

FortiVoice™ v7.31 User Guide

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Introduction

Important information

What happens if the power goes out or if the IP network to VoIP fails?

To ensure a reliable network connection, all elements of the VoIP network should be connected to back-up power supplies (UPS). These elements should include LAN switches, routers, firewalls, broadband connection devices (i.e. cable modems, DSL modems), and VoIP devices. If the power goes out at the Internet Service Provider, no VoIP calls can be made. Calls can still be placed over the telephone lines.

Change password frequently

Setting and frequently changing the system password is recommended to avoid unauthorized changes to system configuration and settings.

PBX fraud

It is recommended that call bridge (DISA) and call back PIN codes are changed frequently to avoid unauthorized users making telephone calls through the phone system.

Emergency service numbers

Ensure emergency service numbers are not blocked by the toll restriction feature. Frequently used emergency service numbers are pre-programmed to avoid blocking, but other local numbers may be used in some location. If your location has an atypical emergency service number, enter it in the routing and blocking rules on the *User Privilege* page to ensure it will be routed to an appropriate phone line.

Before routing any emergency service numbers to a VoIP service provider, check that they do handle emergency service calls and any conditions associated with this service.

Call redirection and service provider billing advisory

Use of the call detail recording and routing and blocking features does not imply any guarantee whatsoever by regulatory authorities, telephone service provider(s), Fortinet or its distributors and resellers, with regard to the accuracy of these features and that the use of such features may not be considered by a telephone company in any disputes which may arise regarding the accuracy of any subscriber's telephone account.

Please read the Start Guide that came with your phone system before reading this user guide. The Start Guide contains critical information about setting up your phone system.

Configuration

Introduction

This chapter contains detailed information about all the features in the Management software, with step-by-step instructions on how to customize these features to best suit your needs.

Starting the Management software



When a unit is being configured, it is locked to prevent other computers or persons using a phone from configuring changes at the same time. If you leave the software open for longer than one hour, the unit unlocks itself to allow configuration changes.

- 1. Plug in the power adapter to turn on your phone system.
- 2. Turn on your computer.
- **3.** Start the phone system Management software. The *Configuration Selection* page appears, and the software attempts to detect your phone system.

FortiVoice Management Software - My Company	
File Tools Options Troubleshooting Help	
	English •
FEBRUNET , FortiVoice Management Software	Français O
	Español O
	Espanor
	Carlos Comment
System discovered at IP address: 192.168.1.12	
(CONFIGURE THE) (CONNECT TO A SYSTEM	OPEN A CONFIGURATION
AUTO-DETECTED SYSTEM	FILE OR TEMPLATE
	ALL STATES AND

- 4. Select your language. You can select *English*, *Français* or *Español*.
- 5. Once the software detects the system, click *Configure Auto-Detected System*. The software loads the configuration from the unit, and the *About* page appears.
- 6. If the software was unable to detect your system, check that all your wires and plugs are securely connected, and then click *Retry Auto-Discovery*. The software loads the configuration from the unit, and the *About* page appears.

- 7. If auto-detection does not work:
 - a. Click Connect to a System Via IP. The Connect Via IP window appears.

obtain the IP address of a any local extension.	s for the system or location. To system on your LAN, dial *56 from
URL / IP addre	ss: 192.168.1.103
	Address Book

- **b.** Enter the URL / IP address and Port of the system, and then click OK. This is the public IP address or public domain name of the system. The software loads the configuration from the system, and the About page appears.
- 8. To open a configuration file:
 - **a.** Click Open a Configuration File or Template. The Open File or Template window appears.

Open File or Template
Select 'Open a File' to load a previously saved configuration. Select 'Create a Template File' to create a new configuration file based on specific hardware combinations.
Open a File
Create a Template File
Cancel

b. Click Open a File. The Open window appears.

open	1805			104-11 11-10-10-10-10-10-	_
Look jn:	FortiVoic	e Management 7.30	~		
My Recent Documents Desktop My Documents	Application Documer ftpServe Greeting Help Languag lib Logs Reseller Utilities & defaultTr & LastSave	onFiles tts r s es Contact emplate.tsd dConfig.tsd			
My Computer					
S	File <u>n</u> ame:				<u>O</u> pe
Places	Files of type:	All Config Files (*.tsd)			Cano

- c. Browse to the .tsd file, and then click Open. The software opens the configuration file.
- **d.** If the selected configuration file is not compatible with your software, you will be prompted to convert the file to a newer format. If you select *OK*, software will first make a backup copy with the filename suffix "_old", then convert the selected file to make it compatible with your software.

About

The About page displays and allows you to change system information.

- About
- 1. Select the *About* page.

The System Information area shows:

- The current time and date programmed into the system.
- The current mode.
- The model numbers of the units in the system.
- The number of new voicemail messages, and the total number of voicemail messages.
- The version number of the Management software.
- When the configuration was last saved to the system.
- The type of connection to the system.
- The region of the system.

System time

System time shows the current date and time programmed into the system. Clicking the *System time* link displays the *Date and Time Properties* window. It allows you to change the date, time, time zone, and NTP server programmed into the system. See "System time" on page 11.

Current mode

Current mode shows the current mode. Clicking the *Current mode* link displays the *Change Mode* window. It allows you to change the mode of the system. See "Change mode" on page 9.

My system

My system shows the model numbers of the units in the system. Clicking the *My system* link displays the *Discovered Network* window. It shows the MAC address, IP address, system ID, model number, firmware version, and length of operation for each unit.

D	iscovered Network
	Network Information
	System ID: 6NKE6HK2T8
	MAC address: 000F4D008F27
	IP address: 10.10.10.3
	Hardware type: FVC-24B
	Firmware version: 7.30.000
	Up time: 1 hrs 48 min 45 sec
	Identify
	Refresh Close

To identify a unit, click the *Identify* button. The *Identify* button will change to a *Stop* button. All the lights on the front panel will start flashing. Click the *Stop* button to end the flashing.

Voicemail messages

Voicemail messages shows the number of new voicemail messages in the system, and the total number of voicemail messages in the system. Clicking the *Voicemail messages* link displays the *Voicemail Memory Usage* window. It shows message statistics for each voice mailbox. See "Mailbox status" on page 85.

Administration

The *Administration* page allows you to set up the system name, system password, numbering plan, region, language, and how to route the call if a user dials an operator number.

1. Select the *Administration* page.

Administration	
System name (optional): My Company	
System password (4-8) digits:	
Confirm password:	
ystem Numbering Plan	
Length of extension, voicemail and speed dial numbers: 3 digits Change	
Length of extension, voicemail and speed dial numbers: 3 digits Change	
Length of extension, voicemail and speed dial numbers: 3 digits Change	
Length of extension, voicemail and speed dial numbers: 3 digits Change anguage System prompt languages currently installed on the system: English Edit French	
Length of extension, voicemail and speed dial numbers: 3 digits Change anguage System prompt languages currently installed on the system: English French Spanish	
Length of extension, voicemail and speed dial numbers: 3 digits Change anguage System prompt languages currently installed on the system: English Edit French Spanish Default language for system prompts to callers: English	
Length of extension, voicemail and speed dial numbers: 3 digits Change anguage System prompt languages currently installed on the system: English Edit French Spanish Default language for system prompts to callers: English	
Length of extension, voicemail and speed dial numbers: 3 digits Change anguage System prompt languages currently installed on the system: English French Spanish Default language for system prompts to callers: English ial 0 Routing-	

Administration

The Administration area allows you to set up the system name and system password.



Use a system password, otherwise the system will be vulnerable to configuration changes, misuse and/or lock-out by callers or users. If the router has port 9393, 8485 and 8486 mapped to the system for remote configuration, the system will also be vulnerable to anyone on the Internet.

Change the system password frequently to prevent unauthorized users from making calls or changing the configuration.

- 1. Optionally enter the *System name*. This should be the company name, or a shortened form suitable for use as caller ID during VoIP calls. If you enter a system name, you will require it when logging in through a browser for status or call detail record (CDR) information.
- 2. Enter the *System password*. It has to be a 4- to 8-digit numeric password, so you can also enter it on a touchtone phone.

The system password allows access to the configuration. It is entered when you start the Management software, and when you access the configuration through a local extension or outside phone.

System numbering plan

You can use a 3, 4 or 5-digit plan for extensions, ring groups, mailboxes and speed dials in your system. The default is 3 digits. To change it, click *Change* and select the desired number of digits. This selection is system-wide; you can't have a mix, for example, of 3 and 5-digit extension numbers.

Numbers available per plan: 3-digit numbers: 100 to 899 4-digit numbers: 1000 to 8999 5-digit numbers: 10000 to 89999

Region selection

The Region Selection area allows you to select the country where your system will operate.

Language

The *Language* area displays the language loaded into the system, and allows you to load and remove language files.

1. To change the language files loaded into the system, click *Edit*. The *Language File Management* window appears, listing loaded language files.

Language File Management	×
English **	Add
	Remove
** Base language file cannot be removed.	
	Close

- **a.** To load a language file, click *Add*, and then select the language file.
- **b.** To remove a language file, select the language, and then click *Remove*.
- 2. Select the default language for prompts heard by callers and users in the *Default language for system prompts to callers* list.

Dial 0 or 9 routing

The *Dial 0 routing* area (or *Dial 9 routing* area in some regions) allows you to select how the system will route the call if a user dials 0 or 9. For example, you can configure the system to connect the user to the receptionist at extension 114.

- 1. Select the action in the Connect to list. Choices are:
 - *perform no action* Does not connect to a resource.
 - go to voicemail Connects to the selected voicemail.
 - go to local extension Connects to the selected local extension.
 - go to remote extension Connects to the selected remote extension.
 - go to ring group Connects to the selected ring group.
 - play announcement Plays the selected announcement.
 - go to auto attendant Connects to the selected auto attendant.
 - queue at ring group Connects to the call queue of the selected ring group.
 - *dial-by-name directory* Accesses the dial-by-name directory.
- 2. Select the resource. Depending on the action, resources are voice mailboxes, extensions, announcements or auto attendants

Scheduling

A mode is a period of time when the system uses a particular call handling setup for incoming calls. Mode 1 is typically office hours, and Mode 2 is typically evening and weekend hours. Holiday Mode is when your office is closed for a statutory holiday or shutdown.

The system can use a different call handling setup for each mode, and can automatically change mode with the time of day, day of week and on holidays. This way it can handle calls in different ways according to a schedule.

The *Scheduling* page allows you to set up Mode 1, Mode 2 and Holiday Mode. You can also adjust the time and date, and can switch modes manually.



Ensure the Management software is closed during automatic mode switching, or if an administrator is manually switching modes using a phone. In these cases the mode cannot switch while the Management software is open.

Note that the mode can switch while the Management software is open, if the user is manually switching modes using the software, as described later.

1. Select the *Scheduling* page.

Indes		
Enable Mode 1 Mode 1 label: Bu	Enable holiday mode Settings Holiday mode label: Holiday mode	
Mode 2 label: Af	ter Hours Current mode: Mode 1 (Business Hours)	Change Mode
Use Automatic Mode Day of Week Monday Tuesday Wednesday Thursday Friday Saturday Sunday	Scheduling Scheduling Copy schedule to other days September 13, 2011 / 11:16 Copy schedule to other days	:44 AM Adjust

Modes

The *Modes* area allows you to enable, name and switch modes, and displays the current mode.

1. Enter the *Mode 1 label* and the *Mode 2 label*. Note that Mode 1 and Mode 2 are always available, so you cannot disable the *Enable Mode 1* and *Enable Mode 2* checkboxes.

- 2. Set up Holiday Mode to handle incoming calls differently during a holiday:
 - **a.** Select the *Enable holiday mode* checkbox to enable Holiday Mode. The window enables the *Settings* button.
 - **b.** Enter the Holiday Mode label.
 - **c.** Click Settings. The Holiday Settings window appears. The calendar shows the current date in green.

-Calend Highligh Canada	ar t in calei a	ndar hol	lidays ok	oserved i Po	in: opulate		Selected Dates
<<		Ju	uly 201	LO		>>	15/07/2010
Sun	Mon	Tue	Wed	Thu 1	Fri 2	Sat 3	
4	5	6	7	8	9	10	
11	12	13	14	15	16	17	
18	19	20	21	22	23	24	
25	26	27	28	29	30	31	
							Remove
f calls ai ise Call	re direct Cascade	ed to an e setting	ny local, i is for Mo	remote o ode 1 or l	r ring g Mode 2	roup froi as selec	n an auto attendant during holiday mode, then the extension will ted below. This step avoids reconfiguring all extension Call

- **d.** Select your country in the *Populate calendar with holidays observed in* list, and then click *Populate*. The calendar will show your country's statutory holidays in grey.
- e. Using the calendar, select the month and then click the date of the holiday. The date is added to the *Selected Dates* area, and the calendar will show the date in blue. Click the date again to remove it.
- f. Repeat the above step until all required holidays have been added.
- **g.** Select the mode in the Use the following Call Cascade settings for Holiday Mode. Choices are Mode 1 and Mode 2. This setting determines whether extensions will use their Mode 1 or Mode 2 call cascades during Holiday Mode.

Change mode

The *Modes* area displays the current mode.

1. To change mode, click Change Mode. The Change Mode window appears.

-Change Mode-	(
Mode 1 (Bus)	siness Hours)
O Mode 2 (Aft	er Hours)
O Holiday mod	le

2. Select the mode. The new mode immediately takes effect. You do not have to save the configuration.

You can also change the mode by phone.

Remotely

- **1.** Ensure the Management software is closed.
- **2.** Remotely call the system.
- 3. At the auto attendant, enter command mode by pressing #.
- 4. Enter the system password, followed by #.
- **5.** Enter one of the following commands:
 - 30 + # to hear the current mode.
 - 31 + # to enable Mode 1.
 - 32 + # to enable Mode 2.
 - 33 + # to enable Holiday Mode.
- 6. Press * to exit command mode.

Locally

- **1.** Ensure the Management software is closed.
- 2. Pick up a local extension.
- 3. Enter one of the following commands:
 - ***30 + # to hear the current mode.
 - ***31 + # to enable Mode 1.
 - ***32 + # to enable Mode 2.
 - ***33 + # to enable Holiday Mode.

Note: a password is not required.

4. Press * to exit command mode.

Scheduling

The Scheduling area allows you to specify when the system will change modes.

- 1. Select the *Use automatic mode switching* checkbox to enable the modes. The window enables the *Scheduling* area.
- **2.** Select the day.
- 3. Select whether to use one mode or multiple modes during that day:
 - Select Continuously run to use the selected mode the whole day.
 - Select *Switch modes depending on time* to switch between Mode 1, Mode 2 and Holiday Mode during the day. The window enables the schedule controls.
- 4. If you selected Continuously run, select Mode 1, Mode 2 or Holiday Mode.
- 5. If you selected *Switch modes depending on time*, set when the mode should change. The default settings are:
 - Mode 1 at 9:00 AM on Monday to Friday.
 - Mode 2 at 5:00 PM on Monday to Friday.
 - Mode 2 continuously on Saturday and Sunday.

To change the mode switch time for more than one day, make the change to one day and click the *Copy schedule to other days* button. In the dialog box, check the desired days to apply the new time.

System time

The Scheduling area shows the date and time programmed into the system. Clicking the Adjust button displays the Date and Time Properties window. It allows you to change the date, time, time zone, and NTP server programmed into the system.



You can configure the system to get the time and date (but not the year or time zone) from caller ID. See "Clock Preferences" on page 136.

1. Click *Adjust*. You can also choose *Options* > Set Date & Time. The Date and Time Properties window appears.

Sun	Mon	Tue	Wed	Thu	Fri	Sat	**``		`* • .	
				1	2	3		1		
4	5	6	7	8	9	10				
11	12	13	14	15	16	17				
18	19	20	21	22	23	24				
25	26	27	28	29	30		·. ·•.	· · · •		
									11	:18:29 AM
e zone	(-05:0	0) Easte	ern (US/	Canada)		/				

Date & time

The Date & Time area allows you to set the date and time programmed into the system.

- 1. Select the month and date for the system.
- 2. Set the time for the system.

Time zone

The *Time Zone* area allows you to set the time zone for the system, and whether daylight savings time is used in your region.

- 1. Select the time zone for the system.
- **2.** If your region uses daylight savings time, select the *Automatically adjust clock for daylight saving changes* checkbox.

NTP server

The *NTP Server* area allows you to set the NTP (Network Time Protocol) server for the system. It provides the same time to the units and to the external IP extensions.

1. Select the *NTP* Server tab.

Date and Time Properties
Date & Time NTP Server
External IP extensions need to receive date and time information from an NTP (Network Time Protocol) server, accessible via the Internet. The system provides clock.sjc.he.net as the default NTP server. You can designate an alternate NTP server using the field below.
A list of alternate NTP servers can be found at: http://support.ntp.org
NTP server: clock.sjc.he.net
Apply <u>OK</u> <u>C</u> ancel

2. Enter the *NTP* server.

IP Configuration

The *IP Configuration* page allows you to set up the system for Internet communications. The Internet can be used for external IP extensions, a VoIP network, a subscription to a VoIP service, and remote configuration.

1. Select the IP Configuration page.

Use configured IF	o and l	ONS inf	ormatio	n .	-										
Unit 1 IP address:	10	. 10	. 10	. 3	Subnet mask:	255	. 255 .	255 .	0						
Unit 2 IP address:	5	. 64	. 14	. 72	Default gateway:	10	. 10 .	10 .	1						
Jnit 3 IP address:	5	. 64	. 14	. 80	Preferred DNS server:	10	. 10 .	10 .	1						
Unit 4 IP address:	5	. 64	. 14	. 88	Alternate DNS server:	0	.0.	D.	0						
10 . 10 .	10	. 10													
10 . 10 . ublic IP Address – Type of publi	10 c addr	10 ess: 5	Static pu	ublic IP ad	dress 🔹										
10 . 10 . ublic IP Address – Type of publi Current public I	10 c addr P addr	. 10 ress: [5 ress: [5	Static pu	ublic IP ad	dress 🗸										
10, 10, 10, 10, 10, 10, 10, 10, 10, 10,	10 c addr P addr ne (FQ	. 10 ress: 5 ress: 5 DN):	Static pu	ublic IP ad	dress •										
10 , 10 , ublic IP Address – Type of public Current public I Public domain nan rewall Settings: –	c addr P addr ne (FQ	. 10 ress: [5 ress: 5 DN): [Static pu	ublic IP ad	dress										
10, 10, 10, 10, 10, 10, 10, 10, 10, 10,	10 c addr P addr ne (FQ <u>10.10</u>	.10 ess: 5 DN): .10.1	Static pu	ublic IP ad	dress .55	nded th	at you p	erform	he fire	ewall tes	t to ens	ure route	r port mapp	ings are corr	ect for

System IP settings

The System IP Settings area shows IP addresses. By default, Obtain IP and DNS information automatically is selected and the area shows IP addresses from the router.

- 1. Change Obtain IP and DNS information automatically to Use configured IP and DNS information in order to lock in the IP addresses.
- In some cases, the System IP Settings area may be blank because your router has not delivered the IP addresses. If so, enter the following IP addresses from your LAN administrator:
 - a. Enter a static IP address for each unit in the Unit IP address boxes.
 - **b.** Enter the *Subnet mask* for the LAN. This address determines the subnet the unit IP addresses belong to.
 - c. Enter the IP address of the *Default gateway* on your network. A gateway is a hardware device that connects the office network to the Internet. The router may act as default gateway.
 - **d.** Enter the IP address of the *Preferred DNS server*. DNS is a service used to resolve a domain name to an IP address. The router may act as DNS server.
 - e. If applicable, enter the IP address of the Alternate DNS server.

PRI Interface Address

The *PRI Interface Address* area allows you to assign an IP address to the PRI interface. This IP address must be different than the system IP address.

Public IP address

The *Public IP Address* section allows you to set up Internet parameters so the system can communicate with other locations over the Internet.

- **1.** Set the *Type of public address*. Choices are:
 - Dynamic public IP address This is the default setting. Your ISP (Internet Service Provider) will assign different public IP addresses to your location. The system will check its public IP address every five minutes. When the public IP address changes, the system will automatically use the new one. This allows it to manage VoIP calls properly.
 - Static public IP address A static IP address is fixed. Your ISP assigns the static IP address.

If *Static public IP address* is selected, the window allows you to enter the *Current public IP address*.

2. If you selected *Dynamic public IP address*, enter the *Public domain name*. A public domain name is only required if this location has external IP extensions.

A DDNS (Dynamic Domain Name Service) provider such as www.dyndns.com can create a public domain name to resolve to your IP address, so your external IP phones will continue to work when the IP address changes.

If you are using DDNS, ensure your router supports your DDNS service, and configure it to update the DNS servers.

If your router does not support DDNS, download one of the applications specified on www.dyndns.com. To update the DNS servers, the application needs to run on a PC connected to the same LAN as the system.

3. If you selected *Static public IP address*, enter the *Current public IP address* from your ISP. Leave the *Public domain name* box blank.

If the unit is not behind a router, or if a private virtual network is used, the public IP address should be set to the local IP address of the unit acting as local proxy.

Note that it will take up to one minute for the new static public IP address to take effect.

Firewall settings

The *Firewall Settings* area displays the IP address of the gateway (i.e. router), and whether router configuration is required.

A router is a gateway between the local area network and the Internet. Most routers have a firewall to block unwanted data from the Internet. For voice data to reach the system through the firewall, port forwarding is required. Port forwarding allows the router to map ports to the IP addresses of the units. Valid Internet data will use the ports to go through the firewall to the units.

If you are setting up external IP extensions, or a VoIP service that doesn't handle port forwarding, port forwarding is required.

If port forwarding is required, and your router supports uPNP (Universal Plug and Play), ensure uPNP is enabled. The system will use uPNP to automatically set up port forwarding, and the *Automatic (uPNP Enabled)* link will appear. No router configuration is required.

If port forwarding is required but your router doesn't support uPNP, or automatic port forwarding doesn't work, the *Manual Port Mapping Required* link will appear. You will need to configure the router as described below.

1. Click the *Manual port mapping required* link. The *Manual Port Mapping* window appears. It lists the packet type, port number, IP address and protocol of each required port.

The system requires the following firew	all ports be mapped	d to the following IP	addresses fo	or
Description	Port	System IP	Protocol	
SIP (VoIP signalling)	5060	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 1)	6100	192.168.1.103	UDP	1
RTP Audio (Unit 1, Port 2)	6102	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 3)	6104	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 4)	6106	192.168.1.103	UDP	٦.
RTP Audio (Unit 1, Port 5)	6108	192.168.1.103	UDP	-13
RTP Audio (Unit 1, Port 6)	6110	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 7)	6112	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 8)	6114	192.168.1.103	UDP	
TFTP (IP Phone Configuration)	69	192.168.1.103	UDP	-

- 2. To access the router configuration:
 - **a.** Click the link containing the IP address of the gateway. The default browser starts, and prompts you for the router's user name and password.
 - b. Enter the router's user name and password. The browser shows a setup screen.
 - c. Navigate to the screen used to set up port forwarding. See your router documentation.
 - **d.** Set up port forwarding using the information from the *Manual Port Mapping* window. See your router documentation for instructions on how to map ports. For information on configuring routers and mapping ports, visit http://www.portforward.com/english/routers/port_forwarding/routerindex.htm.
- **3.** To check the status of each port through the firewall, click *Check Firewall*. The *Firewall Test* window appears.

Service	Status	Service	Status	
Internet Access		🔽 RTP Audio (Unit 2	2, Port 4)	
HTTP (XML Service	9)	🗹 RTP Audio (Unit 2	Port 5)	
RTP Audio (Unit 1,	Port 1)	RTP Audio (Unit 2	?, Port 6)	
RTP Audio (Unit 1,	Port 2)	RTP Audio (Unit 2	2, Port 7)	
RTP Audio (Unit 1,	Port 3)	RTP Audio (Unit 2	2, Port 8)	
RTP Audio (Unit 1,	Port 4)	SIP (VolP signalling	ng)	
RTP Audio (Unit 2,	Port 1)	TFTP (IP Phone C	onfiguration)	
RTP Audio (Unit 2,	Port 2)			
RTP Audio (Unit 2,	Port 3)			
				Details Test Por

- 4. Select the services you want to check.
- 5. Click *Test Ports*. The system will check the ports of the selected services.

Configuring the router manually

If you cannot access the router configuration through the *IP Configuration* page, configure the router manually.

- **1.** Open the router configuration and navigate to the screen used to set up port forwarding. See your router documentation.
- 2. In the Management Software on the *IP Configuration* page, click the *Manual Port Mapping Required* link.
- **3.** Map the port indicated for SIP (VoIP) signaling.

If required, you can map a different port. Select *Troubleshooting* > *VoIP* and enter the port in the *SIP signalling port* field.

Map the rest of the ports to the IP addresses indicated in the Manual Port Mapping window.

roper VoIP communication	all ports be mapped	to the following IP	addresses n	or
Description	Port	System IP	Protocol	
SIP (VoIP signalling)	5060	192.168.1.103	UDP	-
RTP Audio (Unit 1, Port 1)	6100	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 2)	6102	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 3)	6104	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 4)	6106	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 5)	6108	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 6)	6110	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 7)	6112	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 8)	6114	192.168.1.103	UDP	
TFTP (IP Phone Configuration)	69	192.168.1.103	UDP	
				~

If required, you can map different ports. Select *Troubleshooting > VoIP*.

- **4.** Map ports 9393, 8485 and 8486 (Type: TCP) to the unit acting as local proxy to allow remote configuration of the system.
- 5. If available, enable Quality of Service (QoS) to give voice traffic priority over data.
- 6. Save the configuration to the router.

On-Hold/Ringback

Music on hold plays when a caller is on hold, or is being transferred to an extension. The system plays a double beep tone or ringback tone by default, but you can have it play music or a recorded message instead.

The system can play music on hold from an internal .wav file or from an external audio source. An external audio source is a CD player or MP3 player. It must be connected to the MUSIC jack on the back of each unit.

The recording time for internal music on hold, voicemail, and the auto attendants is shared on the unit.

The On-Hold/Ringback page allows you to set up music on hold.

1. Select the On-Hold/Ringback page.

p-Hold Settings-	
Vhen callers are placed on hold, the system can play music loaded on the system or from an external source, or simply play "double beep" tones.	
) Play "double beep" tones	
Play music from source connected to the MUSIC jack	
Play music from file loaded on the system Load Wav File	
Length: 03:07 min Delete Wav File	
Date: April 7, 2010 / 7:48:02 PM	
Playback volume for music file: default 🗸	
ansfer Settings	
When calls are transferred from the auto attendant or from an extension, callers will hear:	
O Music	
Rincback	

On-hold settings

The On-hold Settings area allows you to select the sound to play while the caller is on hold.

- **1.** Select the sound to play while the caller is on hold. Choices are:
 - *Play "double beep" tones* Plays a "beep beep" sound.
 - Play music whose source is connected to the MUSIC jack Plays music from a CD player connected to the MUSIC jack.
 - Play music from file loaded on the system Plays the .wav file loaded into the system.
- 2. If you selected Play music from file loaded on the system:
 - **a.** Obtain an 8 kHz, 8 bit, Mono, u-Law .wav file that contains the music or recorded message.

You can also convert .mp3 and other file formats to .wav with an audio converter utility.

b. Click Load Wav file. The Select Wav File window appears.

Select Wav File	X
Audio files must be in 8 kHz, 8	bit, mono, µ-law format and saved as a .wav file.
Choose File	Browse
	Load Cancel

- **c.** Click *Browse* to select a .wav file. Click *Load* to load the .wav file into the system. The file size and date loaded will be dispayed. *Play* and *Stop* buttons are also provided, if you prefer to listen to the file before loading it.
- **d.** Save the configuration to allow the music on hold to take effect.
- **3.** Adjust the volume of music on hold.
 - a. Check the volume by placing a test call and going on hold.
 - b. Change the volume by setting the Playback volume for music file list. Choices are:
 - loudest
 - louder
 - default
 - quieter
 - quietest
 - c. Choose File > Save.

You must save the configuration for the volume setting to take effect.

d. Repeat Steps a. - c.

Deleting a music on hold file

You can delete a music on hold file. This may be required if you need more space for voicemail messages. In this case you can use an external audio source for music on hold, or you can upgrade the memory capacity of the units.

1. Click Delete Wav File. The system deletes the .wav file from the unit(s).

Playing music on hold on the PA system

You can play music on hold through the PA jack.

1. Dial *80 to play or stop playing music on hold through the PA jack.

If music on hold is playing through the PA jack, and a user dials *0 to make an overhead page, the overhead page will interrupt the music on hold. However, if the PA jack is configured for voicemail screening, and music on hold is playing, voicemail screening will not interrupt the music on hold. See "Paging Options" on page 132.

Transfer settings

The *Transfer Settings* area allows you to select the sound to play while the caller is being transferred from an auto attendant or to another extension.

- 1. Select the sound to play while the caller is being transferred. Choices are:
 - Music Plays music on hold as configured in the On-Hold Settings area.
 - *Ringback* Plays the ringback tone, which is the normal sound heard when the other person's phone is ringing.

Email Service

The system can send an e-mail if a voicemail message has been left in a mailbox. The e-mail includes the caller ID, and can include the voicemail message as an attachment.

The e-mail can optionally include links to delete or save the voicemail message. If you delete the voicemail message, it will be removed from the system. If you save the voicemail message, the system will change its status from "new" to "saved". A voicemail message that is saved will no longer activate the new message indicator on the user's extension.

The system can send notification to up to four e-mail addresses per mailbox. For example, if a caller leaves a message in the general mailbox for the sales group, the system can send an e-mail to four members of the sales group.

An e-mail address can be assigned to multiple mailboxes. For example, the system can send e-mails to a user if callers leave messages in the user's local extension mailbox, remote extension mailbox or a general mailbox.

The *Email Service* page allows you to maintain e-mail addresses, set up the e-mail server parameters, and test the e-mail server.

- Email Service Email Notification Setting ID Labe Full name 2 3 Email address 4 5 Notification option: Email notification only (Full length) v 6 8 10 11 v 12 -Import an Email List 0 Email addresses with their associated names can be imported from a comma-delimited file. The format should be name.email address Select File 0 -Email Server Settings Dedicated email address for sending/receiving emails: Enable Delete/Save Messages Option -Outgoing Email Server When enabled and an account is configured to receive message The outgoing server is used to relay voicemail notification and call detail attachments, the system will include a save/delete option in email logs via email messages notifications so that voice messages can be managed remotely Outgoing mail server (SMTP): Incoming mail server (POP3): More Settings ... More Settings ... Test Account Settings. Connected - Syst
- 1 Select the Email Service page.

Email notification settings

The *Email Notification Settings* area allows you to add up to 255 e-mail addresses.

- 1. Select a slot for the e-mail address.
- 2. Enter the recipient's *Full name*.
- 3. Enter the recipient's *Email address*.

- 4. Select the Notification option. Choices include:
 - Include voice message as attachment Attaches the voicemail message to the e-mail as a .wav file. The e-mail includes options to save or delete the voicemail from the system.
 - *Email notification only (Full length)* Does not attach the voicemail message to the e-mail, but includes full details about the contents of the voicemail, including sender, time sent and length, along with a talley of new and saved messages.
 - Email notification only (SMS length) Does not attach the voicemail message to the e-mail as above, and includes the same information abbreviated to under 130 characters.

Import an email list

The Import an Email List area allows you to import e-mail addresses from a .csv file.

Each entry in the file must be formatted as follows: name,e-mail address. For example:

John Doe,john.doe@email.com Jane Doe,jane.doe@email.com

1. Click Select File. The Import an Email List window appears.



2. Select the .csv file, and then click Open.

E-mail server settings

The *Email Server Settings* area allows you to set up the e-mail server parameters, include links to delete or save the voicemail message, and test the system's ability to send e-mail.

You will need an e-mail address to send notifications from.



Notification may not work properly if the SMTP server uses Microsoft Exchange Server, and/or certain security features are enabled.

Use e-mail account information from your Internet service provider (ISP), system administrator, or alternatively, obtain one from a compatible online e-mail provider.

1. Enter an e-mail address in the *Dedicated email address for sending/receiving emails* box. When e-mail notifications arrive from the system, this e-mail address will show up in the *From* field.

- 2. Enter the name of the Outgoing mail server (SMTP).
- **3.** To configure authentication and server port numbers, click *More Settings*. The *Outgoing Email Settings* window appears. See "Outgoing server authentication" on page 21 and "Server settings" on page 22.
- 4. To include links within the e-mail notification, select the *Enable delete/save messages option* checkbox. The links will allow the recipient to delete or save the voicemail message. The e-mail notification can only include the links if the voicemail message is attached.
- **5.** Deleting the voicemail message will remove it from the unit. Saving the voicemail message will change its status from "new" to "saved". A voicemail message that is saved will no longer activate the new message indicator on the user's extension.
- 6. Enter the name of the *Incoming mail server (POP3)*. The name will automatically appear in the *My Outgoing Server (SMTP) Requires Authentication* area.
- 7. To configure authentication, server port numbers and how often the system should check the POP3 server, click *More Settings*. The *Incoming Email Settings* window appears. See "Incoming server authentication" on page 22, "Incoming server port numbers" on page 22 and "Incoming mail server options" on page 23.

Outgoing server authentication

The *My Outgoing Server (SMTP) Requires Authentication* area allows you to enter account information for the outgoing server. The outgoing server is also referred to as the SMTP server. It can have its own account information, or can use the account information from the incoming server. The incoming server is also referred to as the POP3 server.

Outgoing Email Settings
My Outgoing Server (SMTP) Requires Authentication
This server requires SMTP login information.
This server requires you to login to the incoming mail server before sending mail
Incoming mail server (POP3):
User name:
Password:
Server Settings
Outgoing mail server (SMTP): 25 Use Defaults
Authentication method: PLAIN
QK Cancel

- **1.** If the outgoing server requires authentication, select the *My Outgoing Server (SMTP) Requires Authentication* checkbox.
- **2.** Select the type of login your server requires:
 - If the outgoing server uses its own login information, and can send e-mail without first logging in to the incoming server:
 - i. Select This server requires SMTP login information.
 - ii. Enter the User name of the account from the outgoing server.
 - iii. Enter the *Password* of the account from the outgoing server.
 - If the outgoing server uses login information from the incoming server, and must log in to the incoming server before sending e-mail:
 - i. Select This server requires you to login to the incoming mail server before sending mail.
 - **ii.** Enter the name of the *Incoming mail server (POP3)*. The name will automatically appear in the *Enable Delete/Save Messages Option* area.

- iii. Enter the *User name* of the account from the incoming server. The user name will automatically appear in the *Incoming Server Authentication* area.
- **iv.** Enter the *Password* of the account from the incoming server. The password will automatically appear in the *Incoming Server Authentication* area.

Server settings

The Server Settings area allows you to enter a port number for the outgoing server.

- 1. The default *Outgoing mail server (SMTP)* port is 25. If required, enter a different port number ranging from 1 to 65535.
- 2. Select an acceptable authentication method for your server. If you're uncertain, click the *Test Account Settings* button on the *Email Services* page. The correct authentication method will be set automatically. The Test Email routine will test the server and confirm the authentication method supported by the server and will set the value accordingly.

Incoming server authentication

The *Incoming Server Authentication* area allows you to enter the user name and password of the account from the incoming server. The incoming server is also referred to as the POP3 server.

- 1. Enter the User name of the account from the incoming server. The user name will automatically appear in the My Outgoing Server (SMTP) Requires Authentication area.
- 2. Enter the *Password* of the account from the incoming server. The password will automatically appear in the *My Outgoing Server (SMTP) Requires Authentication* area.

Incoming server port numbers

The Server Settings area allows you to enter a port number for the incoming server.

Incoming Email Settings
Incoming Server Authentication
User name:
Password:
Incoming mail server (POP3): 110 Use Defaults
This server requires secure connection (SSL)
Incoming Mail Server Options
Email request interval: 2 minutes
Delete any unrecognized inbound emails
QK Cancel

- 1. The default *Incoming mail server (POP3)* port is 110. If required, enter a different port number ranging from 1 to 65535.
- 2. If your POP3 server supports secure authentication, check the box. Please refer to your mail server settings.

Incoming mail server options

The *Incoming Mail Server Options* area allows you to specify how often the unit should check the incoming server for e-mail messages from users deleting and saving voicemail messages. The incoming server is also referred to as the POP3 server.

- 1. Select the Email request interval, ranging from 2 minutes to 30 minutes.
- 2. To delete e-mail messages that aren't from users (i.e. spam), select the *Delete any unrecognized inbound emails* checkbox.

Testing the e-mail server settings

- **1.** Click Test Account Settings.
- **2.** Click *Save*. The system saves the configuration, then the *Test Account Settings* window appears.

The system can test the account settings by sending an email to a valid email address.				
est email address: iohndoe@email.com		Start		
est	Status			

3. Enter an e-mail address in the *Test email address* box, and then click *Start*. The following window shows a successful result, with each task completed.

Test Account Settin	gs		×
Test successfull Cli	sk Close to continue.		
Test email address:	iohndoe@email.com		Start
Test		Status	
Connecting to SMTP	server	Passed	
Test complete		Passed	
Test complete		V Passed	
			Close

If a task fails, the system is not able to send e-mail messages. Adjust the e-mail server parameters accordingly:

- If *Find outgoing mail server (SMTP)* fails, check the SMTP server name in the *Outgoing mail server (SMTP)* box.
- If Log onto outgoing mail server (SMTP) fails, check the SMTP server authentication parameters in the Outgoing Email Settings window.
- 4. Ensure the e-mail address has received the test e-mail. Note that the e-mail may have been routed to a junk or spam folder.
Managing voicemail messages

If the *Notification option* is set to *Include voice message as attachment*, the recipient can play, save or delete the voicemail message.

- 1. To play the voicemail message, the recipient double-clicks the attachment. The default .wav player opens the voicemail message.
- 2. To save the voicemail message, the recipient clicks *Save message*. The e-mail program creates a new e-mail message with the *To* and *Subject* fields completed. The recipient sends this e-mail message. Upon receiving the e-mail message, the system will change the status of the voicemail message from "new" to "saved", and the local extension will turn off the new message indicator. However the system will not delete the voicemail message.

Note that the system allows up to 99 voicemail messages per mailbox. Once a mailbox fills up, callers won't be able to leave voicemail messages for that user. Therefore users should delete voicemail messages before the mailbox fills up.

3. To delete the voicemail message, the recipient clicks *Delete message*. The e-mail program creates a new e-mail message with the *To* and *Subject* fields completed. The recipient sends this e-mail message. Upon receiving the e-mail message, the unit will delete the voicemail message.

Setting up POP3 service with Microsoft Exchange

If your system uses Microsoft Exchange, and you wish to include voicemail messages as attachments within the notification emails, you must set up POP3 service.

For instructions on how to download, install and configure POP3 service, refer to the Microsoft documentation included with your version of Exchange. See http://technet.microsoft.com/en-us/library/aa998454(EXCHG.65).aspx for more information.

Risks of enabling POP3 service

- The default configuration does not use encryption when sending the username and password.
- Unauthorized clients can use the POP3 service.
- E-mails from users for saving and deleting voicemail messages are not stored on the Exchange server, but are pulled through to the client (i.e. the system).

Security recommendations

- 1. Install the system on the same LAN as the Exchange server.
- 2. Follow the Microsoft procedures for tightening security.
- **3.** The POP3 service is a virtual server. You can use the following techniques to control incoming access to a virtual server. Refer to the Microsoft documentation included with your version of Exchange.
 - Grant or deny access using IP addresses or Internet domain names.
 - Require authentication for incoming connections.
 - Restrict concurrent connections.
 - Set connection time-out values.

Global Dial Plan

Configuring multiple locations with the global dial plan

FortiVoice FVC-40, 70 and 100 systems in multiple locations can be connected over the public internet or a closed network such as a VPN. Extensions in any office can call any other office by using a location code prefix. Calls can be transferred and conferenced between locations.

The global dial plan is set by the central administrator and instantly broadcast to all other locations.

Setting up the global dial plan: master system

1. Select the *Global Dial Plan* page in the FortiVoice Management software.

Activate Global Dial Plan Automatic Synchronization Image: Create Dial Plan File User Key: User1234 Password: Password Password: Password Location Name: View Directory Location Code: Global dial plan configuration date: Feb 15 09:53 Call Handling Mode 1 Mode 2 Holiday Mode When a call comes in on this phone number, perform the following action: Ing extensions are busy or the call is not answered: perform no action If all extensions are busy or the call is not answered: perform no action If all extensions are busy or the call is not answered:	obal Dial Plan			
Automatic Synchronization This site is the master location Server Address: 64.26.140.142 User Key: User1234 Password: Password Location Name: Location Name: Location Code: Global dial plan configuration date: Feb 15 09:53 Call Handling Mode 1 Mode 2 Holday/Mode When a call comes in on this phone number, perform the following action: ring extensions are busy or the call is not answered: perform no action f all extensions are busy or the call is not answered: perform no action f after 5 rings.	🖉 Activate Global Di	al Plan		(
This site is the master location Server Address: 64 26.140.142 User Key: User1234 Password: Password Location Name: Location Name: Location Code: Global dial plan configuration date: Feb 15 09:53 Call Handling Mode 1 Mode 2 Holiday Mode When a call comes in on this phone number, perform the following action: ing extensions are busy or the call is not answered: perform no action f all extensions are busy or the call is not answered: perform no action f all extensions are busy or the call is not answered: perform no action f after 5 rings.	CAutomatic Synchr	onization		0
Server Address: 64.26.140.142 User Key: User1234 Password: Password Location Name: Location Name: Location Code: Global dial plan configuration date: Feb 15 09:53 Call Handling Mode 1 Mode 2 Holday Mode When a call comes in on this phone number, perform the following action: fing extensions Edit If all extensions are busy or the call is not answered: perform no action	This site is the	e master location		
User Key: User1234 Password: Password Location Name: View Directory Location Code: Global dial plan configuration date: Feb 15 09:53 Call Handling Mode 1 Mode 2 Holiday Mode When a call comes in on this phone number, perform the following action: ring extensions • Edit If all extensions are busy or the call is not answered: perform no action • • after 5 • rings.	Server Address:	64.26.140.142	Create Dial Plan File	
Password: Password Location Name: Location Name: Location Code: Global dial plan configuration date: Feb 15 09:53 Call Handling Mode 1 Mode 2 Holday/Mode When a call comes in on this phone number, perform the following action: ing extensions are busy or the call is not answered: perform no action after 5 rings.	User Kev:	User1234		
Location Name: Location Code: Global dial plan configuration date: Feb 15 09:53 Call Handling Mode 1 Mode 2 Holday Mode When a call comes in on this phone number, perform the following action: ring extensions Edtt If all extensions are busy or the call is not answered: perform no action a the state of	Dassword	Password		
Location Name: Location Code: Global dial plan configuration date: Feb 15 09:53 Call Handling Mode 1 Mode 2 Holiday Mode When a call comes in on this phone number, perform the following action: ring extensions • Edit If all extensions are busy or the call is not answered: perform no action • • after 5 • rings.	rassword.	1 435 WOIL		
Location Name: Location Code: Global dial plan configuration date: Feb 15 09:53 Call Handling Mode 1 Mode 2 Holiday Mode When a call comes in on this phone number, perform the following action: ring extensions Edit If all extensions are busy or the call is not answered: perform no action after 5 rings.				
Location Code: Global dial plan configuration date: Feb 15 09:53 Call Handling Mode 1 Mode 2 Holiday Mode When a call comes in on this phone number, perform the following action: ring extensions Edit If all extensions are busy or the call is not answered: perform no action after 5 rings.		Location Name:	View Directory	
Global dial plan configuration date: Feb 15 09:53		Location Code:		
Call Handling Mode 1 Mode 2 Holday Mode When a call comes in on this phone number, perform the following action: ring extensions Edit If all extensions are busy or the call is not answered: perform no action after 5 rings.	Global dial plan c	onfiguration date: Feb 15 09:53		
Mode 1 Mode 2 Holiday Mode When a call comes in on this phone number, perform the following action: ring extensions Edit If all extensions are busy or the call is not answered: perform no action after 5 rings. 	Call Haadling			
Mode 1 Mode 2 Holday Mode When a call comes in on this phone number, perform the following action:	Call Handling			C
When a call comes in on this phone number, perform the following action: ring extensions Fidit If all extensions are busy or the call is not answered: perform no action Fidit for the call is not answered: Fidit for the call is not answered	Mode 1	Mode 2 Holiday Mode		Ϋ́,
ring extensions Edit If all extensions are busy or the call is not answered: perform no action after 5 rings. 	When a call c	omes in on this phone number, perfo	orm the following action:	
If all extensions are busy or the call is not answered: perform no action Image: Control of the call is not answered: after 5 after 5 Image: Control of the call is not answered:	ring extension	ns 🔹 Edit		
perform no action - after 5 - rings.	If all extension	ns are busy or the call is not answer	red:	
	perform no a	ction 👻	✓ after 5 ✓ rings.	

