FortiVoice™ v7.31
User Guide

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Table of Contents

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

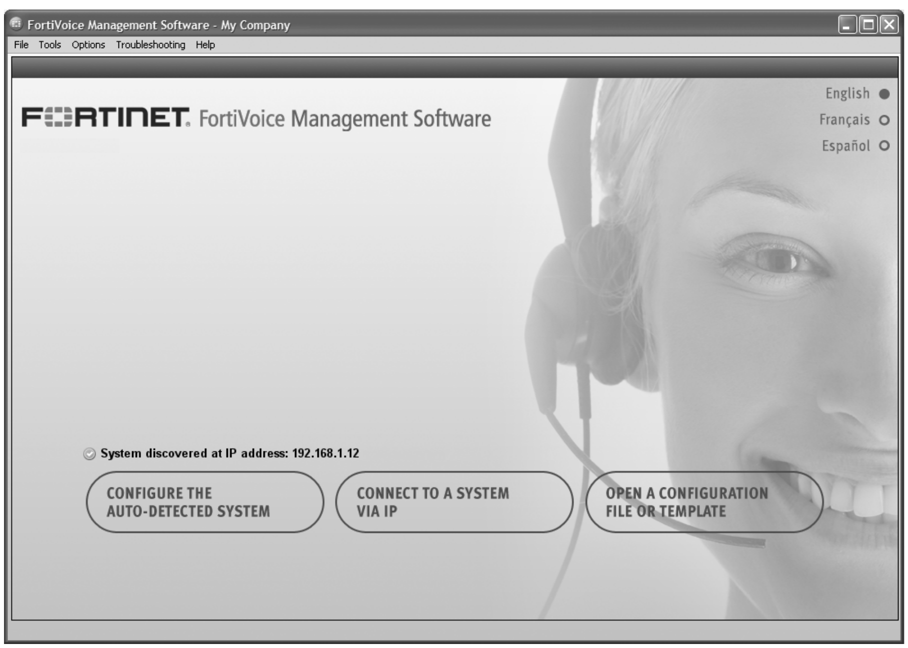
Introduction

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

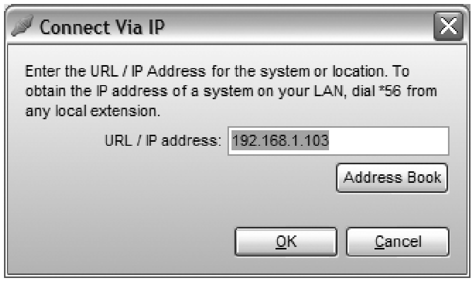
When a unit is being configured, it is locked to prevent other computers or persons using a phone from configuring changes at the same time. If you leave the software open for longer than one hour, the unit unlocks itself to allow configuration changes.

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.

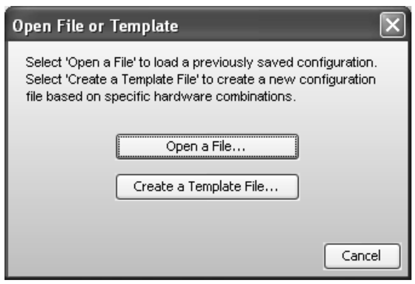


4. Select your language. You can select *English*, *Français* or *Español*.
5. Once the software detects the system, click *Configure Auto-Detected System*. The software loads the configuration from the unit, and the *About* page appears.
6. If the software was unable to detect your system, check that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