- 2. Check the Activate Global Dial Plan checkbox.
- **3.** Check the *This site is the master location* checkbox. Enter your public IP address or the IP address of the system if you are using a VPN.
- 4. Create an 8-character User Key. The user key will be shared among the locations.
- 5. Create a *Password*. The password is for authentication of the plan and should be known to the central admin only.

Add the locations

1. Click on the Create Dial Plan File button. The Create Dial Plan File window opens.

Location name	Loc Code	IP/FQDN	SIP Port	HTTP Port	Numbering Plan
			A	id Edi	it Remove

2. Click the Add button. The Add Location window opens.

Name:	Location Code:
IP/FQDN:	Numbering Plan: 3 digits 👻
SIP Port:	HTTP Port:

- 3. Enter the Name of the master location.
- **4.** Assign a *location Code*. Location codes can be 2 or 3 digits. Each location must have the same number of digits.
- 5. Enter the IP address or Fully Qualified Domain Name of the master location
- 6. Select the number of digits in the master location's dialing plan. For optimum usability, each location should use the same number of digits in extensions, but the plan will work if they differ.
- 7. The *SIP Port* and *HTTP Port* values are prepopulated. If you have changed the defaults for the site, you must enter the correct values.
- 8. Click *Done* and repeat for all of the sites.
- 9. Save settings to the system.

Setting up the global dial plan: all other systems

1. Select the Global Dial Plan page in the FortiVoice Management software at each location.

			_					
Activate Global Di	ıl Plan ———							
Automatic Synchr	onization ——							@
This site is the	master location							
Server Address:	64.26.140.142			Crea	te Dial Plan File			
User Key:	User1234							
Password:	Password							
	Location Name				View Direc	tory		
	Location Name				View Direc	tory		
0	Location Name				View Direc	tory		
Global dial plan co	Location Name Location Code nfiguration date	: Feb 15 09:53			View Direc	story		
Global dial plan cr Call Handling	Location Name Location Code nfiguration date	: : Feb 15 09:53			View Direc	story		(c
Global dial plan co Call Handling —— Mode 1	Location Name Location Code nfiguration date	: : Feb 15 09:53			View Direc	tory		@
Global dial plan co Call Handling	Location Name Location Code nfiguration date	: : Feb 15 09:53 day Mode			View Direc	<u>ttory</u>		@
Global dial plan co Call Handling Mode 1 When a call co	Location Name Location Code Infiguration date Wode 2 Holi mes in on this p	: : Feb 15 09:53 day Mode hone number,	perform the 1	following action:	View Direc	tory)		@
Global dial plan co Call Handling Mode 1 When a call co ring extension	Location Name Location Code Infiguration date Mode 2 Holi mes in on this p S •)	: Feb 15 09:53 day Mode hone number, Edit	perform the 1	iollowing action:	View Direc	tory		
Global dial plan co Call Handling Mode 1 When a call co (ring extension If all extension	Location Name Location Code Infiguration date Mode 2 Holi mes in on this p s s are busy or th	: Feb 15 09:53 day Mode hone number, Edit e call is not an	perform the t	iollowing action:	View Direc	<u></u>		
Global dial plan co Call Handling Mode 1 When a call co ring extension If all extension perform no ac	Location Name Location Code Infiguration date Mode 2 Holi mes in on this p s v (s are busy or th tion v)	: Feb 15 09:53 day Mode) hone number, Edit e call is not an	perform the 1) swered:	following action:	View Direc	<u>tory</u>		
Global dial plan co Call Handling Mode 1 When a call co ring extension If all extension perform no ac	Location Name Location Code Infiguration date Mode 2 Holin mes in on this p s v (s are busy or th tion v)	: : Feb 15 09:53 day Mode) hone number, Edit e call is not an	perform the f	following action:	View Directer S - rings.			
Global dial plan co Call Handling Mode 1 When a call co (ring extension If all extension perform no ac	Location Name Location Code Infiguration date Mode 2 Holin mes in on this p s	: : Feb 15 09:53 day Mode) hone number, Edit e call is not an	perform the t	following action:	View Direc			

- 2. Check the Activate Global Dial Plan checkbox.
- 3. Enter the public IP address or the IP address of the master system if you are using a VPN.
- **4.** Enter the User Key.
- 5. Save settings to the system.

Using the directory

Click the *View Directory* button for the full directory of all extensions in all locations. Each location's directory can be exported as a comma-separated file by clicking the *Export* button.

Configuring Call Handling (optional)

If a user dials a location code without an associated extension number, the call will go through to that location. You can set the way a location handles those calls in the *Call Handling* section.

For each available mode, select the desired call handling from the first pull-down menu, and select a resource, if applicable, from the second.

VoIP Configuration

The *VoIP Configuration* page allows you to set up to four service provider profiles. You can also view registration status, set codec options, and reserve VoIP lines for specific setups.

Setting up a service provider profile

A service provider profile contains the settings that allow your system to register with the provider.

Service configuration guides for certified VoIP service providers are available in the *Windows* program folder under *Service Configuration Guides*.

- 1. Select the VoIP Configuration page.
- 2. Select a Profile (SP 1 to SP 4).

Profile	Profile Name	Activate Profile
SP 1	Service Provider 1	
SP 2	FortiCall	Profile name: FortiCall Click to register for an account or free trial
SP 3	Service Provider 3	
SP 4	Service Provider 4	Additional Settings
		Registration Method Username and Password Your VoIP Provider will have a preferred method of registration. Select a method below: Registration Method: By Account Number - Master

Activate profile

You can set up a service provider profile automatically or manually.

Automatic configuration

- 1. Select the Activate Profile checkbox.
- 2. The Service Provider menu offers a list of certified VoIP service providers. If your service provider appears in the menu, click on the name. The name is then displayed in the Service *Provider* field.

3. Click the *Update Configuration* button. The essential settings for communication with the service provider's registration server will be completed automatically.

/oIP Configuration	
Profile Profile Name SP1 Service Provider 1 SP2 FortCall SP3 Service Provider 3 SP4 Service Provider 4	Activate Profile Service Provider: FortiCal Profile name: FortiCal Click to register for an account or free trial Additional Settings r. Registration Method
	Your VolP Provider will have a preferred method of registration. Select a method below: Registration Method: By Account Number - Master
	VoIP Provider Proxy/Registrar server name: forti1.secureto.us Registrar server name: forti1.secureto.us Outbound proxy: forti1.secureto.us
	Realm/domain: forti1.secureto.us
- System VoIP Options	D information for all outgoing VoIP calls Iter ID information for all outgoing VoIP calls Reserve VoIP Lines



For selected service providers in your region, a link may appear beside the *Profile name* to permit easy access to the provider's website.

4. If you want to customize other aspects of your VoIP lines, you may do so in the System VoIP Options area. See "System VoIP options" on page 32.

Account-specific and number-specific settings are not automatically configured. These must be entered on the *VoIP Numbers* page.

Manual configuration

- 1. Select the Activate Profile checkbox.
- **2.** Enter the *Profile name*. The default profile name is *Service provider n* (e.g. *Service provider 1*).

Additional settings

Public IP

If your service provider requires you to register using your private IP address, select the *Disable public IP address substitution* checkbox. Check with your service provider.

NAT

1. If your service provider requires keep alive messages, and if your router does not support uPNP, check the *Enable NAT keep alives* checkbox.

- a. Select the method used to keep ports open. Choices are:
 - Simple ping A standard ping message that works with all SIP servers.
 - Nortel ping A ping message that works with Nortel SIP servers (e.g. Nortel MCS 5200).
- **b.** If necessary, change the ping frequency. The default setting is 45 seconds.

Preferred ID

Preferred Identity is a supported service provider feature that allows your system to make calls with an anonymous caller ID. Please ensure your service provider offers this feature before enabling it, or you might be unable to make calls.

Codec

You can specify which codecs to use by clicking the *Additional Settings* button. See "Setting codec options" on page 31.

Registration method

Some providers require the system to register using the username or account information rather than the VoIP number(s) provided. If so, check the *Register with authentication username* box to have the system register with the username information provided in the *VoIP numbers* page. Check with your VoIP service provider if you're uncertain which method of registration is required.

VoIP Provider

Enter the IP addresses or public domain names, as provided by the service provider, into the following boxes. If the service provider does not specify a value, leave the box blank.

- Proxy/Registrar server name
- Registrar server name
- Outbound proxy
- Realm/domain

The *View All Registrations* button will allow you to confirm connection to your service provider once you have completed the configuration of your VoIP numbers. See "Viewing registration status" on page 31.

If you want to customize other aspects of your VoIP lines, you may do so in the *System VoIP Options* area. See "System VoIP options" on page 32.

Viewing registration status

Clicking the *View All Registrations* button shows a window with a list of VoIP numbers, their registration status, and the number of seconds until their registrations with the SIP server will expire. This confirms the system is registered with a SIP server.

1. Click View All Registrations. The Registration status window appears.

Reg	gistrati	on status		2
R	egistrat	ion status —		
	Client	Chabur	Evolvor	
		Dedictored	170	
	350	Registered	170	
	351	Registered	171	
	352	Registered	1/2	
	353	Registered	162	
	354	Registered	163	
			my Service Provider	
			Refresh	

- 2. Choose All Registrations or an active profile.
 - Client lists the VoIP numbers set up within the system.
 - Status is the registration status (Registered or Not registered).
 - *Expires* is the amount of time, in seconds, until the client has to re-register with the SIP server.

Setting codec options

A codec is a method of compressing and decompressing audio signals for communication across a network. The system supports the G.729 and G.711 (μ -law or A-law) codecs for VoIP calls. If your service provider or equipment requires specific codecs for VoIP or Fax over IP calls, you can restrict the system to use the required codec. The following codes are supported:

If your Internet co bandwidth). The c	nnection can su lefault setting is	pport high bandwidth G.729 (25 kbps band	content, you can select G.711 (100 kbps width).
	☑ G.711µ	👿 G.711A	Voice activity detection (VAD)
Preferred codec:	G.729 👻		

- *G.729* This codec provides good quality. It requires the least bandwidth and accommodates the highest number of concurrent calls.
- G.711μ This codec provides high quality and supports Fax over IP. It requires the most bandwidth and accommodates the fewest number of concurrent calls. G.711μ is used in North America and Japan.
- *G.711A* This codec provides high quality and supports Fax over IP. It requires the most bandwidth and accommodates the fewest number of concurrent calls. *G.711A* is used worldwide outside North America and Japan.
- Voice activity detection (VAD) Enabling VAD reduces voice bandwidth when no speech is detected, and reduces transmission of background noise. We recommend disabling VAD to keep bandwidth available for speech.

You may select which codecs to use or clear the unsupported codecs as well as select a *Preferred codec*.

System VoIP options

VoIP Caller ID

The VoIP Caller ID area allows you to set up the source for the caller ID name for outbound VoIP calls. The same setting is used for Global Dial Plan and all service provider profiles. Extension names are used by default.

- **1.** Set the caller ID information for outgoing VoIP calls:
 - To use the System name from the Administration page, select Use system name in Caller ID information for all outgoing VoIP calls.
 - To use the First name and Last name from the Local Extensions/Fax page, select Use extension names in Caller ID information for all outgoing VoIP calls.

See "Caller ID settings" on page 49 to select the phone number that will appear on the other phone.

Line reservation

By default, all VoIP lines are available for external IP extensions, calls to other locations in a multi-branch network, and/or service provider calls on a first-come first-served basis. You can also reserve VoIP lines for a specific use. For example, you could set aside two lines for external IP extensions.

Reserving VoIP lines guarantees resources for a specific network. Sharing VoIP lines uses them most efficiently.

When reserving VoIP lines for a service, the choice ranges from 1 to the number of unreserved VoIP lines at this location. *Shared* means do not reserve any VoIP lines for this service. Only unreserved lines will be available.

1. Click Reserve VoIP Lines. The VoIP Resource Reservation window appears.

- VolP Resource Reservation	
External IP extension calls: Shared	
Global Dial Plan calls: Shared	
Service Provider 1 calls: Shared	
FortiCall calls: Shared	
Service Provider 3 calls: Shared	
Service Provider 4 calls: Shared	
Total VoIP lines available on system: 8	
Total VoIP lines shared: 8	
Total VolP lines reserved: 0	

2. Select the number of VoIP lines to reserve for *External IP extension calls* using its pull-down menu.

- **3.** Select the number of VoIP lines to reserve for *Global Dial Plan calls* using its pull-down menu.
- 4. Select the number of VoIP lines to reserve for *Service Provider calls* for each provider using its pull-down menu.

User Privileges

User privileges control the lines and features an extension can use. You can set different profiles to apply to management and staff, for example, or for different departments. You can also route or block calls to specific numbers. Each extension must have a user privilege profile. This is also known as class of service.

You can configure user privilege profiles on the User Privileges page. By default, Profile 1 is active. Select any other profile and check the Activate User Privileges checkbox to activate it, or click on an active profile to edit its settings.

Privileges	
e Name: Employee Level 1 Default Outgoing Access Ø Hunt Group Access 9,81,82 Edit PIN required for outbound access Ø Call Bridge (DISA) Access Ø Intercom Calls Ø Ø Intercom Calls Ø	
tional Settings	

Outgoing Access

The Outgoing Access area controls the hunt groups an extension can use. If a user dials a hunt group without access, a PIN code will be required to gain access. By default, Profile 1 has access to all hunt groups.

To assign hunt group access:

- 1. Check the Hunt Group Access checkbox
- 2. Click the *Edit…* button
- 3. Check the boxes for the hunt groups this user profile may access.

Blocking outbound access

If you wish all users in a profile to use a PIN code in order to access a hunt group, check the *PIN* required for outbound access checkbox.

Call Bridge (DISA) access

Call bridge access allows the user to call into the system and seize an outbound line for calls. If you wish the users in a profile to have access to call bridge, check this checkbox. For more information, see "Using call bridge" on page 169.

Feature Access

Check the features that you wish this profile to have access to.

Intercom Calls — Allows users to call another extension. Typically, courtesy phones aren't allowed to call other extensions.

Remote Extensions – Allows users to call remote extensions.

Ring Groups – Allows users to call or transfer calls to Ring Groups.

Speed Dial - Allows users to dial system speed dials

Command Mode — Allows users to enter command mode to make changes to the system.

Paging – Allows users to page other extensions or use the overhead paging system.

Global Dial Plan – Allows users to call or transfer calls to other locations.

Log in/out of Ring Groups - Allows users to join or leave ring groups.

Call Barge — Allows a user to join another user's call. This feature is restricted to calls connected via PSTN telephone line to an outside party.

Additional Settings

Personal Identification Numbers

If your user profiles have access to call bridge or require a PIN code to get outbound access, you must assign PIN codes to the users. PIN codes must be 4 to 8 digits. PINs can be assigned a different user privilege profile to supersede the current active profile to provide additional access if required.



Enable password protection when PINs are used. Change the system password frequently to prevent unauthorized users from making calls or changing the configuration. See "Administration" on page 6.

The PIN Code Configuration area allows you to add and remove access codes.

Adding a PIN

- 1. Select an ID.
- 2. Enter a *Name*. This is typically the name of the user, or the name of the office where the extension is located.

Alternatively, you can click the *Browse* button to display the *Browse for Extension* window. It allows you to select an extension number and the associated first name and last name as the name.

Browse for Extension
Extension Name: 111 - John Doe 🗸
<u>O</u> K <u>C</u> ancel

If a call is made from a restricted local extension, the call detail record (CDR) will display the name associated with the PIN, and not the PIN itself.

- **3.** Enter a *Code*. This is a numeric code that can be dialed using a telephone keypad. Each PIN works with all restricted extensions.
- 4. Assign a User Privilege to this PIN.

Removing a PIN

- 1. Select the PIN.
- 2. Click Clear Code.

Routing and blocking

The routing and blocking rules control the numbers that can be dialed by users from the system and the lines they access. Each rule can be applied to any of the user privilege profiles to control which extensions are allowed to dial long distance or international calls.



Enable password protection when routing and blocking is used. Change the system password frequently to prevent unauthorized users from making calls or changing the configuration. See "Administration" on page 6.

Routing handles outgoing calls depending on the leading digits. It can also prefix phone numbers with certain carrier codes depending on the leading digits and time of day so calls can use specific telephone lines, alternative carriers, or VoIP lines for least-cost routing. For example, you can route international calls to a VoIP service provider and local calls to a carrier offering discounted rates.

If the user dials a hunt group number, routing and blocking rules can override the selection and use a different hunt group depending on the leading digits.

The leading digits are the first numbers dialed when placing a telephone call. For example, "1900" are the leading digits of 1-900-555-1234.

Calls with leading digits that do not match the routing and blocking rules will be routed to the hunt group originally dialed by the caller.

Routing and blocking acts on the longest leading digits entry matching the dialed number. For example, an entry in the routing and blocking rules blocks numbers with leading digit "1" to prevent long-distance calls. However a second entry routes numbers with leading digits "1800" through a hunt group because they are toll-free. Because the second entry (1800) is longer than the first entry (1), the second entry has precedence. Therefore a call with leading digits "1800" is routed, even though leading digit "1" would otherwise cause it to be blocked. This reduces the number of rules required and can restrict calls to all but specific area codes or countries.

- Leading digits can be 1–11 numbers in length.
- The leading digits exclude the hunt group number dialed by the user.

Carrier codes

A carrier code is prefixed to the phone number dialed by the user. It tells the telephone company to route the call to an alternative carrier. For example, the carrier code could be a calling card number and PIN number. You can set the system to use different carrier codes based on the leading digits and the time of day.

- Carrier codes can be up to 1-24 characters in length.
- Carrier codes can include numbers, *, #, "," (comma for 2-second pause), and *w* (for wait-for-dial-tone). A 2-second pause is automatically added after dialing a carrier code.

Each entry in the routing and blocking table can have one or two carrier codes. *Carrier code 1* is either used all day, or from *Start time 1* until *Start time 2*. *Carrier code 2* is used from *Start time 2* until *Start time 1*.

Use of Carrier Selection Prefix and Carrier Codes may require a subscription to a carrier's discount calling plan. This service may not be available in some countries and on some

telephone companies' telephone lines. Carrier Selection Prefix may not be required if a "Pre-Selection" service is provided by the telephone company.

3-Way Calling/Conference service

The use of the telephone company's 3-Way Calling/Conference service is not recommended on telephone lines when routing and blocking rules are used. The system can't control routing through these services.

Setting up routing and blocking

- 1. Configure the system password and hunt groups before setting up routing and blocking.
- **2.** Click the User Privileges page. Select the Edit Rules button in the Routing and blocking section. The Edit Routing and Blocking Rules window appears.

Edit Rou	uting and Bloc	k Rules		
The sy	ystem can route o	r block outgoing	calls ba	sed on the digits captured while a user dials an external number.
ID	Number	Rule		_ Rule
1	12321321321	Block calls		
2	011	Block calls		Number: 12321321321
3	1	Block calls		Pule: Block calls
4				
5			=	Carries Carda Das fa
6				
7				Carrier code 1:
8				
9				Start time 1: 09:00 AM 💲
10				
11			_	Carrier code 2:
12			_	
13				Start time 2: 05:00 PM
14	1		- 1	
15	1			r Applicable User Privileges profile
16	1		- 1	
1/	1			V 1 - Employee Level 1 0
18			- 1	
19				
20				3 - Manager 8
21	1			
22				4 - Director 9
24				
25				5 - Chris Lee 10
26				
27			~	
				Done Cancel

Assigning routing and blocking rules

- **1.** Select a free table entry, or edit an existing entry.
- 2. Enter the leading digits in the *Number* box. Omit the hunt group number from the leading digits. The system will only block or redirect calls where there is a matching leading digit match.

- 3. Select the Action. Choices are:
 - *Hunt group (n)* Routes the call to the hunt group.
 - Block calls Blocks the call.
- 4. Assign the rule to applicable user privilege profiles.

Carrier code prefixes

The *Carrier Code Prefix* area allows you to set carrier codes the system will automatically use when the leading digits are dialed. A carrier code is prefixed to the number dialed by the user. It tells the service provider to route the call to an alternative carrier. For example, the carrier code could be a calling card number and PIN number.

- 1. If the route requires one carrier code for the whole day:
 - a. Enter the carrier code in Carrier code 1.
 - **b.** Clear the *Carrier code 2* checkbox.
- 2. If the route requires two carrier codes, one for each part of the day:
 - a. Enter the first carrier code in Carrier code 1.
 - **b.** Enter Start time 1.
 - c. Select the Carrier code 2 checkbox.
 - d. Enter the second carrier code in Carrier code 2.
 - e. Enter Start time 2.

Regulatory advisory notice



Call Redirection & Service Provider Billing Advisory

Use of the routing and blocking and call detail recording features does not imply any guarantee whatsoever by regulatory authorities, your telephone service provider(s), Fortinet or its distributors and resellers, with regard to the accuracy of these features and that the use of such a features may not be considered by a telephone company in any disputes which may arise regarding the accuracy of any subscriber's telephone account.

Conference Bridge

The system supports up to 10 conference bridge accounts that will allow up to 8 participants to join a conference. A conference bridge number is required for users to dial when they call into the system to enter the bridge. Each caller enters an access code to join the correct conference. The participant access codes are created by the moderator.

Each moderator is assigned a unique access code to allow them to create and control their conference bridge. By default, participant codes are active for 24 hours, but can be extended to 1 week or indefinitely.

Local Extensions/Fax

A local extension can either be an analog extension or an IP extension.

An analog extension is a device (standard phone, cordless phone, fax machine or modem) connected to an extension jack on the unit.

An IP extension is an IP phone connected through a network to a system. An internal IP extension is a phone connected on the same LAN as the system. An external IP extension is a phone connected outside the LAN. See "Adding IP phones" on page 55 for more information.

The Local Extensions/Fax page allows you to set up a local extension.

Local Extensions/Fax page

The *Local Extensions/Fax* page shows the extension type, user name, make and model, prompt language, user privileges, extension details, programmable keys, direct line access, voicemail, hotline, caller ID, and call cascade for each local extension.

In the list of extensions on the left side of the page, manually configured IP extensions will be shown in bold, and unregistered or disconnected extensions will be shown in red.

Before you begin, decide the number of digits you want your system to use for extensions, ring groups, mailboxes and speed dials. You can use 3, 4 or 5-digit numbers. The default is 3. If you want to use 4 or 5-digit numbers, you must change the *System Numbering Plan* setting on the *Administration* page, or your extension numbers will be rejected by the system.

Numbers available per plan: 3-digit numbers: 100 to 899 4-digit numbers: 1000 to 8999 5-digit numbers: 10000 to 89999

Extension tab

1. Select the Local Extensions/Fax page.

ocal Extensions	are telephones or fa	ax machines connected to the system.	
Add Ext First nam	Remove Last name	Extension Additional Settings Voicemail	
IOO Don	Jones	Extension 100	
14 Andy	Parker		e
115 Joe	Smith	First name: Don Last name: Jones	
133 Jason	McKenna	Extension number: 100 Extension type: IP Extension	
Jackie	Harrison	Manufacturer: Fortinet/TalkSwitch Model: 550i	
		Unar Drivitanes (A. Employee Level 4	
		P Extension Details	
		Location: Internal Extension status: Registered	
		MAC address: 001A7EAB02E9 Select ID address: 10.10.10.37	
		Unit: 1	
		Configure K	(eys
		Call Handling	(?
		Made 4 (devillede)	
		Mode I (daymode) Mode 2 (nightMode)	1
		Busy No Answer Answered Do Not Disturb	
		If extension is busy: go to voicemail 🗸 100 - Don Jones	-
		If busy or not answered after: 4 🚽 rings go to voicemail 🚽 100 - Don Jones	ΞH
		If busy or not answered after: 4 rings baco up	- 11
			- 11

2. Click Add.

3. In the *Add User / Extension* dialog box, select the *Extension type*. Choices are *Regular phone or fax* and *IP Extension*. *Regular phone or fax* allows you to set up an analog extension.

Extension type:			
	IP Extension	~	
Unit:	1 💌		

- **4.** In a multi-unit system, select the unit the extension will be associated with. This unit will hold the voicemail for the extension. For analog extensions, this must be the unit that the analog phone is plugged into.
- 5. Analog extensions: select the jack the analog phone is plugged into. Click Next.
- 6. Enter the user's *First name* and *Last name*. The names are used in the dial-by-name directory and caller ID.

Add User / Extens	ion		X
A suggested range	is 100-199		
First name:			
Last name:			
Extension number:			
Manufacturer:	Other		~
Model:	Other		~
MAC address:			Select
	(Finish	<u>C</u> ancel

- 7. Enter the Extension number.
- 8. Select the Manufacturer.

We strongly recommend you only use supported IP phones. However, if you have an unsupported IP phone, you can select *Other*. Other IP phones that support the G.711 codec (μ -law or A-law) may work with the system, but some features may not work. As the system cannot enable special features, or customization of these IP phones, further configuration will be limited to the programmable options on the phone itself.

- 9. Select the phone *Model*. Supported models and manufacturers vary by market.
- **10.** If you're setting up an IP phone, enter the *MAC address* of the phone. You can click *Select* to choose from addresses of phones connected to the system, or enter the MAC address manually. The MAC address is printed on the bottom of the phone and on the phone's shipping box.
- 11. Click Finish.

The information you configured now appears in the *Extension* tab, and you can edit it directly there if you need to make changes. Select any extension in the menu under the *Add* and

Remove buttons to access the configuration details. In the *Extension* tab, assign a User Privilege for each extension. The default is *1 - Default*.

The Voicemail tab is now visible for each extension. See "Voicemail tab" on page 49.

Regular extension details

If you set *Extension type* to *Regular phone or fax*, the *Regular Extension Details* area is enabled. You can edit the unit number and jack to which the extension is connected.

IP extension details

If you set Extension type to IP extension, the IP Extension Details area is enabled.

Once you save the configuration and reboot the IP phone, the system will configure the IP phone. *Status* will change to *Registered*, and the IP address of the phone will appear.

Location:	External	~		Extension sta	atus: Registered	
MAC address:	001A7EAEC53E		Select	IP addr	ress: <u>192.168.1.102</u>	
TalkSwitch unit:	1	~		Time zone:	(-05:00) Eastern (US/Canada)	~

If you connected a proprietary IP phone to the LAN and used it to select an extension number, the *Location* and *MAC address* for that extension will be complete.

- 1. Set the *Location* of the phone. Choices are:
 - Internal An internal IP extension is located within the office, and is connected to the same LAN as the unit.
 - External An external IP extension is located outside the office, and is connected to the unit via Internet.
 - *Both* The phone can be used as an internal or external IP extension. This is available for the Counterpath eyeBeam softphone only.



Using an external IP extension to call an emergency service number will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP extension stating:

If an emergency call is made from this phone, you must provide your address to the emergency operator.

- 2. If you set *Location* to *External*, select the *Time zone* that matches the location of the IP phone.
- **3.** Use one of the procedures below to identify the IP phone, depending on the make and model.

Configuring IP phone keys

- 1. If the *Configure Keys* button is visible and active, you can click it to set up the actions of your phone's keys. See "About programmable function keys" on page 58 for further instructions.
- 2. Choose *File* > *Save*. The system will create a configuration file that the phone will download when the phone is restarted.

Setting handset ID for 850i or 860i phones

- 1. Select the Handset ID for the extension. Use the handset name on the handset screen.
- 2. Choose *File* > *Save*. The system will create a configuration file that the phone will download when the phone is restarted.

Counterpath or other IP phone

1. Enter the *Username* and *Password*. The default settings are user[extension number] and pass[extension number]. These are used to manage calls between systems and the IP extension and must match the username and password of the IP phone. Change only if required.

About call cascades

The *Call Handling* area allows you to set up call cascades for a local extension. A call cascade routes a call to an alternative or a series of alternatives when the target extension doesn't answer. You can set up different call cascades for Mode 1 and Mode 2.

For example, if a call reaches your desk when you're away, it can be sent to another local extension. If that extension isn't answered, the call can be routed to your cell phone. If you don't answer your cell phone, the call can go to your voicemail.

Note that the system can be configured to ignore the call cascade for an unanswered intercom call, or for an unanswered transferred call. See "Transfer Preferences" on page 132.

Setting up call handling

Busy call cascade

The busy call cascade is used when the extension is busy.

- 1. Select the *Mode 1* tab or the *Mode 2* tab.
- 2. Select the *Busy* tab.

de 1 Mode 2					
Busy No Answer Answered	Do No	t Disturb			
W 1					
If extension is busy:		go to voicemail	*	111	~
If extension is busy: If busy or not answered after:	4 🗸 ri	go to voicemail	*	111 111	✓

- 3. Set up the first alternative.
 - a. Select the action in the If extension is busy list. Choices are:

- go to voicemail Transfers the call to the selected voicemail.
- go to local extension Attempts to transfer the call to the selected local extension.
- go to remote extension Attempts to transfer the call to the selected remote extension.
- go to ring group Attempts to transfer the call to the selected ring group.
- play announcement Plays the selected announcement.
- invoke call waiting Notifies the user that a caller is attempting to reach them.
- go to auto attendant Routes the call to the selected auto attendant.
- queue at extension Transfers the call to the local extension's call queue.
- *play busy tone* Plays a busy tone in the caller's phone.
- hang up Disconnects the telephone line.
- **b.** Select the resource, if applicable.
- c. If you selected *queue at extension*, the caller will hear a prompt each time the reminder timer expires. The prompt says *"If you wish to continue holding, please remain on the line. To leave a voicemail message, press 1".* You can set the duration of the reminder timer, to control how often the caller hears the prompt. See "Call Reminders" on page 135.
 - i. Choose Options > Call Reminders. The Call Reminder Timers window appears.

fold ringback reminder timer:	90 seconds	~			
ark ringback reminder timer:	90 seconds	۷			
Queue options reminder timer	2 minutes	~			

ii. Set the Queue options reminder timer.

- 4. If permitted, set up the second alternative.
 - **a.** Set the *If busy or not answered after* list to the number of rings to try the first alternative. Choices range from 1 to 9.
 - **b.** Set the action.
 - c. Select the resource, if applicable.
- 5. If permitted, set up the third alternative.
 - **a.** Set the *lf busy or not answered* list to the number of rings to try the second alternative. Choices range from 1 to 9.
 - **b.** Set the action. Choices are:
 - go to voicemail
 - go to auto attendant
 - play announcement
 - hang up
 - **c.** Select the resource, if applicable.

No answer call cascade

The no answer call cascade is used when the extension is not answered.

1. Select the *Mode 1* tab or the *Mode 2* tab.

2. Select the No Answer tab.

Busy No Answer Answered	Do	Not Dis	turb			
If this extension is not answered after:	4	rings	go to voicemail	*	111	~
If busy or not answered after:	4 🔨	rings	go to voicemail	~	111	¥.
If busy or not answered after:	4 🔨	rings	go to voicemail	~	111	~

- 3. Set up the first alternative.
 - **a.** Set the *If this extension is not answered after* list to the number of rings to try the extension. Choices range from 1 to 9.
 - **b.** Select the action. Choices are:
 - go to voicemail
 - go to local extension
 - go to remote extension
 - go to ring group
 - play announcement
 - go to auto attendant
 - go to extended ringing
 - hang up
 - c. Select the resource, if applicable.
- **4.** If permitted, set up the second alternative.
 - **a.** Set the *lf busy or not answered after* list to the number of rings to try the first alternative. Choices range from 1 to 9.
 - b. Set the action. Choices are:
 - go to voicemail
 - go to local extension
 - go to remote extension
 - go to ring group
 - play announcement
 - go to auto attendant
 - hang up
 - c. Select the resource, if applicable.
- 5. If permitted, set up the third alternative.
 - **a.** Set the *If busy or not answered after* list to the number of rings to try the second alternative. Choices range from 1 to 9.
 - **b.** Set the action. Choices are:
 - go to voicemail
 - go to auto attendant
 - play announcement
 - hang up
 - **c.** Select the resource, if applicable.

Answered call cascade

The answered call cascade is used to enable call screening, typically for cell phone remote extensions.

- 1. Select the *Mode 1* tab or the *Mode 2* tab.
- 2. Select the Answered tab.

Busy No Answer Answered	Do	Not Dis	turb			
If this extension is not answered after:	4 🗸	rings	go to voicemail	~	111	~
If busy or not answered after:	4 ~	rings	go to voicemail	~	111	✓
If hugu as not an unread offers	4 ~	rings	go to voicemail	~	111	×

- 3. Set the When a call is answered list. Choices are:
 - *stay connected* Transfers the call to the extension. This option disables the remaining controls in the tab. This is the default.
 - play caller's name first Performs a screened transfer. The caller is prompted to say his
 or her name. When the user answers the phone, the system plays the caller's name, and
 prompts the user to accept the call. If the user accepts the call by pressing #, it gets
 connected. If the user rejects the call by pressing * or hanging up, the call is routed to
 the first alternative.
- 4. Set up the first alternative.
 - a. Set the If a call is rejected from this extension action.
 - **b.** Select the resource, if applicable.
- 5. If permitted, set up the second alternative.
 - **a.** Set the *If busy or not answered after* list to the number of rings to try the first alternative. Choices range from 1 to 9.
 - **b.** Set the action.
 - c. Select the resource, if applicable.
- 6. If permitted, set up the third alternative.
 - a. Follow the same steps as the second alternative.

Do not disturb cascade

The do not disturb call cascade is used when the extension is in Do Not Disturb mode.

- 1. Select the *Mode 1* tab or the *Mode 2* tab.
- 2. Select the Do Not Disturb tab.

Busy No Answer Answered Do Not D	isturb		
Dial *62 from this extension to toggle Do N	ot Disturb on or off.		
If Do Not Disturb is enabled for this extension:	go to voicemail	✓ 111	~
If busy or not answered after: 4 😒 ring	s go to voicemail	✓ 111	×

3. Set up the first alternative.

- a. Set the If do not disturb is on for this extension list. Choices are:
 - go to voicemail
 - go to local extension
 - go to remote extension
 - go to ring group
 - play announcement
 - go to auto attendant
 - hang up
- **b.** Select the resource, if applicable.
- 4. If permitted, set up the second alternative.
 - **a.** Set the *If busy or not answered after* list to the number of rings to try the first alternative. Choices range from 1 to 9.
 - **b.** Set the action.
 - c. Select the resource, if applicable.
- 5. If permitted, set up the third alternative.
 - **a.** Set the *If busy or not answered after* list to the number of rings to try the second alternative. Choices range from 1 to 9.
 - **b.** Set the action.
 - **c.** Select the resource, if applicable.

Setting do not disturb mode

The user can toggle Do Not Disturb mode by dialing *62 on their local extension.

The administrator can enable or disable Do Not Disturb mode for a local extension by phone.

- 1. Pick up a local extension, or remotely call the system.
- 2. Enter command mode by pressing # on an analog extension phone, or *55# on an IP phone (other brands may use *55 Send or *55 Dial).
- **3.** Enter the system password, followed by #.
- 4. Enter one of the following commands:
 - #60 [Local extension] # to disable Do Not Disturb mode for the local extension.
 - #61 [Local extension] # to enable Do Not Disturb mode for the local extension.
- 5. Press * to exit command mode.

Ignoring call cascades for unanswered intercom calls

An intercom call is an internal call from a local extension to another local extension, remote extension or ring group. If the call is unanswered, the call follows that extension's call cascade by default. However you can configure the system to ignore call cascades for unanswered intercom calls. See "Internal Calls" on page 138.

1. Choose Options > Internal Calls. The Intercom Calls Options window appears.

Intercom Calls Options	
-Intercom calls	© settings
	Use Defaults QK Cancel

- 2. To have the system ignore call cascades for unanswered intercom calls, clear the *Intercom calls follow cascade settings* checkbox.
 - If the extension is busy, the caller will hear a busy tone.
 - If the extension is not answered, it will continue to ring.
 - If a local extension has call screening enabled, the caller won't be prompted for their name. However, if a remote extension or ring group has call screening enabled, call screening will occur. If the call is rejected, the system will follow the call cascade.
 - If a local extension has do not disturb mode enabled, it won't ring, and the caller will hear "*I'm sorry, that extension is unavailable at this time. Do not disturb on. Please try again later*". If all the local extensions in a ring group have do not disturb mode enabled, the caller will hear a busy tone.

Ignoring call cascades for calls transferred from local extensions

A user at a local extension can transfer a call to another local extension, remote extension or ring group. If the call is unanswered, the call follows that extension's call cascade by default. However you can configure the local extensions, remote extensions and/or ring groups to ignore their call cascades for calls that are transferred from a local extension but unanswered. In this case, the system will ring back the local extension that transferred the call, reconnecting the user with the caller they transferred.

1. Choose Options > Transfer Preferences. The Transfer Preferences window appears.

Specify the options for transferring a caller:	
To Local Extension To Remote Extension To Ring Group From Auto Attendant From Home Phone	
_If the Extension is Busy	-0
◯ Ring back the extension that transferred the call	
O Perform the Busy action	
If the Extension Does Not Answer-	-0
Ring back the extension that transferred the call	
⊙ Perform the No Answer action	

- **2.** Select the appropriate tab.
 - To configure call handling for calls transferred to local extensions, select the *To Local Extension* tab (see "To local extension tab" on page 133).
 - To configure call handling for calls transferred to remote extensions, select the *To Remote Extension* tab (see "To remote extension tab" on page 133).
 - To configure call handling for calls transferred to ring groups, select the *To Ring Group* tab (see "To ring group tab" on page 134).
- 3. Select the call handling setup for calls transferred from a local extension but unanswered:
 - To have the system ring back the local extension that transferred the call, select *Ring* back the extension that transferred the call.
 - To use the Busy call cascade, select *Perform the Busy action*.
 - To use the No Answer call cascade, select Perform the No Answer action.

Additional Settings

The *Additional Settings* tab enables you to set up or customize access to lines or hunt groups for outbound calls, designate the extension as a hotline and select CallerID display options.

Prompt Language

In the *Additional Settings* tab, you can select the language for prompts heard by the user of the extension in the *System prompt language* list.

Direct line access

Under the *Additional Settings* tab, the *Direct Line Access* area allows you to select the hunt group that the local extension will use automatically. When an analog phone goes off-hook or a number is dialed on an IP phone, the system will automatically find an available line within that hunt group.

Example: You have a fax machine connected to the local extension and don't want to reprogram the speed dial numbers with hunt group numbers. Enable direct line access and select the hunt group. As soon as the fax goes off-hook, it finds an available line within the line hunt group.

- 1. Under the *Additional Settings* tab, select the *Use direct line access on* checkbox. The window enables the list of line hunt groups.
- **2.** Select the hunt group.



When using direct line access, you hear the telephone company dial tone when you pick up the handset. You do not hear the internal dial tone. The following features are only available from the internal dial tone:

- Intercom calls
- System speed dial numbers
- Calling the receptionist
- Attaching an account code
- Intercom paging
- Group paging
- Overhead paging
- Hunt groups
- Stutter dial tone for new voicemail
- Voicemail retrieval/access
- Entering command mode
- Call pickup
- Retrieving a parked call
- Do Not Disturb functions

To obtain an internal dial tone on an analog extension that is set for direct line access, pick up the handset, and then press *Flash* or *Recall*. On an IP extension, press **, and then dial the number or function.

Hotline access

The *Hotline Access* area allows you to select the resource that the extension will connect to. This restricts the extension to one special task, and you cannot use the extension for any other purpose.

An analog phone or a corded FortiFone IP phone will automatically connect to the resource when you lift the handset or press the speaker or headset button. Other IP phones do not support hotline access.

1. Click the Additional Settings tab.

- 2. Select the action in the *Connect to the following resource* list. Choices are:
 - None Selected Does not automatically connect to a resource.
 - go to voicemail Connects to the selected voicemail.
 - go to local extension Connects to the selected local extension.
 - go to remote extension Connects to the selected remote extension.
 - go to ring group Connects to the selected ring group.
 - go to auto attendant Connects to the selected auto attendant.
- **3.** Select the resource. Depending on the action, resources are voice mailboxes, extensions, or auto attendants.

Caller ID settings

The *Caller ID Settings* area allows you to select the phone number that will appear on the other phone when the local extension is used to make an outbound call on a PRI trunk or VoIP trunk.

- 1. Click the Additional Settings tab.
- 2. Set each Caller ID to be used for each service.