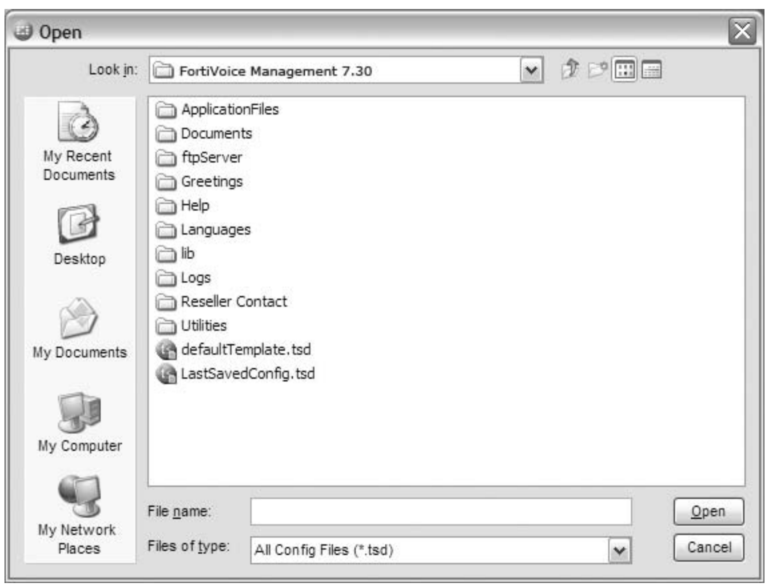
7. If auto-detection does not work:
   a. Click *Connect to a System Via IP*. The *Connect Via IP* window appears.



   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.



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



   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.

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.

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.

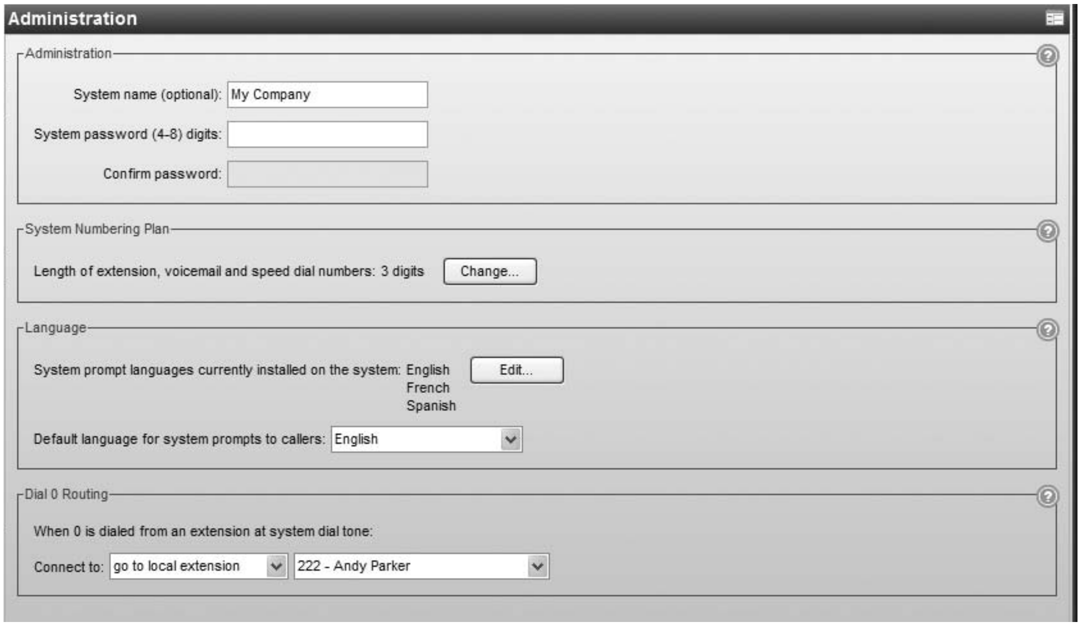
Voicemail messages

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

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.

   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.