Choices include *Default number*, and the numbers set up in the *PRI Numbers* page or *VoIP Numbers* page. If *Default number* is selected, the phone number associated with the line is used. This selection makes the most sense if all your numbers have been configured to handle all inbound calls the same way.

See "System VoIP options" on page 32 to select the name that will appear on the other phone for VoIP calls.

Voicemail tab

The *Voicemail* tab allows you to activate the voice mailbox or announcement, load the greeting, set the action to perform if the caller dials *0* (9 in some regions), and set up voicemail notification.

By default, each local extension mailbox is activated.

The recording time for internal music on hold, voicemail, and the auto attendants is shared on the unit.

Note that the system allows up to 99 voicemail messages per mailbox. Once a mailbox fills up, callers won't be able to leave voicemail messages for that user. Therefore users should delete voicemail messages before the mailbox fills up.

1. Select the *Local Extensions/Fax* page.

2. Select the Voicemail tab.

	Add	Remove		
Ext	First Name	Last Name	Extension Additional Settings Voicemail	
23	Fax	Machine	Activate Voice mailbox 222	
22	Andy	Parker		
3	Jackie	Harrison	Mailbox name: Andy Parker	
24	Karen	Allidina	-Mailbox Greeting	
5	Jason	McKenna		0
			Action to perform when caller presses 0 while in voicemail. Mode 1: go to local extension v 224 - Karen Allidina v	
			Mode 2: perform no action	
			Notification Options	0
			Email Notification Call / Dansa Natification Manager Matter Links	

Note: Depending on the region, an operator may be dialed using 9 or 0.

3. If necessary, select the *Activate Voice mailbox* checkbox.

Mailbox greeting

The *Mailbox Greeting* area allows you to load a greeting, and configure the voice mailbox as a mailbox or announcement. Note that you can also record a greeting using a local extension connected to the unit.

The greeting should tell the caller to dial 0 (9 in some regions) to perform the action selected in "Caller options", if you will configure these options.

- **1.** To record a greeting:
 - a. Pick up a local extension connected to the unit.
 - **b.** Press ** <extension number>, and then follow the prompts to record a greeting.
- **2.** To load a greeting:
 - **a.** Obtain an 8 kHz, 8 bit, Mono, u-Law .wav file that contains the greeting. The maximum file size is 5 minutes.
 - **b.** Click *Load Greeting*. The Select Wav File window appears.

elect Wav File		
Audio files must be in 8 kHz, 8 b	bit, mono, μ-law format and save	ed as a <i>.wav</i> file.
Choose File		Browse
	Load	Cancel

- **c.** Click *Browse* to select a .wav file. Click *Load* to load the .wav file into the system. *Play* and *Stop* buttons are also provided, if you prefer to listen to the file before loading it.
- **3.** To configure the voice mailbox as a mailbox, leave the *Play announcement only* checkbox cleared. The unit activates the greeting and the mailbox, and enables all the controls on the page. The caller will hear a greeting, and will be able to leave a message in the mailbox.

4. To configure the voice mailbox as an announcement, select the *Play announcement only* checkbox. The unit activates the greeting, disables the ability to leave a message, and disables the *Notification Settings* area. The caller will hear a greeting, but will not be able to leave a message.

Caller options

The *Caller Options* area sets the action to perform if the caller dials 0 (9 in some regions) during the voicemail greeting. You can set up different actions for Mode 1 and Mode 2.

- **1.** For each mode, select the action to perform if a caller dials *0* (9 in some regions). Choices are:
 - go to auto attendant
 - go to local extension
 - go to remote extension
 - go to ring group
 - go to voicemail
 - play announcement
 - *dial-by-name directory*
 - perform no action
- **2.** Depending on the action, enter the resource.

Notification options

The *Notification Options* area allows you to set up voicemail notification, which tells the user if a caller leaves a message. The system can:

- Notify up to four users by e-mail, with the voicemail message included as an attachment.
- Notify a user by phone and/or pager.
- Activate the message waiting light and stutter dial tone on one or more local extensions.
- Perform voicemail screening by routing audio to the PA jack.

Setting up notification by e-mail

Note that e-mail notifications might be routed to the user's junk or spam folder. Advise each user to set up their e-mail accounts to allow e-mail notifications to go to their inboxes.

- 1. Set up the e-mail addresses. See "Email Service" on page 19.
- 2. Select the Email Notification tab.

nail Notificatio	n Cell / Pager Notifica	ition Message Waiting L	ight	
Enable Ema	I Notification			
Email Addres:	ses			

3. Select the Enable email notification checkbox.

4. Click *Edit*. The *Select Email Addresses* window appears.



5. Select up to four e-mail addresses.

Setting up notification by phone

1. Select the Cell/Pager Notification tab.

Remote Phone	r Notification Message Wait	ing Light Pager	7
🗹 Enable Remote Phone no	tification	Enable Pager Notification	
Phone number:		Pager number:	1
Notification Settings		Numeric message:	j
	Configure Notificat	tion Options	_

- 2. Select the Enable remote phone notification checkbox.
- **3.** Enter the *Phone number*. Enter the number as you would normally dial it (i.e. without the hunt group number). You can enter digits 0–9, space, dash, comma, # and *. A comma pauses dialing for two seconds.
- 4. Click Notification Settings. The Notification Settings window appears.

Notification Settings
When notification calls are made to a remote phone number, there are two options for how the notification process should work:
\odot The notification call goes to a remote phone with voicemail. Once the call has been answered the system will play the notification message.
The notification call goes to a phone that may be roaming or answered by an announcement Such as "subscriber unavailable". Once the call has been answered the system will prompt the recipient to press # to hear the notification message, or press * to skip the notification message.
Note: For all cases use the Configure Notification Options button to set how the system responds if a notification call is not answered.
<u>QK</u> <u>Cancel</u>

- **5.** Select the notification setting. Choices are:
 - Select the first option to have the system play the notification message once, when the phone is answered. It will then consider notification to be successful. Note that if there is any answer other than a busy tone, (e.g. voicemail, *"subscriber not available"* message, etc.) It will consider notification to be successful.
 - Select the second option to have the system repeat notification until the user either dials
 * to skip the message, or dials # to play the message. It will only consider notification to
 be successful once the user acknowledges notification by dialing a key. This is useful for
 cell phones where the telephone company plays a "subscriber not available" message
 instead of a busy tone.

Setting up notification by pager

- 1. Select the Cell/Pager Notification tab.
- 2. Select the Enable pager notification checkbox.
- **3.** Enter the *Pager number*. Enter the number as you would normally dial it (i.e. without the hunt group number). You can enter digits 0–9, space, dash, comma, # and *. A comma pauses dialing for two seconds.
- 4. Enter the *Numeric message* that will appear on the user's pager.

Setting up the notification options

1. Click Configure Notification Options. The Notification Settings window appears.

Notification Settings	X
Dial notification phone number using	Hunt Group 9 - General 🗸 🗸
Number of rings between attempts:	after 6 rings
Number of attempts:	3 more times
Interval between attempts:	5 minutes
Perform message notification for:	All Modes
	<u>QK</u> <u>C</u> ancel

- 2. Select the hunt group the system will use to phone or page the user.
- 3. Select the number of rings before aborting attempt, ranging from 1 ring to 9 rings.
- 4. Select the number of attempts the system should make, ranging from once to 10 times.
- 5. Select the interval between attempts, ranging from 5 minutes to 60 minutes.
- 6. Select the modes when the system will perform notification. Choices include *Mode 1*, *Mode 2* and *All Modes*.

Setting up message waiting light activation

If a local extension has a new message in its voice mailbox, the system activates notification on that extension, by default. The user hears a stutter dial tone when they pick up the handset (not applicable to the 850i or 860i phone models). If the extension supports an FSK message waiting indicator (not applicable to IP phones), the message waiting light flashes. Some phone models support a message waiting counter. The display on the extension shows the number of new messages stored in the mailbox(es) associated with that extension.

You can also have the system activate notification on a local extension if a message is received in any other local extension, remote extension, or general mailbox. 1. Select the Message Waiting Light tab.



2. Click the *Edit* button. The *Message Waiting Indication* window appears.

☑ 111 □ 112	~
□ 112 □ 113	
114	
115	
116	
117	
118	~

3. Select the extensions.

Voicemail screening

The system can be configured to perform voicemail screening in addition to overhead paging. In this case it will route audio to the PA jack when a caller leaves a voicemail message or a user accesses a voice mailbox. See "Paging Options" on page 132



If a user has dialed *80 to route music on hold through the PA jack, voicemail screening will not interrupt music on hold.

 Choose Options > Paging. The Paging Options window appears. The PA output option is set to PA Announcements by default, which allows overhead paging but not voicemail screening.

-Audio Output Options	
PA output option: PA A	nnouncements 🗸
-Overhead Paging	
If performing overhead p	paging and want to avoid feedback, then we Delayed Paging feature. The message will be played
over the PA after the me	essage has been recorded.
over the PA after the me	essage has been recorded. ng
over the PA after the me	essage has been recorded. ng

2. To enable voicemail screening, select *Voicemail screening*. The system will continue to allow overhead paging, but will also perform voicemail screening.

Adding IP phones

This section describes how to configure an IP phone as a local extension. An IP phone can be internal (located in the office) or external (located outside the office).

We recommend our IP phones and supported third-party IP phones for ease of configuration. The system currently supports the following third-party IP phones:

- Polycom IP 301, IP 320, IP 330, IP 430, IP 501, IP 550, IP 601 and IP 650. See "Adding Polycom IP Phones" on page 62.
- Grandstream GXP2000 and GXP2020. See "Adding Grandstream IP phones" on page 65.
- Counterpath eyeBeam softphone. See "Adding Counterpath IP phones" on page 68.

Overview

For each supported IP phone, this guide describes how to:

- Add the IP phone to the system configuration.
- Configure the router, if setting up an external IP extension.
- Connect the IP phone to the network.
- Check and update the IP phone firmware version, if required.
- Program the IP phone.

Refer to "Adding Other IP Phones" on page 72 for configuration information of non-supported IP phones.

Adding FortiFone IP phones

Connecting the IP phone to the network

- 1. Connect a network cable between the LAN port on the phone and your network (i.e. router or LAN connection) and then connect power to the phone. Refer to the phone's quickstart guide for specific instructions.
- 2. If the phone is connected to the same network as the system, and an extension has been configured with the phone's MAC address, the system will automatically register and configure the phone. When complete, the phone will display the extension name and extension number.

If the phone is connected to the network and no extension has been configured with the phone's MAC address, the phone will bootup with the model number showing in the display.

Adding the extension to the system

- 1. Launch the Management software.
- 2. Select the Local Extensions/Fax page.
- 3. Click the Add button. The Add User / Extension window appears.

ld User / Extension	
Extension type: P Extension Unit: 1	V
	Next > Cancel

- 4. Set the Extension type to IP Extension.
- 5. In a multi-unit system, select the unit the extension will be associated with. This unit will hold the voicemail for the extension. Click *Next*.
- 6. Enter the user's *First Name* and *Last Name*. The names are used for caller ID and the dial-by-name directory.

A suggested range	is 100-199			
First name:	Jackson			
Last name:	Hellman			
Extension number:	246			
Manufacturer:	Fortinet	~		
Model:	FON-350i	~		
MAC address:			Select	
		 		_

- 7. Assign an Extension number.
- 8. If necessary, select the Manufacturer, and then select the Model.
- 9. Enter the MAC address of the phone:
 - You can select the phone/base *MAC Address* from a list of automatically-detected phones connected to your LAN. To use this method:

i. Click the Select button. A MAC Selection window appears and lists IP phones of the selected type.



- **ii.** Select the MAC address of the IP phone associated to the extension, and then click *Select*.
- You can enter the phone/base *MAC Address* manually. This is the required method if you are setting up an external IP extension. The MAC Address is a 12-digit alphanumeric string located in the barcode on the bottom of the phone and the box the phone came in.



Click Finish.

- 10. For the 850i/860i/870i, select the Handset ID for the extension.
- **11.** In the *Extension* tab, select the language for prompts heard by the user of the extension in the *System Prompt Language* list. Setting the prompt language will also change the language for text displayed on the phone itself.
- **12.** Choose *File* > *Save*. The system will create a configuration file that the phone will download when the phone is restarted.

IP extension details area

Location:	External	~		Extension st	atus: Registered	0
MAC address:	001A7EAEC53E		Select	IP addr	ress: <u>192.168.1.102</u>	
TalkSwitch unit:	1	~		Time zone:	(-05:00) Eastern (US/Canada)	~
					Configure K	eys

- **1.** Set the *Location*:
 - Internal The phone is an internal IP extension located within the office, and is connected to the same LAN as the unit.
 - External The phone is an external IP extension located outside the office, and is connected to the unit over the internet. See the phone's user guide for further details.



Calls to an emergency service number using an external IP extension will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP phone:

If an emergency call is made from this phone, you must provide your address to the emergency operator.

2. If you set *Location* to *External*, select the *Time Zone* that matches the location of the IP phone.

About programmable function keys

Programmable functions keys allow the user to access features, and to monitor and engage lines, extensions and queued calls (i.e. line appearance).

The keys cannot be programmed if the extension has hotline access enabled. In this case, the phone will automatically connect the preconfigured resource (external number, extension etc.)

Model	Programmable function keys
FON-260i	4
FON-350i/360i	6
FON-450i/460i	10 (34)*
FON-550i/560i	22 (46)*

*with two FF-50E expansion modules attached

Programming function keys

- 1. Click Configure Keys.
- 2. For each key, select the *Function* and the *Resource* (if applicable). The function keys can be assigned for Line Appearance, Extension Appearance, Queue Appearance, Voicemail, Do Not Disturb (DND), Speed Dial, Park, Unpark, Call Pickup (any or specific extension), Group Page, Overhead Page, Phone Book configuration, Call Recording (360i/460i/560i only) or User Defined (phone). For further details, see "Phone programmable key functions" on page 60.
- **3.** Optionally, you can apply predefined key assignments from a template file using the *Open Template* button. See "Using a key assignment template" on page 59.

FortiFone 350i					×
	Option	Function		Resource	
	1:	Reserved	Ŧ		*
	2:	Reserved	Ŧ		
	3:	Phonebook	•		
	4:	Intercom	•		
	5:	Line Appearance	•	Unit 1 Line 1	•]
	6:	Line Appearance	-	Unit 1 Line 2	•
	7:	Line Appearance	•	Unit 1 Line 3	•
	8:	Line Appearance	•	Unit 1 Line 4	•
The FON-350i has 6 flexible keys that can be assigned to engage system features or monitor and engage lines, extension or queues.					
Default	en Tem	plate] Save Template	e As	Print <u>O</u> K	<u>Cancel</u>

If necessary, use the *Default* button to restore the keys to their default settings.

4. Use the *Print* button to print a label showing the key configuration. Cut out the printed label and insert it in the phone next to the key lights.

5. Your key assignments can be saved as a template for programming additional phones. See "Saving a key assignment template" on page 59.

Saving a key assignment template

You can use the template file as a starting point for the key assignments for another extension. After setting up the key assignments:

- 1. Click *Save Template As* to save the key assignments to a template file. The *Save Template As* window appears.
- 2. Enter the filename of the template file, and then click *Save*.

Using a key assignment template

- **1.** To display the key assignments from a template file, click *Open Template*. The *Open Template* window appears.
- 2. Select the template file, and then click Open Template.

Further configuration

The steps above will configure your IP phone and will enable the programmable keys. Click the *Local Extension/Fax* Help @ icons for instructions on configuring direct line access, Caller ID settings (VoIP only), call handling, and voicemail.

For phone-specific configuration, refer to your phone user guide.

Contact your reseller or Customer Support if you require further assistance with special ports or network settings.
Phone programmable key functions

Some TalkSwitch/FortiFone IP phone models have programmable keys. The function and associated resources are assigned using Management software in the *Local Extensions/Fax* > *IP Extension Details* > *Configure Keys* page. Supported functions for a phone model typically include most of the items listed below.

• *Line appearance* — Select a telephone line or VoIP number as the resource. The corresponding button or softkey will display the status of the line, and allow you to make calls with a single press of the button or softkey.

For phones equipped with button lights, the button will light up when the line is in use, flash if the line is ringing, or be off when the line is available.

For phones with softkeys, a status icon and line ID will be displayed beside the softkey. The display will show an off-hook icon when the line is in use, show a ringing icon when the line is ringing, and show an on-hook icon when the line is available.

• *Extension appearance* — Select a local extension as the resource. The corresponding button or softkey will display the status of the selected extension, and allow you to call the extension with a single press of the button or softkey.

For phones equipped with button lights, the button will light up when the selected extension is in use, flash if the extension is ringing, or be off when the extension is available.

For phones with softkeys, a status icon and extension call ID will be displayed beside the softkey. The display will show an off-hook icon when the selected extension is in use, show a ringing icon when the extension is ringing, and show an on-hook icon when the extension is available. The extension call ID is also be displayed.

• Queue appearance — Select a local extension as the resource. The corresponding button or softkey will indicate whether calls are queued at the selected extension, and allow you to pick up the oldest queued call with a single press of the button or softkey.

For phones equipped with button lights, the button will flash if calls are queued for the extension, or be off when there are no queued calls.

For phones with softkeys, a status icon, extension number and "Q" will be displayed beside the softkey. The display will show a ringing icon if calls are queued for the extension, or an on-hook icon when there are no queued calls.

- Voicemail Do not select a resource. Press the button or softkey to access the voice mailbox of the local extension. Lights or icons are not used for voicemail keys. Note: You can also access Voicemail by pressing **#.
- DND Do not select a resource. Press the button or softkey to toggle Do Not Disturb mode on or off. Lights or icons are not used for DND keys. Note: You can also toggle DND mode by pressing *62#.
- Park Do not select a resource. Press the button or softkey to put the call on hold, in the next available park orbit. The system will respond with the park orbit number (500 to 509). Lights or icons are not used for Park keys. Note: You can also Park a call by pressing
 *510# on a 350i, or *510# on a 450i or 550i.
- Un-park Do not select a resource. Press the button or softkey, select the park orbit number (500 to 509), then press Unpark to retrieve the call. Softkey displays will show the parked call ID beside the orbit number. Lights or icons are not used for Unpark keys. Note: You can also Unpark a call by lifting the handset, pressing **, dialing the park orbit number, then pressing #.
- Pickup any Do not select a resource. Press the button or softkey to answer a call from an outside number ringing any extension. Lights or icons are not used for Pickup keys. Note: You can also pick up a call by pressing *9#.

- Pickup ext Do not select a resource. Press the button or softkey, dial an extension, then dial # (or softkey again) to answer a call from an outside number or extension ringing the selected extension. Lights or icons are not used for Pickup keys. Note: You can also pick up a call at a selected extension by pressing *7, dialing the extension, then pressing #.
- Intercom Do not select a resource. Press the button or softkey, dial an extension, then
 press # to page the extension in Intercom mode. The intercom page function can only page
 proprietary phones. Lights or icons are not used for Page keys. Note: You can also page a
 proprietary phone by pressing *84, dialing the extension, then pressing #, or the *Dial* key or
 softkey.
- Overhead page Do not select a resource. Press the button or softkey to connect to an attached external PA system. Lights or icons are not used for Page keys. Note: You can also page the external PA system by pressing *0#.
- *Queue (show list)* Do not select a resource. Press the *Queue* softkey, select a queued call, then press *Retrieve* to connect to the selected call. This function is only available when programmed as a function key on supported phone models.
- *Phone book* Select the phone book record two-digit speed dial number as the resource. Press the button or softkey to place a call using the contact information from the associated phone book record. When the button is pressed, it will light up for the duration of the call. This function is available on the 350i, 450i and 550i models.
- User Defined (phone) The button or softkey function is assigned using the phone's configuration options, and is not assigned using Management software.
- None The button or softkey is not programmed.
- System speed dial Select the desired resource from speed dials configured on the System Speed Dials page. Press the button or softkey to call the resource.
- *Group page* Select Ring/Page group. Press the button or softkey to page the phones in the group. Not all phones can receive pages.
- *Call Recording (360i/460i/560i only)* Select *Call Recording* from the pulldown menu. Press the button to start call recording during a call. The light will flash and a tone will be played when call recording is started. Press the button to stop recording. Recorded calls will be saved to the extension's voice mailbox.

Adding Polycom IP Phones

Adding the extension to the system

- 1. Launch the Management software.
- 2. Select the Local Extension/Fax page.
- 3. Click the Add button. The Add User / Extension window appears.

id User / Exte	nsion		
Extension type: Unit:	IP Extension	M	

- 4. Set the Extension type to IP Extension.
- 5. In a multi-unit system, select the unit the extension will be associated with. This unit will hold the voicemail for the extension. Click *Next*.
- 6. Enter the user's *First Name* and *Last Name*. The names are used for caller ID and the dial-by-name directory.

Add User / Extens	sion 🛛 🔀
A suggested range	e is 100-199
First name:	Audrey
Last name:	Winston
Extension number:	222
Manufacturer:	Polycom
Model:	Polycom 301
MAC address:	Select
	Finish <u>C</u> ancel

- 7. Assign an Extension number.
- 8. Select the Manufacturer, and then select the Model.
- **9.** Enter the *MAC address* of the phone. The MAC Address is a 12-digit alphanumeric string located in the barcode on the bottom of the phone and the box the phone came in.

0004F2060597	
	Rev.A OSA#1015
Made in Thailand	02140798

10. Click Finish.

- **11.** In the *Extension* tab, select the language for prompts heard by the user of the extension in the *System Prompt Language* list. Setting the prompt language will also change the language for text displayed on the phone itself.
- **12.** Choose *File* > *Save*. The system will create a configuration file that the phone will download when the phone is restarted.

Click the Additional Settings button to set up:

- Direct line access (see "Direct line access" on page 48)
- Caller ID settings (see "Caller ID settings" on page 49)

IP extension details area

Ì	-IP Extension Deta	ails				
	Location:	Internal	~	Status:	Registered	
	MAC Address:	0123456789ab	Select) IP Address:	111@192.168.1.101:5060	
	Handset ID:	1	~		Configure Keys	

- 1. Set the *Location*. Choices are:
 - Internal The phone is an internal IP extension located within the office, and is connected to the same LAN as the unit.
 - *External* The phone is an external IP extension located outside the office, and is connected to the unit over the internet.



Using an external IP extension to call an emergency service number will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP extension stating:

If an emergency call is made from this phone, you must provide your address to the emergency operator.

- 2. If you are setting up an external IP extension, select the *Time Zone* of the IP phone.
- **3.** See "Setting up call handling" on page 42 and "Voicemail tab" on page 49 for instructions on configuring call handling and voicemail.
- 4. Choose *File* > *Save*. The system will create a configuration file that the phone will download when the phone is restarted.

Configuring the system for external IP extensions

If you are setting up an external IP extension:

- 1. Ensure your system is connected to a network. See "Connecting to a network" on page 176.
- Ensure that you have set up a public IP address for the system. See "Public IP address" on page 14.
- 3. Ensure the router is configured at the location. There should be no need to make any adjustments to the firewall at the remote location. See "Firewall settings" on page 14.
- All VoIP lines are shared by default. You can also reserve VoIP lines for external IP extensions. See "Line reservation" on page 32.
- External IP extensions will use the preferred codec set in the *Extensions* > *IP Extensions* window. See "Extensions > IP Extensions" on page 141. The default preferred codec is G.729.

Connecting the Polycom phone to the network

- 1. Connect a network cable between your router and the port marked LAN on the back of the phone. The PC port on the phone can be used to connect the PC if only one LAN port is available.
- **2.** Connect the power adapter to the phone.

Programming the Polycom IP phone

The system requires the IP phone to have the latest firmware version.

The system automatically configures the necessary parameters required for the proper operation of the Polycom phone. On the phone:

- 1. Press the *Menu* key and select *Settings* > *Advanced*.
- 2. Enter the password, and then press the *Enter* key. By default this is 456. When the unit configures the phone, it changes the password to 23646 (which spells "admin" on the telephone keypad).
- **3.** Select Admin Settings > Network Configuration.
- 4. Select Server Menu and change ServerType to TrivialFTP.
- 5. Select Server Address and press the Edit key. Press the 1/A/a key. If you are setting up an internal IP extension, enter the IP address of the unit acting as local proxy. If you are setting up an external IP extension, enter the public IP address or public domain name of the system. Use the * key to enter decimal points.
- 6. Press the *Exit* button three times. *Advanced* appears at the top of the menu.
- 7. Select Restart Phone and confirm by selecting Yes.

The phone connects to the local proxy and downloads the configuration file. If the configuration is successful, the phone will reboot and show the time, date and extension number. To ensure full functionality, dial another extension and check the audio transmission in both directions.

Configuring the phone to point to the firmware files

Changing the phone *Server Address* setting to the IP address of the PC with the TFTP program and the firmware files enables the update process. On the phone:

- 1. Press the *Menu* key and select *Settings* > *Advanced*.
- 2. Enter the password, and then press the *Enter* key. By default this is 456. When the unit configures the phone, it changes the password to 23646 (which spells "admin" on the telephone keypad).
- **3.** Select Admin Settings > Network Settings.
- 4. Select Server Menu and change ServerType to TrivialFTP.
- 5. Select *Server Address* and press the *Edit* key. Press the *1/A/a* key and enter the IP address of the PC running the TFTP server, using the * key to enter decimal points.
- 6. Press the Exit button three times. Advanced appears at the top of the menu.
- 7. Press 3 for *Restart Phone* and confirm by pressing the Yes button.
- 8. Once the firmware is updated, reconfigure the phone according to "Programming the Polycom IP phone" on page 64. Note that some errors will occur for files not found. These are expected and can be ignored.

For more information, refer to the "Troubleshooting" section in your phone user guide.

Adding Grandstream IP phones

Adding the extension to the system

- 1. Launch the Management software.
- 2. Select the Local Extension/Fax page.
- 3. Click the Add button. The Add User / Extension window appears.

dd User / Extension	
Extension type: IP Extension Unit: 1	
	Next > Cancel

- 4. Set the Extension type to IP Extension.
- 5. In a multi-unit system, select the unit the extension will be associated with. This unit will hold the voicemail for the extension. Click *Next*.
- 6. Enter the user's *First Name* and *Last Name*. The names are used for caller ID and the dial-by-name directory.

Add User / Extens	ion	\mathbf{X}
A suggested range	is 100-199	
First name:	Audrey	
Last name:	Winston	
Extension number:	222	
Manufacturer:	Grandstream	*
Model:	GXP-2000/2020	*
MAC address:		Select
	Finish	<u>Cancel</u>

- 7. Assign an Extension number.
- 8. Select the Manufacturer and the Model.
- **9.** Enter the *MAC* address of the phone. The MAC Address is a 12-digit alphanumeric string located in the barcode on the bottom of the phone and the box the phone came in.



Click Finish.

- **10.** In the *Extension* tab, select the language for prompts heard by the user of the extension in the *System Prompt Language* list. Setting the prompt language will also change the language for text displayed on the phone itself.
- **11.** Choose *File* > *Save*. The system will create a configuration file that the phone will download when the phone is restarted.

Click the Additional Settings button to set up:

- Direct line access (see "Direct line access" on page 48)
- Caller ID settings (see "Caller ID settings" on page 49)

IP extension details area

-IP Extension	n Detalls				0
Location:	Internal	~	Status:	Registered	
MAC Addr	ess: 0123456789ab	Selec	t IP Address:	111@192.168.1.101:5060	
Handset ID	u 1	¥		Configure Keys	

- 1. Set the *Location*. Choices are:
 - Internal The phone is an internal IP extension located within the office, and is connected to the same LAN as the unit.
 - *External* The phone is an external IP extension located outside the office, and is connected to the unit over the internet.



Using an external IP extension to call an emergency service number will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP extension stating:

If an emergency call is made from this phone, you must provide your address to the emergency operator.

- 2. If you are setting up an external IP extension, select the *Time Zone* of the IP phone.
- **3.** See "Setting up call handling" on page 42 and "Voicemail tab" on page 49 for instructions on configuring call handling and voicemail.
- 4. Choose *File* > *Save*. The system will create a configuration file that the phone will download when the phone is restarted.

Configuring the system for external IP extensions

If you are setting up an external IP extension:

- 1. Ensure your system is connected to a network. See "Connecting to a network" on page 176.
- Ensure that you have set up a public IP address for the system. See "Public IP address" on page 14.
- **3.** Ensure the router is configured at the system's location. There should be no need to make any adjustments to the firewall at the remote location. See "Firewall settings" on page 14.
- 4. All VoIP lines are shared by default. You can also reserve VoIP lines for external IP extensions. See "Line reservation" on page 32.
- External IP extensions will use the preferred codec set in the *Extensions* > *IP Extensions* window. See "Extensions > IP Extensions" on page 141. The default preferred codec is G.729.

Connecting the Grandstream phone to the network

- 1. Connect a network cable between your router and the port marked LAN on the back of the phone. The PC port on the phone can be used to connect the PC if only one LAN port is available.
- 2. Connect the power adapter to the phone.
- 3. Once the phone boots up, it will attempt to obtain an IP address from a router DHCP server.

Programming a Grandstream IP phone

- 1. Note the IP address that appears on the screen of the Grandstream phone.
- 2. In a web browser, enter the IP address in the *Address* field.
- **3.** Enter the password. By default this is *admin*. When the unit configures the phone, it changes the password to 23646 (which spells "admin" on the telephone keypad). The *Grandstream Device Configuration* page appears.

Grandstream Device Configuration
<u>STATUS</u> <u>BASIC</u> <u>ADVANCED ACCOUNT ACCOUNT ACCOUNT ACCOUNT</u>

4. On the Grandstream Device Configuration page, click Advanced.

Firmware Upgrade and Provisioning:	Upgrade Via	IFTP	○ HTTP	
	Firmware	Server Path	192.168.1.101	
	Config Se	rver Path 192	2.168.1.101	

- 5. In the section for Firmware Upgrade and Provisioning, set Upgrade Via to TFTP.
- 6. If you are setting up an internal IP extension, set *Config Server Path* to the IP address of the unit acting as local proxy. If you are setting up an external IP extension, set *Config Server Path* to the public IP address or public domain name of the system.
- 7. Scroll to the bottom of the page and click the Update button to save the settings.

Update Cancel Reboot

8. On the following screen, click the *Reboot* button to apply the settings.

```
Your configuration changes have been saved.
They will take effect on next reboot.
Reboot
```

The reboot process will be completed when you see the extension number and IP address on the display of the phone. When the phone successfully registers, you will see the filled-in Ethernet icon.



9. Test to ensure the phone is configured properly by dialing another local extension.

Adding Counterpath IP phones

Adding the extension to the system

- 1. Launch the Management software.
- 2. Select the Local Extension/Fax page.
- 3. Click the Add button. The Add User / Extension window appears.

~

- 4. Set the Extension type to IP Extension.
- 5. In a multi-unit system, select the unit the extension will be associated with. This unit will hold the voicemail for the extension. Click *Next*.
- 6. Enter the user's *First Name* and *Last Name*. The names are used for caller ID and the dial-by-name directory.

Add User / Extens	sion 🛛
A suggested range	e is 100-199
First name:	Audrey
Last name:	Winston
Extension number:	222
Manufacturer:	Counterpath 🗸
Model:	Counterpath Eyebeam
MAC address:	Select
	Finish Cancel

- 7. Assign an Extension number.
- 8. Select the Manufacturer and Model. Click Finish.
- **9.** In the *Extension* tab, select the language for prompts heard by the user of the extension in the *System Prompt Language* list. Setting the prompt language will also change the language for text displayed on the phone itself.
- **10.** Choose *File* > *Save*. The system will create a configuration file that the phone will download when the phone is restarted.

Click the Additional Settings button to set up:

- Direct line access (see "Direct line access" on page 48)
- Caller ID settings (see "Caller ID settings" on page 49)

IP extension details area

[IP Extension De	etails		0
Location:	Both 🗸	Status: Registered	
		IP Address: <u>111@192.168.1.101:5060</u>	
User Name:	user111	Password pass111	
		Time Zone: (-05:00) Eastern (US/Canada) 🗸	
Handset ID:	1	Configure Keys	

- 1. Set the *Location*. Choices are:
 - Internal The phone is an internal IP extension located within the office, and is connected to the same LAN as the unit.
 - *External* The phone is an external IP extension located outside the office, and is connected to the unit over the internet. A VoIP-enabled unit is required.
 - *Both* The phone can be used as an internal or external IP extension.



Using an external IP extension to call an emergency service number will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP extension stating:

If an emergency call is made from this phone, you must provide your address to the emergency operator.

- 2. If the phone can be used as an external IP extension, select the *Time Zone* of the IP phone.
- **3.** Enter the *Username* and *Password*. The default settings are user[extension number] and pass[extension number].
- **4.** See "Setting up call handling" on page 42 and "Voicemail tab" on page 49 for instructions on configuring call handling and voicemail.
- 5. Choose *File* > *Save*. The system will create a configuration file that the phone will download when the phone is restarted.

Configuring the system for external IP extensions

If you are setting up an external IP extension:

- 1. Ensure your system is connected to a network. See "Connecting to a network" on page 176.
- Ensure that you have set up a public IP address for the system. See "Public IP address" on page 14.
- **3.** Ensure the router is configured at the system's location. There should be no need to make any adjustments to the firewall at the remote location. See "Firewall settings" on page 14.
- 4. All VoIP lines are shared by default. You can also reserve VoIP lines for external IP extensions. See "Line reservation" on page 32.
- External IP extensions will use the preferred codec set in the *Extensions* > *IP Extensions* window. See "Extensions > IP Extensions" on page 141. The default preferred codec is G.729.

Configuring a new SIP account

The following procedure describes setting up the eyeBeam software to register on the network and to make itself available as a client for sending and receiving calls.

1. Install the eyeBeam software. The SIP Accounts window appears.

Enabled	Domain	Username	Display Name	Add
				Remove
				Properties
				Make Defaul
-				

2. Click the Add button. The Properties of Account1 window appears.

count	Voicemail	Topology	Presence	Storage	Security	Advanced
User [Details					
Displa	y Name	3	Eye Beam			
User n	ame	l.	115			
Passw	ord		•••••			
Autho	rization use	r name	user115			
Domain			192.168.1.102			
Domai Domai VRe	n in Proxy gister with	domain and r	192.168.1. eceive incon	102 ning calls		
Domai Doma VRe Send (O O	n gister with outbound vi domain proxy target dom	domain and r a: Address ain	192.168.1.	102 ning calls		

- 3. Enter the User Details in the Account tab.
 - a. Set the *Display Name* to the name for Caller ID.
 - **b.** Set the *User name* to the local extension number.
 - c. Set the Password to the Password set up in step 3 of "IP extension details area" on page 69.
 - **d.** Set the Authorization user name to the Username set up in step 3 of "IP extension details area" on page 69.

e. If setting up an internal IP extension, set *Domain* to the IP address of the unit acting as local proxy. If setting up an external IP extension, set *Domain* to the public IP address or public domain name of the system.

Eye Beam
115
•••••
user115
192.168.1.102

- 4. Set up the *Domain Proxy* area in the *Account* tab.
 - a. Select the Register with domain and receive incoming calls checkbox.
 - **b.** Select the *target domain* option.

Domain Proxy		
Register wit	h domain and	receive incoming calls
Send outbound	via:	
🔿 domain		
Oproxy	Address	
⊙target do	omain	

- 5. Set up the Voicemail tab.
 - a. Select Check for voice mail.
 - b. Enter ** in the Number to dial for checking voicemail field.
 - c. Enter *<voicemail number> in the Number for sending calls to voicemail field.
- 6. Click OK to save settings and close the Properties of Account1 window.
- 7. Click *Close* to close the *SIP Accounts* window and enable the newly configured SIP account. Once closed, the phone display will show:
 - Discovering network...
 - Registering...
 - Ready, Your username is: [local extension number].

Updating eyeBeam software

The latest release of eyeBeam can be found at www.counterpath.net/eyebeam.html.

Adding Other IP Phones

Other IP phones with the G.711 codec may work with the system but not all features may be supported. We strongly recommend you use only the supported IP phones.

If you connect an unsupported IP phone, select *Other* as the *Manufacturer*. As you cannot enable features, or customizations of these phones, further configuration will be limited to the programmable options on the IP phone itself.

Extension Voicemail	1			
Extension 225				0
First name:	Jason	Last name:	McKenna	
Extension number:	225	Extension type:	IP Extension	*
Manufacturer:	Other 🗸	Model:	Other	×
System prompt language:	Default 💌		Additional Settings	

Refer to the specific IP phone user guide's details for information on configuration.

External IP extensions

An external IP extension is an IP phone located outside the office that is configured as a local extension.

To enable external IP extensions, the feature must be supported on your model of phone system. Your system and the external location must both have a high-speed Internet connection.



Calls to an emergency service number using an external IP extension will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP extension stating:

If an emergency call is made from this phone, you must provide your address to the emergency operator

Configuring the router

If you are setting up an external IP extension, you must ensure the router at the phone system's location is configured.

The *Firewall Settings* area of the *IP Configuration* page displays the type of gateway device (i.e. the type of router), the IP address of the gateway (i.e. router), and whether router configuration is required.

A router is a gateway between the local area network and the Internet. Most routers have a firewall to block unwanted data from the Internet. For voice data to reach the system through the firewall, port forwarding is required. Port forwarding allows the router to map ports to the IP addresses of the units. Valid Internet data will use the ports to go through the firewall to the units.

If you are setting up external IP extensions, or a VoIP service that doesn't handle port forwarding, port forwarding is required.

If port forwarding is required, and your router supports uPNP (Universal Plug and Play), ensure uPNP is enabled. The system will use uPNP to automatically set up port forwarding, and the *Automatic (uPNP Enabled)* link will appear. No router configuration is required.

If port forwarding is required but your router doesn't support uPNP, or automatic port forwarding doesn't work, the *Manual port mapping required* link will appear. You will need to configure the router as described below.

- 1. Select the IP Configuration page.
- 2. If required, click the *Manual port mapping required* link. The *Manual Port Mapping* window appears. It lists the packet type, labels to aid in port identification, port number, IP address and protocol of each required port.

The system requires the following firew proper VoIP communication	all ports be mapped	to the following IP	addresses f	or
Description	Port	System IP	Protocol	
SIP (VoIP signalling)	5060	192.168.1.103	UDP	-
RTP Audio (Unit 1, Port 1)	6100	192.168.1.103	UDP	1
RTP Audio (Unit 1, Port 2)	6102	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 3)	6104	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 4)	6106	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 5)	6108	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 6)	6110	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 7)	6112	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 8)	6114	192.168.1.103	UDP	-L
TFTP (IP Phone Configuration)	69	192.168.1.103	UDP	-

- **3.** To access the router configuration:
 - **a.** Click the link containing the IP address of the gateway. The default browser starts, and prompts you for the router's user name and password.
 - **b.** Enter the router's user name and password. The browser shows a setup screen.
 - c. Navigate to the screen used to set up port forwarding. See your router documentation.
 - **d.** Set up port forwarding using the information from the *Manual Port Mapping* window. See your router documentation for instructions on how to map ports. For information on configuring routers and mapping ports, visit www.portforward.com/english/routers/port_forwarding/routerindex.htm.
- **4.** To check the status of each port through the firewall, click *Check Firewall*. The *Firewall Test* window appears.

Callings						
rewall Settings						
Service	Status	Servi	ce	Status		
✓ Internet Access		V RT	P Audio (Unit 1, Port 6)			
HTTP (XML Service)		🗹 RT	P Audio (Unit 1, Port 7)			
RTP Audio (Unit 1, Port 1)		🗹 RT	'P Audio (Unit 1, Port 8)			
RTP Audio (Unit 1, Port 2)		SIF	(VolP signalling)			
RTP Audio (Unit 1, Port 3)		TF	TP (IP Phone Configuration)			
RTP Audio (Unit 1, Port 4)						
RTP Audio (Unit 1, Port 5)						
					Details >>	Test Ports

- 5. Select the services you want to check.
- 6. Click *Test Ports*. The system will check the ports for the selected services.

Configuring the router manually

If you cannot access the router configuration through the *IP Configuration* page, configure the router manually.

- **1.** Open the router configuration and navigate to the screen used to set up port forwarding. See your router documentation.
- 2. In the Management Software on the *IP Configuration* page, click the *Manual Port Mapping Required* link.
- **3.** Map the port indicated for SIP (VoIP) signaling.

If required, you can map a different port. Select *Troubleshooting* > *VoIP* and enter the port in the *SIP signalling port* field.

Map the rest of the ports to the IP addresses indicated in the Manual Port Mapping window.

he system requires the following firewa proper VoIP communication	all ports be mapped	to the following IP	addresses to	or
Description	Port	System IP	Protocol	
SIP (VoIP signalling)	5060	192.168.1.103	UDP	-
RTP Audio (Unit 1, Port 1)	6100	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 2)	6102	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 3)	6104	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 4)	6106	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 5)	6108	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 6)	6110	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 7)	6112	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 8)	6114	192.168.1.103	UDP	
TFTP (IP Phone Configuration)	69	192.168.1.103	UDP	

If required, you can map different ports. Select *Troubleshooting > VoIP*.

- **4.** Map ports 9393, 8485 and 8486 (Type: TCP) to the unit acting as local proxy to allow remote configuration of the system.
- 5. If available, enable Quality of Service (QoS) to give voice traffic priority over data.
- 6. Save the configuration to the router.

Remote Extensions

A remote extension reaches an external phone by automatically selecting a line from a hunt group and dialing the phone number. For example, a remote extension could reach an employee's cell phone or home phone, or a phone at a branch office.

A caller can connect to a remote extension through the auto attendant, or can be transferred to a remote extension by a call cascade. A user at a local extension can manually transfer a caller to a remote extension, or can dial a remote extension directly. If the remote extension is busy or unanswered, the system can route the call using the remote extension's call cascade.

For example, a caller reaches the auto attendant and dials a local extension. The user is not there, so the call is unanswered. The call cascade of the local extension can be configured to transfer unanswered calls to a remote extension. The remote extension can be configured to dial the user's cellular phone. This way the user is available outside the office.

There are three ways the system can transfer calls to a remote extension:

- If the telephone line has the Transfer and Clear service available and activated, the system can use the transfer and clear feature. When an outside caller is being routed to a remote extension, the system directs the telephone company to put the caller on hold. It then uses the same line to reach the remote extension. The telephone company connects the outside caller to the remote extension, and then frees the line.
- If the telephone line has the 3-Way Calling/Conference service available and activated, the system can use the same line connect feature. When an outside caller is being routed to a remote extension, the system directs the telephone company to put the caller on hold. It then uses the same line to reach the remote extension. The telephone company connects the outside caller to the remote extension, but the line to the unit remains occupied until the call is complete.
- If the telephone line doesn't have these services, and an outside caller is being routed to a remote extension, the system puts the caller on hold. It then uses a second line to reach the remote extension. Both lines remain occupied until the call is complete.