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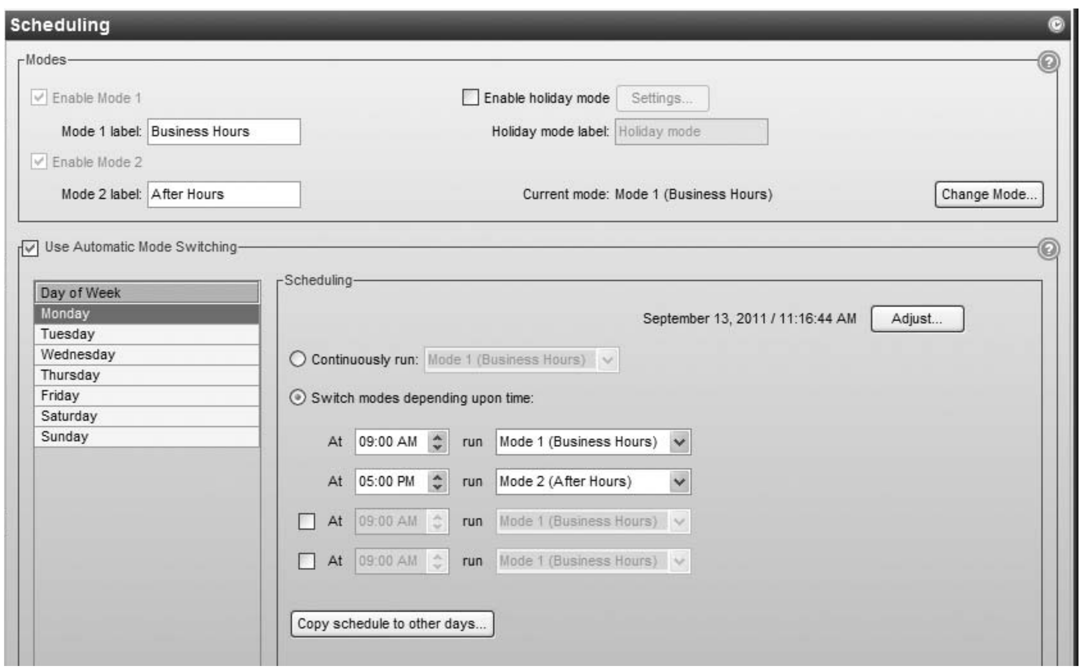
1. Select the *Scheduling* page.

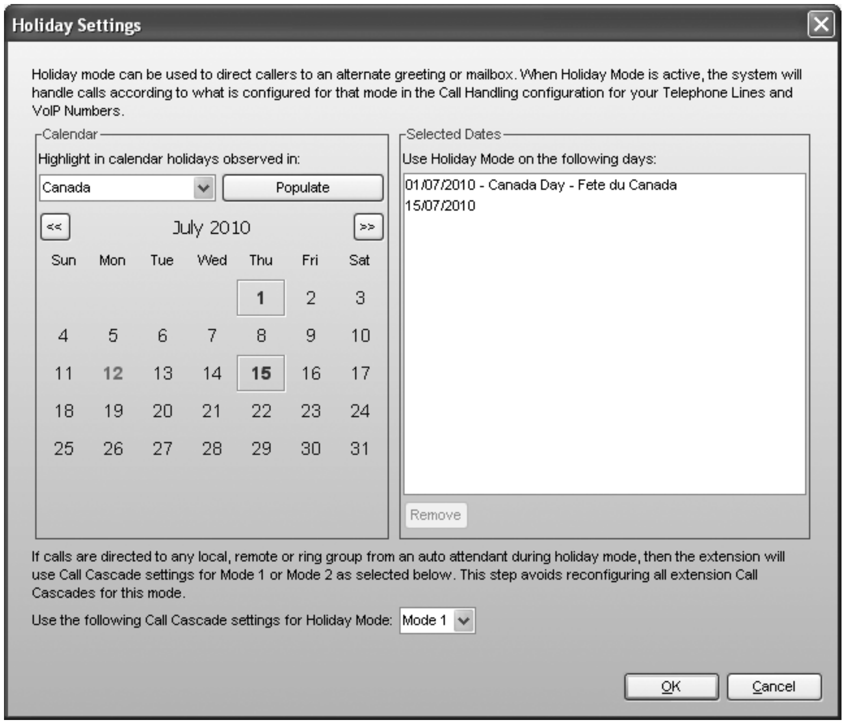


Modes

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.

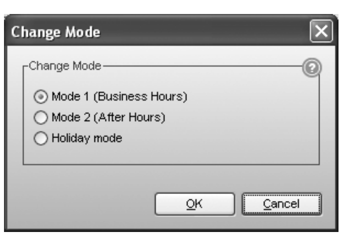


   d. Select your country in the *Populate calendar with holidays observed in* list, and then click *Populate*. The calendar will show your country's statutory holidays in grey.
   e. Using the calendar, select the month and then click the date of the holiday. The date is added to the *Selected Dates* area, 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.