If two lines are used to connect an outside caller to a remote extension, the user at the remote extension can:

- Place the call on hold by dialing **.
- Retrieve the call on hold by dialing **.
- Transfer the call by dialing ** + the extension number. The call can be transferred to a local extension, remote extension, ring group, or another location in a multi-branch network.
- Transfer the call to a voice mailbox by dialing *** + the voice mailbox number.

The remote extensions can be configured to perform a blind transfer or a screened transfer. However a transfer to another remote extension is always a blind transfer to avoid tying up a third line.

If the system is configured to perform a blind transfer, it plays "*Call transferred. Goodbye.*" and then hangs up. The call is transferred to the dialed extension and follows its call cascade if busy or unanswered.

If the system is configured to perform a screened transfer, the user is connected to the other extension and asks whether they want the call. If so, the user dials ** + 4 to complete the transfer. If the other person doesn't want the call, the user dials ** + 5 to cancel the transfer and return to the caller.

The *Remote Extensions* page allows you to set up a remote extension.



Remote extensions are designed to operate with local major telephone service providers. The feature may not function correctly with some telephone and mobile operator's networks, especially for international phone numbers and mobile phones roaming internationally.

Remote Extension tab

1. Select the Remote Extensions page.

Add	Remove					
First name	Last name	Remote Extension	Voicemail			
Valerie	Cho	Extension 137				
Peter	Altman	First name:	Valerie	Last name	Cho	
		Extension number:	137			
		Remote phone number:	6152371067	Connect using	Hunt Group 9	
					Use same line connect	
	_			System prompt language	Default 🖌	
		Call Handling				<u></u>
		Mode 1 (Business I	Hours) Mode 2 (Afte	r Hours)		
		Busy No Ans	swer Answered			
		If extension is hus	v.	hang up	-	
		If buoy or not anou	vorod after: // ringe	hang up		
		If busy of not answ	vereu alter. 4 V nings	nang up	Y	X

2. Click the Add button. The Add Remote Extension window appears.

Add Remote Extensio	n 🔀
A suggested range is 2	00-263
First name:	
Last name:	
Extension number:	
Remote phone number:	
Connect using:	Hunt Group 9
	Use same line connect
	<u>O</u> K <u>C</u> ancel

- **3.** Enter the user's *First Name* and *Last Name*. The names are used for caller ID and the dial-by-name directory.
- 4. Assign an Extension number.
- 5. Enter the *Remote Phone Number*. Enter the number as you would normally dial it (i.e. without the hunt group number). You can enter digits 0–9, space, dash, comma, # and *. A comma pauses dialing for two seconds.
- 6. Select the User Privilege. See "User Privileges" on page 34.
- 7. Select the hunt group in the *Connect Using* list. The unit will use a line from this hunt group to connect with the remote extension. We recommend the default *Hunt Group* 9 (or *Hunt*

Group 0 in certain regions) unless you have set up a different hunt group for calling remote extensions.

8. If you have the 3-Way Calling/Conference service from the telephone company, you can select the *Use same line connect* checkbox. The same line connect feature will direct the telephone company to put the caller on hold, and will then use the same line to try the remote extension. Because the same line is used, the hunt group setting from Step 7 is ignored.

If you do not select the *Use same line connect* checkbox, the unit will put the caller on hold, and then use a second line to try the remote extension. The second line will be from the hunt group selected in Step 7.

If you enable the same line connect feature, the user at the remote extension will not be able to transfer the call to another extension.

When the same line connect feature is in use, the caller hears silence while they are on hold. This is because the caller is on hold at the telephone company and not at the unit. If you want the caller to hear music on hold while their call is being forwarded to the remote extension, do not use the same line connect feature.

The same line connect feature is not compatible with some telephone company lines.

Click OK.

- **9.** In the *Remote Extension* tab, select the language for prompts heard by the user of the extension in the *System prompt language* list.
- **10.** Set the type of transfer performed by users at remote extensions. This affects transfers to local extensions and ring groups. Transfers to other remote extensions are always blind transfers. See "To remote extension tab" on page 133.
 - a. Choose Options > Transfer Preferences. The Transfer Preferences window appears.

Transfer Preferences	×
Specify the options for transferring a caller:	
To Local Extension To Remote Extension To Ring Group From Auto Attendant	
If the Extension is Busy-	-0
O Ring back the extension that transferred the call	
⊙ Perform the Busy action	
Lif the extension does not answer-	-0
O Ring back the extension that transferred the call	
O Perform the No Answer action	
Use Defaults QK	Cancel

- **b.** Select the To Remote Extension tab.
 - To allow screened transfers, select *Allow screening of calls*.
 - To allow blind transfers, select Perform a blind transfer. This is the default selection.
- **11.** If you have the Transfer and Clear service from the telephone company, you can enable the transfer and clear feature. See "To remote extension tab" on page 133.
 - a. Choose Options > Transfer Preferences. The Transfer Preferences window appears.
 - **b.** Select the *To Remote Extension* tab.
 - **c.** To enable the transfer and clear feature, select the *Clear telephone line after call transferred* checkbox.

About call cascades

See "About call cascades" on page 42.

Setting up call handling

The procedure for setting up a call cascade is similar to that described in *Local Extension* with the differences described below. See "Setting up call handling" on page 42.

- The Remote Extensions page does not have the Do Not Disturb tab.
- The *If Extension is busy* list in the *Busy* tab does not have the *invoke call waiting* or *queue at extension* options.
- The When a call is answered list in the Answered tab has the play accept/reject prompt option, in addition to the stay connected and play caller's name first options.

If *play accept/reject prompt* is selected, the system will place the caller on hold and will then call the remote extension. When the user answers, the system will say: *"You have a forwarded call. To accept the call, press #. To reject the call, press *."* If the user accepts the call, the system will route the call to the remote extension. If the user rejects the call, the system will try the next alternative in the remote extension's call cascade.



When setting the number of rings to wait for a remote extension to connect, it is very important to allow enough time for the telephone network to connect to the phone. For example, a call to a cell phone or PDA may take 12 seconds (i.e. 2 rings) just to connect.

In the following circumstances, the system will start by saying "*This is call cascade*", instead of "*This is call forward*" or "You have a forwarded call".

- An extension's call cascade uses play caller's name first or play accept/reject prompt, and
- The previous user answered but rejected the call, and
- The extension's call cascade is now attempting to transfer the call to a remote extension.

This notifies the user that the call was rejected. If the user also rejects the call, it will be routed according to the other extension's call cascade.

Voicemail tab

The procedure for setting up voicemail is identical to that described in *Local Extension/Fax*. See "Voicemail tab" on page 49.

Mailbox greeting

The procedure for setting up the mailbox greeting is identical to that described in *Local Extension/Fax*. See "Mailbox greeting" on page 50.

Caller options

The procedure for setting up the caller options is identical to that described in *Local Extension/Fax*. See "Caller options" on page 51.

Notification settings

The procedure for setting up the notification settings is identical to that described in *Local Extension/Fax*. See "Notification options" on page 51.

Ring and Page Groups

A ring group is a group of local extensions that ring in unison. Local extensions and auto attendants can dial a ring group. A page group is a group of extensions that can be paged in unison. Page groups require telephones that support group paging.

There are 10 ring/page groups available in the system.

Ring/page groups have two main uses:

- A ring/page group can reach a group of employees. For example, ring/page group 301 can ring the sales group at extensions 111, 112, 113, and 114. When a customer calls the sales group, the first available salesperson answers for the group.
- A ring/page group with a different ring pattern can differentiate callers. For example, the president doesn't want to answer calls from the general public, but wants to be alerted when important colleagues are on the line. His local extension is 111, but you can configure ring/page group 305 to ring his phone with a different ring pattern. The president tells his colleagues to reach him at extension 305, which is his private extension. He can tell by the ring pattern if the caller dialed extension 111 or 305, and can react accordingly.
- 1. Select the *Ring/Page Groups* page.

Group Name Group name: Sales Department Group number: 100 Extensions to ring / page: 223 - Jackie Harrison 222 - Andy Parker	
Sales Department Group name: Sales Department Group number: 100 Ring pattern: Atternate 1 Extensions to ring / page: 223 - Jackie Harrison 222 - Andy Parker	
Group number: 100 Ring pattern: Alternate 1 💌 Extensions to ring / page: 223 - Jackie Harrison 222 - Andy Parker	
Extensions to ring / page: 223 - Jackie Harrison 222 - Andy Parker	
Edit	
Mode 1 (Business Hours) Mode 2 (After Hours)	
Busy No Answer Answered	
IT extension is busy: play busy tone	~
If busy or not answered after: 4 v rinos hand up	~
	1.4
If busy or not answered after: 4 🗸 rings hang up	

2. Click the Add button. The Add Ring/Page Group window appears.

A suggested range is 3	00-309
Group name:	
Group number:	
Ring pattern:	Regular

- 3. Enter the Group name.
- **4.** Assign a *Group number*.
- 5. Select a *Ring pattern* to indicate the call is for the ring group. Click *Next*.
- 6. Use the checkboxes to select the local extensions you want to ring in unison. Click Finish.

Add Ring / Page Group
Extensions to ring / page:
134 - Jason Staples
245 - Trish Kowslowski
255 - James Cordoba
256 - Mila Pelligrino
257 - Fax Machine
Select All Deselect All Finish Cancel

About call cascades

See "About call cascades" on page 42.

Setting up call handling

The procedure for setting up a call cascade is similar to that described in *Local Extension/Fax* with the differences described below. See "Setting up call handling" on page 42.

- The Ring/Page Groups page does not have the Do Not Disturb tab.
- The *If Extension is busy* list in the *Busy* tab does not have the *invoke call waiting* option.
- The *queue at ring group* option replaces *queue call*, and places the call in the ring group's call queue.

Caller ID options

The *Caller ID Options* area allows you to set up the system so users will know the caller is trying to contact the ring group, and not them personally.

Two options can be selected or combined:

- The ring group number can be displayed in front of the caller ID information. For example, if John Smith is calling ring group 300, each extension in ring group 300 will display "300 -John Smith".
- **2.** The ring group name can be displayed in place of the caller ID information. For example, if John Smith is calling ring group 300, which has the name "Sales", each extension in ring group 300 will display "Sales".

General Voice Mailboxes

General voicemail is not associated with any extension, but is for general use or for a group.

There are 10 general voice mailboxes per unit.

The General Voice Mailboxes page allows you to set up general voicemail.

1. Select the General Voice Mailboxes page.

Add Remove	voice mailboxes per unit. General mailboxes are useful for linking to ring groups or using as virtual mailboxes.	
Label		
3 Sales inquiries	Mailbox name: General inquiries	
General inquiries	Number: 334	
	Mailbox Greeting	
	Record a greeting using a phone or load a wav file. Load Wav File	
	Caller Options	6
	Action to perform when caller presses 0 while in voicemail	6
	Mode 2: perform no action	
	Notification Options	
	Email Notification	
	Enable amail potification	[
	Email Addresses:	
	Edit	
	Let a la construction de la constru	

Note: In the Caller Options area, the screen may refer to when a caller presses 9 depending on the region.

2. Click the Add button. The Add General Mailbox window appears.

dd General Mailbox				2
A suggested range is 450 -	499			
Mailbox name:				
Number:				
	_			
		Next	<u>Cancel</u>	

- 3. Enter a Mailbox name.
- 4. Assign a Number. Click Next.

5. Use the checkboxes to select the local extensions you want to receive notification of new voicemail in this box. Click *Finish*.

Extensions to Notify:	
134 - Jason Staples	
245 - Trish Kowslowski	
255 - James Cordoba	
256 - Mila Pelligrino	
257 - Fax Machine	
Select All Deselect All Finish Cancel	

Mailbox greeting

The procedure for setting up the mailbox greeting is identical to that described in *Local Extension/Fax*. See "Mailbox greeting" on page 50.

Caller options

The procedure for setting up the caller options is identical to that described in *Local Extension/Fax*. See "Caller options" on page 51.

Notification options

The procedure for setting up the notification settings is identical to that described in *Local Extension/Fax*. See "Notification options" on page 51.

Voicemail Preferences

The *Voicemail Preferences* page allows you to set up the voice mailbox settings. Any changes entered on this window affect all mailboxes. You can also view mailbox data, reset mailboxes, delete the password for a mailbox and disable the password requirement for internal mailbox access.

1. Select the Voicemail Preferences page.

Voicemail Preferences
r Mailbox Setting-
When listening to voicemail messages, always start with: Newest message
Notify the pager/remote phone after receiving: Only 1st new message
Maximum recording length for each message: 2 minutes
Blank Message Setting-
Automatically remove blank and 'hang up' messages
Only remove messages shorter than: 0 ms
CMailbox Access
If the system is used in a trusted environment, then access to voice mailboxes can be simplified by eliminating the password request when accessing voicemail from any local extension. Passwords will still be required when checking voicemail remotely.
Disable password request
r Voicemail Management
Vajaamai Mamany State Reset Mailhaves Delete a Mailhav Baseward
Volceman memory stats reset manuoxes Delete a manuox Password

Mailbox setting

The Mailbox Setting area allows you to set up general mailbox settings.

- 1. Set the *When listening to voicemail messages, always start with* list to which message should be played first. Choices are *Newest message* or *Oldest message*.
- 2. Set the *Notify the pager/remote phone after receiving* list to how often the system should send notification. Choices are *Every new message* or *Only 1st new message*.
- **3.** Set the *Maximum recording length for each message* list to how long a message can be. Choices range from 1 *minute* to 8 *minutes*.
- 4. Select the Automatically remove blank and 'hang up' messages checkbox, if you want the system to delete blank messages. A blank message occurs when the caller hangs up upon reaching your voicemail. If you have the caller ID service from your telephone company and choose to keep blank messages, you will be able to check the caller's number when they leave a blank message.
- 5. If you selected the *Automatically remove blank and 'hang up' messages* checkbox, set the *Only remove messages shorter than* list. The system will automatically remove messages that are shorter than the selected length. Choices range from 1 second to 5 seconds.

Mailbox access

Local extensions can access voicemail without a password if you check *Disable password request* here. Passwords will still be required when checking voicemail remotely.

Voicemail management

The *Voicemail Management* area has buttons that allow you to view "Mailbox status", "Reset mailboxes", and "Delete password" for a mailbox.

Mailbox status

The *Voicemail Memory Usage* window shows how much recording time is taken up by voicemail greetings and messages.

The recording time for internal music on hold, voicemail, and the auto attendants is shared on the unit.

Note that the system allows up to 99 voicemail messages per mailbox. Once a mailbox fills up, callers won't be able to leave voicemail messages for that user. Therefore users should delete voicemail messages before the mailbox fills up.

1. Click the Voicemail Memory Stats button. The Voicemail Memory Usage window appears.

Alternatively, you can choose Tools > Voicemail Manager > Voicemail Memory Stats.

failbox 🔺	Mailbox Name	Туре	Greeting Time	New Messages	Total Messages	Message Time
23	Fax Machine	Voicemail	3 sec	0	0	0 sec
37	kdkd dkdk	Voicemail	0 sec	0	0	0 sec
D1	Peter Parker	Voicemail	0 sec	0	0	0 sec
22	Andy Parker	Voicemail	0 sec	0	0	0 sec
23	Jackie Harrison	Voicemail	0 sec	0	0	0 sec
24	Karen Allidina	Voicemail	0 sec	0	0	0 sec
33	whappo	Voicemail	0 sec	0	0	0 sec
34	whammo	Voicemail	0 sec	0	0	0 sec
/A	Unassigned	Deactivated	9 sec	0	0	0 sec

Unassigned messages

Due to deactivation of mailboxes or errors, the system may display messages currently not assigned to a mailbox. Clicking on the *Unassigned* link will allow you to either download these files to your PC as .wav files, or delete them. The files will be stored in a folder in the Management software program directory.

Reset mailboxes

When a user leaves an organization, you can reset the mailbox to return it to its default state for the next user. Resetting mailboxes deletes greetings, messages and passwords from the selected local extension, remote extension and general mailboxes. You can optionally delete the names recorded for the dial-by-name directory from these mailboxes as well. If you need to listen to the voicemail messages of an extension before resetting the mailbox, you can download them from the *Voicemail Memory Usage* window (see "Mailbox status" on page 85), or disable the password to listen to them from an extension (see "Delete password" on page 86).

1. Click Reset Mailboxes. The Reset Mailboxes window appears.

Alternatively, you can choose *Tools > Voicemail Manager > Reset Mailboxes*.

Reset Mailboxes		×
WARNING! Reset	ting a mailbox will delete the ages and remove the passv	vord.
1111 112 113		
114 115		
117 118 121		~
Delete name from dia	-by-name directory	
	<u>Q</u> K <u>C</u> an	cel

- 2. Select one or more mailboxes to delete their greetings, messages and passwords.
- **3.** Select the *Delete name from dial-by-name directory* checkbox to delete the names recorded for the dial-by-name directory from these mailboxes.
- **4.** Click *OK*. The greetings, messages and passwords (and optionally names for the dial-by-name directory) are deleted from the selected mailboxes.

Delete password

If a user has left your organization or is temporarily unavailable, you might need to listen to voicemails on a password-protected voice mailbox. You can delete the password from the selected local extension, remote extension or general mailbox. Once the password is deleted, voicemails can be listened to without a password. Note that anyone can listen to them, from a local extension or by calling in to the system, if they know the mailbox number.

The password can be reset by following the normal procedure. See "Retrieving messages and accessing a voice mailbox" on page 165.

You can also download messages from a mailbox without disrupting the password protection. See "Mailbox status" on page 85.

1. Click Delete Password. The Delete Password window appears. Alternatively, you can choose Tools > Voicemail Manager > Delete Mailbox Password.

Delete Password	X
Delete password for mailbox:	111 💌
	QK Cancel

2. Select the mailbox to delete its password.

Voicemail Broadcast

Voicemail broadcast distributes a recorded message to a group of voice mailboxes. You can set up to ten voicemail broadcast groups system-wide.

Add Broadcast Group

- 1. Open the Voicemail Broadcast page.
- 2. Click Add.
- **3.** Enter a name for the group.
- 4. Assign a number for the group. Click Next.
- 5. Check the boxes next to the extensions you want to include in the group.
- 6. Click Finish.

You can set a password for the broadcast group by dialing ** + the group number from a local extension and following the prompts. The extensions in the group can also be added and deleted in the same way.

System Speed Dials

System speed dials allow the user to quickly dial an outside phone number from any local extension by dialing a speed dial number. Because the speed dial numbers are maintained within the system, they don't need to be programmed into each individual phone. If a new client is added or an existing client's phone number changes, you can add or modify the speed dial number, and all users will have immediate access to the new phone number.

You assign system speed dials a 3, 4 or 5-digit number, based on the numbering plan you have specified for your system. See "System numbering plan" on page 7 for more information.

The System Speed Dials page allows you to define up to 100 speed dial numbers. Each speed dial number includes the first name, last name, phone number and hunt group. You can create a .csv file that contains speed dial numbers, and then import them into the system. You can also export the speed dial numbers to a .csv file. Caller ID name tagging can be enabled, which will substitute the caller ID name with the system speed dial name.

Activate speed dial

1. Select the System Speed Dials page.

Syste	m Speed	Dials						14
Up to '	100 system-w	vide speed dial nu	imbers can be progra	immed to dial any phone	e number.			
	Add	Remove]					
		-11-	, ⊓ ⊢Speed Dial 101					
SD	First name	Last name						
101	Dave	Jefferson	First name:	Dave	Last name:	Jefferson		
104	Joe	Bowie						
105	Carly	Saby	Speed dial:	101				
106	Ling	Dewaparna						
111	Kirk	Desmond	Phone number:	1232344444	Connect using:	Hunt Group 9	~	
144	Karen	Hao		<u>بال</u>	78			
			Select File					
			Export System S	peed Dial List				@
			The speed dial I shared with use Export	ist can be exported to a prs of the system.	csv file for easy importing in	ito applications like E:	xcel. The list can then	be printed or

You can define up to 100 speed dial numbers. Each speed dial number includes the first name, last name, phone number and hunt group.

2. Click the Add button. The Add Speed Dial window appears.

Add Speed Dial	ı X
A suggested ra	nge is *300-*399
First name:	
Last name:	
Speed dial:	
Phone number:	
Connect using:	Hunt Group 9
	<u>O</u> K <u>Cancel</u>

3. Enter a *First name* and *Last name*.

- 4. Assign a Speed dial number.
- Enter the *Phone number*. Valid characters are 0–9, -, , (pauses dialing for two seconds), w (pauses dialing until a dial tone is detected), * and #. The phone number can be up to 19 characters long.

The phone number can be a regular phone number or a VoIP number from a service provider.

If the phone number is a regular phone number, it can include a pause followed by an extension number. For example, the phone number is 555-1212 and the desired extension is 123. The phone number can be set to 555-1212,123.



The phone number cannot include an extension number if it is a VoIP number from a service provider.

If auto route selection blocks the phone number, the phone number will not be blocked if a user dials the speed dial number.

6. Select the hunt group. Choices are the active hunt groups.

If auto route selection or permissions block the hunt group, the hunt group will not be blocked if a user dials the speed dial number.

If all lines in the hunt group are busy, but an overflow hunt group is set up, the system will try the overflow hunt group.

Click Finish.

Import system speed dial list

You can import a speed dial list created with a text editor like Microsoft Notepad or a spreadsheet program like Microsoft Excel. Each entry in the file requires the speed dial number, first name, last name, and phone number. If you're using a text editor, The values must be separated by commas. The file must be saved as a .csv file.

If an entry has the same speed dial number as an existing speed dial number in the system, the imported entry will overwrite the existing speed dial number.

If you're using a text file, it should look like this:

300, John, Doe, 1235551212 301, Jane, Doe, 1235551213

If you're using a spreadsheet program, put the speed dial number, first name, last name and phone number each in their own column.

R 1	🖾 Microsoft Excel - speed dial list.csv								
	<u>File E</u> dit <u>V</u>	jew <u>I</u> nsert	F <u>o</u> rmat <u>T</u> ool	s <u>D</u> ata <u>W</u> indo	w <u>H</u> elp Ad	o <u>b</u> e PDF			
	🖻 🖬 🔒) <i>6</i> Q.	** X 🖻	n 🛍 💅 🗠	* CH *	🔒 Σ 🖡			
	R2C5	-	=						
	1	2	3	4	5	6			
1	*300	John	Doe	1235551212					
2	*301	Jane	Doe	1235551313					
3									

Using a * prefix

You can add a * in front of speed dial numbers to differentiate them from other numbers in the system.

1. Click Select File. The System Speed Dial List (Import) window appears.

System Spee	d Dial List (lı	nport)					\mathbf{X}
Look in:	Difference	Management 7	7.30	[• 1	? 	
My Recent Documents	Applicatio Documeni ftpServer Greetings Help Language ib Logs Reseller C Utilities	nFiles s s contact eed_Dials.csv					
My Computer My Network Places	File <u>n</u> ame: Files of <u>type</u> :	Office_Speed_ Comma Separa	_Dials.csv ated Values (*.c	csv)			<u>Open</u> Cancel

- 2. Select the .csv file, and then click Open. A progress bar appears.
- 3. Click Close.
- 4. Choose *File > Save* to save the imported speed dial list to the unit. Imported speed dial numbers will use the first available hunt group (i.e. 9, or 0 in some regions).

Export system speed dial list

The *Export System Speed Dial List* area allows you to export the speed dial numbers to a .csv file. You can then share the .csv file as required, or print it for your users. The first line of the file is a title line, and subsequent lines contain entries. Each entry in the file has the following information: Index number, speed dial number, first name, last name, and phone number. The index entry represents the order of the list within the system.

If you open the .csv file in a text editor (e.g. Microsoft Notepad), each item will be surrounded by quotes. For example:

```
"INDEX", "SSD", "FIRST NAME", "LAST NAME", "PHONE NUMBER"
"1", "300", "John", "Doe", "1235551212"
"2", "301", "Jane", "Doe", "1235551313"
```

If you open the .csv file in a spreadsheet program (e.g. Microsoft Excel), the quotes will be omitted. For example:

R 1	dicrosoft	Excel - speed di	al list.csv	_		
	<u>File E</u> dit	<u>V</u> iew <u>I</u> nsert F <u>o</u> rm	at <u>T</u> ools <u>D</u> ata	Window Help Adobe	PDF	
	🖻 🖪 🖁	3 5 0.♥	አ 🖻 🖻 🔇	🌮 🖓 🖌 🖓	Σ <i>f</i> ∗ ^A Z↓	ZI 🛍 4
	R3C5	▼ =				
	1	2	3	4	5	6
1	SSD	FIRST NAME	LAST NAME	PHONE NUMBER		
2	*300	John	Doe	1235551212		
3	*301	Jane	Doe	1235551313		
4						



If the phone number has more than 15 digits and does not have any special characters (-, ,, w, * or #), Microsoft Excel will convert it to scientific notation. Do not attempt to import a .csv file that contains a phone number in scientific notation.

1. Click Export. The System Speed Dial List (Export) window appears.

System Speet	ed Dial List (Export)
Save in:	🔁 FortiVoice Management 7.30 💌 🤌 🍽 📰
My Recent Documents Desktop	ApplicationFiles Documents ftpServer Greetings Help Languages Ib Coss Reseller Contact Uitities Docs Second Disk provides
My Documents My Computer My Network Places	File name: Save Files of type: Comma Separated Values (*.csv)

- 2. Enter the name of the .csv file, and then click Save. A progress bar appears.
- 3. Click Close.

Caller ID name tagging

The *Caller ID Name Tagging* area allows you to enable the substitution of the caller ID name with the name from the *System Speed Dials* page.

Caller ID name tagging compares the phone number from the caller ID of an incoming call with the phone numbers from the *System Speed Dials* page. If a matching phone number is present, the system will replace the name from the caller ID with the first name and last name from the *System Speed Dials* page. These names will appear on the extension instead of the caller ID name.

For more information, see "Caller ID (or CLID) Based Routing" on page 103.

- 1. To enable caller ID name tagging of the entries within the System Speed Dials page, select the Use the first and last names specified above to replace the incoming Caller ID names checkbox.
- **2.** Select the order for the names. Choices are *First name followed by last name*, and *Last name followed by first name*.

Telephone Lines

The *Telephone Lines* page allows you to set up the telephone numbers, telephone company services, and call handling for each telephone line. You can also optimize the unit to match the telephone lines.

erm .	Line	Label		
	1			Line detected at initialization: Ye
	2		Phone Numbers	Line Optimization-
	3			M
	4		Main number:	Line optimization should be done whenever a line
	6		Distinctive ring 1:	is added of changed.
	7			Optimize Lines
	8		Distinctive ring 2:	
			Caller ID Call Waiting	Conference Hunt/Rollover/Busy Forwarding
			Call Handling	Conference Hunt/Rollover/Busy Forwarding
			Call Handling	Conference Hunt/Rollover/Busy Forwarding Distinctive Ring 2
			Call Handling Main Number Distinctive Ring 1 Mode 1 (Business Hours) Mode 2	Conference Hunt/Rollover/Busy Forwarding Distinctive Ring 2 2 (After Hours) Holiday Mode
			Call Handling Main Number Distinctive Ring 1 Mode 1 (Business Hours) Mode 2 When a call comes in on this phone num	Conference Hunt/Rollover/Busy Forwarding Distinctive Ring 2 2 (After Hours) Holiday Mode nber, perform the following action:
			Call Handling Call Handling Main Number Distinctive Ring 1 Mode 1 (Business Hours) Mode 2 When a call comes in on this phone num ring extensions	Conference Hunt/Rollover/Busy Forwarding Distinctive Ring 2 2 (After Hours) Holiday Mode mber, perform the following action:
			Call Handling Call Handling Call Handling Call Handling Main Number Distinctive Ring 1 Mode 1 (Business Hours) Mode 2 When a call comes in on this phone num ring extensions Call Edit. If all extensions are busy or the call is n	Conference Hunt/Rollover/Busy Forwarding Distinctive Ring 2 2 (After Hours) Holiday Mode mber, perform the following action:
			Call Handling Call Handling Call Handling Main Number Distinctive Ring 1 Mode 1 (Business Hours) Mode 2 When a call comes in on this phone num ring extensions Edit If all extensions are busy or the call is n perform no action	Conference Hunt/Rollover/Busy Forwarding Distinctive Ring 2 2 (After Hours) Holiday Mode mber, perform the following action: not answered: after 5 y rings.

1. Select the *Telephone Lines* page.

Activate line

The *Activate Line* area allows you to activate a telephone line. All telephone lines are activated by default.

Line detected at initialization: Yes appears if the telephone line was connected when you started the Management software. Line detected at initialization: No appears if the telephone line was not connected.

- **1.** Select a telephone line.
- 2. If necessary, select the Activate Line checkbox.
- **3.** In certain countries, a *Service Provider* menu is displayed. Use the menu to identify the company that provides service for the selected telephone lines.

Phone numbers

The Phone Numbers area allows you to enter the main and distinctive ring phone numbers.

The Distinctive Ring service provides one or two additional phone numbers to ring the same telephone line. Each phone number has a different ring pattern. You can set up separate call handling for the main number, distinctive ring 1 number, and distinctive ring 2 number.

- Main number corresponds to the normal ring pattern.
- Distinctive ring 1 corresponds to a double ring pattern.
- Distinctive ring 2 corresponds to a triple ring pattern.

Ringing patterns vary based upon region. Check with your service provider.

- 1. Enter the phone number of the telephone line in the *Main number* box.
- **2.** If the telephone line has the Distinctive Ring service, and you want to set up different call handling for the distinctive ring numbers:
 - a. Select the Distinctive ring 1 and Distinctive ring 2 checkboxes.
 - **b.** Enter the distinctive ring numbers.
- **3.** If the telephone line has the Distinctive Ring service, but you want the same call handling for each distinctive ring number, clear the *Distinctive ring 1* and *Distinctive ring 2* checkboxes.

Phone line services

The *Phone Line Services* area allows you to select the telephone company services that are active on the telephone line.

- *Transfer and Clear* allows the unit to release the line after transferring a call from an outside caller to a remote extension. You can enable this feature if the Transfer and Clear service will allow a call to remain in progress between the outside caller and the remote extension, after the unit hangs up. Enabling this feature without the service will result in the caller being disconnected when their call is transferred to a remote extension. To enable the feature, select the *Clear telephone line after call transferred* checkbox in the *Transfer Preferences* window. See "To remote extension tab" on page 133.
- 3-Way Calling/Conference allows the same line connect feature to work with the line's 3-Way Calling/Conference service for call bridge (DISA) calls, and with incoming calls that are routed to remote extensions.
- *Hunt/Rollover/Busy Forwarding* indicates the line has a hunting facility or call forward on busy service. If an incoming call reaches a busy line, it will automatically try another line.
- *Caller ID* allows the unit to forward caller ID information to the local extensions. The unit can also use the time information provided by caller ID to update its internal clock.
- *Call Waiting* sounds a tone if an incoming call reaches a busy line. Call waiting is not recommended, since the system cannot answer the second call. If your installation has more than one line consider using the telephone company's Hunt/Rollover service. The system has call waiting built-in, and does not require the Call Waiting service from the telephone company.
- When the Call Waiting service checkbox is enabled for an external telephone line, a user can pick up a waiting call on that line by pressing *Flash* or *Recall* then 89 on an analog phone, or *Xfer/Transfer/TRNF* then 89 on an IP phone.
- *Line Reversal* is used in some regions to exchange signals between the telephone company and your telephone equipment. If you select the *Line Reversal* checkbox, click the *Line Reversal Settings* button to select the reversal patterns used by your telephone company.

Note that line reversal configuration is not offered in the United States or Canada.

Line reversal settings

Line reversal is used in some regions to exchange signals between the telephone company and your telephone equipment. Line reversal can improve performance and lower costs on busy telephone lines. Check with your telephone company to see whether line reversal is used, and when. For example, the unit can send a signal when it:

- Seizes the line. The unit seizes a line when a user dials a hunt group number.
- Answers the line.
- Releases the line. The unit releases the line when the user hangs up.

Note that line reversal configuration is not offered in the United States or Canada.

1. If line reversal is used, click *Line Reversal Settings*. The *Line Reversal Settings* window appears.

ine R	eversal Settings	
Г ^{Саll :}	Settings	
Inco	oming call reversal pattern:	Ignore reversal 🗸 🗸
Out	going call reversal pattern:	Answer and release 🗸

- **2.** Select the reversal patterns specified by your telephone company for incoming and outgoing calls. The reversal patterns include:
 - Ignore reversal Do not send or respond to line reversal signals.
 - Seizure and release Send a signal when a line is seized, and again when the line is released.
 - Seizure and answer Send a signal when a line is seized, and again when the line is answered.
 - Answer and release Send a signal when a line is answered, and again when the line is released.

Call handling

The *Call Handling* area allows you to set up what happens when a call comes in from this line. Calls can be handled differently in each of the three modes. Calls can be sent directly to an extension, or they can ring extensions, go to an auto attendant, ring group, voice mailbox, announcement or VoIP location.

For example, the system can ring the receptionist when Mode 1 is active during the day. If the receptionist doesn't answer, it can start ringing other users as well. If there is still no answer, it can play an auto attendant. The auto attendant provides the dial-by-name directory, and allows the caller to dial an extension. A voice mailbox can immediately answer the VoIP number when Mode 2 is active at night or on weekends. Another auto attendant can immediately route the call to a remote extension during Holiday Mode.

- 1. Select the Mode tab. Choices are Mode 1, Mode 2 and Holiday Mode.
- 2. Select the action you wish to apply:
 - go to auto attendant plays the selected auto attendant.
 - go to local extension goes to the specified extension. Busy or unanswered calls will follow the call cascade for that extension.
 - go to remote extension goes to the specified extension. Busy or unanswered calls will follow the call cascade for that extension.
 - go to ring group goes to the specified ring group. The call will follow the call cascade for that ring group.
 - go to voicemail Accesses the selected voice mailbox.
 - play announcement Plays the selected announcement.
 - *ring extensions* Sets extensions to ring.

You can set up a ring sequence. The ring sequence determines when each local extension will start ringing. The extensions will not use their call cascades during the ring sequence. The system transfers the call to the first extension that is answered.

i. Click *Edit*. The *Ring Sequence* window appears.

Extension to Ring.	At	ring:
123 - Fax Machine	1	~
222 - Andy Parker	1	~
223 - Jackie Harrison	1	~

- ii. Select the local extensions to ring.
- iii. For each selected extension, set when the local extension will start ringing. All extensions ring after the first ring by default.
- iv. If you assign a different value to each extension, the extension with the lowest number of rings will ring first. The other extensions will start ringing as the number of rings goes up. For example, the system can ring the receptionist first, and then ring other users if the receptionist doesn't answer.
- v. Set up call handling to route an unanswered call. *Perform no action* causes a generic auto attendant to answer after 15 rings to allow authorized callers to make configuration changes, access voicemail or dial extensions.
- vi. Select when the system will perform call handling.

Line optimization

The *Line Optimization* area allows you to calibrate the unit to match the telephone lines. Optimization calibrates transmission for the type of telephone line or equipment connected to the unit by setting the far end impedance. Optimization improves performance for call bridge (DISA) and call forwarding.

You should calibrate the lines:

- after adding or changing a telephone line.
- after connecting a line to equipment other than a telephone company line.
- if poor audio performance is encountered.
1. Click *Optimize Lines*. The *Optimize Lines* window appears.



2. Select the line(s) to optimize, and then click *Run Optimization*.

PRI Numbers

The *PRI Numbers* page allows you to set up the PRI numbers. A PRI number is a telephone number delivered over your T1/E1 PRI line. You will be provided a primary or main number, as well as DID numbers from the provider.

You can set up unique call handling for each number. You can set up call handling for Mode 1, Mode 2 and Holiday Mode.

If a PRI number is called, it can be sent directly to one extension or a sequence of extensions, to an auto attendant, ring group, voice mailbox or announcement.

1. Select the *PRI Numbers* page.

Number	Activate Main Number	
613555555	Main Number: 613555555	
	r Call Handling	(
	Mode 2 Holday mode	1
	When a call comes in on this phone number, perform the following action:	
	ring extensions	
1	If all extensions are busy or the call is not answered:	
	perform no action 🗸 a	fter 5 🚽 rings.
		5
	_ *	
ttings		
D Chan		
D Onlan		
umber of active PRI channel	s: 23 👻	

Activate number

The Activate Number area allows you to activate a PRI number.

- **1.** Select an ID number.
- 2. Select the Activate Number checkbox.
- **3.** Enter the phone *Number*.

Call handling

The *Call Handling* area allows you to set up what happens when a call comes in from this line. Calls can be handled differently in each of the three modes. Calls can be sent directly to an extension, or they can ring extensions, go to an auto attendant, ring group, voice mailbox, announcement or VoIP location. For example, the system can ring the receptionist when Mode 1 is active during the day. If the receptionist doesn't answer, it can start ringing other users as well. If there is still no answer, it can play an auto attendant. The auto attendant provides the dial-by-name directory, and allows the caller to dial an extension. A voice mailbox can immediately answer the PRI number when Mode 2 is active at night or on weekends. Another auto attendant can immediately route the call to a remote extension during Holiday Mode.

- 1. Select the Mode tab. Choices are Mode 1, Mode 2 and Holiday Mode.
- 2. Select the action you wish to apply:
 - go to auto attendant plays the selected auto attendant.
 - go to local extension goes to the specified extension. Busy or unanswered calls will follow the call cascade for that extension.
 - go to remote extension goes to the specified extension. Busy or unanswered calls will follow the call cascade for that extension.
 - go to ring group goes to the specified ring group. The call will follow the call cascade for that ring group.
 - go to voicemail Accesses the selected voice mailbox.
 - play announcement Plays the selected announcement.
 - ring extensions Sets extensions to ring.

You can set up a ring sequence. The ring sequence determines when each local extension will start ringing. The extensions will not use their call cascades during the ring sequence. The system transfers the call to the first extension that is answered.

i. Click Edit. The Ring Sequence window appears.

Extension to Ring.	,	At ring:
123 - Fax Machine	1	~
222 - Andy Parker	1	~
223 - Jackie Harrison	1	~

- **ii.** Select the local extensions to ring.
- iii. For each selected extension, set when the local extension will start ringing. All extensions ring after the first ring by default.

If you assign a different value to each extension, the extension with the lowest number of rings will ring first. The other extensions will start ringing as the number of rings goes up. For example, the system can ring the receptionist first, and then ring other users if the receptionist doesn't answer.

- iv. Set up call handling to route an unanswered call. *Perform no action* causes a generic auto attendant to answer after 15 rings to allow authorized callers to make configuration changes, access voicemail or dial extensions.
- v. Select when the system will perform call handling.

Settings

The Settings area allows you to configure the PRI service.

D Channel

The D Channel is the signalling channel for your PRI service. This channel is configured to provide the system with inbound call notification and caller ID information. The default channel is 24 for North American and channel 16 elsewhere.

Number of active PRI channels

If subscribing to a PRI service with fewer PRI channels, set the number of channels to match the amount being provided.

VoIP Numbers

A VoIP number is like a telephone number. It allows a caller to dial the system through a VoIP service provider.

You can set up unique call handling for each VoIP number. Call handling for VoIP numbers is identical to call handling for telephone numbers. You can set up call handling for Mode 1, Mode 2 and Holiday Mode.

If a VoIP number is called, it can be sent directly to one extension or a sequence of extensions, to an auto attendant, ring group, voice mailbox or announcement.

The VoIP Numbers page allows you to set up VoIP numbers.

1. Select the VoIP Numbers page.

VoIP Numbers	篇 [1]
D VolP Number 1 2 3 4 5 6 7 8	Activate VolP Number Select a VolP Profile: FortiCall Phone Number Country code: City or area code: Number: 13438821545 User/Account: System D Password: Password
9 10 11 12 13 14 15 56	Registration Status (View All Registrations)
16 17 18 19 20 21 22 23 24 13438821545	Mode 1 (dayMode) Mode 2 (nightMode) Holiday Mode When a call comes in on this phone number, perform the following action: go to local extension 115 - Joe Smith go to local extension 115 - Joe Smith Busy or unanswered calls will follow the call handling options configured for this extension

Activate VoIP number

The Activate VoIP Number area allows you to activate a VoIP number and select its profile.

- **1.** Select a VoIP number.
- 2. Select the Activate VoIP Number checkbox.
- 3. Select the VoIP Profile. Choices are:
 - Service provider (1 to 4) The VoIP number will use the service provider. The Phone Number area will contain the Country code box, City or area code box and Number box, and the Username and Password area will be enabled.

Phone number

If a service provider profile is used, use the country code, area code and phone number, as provided by the service provider.

- 1. Enter the *Country Code*, if required.
- 2. Enter the City or Area Code.
- **3.** Enter the *Number*.

Username and password

If a service provider profile is used, the *Username and Password* area allows you to enter the authentication information required to access the service provider's SIP server.

- 1. Enter the *User/Account*, as provided by the service provider.
- 2. Enter the *Password*, as provided by the service provider.

Registration status

See "Viewing registration status" on page 31.

Call handling

The *Call Handling* area allows you to set up what happens when a call comes in from this line. Calls can be handled differently in each of the three modes. Calls can be sent directly to an extension, or they can ring extensions, go to an auto attendant, ring group, voice mailbox, announcement or VoIP location.

For example, the system can ring the receptionist when Mode 1 is active during the day. If the receptionist doesn't answer, it can start ringing other users as well. If there is still no answer, it can play an auto attendant. The auto attendant provides the dial-by-name directory, and allows the caller to dial an extension. A voice mailbox can immediately answer the VoIP number when Mode 2 is active at night or on weekends. Another auto attendant can immediately route the call to a remote extension during Holiday Mode.

- 1. Select the *Mode* tab. Choices are *Mode* 1, *Mode* 2 and *Holiday Mode*.
- 2. Select the action you wish to apply:
 - go to auto attendant plays the selected auto attendant.
 - go to local extension goes to the specified extension. Busy or unanswered calls will follow the call cascade for that extension.
 - go to remote extension goes to the specified extension. Busy or unanswered calls will follow the call cascade for that extension.
 - go to ring group goes to the specified ring group. The call will follow the call cascade for that ring group.
 - go to voicemail Accesses the selected voice mailbox.
 - *play announcement* Plays the selected announcement.
 - *ring extensions* Sets extensions to ring.

You can set up a ring sequence. The ring sequence determines when each local extension will start ringing. The extensions will not use their call cascades during the ring sequence. The system transfers the call to the first extension that is answered.

i. Click *Edit*. The *Ring Sequence* window appears.

Extension to King.	At	ring:
123 - Fax Machine	1	~
222 - Andy Parker	1	~
223 - Jackie Harrison	1	~

- ii. Select the local extensions to ring.
- **iii.** For each selected extension, set when the local extension will start ringing. All extensions ring after the first ring by default.

If you assign a different value to each extension, the extension with the lowest number of rings will ring first. The other extensions will start ringing as the number of rings goes up. For example, the system can ring the receptionist first, and then ring other users if the receptionist doesn't answer.

- iv. Set up call handling to route an unanswered call. *Perform no action* causes a generic auto attendant to answer after 15 rings to allow authorized callers to make configuration changes, access voicemail or dial extensions.
- v. Select when the system will perform call handling.

Caller ID (or CLID) Based Routing

Incoming calls include caller ID information. The caller ID (referred to as CLID in some regions) includes the phone number and perhaps the name of the caller. The *Caller ID Based Routing* page allows you to set up call handling based on the caller ID information. The system will check the caller ID of each incoming call. If the caller ID matches a caller ID entry, the system will route the call accordingly.

For example, if the call is from an important client, the call can be routed directly to the president's extension. If no caller ID is present, the call can be routed to voicemail.

You can define up to 200 caller ID entries. Each caller ID entry has an optional name, a phone number and a routing assignment. The routing assignment can use call handling set up for a group, or can display an alternate name on the user's extension, instead of the name from the caller ID.

If caller ID entries have phone numbers with overlapping digits, the system will use the best (longest) match to route the call. For example, the first entry routes calls from phone numbers that start with *5551*. The second entry routes calls from phone numbers that start with *5551*. The second entry routes calls from phone numbers that start with *555*. If the incoming call is from 555-1234, both entries match. However the system will use the first entry because it is a better match with a longer set of matching digits.

You can set up call handling for up to ten groups for Mode 1, Mode 2 and Holiday Mode. Call handling for a group is identical to call handling for telephone numbers, except you can set the ring pattern for each group. Depending on the caller ID, an incoming call can ring selected local extensions in sequence, and then play an auto attendant or announcement, or go to a voice mailbox. Alternatively, it can immediately play an auto attendant or announcement, or go to a voice mailbox, without ringing the extensions.

1.	Select the	e Caller ID	Based	Routing page.	
----	------------	-------------	-------	---------------	--

D	Name		100 A C 100 A C 100 A C 1					
1			Up to 200 matchin	g options can be co	onfigured to route calls acc	ording to one of	he 10 groups config	gured below
3			anu/or replace the	Caller ID name.				
4			Replace	Caller ID name with	n:	1		
5 6		_	Ca	ller ID type to match	n: <select></select>	~		1
7			When matched in	erform the following	use original Caller ID	~		
9		_	rmon materioù, pi	anoth the relie wing	g. dee original callel 10			
10		v						
ller ID R	Routing Groups							
ller ID R	Routing Groups	ed to direct ca	alls when the Call	er ID of an inbound o	call is matched to one of t	ne 200 entries in	he Caller ID Lookup	List. To assign the
ller ID R en rout revious	Routing Groups	ed to direct ca	alls when the Call , dial *81 + a group	er ID of an inbound ((1-10) then # at sy	call is matched to one of ti /stem dialtone.	ne 200 entries in	he Caller ID Lookup	List. To assign the
ller ID R en rout revious Group	Routing Groups	ed to direct ca	alls when the Call dial *81 + a group troup 1	er ID of an inbound ((1-10) then # at sy	call is matched to one of the vistem diatione.	ne 200 entries in	he Caller ID Lookup	List. To assign the
ller ID R en rout revious Group 1	Routing Groups	ed to direct ca in extension,	alls when the Call , dial *81 + a group iroup 1	er ID of an inbound ((1-10) then # at sy	call is matched to one of t ystem dialtone.	ne 200 entries in	he Caller ID Lookup	List. To assign the
lier ID R en rout revious Group 1 2 3	Routing Groups ing groups can be use a call to a group from a	ed to direct ca	alls when the Call , dial *81 + a group iroup 1 Group Label: Mode 1 (Busin	er ID of an inbound o (1-10) then # at sy	call is matched to one of the state of the s	ne 200 entries in	he Caller ID Lookup	List. To assign the
ller ID R en rout revious Group 1 2 3 4	Routing Groups	ed to direct ca	alls when the Call , dial *81 + a group iroup 1 Group Label: Mode 1 (Busin	er ID of an inbound o (1-10) then # at sy ess Hours) Mo	call is matched to one of ti ystem dialtone.	ne 200 entries in oliday Mode	he Caller ID Lookup	List. To assign the
ller ID R en rout revious Group 1 2 3 4 5 5	couting Groups	ed to direct co	alls when the Call dial *81 + a group iroup 1 Group Label: Mode 1 (Busin When a call is r	er ID of an inbound o (1-10) then # at sy ess Hours) Mo outed to this group,	call is matched to one of ti ystem dialtone. ode 2 (After Hours) H , perform the following act	ne 200 entries in oliday Mode on:	he Caller ID Lookup	List. To assign the
ller ID R en rout revious Group 1 2 3 4 5 6 7	couting Groups	ed to direct ca	alls when the Call dial *81 + a group iroup 1 Group Label: Mode 1 (Busin When a call is r ring extensions	er ID of an inbound o (1-10) then # at sy ess Hours) Mo outed to this group, s V	call is matched to one of ti ystem dialtone. ode 2 (After Hours) H , perform the following act Edit	oliday Mode	he Caller ID Lookup	List. To assign the
ller ID R en rout revious Group 1 2 3 4 5 5 6 7 7 8	couting Groups	ed to direct ca	alls when the Call dial *81 + a group iroup 1 Group Label: Mode 1 (Busin When a call is r ring extensions If all extensions	er ID of an inbound o (1-10) then # at sy ess Hours) Mo outed to this group, are busy or the cal	call is matched to one of ti ystem dialtone. ode 2 (After Hours) H , perform the following act Edit	oliday Mode	he Caller ID Lookup	List. To assign the
ller ID R en rout revious Group 1 2 3 4 5 5 6 6 7 8 9 9	couting Groups	ed to direct ci in extension,	alls when the Call dial *81 + a group iroup 1 Group Label: Mode 1 (Busin When a call is r ring extensions If all extensions perform no act	er ID of an inbound o (1-10) then # at sy ess Hours) Mo outed to this group, are busy or the cal on	call is matched to one of ti ystem dialtone. ode 2 (After Hours) H , perform the following act Edit	oliday Mode	he Caller ID Lookup	List. To assign the

Note: Caller ID is referred to as CLID in some regions.

Caller ID lookup list or CLID matching list

The *Caller ID lookup list* area (or *CLID matching list* in some regions), allows you to define up to 200 caller ID entries. Each entry has an optional name, a phone number and a routing assignment. The routing assignment can use call handling set up for a group, or can display an alternate name on the user's extension, instead of the name from the caller ID.

- 1. Optionally, enter the name in the *Replace Caller ID name with* box. The extension will display this alternate name, instead of the name from the caller ID.
- 2. Select the Caller ID type to match or CLID type to match. Choices are:
 - *Private/blocked* Defines an entry for incoming calls that have caller ID blocked for privacy reasons.
 - Long distance/unknown Defines an entry for incoming calls that don't have caller ID.
 - *Phone number ends with* Defines an entry for an incoming call from a phone number that ends with the specified digits. For example, you can enter an important client's phone number if you want to route their calls directly to the president's extension.
 - All incoming calls from phone numbers with ending digits that match the specified digits will be routed the same way. For example, the specified digits are 5551212. The system will use the same routing if the phone number is 555-1212, 613-555-1212 or 416-555-1212.
 - Phone number starts with Defines an entry for an incoming call from a phone number that starts with the specified digits. For example, you can enter an area code if you want to route long distance calls to a particular extension for faster processing, or if you want to route telemarketers with 1-800 numbers directly to voicemail.
 - All incoming calls from phone numbers with starting digits that match the specified digits will be routed the same way. For example, the specified digits are 1800. The system will use the same routing if the phone number is 180-0121, 1-800-123-4567 or 1-800-555-1234.



In North America, the 1 prefix is not present on PSTN lines. For PSTN in North America, use 800-555-1234.

- 3. If you selected Phone number ends with or Phone number starts with, enter the digits.
- 4. Select the routing in the When matched, perform the following list. Choices are:
 - use original Caller ID The caller ID will not affect call routing.
 - *replace Caller ID name* The caller ID will not affect call routing, but the extension will display the alternate name, instead of the name from the caller ID.
 - route using Group [1–10] The caller ID is used to route the call according to the specified group. The extension will display the alternate name, instead of the name from the caller ID.

Caller ID or CLID Routing Groups

The Caller ID Routing Groups area (or CLID Routing Groups in some regions), allows you to set up call handling for up to 10 groups for Mode 1, Mode 2 and Holiday Mode. Call handling for a group is identical to call handling for telephone numbers, except you can set the ring pattern for each group. Depending on the caller ID, an incoming call can be sent directly to an extension, or it can ring extensions in a sequence, go to an auto attendant, ring group, voice mailbox, announcement or VoIP location.

For example, when an important client calls, the system can ring the president during the day. If the president doesn't answer, the call can be routed to voicemail. A voice mailbox can immediately answer the call when Mode 2 is active at night or on weekends. The call can go directly to the president's remote extension during Holiday Mode.

- 1. Enter the Group Label to name the group.
- 2. Select the *Mode* tab. Choices are *Mode* 1, *Mode* 2 and *Holiday Mode*.
- 3. Select the action you wish to apply:
 - go to auto attendant plays the selected auto attendant.
 - go to local extension goes to the specified extension. Busy or unanswered calls will follow the call cascade for that extension.
 - go to remote extension goes to the specified extension. Busy or unanswered calls will follow the call cascade for that extension.
 - go to ring group goes to the specified ring group. The call will follow the call cascade for that ring group.
 - go to voicemail Accesses the selected voice mailbox.
 - *play announcement* Plays the selected announcement.
 - ring extensions Sets up a ring sequence.

The ring sequence determines when each local extension will start ringing. The extensions will not use their call cascades during the ring sequence. The system transfers the call to the first extension that is answered.

i. Click Edit. The Ring Sequence window appears.

 Extension to Ring:	At	ring
123 - Fax Machine	1	1
222 - Andy Parker	1	1
223 - Jackie Harrison	1	1

- ii. Select the local extensions to ring.
- **iii.** For each selected extension, set when the local extension will start ringing. All extensions ring after the first ring by default.

If you assign a different value to each extension, the extension with the lowest number of rings will ring first. The other extensions will start ringing as the number of rings goes up. For example, the system can ring the receptionist first, and then ring other users if the receptionist doesn't answer.

- **iv.** Set up call handling to route an unanswered call. *Perform no action* causes a generic auto attendant to answer after 15 rings to allow authorized callers to make configuration changes, access voicemail or dial extensions.
- v. Select when the system will perform call handling.

Setting up caller ID routing with an extension

A user can set up caller ID routing with their extension, once the call is terminated. For example, a telemarketer can be added to a group that routes calls directly to an announcement.

- 1. To add the caller to a group, dial *81 [group number 1–10] #. The system adds the caller ID entry to the *Caller ID lookup list* (or *CLID matching list* in some regions). Future calls will be routed according to the group.
- 2. To display the alternate name instead of the caller ID name, dial *810#. The system adds the caller ID entry to the *Caller ID lookup list* (or *CLID matching list* in some regions). Future calls will display the name from the *Replace Caller ID name with* box.
- **3.** To delete the caller ID entry from the *Caller ID lookup list* (or *CLID matching list* in some regions), dial *81255#. The system will no longer use the caller ID to route the call or display the alternate name.

Line Hunt Groups



Ensure that hunt group 9 or 0 is assigned to the group of telephone lines or VoIP trunks used for calls to emergency services.

A line hunt group is a set of lines that are available for making an outbound call. It can use selected telephone lines, or all VoIP lines associated with a service provider VoIP network.

When placing an outbound call, the user first dials the hunt group number. The system selects an available line from the group. However, the user does not have to dial a hunt group before VoIP numbers. Nor does the user have to dial a hunt group before remote extension numbers. These automatically use the hunt group configured for the remote extension.

You can set the hunt order for telephone lines, but the system automatically determines the hunt order for VoIP lines.

You can set up nine different hunt groups. If you are using multiple service provider VoIP networks, set up a hunt group for each service provider.

A local extension can be restricted to a set of hunt groups, in order to reserve telephone lines for high-priority users, or to control access to VoIP networks. See "User Privileges" on page 34.

The unit uses a hunt group when placing a call from a local extension, to a remote extension, or with the call bridge (DISA) feature. The hunt groups do not affect incoming calls.

The system has the following hunt groups by default. If default settings are used, the system will first search the higher line numbers (e.g. line 4) for outgoing calls, since the lower line numbers are more heavily used for incoming calls. You can modify the default hunt groups as required.

Hunt Group 9 (0 in some regions	Selects line 4, line 3, line 2, then line 1.
Hunt Group 81–88	No lines are assigned.

The *Line Hunt Groups* page allows you to set up the name, line type, set of telephone lines, hunting order, overflow hunt group and overflow notification for each hunt group.

1. Select the *Line Hunt Groups* page.

Name	Activate Hunt Group 9	
	Hunt Group name:	
	Hunt Group Line Assignments-	
	Line type: Phone Lines	
	Unit 1 Line 8 Unit 1 Line 7 Add/Remove Lines	
	Unit 1 Line 6 Unit 1 Line 5 Move Up	
	Unit 1 Line 4 Unit 1 Line 3 Unit 1 Line 2 Move Down	
	Unit 1 Line 1	
	-Hunting Order for Outgoing Calls-	(
	Hunt lines in the following order: Order specified above	
		(
	If all lines are busy in this hunt group, use: no overflow	
	If all lines are busy in the previous hunt group, use: no overflow	

Note: Depending on the region, an operator may be dialed using 9 or 0.

Activate hunt group

The Activate Hunt Group area allows you to activate and name a hunt group.

- **1.** Select a hunt group.
- 2. If necessary, select the *Activate Hunt Group* checkbox. All hunt groups are active by default.
- **3.** Enter a name in the *Hunt Group name* box. The name will identify the hunt group elsewhere in the Management software.

Hunt group line assignments

The *Hunt Group Line Assignments* area allows you to set up a hunt group for telephone lines or VoIP lines.

- 1. Set the *Line type*. Choices are:
 - Phone Lines Allows you to select telephone lines.
 - SPn VoIP Service Uses the service provider VoIP lines in the hunt group.
 - *PRI Lines* use the PRI service in the hunt group.

- 2. If you set *Line type* to *Phone lines*:
 - a. Click Add/Remove Lines. The Add/Remove Telephone Lines window appears.

Unit 1 Line 1	^
🔽 Unit 1 Line 2	
🗹 Unit 1 Line 3	
🗹 Unit 1 Line 4	
🗹 Unit 1 Line 5	
🗹 Unit 1 Line 6	
🗹 Unit 1 Line 7	
🗹 Unit 1 Line 8	~

- **b.** Select the telephone lines for the hunt group, and then click *OK*.
- **c.** Change the order of the lines by selecting a line, and then clicking *Move Up* and *Move Down*.

If you set Line type to SPn VoIP Service, VoIP lines are selected automatically.

Hunting order for outgoing calls

The *Hunting Order for Outgoing Calls* area allows you to specify how the system will hunt for an available line within the hunt group.

- 1. Set the Hunt lines in the following order list. Choices are:
 - Order specified above The system will start with the first line in the hunt group.
 - Search lines on same unit first The system will hunt for lines on the unit that the extension is connected to, before hunting on the other units. This minimizes network traffic between units.
 - Uniform distribution (Circular) The system will start with the line after the last hunted line in the hunt group. This way telephone lines are used more evenly for outbound calls.

For example, the hunt group has lines 1, 2, 3 and 4. The last call with the hunt group used line 2. The system will first check line 3 for availability.

The circular hunt mode can reduce costs if the line toll increases after utilization exceeds a threshold. For example, assume each line costs 5¢ per minute for the first 100 minutes, and then costs 10¢ per minute. The circular hunt mode makes it more likely users will finish the first 100 minutes on each line before the thresholds are exceeded and higher charges are incurred.

Hunt group busy overflow for outgoing calls

The *Hunt Group Busy Overflow for Outgoing Calls* area allows you to select overflow hunt groups. If the user dials a hunt group, but there are no lines available, the system will play a warning tone while hunting for a free line in the first overflow hunt group. If there are no lines available in the first group, the system will hunt for a free line in the second overflow group. The warning tone indicates the call may be more expensive or of different quality than expected. The user can then remain on the line, or hang up and wait for a line to become available in the original hunt group.

If the dialed hunt group and the overflow hunt group are both busy, the user will hear the busy tone.

A hunt group that contains telephone lines can have overflow hunt groups that contain telephone lines or VoIP lines. Similarly, a hunt group that contains VoIP lines can have overflow hunt groups that contain telephone lines or VoIP lines.

The first overflow hunt group must contain different lines than the original hunt group, and the second overflow group must contain different lines than the first overflow group but all lines must support the same dialed number structure.

For example, assume the original hunt group contains telephone lines and the first overflow hunt group contains VoIP lines. If the lines within the original hunt group require you to dial *1* before the area code and phone number (e.g. 1 613 555 1212), the lines within the overflow hunt group cannot require you to omit the *1* (e.g. 613 555 1212).

- 1. Select the overflow hunt group in the *If all lines are busy in this hunt group* list. Choices are no overflow and the activated hunt groups.
- **2.** Select the overflow hunt group in the *If all lines are busy in the previous hunt group, use* list. Choices are no overflow and the activated hunt groups.

Overflow tone notification

The Overflow Tone Notification area allows you to disable the warning tone that sounds when an overflow hunt group is being used. The warning tone sounds by default.

1. To disable the warning tone, clear the *Play notification tone when overflowing to another hunt group* checkbox.

Auto Attendants

An auto attendant can answer a telephone line or VoIP number, and can be included in the call cascade of a local extension, remote extension or ring group.

An auto attendant can answer a call if the receptionist is away or if you don't have a receptionist. Each auto attendant has a message and up to six options. The message tells the caller what the options are. You can load a professionally pre-recorded message, or can record a message using a handset. The caller selects an option by dialing 0 (9 in some regions), 1, 2, 3, 4 or 5. The auto attendant then performs the action programmed for the option. The auto attendant can:

- Transfer the call to a local extension, remote extension or ring group. The call then follows the extension's call cascade.
- Transfer the call to the call queue of a ring group. The call is placed on hold. The system will ring the next available local extension in the ring group.
- Transfer the call to a voice mailbox, allowing the caller to leave a message. The call can be transferred to a local extension, remote extension or general voice mailbox. Pressing * during the greeting returns to the auto attendant. If the 0 option is programmed (9 in some regions), pressing 0 during the greeting (9 in some regions) can route the call to an extension, voice mailbox, announcement or auto attendant. An authorized caller can retrieve messages and perform other voicemail activities by pressing 8 during the greeting. If no digit is pressed during the greeting, the system hangs up after the caller leaves a message.
- Play an announcement with directions, business hours, etc. The announcement can have the *0* option programmed (*9* in some regions) to route the call to an extension, voice mailbox, announcement or auto attendant. The system hangs up after the announcement if no digit is dialed. Pressing * returns to the auto attendant.
- Access the dial-by-name directory so the caller can find a user's extension number. The dial-by-name directory prompts the caller to enter the first three letters of the user's last name. Pressing * returns to the auto attendant. See "Setting up the dial-by-name directory" on page 116.
- Route the call to another auto attendant, which allows actions to be nested into a powerful call routing system. For example, the main auto attendant can say *"Press one for English. Oprima dos para Español."* Option 1 goes to the English auto attendant and option 2 goes to the Spanish auto attendant.

In addition to the six auto attendant options, the caller can:

- Reach an extension by dialing an extension number.
- Access voicemail by pressing ** then the mailbox number.
- Access call back by dialing 6.
- Access call bridge/DISA by dialing a hunt group number.
- Enter command mode by pressing #.

The caller is able to dial an extension, even if the first number of the extension is the same as an auto attendant option. For example, the caller can dial *111* even though the first "1" is the same as auto attendant option 1. This is because the system waits after the caller dials a digit, before following the action for that digit. The default wait time is 1.5 seconds. To change the wait time, go to *Troubleshooting > Auto Attendants*. The *Auto Attendants* window will appear. Select the desired time from the pull-down menu *Single digit fall through time*.

The system automatically copies the auto attendants to each unit on the network. This reduces network traffic and allows the system to continue functioning even if a unit loses power or is disconnected from the LAN.

The recording time for internal music on hold, voicemail, and the auto attendants is shared on the unit.

Configuring auto attendants

1. Select the *Auto Attendants* page.

and Hours					
Auto Atte	ndant label: Business H	ours			
CActions d	uring auto attendant pla	yback			
If caller					
presses	Action		Resource	Language of syster	m prompts
0:	perform no action	~		Default (English)	~
1:	perform no action	~		Default (English)	~
2:	perform no action	×		Default (English)	×
3:	perform no action	~		V Default (English)	~
				Default (Fastish)	
	perform no action			Denault (English)	<u> </u>
5:	perform no action	~		Default (English)	~
	If a fax call is dete	ected: perf	orm no action 🖌 🖌		~
-Action Pe	rformed After Auto Atte	ndant Playt	ack		
If no sele	ection is made, perform t	he followin	g action:		
Wait 5 s	econds 🔽 then ha	ng up	~	*	
					<u> </u>
CAttendan	Greetings				
Greeting	files must be in 8 KHz	8 hit mono	u-law way format		
Length:	00:05 min	a an, mono,	Record	Load Wav	rile
			Play Gr	eeting	

Note: In the Actions During Auto Attendant Playback section, depending on the region, callers can press either 0 or 9.

The Activate Auto Attendant area allows you to activate an auto attendant. All auto attendants are activated by default.

- **2.** Select an auto attendant.
- 3. If necessary, select the Activate Auto Attendant checkbox.
- **4.** Enter the *Auto Attendant label*. The name will identify the auto attendant elsewhere in the Management software.

Actions during auto attendant playback

The Actions During Auto Attendant Playback area allows you to configure the auto attendant with up to six options, associated with the caller pressing 0 (9 in some regions), 1, 2, 3, 4 or 5.

- **1.** Select the *Action*. Choices are:
 - go to voicemail Accesses the selected voice mailbox.
 - go to local extension Transfers the call to the selected local extension.
 - go to remote extension Transfers the call to the selected remote extension.
 - go to ring group Transfers the call to the selected ring group.
 - play announcement Plays the selected announcement.
 - go to auto attendant Routes the call to the selected auto attendant.
 - queue at ring group Transfers the call to the call queue of the selected ring group.
 - dial-by-name directory Accesses the dial-by-name directory. See "Setting up the dial-by-name directory" on page 116.
 - *perform no action* The option is unused.

- 2. Select the *Resource*. Depending on the action selected in Step 1, resources are voice mailboxes, extensions, announcements or auto attendants.
- **3.** Select the language in the *Language of system prompts* list. If the caller selects this option, they will hear all subsequent prompts in the selected language.
- **4.** Select the action *If a fax call is detected*. The system can automatically detect a fax machine that plays a CNG tone. Choices are:
 - go to local extension
 - go to remote extension
 - perform no action
 - hang up
- **5.** Depending on the action selected in Step 4, select the extension the fax machine is connected to.
- 6. If the caller dials an option that causes the auto attendant to transfer their call, the system will play the *"One moment please"* prompt by default. You can disable the prompt. See *"From auto attendant tab"* on page 135.

Transfer Preferences	×
Specify the options for transferring a caller:	
To Local Extension To Remote Extension To Ring Group From Auto Attendant	
LIL nother bar bar another bar and the bar	-0
The system can act as an auto attendant for other PBXs. When doing this, the PBX extension length of the other system needs to be specified.	
PBX extension length: not used	
CAuto Attendant Transfer Prompts	-0
Disable playing of transfer prompts to callers when redirecting calls from the Auto Attendant	

- a. Choose Options > Transfer Preferences. The Transfer Preferences window appears.
- **b.** Select the From Auto Attendant tab.
- c. To disable the prompt, select the Disable playing of transfer prompts checkbox.

Action performed after auto attendant playback

The Action Performed After Auto Attendant Playback area allows you to set up how the system will react if the user makes no selection. This can occur if the caller does not understand the prompts, does not have tone dialing on their phone, or prefers to speak with a person.

- 1. Select the time limit, ranging from 0 seconds to 30 seconds.
- 2. Select the action. Choices include:
 - go to voicemail
 - go to local extension
 - go to remote extension
 - go to ring group
 - play announcement
 - go to auto attendant
 - queue at ring group
 - *dial-by-name directory* See "Setting up the dial-by-name directory" on page 116.
 - hang up
- **3.** Depending on the action selected in Step 2, select the voice mailbox, extension, announcement, or auto attendant.

Working with auto attendant messages

You can record a new message, load a professionally recorded message, or erase a message from the system.



If you don't set up an auto attendant message, the system will answer without playing a message. The caller will hear silence and might assume the call did not get connected.

Recording a new message

You can record a new message from the Management software or from a local extension or remote phone.

To record from the software

- 1. Open the *Auto Attendants* page. Select the auto attendant that you wish to record and click the *Record Greeting* button in the *Attendant Greeting* section.
- 2. Select the local extension that you wish to record from and click OK. The system will call the extension.
- **3.** Pick up the extension and follow the prompts to record the greeting.

To record from a local extension or remote phone

- 1. Pick up a local extension, or dial into the system from a remote phone. If you pick up a local extension, you will hear the dial tone. If you dial in from a remote phone, the auto attendant will answer.
- **2.** Enter command mode by either pressing # on an analog extension phone, or *55# on a proprietary IP phone (note: other brands may use *55 Send or *55 Dial).
- 3. Enter the system password, followed by #.
- **4.** Dial *4* [auto attendant number] # to record the message. For example, dial *41*# to record the message for auto attendant 1.
- 5. Press # when you have completed saying the message.
- 6. Dial 5 [auto attendant number] # to listen to the message. For example, dial 51# to listen to the message for auto attendant 1.
- 7. Repeat Steps 4 to 6 to re-record the message, or hang up to keep the message.

Listening to the recorded message

In the Management software

- 1. Select the auto attendant that you wish to listen to and click on the *Play Greeting* button.
- 2. Choose a local extension then click *OK*. The system will call the extension and play the auto attendant greeting.

Loading a message

The Attendant Greetings area allows you to load the message for the auto attendant.

You can load an 8 kHz, 8 bit, Mono, u-Law .wav file that contains the auto attendant message. If using a professionally recorded message, ensure it is in this format. Maximum message length is 5 minutes.

1. Select an auto attendant.

2. Click Load Greeting. The Select Wav File window appears.



3. Click *Browse* to select the .wav file. Click *Load* to load the .wav file into the system. *Play* and *Stop* buttons are also provided, if you prefer to listen to the file before loading it.

Erasing a message

You can erase a recorded or loaded message using a local extension or remote phone. Erase unused auto attendant messages to free up space for voicemail.

- 1. Pick up a local extension, or dial into the system from a remote phone. If you pick up a local extension, you will hear the dial tone. If you dial in from a remote phone, the auto attendant will answer.
- 2. Enter command mode by either pressing # on an analog extension phone, or *55# on a proprietary IP phone (note: other brands may use *55 Send or *55 Dial).
- 3. Enter the system password followed by #.
- **4.** Dial *04 [auto attendant number]* # to erase the message. For example, dial *041* # to erase the message for auto attendant 1.

Example auto attendant

The following auto attendant answers incoming calls, and plays the following recorded message:

"Welcome to ABC Company. If you know your party's three-digit extension, you may dial it now. To reach the receptionist, dial 0 (9 in some regions) or stay on the line. Dial 1 to find your party's extension in our dial-by-name directory. Dial 2 to hear our business hours and directions to our location. Dial 3 to talk to the next available customer support representative, or dial 4 to leave them a message."

If the caller is a fax machine that plays the CNG tone, the system will route the call to the fax machine at extension 118. If no selection is made, the call is routed to the receptionist at extension 114.

Jusiness Hours				
Auto Att	endant label: Business Hours			
Actions	during auto attendant playback			
If caller				
presse	s Action	Resource	Language of system prompts	
	go to local extension 👻	115 - Joe Smith 🗸	Default (English) 🚽	
		200	Default (Destable)	
1:	go to ring group 👻	300 - myringgroup 🗸 🗸	Default (English) 👻	
2:	queue at ring group 🚽	300 - myringgroup	Default (English)	
3 :	perform no action 👻		Default (English) 👻	
A.	dial by name directory		Default (English)	
		· · · · · · · · · · · · · · · · · · ·		
5:	perform no action 🚽		Default (English)	
	If a fax call is detected:	perform no action		
- Action F	Performed After Auto Attendant	Playback-		
IT NO SE	ection is made, perform the fol	lowing action:		
Wait 1	5 seconds 👻 then hang up	▼	· ·	
Attenda	nt Greetings			
Greetin	n files must be in 8 KHz 8 bit	mono ulaw way format		
Length	00.07 min	Record Gree	Load wav File	
Lengu	. 00.07 min	Play Greet	ina	

Note: In the Actions During Auto Attendant Playback section, depending on the region, callers can press either 0 or 9.

Setting up the dial-by-name directory

The dial-by-name directory allows a caller to find a user's extension number, and connect to their local extension or remote extension. This way the caller can reach their party without speaking to the receptionist.

When prompted by the auto attendant, the caller selects the dial-by-name directory, and then dials the first three letters of the user's first or last name. Alternatively, the caller can dial only one or two letters. If a matching entry is found, the system will play the user's name and extension number. The caller can then dial *1* to connect to the user's extension. If there are multiple matches, the caller can dial *2* to hear the next matching name and extension number.

- 1. The directory can search by last name only, first name only or first and last name. Under the *Options* menu, select *Dial-by-name Directory* to specify the search method.
- 2. Set up the local extensions and remote extensions. The name fields from the extensions will be used in the directory.

3. Have each user set up their voice mailbox by dialing **# and then following the prompts. They will record their names for the dial-by-name directory. Alternatively, you can set up their voice mailboxes by dialing ** + voice mailbox number + # and then following the prompts.



The name must be recorded for the entry to be available.

- 4. Configure the telephone line or VoIP number to be answered by an auto attendant. See "Call handling" on page 94.
- 5. Configure the auto attendant with an action to access the dial-by-name directory. See "Actions during auto attendant playback" on page 112.
- 6. Record an announcement for the auto attendant that tells the caller how to access the dial-by-name directory. See "Working with auto attendant messages" on page 114.
- 7. To make the dial-by-name directory available to users as well as callers, configure the Dial 0 Routing feature (Dial 9 in some regions), to connect with the dial-by-name directory. See "Dial 0 or 9 routing" on page 7.

You can delete the names recorded for the dial-by-name directory by resetting the voice mailbox. See "Reset mailboxes" on page 86.

Setting the PBX extension length

You can configure a unit with an auto attendant, and then connect the unit to another PBX. This way the unit can route callers as requested.

If you will connect the unit to another PBX, set the extension length used by the other PBX. See "From auto attendant tab" on page 135.

- 1. Choose Options > Transfer Preferences. The Transfer Preferences window appears.
- 2. Select the From Auto Attendant tab.

cify the options for tr	ansferring a caller:	
To Local Extension	To Remote Extension To Ring Group From Auto Attendant	
		-
f used with another F	BX	
If used with another F	BX-	
If used with another F The system can act i	BX-	_0
rr used with another F The system can act a PBX extension length	BX- as an auto attendant for other PBXs. When doing this, the PBX extension length of the other system needs to be specified.	-0
If used with another F The system can act a PBX extension length	BX-	_@

3. Set the *PBX extension length* to the number of digits in an extension number on the other PBX.

Auto Call Back

Auto call back lets a travelling user gain access to the system without completing the call to the telephone line or VoIP number. This minimizes long-distance and hotel telephone costs. You can set up four auto call back accounts per unit.

The *Auto Call Back* page allows you to set up auto call back accounts. You can set up four auto call back accounts per unit. Each auto call back account includes the name, call back number, telephone line or VoIP number that will trigger auto call back, announced message, password, dialing prefix, and hunt groups.



You must ensure the auto attendant for the telephone line or VoIP number is set to answer after four rings. In order to trigger auto call back, the user must call the telephone line or VoIP number, and then hang up after two rings, but before the auto attendant answers.

Auto Call Back		٢
Auto Call Back	Activate Auto Call Back Label: Number to call back: Use Call Back prefix: Number to Trigger Call Back on Unit 1: Unit 1 Line 1 Account Options Use announced message Use password (4-8 digts): ✓ Allow access to Hunt Groups Hunt Group 81 Hunt Group 82 Hunt Group 83 Hunt Group 84 Hunt Group 87 Hunt Group 9	© (©
	Hunt Group 81 Hunt Group 82 Hunt Group 83 Hunt Group 84 Hunt Group 87 Hunt Group 88 Hunt Group 9 Edit	

1. Select the Auto Call Back page.

Using auto call back

- 1. Call the telephone line or VoIP number, let it ring twice, and then hang up. The unit will dial the call back number configured for that line.
- 2. Press # to accept the call back.
- **3.** Enter the auto call back password + # to enter the system menu. From the system menu you can:
 - Dial an extension. After you complete the call, you will be disconnected.
 - Check your voicemail by dialing ** + extension. After you check your voicemail, press * to return to the system menu.
 - Call an outside number by dialing 9 (0 in some regions), or 81–88 + phone number. To complete the call, press #* to return to the system menu. You can also press ## to call another outside number using the same hunt group.
 - Configure the system by dialing #, entering the system password + #, and then entering the command.

When the system returns your call, it can play a default message that tells the recipient to press # to accept the auto call back. You can record and enable an announced message that will be played instead of the default message. For example, the announced message can tell the front desk at your hotel to transfer the call to your room. You can also disable the announced message to use the default message.

To configure auto call back, call the telephone line or VoIP number, and let the auto attendant answer. Dial 6, and then follow the prompts. You can change the call back number, change the announced message, enable/disable the announced message, and enable/disable the auto call back account.

Activate auto call back

The Activate Auto Call Back area allows you to set up an auto call back account, with the name, call back number, and the telephone line or VoIP number that will trigger auto call back.

- 1. Select an available auto call back account.
- 2. Select the Activate Auto Call Back checkbox.
- 3. Enter the name of the auto call back account in the Label box.
- 4. Enter the call back number in the *Number to call back* box. This is the telephone number at the traveller's destination. Do not enter a hunt group number. Note that the travelling user can change the call back number. The new call back number will replace that entered into the *Auto Call Back* page.
- **5.** To have the unit dial a number before the call back number, select the *Use call back prefix* checkbox, and then enter the prefix. The prefix can include numbers 0–9, *, #, and "," (comma for 2-second pause). For example, you can enter a calling card number, a pause, a PIN number, and another pause.
- 6. Select the telephone line or VoIP number in the *Number to trigger call back* list. Each auto call back account can be associated with one telephone line or VoIP number. Calling this telephone line or VoIP number and then hanging up will trigger the call back.
- 7. Assign a user privilege to this account.
- 8. After the unit dials an auto call back number or the prompted call back number, it will wait for a configurable number of rings for someone to answer. By default, it will wait six rings before hanging up. You can change the number of rings. See "Call Back" on page 138.
 - a. Choose Options > Call Back. The Call Back Options window appears.

Call Back Attempt Timer-	
During Call Back, number of rings bef	ore the system hangs up: 6 rings 💌

b. Select the number of rings the unit should wait before hanging up.

Account options

To have the unit play a recorded message when the call back is answered, select the *Use announced message* checkbox. This is required if a receptionist will answer the phone after the unit dials the call back number. For example, the announced message can tell the hotel front desk: *"Please transfer this call to Bob in room 307"*. You can record the announced message using a local extension connected to the unit, or remotely by calling a telephone line connected to the unit. See "Recording an announced message" on page 120.

Recording an announced message

This procedure describes how to record the announced message played by the unit when the call back is answered.

- 1. Pick up a local extension connected to the unit, or remotely call a telephone line or VoIP number connected to the unit. You must record the announced message on the same unit that has the auto call back account.
- 2. Enter command mode by either pressing # on an analog extension phone, or *55# on a proprietary IP phone (note: other brands may use *55 Send or *55 Dial).
- 3. Enter the system password, followed by #.
- 4. Enter the command listed below, and follow the prompts.

Auto call back account number	Record message	Erase message	Play message
1	61#	061#	71#
2	62#	062#	72#
3	63#	063#	73#
4	64#	064#	74#

5. Press * to exit command mode.

You can also record, enable or disable an announced message after dialing 6 at the auto attendant.

Prompted Call Back

Prompted call back lets a travelling user gain access to the system. This minimizes long-distance and hotel telephone costs. You can set up one prompted call back account for the system.

The *Prompted Call Back* page allows you to set up prompted call back for the system, including the name, call back number, announced message, password, dialing prefix, and hunt groups.



You must ensure an auto attendant will answer. In order to trigger prompted call back, the user must call the system, wait for the auto attendant to answer, dial 61, and then hang up.

1. Select the *Prompted Call Back* page.

Number to c	all back:				
🗌 Use Call Bac	k prefix:				
			0-101		
ote: The call bac	k number can be changed remote	ely when using Prompted	Call Back.		
Account Option:	3		0		
Use annour	ced message				
	and (4. 9 diatta):				
	ora (4-o aigits).				
Allow acce	ss to Hunt Groups				
Hunt Groups:	Hunt Group 81				
	Hunt Group 82				
	Hunt Group 83				
	Hunt Group 84				
	Hunt Group 88				
	Hunt Group 9				
		201			

Using prompted call back

- **1.** Call the system, and let the auto attendant answer.
- 2. Dial 61, and then hang up. The system will dial the call back number.
- 3. Press # to accept the call back.
- **4.** Enter the prompted call back password + # to enter the system menu. From the system menu you can:
 - Dial an extension. After you complete the call, you will be disconnected.
 - Check your voicemail by dialing ** + extension. After you check your voicemail, press * to return to the system menu.
 - Call an outside number by dialing 9 (0 in some regions), or 81–88 + phone number. To complete the call, press #* to return to the system menu. You can also press ## to call another outside number using the same hunt group.
 - Configure the system by dialing #, entering the system password + #, and then entering the command.

When the system returns your call, it can play a default message that tells the recipient to press # to accept the prompted call back. You can record and enable an announced message that will be played instead of the default message. For example, the announced message can tell the front desk at your hotel to transfer the call to your room. You can also disable the announced message to use the default message.

To configure prompted call back, call the system, and let the auto attendant answer. Dial 6, and then follow the prompts. You can change the call back number, change the announced message, and enable/disable the announced message.

You may incur long-distance and hotel telephone charges each time you access prompted call back from a remote location.

Activate prompted call back

The Activate Prompted Call Back area allows you to enter the name and call back number.

- 1. Select the Activate Prompted Call Back checkbox.
- 2. Enter the call back number in the *Number to call back* box. This is the telephone number at the traveller's destination. Do not enter a hunt group number.
- **3.** Note that the travelling user can change the call back number. The new call back number will replace that entered into the *Prompted Call Back* page.
- **4.** To have the unit dial a number before the call back number, select the *Use call back prefix* checkbox, and then enter the prefix. The prefix can include numbers 0–9, *, #, and "," (comma for 2-second pause). For example, you can enter a calling card number, a pause, a PIN number, and another pause.
- 5. Assign a user privilege to this account.
- 6. After the unit dials an auto call back number or the prompted call back number, it will wait for a configurable number of rings for someone to answer. By default, it will wait 6 rings before hanging up. You can change the number of rings. "Call Back" on page 138.
 - **a.** Choose Options > Call Back. The Call Back Options window appears.

Back Options	X
all Back Attempt Timer	
During Call Back, number of rings before the syst	tem hangs up: 6 rings 💌
	Use Defaults OK Cancel

b. Select the number of rings the unit should wait before hanging up.

Account options

To have the unit play a recorded message when the call back is answered, select the *Use announced message* checkbox. This is required if a receptionist will answer the phone after the unit dials the call back number. For example, the announced message can tell the hotel front desk: *"Please transfer this call to Bob in room 307"*. You can record the announced message using a local extension connected to the unit, or remotely by calling a telephone line connected to the unit. *"Recording an announced message"* on page 122.

Recording an announced message

This procedure describes how to record the announced message played by the unit when the call back is answered.

- **1.** Pick up a local extension, or remotely call the system.
- **2.** Enter command mode by either pressing # on an analog extension phone, or *55# on a proprietary IP phone (note: other brands may use *55 Send or *55 Dial).
- 3. Enter the system password, followed by #.
- **4.** Enter 65# to record the message, 065# to erase the message, or 75# to play the message, and then follow the prompts.
- **5.** Press * to exit command mode.

You can also record, enable or disable an announced message after dialing 6 at the auto attendant.

File menu

The *File* menu has commands for working with configuration files. A configuration file contains parameter settings for the system and has the .tsd extension. The unit uses parameter settings to control operation. The default location for storing configuration files is C:\Program Files, in the folder for the system.

The commands available within the *File* menu change depending upon whether a configuration is open.

If no configuration is open, and the *Configuration Selection* page is displayed, you can:

- Create a new configuration file from the template, with default parameter settings for your system. This way you can set up a configuration without a unit.
- Open the configuration from a unit.
- Open an existing configuration file.

If a configuration is open, you can:

- Save the configuration to a unit. The unit can be local or at a remote location. This is only available if you opened a configuration file.
- Save a new configuration file.
- Save a configuration to an existing configuration file, overwriting it.
- Retrieve the current settings of a connected system.

For more information about the *Configuration Selection* page, see "Starting the Management software" on page 2.

New Template

The *New Template* command creates a configuration file from the template, with default parameter settings for your system. You can then modify the file using the Management software.

1. Choose File > New Template. The Create a new template file window appears.

Alternatively, you can click the *Open a Configuration File or Template* button in the *Configuration Selection* page, and then click the *Create a Template File* button.

Create a	ı new template file 🛛 🗙
Linit 1:	FortiVoice 24P
Linit 2	None Selected
Unit 3:	None Selected
Unit 4:	None Selected V
Templa	te file location: ement 7.00\defaultTemplate.tsd
	Browse

- 2. Select the types of units in your system.
- 3. Click the Browse button to select the location and name of the configuration file.
- 4. Click OK. The system creates the file, and opens it in the Management software.

Open > Location

The *Open > Location* command connects to the selected phone system and retrieves the configuration. The unit can be local or at a remote location. You can then modify the file using the Management software. The command is only available when no configuration is open.