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.

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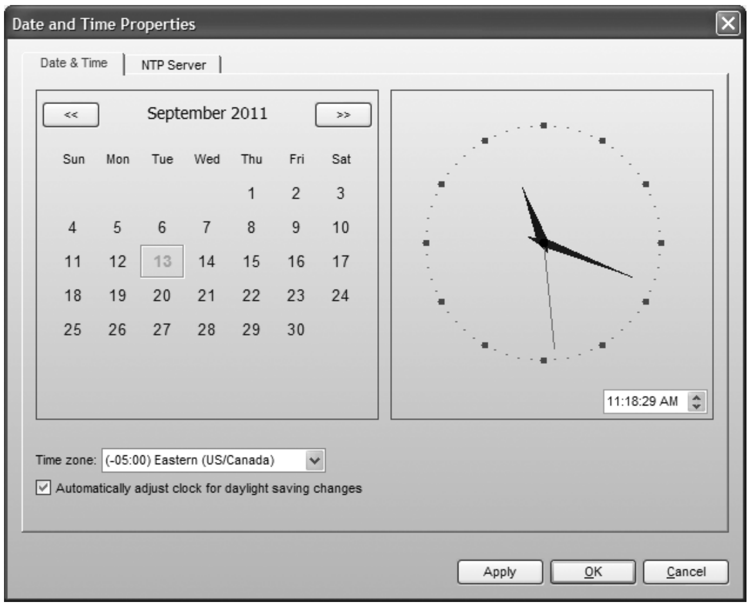
1. Click *Adjust*. You can also choose *Options* > *Set Date & Time*. The *Date and Time Properties* window appears.



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

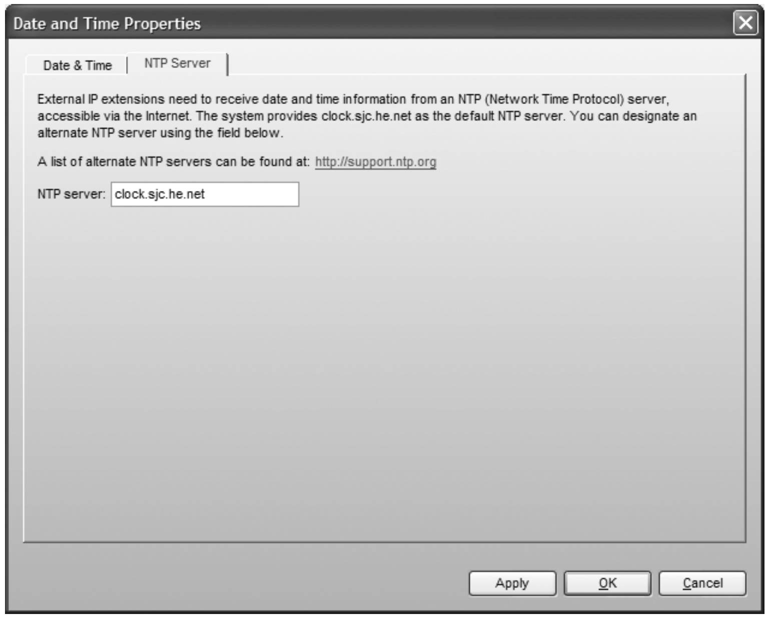
The *Time Zone* area allows you to set the time zone for the system, and whether daylight savings time is used in your region.

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.