1. Choose *File > Open > Location*. The *Connect Via IP* window appears.

Alternatively, you can also click the *Connect to a Different System* button or the *Connect to a System Via IP* button in the *Configuration Selection* page.

🖉 Connect Via IP	X
Enter the URL / IP Address fo obtain the IP address of a sys any local extension.	r the system or location. To stem on your LAN, dial *56 from
URL / IP address:	192.168.1.103
	Address Book
	OK Cancel

2. Enter the *URL/IP Address* of the unit, and then click *OK*. You can also choose the system from the address book.

Open > Configuration File

The *Open* > *Configuration File* command opens a saved configuration file. You can then use the Management software to modify the configuration. The command is only available when no configuration is open.

1. Choose File > Open > Configuration File. The Configuration File window appears.

Alternatively, you can also click the *Open a Configuration File or Template* button in the *Configuration Selection* page, and then click the *Open a File* button.

- prov	0			
LOOK IN:	- FortiVoic	e Management 7.30	*	
My Recent Documents	Application	nFiles ts s		
Desktop	Help	25		
My Documents	Cogs Reseller (Utilities C defaultTe	Contact mplate.tsd		
My Computer	CastSave	dConfig.tsd		
9	File <u>n</u> ame:			<u>O</u> pe
My Network Places	Files of type:	All Config Files (*.tsd)		Cano

2. Choose the configuration file, and then click Open.

Save

The *Save* command saves the configuration to the open configuration file or to the connected unit. The unit can be local or at a remote location.



New settings must be saved to the unit to take effect.

Save To > File

The Save To > File command saves the configuration to a new configuration file, or overwrites an existing configuration file. It does not save the parameter settings to the unit.

1. Choose *File* > *Save To* > *File*. The *Save* window appears.

Save in:	FortiVoice	e Management 7.30	*	1 2	
	Applicatio	nFiles			
9	🗀 Documen	ts			
My Recent	🛅 ftpServer				
Documents	Greetings				
-	🗀 Help				
	🗀 Language	s			
Deskton	ib lib				
Desktop	Logs				
	Reseller O	Contact			
2	🔂 Utilities				
ly Documents	defaultTe	mplate.tsd			
	LastSave	dConfig.tsd			
-	-				
My Computer					
		2			
7498	Eile name:				Save

2. Enter the name of the configuration file, and then click Save.

Save To > Location

The *Save To > Location* command saves the configuration from the open configuration file to the selected system. The system can be local or at a remote location. The command is only available if you opened a configuration file.

1. Choose *File* > *Save To* > *Location*. The *Location* window appears. If the Management software auto-discovered your local system, the window shows the unit configured as proxy.

catio Syste	n ems discove	ered:	2	~		×
10.1	0.10.3:9393	3 - Firm	ware: 7	.30.000		
Othe	er system		<u>о</u> к		<u>C</u> ancel	_

- 2. To save the configuration to the auto-discovered local system, select the unit configured as proxy, and then click *OK*.
- 3. To save the configuration to a remote system:
 - a. Click Other. The Connect Via IP window appears.

Connect Via IP	$\overline{\mathbf{X}}$
Enter the URL / IP Address for obtain the IP address of a sys any local extension.	r the system or location. To tem on your LAN, dial *56 from
URL / IP address:	192.168.1.103
	Address Book
	<u>O</u> K <u>C</u> ancel

b. Enter the *URL/IP* address of the remote system, and then click *OK*. You can also choose the system from the address book.

Retrieve Settings

The *Retrieve Settings* command opens the configuration from the current unit. You can then modify the configuration using the Management software.

Alternatively, you can also click the *Configure Auto-Detected System* button or the *Retry Auto Discovery* button on the *Configuration Selection* page.

Close

The Close command closes the configuration, and returns to the Configuration Selection page.

Exit

The Exit command closes the Management software.

Tools menu

Memory Usage

1. Choose *Tools > Memory Usage*. The *Memory Usage* window appears. It shows the amount of time provided by the units, and the amount of time used by voicemail and auto attendant messages.

Unit 1: 2 hr						
mmary-						
		9%			1 hr 4	19 min 43 sec remain
ory Usage						
Label		Туре	Greeting Time	New Messages	Total Messages	Message Time
Fax Machine		Voicemail	3 sec	0	0	0 sec
Valerie Cho		Voicemail	0 sec	0	0	0 sec
Peter Altman		Voicemail	0 sec	0	0	0 sec
Andy Parker		Voicemail	0 sec	0	0	0 sec
Jackie Harrison		Voicemail	0 sec	0	0	0 sec
Karen Allidina		Voicemail	0 sec	0	0	0 sec
Jason McKenna		Voicemail	0 sec	0	0	0 sec
		Voicemail	4 sec	0	0	0 sec
Sales inquiries		Voicemail	0 sec	0	0	0 sec
General inquiries		Voicemail	0 sec	0	0	0 sec
Time Usage tess Hour: 5 sec ic éc	AA5 - : 34 sec AA6 - : 34 sec AA7 - : 2 sec AA7 - : 2 sec	AA9 - : 34 sec AA10 - : 0 sec AA11 - : 0 sec	A A A	A13-:0 sec A14-:0 sec A15-:0 sec A15-:0 sec	AA17 - : 0 AA18 - : 0 AA19 - : 0	sec Sec
n 10 sec	AA8 - : / SEC	AA12 - : 0 Sec	A	A16 - 10 Séč	AA20-:0	sec
	Vort 1: 2 hr mary wy Usage Label Fac Marcha Peter Alman, Andry Peter Jackie Harman Kreen Aldina Jackie Harman Kreen Aldina Jackie Harman Sales Inquiries Ceneral Inquiries Ceneral Inquiries	Vinit 1: 2 hr mmny vy Uage Label Fax Machine Valeric Cn Peter Adman Andy Parker Andy Park	Init 1: 2 hr Init 2 hr Ini	Int 1: 2 hr Int 1: 2 hr International Intern	Unit 1: 2 hr wmmy and Add Add high Partier Address Volcensal Address Valence Cine Addy Partier Valence Cine Valence Cine	Unit 1: 2 hr mmry 6% 1 hr / ry Usage 1 hr / 1 hr / Label Type Greeting Time New Messages Total Messages Label Type Greeting Time New Messages Total Messages Fact Machine Violemal 3 acc 0 0 Valence Lin 0 acc 0 0 Andry Parker Violemal 0 acc 0 0 Andry Parker Violemal 0 acc 0 0 0 Stace Industries Violemal 0 acc 0 0 0 Stace Industries Violemal 0 acc 0 0 0 General Inquiries Violemal 0 acc AA13 -: 0 acc AA41 -: 0 Co

The recording time for internal music on hold, voicemail, and the auto attendants is shared on the unit.

Due to deactivation of mailboxes or errors, the system may display messages currently not assigned to a mailbox. Clicking on the *Unassigned* link will allow you to either download wav files to your PC to listen to, or delete them. Saving the files will create a folder in the program directory.

Voicemail Manager > Mailbox Status

See "Mailbox status" on page 85.

Voicemail Manager > Delete Mailbox Password

See "Delete password" on page 86.

Voicemail Manager > Reset Mailboxes

See "Reset mailboxes" on page 86.

Terminal Window (CLI)

The *Terminal Window (CLI)* command displays the *Command Console* window. Use this as directed by customer support.

Call Logging Output (CDR)

The *Call Logging Output (CDR)* command allows you to set up logging of call detail records. For information about call logging, see "Enabling call detail record logging" on page 172.

Defaults > Entire Configuration

The *Defaults > Entire Configuration* command resets the entire configuration to default values.

Defaults > Current Page

The *Defaults > Current Page* command resets the current page of the configuration to default values.

Uninstall License

The Uninstall License command facilitates the removal of licensed add-ons from systems.

To deactivate a licensed add-on, you will require a deactivation key from Technical Support. Technical Support will ask you for your System ID.

- 1. In the Uninstall License window, select the feature you wish to deactivate.
- 2. Enter the deactivation key from Technical Support in the bottom field. Click *Deactivate*.

Click-to-Dial

Users can dial their Microsoft Outlook contacts with a single click.

The *Click-to-Dial* utility must be installed on the PC of the user. Get the free utility from your reseller.

The utility will require an internal IP address or Wan IP address and password for configuration. This information for configuring each extension is listed in the *Click-to-Dial* window.

Admin Observed Events

Admin Observed Events sends notifications to the administrator when certain events occur. The notification options are:

- *EMS Dialed* (emergency services was dialed)
- Call Blocked (attempted call to a blocked number)

To activate event notifications, enter the administrator's e-mail address in the Admin Observed *Events* window.

Update Phone

This launches the phone updater tool, which will allow you to update selected IP phones connected to your system to the latest version of phone firmware you have installed on your PC.

Update Firmware

The *Update Firmware* command shows the firmware version of the units and the latest firmware version available with the Management software. It allows you to update the units with a newer firmware version.

1. Choose Tools > Update Firmware. The System Firmware Update window appears.

stem Firmware Update	
System Information	- System File Information
Firmware on system: 7.10.000	Firmware file selected: 7.10.000
	Cancel Update

- 2. If the Firmware detected is lower than the Firmware file:
 - a. Set the Auto-Reboot checkbox.
 - Select *Auto-Reboot* to automatically reboot the units once the firmware has been updated. This will cancel any calls in progress.
 - Clear Auto-Reboot to prevent the units from rebooting automatically. You will have to manually reboot the units later, when there are no calls in progress. See "Reboot System" on page 130.
 - **b.** Click the Update button.

Update progress will be indicated in the software window, while the lights on the front of the unit(s) will display progress patterns. A dialog will appear to report successful completion of the update. Click *OK* to continue.

System Firmware Update	×
Success! Firmware updated to 7.00.	000.

Reboot System

The *Reboot System* command reboots the units at the specified time. This is required after updating the firmware, unless you selected the *Auto-Reboot* checkbox, and the units rebooted automatically.

1. Choose Tools > Reboot System. The Rebooting System window appears.

Rebooting system			X
Reboot system now (Warning! All	calls in prog	ress will be disconne	cted)
O Lock down system and reboot wi	hen all calls a	are complete	
O Reboot after the system has been	idle for:	minute(s)	
O Reboot system at a specific time:	<no sp<="" td="" time=""><td>ecified></td><td>Set Time</td></no>	ecified>	Set Time
		Reboot	<u>C</u> ancel

- 2. Select when the units should reboot. They can reboot immediately, when all calls are complete, once the system has been unused for a period of time, or once a certain time arrives.
- 3. Click Reboot.

Print > Print Labels

The Tools > Print > Print Labels command allows you to print the labels for IP extensions.

1. Choose *Tools > Print > Print Labels*. The *Select Extensions* window appears.

🖉 101 - Jane Smith		

- 2. Choose the extensions that you wish to print the labels for.
- 3. Click Print.

Print > Print Resources

The *Tools > Print > Print Resources* command allows you output the resources configured within the system into Excel to save or print.

1. Choose Tools > *Print* > *Print Resources*. The *Select Resources* window appears.

Select Resources	
Selected resources will be exported to an Excel file from which you can print.	
Cal / Remote Extensions	1
Ring / Page Groups	
Broadcast / General Voice Mail	
System Speed Dials	
User Privileges	
Network Settings	
Email Service	
Auto Attendants	
Telephone Lines / PRI	
VolP Configuration	
Caller ID Based Routing	
Line Hunt Groups	
	1
Select All Deselect All Proceed Cancel)

- 2. Choose the resources that you wish to print:
 - Local/Remote Extensions
 - Ring/Page Groups
 - Broadcast/General Voicemail
 - System Speed Dials
 - User Privileges
 - Network Settings
 - Email Service
 - Auto Attendants
 - Telephone Lines/PRI
 - VoIP Configuration
 - Caller ID Based Routing
 - Line Hunt Groups
- **3.** Click *Proceed*. The selected resources will be exported to an Excel file from which you can print.

Configuration Assistant

This opens the simplified assisted configuration tool.
Options menu

The Options menu has commands for configuring the system.

Paging Options

The Audio Output Options section allows you to set the audio from the PA jack.

1. Choose Options > Paging. The Paging Options window appears.

Audio Output Options		
PA output option: PA	Announcements 🗸	
		6
Overhead Paging		6
Overhead Paging- If performing overhea recommend enabling over the PA after the	ad paging and want to avoid feedback the Delayed Paging feature. The mess message has been recorded.	, then we age will be played
Overhead Paging If performing overhea recommend enabling over the PA after the Enable Delayed P	ad paging and want to avoid feedback the Delayed Paging feature. The mess message has been recorded. aging	, then we age will be played

- 2. Select the audio that is output from the PA jack. Choices are:
 - PA announcements Routes a PA announcement to the PA jack when a user dials *0.
 If a user has dialed *80 to route music on hold through the PA jack, and a user dials *0 to make a PA announcement, the PA announcement will interrupt the music on hold.
 - Voicemail screening Routes a call to the PA jack if a caller reaches a voice mailbox, or if a user accesses a voice mailbox. In addition, the system will route a PA announcement to the PA jack if a user dials *0.



Unlike a PA announcement, voicemail screening will not interrupt music on hold.

- Demo with router Intended for demonstration purposes only.
- Static VoIP demo Intended for demonstration purposes only.

Overhead Paging

If overhead paging results in feedback, you can enable delayed paging. The page will be played over the PA after the message has been recorded.

Check the box to enable delayed paging.

Transfer Preferences

The *Transfer Preferences* command allows you to set options for transferring callers. You can set options for calls transferred to local extensions, remote extensions and ring groups, and from auto attendants and home phones.

1. Choose Options > Transfer Preferences. The Transfer Preferences window appears.

To local extension tab

The *To Local Extension* tab allows you to set call handling for calls manually transferred to a local extension, if the extension is busy or unanswered.

1. Select the To Local Extension tab.

Transfer Preferences	X
Specify the options for transferring a caller: To Local Extension To Ring Group From Auto Attendant	_ 1
If the Extension is Busy-	
If the extension does not answer-	
O Ring back the extension that transferred the call	
O Perform the No Answer action	

- **2.** Select the option if the extension is busy:
 - *Ring back the extension that transferred the call* If the extension is busy, transfer the call back to the source extension.
 - *Perform the Busy action* If the extension is busy, perform the target extension's Busy call cascade.
- 3. Select the option if the extension is unanswered:
 - *Ring back the extension that transferred the call* If the extension is unanswered, transfer the call back to the source extension.
 - *Perform the No Answer action* If the extension is unanswered, perform the target extension's No Answer call cascade.

To remote extension tab

The *To Remote Extension* tab allows you to set call handling for calls manually transferred to a remote extension, if the extension is busy or unanswered. It also sets transfer and clear, and screened or blind transfer.

1. Select the To Remote Extension tab.

ansfer Preferences	×
Specify the options for transferring a caller:	
To Local Extension To Remote Extension To Ring Group From Auto Attendant	
rIf the extension is busy-	
Rinα back the extension that transferred the call	
O Perform the Busy action	
rlf the extension does not answer	0
Piny back the extension that transferred the cell	
O Perform the No Answer action	
L	0
Clear telephone line after call transferred	
Remote Transfer	
TransferPreferencesForm.remoteBlindTransfer.label	
O Allow screening of calls	
⊙ Perform a blind transfer	

- 2. Select the option if the remote extension is busy:
 - *Ring back the extension that transferred the call* If the remote extension is busy, transfer the call back to the source extension.
 - *Perform the Busy action* If the remote extension is busy, perform the remote extension's Busy call cascade.

- 3. Select the option if the remote extension is unanswered:
 - *Ring back the extension that transferred the call* If the remote extension is unanswered, transfer the call back to the source extension.
 - *Perform the No Answer action* If the remote extension is unanswered, perform the remote extension's No Answer call cascade.
- 4. Select the *Clear telephone line after call transferred* checkbox to clear the telephone line after the call is transferred to a remote extension. The telephone line must have the Transfer and Clear service available and selected on the *Telephone Lines* page. See "Phone line services" on page 93.
- 5. Select how the system transfers calls from a remote extension to a local extension, ring group. Choices are:
 - Allow screening of calls When the user at a remote extension transfers a call, allows them to determine whether the other user wants it.
 - *Perform a blind transfer* When the user at a remote extension transfers a call, the system immediately connects the caller to the other user and releases the telephone line connected to the remote extension.



Transferring a call from a remote extension requires telephone lines without the 3-Way Calling/Conference service or the Transfer and Clear service.

When a call is transferred from one remote extension to another remote extension, it is always a blind transfer to avoid tying up a third line.

To ring group tab

The *To Ring Group* tab allows you to set call handling for calls manually transferred to a ring group, if all the extensions are busy or unanswered.

1. Select the *To Ring Group* tab.

eony the options h	or transferring a caller:	
o Local Extension	To Remote Extension To Ring Group From Auto Attendant	
If all the extension	are busy-	
Ring back the	extension that transferred the call	
 Perform the E 	usy action	
	swers-	
If no extension an		
If no extension an	and a stand down a dama of the stand	

- 2. Select the option if all the extensions are busy:
 - *Ring back the extension that transferred the call* If all the extensions are busy, transfer the call back to the source extension.
 - *Perform the Busy action* If all the extensions are busy, perform the ring group's Busy call cascade.
- 3. Select the option if all the extensions are unanswered:
 - *Ring back the extension that transferred the call* If all the extensions are unanswered, transfer the call back to the source extension.
 - *Perform the No Answer action* If all the extensions are unanswered, perform the ring group's No Answer call cascade.

From auto attendant tab

The *From Auto Attendant* tab allows you to set the PBX extension length and disable the transfer prompts.

1. Select the From Auto Attendant tab.

pecify the options for tr	ansferring a caller	
To Local Extension	To Remote Extension To Ring Group From Auto Attendant	
If used with another I The system can act PBX extension lengt	BX	0
NUMBER OF STREET, STRE	fer Promots	-0
- Auto Attendant Trans	Del 1 Della del 1	(6)

2. If the system is used as an auto attendant for another PBX, set the *PBX extension length* to the number of digits the PBX uses for its extensions. When the caller dials 7 followed by an extension number, the unit will capture the extension number, flash the line and then dial the extension number to complete the transfer. Choices range from *not used* to 7 *digits*.



This feature only works with PBXs that use hook flash as a transfer method. Please contact your PBX manufacturer for transfer details.

3. Select the *Disable playing of transfer prompts* checkbox to disable the "*one moment please*" prompt when transferring a call.

Call Reminders

The *Call Reminders* command allows you to set reminder timers. When a reminder timer expires, the phone rings indicating a caller is on hold or parked. In addition, a queued caller hears a prompt when the *Queue options reminder timer* expires.

1. Choose Options > Call Reminders. The Call Reminder Timers window appears.

Call Reminder Timers		_	×
Call Reminder Timers			0
Hold ringback reminder timer:	90 seconds	~	
Park ringback reminder timer:	90 seconds	~	
Queue options reminder timer:	2 minutes	~	
			Use Defaults OK Cancel

- 2. Set the *Hold ringback reminder timer* to how often the phone should ring when a caller is on hold.
- **3.** Set the *Park ringback reminder timer* to how often the phone should ring when a caller is parked.
- 4. Set the Queue options reminder timer to how often a queued caller should hear a prompt.

VoIP Trunking

The VoIP Trunking command allows you to set the VoIP ports and choose the local Proxy.

1. Choose Options > VolP Trunking. The VolP Settings window appears.

JP Settings	×
VoIP ports	
SIP signalling port: 5060]
Base RTP port: 6000	
Local Proxy Unit acting as proxy: Unit 1	Change Proxy

VoIP ports

The VoIP Ports area allows you to set the SIP server port, and the starting RTP port.

- 1. The *SIP signalling port* box allows you to change the SIP server port. The default SIP server port is 5060. If required, enter a different port number ranging from 1024 to 65535.
- **2.** The default *Base RTP port* is 6000. If required, enter a different port number ranging from 1024 to 65535.

Each unit requires 8 RTP ports. The RTP ports are evenly numbered from the starting port as follows:

- unit 1 6100 to 6114
- unit 2 6200 to 6214
- unit 3 6300 to 6314
- unit 4 6400 to 6414

These are the RTP ports the system listens to for audio. If these are not opened correctly, users will not hear their callers.

3. Ensure your router is set up to perform port forwarding for the SIP signalling and RTP ports. See "Firewall settings" on page 14.

Local proxy

If your system has multiple units, the *Local Proxy* area displays and allows you to change the unit acting as local proxy. The local proxy handles Internet communications for the system. It establishes connections across the Internet, and then routes Internet data to and from each unit. The lowest numbered unit acts as local proxy by default.

The router must map the SIP, TFTP and HTTP ports to the local proxy. See "Firewall settings" on page 14.

1. If necessary, click the *Change Proxy* button, then select the unit.

Clock Preferences

The *Clock Preferences* command allows you to have the system get the time and date (but not the year or time zone) from caller ID. This allows the system to refresh the clock automatically. It is beneficial if your area has frequent power outages that cause the system time to become inaccurate. However it is not recommended if you have more than one telephone service (e.g. regular telephone and VoIP) because the telephone services could have conflicting times.

1. Choose Options > Clock Preferences. The Caller ID / Time Set Options window appears.



2. Select the Obtain and set system time from Caller ID checkbox to have the unit update its internal clock using the timestamp within caller ID. If it is not selected, set the time using the Date and Time Properties window. See "System time" on page 11.

PRI Settings

The *PRI Settings* command allows you to set the active number of channels, D channel and the Line Build Out level.

1. Choose Options > PRI Settings. The PRI Settings window appears.

PRI Settings	x
PRI Channels Number of active PRI channels: 23 🗸	0
Advanced Settings	0
Line Build Out	0
Use Defaults OK	cel

PRI Channels

Select the number of channels being supplied by your PRI service provider.

Advanced Settings

Select the signalling channel used by your PRI service provider. The default D Channel is 24 for North America, or 16 elsewhere.

Line Build Out

The *Line Build Out* area allows you to set the signal strength if connected to a service with a long circuit length. The PRI provider will provide details about the circuit length and the line build out value that should be configured. The default level is *SHORT*.

Internal Calls

The *Internal Calls* command allows you to set whether intercom calls use the call cascade of the dialed extension if the extension is busy, unanswered or in do not disturb mode. An intercom call is a call from a local extension to another local extension, remote extension or ring group.

1. Choose Options > Internal Calls. The Intercom Calls Options window appears.

Intercom Calls Options	\mathbf{X}
Intercom calls	
Intercom calls follow cascad	le settings
	Use Defaults <u>QK</u> <u>Cancel</u>

2. Select the *Intercom calls follow cascade settings* checkbox to have an intercom call use the call cascade of the dialed extension. Otherwise the intercom call will ring continuously.

Call Back

The *Call Back* command allows you to set the number of rings the unit will allow before hanging up, during an auto call back or prompted call back.

1. Choose Options > Call Back. The Call Back Options window appears.

ll Back Options	
Call Back Attempt Timer	6
During Call Back, number of rings before the system hang	s up: 6 rings 🔽
	Use Defaults <u>O</u> K <u>C</u> ancel

2. Select the number of rings the unit will allow during a call back before hanging up. Choices range from *1 ring* to *9 rings*.

Caller ID

You can turn off the missed calls indicators here. In offices with busy ring groups, the missed call indicators can be an annoyance. To disable missed call indication, uncheck the box and press *OK*.

Dial-by-name Directory

The *Dial-by-name Directory* command allows you to specify whether the dial-by-name directory search will provide results based on first, last, or both names.

The dial-by-name directory is an auto attendant option. Callers can connect to an extension using their telephone dialpad to spell the name of the party they wish to speak with.

1. Choose Options > Dial-by-name Directory. The Dial-by-name Directory window appears.

-Dial-by-name Director	у
Callers can search fo Select an option belo	or people via the dial-by-name directory. w for directory searches.
Dial-by-name search	: last name only

2. Select the desired search option. Choices include first name, last name, or both names.

Call Park

When you park a call, by default the system prompt tells you "*Call parked at...*" and the number of the park orbit. Check the *Disable 'Call Parked at' prompt* box to have the system prompt with the number of the orbit only.

Set Date & Time

The Set Date & Time command allows you to set the time, date, time zone, and NTP server. See "System time" on page 11.

Dat	e and Ti	ime Pr	opertie	es				×
	Date & Ti	me	NTP Se	rver				1
	<<]	Septe	ember	2011		>>	
	Sun	Mon	Tue	Wed	Thu	Fri	Sat	
					1	2	3	
	4	5	6	7	8	9	10	
	11	12	13	14	15	16	17	
	18	19	20	21	22	23	24	
	25	26	27	28	29	30		1
								11:18:29 AM
	Time zone	(-05:0	0) Easte	ern (US/(Canada)	•		
	✓ Autom	atically	adjust cli	ock for d	laylight :	saving	changes	
								Apply <u>OK</u> <u>C</u> ancel

Troubleshooting menu

The *Troubleshooting* menu allows access to diagnostics and advanced settings. Do not adjust these settings unless advised to do so by your local dealer or technical support centre. The default settings should provide correct operation.



If your system does not appear to be functioning properly, first ensure the country where the system operates is set correctly in the *Registration Selection* field of the *About* page, then save your configuration to the system. If this does not resolve the problem, please contact your local dealer or technical support centre for assistance.

Auto Attendants

The Auto Attendants window allows you to set auto attendant timers.



Do not adjust these settings unless directed by your local dealer or technical support centre. The default settings should provide correct operation.

If your system does not appear to be functioning properly, first ensure the country where the systemh operates is set correctly in the *Region Selection* field of the *About* page, then click the *Use Defaults* button on this window. If this does not resolve the problem, please contact your local dealer or technical support centre for assistance.

1. Choose Troubleshooting > Auto Attendants. The Auto Attendants window appears.

uto Attendants	X
Single Digit Action Timer Time allowed for pressing another digit after pressing 1,2 or 3 at the auto attendant. Setting this value to immediate means that caller will not be able to directly dial a local extension, remote extension or Ring Group. Single digit fall through time: 1.5 seconds	0
Use Defaults	

2. The Single digit fall through time sets a time-out period at the auto attendant for detection of additional digits after the first digit — e.g. an extension number. Once the wait time elapses, the auto attendant processes the received digits. Selecting *immediately* limits each entry to a single number, and prevents callers from dialing extensions directly. The other options allow callers to enter extension and ring group numbers. If the caller dials three digits, the auto attendant immediately processes the entry.

Extensions > Regular Analog Extensions

The *Extensions* > *Regular Analog Extensions* window allows you to set analog extension timers and the volume level.



Do not adjust these settings unless directed by your local dealer or technical support centre. The default settings should provide correct operation.

If your system does not appear to be functioning properly, first ensure the country where the systemh operates is set correctly in the *Region Selection* field of the *About* page, then click the *Use Defaults* button on this window. If this does not resolve the problem, please contact your local dealer or technical support centre for assistance.

1. Choose *Troubleshooting* > *Extensions* > *Regular Analog Extensions*. The *Regular Analog Extensions* window appears.

Regular Analog Extensions	×
Flash/Recall Key Operation If flash/Recall Key Operation If flash/recall signals are being missed by the phone system, you may need to adjust the detection range. Try lowering the minimum value and increasing the maximum value in increments of 100 ms until it works Minimum length: 400 ms Maximum length: 600 ms	0
Phantom Rings— If some extensions are ringing shortly after finishing a call, then increase this value until the problem stops. Ignore flash/recall signals detected if a hang-up follows the flash/recall signal within: 500 ms	0
Audio If the volume for all analog extensions is too loud or too soft, try adjusting this control. Volume heard at local extensions: Default	0
Use Defaults)	;el

- 2. The *Minimum length* sets the minimum accepted duration of a valid Flash/Recall signal from a local extension. If there are false ring backs after a call, try increasing the *Minimum length*. If the system ignores genuine Flash/Recall signals, try reducing the *Minimum length*.
- **3.** The *Maximum length* sets the maximum accepted duration of a valid Flash/Recall signal from a local extension. If the system disconnects on genuine Flash/Recall signals, try increasing the *Maximum length*. If some calls are placed on hold when the user hangs up and quickly places a subsequent call, try reducing the *Maximum length*.
- 4. The *Phantom Rings* timer is used to ignore false Flash/Recall signals before the end of a call. If extensions experience false ring backs after users hang up, try increasing the *Ignore flash/recall signals* timer.
- 5. The *Volume heard at local extensions* setting controls the inbound volume level for calls and system prompts heard on extensions.

Extensions > IP Extensions

The *Extensions > IP Extensions* window allows you to set the amount of time between registration messages from IP extensions. You can also change the Starting SIP, Base RTP and HTTP ports.



Do not change the following controls unless directed by your local dealer or technical support centre.

Extensions	
Internal Registration Interval If your internal IP extensions are having re-registration problems, then try adjusting this value. Internal IP extension registration intervat External Registration Interval If your external IP extensions are having trouble re-registering with the phone system, then try adjusting this value. WARNINGI Increasing this value may affect keeping ports open on the router/firewall for calls. We recommend you only increase the value if VoIP ports are manually mapped from your router/firewall to the system. External IP extension registration interval Source State St	HTTP Port Change ports only if there is a conflict with other devices. HTTP port: 8484 IP Extension VolP Ports Change ports only if there is a conflict with other devices. Starting SIP port: 5000 Base RTP port: 4000
Line Appearance Optimization	Time Format Choose the time format to be displayed on your phones. Time Format: 12 Hour Phone Configuration View Phone Configuration Files
External IP Extension Codec Options If your internet connection cannot support calls using the G.711 codec (100 kbps bandwidth), then license to have access to the G.729 codec. G.729 V G.711µ G.711A Preferred codec: G.711µ V	enable the G.729 (25 <i>kbps bandwidth</i>) codec. You must have purchased the VoIP Trunking

Internal Registration Interval

The interval for registration messages from IP extensions within the office.

External Registration Interval

The interval for registration messages from IP extensions outside the office.

Line Appearance Optimization

Activate this if your 350i, 450i and/or 550i phones are experiencing line appearance inconsistencies.

External IP Extension Codec Options

Control the audio quality of external IP extensions. For more information on codecs, see "Setting codec options" on page 31.

HTTP Port

The default HTTP port is 8484. If required, enter a different port number ranging from 1024 to 65535. If your system will use a proprietary IP phone as an external IP extension, ensure the router maps this port to the unit acting as local proxy.

IP Extension VoIP Ports

The default Starting SIP port is 5000. If required, enter a different port ranging from 1024 to 65535.

The default Base RTP port is 4000. If required, enter a different port number ranging from 1024 to 65535.

Time Format

Choose the time format to be displayed on your phones, either 12 Hour (default) or 24 Hour.

Phone Configuration

Click to open a browser with detailed configuration files for each IP extension.

Fax Detection

The Fax Detection window allows you to set up detection of Fax Over IP and CNG tones.



Do not change the following controls unless directed by your local dealer or technical support centre.

1. Choose *Troubleshooting > Fax Detection*. The *Fax Detection* window appears.



- 2. Select the Automatically detect and switch to G.711 for fax over IP checkbox to have the unit detect fax over IP and then switch to G.711.
- **3.** Set the number of CNG tones the unit must detect before the auto attendant routes the call to the fax machine. Choices are:
 - Detect 1 tone The auto attendant will route the call to the fax machine if the unit detects one CNG tone. Select this if fax calls are not being routed to the fax machine.
 - Detect 2 tones The auto attendant will route the call to the fax machine if the unit detects two CNG tones. Select this if regular calls are being routed to the fax machine.

Telephone Lines > Audio

The *Telephone Lines > Audio* window allows you to automatically calibrate the unit to match the telephone lines and select the volume level on the lines.

For the best call audio quality and volume levels, your system parameters must be matched to your telephone lines. The *Telephone Line Audio Optimization* area offers calibration to match your telephone lines.

1. Choose *Troubleshooting* > *Telephone Lines* > *Audio*. The *Telephone Lines* - *Audio* window appears.

If called button the line If you e proper and if y	rs compi below. I volume encounte ly calibra you still h	am that your phone volume f further adjustment is nee er echo problems when on ted to match the impedance nave a problem, contact Te	e is too quiet or too loud, use the Uptimiz ded, use the Transmit level menu below calls, some line ports on the system ma ce of the lines. Please run the optimizatio likSwitch Support for further instructions	to adjust ny not be n routine s.
Unit	Line	Phone Number	Optim Transmit level to callers	nize Lines
1	1		Default	~

2. Optimize the unit to match the far end impedance of the telephone lines. Click *Optimize Lines*. The *Optimize Lines* window appears. Click *Run Optimization*.

Optimize Lines	×
Select line(s) for optimization:	
Unit 1 Line 1 (OPTIMIZED)	
	Run Optimization Cancel

Telephone Lines > Detection and Timers

The *Telephone Lines > Detection and Timers* window allows you to set telephone line timers, and perform the ring pattern test.



Do not adjust these settings unless directed by your local dealer or technical support centre. The default settings should provide correct operation.

If your system does not appear to be functioning properly, first ensure the country where the system operates is set correctly in the *Region Selection* field of the *About* page, then click the *Use Defaults* button on this window. If this does not resolve the problem, please contact your local dealer or technical support centre for assistance.

elephone Lines - Detection and Timers	
Line Disconnect- If you encounter problems where the line is staying off hook long after a call is completed, then try lowering the CPC timer in 50 ms increments. If calls are being accidentally disconnected, try raising this value in 50 ms increments. Adjust and test until the problem is resolved. 250 ms Ring Detection This test will measure the ring cadence of an incoming call. Ring Pattern Test	Call Waiting and Other Flash/Recall Based Services () If using Flash/Recall to engage telephone company features and features are not being engaged, then the transmitted flash timing may need to be increased or decreased. Adjust in 50 ms and re-test until the problem is resolved. 450 ms
	Use Defaults QK <u>C</u> ancel

The *Line Disconnect* timer controls the duration a telephone line may be disconnected before the system clears the telephone call. In some cases this may be used by the telephone network to indicate a calling party has hung up and the call has ended (also known as Disconnect Clear, K-Clear or CPC signal). Incidents of disconnected calls may indicate a faulty telephone line, or may be resolved with a longer *Line disconnect* time setting.

To test the ring pattern for incoming calls on telephone lines:

- 1. Click *Ring Pattern Test*. The *Ring Pattern Test* window appears.
- 2. Click *Test Ring Pattern*. Call one of the telephone lines. After three rings the window will adjust to reflect the results of the test. The time is displayed in milliseconds.



The *Call waiting and other Flash/Recall Based Services* timer sets the time required for a Flash/Recall signal to activate calling features on a telephone line. If a Flash/Recall signal is not successfully engaging telephone company features, try adjusting the timer.

The 3-way calling/Centrex Transfer timer sets the required delay after a Flash/Recall signal before the first digit of a telephone number is dialed. This setting may need to be increased if same line connect calls are misdirected and the initial digits are not detected by the telephone company. The same line connect feature may not be compatible with some telephone companys' 3-way calling features.

VolP

The *VoIP* window permits adjustment of ports, timers and network-related settings affecting VoIP communications.



Do not change the following controls unless directed by your local dealer or technical support centre.

VoIP ports

The SIP server port and starting RTP port can be specified for the SIP server.

UPnP

By default, the system uses Universal Plug'n Play (UPnP) to attempt to automatically open the required ports for VoIP communication through supported firewalls. By disabling this feature, the ports must be manually configured within your firewall in order for VoIP communication to operate.

For information on configuring your router ports for VoIP service providers and external IP extensions, see "Firewall settings" on page 14.

Global Dial Plan Codec Options

Control the audio quality for your multiple locations. For more information on codecs, see "Setting codec options" on page 31.

Firewall Test

The *Troubleshooting > Firewall Test* window allows you to test the status of each port through the firewall. See "Firewall settings" on page 14.

1. Choose *Troubleshooting > Firewall Test*. The *Firewall Test* window appears.

Internet Access	RTD Audio (Lipit 2		
U HTTD (VML Convice)	Im RTP Audio (Unit 2,	Port 4)	
✓] HTTP (XIVE Service)	RTP Audio (Unit 2,	Port 5)	
RTP Audio (Unit 1, Port 1)	🗹 RTP Audio (Unit 2,	Port 6)	
RTP Audio (Unit 1, Port 2)	RTP Audio (Unit 2,	Port 7)	
RTP Audio (Unit 1, Port 3)	RTP Audio (Unit 2,	Port 8)	
RTP Audio (Unit 1, Port 4)	SIP (VolP signalling	1)	
RTP Audio (Unit 2, Port 1)	TFTP (IP Phone Cor	nfiguration)	
RTP Audio (Unit 2, Port 2)			
RTP Audio (Unit 2, Port 3)			

- 2. Select the services you want to check.
- 3. Click *Test Ports*. The system will check the ports of the selected services.

System Information and Diagnostics

This window provides system information and diagnostics for the system as a whole, for local extensions, VoIP numbers, your network and memory usage.

tem Information an	d Diagnostics				
System Information Local Extensions VoP Numbers Discovered Network Memory Usage					
Information					
Unit ID	Hardware Type	MAC Address	IP Address	Firmware Version	Uptime
Unit 1	848 VS	000F4D0003F1	192.168.1.103	7.00.000	7 sec
-Language					
Languages installed o	on the system: English				
	French				
	Spanish				
Default system pron	npts to callers: English				
System Information-					
System time: S	September 19, 2011 / 3:14:30 PM				
Application version: 7	7.00.000				
Last config change: S	September 19, 2011 / 2:57:00 PM				
Connection type: I	P				
Region: 1	North America				
				Export	Reload Close

System Information

The System Information tab includes general system information, MAC and IP addresses, system ID software and firmware versions, languages and uptime.

Local Extensions

The *Local Extensions* tab includes status, type, MAC address, IP address and firmware for each extension.

VoIP Numbers

The *VoIP Numbers* tab includes service provider profile information and username, password and status for VoIP numbers.

Network

The *Network* tab includes LAN and WAN configuration details, gateway device information and port configuration details. You can run the Check Firewall test from this tab.

Memory Usage

The *Memory Usage* tab includes memory statistics by function and voicemail memory usage by mailbox.

Support Tools > Call Logging Options



This feature is intended for the exclusive use of technical support personnel. Do not use this feature unless directed by your local dealer or technical support centre.

The *Troubleshooting* > *Support Tools* > *Call Logging Options* window provides diagnostic logging tools to capture call data and phone system events. These tools are intended for the exclusive use of technical support personnel.

1. Choose *Troubleshooting > Support Tools > Call Logging Options*. The *Logging* window appears.



2. Select the *SIP Trace Capturing* url to open a browser interface to permit detailed logging of SIP messages for calls to IP phones and/or VoIP lines.

	Web Admin	
Home	Message Log	
SIP Status	Mer Time En/Dis Remote ID Address Remote Dert	
RTP Status	SIP 9394	
UPnP Status		
Trace Info	Enabled Message Type	
	✓ INVITE_REQUEST	
	ACK_REQUEST	
	REGISTER_REQUEST	
	BYE_REQUEST	
	OPTIONS_REQUEST	
	□ INFO_REQUEST	
	CANCEL_REQUEST	

3. Select the *CP Logging* url to open a browser interface to permit detailed logging of phone system events.

		,	Web Admin		
Home IP Status SIP Status RTP Status UPnP Status Trace Info	Trace Modules WARNING: This disruption and/or <u>Callp Log Setup</u> UDP Monitoring Rem IP Addres Enable all module will be disabled) [Major 8 Critica] Core Modules	is for authorized and resets to the system a s Rem Port En/1 9395 s (on all boxes) and tu	rained users only. The misuse of this ad should be used with caution. Dis [save UDP settings] m on critical and major levels only (r	: feature can cause ninor and trace levels	
	Name	Levels Ena	ble all/Disable all		
	DTD	Ch An	[

4. Click OK to exit the Logging window.

Support Tools > Email Software Configuration Logs

The *Troubleshooting* > Support Tools > Email Software Configuration Logs window allows you to automatically e-mail log files to Customer Support. Only use this feature as directed, after receiving a Case ID from Customer Support.

Automatically e-mailing log files requires Microsoft Outlook to be configured as your default e-mail client. Note that the e-mail will only include information related to your system.

1. Choose *Troubleshooting* > *Support Tools* > *Email Software Configuration Logs*. The *Support* window appears.

Support	X
Support Select the files you would like to send to support. Configuration File Log Files	0
Next Canc	xel

- 2. Select the files you would like to send to support. Click Next.
- 3. Enter the Case ID supplied by Technical Support.

Support X
Support-O
FortiVoice Management Software will capture your current system information and generate an email to Technical Support.
Note: No personal information is captured in the file.
Case ID provided by Technical Support
To use an email client other than your default client, please send an email to <u>fv_support@fortinet.com</u> with a description of the case scenario and attach the file at the link below:
rtiVoice Management 7.00%Logs\TSPackage_7.00001_1318357166554.zip
Next Cancel

If you use a different e-mail client, or don't have one configured (e.g. you use a web-based e-mail service like Hotmail) you can manually e-mail the log files. Enter the Case ID in the Subject line. Attach the .zip file listed in the *Support* window.

Using your phone system

Receiving calls

With the auto attendant

If you don't have a receptionist, the auto attendant helps you handle all your inbound calls. Once an auto attendant has been configured and recorded, it will follow the instructions given in the configuration.

When calls are answered by an auto attendant, callers can:

- Dial any local or remote extension
- Dial any ring group
- Dial voicemail directly
- Use the dial-by-name directory
- Choose options 0 (9 in some regions), 1, 2, 3, 4, 5 (if configured to perform specific actions)
- Transmit a fax
- Go to a different auto attendant
- Remain on the line to follow an action programmed for that auto attendant

System users can access other options at the auto attendant:

- Call back or call bridge
- Retrieve voicemail
- Enter command mode to make system changes

All these features can be password-protected to avoid unauthorized access.

The system also provides the option of ringing specific extensions or ring groups prior to the engagement of an auto attendant.

Using an analog extension

This section describes how to use a third party analog phone with the system.

Making calls

Placing an intercom call

An intercom call is a call from a local extension to another local extension, remote extension or ring group.

1. Dial the extension number.

If the *Intercom calls follow cascade settings* checkbox is selected in the configuration, an intercom call will follow the call cascade of the dialed extension. Otherwise the intercom call will ring continuously.

The dialed extension rings as follows:

- A local extension uses two short rings to identify the intercom call.
- A remote extension rings normally.
- A ring group uses the ring pattern selected in the configuration. The ring group uses the same ring pattern for intercom calls and outside calls.

Overhead paging

Overhead paging is a call from a local extension through the optional PA amplifier connected to the phone system.

1. Dial *0.

Two-way paging

Two-way paging is a call from a local extension to another local extension that activates the speakerphone of the dialed extension. The user at the dialed extension can reply through the speakerphone. You can perform two-way paging from a local extension to a FortiFone telephone.

- **1.** Dial *84.
- 2. Dial the extension number.

One-way group paging

One-way group paging is a call from a local extension to another local extension or to a ring/page group that activates the speakerphones of the dialed extensions. The users at the dialed extensions can't reply through the speakerphone. You can perform one-way group paging from a local extension to FortiFone telephones.

- 1. Dial *85.
- **2.** Dial the extension or ring group number.

Place out-of-office calls from a local extension

- 1. Dial 9 (0 in some regions) or 81–88 to choose a hunt group. The hunt group gives you access to an available line to dial out.
- 2. If your system requires permissions for outgoing calls, enter your access code.
- **3.** Dial the phone number.

Extensions can be restricted to specific hunt groups by using User Privileges. See "Routing and blocking" on page 36.

Your extension can also be configured with direct line access. When you pick up the handset, you hear the telephone company dial tone and can dial an outside number without dialing a hunt group number. See "Direct line access" on page 48.



If direct line access is enabled for a local extension, you need to press *Flash* or *Recall* before dialing any extensions or voice mailboxes.

If you dial your own extension you will be directed to your voicemail. If your voicemail is not set up, you will get a busy signal.

Click-to-Dial from Microsoft Outlook

Dialing directly from Outlook requires the Click-to-Dial utility to be installed on your computer. Visit http://support.fortinet.com to download and install the software.

To call a contact from Outlook, follow the Outlook instructions for initiating calls.

Receiving calls

The steps below apply when an analog extension is ringing. If an auto attendant engages before you are able to pick up, the caller can select an option at the auto attendant.

For your local extension

You can answer a call that is ringing your local extension.

- 1. If the call went through the auto attendant and call screening is enabled:
 - Press # to accept the call.
 - Press * to reject and re-route the call to its extension's call cascade, and then hang up.

For any other local extension

You can intercept a call that is ringing another local extension.

1. Press *9.

For a specific local extension

You can intercept a call that is ringing a specific extension.

- **1.** Press *7.
- 2. Dial the extension number.

While you're on a call

You can intercept a call while connected to another caller.

- 1. Press Flash or Recall to place the first call on hold.
- 2. Press *9, or *7 and the extension number to intercept the second call.
- **3.** Once you are finished with the second call or would like to return to the first caller, press *Flash 5* or *Recall 5*.

Placing calls on hold

- 1. Press Flash or Recall to place the call on hold.
- 2. Press Flash or Recall again to take the call off hold.
- **3.** If you have multiple callers on hold or in queue at your extension, press *Flash 7* or *Recall 7* to retrieve callers on a first in/first out basis.

If you put a caller on hold using the *Hold* button on some third party analog phones, you won't be able to access the phone system features while the caller is holding, and the caller will not hear music while on hold.

While a call is on hold, your phone will ring once each time the reminder timer expires. See "Call Reminders" on page 135.

Transferring calls to an extension

You can transfer a call from a local extension to another local extension, remote extension or ring group.

Unscreened transfer

- 1. Press Flash or Recall to put the caller on hold.
- 2. Dial the extension you want to transfer to, and then hang up. The caller is taken off hold when the receiving extension answers. If the extension is busy or the call not answered, the system can either perform the call cascade's Busy action or No Answer action, or can ring back the extension that transferred the call. See "To local extension tab" on page 133.

Screened transfer

- 1. Press *Flash* or *Recall* to put the caller on hold.
- 2. Dial the extension number you want to transfer to, and wait for them to answer.
 - If the person at the receiving extension wishes to take the call, complete the transfer by hanging up or pressing *Flash 4* or *Recall 4*. The caller is immediately taken off hold and connected to the receiving extension. You hear an internal dial tone.
 - If the person at the receiving extension does not wish to take the call, return to the caller by pressing *Flash 5* or *Recall 5*.
 - If you dial an extension that is busy or not answered, you can complete the transfer by hanging up or pressing *Flash 4* or *Recall 4*. The call will be routed according to that extension's call cascade settings. You hear an internal dial tone.



If you handle many calls, you may want to complete or cancel a transfer without hanging up. *Flash 4* or *Recall 4* will complete the transfer and provide an internal dial tone. *Flash 5* or *Recall 5* cancels the transfer and connects you back to the caller.

Transferring calls to an outside number

You can transfer a call from a local extension to an outside number.

If the telephone line has the 3-Way Calling/Conference service or the Transfer and Clear service, and the local extension has enabled *Allow the use of same line connect (by dialing 80)*, the user can select hunt group 80 to dial the outside number on the same line the caller is on.

Unscreened transfer

- 1. Press Flash or Recall to put the caller on hold.
- **2.** Dial 9 (0 in some regions) or 81–88 to choose a hunt group.
- 3. Dial the outside phone number.
- 4. Press *Flash 4* or *Recall 4* to complete the transfer. Do not hang up to complete the transfer, because this disconnects the call.

Screened transfer

Instead of completing the transfer once you have dialed the outside number, wait until the call is answered and ask the recipient if he or she wishes to take the call. If the recipient accepts the call, press *Flash 4* or *Recall 4* to complete the transfer. If not, press *Flash 5* or *Recall 5* to return to the caller.

Parking and retrieving calls

Call park is a feature for placing a call on hold and then retrieving it from another local extension. The system has 10 park orbits (500–509).

Parking a call

- 1. Press *Flash* or *Recall* to put the caller on hold.
- 2. Press *510 or the *Park* button or softkey. The system selects the first available park orbit (500–509).
- **3.** Provide the park orbit to the person with the parked call (e.g. *"Mary, there is a call parked for you in 500"*).

While a call is parked, your phone will ring once each time the reminder timer expires. See "Call Reminders" on page 135.

Retrieving a parked call

- 1. If direct line access is enabled at your extension, press *Flash* or *Recall* to get an internal dial tone.
- 2. Press ** or the Un-park button or softkey and the appropriate park orbit.

Queuing and retrieving callers

Call queue is a useful feature for placing multiple callers on hold at your extension while you are on an existing call.

Queuing callers at a local extension

If an extension is busy and the *Busy* call cascade is set to *queue at extension*, incoming callers hear one of the prompts below, followed by music (if enabled) while they are on hold:

- If the call came from the auto attendant and the extension has a voice mailbox: "The extension you have reached is currently busy. If you wish to continue holding, please remain on the line. To leave a voicemail message, press 1, and to return to the main menu, press star."
- If the call came from the auto attendant and the extension has no voice mailbox: "The extension you have reached is currently busy. If you wish to continue holding, please remain on the line. To return to the main menu, press star."

- If the call was transferred from another extension and the extension has a voice mailbox, but there is no auto attendant on this line with a main menu to return to: "The extension you have reached is currently busy. If you wish to continue holding, please remain on the line. To leave a voicemail message, press one."
- If the call was transferred from another extension and the extension has no voice mailbox and there is no auto attendant on this line with a main menu to return to: *"The extension you have reached is currently busy. If you wish to continue holding, please remain on the line."*

While the caller is on hold at your extension, you hear a call waiting beep every 75 seconds. If you want to rotate through queued callers, press *Flash 7* or *Recall 7*. The caller you are speaking to is placed on hold, and the first caller in the queue is retrieved. If you have multiple callers queued, press *Flash 7* or *Recall 7* to retrieve callers on a first in/first out basis.

If an extension is busy and the *Busy* call cascade is set to *invoke call waiting*, callers hear ringing (or music if enabled) while the local extension hears a call waiting beep and receives the caller ID for the new call. To connect to the new caller, press *Flash 7* or *Recall 7*. To let the caller fall through to the next level of the cascade, ignore the beep.

Queuing callers at a ring group

If you want to queue callers at a ring group, configure this action using an auto attendant.

For example, configure auto attendant 1 to queue callers to ring group 301 for technical support if they have pressed 5. When a caller presses 5 for technical support, they enter the queue immediately.

Every 60 seconds (configurable) the caller hears the following prompt: "Your call is important to us. Please remain on the line. To return to the previous menu, press star." Extensions that are part of the ring group and are available will start ringing within 10 seconds of hanging up the phone from a previous call. The delay allows time to make a new call or activate *Do not Disturb* as a log out option. After 5 extension rings, available phones will ring once every 16 seconds. When you pick up, you hear a prompt that provides the option to accept the queued call by pressing # or leave the caller in the queue by pressing * and returning to the dial tone to make a new call.

Using call waiting

The call waiting feature is activated by setting the *If extension is busy* list to *invoke call waiting*. This control is in the *Busy* tab of the *Local Extensions/Fax* page. When the dialed local extension is busy, the caller hears ringing or music depending on the settings.

If the phone supports call waiting caller ID (type II signaling), you hear a beep followed by a short blip. Depending on your phone, you may hear something similar to a modem noise. This is normal. During this notification, the caller ID information is displayed on the phone. The caller does not hear the beep. The phone mutes the other side.

If the phone does not support call waiting and caller ID, you hear a beep. When the call waiting notification beep is heard, you can either press *Flash 7* or *Recall 7* to put the current caller on hold and answer the new call, or press *Flash 5* or *Recall 5* to terminate the current call and answer the new call. If you do not want to answer the new call, ignore the call waiting beeps. The system will direct the caller to the next level of the call cascade for your extension.

Conference calling

You can set up conference calling between three local extensions, two local extensions and one outside caller, or two outside callers and one local extension.

Two local extensions and one outside caller

You do not require access to the telephone company's 3-Way Calling/Conference service to use the conference calling capabilities.

- **1.** Establish a call with an outside party.
- 2. Press Flash or Recall to place the outside caller on hold.
- 3. Dial the number of the local extension you wish to conference with.
- 4. When the extension is picked up, press Flash 6 or Recall 6 to establish the 3-way call.

Two outside callers and one local extension

There are two different methods for this type of conference call. The first method is similar to the one above, using only the phone system for the conference function.

- **1.** Establish a call with an outside party.
- 2. Place the outside caller on hold.
- 3. Dial 9 (0 in some regions) or 81–88 to choose a hunt group.
- 4. Dial the number of the other outside party.
- 5. When the call is answered, press Flash 6 or Recall 6 to establish the 3-way call.

The second method requires the same line connect feature and uses the telephone company's 3-Way Calling/Conference service.

- **1.** Establish a call with an outside party.
- 2. Place the outside caller on hold.
- 3. Dial 80 to choose same line connect.
- 4. Dial the number of the other outside party.
- 5. When the call is answered, press *Flash* 6 or *Recall* 6 to establish the 3-way call.



The conference initiator can disengage the second conferenced party at any time by pressing *Flash 5* or *Recall 5*.

Call Barge

Call Barge enables a user to join another user's call. This feature is restricted to calls connected via PSTN telephone line to an outside party.

- 1. Dial *82.
- 2. Enter the local extension you wish to join.
- 3. Enter your privilege PIN.

All 3 parties are now joined in a conference.



Call Barge must be enabled in *User Privileges* and requires temporary access to a conference room.

Using an IP extension

This section describes how to use a third party IP phone with the system.

Making calls

Placing an intercom call

An intercom call is a call from a local extension to another local extension, remote extension or ring group.

1. Dial the extension number.

If the *Intercom calls follow cascade settings* checkbox is selected in the configuration, an intercom call will follow the call cascade of the dialed extension. Otherwise the intercom call will ring continuously.

Overhead paging

Overhead paging is a call from a local extension through the optional PA amplifier connected to the phone system.

1. Dial *0.

Two-way paging

Two-way paging is a call from a local extension to another local extension that activates the speakerphone of the dialed extension. The user at the dialed extension can reply through the speakerphone. You can perform two-way paging from a local extension to a FortiFone telephone.

- 1. Dial *84.
- 2. Dial the extension number.

One-way group paging

One-way group paging is a call from a local extension to another local extension or to a ring/page group that activates the speakerphones of the dialed extensions. The users at the dialed extensions can't reply through the speakerphone. You can perform one-way group paging from a local extension to FortiFone telephones.

- 1. Dial *85.
- **2.** Dial the extension or ring group number.

Place out-of-office calls from a local extension



Using an external IP extension to call an emergency service number will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP extension stating:

If an emergency call is made from this phone, you must provide your address to the emergency operator.

- 1. Dial 9 (0 in some regions) or 81–88 to choose a hunt group. The hunt group gives you access to an available line to dial out.
- 2. If your system requires permissions for outgoing calls, enter your access code.
- **3.** Dial the phone number.
- 4. Depending on your IP phone, press Dial, Send or #.

Extensions can be restricted to specific hunt groups by using User Privileges. See "Routing and blocking" on page 36.

Your extension can also be configured with direct line access. When you pick up the handset, you hear the telephone company dial tone and can dial an outside number without dialing a hunt group number. See "Direct line access" on page 48.



If direct line access is enabled for a local IP extension, you need to press ** before dialing any extensions or voice mailboxes.

Receiving calls

If an auto attendant engages before you are able to pick up, the caller can select an option at the auto attendant. The steps below apply when you pick up an extension that is ringing.

For your local extension

You can answer a call that is ringing your local extension.

- 1. If the call went through the auto attendant and call screening is enabled:
 - Press #, then press Dial, Send or # to accept the call.
 - Press *, then press *Dial, Send* or # to reject and re-route the call to its extension's call cascade.

For any other local extension

You can intercept a call that is ringing on any other local extension.

1. Press *9, then press *Dial*, *Send* or #.

For a specific local extension

You can answer a call that is ringing at a specific extension.

- 1. Press *7.
- 2. Dial the extension, then press Dial, Send or #.

While you're on a call

You can intercept a call while connected to another caller.

- 1. Press *Hold* to place the first call on hold.
- 2. Press *9, or *7 and the extension number, then press Dial, Send or #.
- **3.** Once you are finished with the second call or would like to return to the first caller, press *Cancel*.

Placing calls on hold

- 1. Press Hold.
- **2.** If you have multiple callers on hold or in queue at your extension, you can retrieve them on a first in/first out basis. Depending on your phone, you can either:
 - Press the flashing line key that corresponds to the caller on hold.
 - Press Hold and 7, then press Dial, Send or #.

Transferring calls to an extension

Unscreened transfer

- **1.** Press *Transfer* or *xfer* to put the caller on hold.
- **2.** Dial the extension number.
- **3.** Press *Transfer* or *xfer*. If the IP phone supports the hang-up transfer feature, hang up the phone. Please check the documentation of the IP phone to see if it supports this feature.

The caller is taken off hold when the receiving extension answers. If the extension is busy or the call not answered, the system can perform the call cascade's Busy action or No Answer action, or can ring back the extension that transferred the call. See "Transfer Preferences" on page 132.

Screened transfer

To initiate a screened transfer from an IP extension to another local extension, remote extension or ring group:

- 1. Press *Transfer* or *xfer* to put the caller on hold.
- 2. Dial the extension number.
 - If the person at the receiving extension wishes to take the call, press *Transfer* or *xfer* to complete the transfer. If the IP phone supports the hang-up transfer feature, hang up the phone. Please check the documentation of the IP phone to see if it supports this feature. The caller is immediately taken off hold and connected to the receiving extension.
 - If the person at the receiving extension does not wish to take the call and you want to return to the caller, press *Cancel*. If this softkey is unavailable, press the flashing line button that corresponds to the caller on hold. You can also press *Hold* and *7*, then press *Dial*, *Send* or #.
 - If you dial an extension that is busy or not answered, you can complete the transfer. The caller will be directed according to that extension's call cascade settings.

Transferring calls to an outside number

You can initiate a transfer from an IP extension to an outside number.

If the telephone line has the 3-Way Calling/Conference service or the Transfer and Clear service, and the local extension has enabled *Allow the use of same line connect (by dialing 80)*, the user can select hunt group 80 to dial the outside number on the same line the caller is on.

Unscreened transfer

- 1. Press *Transfer* or *xfer* to put the caller on hold.
- 2. Dial 9 (0 in some regions) or 81–88 to choose a hunt group.
- **3.** Dial the outside phone number.
- **4.** Press *Transfer* or *xfer* to complete the transfer. Do not hang up to complete the transfer, because this disconnects the call.

Screened transfer

- 1. Press *Transfer* or *xfer* to put the caller on hold.
- 2. Dial 9 (0 in some regions) or 81–88 to choose a hunt group.
- **3.** Dial the outside phone number.
- 4. Wait until the call is answered and ask the recipient if he or she wishes to take the call. If the recipient accepts the call, press *Transfer* or *xfer* to complete the transfer. If not, press *Cancel* to return to the caller. If this softkey is unavailable, press the flashing line button that corresponds to the caller on hold.

Call park – parking and retrieving callers

Call park is a feature for placing a call on hold and then retrieving it from any other local extension. The system has 10 park orbits, 500–509.

Parking a caller

- 1. Press *Transfer* or *xfer* to put the caller on hold.
- **2.** Press *510, then press *Dial, Send* or #. The system selects the first available park orbit (500–509). You hear a confirmation indicating the caller has been parked successfully and into which park orbit.
- **3.** Provide the park orbit to the person with the parked call (e.g. *"Mary, there is a call parked for you in 500"*).

Retrieving a parked call

1. Press **, the park orbit, then press *Dial, Send* or #.

Queuing and retrieving callers

This feature is used the same way on analog and IP extensions except for the slight differences outlined in the next paragraph.

Some IP extensions display a notification on the screen while the caller is on hold at your extension. If you want to rotate through queued callers while you are on the phone, dial 7 and press *Dial, Send* or #. The caller you are speaking to is placed on hold, and the first caller in the queue is retrieved. If several callers are queued, dial 7 and press *Dial, Send* or # to retrieve them on a first in/first out basis.

Using call waiting

If the call waiting feature is activated as a *Busy* option in the call cascade routing for a local extension and the dialed extension is in use, the caller hears ringing or music depending on the music on hold settings. You hear the following:

- If the phone supports call waiting caller ID, you hear a beep followed by a short blip.
- If the phone does not support call waiting caller ID, you hear a beep. When the call waiting notification beep is heard, you can pick up another line and dial 7 and press *Dial, Send* or # to put the current caller on hold and answer the new caller. To return to the previous call, press the flashing line key that corresponds to the previous call. On most IP extensions, this will put the current call on hold and connect to the other caller.
- To terminate the current call and go back to the other caller on hold, hang up, then go off-hook and press the button that corresponds to the flashing line of the caller on hold. If you do not want to take the new call, ignore the call waiting beeps. The system will direct the caller to the next level of the call cascade for that extension (i.e. send the caller to the associated voice mailbox).

Conference calling



If you try to initiate a conference call with a 9112i IP phone, but there is no line available, you will not hear a busy signal. You will instead be reconnected with the other caller.

Two local extensions and one outside caller

You do not require access to the telephone company's 3-Way Calling/Conference service to use the conference calling capabilities.

- **1.** Establish a call with an outside party.
- 2. Press *Conf* or *Hold* to place the outside caller on hold.
- **3.** Dial the local extension you wish to conference with.
- 4. When the extension answers, press *Conf* to establish the 3-way call.

Two outside callers and one local extension

There are two different methods for this type of conference call. The first method is similar to the one above, using only the phone system for the conference function.

- **1.** Establish a call with an outside party.
- 2. Press *Conf* or *Hold* to place the outside caller on hold.
- **3.** Dial 9 (0 in some regions) or 81–88 to choose a hunt group.
- 4. Dial the number of the other outside party.
- 5. When the call is answered, press *Conf* to establish the 3-way call.

The second method uses the same line connect feature and the telephone company's 3-Way Calling/Conference service.

- 1. Establish a call with an outside party.
- 2. Press *Hold* to place the outside caller on hold.
- 3. Dial 80 to choose the hunt group.
- 4. Dial the number of the other outside party and press *Dial, Send* or #.
- 5. When the call is answered, press *Transfer* to establish the 3-way call.

Call Barge

Call Barge enables a user to join another user's call. This feature is restricted to calls connected via PSTN telephone line to an outside party.

- 1. Dial *82.
- 2. Enter the local extension you wish to join.
- **3.** Enter your privilege PIN.

All 3 parties are now joined in a conference.



Call Barge must be enabled in *User Privileges* and requires temporary access to a conference room.

Programmable phone key functions

FortiFone IP phones have programmable keys. The function and associated resources are assigned using the Management software in the *Local Extensions/Fax > IP Extension Details > Configure Keys* page. Supported functions for a phone model typically include most of the items listed below.

• *Line appearance* — Select a telephone line or VoIP number as the resource. The corresponding button or softkey will display the status of the line, and allow you to make calls with a single press of the button or softkey.

For phones equipped with button lights, the button will light up when the line is in use, flash if the line is ringing, or be off when the line is available.

For phones with softkeys, a status icon and line ID will be displayed beside the softkey. The display will show an off-hook icon when the line is in use, show a ringing icon when the line is ringing, and show an on-hook icon when the line is available.

• *Extension appearance* — Select a local extension as the resource.

The corresponding button or softkey will display the status of the selected extension, and allow you to call the extension with a single press of the button or softkey.

For phones equipped with button lights, the button will light up when the selected extension is in use, flash if the extension is ringing, or be off when the extension is available.

For phones with softkeys, a status icon and extension call ID will be displayed beside the softkey. The display will show an off-hook icon when the selected extension is in use, show a ringing icon when the extension is ringing, and show an on-hook icon when the extension is available. The extension call ID is also be displayed.

 Queue appearance — Select a local extension as the resource. The corresponding button or softkey will indicate whether calls are queued at the selected extension, and allow you to pick up the oldest queued call with a single press of the button or softkey.

For phones equipped with button lights, the button will flash if calls are queued for the extension, or be off when there are no queued calls.

For phones with softkeys, a status icon, extension number and "Q" will be displayed beside the softkey. The display will show a ringing icon if calls are queued for the extension, or an on-hook icon when there are no queued calls.

- Voicemail Do not select a resource. Press the button or softkey to access the voice mailbox of the . Lights or icons are not used for voicemail keys. Note: You can also access Voicemail by pressing **#.
- DND Do not select a resource. Press the button or softkey to toggle Do Not Disturb mode on or off. Lights or icons are not used for DND keys. Note: You can also toggle DND mode by pressing *62#.
- Park Do not select a resource. Press the button or softkey to put the call on hold, in the next available park orbit. The system will respond with the park orbit number (500 to 509). Lights or icons are not used for Park keys. Note: You can also Park a call by pressing
 *510# on a 350i, or + *510# on a 450i or 550i.
- Un-park Do not select a resource. Press the button or softkey, select the park orbit number (500 to 509), then press Unpark to retrieve the call. Softkey displays will show the parked call ID beside the orbit number. Lights or icons are not used for Unpark keys. Note: You can also Unpark a call by lifting the handset, pressing **, dialing the park orbit number, then pressing #.
- Pickup any Do not select a resource. Press the button or softkey to answer a call from an outside number ringing any extension. Lights or icons are not used for Pickup keys. Note: You can also pick up a call by pressing *9#.

- *Pickup ext* Do not select a resource. Press the button or softkey, dial an extension, then dial # (or softkey again) to answer a call from an outside number or extension ringing the selected extension. Lights or icons are not used for Pickup keys. Note: You can also pick up a call at a selected extension by pressing *7, dialing the extension, then pressing #.
- Intercom page Do not select a resource. Press the button or softkey, dial an extension, then press # to page the extension in Intercom mode. The intercom page function can only page FortiFone telephones. Lights or icons are not used for Page keys. Note: You can also page a FortiFone telephone by pressing *84, dialing the extension, then pressing #, or the Dial key or softkey.
- Overhead page Do not select a resource. Press the button or softkey to connect to an attached external PA system. Lights or icons are not used for Page keys. Note: You can also page the external PA system by pressing *0#.
- *Queue (show list)* Do not select a resource. Press the *Queue* softkey, select a queued call, then press *Retrieve* to connect to the selected call. This function is only available when programmed as a function key on supported phone models.
- *Phone book access* Select the phone book record two-digit speed dial number as the resource. Press the button or softkey to place a call using the contact information from the associated phone book record. When the button is pressed, it will light up for the duration of the call. This function is available on the 350i, 450i and 550i models.
- User Defined (phone) The button or softkey function is assigned using the phone's configuration options, and is not assigned using the Management software.
- *Call Recording (360i/460i/560i only)* Select *Call Recording* from the pulldown menu. Press the button to start call recording during a call. The light will flash and a tone will be played when call recording is started. Press the button to stop recording. Recorded calls will be saved to the extension's voice mailbox.
- *None* The button or softkey is not programmed.

Forwarding calls out of the office

Automatic call forwarding

The auto attendant can forward an office call to a remote extension. Use the auto attendant message to prompt callers to select your remote extension number(s). (e.g. *"To speak to John Smith, dial 211."*)

Manual call forwarding

If you have answered a call at a local extension, you can forward the call to a remote extension.

- 1. Put the call on hold at the local extension.
- 2. Dial the remote extension number (e.g. 211) to which you want the call forwarded.
- **3.** Hang up an analog extension or press *Transfer* or *xfer* on an IP extension.

The call is handled according to the call cascade options of that remote extension.

Conditional call forwarding

If the auto attendant answers a call and the caller selects an extension or ring group number that is busy or is not answered, the auto attendant can forward the call to a remote extension. Conditional call forwarding is configured in the Management software using the call cascade's *No answer* or *Busy* settings for each local extension, remote extension and ring group.

Transferring calls from a remote extension

You can forward calls from a remote extension without using Centrex or 3-way calling services. Calls can be transferred to any local or remote extension, ring group or voice mailbox. This feature also applies to VoIP numbers.

- **1.** Press ** to place a call on hold at a remote extension anytime during a conversation. You hear the same dial tone as at a local extension and have the following three possibilities:
 - Press ** to retrieve the call placed on hold.
 - Dial any local extension, remote extension, voice mailbox, VoIP location or ring group.
 - If the system is configured to perform a blind transfer, the system plays "Call Transferred Goodbye." and then hangs up.
 - If the system is configured to allow call screening, determine whether the other person wants the call, and then press **4 to complete the transfer or **5 to cancel the transfer and return to the caller.

Screening options for forwarded calls

The screening options for call forwarding are configured in the Management software using the *Answered* tab in the *Remote Extensions* page.

Forwarding calls with screening

When *play accept/reject prompt* is configured in the *Answered* call cascade option in the *Remote Extensions* page, the system plays a pre-recorded prompt to callers asking them to hold and then dials your call forwarding number. When you answer the forwarded call, you hear a prompt saying: "*You have a forwarded call. To accept the call, press #. To reject the call, press *.*" If you reject the call, it is routed according to the call cascade option specified next to *If a call is rejected from this ext.*

When *play caller's name first* is configured in the *Answered* call cascade option, the system plays a pre-recorded prompt to callers asking them to record their name at the sound of the tone. The system asks the caller to hold and dials the call forwarding phone number. When you answer the forwarded call, the system plays a pre-recorded prompt saying: *"This is call forward, you have a call from..."* and plays the recording of the caller's name. To accept the forwarded call, press # on your telephone keypad, re-route the call to its call screening call cascade options by pressing *.

When either screening option is used, music on hold is enabled and you are not using the 3-way calling feature (*Same Line Connect*) to forward calls, the caller hears music while the system is waiting for the remote extension to accept the call.

If a forwarded call is not answered or the line is busy, the system:

follows the remote extension's *No Answer* or *Busy* call cascade action. If the call was
manually forwarded, this action can be changed to ring the extension back to the person
who performed the transfer.

Using the voicemail system

Internal music on hold, voicemail, and the auto attendants use the memory of the phone system. If you need to expand the memory, talk to your reseller.

There is a limit of 99 voicemail messages per mailbox, so users should delete messages before the mailbox fills up.

Activating voice mailboxes

You can activate voicemail with a mailbox, as an announcement, or you can deactivate it. There are three types of voice mailboxes:

- Local extension mailboxes are associated with local extensions. All local extensions are activated with a respective mailbox by default.
- *Remote extension mailboxes* are associated with the remote extensions. If you activate a remote extension, the mailbox is activated by default.
- General voice mailboxes are not associated with extensions.

Callers can reach voice mailboxes from a call cascade, be transferred from a local extension or have announcements played through the auto attendant.

If you want incoming calls to go straight to a voice mailbox after a specified number of rings, program an auto attendant to answer the line and then route the call to voicemail.

To leave a message from an analog extension:

1. Press * and the mailbox number.

To leave a message from an IP extension:

1. Press * and the mailbox number and press Dial, Send or #.

To transfer a caller directly to voicemail from an analog extension:

1. Press *Flash* or *Recall* *, the mailbox number and hang up.

To transfer a call to a voice mailbox from an IP extension:

- 1. Press Transfer or xfer *.
- **2.** Enter the mailbox number.
- 3. Press *Transfer* or *xfer* to complete.
- 4. Press *Dial*, *Send* or #.

When a caller presses # after leaving a message or after 2 minutes have elapsed, the system plays a prompt asking callers if they wish to keep the message, listen to the message or re-record the message. Callers can dial *0* (9 in some regions) and the system directs the call according to the settings of that mailbox.

Retrieving messages and accessing a voice mailbox

The first time you access your voice mailbox, you are prompted to assign a password, record a greeting and record your name for the dial-by-name directory.

For the general mailboxes, there are no directory listings and you are not prompted to record a name. If you wish to have your name in the company directory, use the mailbox associated with its extension, as the directory is related to the extension's mailbox. When your mailbox has been set up, you access it through the prompt instructions.

If a new message is left in a local extension mailbox, the system plays a stutter dial tone when the handset is picked up (not applicable to the 850i). On some phones, the message waiting light flashes. Selected phones will also show a message waiting counter, and some display the number of new messages stored in the mailbox(es) associated with that extension.

- 1. If direct line access is enabled, be sure to press *Flash* or *Recall* before trying to access your mailbox. From a Grandstream or Polycom IP extension, press *MSG* or *Messages* button.
- 2. Press **# from a local extension or ** and the mailbox number at the auto attendant if you are dialing from an outside location. From an IP extension, press *Dial*, *Send* or #.

3. Dial *1* to listen to messages.

By default, the system starts with newest message before it plays the older ones. While the message is playing or after the message has finished, you can dial the following:

1	to rewind 10 seconds
11	to rewind to beginning of the message
3	to skip ahead 10 seconds
33	to skip to the end of the message
5	to listen to the time and date stamp
6	to forward the message
7	to delete a message
9	to save a message
*	to back up one level
#	to skip to next message, leaving new messages as new

4. Dial 2 to change greeting options.

This allows you to record a new personal voicemail greeting. The default greeting is: "The extension you have reached is unavailable at this time. Please leave a message after the tone".

While recording your greeting or announcement, remember that callers can press * to return to the auto attendant, if the call came from the auto attendant. When they are done recording, they can press # for more options. You may want to give your callers these options for their convenience. In local extension mailbox options, 0 (9 in some regions) can be configured as an option for additional routing.

- 5. Dial 3 to change personal options.
 - change your current password
 - turn the auto date and time stamp on/off
 - set up or change pager and remote phone notification

By default, the auto date and time stamp is on. This information is played at the end of each message. If you have caller ID service, the phone number is displayed with the time and date.

6. Dial 4 to record your name for the dial-by-name directory.

If you haven't recorded your name for the directory or if you wish to change the recording, dial 4. The dial-by-name directory is accessed at the auto attendant according to your configuration. Callers are directed to enter the first 3 letters of the employee's name. The system finds the exact or closest match and plays the recorded name with the extension number, giving callers the option to connect to that extension.



You must record your name in order for the extension to be included in the dial-by-name directory.

7. You can transfer a caller to voicemail by pressing *Flash* or *Recall* ** and entering the mailbox number. From an IP extension, press *Transfer* or *xfer* + mailbox + *Dial*, *Send* or #.

Using Voicemail broadcast

Voicemail broadcast distributes a recorded message to a group of voice mailboxes. You can set up to ten voicemail broadcast groups system-wide. Voicemail broadcast groups must be configured before use. See "Voicemail Broadcast" on page 87 for configuration instructions.

To send a broadcast voicemail, dial * + the voicemail broadcast group number and follow the prompts.

Recording an announcement on a local extension

An announcement is a recorded message providing information to callers. It can be recorded using your phone as a voicemail message or loaded as an audio file. The system hangs up after playing an announcement. Callers have the option of pressing * during an announcement to return to the auto attendant. Callers can also dial *0* (9 in some regions) to perform other actions if this is configured in the voicemail options.

You might include in the announcement a mention of the selections a caller may press and leave a section of silence at the end to give the caller time to make a selection.

To record an announcement, access a voice mailbox and select option 2 to change greeting options as described in "Retrieving messages and accessing a voice mailbox" on page 165.

Pager and cell phone notification

When a new message is left in a voice mailbox, it can notify a pager, cell phone or a remote number.

You can configure the system to notify you for every new message or for only the first new message received since the last time you accessed new messages.

If you use remote phone notification, you can accept or reject the notification call when you answer.

- Press # to listen to messages.
- Press * to postpone listening to messages.

Music on hold

The system can play music to callers when they are on hold, parked or queued. The music played is provided by the source you have connected to the music jack or from a sound file (8 KHz, 8 bit, mono, u-law, .wav format) stored on the unit(s). When a sound file is loaded on one of the systems, it is duplicated to all units on the LAN.

Music on hold and call forwarding to remote extensions

If you are using the telephone company's 3-Way Calling/Conference service with the same line connect feature to activate the phone system's call forwarding feature, callers do not hear music on hold while they are being forwarded to a remote extension. The caller is put on hold by the telephone company's central office switch rather than by the phone system, and the caller hears silence while being transferred.

A system prompt indicates that callers will hear silence while the system is locating their party. This prompt explains to callers why there is silence for an extended period of time before connecting to the remote extension. This prompt cannot be disabled.



If you have two or more units on a LAN, we recommend that you use a .wav file. If you use an external audio source, connect it to each phone system unit. Use 1/8" audio splitters, which are available at most electronics stores.
Switching modes



Ensure the Management software is closed during automatic mode switching, or if an administrator is switching modes remotely. In these cases, the mode cannot switch while the Management software is open.

Note that the mode can switch while the Management software is open, if you are switching modes using the software.

Switching modes manually

Use telephone keypad commands at a local extension or at the auto attendant to switch modes.

Locally

- **1.** Enter one of the following commands:
 - ***30 + # to hear the current mode.
 - ***31 + # to enable Mode 1.
 - ***32 + # to enable Mode 2.
 - ***33 + # to enable Holiday Mode.

Remotely

- **1.** Call into the system.
- 2. Enter command mode by pressing # during the auto attendant.
- 3. Enter the system password, followed by #.
- **4.** Enter one of the following commands:
 - 30 + # to hear the current mode.
 - *31* + # to enable Mode 1.
 - *32* + *#* to enable Mode 2.
 - 33 + # to enable Holiday Mode.

Switching modes automatically

Use the Management software to set up automatic mode switching. See "Scheduling" on page 10.

If you switch modes manually, the system will automatically switch modes during the next scheduled mode change.

Using call bridge and call back

Working together, call back and call bridge act as your personal long-distance operator. Whether you are across town or around the world, these two features allow you to place calls from your office telephone line.

You can make a call to the phone system, access a telephone line connected to it and enter the number you want to dial. This is especially useful when you are out of the office with your cell phone and need to make a long-distance call. You can avoid the long-distance cell rates by making a local call to the system and access your office savings plan through call bridge.

Using call bridge

- **1.** Dial into a telephone line.
- **2.** When the auto attendant answers, select a hunt group 9 (0 in some regions), 81–88 or Same Line Connect 80.
- **3.** Enter your account password. If you are using *Same Line Connect* and your line supports the 3-way calling/conference service, you are prompted to enter the phone number.
- 4. When you have completed your call, do one of the following:
 - a. Make another call by pressing ##.

The system disconnects you from the call in progress. You can dial another number or redial the same number.

b. Activate the auto attendant by pressing #*.

The system ends the call bridge session and activates the auto attendant. When the auto attendant message begins to play, you have the following options:

- Press # to enter command mode
- Dial 6 to change/enter call back settings
- Dial a local or remote extension or a ring group. This option allows you to contact someone in your office, check your voicemail, etc.
- **c.** End the call bridge session by hanging up. This disconnects you from the system. To ensure the system disconnects at the end of a call bridge session, press ## before hanging up.

The call bridge phone number you dial can be local, long-distance, toll free (800 and 888) or international. Enter the phone number as you would in your office. For long-distance calls, include the 1, country code and area code.

Using call back

The call back feature allows you to initiate the system to call you at a specified phone number. This gives you access to the following:

- Call bridge
- Local extensions, remote extensions and ring groups
- Voicemail
- Configuration settings

Using call back involves 3 steps:

- 1. Call your system phone number to activate call back.
- 2. Answer and accept call back. The system calls you back within 30 seconds.
- **3.** At the auto attendant you can choose to perform call bridge, dial an extension, access and retrieve voicemail or configure your system.

Activating call back

There are two ways to activate call back from your out-of-office location.

Using prompted call back activation

- 1. Dial your system's phone number.
- 2. Let the line ring until the auto attendant answers.
- **3.** To initiate call back at the current call back number, dial *61* on the telephone keypad, and then hang up. The system will call you back and then prompt you for your password.
- 4. To initiate call back at a new call back number, dial 62 on the telephone keypad. The system will prompt you for your password. Enter your password and the new call back number, and then hang up. The system will call you back and then prompt you for your password.



For prompted call back, the system always dials the last prompted call back number entered. If you do not have the auto attendant answer a line, a generic system auto attendant will answer after 15 rings. You can then dial 6 to access the call back settings.

Using auto call back activation

- 1. Dial one of the call back phone numbers that has been configured for auto call back activation.
- **2.** Let the line ring at least once but no more than three times and hang up before the call is answered.
- 3. Within 20 seconds, the system calls your auto call back number.



It is important to remember that all call back settings must be configured to use auto call back. Please ensure the auto attendant for the line using call back does not answer before 4 rings.

Accepting the call back

There are two ways to accept a call back.

Directly answering the call back

1. Answer the call and press #.

If required, enter your password. This option is used when you have a direct line to your call back phone number.

Using the announced message option

The Use announced message option allows the call back from the system to reach you when you are in a hotel or an office where calls are intercepted by a receptionist or switchboard operator. When the call back is answered, the system begins playing your pre-recorded message (e.g. *"Please forward this call to Bob Smith in room 312"*). The message is repeated for 2 minutes.

1. When the call is forwarded to you, press # and enter your password on the telephone keypad to accept the call back. If the call back is not accepted within the initial 2-minute period, the system disconnects the call back.

When you accept the call, you hear the auto attendant. You can select to check your voicemail, ring an extension, use call bridge or enter command mode.

Using Conference Bridge

The system supports 10 conference bridge accounts that will allow up to 8 participants to join a conference.

All participants get a number to dial when they call into the system to enter the conference area. The moderator creates another access code for participants to enter their particular conference.

Create a conference bridge number

1. Go to the *Conference Bridge* page.

Conference Bridge	44. 44.
Activate Conference Bridge	@
Assign a conference bridge number.All users will dial this number to access	a conference. Each conference supports up to 8 participants
Conference Bridge Number:	
-Moderator Codes	
D Label Assign an access code for each mode Moderators can create conference brid	ator. Access codes must be 4 to 8 digits. ges from any local extension
3 Label:	
5 Moderator Access Code:	
The moderator issues codes to particip	ints. Set the lifespan of the participant access code for
Participant access code expires in: 6 H	ours 🗸

- 2. Check the Activate Conference Bridge checkbox.
- **3.** Assign a conference bridge number. This is a unique 3-, 4- or 5-digit number, depending on your dial plan.
- 4. Assign moderator access codes. Under *Moderator Codes*, select 1 of the 10 conference numbers and assign a code. Give the code a label so you can remember what it's for.
- **5.** Assign a lifespan to participant access codes in the *Participant access code expires in:* pull-down menu. The participant access codes are assigned by the moderators.

Assigning participant access codes

- **1.** The moderator calls into the conference area by entering the conference bridge number from a local extension.
- 2. Follow the prompts to create a new participant access code.

The moderator distributes the conference bridge number and the participant access code to the participants prior to the conference.

Call Detail Record Logging

Introduction

This chapter covers the setup and retrieval of call detail records (CDRs), which are records generated by calls through the phone system.

The unit can store the CDRs. The system can automatically e-mail the CDRs, or you can manually retrieve them with the Call Reporting licensed add-on application or a web browser.

If multiple units are networked on a LAN, the CDRs from all the units are stored on one unit. The unit stores approximately 6000 records in a circular buffer. This means that after about 6000 calls, the oldest records will be deleted and new ones created. On a very busy system, 6000 records may only cover a couple of days, but on a system with less traffic, 6000 records may cover more than a month.

Enabling call detail record logging

1. Choose Tools > Call Logging Output (CDR). The Call Logging window appears.

tore to file on Unit	~		
Unit			
Select a unit that v	vill handle all CDR data	1	
③ Unit 1	🔿 Unit 2	🔿 Unit 3	🔿 Unit 4
Z Email Log File-			
Email address:			
Time to send: 1	1:00 PM		
-Davs to Send-	·····		
	—		
monday			
	Saturday	Sunday	
Logging			

- 2. Select how the unit will process CDRs. Choices are:
 - Deactivated CDRs are neither stored nor sent to the computer.
 - Store to file on unit CDRs are stored in the unit. Selecting this enables the Unit area and the Email Log File area.

Unit

The *Unit* area allows you to select the unit that will store the CDRs. It is enabled when you select *Store to file on unit*.

1. Select the unit that will process the CDRs. Only active units are available.

Email log file

The *Email Log File* area allows you to have the system regularly e-mail a log file containing the CDRs. It is enabled if you selected *Store to file on unit*.

Note that if you want the system to e-mail a log file, you must configure the outgoing mail server (SMTP). See "Email Service" on page 19.

- 1. To have the system e-mail the log file, select the *Email log file* checkbox.
- 2. Enter the Email Address.
- **3.** Select the *Time to send*.
- 4. Select the Days to send.

Retrieving call data records

There are two methods for retrieving CDRs:

- · Call Reporting licensed add-on application
- Browser

Call reporting

The Call Reporting licensed add-on application automatically connects to the unit, downloads the CDRs, and builds detailed reports of call activity. The application requires the *Store to file on unit* setting in the *Call Logging* window.

Browser

You can connect to the unit and download the CDRs using a browser. Retrieving the CDRs using a browser requires the *Store to file on unit* setting in the *Call Logging* window.

- 1. Select the *IP Configuration* page in the Management software, and note the *Unit IP Address* (e.g. 192.168.1.102) where the CDR is being stored.
- **2.** Select the *Administration* page in the Management software, and note the *System name*. You will also require the *System password*.
- **3.** Enter the following into the *Address* field of a web browser: http://, followed by the unit IP address, followed by :8484/ (e.g. http://192.168.1.102:8484/). The *Log in* window appears.