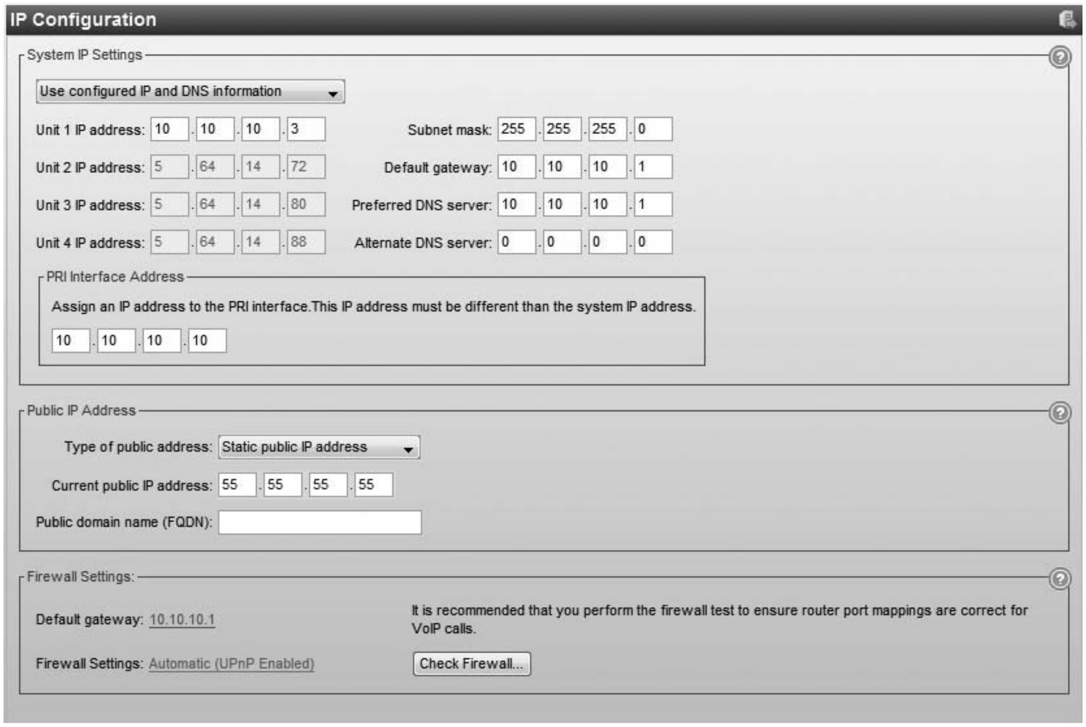


2. Enter the *NTP server*.

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.



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.
2. 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.

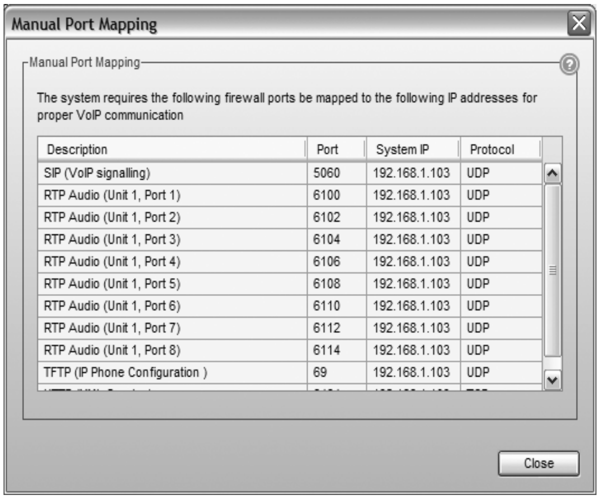
A router is a gateway between the local area network and the Internet. Most routers have a firewall to block unwanted data from the Internet. For voice data to reach the system through the firewall, port forwarding is required. Port forwarding allows the router to map ports to the IP addresses of the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

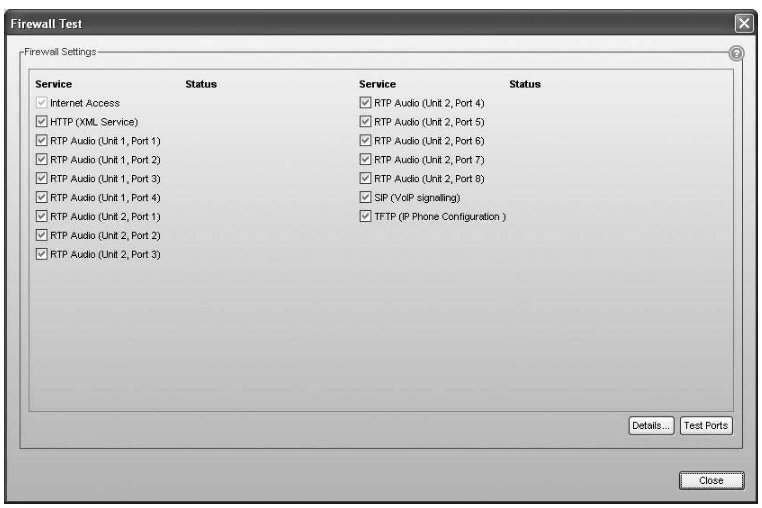
If port forwarding is required, and your router supports uPNP (Universal Plug and Play), ensure uPNP is enabled. The system will use uPNP to automatically set up port forwarding, and the *Automatic (uPNP Enabled)* link will appear. No router configuration is required.