User name:	🖸 phonesystem	*
Password:	••••	

4. Enter the system name in the *User Name* field of the *Log in* window, and the system password in the *Password* field. The browser then shows the *Welcome* screen.

	Web Admin
Home	Welcome to 284 VS (Unit 1)
IP Status	Firmware version: 284 VS 7.00.00 last reheat at: 2011/00/27.14/36/38
SIP Status	Filliware version. 284 v.5 7.00.00 last reboot al. 2011/09/27 14.50.58
RTP Status	System Status
UPnP Status	Trace Information
Trace Info	Can Detan Record (CDR) Management
	The system is currently connected to 1 configuration tool(s) located at IP: 192.168.1.5:54895
	No active attendant console TCP link detected

5. Click the Call Detail Record (CDR) Management link. The browser shows the Call Detail Record (CDR) Management screen.

6. Scroll down to the Download CDR section.

Download CDR

```
    To view the CDR in the browser, simply (left) click on the link below.
    To save the CDR to a file:

        Microsoft Internet Explorer

        Right click on the link below, select "Save Target As..." and save to a convenient

        location.

        Netscape, Mozilla Firefox

        Left click on the link below, the CDR will be displayed in the browser. Next, select

        "File" - "Save Page As..." from the menu bar and save to a convenient location.
```

Download CDR (tab delimited)

- 7. You can either right-click the *Download CDR* link to save the information to your PC, or can click the link to view the current information.
- 8. Once the CDR file has been downloaded, scroll down to the Clear CDR section.

Clear CDR

• To clear the CDR on the unit, click here.

Analyzing the data



We recommend that you clear the unit immediately after downloading the CDRs. The unit will then start saving data to a new file. This allows easier viewing and analyzing of the captured information.

The data can be viewed either through the web interface or downloaded into a spreadsheet (e.g. Excel) as a tab delimited text file. The log is divided into 12 columns identified by numbers:

1. Туре	5. Time	9. Caller ID Name
2. Log	6. Duration	10. Line
3. Event	7. Connection	11. Account Number
4. Date	8. Phone Number	12. User

Each column contains specific information related to the current state of the call. The following is a list of the columns with a description:

- **1.** *Type* This column can contain the following characters:
 - I Inbound call
 - T Transfer state
 - *O* Outbound call
 - *B* Bridged call. For example, a call in on one line that is forwarded over another line. This can include calls forwarded to remote extensions.
 - X Blocked call
 - A Account number assigned to the call with the same log #
 - *D* Dropped / Abandoned call.
- **2.** Log This column indicates log number for the call. For the duration of the call, the log number remains the same. The event number will increment with each change of state. The first digit of the log number indicates on which unit the event originated.
- **3.** *Event* This column tracks various state changes of the call. For example, when an auto attendant answers, the counter might be 1. When the caller selects an extension, the counter increments to 2, etc.
- 4. Date This column shows the date of the call (MM/DD/YYYY).

- **5.** *Time* This column contains the time that the call entered a specific state. The time is displayed in 24-hour format (HH:MM).
- 6. Duration This column indicates the total time the call was in this state/event. The time is displayed in 24-hour format (HH:MM:SS).
- **7.** *Connection* This column indicates where the call was for each event. This includes auto attendants, voicemail, extension ringing or queuing.
 - Ex = Connected to extension x
 - A0x = Connected to auto attendant x (x = 01 to 20)
 - Rx = Ringing at extension x
 - RCO = Ringing at telephone line.
 - Qx = Queued at extension x
 - Hx = Call on hold at extension x
 - Mx = Connected to voice mailbox x
 - Cxy = Forwarded out on unit x line y. (See "10. Line" below for more details).
- **8.** *Phone Number* This column displays the phone number of the inbound or the outbound caller if available.
- 9. Caller ID Name This column displays the name of the caller if available.
- **10.** *Line* This column indicates on which unit and line the call came in or out. The format of this information is unit and line number. Lines 01–08 are telephone lines and 09–16 are VoIP lines.

Example: 101 — unit 1, line 1 203 — unit 2, line 3 110 — unit 1, VoIP 2

11. Account Number — This column contains an account number that the user can assign after a call. This allows you to group calls together that may have been placed over time to calculate the total amount of time spent with a particular customer. The account number can be a numerical code up to 15 digits long.

To assign an account number after a call has been completed, dial *88. When prompted, dial <Account Number> + #.

If Direct Line Access is enabled on an extension:

- Analog extension press *Flash* or *Recall*, then dial *88. When prompted, dial <Account Number> + #.
- IP extension press **, then dial *88. When prompted, dial <Account Number> + #.
- **12.** *User* If Permissions are enabled, this column shows the code name associated with the access code used to dial out.

The data can be imported into any call management software or an Excel spreadsheet as a tab delimited text file. After the import, the data can be grouped and tallied based on the criteria set forth.

Example:

1. Total line usage for a day	4. Average call times
2. Number of inbound calls	5. Number of calls per line
3. Wait times	

VoIP Information

Connecting to a network

Each location requires a high-speed connection to the Internet or private IP network that is sufficient for VoIP calls.

Connecting to local and IP networks

Set up a LAN (local area network) with an Ethernet switch, router, and modem or gateway. The Ethernet switch can be integrated into the router. Connect a computer to the LAN for configuring the phone system and other equipment. Use the provided RJ-45 Ethernet cable to connect the phone system to the LAN. To ensure reliability, all equipment should be connected to a UPS (uninterrupted power supply).

Figure 1: Local and IP network setup



Confirming network capacity

Ensure the LAN in each location has a reliable high-speed broadband connection to the Internet or private IP network. The quality of VoIP calls, especially on mixed voice and data networks, depends on high data-transfer rates across the network. The limiting factor is the upstream bandwidth to the ISP (Internet Service Provider). 'Lite' broadband connections (128 Kbps or less) are not suitable for simultaneous voice and data traffic.

Configuring IP addresses

Setting the system IP settings

Each system must be configured with a local IP address. These addresses are used to direct VoIP calls to the appropriate location.

DNS inf	ormatic	n	•													
. 10	. 10	. 3	Subnet mask:	255	. 255	. 255	. 0]								
. 64	. 14	. 72	Default gateway:	10	. 10	. 10	.1									
. 64	. 14	. 80	Preferred DNS server:	10	. 10	. 10	.1									
. 64	. 14	. 88	Alternate DNS server:	0	. 0	. 0	. 0]								
Iress: S	tatic p	ublic IP ad	idress 👻													
Iress: 5	5.5	55 . 55	. 55													
QDN):																

- 1. In the Management software, select the *IP Configuration* page. By default, *Obtain IP and DNS information automatically* is selected and the area shows IP addresses from the router.
- 2. Change Obtain IP and DNS information automatically to Use configured IP and DNS information in order to lock in the IP addresses.
- In some cases, the System IP Settings area may be blank because your router has not delivered the IP addresses. If so, enter the following IP addresses from your LAN administrator:
 - a. Enter a static IP address for each system in the Unit IP address boxes.
 - **b.** Enter the *Subnet mask* for the LAN. This address determines the subnet that the unit IP addresses belong to.
 - **c.** Enter the IP address of the *Default gateway* on your network. A gateway is a hardware device that connects the office network to the Internet. The router may act as default gateway.
 - **d.** Enter the IP address of the *Preferred DNS server*. DNS is a service that is used to resolve a domain name to an IP address. The router may act as DNS server.
 - e. If applicable, enter the IP address of the Alternate DNS server.

Setting the public IP address

If you are setting up an external IP extension or multiple locations, you must configure a public IP address for the system. Some service provider VoIP networks also require the system to have a public IP address.

DNS information		
. 10 . 10 . 3	Subnet mask: 255 . 255 . 0	
.64 .14 .72	Default gateway: 10 . 10 . 10 . 1	
. 64 . 14 . 80	Preferred DNS server: 10 . 10 . 10 . 1	
. 64 . 14 . 88	Alternate DNS server: 0 . 0 . 0 . 0	
š		
s to the PRI interface.Th	is IP address must be different than the system IP address.	
. 10		
		0
ress: Static public IP a	ddress 🗸	
ress: 55 . 55 . 55	55	
2DN):		

- 1. Set the *Type of public address*. Choices are:
 - Dynamic public IP address This is the default setting. Your ISP (Internet Service Provider) will assign different public IP addresses to your location. The system will check its public IP address every few minutes. When the public IP address changes, the system will automatically use the new one, in order to manage VoIP calls properly.
 - Static public IP address A static IP address doesn't change. It's assigned by your ISP.

If *Static public IP address* is selected, the window allows you to enter the *Current public IP address*.

2. If you selected *Dynamic public IP address*, enter the *Public domain name*. Get the public domain name from your ISP.

A DDNS (Dynamic Domain Name Service) provider such as www.dyndns.com matches your dynamic IP addresses to your public domain name, so your multiple locations and external IP phones will continue to work when the IP address changes.

If your router supports DDNS, ensure it supports your DDNS provider, and configure it to update the DNS servers.

If your router does not support DDNS, download one of the applications specified on www.dyndns.com. To update the DNS servers, the application needs to run on a PC connected to the same LAN as the phone system.

3. If you selected *Static public IP address*, enter the *Current public IP address* from your Internet Service Provider. Leave the *Public domain name* box blank.

If the system is not behind a router, or if a private virtual network is used, the public IP address is the local IP address of the system acting as local proxy.

Note that it will take up to one minute for the new static public IP address to take effect.

4. Record the Current public IP address or the Public domain name.

If you are setting up an external IP extension, you will need the public IP address or public domain name of the system.

Configuring the router

The *Firewall Settings* area of the *IP Configuration* page displays the type of gateway device (i.e. the type of router), the IP address of the gateway (i.e. router), and whether router configuration is required.

A router is a gateway between the local area network and the Internet. Most routers have a firewall to block unwanted data from the Internet. For voice data to reach the phone system through the firewall, port forwarding is required.

If you are setting up external IP extensions, or a VoIP service that doesn't handle port forwarding, port forwarding is required.

If port forwarding is required, and your router supports uPNP (Universal Plug and Play), ensure uPNP is enabled. The system will use uPNP to automatically set up port forwarding, and the *Automatic (uPNP Enabled)* link will appear. No router configuration is required.

If port forwarding is required but your router doesn't support uPNP, or automatic port forwarding doesn't work, the *Manual Port Mapping Required* link will appear. You will need to configure the router as described below.

- **1.** Select the *IP Configuration* page.
- 2. If required, click the *Manual Port Mapping Required* link. The *Manual Port Mapping* window appears. It lists the packet type, port number, IP address and protocol of each required port.

he system requires the following firew	all ports be mapped	l to the following IP	addresses fo	or
Description	Port	System IP	Protocol	
SIP (VoIP signalling)	5060	192.168.1.103	UDP	1
RTP Audio (Unit 1, Port 1)	6100	192.168.1.103	UDP	-17
RTP Audio (Unit 1, Port 2)	6102	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 3)	6104	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 4)	6106	192.168.1.103	UDP	٦.
RTP Audio (Unit 1, Port 5)	6108	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 6)	6110	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 7)	6112	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 8)	6114	192.168.1.103	UDP	
TFTP (IP Phone Configuration)	69	192.168.1.103	UDP	- -

- **3.** To access the router configuration:
 - **a.** Click the link containing the IP address of the gateway. The default browser starts, and prompts you for the router's user name and password.
 - **b.** Enter the router's user name and password. The browser shows a setup screen.
 - c. Navigate to the screen used to set up port forwarding. See your router documentation.
 - **d.** Set up port forwarding using the information from the *Manual Port Mapping* window. See your router documentation for instructions on how to map ports. For information on configuring routers and mapping ports, visit http://www.portforward.com/.

4. To check the status of each port through the firewall, click *Check Firewall*. The *Firewall Test* window appears.

vall Test						
ewall Settings						
Service	Status	Service		Status		
Internet Access		RTP Audio (L	Init 1, Port 6)			
HTTP (XML Service)		RTP Audio (U	Init 1, Port 7)			
RTP Audio (Unit 1, Port 1)		RTP Audio (U	Init 1, Port 8)			
RTP Audio (Unit 1, Port 2)		SIP (VoIP sig	nalling)			
RTP Audio (Unit 1, Port 3)		TFTP (IP Pho	ne Configuration)			
RTP Audio (Unit 1, Port 4)						
RTP Audio (Unit 1, Port 5)						
					2	191
					Details >>	Test Ports

- 5. Select the services you want to check.
- 6. Click *Test Ports*. The system will check the ports for the selected services.

Configuring the router manually

If you cannot access the router configuration through the *IP Configuration* page, configure the router manually.

- **1.** Open the router configuration and navigate to the screen used to set up port forwarding. See your router documentation.
- 2. In the Management software on the *IP Configuration* page, click the *Manual Port Mapping Required* link.
- 3. Map the port indicated for SIP (VoIP) signaling.

If required, you can map a different port. Select *Troubleshooting* > *VoIP* and enter the port in the *SIP signalling port* field.

Map the rest of the ports to the IP addresses indicated in the Manual Port Mapping window.

roper VoIP communication				
Description	Port	System IP	Protocol	
SIP (VoIP signalling)	5060	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 1)	6100	192.168.1.103	UDP	1
RTP Audio (Unit 1, Port 2)	6102	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 3)	6104	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 4)	6106	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 5)	6108	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 6)	6110	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 7)	6112	192.168.1.103	UDP	
RTP Audio (Unit 1, Port 8)	6114	192.168.1.103	UDP	
TETP (IP Phone Configuration)	69	192.168.1.103	UDP	

If required, you can map different ports. Select *Troubleshooting > VoIP*.

- **4.** Map ports 9393, 8485 and 8486 (Type: TCP) to the unit acting as local proxy to allow remote configuration of the system.
- 5. If available, enable Quality of Service (QoS) to give voice traffic priority over data.
- 6. Save the configuration to the router.

External IP extensions



Using an external IP extension to call an emergency service number will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP extension stating:

If an emergency call is made from this phone, you must provide your address to the emergency operator.

An external IP extension is an IP phone located outside the office. It is configured as a local extension of a system, but connects over the Internet or private data network. A user can receive or place a call with their external IP extension through the system to the standard telephone network, or to a VoIP network.

For example, the following illustration shows external IP extension 151 at a home office, and 152 at a branch office. The external IP extensions connect to the phone system at the main office over the Internet. In this example, a user can receive or place a call with their external IP extension through the phone system to the standard telephone network.



Figure 2: External IP extensions

Those system

Before proceeding, ensure you have:

- Connected the system at each location to a network (page 176).
- Configured local and public IP addresses (page 177).
- Configured the router (page 179).

To set up an external IP extension:

- 1. Add the external IP extension, as described in "Adding IP phones" on page 55.
- 2. Verify operation of the external IP extension:
 - a. Select the Extension/Fax page.
 - **b.** Select the external IP extension.
 - **c.** Confirm *Extension status* is *Registered*, and the phone's *IP address* appears in the *IP Extension Details* area.

[IP Extension Details					0
Location:	External	~		Extension status: Registered	
MAC address:	001A7EAEC53E		Select	IP address: <u>192.168.1.102</u>	
TalkSwitch unit:	1	~		Time zone: (-05:00) Eastern (US/Canada)	-
				Configure Keys	

3. Optionally, set the *Time zone* of the extension.

- 4. Optionally, perform advanced VoIP configuration:
 - **a.** All VoIP lines are shared by default. You can reserve VoIP lines for the external IP extensions. See "Reserving VoIP lines" on page 190.
 - **b.** External IP extensions use G.711µ as their preferred codec. This can be changed under *Troubleshooting > Extensions > IP Extensions*.

Service provider VoIP network

The service provider acts as the SIP server. The service provider assigns the VoIP numbers and VoIP configuration parameters.

For example, the following illustration shows a service provider VoIP network.



Figure 3: Service provider VoIP network

To reach an outside party, the user dials the hunt group number associated with the service provider profile, and then dials the phone number.

To reach a local extension, an outside party dials the VoIP number that was assigned to the system by the service provider.

Before proceeding, ensure you have:

- Connected the system at each location to a network (page 176).
- Configured local IP addresses (page 177), and public IP addresses if required.
- Configured the router (page 179).

Setting up a service provider profile

A service provider profile contains the registration details for connecting to the service provider's SIP server.



If you are using an approved VoIP service provider, visit the "Support" section at http://www.fortivoice.com to access the service configuration guide for your VoIP service provider. Otherwise use the procedures in this section.

- 1. Select the VoIP Configuration page.
- 2. Select a Profile (SP 1 to SP 4).

ofile Profile Name	Activate Profile
P 1 Service Provider 1	Service Provider: FortiCall Update Configuration
P 2 FortiCall P 3 Service Provider 3 P 4 Service Provider 4	Profile name: FortiCall Click to register for an account or free trial
	Registration Method Vour VolP Provider will have a preferred method of registration. Select a method below: Registration Method: By Account Number - Master
	r VoIP Provider Proxy/Registrar server name: forti1.secureto.us Registrar server name: forti1.secureto.us
	Outbound proxy: forti1.secureto.us Realm/domain: forti1.secureto.us
	View All Registrations
ystem VoIP Options	
VoIP Caller ID	n Caller ID information for all outgoing VoIP calls

Activate profile

You can set up a service provider profile automatically or manually.

Automatic configuration

- 1. Select the Activate Profile checkbox.
- 2. The Service Provider menu offers a list of approved VoIP service providers. If your service provider appears in the menu, click on the name. The name is then displayed in the Service *Provider* field.

3. Click the *Update Configuration* button. The essential settings for communication with the service provider's registration server will be completed automatically.

VoIP Configuration		Ø
Profile Profile Name SP 1 Service Provider 1 SP 2 FortiCell SP 3 Service Provider 3 SP 4 Service Provider 4	Activate Profile Service Provider: FortiCall Profile name: FortiCall Click to register for an account or free trial Additional Settings Registration Method Your VoIP Provider will have a preferred method of registration. Select a method below: Registration Method: By Account Number - Master	0
	VoIP Provider	
System VolP Options	D information for all outgoing VoIP calls lier ID information for all outgoing VoIP calls	-@
eserved.	Connected	- System

4. If you want to customize other aspects of your VoIP lines, you may do so in the System VoIP Options area. See "System VoIP options" on page 32.

Account-specific and number-specific settings are not automatically configured. These must be entered on the *VoIP Numbers* page.

Manual configuration

- 1. Select the Activate Profile checkbox.
- 2. Enter the *Profile name*. The default profile name is *Service provider n* (e.g. *Service provider 1*).

Additional settings

Public IP

If your service provider requires you to register using your private IP address, select the *Disable public IP address substitution* checkbox. Check with your service provider.

NAT

- 1. If your service provider requires keep alive messages, and if your router does not support uPNP, check the *Enable NAT keep alives* checkbox.
 - a. Select the method used to keep ports open. Choices are:
 - Simple ping A standard ping message that works with all SIP servers.
 - Nortel ping A ping message that works with Nortel SIP servers (e.g. Nortel MCS 5200).
 - **b.** If necessary, change the ping frequency. The default setting is 45 seconds.

Preferred ID

Preferred Identity is a supported service provider feature that allows your system to make calls with an anonymous caller ID. Please ensure your service provider offers this feature before enabling it, or you might be unable to make calls.

Codec

You can specify which codecs to use by clicking the *Additional Settings* button. See "Setting codec options" on page 31.

Registration method

Some providers require the system to register using the username or account information rather than the VoIP number(s) provided. If so, check the *Register with authentication username* box to have the system register with the username information provided in the *VoIP numbers* page. Check with your VoIP service provider if you're uncertain which method of registration is required.

VoIP Provider

Enter the IP addresses or public domain names, as provided by the service provider, into the following boxes. If the service provider does not specify a value, leave the box blank.

- Proxy/Registrar server name
- Registrar server name
- Outbound proxy
- Realm/domain

The *View All Registration* button will allow you to confirm connection to your service provider once you have completed the configuration of your VoIP numbers.

If you want to customize other aspects of your VoIP lines, you may do so in the *System VoIP Options* area. See "System VoIP options" on page 32.

Setting codec options



If you are using an approved VoIP service provider, visit the "Support" section at http://www.fortivoice.com to access the service configuration guide for your VoIP service provider. The service configuration guide lists supported codecs.

A codec is a method of compressing and decompressing audio signals for communication across a network. The system supports the G.729 and G.711 (μ -law or A-law) codecs for VoIP calls. If your service provider or equipment requires specific codecs for VoIP or Fax over IP calls, you can restrict the system to use the required codec. The following codes are supported:

If your Internet co bandwidth). The o	nnection can su default setting is	upport high bandwidth G.729 (25 kbps band	content, you can select G.711 (100 kbps width).
☑ G.729	[√] G.711µ	👿 G.711A	Voice activity detection (VAD)
Preferred codec:	G.729 👻		

- *G.729* This codec provides good quality. It requires the least bandwidth and accommodates the highest number of concurrent calls.
- G.711μ This codec provides high quality and supports Fax over IP. It requires the most bandwidth and accommodates the fewest number of concurrent calls. G.711μ is used in North America and Japan.
- *G.711A* This codec provides high quality and supports Fax over IP. It requires the most bandwidth and accommodates the fewest number of concurrent calls. G.711A is used worldwide outside North America and Japan.
- Voice activity detection (VAD) Enabling VAD reduces voice bandwidth when no speech is detected, and reduces transmission of background noise. We recommend disabling VAD to keep bandwidth available for speech.

You may select which codecs to use or clear the unsupported codecs as well as select a *Preferred codec*.

Configuring VoIP numbers for a service provider VoIP network



If you are using an approved VoIP service provider, visit the "Support" section at http://www.fortivoice.com to access the service configuration guide for your VoIP service provider. The service configuration guide lists supported codecs.

Your service provider will assign you VoIP numbers.

1. Select the VoIP Numbers page.

VoIP Numbers		
D VolP Number 1 1-224-5678 2 117 3	Activate VoIP Number Select a VoIP Profile: My Service Provider Phone Number Country code: 1 Username and Password User/Account: My Account	0
6 7 8 9 10	City or area code: 234 Password: My Password Number: 5678	
12 13 14 15	Registration Status Status: Unregistered View All Registrations Cell Handline	_@
10 17 18 19 20 21	Mode 1 (Business Hours) Mode 2 (After Hours) Holiday Mode When a call comes in on this phone number, perform the following action:	
22 23 24	If all extensions are busy or the call is not answered:	

- **2.** Select a VoIP number slot.
- 3. Select the Activate VoIP Number checkbox.
- 4. Set the VoIP profile to the service provider (e.g. My Service Provider).
- 5. Enter the Country Code (if required), the City or Area Code and the Number.
- 6. Enter the User/account and the Password (if required) for this number.
- 7. Set up call handling for the VoIP number. For more information, click the *Help* icon (
) in the *Call Handling* area.
- 8. Repeat steps 2 to 7 for each additional VoIP number.

Setting up line hunt groups



Ensure that hunt group 9 or 0 is assigned to the group of telephone lines or VoIP trunks used for calls to emergency services. Failure to properly configure hunt groups could prevent emergency calls.

A line hunt group is a set of lines that are available for making an outbound call. It can use selected telephone lines, all VoIP lines associated with a service provider VoIP network.

The configuration has nine different hunt groups. You can modify these default settings as required. If you are using multiple service provider VoIP networks, set up a hunt group for each service provider.

A local extension can be restricted to a set of hunt groups in order to control access to VoIP networks by using User Privileges. See "Routing and blocking" on page 36.

1. Select the Line Hunt Groups page.

Line Hunt Groups		2
HG Name 9 81 81 82 83 84 85 86 87 My Service Provider 88 9	Activate Hunt Group 87 Hunt Group Line Assignments Line type: SP1 VoIP Service (My Service Provider) VoIP lines will be selected automatically. To modify the number of VoIP lines available for each Service Provider, go to the VoIP Configuration page. Number of lines available: 8 Number of lines reserved: 0	9
	Hunting Order for Outgoing Calls	0
	Hunt Group Busy Overflow for Outgoing Calls If all lines are busy in this hunt group, use: no overflow If all lines are busy in the previous hunt group, use: no overflow	0
Overflow Tone Notification	flowing to another hunt group.	

- 2. Select a hunt group.
- 3. If necessary, select Activate Hunt Group. All hunt groups are active by default.
- 4. Enter a Hunt Group name. The default name is Hunt Group.
- 5. Set Line type to SPn VoIP Service.
- 6. Select the overflow hunt group. Choices are *no overflow* and the other activated hunt groups. If you select an overflow hunt group, ensure it contains different lines than the original hunt group.

Advanced VoIP configuration

Setting up caller ID

The *VoIP Caller ID* area allows you to set up the source for caller ID for outbound VoIP calls. The same setting is used for all service provider profiles. Extension names are used by default.

- **1.** Select the *VoIP Configuration* page.
- 2. Set the caller ID for outbound VoIP calls.
 - To use the System name from the Administration page, select Use System Name in caller ID information for all outgoing VoIP calls.
 - To use the First name and Last name from the Local Extension/Fax page set up for each extension, select Use Extension Names in caller ID information for all outgoing VoIP calls.

VoIP Caller ID
O Use System Name in caller ID information for all outgoing VoIP calls
⊙ Use Extension Names in caller ID information for all outgoing VoIP calls

Reserving VoIP lines

By default, all VoIP lines are available for external IP extensions, and/or service provider calls on a first-come first-served basis. You can also reserve VoIP lines for a specific use. For example, you can set aside two lines for external IP extensions.

Reserving VoIP lines guarantees resources for a specific network. Sharing VoIP lines uses them most efficiently.

- 1. Select the VoIP Configuration page.
- 2. Click Reserve VoIP Lines. The VoIP Resource Reservation window appears.

VolP Resource Reservation		(
External IP extension calls	Shared 👻	
Global Dial Plan calls	Shared 👻	
Service Provider 1 calls	Shared 👻	
FortiCall calls	Shared 👻	
Service Provider 3 calls	Shared 👻	
Service Provider 4 calls	Shared 👻	
Total VolP lines available on	system: 8	
Total VolP lines	shared: 8	
Total VolP lines re	served: 0	

- **3.** Select the number of VoIP lines to set aside for each service. Choices range from 1 to the number of unreserved VoIP lines at this location. *Shared* means no VoIP lines are reserved for this service. Only unreserved lines will be available.
- 4. To transfer settings from your computer to the system, choose *File > Save*. A window appears indicating the configuration is being sent.

Troubleshooting and Support

Troubleshooting

- Configuration
- Auto attendant
- Music on hold
- Call routing
- Answering and fax machines
- Local extensions
- Multiple units
- VolP

Some problems might be due to physical connections such as loose cables.

Configuration

I am unable to configure the system using a touchtone phone

- 1. You cannot configure the system using a phone sharing the same line in parallel. You have to configure it using one of the extensions plugged into the back panel of the system.
- 2. You cannot configure the system using touchtone keypad commands while the Management software is open. If the software is closed and you hear the prompt *"I'm sorry, the system is currently being configured",* reboot the system.
- **3.** If you are using a local extension, check if the extension has direct line access to your telephone lines. If direct line access is enabled, press *Flash* or *Recall* on an analog extension or ** on a IP extension to receive an internal dial tone. Press # to enter command mode. Enter the correct password, if you have password protection.
- **4.** From an out-of-office phone, call into the system and wait for the auto attendant. Enter command mode by pressing # on an analog extension phone, or *55# on a proprietary IP phone (note: other brands may use *55 Send or *55 Dial). Enter the correct password, if you have password protection.

Auto attendant

The auto attendant does not play when calls come in

- **1.** Check if the line light on an incoming call is flickering while the phone is ringing. If not, replace the phone cord.
- 2. Make sure you have an auto attendant message recorded.
- **3.** Ensure the system is running the correct mode. Open the Management software. Select *Scheduling*. Check the *Current mode* in the *Modes* area.
- **4.** Ensure the telephone line has the correct auto attendant setup. Select *Telephone Lines*. Ensure the right auto attendant is selected for the mode, telephone number, and telephone line. Try setting the auto attendant to answer *Immediately*.

The auto attendant message is broken up or very faint

- 1. Open the Management software. Choose *Troubleshooting* > *Telephone Lines* > *Audio*. Adjust the *Transmit levels* volume setting.
- 2. The quality of the microphone in the telephone handset you used to record your auto attendant can affect the quality of your recording. Try recording the auto attendant from another extension/telephone.

The auto attendant answers calls before any of the extensions ring

If users wish to have their local extensions ring before the auto attendant picks up incoming calls, do the following:

1. Open the Management software. Select *Telephone Lines*. Select the telephone line, telephone number, and mode. Click *Edit*. Increase the number of rings.

The auto attendant is transferring calls to the wrong extension

1. Check if the extension jacks of your devices (phones, faxes) are plugged into the corresponding jacks on the back panel (i.e. extension 111 is plugged into E1).

The auto attendant answers calls, but does not transfer them to the extensions

- 1. Make sure your extensions are plugged into the extension jacks on the back panel of the system (instead of your telephone wall jacks).
- 2. Open the Management software. Expand *Auto Attendants*. In the *Actions During Auto Attendant Playback* area, check if the actions and resources point to the correct extensions.

Calls are going to my phone company voicemail instead of being answered by the system

This may be a result of two different situations:

- 1. If the line is currently busy and you do not have the hunt/rollover service from the phone company, then the call will go into voicemail.
- 2. If calls are ringing, then going into voicemail, the system may be set to answer after a predetermined number of rings that is larger than the number of rings set for your phone company voicemail.

To remedy this situation, decrease the number of rings before the auto attendant answers or increase the number of rings before your phone company voicemail answers.

To decrease the number of rings before an auto attendant answers:

1. Open the Management software, select *Lines and Greetings > Telephone Lines*. The *Telephone Lines* section appears.

Under the *Call Handling* section, set the *Routing option* to less than the phone company's voicemail number of rings, or to *immediately*.

How do I access my voicemail remotely?

To access your voicemail remotely:

1. Call into the system. The auto attendant answers.

2. Press **, followed by your extension number. This will take you into the standard voicemail options and allow you to listen to your messages and change voicemail options.



If your inbound call is answered by a, they can transfer you to your voicemail box by transferring your call by pressing **, followed by your extension number.

Music on hold

Callers hear the "One moment please" message and then total silence when the auto attendant transfers their call to an extension

1. There is a problem with the music source. Check all connections and the power.

Callers hear only silence when put on hold at an extension

- 1. Make sure you use the *Flash* or *Recall* or *Link* button on an analog extension, or the *Hold* button on an IP extension, to put callers on hold. If you use the *Hold* button on an analog extension, callers will be placed on hold at the extension itself and not through the system.
- 2. You have enabled the music on hold feature, but have not attached an audio device to the music jack.
- **3.** You have enabled play the *Play music from file* feature but have not loaded a .wav music file.

Call routing

Callers are disconnected when transferring calls from one extension to another

1. If you are using the hook switch rather than the *Flash* or *Recall* button to put callers on hold, you may be holding the hook switch down too long and disconnect.

I can't use my conference/3-way calling feature from the phone company

1. You need to make sure you have checked the box indicating you have 3-Way Calling/Conference service or the Transfer and Clear service on the appropriate lines in the *Telephone Lines* page.

I'm unable to place intercom calls from a local extension

1. If the local extension has been set up for direct line access to your telephone lines, you need to press *Flash* or *Recall* on an analog extension, or ** on an IP extension, before intercom calling.

Answering and fax machines

The answering machine and/or fax machine picks up calls before they can be answered by telephone extensions

1. Check the ring sequence for the telephone line, telephone number and mode. Ensure the extension for your fax machine and/or answering machine is not set to answer calls at the same time as your telephone extensions. If necessary, remove the extension for the fax machine and/or answering machine from the list of extensions to ring on an incoming call.

Incoming faxes are not automatically detected and routed to the fax machine

- 1. Ensure you selected the correct extension in the *If a fax call is detected* list in the *Auto Attendants* page.
- 2. Not all fax machines emit a CNG tone that the system can detect and route. Therefore, you may consider incorporating an additional method of fax routing. This could include the use of a dedicated fax line or a distinctive ring number.
- **3.** Many fax machines/modems will time-out or disconnect after 25 seconds on average if they have not connected with another fax machine/modem. If you set up the *Action Performed After Auto Attendant Playback* area, the fax call may time-out before it reaches your fax extension. If you have a long auto attendant message, you may want to shorten it or use another fax routing option.
- **4.** The level of the auto attendant is too loud and is overpowering the CNG tone, affecting the detection. Try lowering the volume level of the auto attendant.

Local extensions

I am unable to access lines with a local extension

- 1. Check your direct line access setting in the *Additional Settings* window of the *Local Extensions/Fax* page. If a local extension has not been given direct line access to your telephone lines, you must dial 9 (0 in some regions) or 81–88 to access a line.
- 2. Check if routing and blocking is enabled in the *User Privilege* page. This can re-direct or block calls.

One of the local extensions (telephones) does not ring

- 1. Some telephones require more voltage in order to ring. These phones are usually the older 'Bell' phones with mechanical ringers. These phones should not be used as extensions.
- 2. Make sure the telephone cord you are using between the phone and the system is working properly.
- **3.** Ensure the ringer is turned on.

Extension(s) ring, but there is no caller

This can occur when a caller hangs up after the auto attendant has begun to play. In some areas, the telephone company's disconnect/clear signal is weak. The auto attendant may not be able to pick up the disconnect signal as soon as a caller hangs up. The auto attendant will not receive a response after playing its message and the call will fall through to your settings in the *Action Performed After Auto Attendant Playback* area in the *Auto Attendants* page.

When I try to access voicemail, I hear a busy tone

 Check your direct line access settings in the Additional Settings window of the Local Extensions/Fax page in the Management software. If a local extension has been given direct line access to your telephone lines, you must first press Flash or Recall on an analog extension, or ** on an IP extension, before dialing the mailbox. The same applies for dialing other extensions to initiate intercom calls.

Multiple units

The configuration does not show the other unit(s)

- 1. Check the LAN connection on the system.
- 2. Check the front LEDs for error codes. Refer to the *FVC Hardware Specification Guide* for an explanation of the light sequences.

After recording an auto attendant greeting, I can't play it back

1. Try again in a few minutes. After recording an auto attendant on one of the units, it will copy it to other units on the LAN. During this process, you cannot listen to that particular auto attendant.

VolP

Callers complain the sound is distorted or choppy

- 1. Your broadband connection may not have enough upstream bandwidth to support many simultaneous VoIP calls. In the *Codec Options* area, you may need to disable the G.711 and G.726 codecs and only use G.729.
- 2. You may have too much data traffic on your Internet connection at the same time you are trying to make voice calls. We suggest you use a router that supports QoS (Quality of Service) for VoIP.

When I call someone or they call me, voice is only heard in one direction

- 1. The cause of this problem is usually a result of a router being misconfigured with respect to port mappings. Ensure all required VoIP ports are mapped to your unit. Also, ensure you use a static private IP address when connected to your router as this can also affect port mappings from the firewall/router.
- **2.** Use a router that supports uPnP.

For more VoIP information, see "VoIP Information" on page 176.

Support

If you are having problems with the configuration or operation of your system:

- 1. Contact your authorized reseller or point of purchase.
- 2. In the Management software, go to *Help > Online Support*.

Appendix A: Functions and Commands

Functions you can enter from local extensions

On an analog extension, press Flash or Recall. On an IP extension, press Hold.

Keys	Function performed
Flash or Recall/Hold	If you are connected to a caller, <i>Flash</i> or <i>Recall/Hold</i> puts the call on hold and you hear the internal dial tone.
Flash or Recall/Hold	Pressing <i>Flash</i> or <i>Recall/Hold</i> again at the internal dial tone retrieves the most recent call on hold.
<i>Flash</i> or <i>Recall/Hold</i> at external dial tone	When you are connected to an outside line, pressing <i>Flash</i> or <i>Recall/Hold</i> switches you to the internal dial tone.
* and mailbox number	Dials into a local, remote or general voice mailbox to leave a message.
** and mailbox number	Accesses a local, remote or general voice mailbox. You can retrieve messages, change greetings, password, etc.
** #	Accesses the mailbox associated with your extension.
Flash or Recall/Hold 4	Completes a transfer and returns to the internal dial tone.
On an analog extension, press <i>Flasi</i> by <i>Dial, Send</i> or # to complete the t	h 4 or Recall 4. On an IP extension, press Hold 4 followed ransfer.
0 (9 in some regions)	Dials the receptionist.
*0	External PA access to make an announcement through the attached external PA system.
* and the speed dial number	Dials the system speed dial number.
* and voicemail broadcast group	Dials the voicemail broadcast group to broadcast a message.
Flash or Recall/Hold 5	Disconnects the current caller and reconnects to the last caller on hold. A conference-call initiator can disengage the second conferenced party by pressing <i>Flash 5</i> or <i>Recall 5</i> .
Flash or Recall/Hold *500–509	Press <i>Flash</i> or <i>Recall/Hold</i> to place a call on hold and then press *500–509 to place the call in one of ten park orbits.
Flash or Recall/Hold *510	Press Flash or Recall/Hold to place a call on hold and

**500–509

orbit, starting at 500.

Retrieves a parked call.

then press *510 to place the call in the next available park

Keys	Function performed	
Flash or Recall/Hold 6	Conference call for 3 parties. Press <i>Flash</i> or <i>Recall</i> to put the first caller on hold, dial another extension or external number and press <i>Flash</i> 6 or <i>Recall</i> 6.	
**050	Plays the music on hold	
*60	Disables Do Not Disturb at the current extension.	
*61	Enables Do Not Disturb at the current extension.	
*62	Toggles Do Not Disturb at the current extension.	
*7 and extension	Call pick-up at a specific extension. Dial from dial tone to pick up a call ringing at another extension.	
Flash or Recall/Hold 7	Press <i>Flash</i> 7 or <i>Recall</i> 7 to cycle through queued calls on a first in/first out basis.	
80	Accesses Same Line Connect when transferring an outside caller to another outside number.	
9 (0 in some regions), 81–88	Dial to access an outside line.	
91#	The phone system unit will announce its IP address.	
On an IP extension, press <i>Dial, Send</i> or # after your entry for the following options:		
*80	Play music on hold from an internal .wav file or an external audio source through the PA jack. If music on hold is playing through the PA jack, and a user dials *0 to make an overhead page, the overhead page will interrupt the music on hold. However, if the PA jack is configured to perform voicemail screening, and a user or caller accesses a voice mailbox, music on hold will not be interrupted.	
*810#	Add the last caller's ID information to the Caller ID routing table.	
*81 [1–10] #	Route future calls from the last caller according to the specified Caller ID-based routing group.	
*81255#	Delete the last caller from the caller ID lookup list.	
*84 and extension	Hands-free intercom call to an extension, automatically answered in speaker mode, enabling instant hands-free 2-way communication (for supported sets).	
*85 and extension or ring group number	Page a phone at an extension or the phones of a ring group.	
*88 and account code	Attach account code to call detail record (CDR) for last call.	
*9	Call pick-up of any ringing line.	
# on an analog phone, or *55# on IP phones (others may use *55 <i>Send</i> or *55 <i>Dial</i>).	Enters command mode.	

Functions you can enter from outside phones

Enter these keys after the auto attendant answers.

Keys	Function performed
0 (9 in some regions), 1, 2, 3, 4 and 5	Performs the action defined in the auto attendant configuration.
Local extension	Rings at the local extension (e.g. 111).
Remote extension	Rings at the remote extension (e.g. 211).
Ring group	Rings all the extensions in the ring group (e.g. 301). There are 10 ring groups for the entire system.
61	Causes the system to call you back at the current prompted call back number, after you hang up.
62	Causes the system to prompt you for your password and then the new prompted call back number. After you hang up, the system calls you back.
7 and extension	Transfers a caller on hold to another phone system used in conjunction with the system. As soon as the call connects to the other system, it disconnects from the system.
9 (0 in some regions) and 81–88	Accesses a hunt group for call bridge and is protected with the call bridge password.
80	Accesses call bridge using Same Line Connect. Requires 3–way calling/conference on the line.
* and mailbox number	Dials into a local, remote or general voice mailbox to leave a message.
** and mailbox number	Accesses voicemail of a local, remote or general voice mailbox. You can retrieve messages, change greetings, password, etc.

Functions you can enter in command mode

Enter command mode by either pressing # on an analog extension phone, or *55# on some IP phones (others may use *55 Send or *55 Dial). When prompted, enter the system password + #. You can then enter the command.

Keys	Function performed
00 + #	The system tells you the Unit ID of the unit you are connected to.
0 and ID number + #	Assigns a Unit ID (1 to 4).
30 + #	The system tells you which mode it is currently using.
3 and mode number + #	Switch modes (1 = mode 1, 2 = mode 2, 3 = holiday mode).
4 and auto attendant number + #	Record auto attendant message.
04 and auto attendant number + #	Erase auto attendant message.
5 and auto attendant number + #	Play auto attendant message.
6 and auto call back number + #	Record auto call back announced message (1 to 4).
06 and auto call back number + #	Erase auto call back announced message (1 to 4).
7 and auto call back number + #	Play auto call back announced message (1 to 4).
65 + #	Record prompted call back announced message.
065 + #	Erase prompted call back announced message.
75 + #	Play prompted call back announced message.
91 + #	Play IP address of unit 1.
92 + #	Play IP address of unit 2.
93 + #	Play IP address of unit 3.
94 + #	Play IP address of unit 4.
80 + Remote extension number + * + desired external phone number	Set a remote extension phone number.
61 + number + #	Enable Do Not Disturb at the local extension.
60 + number + #	Disable Do Not Disturb at the local extension.

Appendix B: Power Interruptions

Settings and configurations

In the event of a power failure, your settings and configuration will not be lost.

The configuration is stored in non-volatile memory. Non-volatile memory means your feature settings (configuration) remain in the memory despite interruptions in the power supply.

Calling features

In the event of a power failure, each unit will connect extension E4 to phone Line 1 to permit calls (except-AU models).

Calls must be dialed without a hunt group. All other functions will not be available until power is restored. We recommend that a telephone that does not depend on mains power be available for emergency use.

Mode scheduling and power interruptions

When the power supply to the system has been interrupted, the system's internal clock will continue to run for up to 12 hours. If you have enabled the automatic mode scheduling feature and it has been longer than 12 hours, the system will not function properly until the internal clock is reset via a caller ID call or with the Management software. When power is restored, it will check the time and determine the mode it should be running according to the configuration and set itself to that mode.

To check the current mode of operation:

- 1. Enter command mode by either pressing # on an analog extension phone, or *55# on IP phones (others may use *55 Send or *55 Dial).
- 2. Dial the system password + #.
- **3.** Dial 30#. The system will report the current mode.

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