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.



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.



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

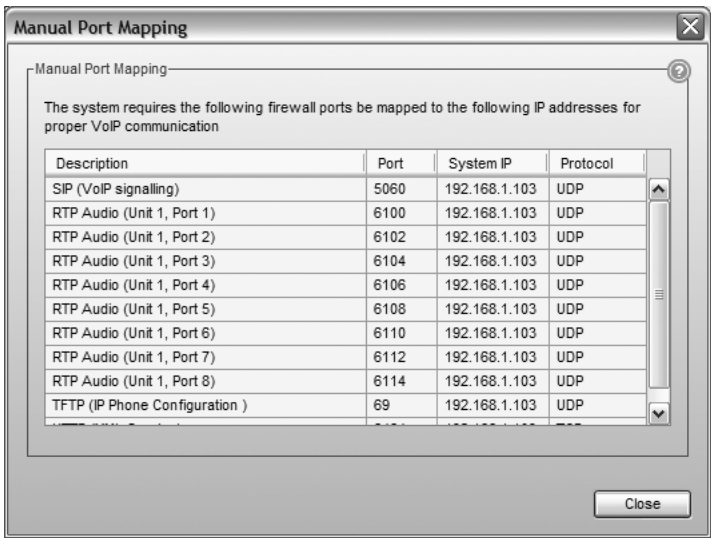
Configuring the router manually

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.



   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

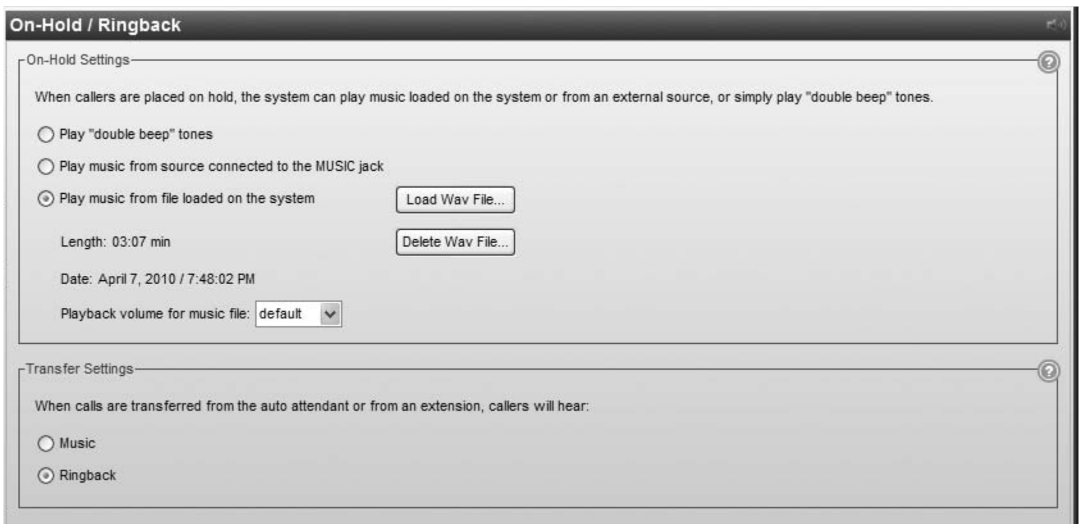
Music on hold plays when a caller is on hold, or is being transferred to an extension. The system plays a double beep tone or ringback tone by default, but you can have it play music or a recorded message instead.

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.



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:

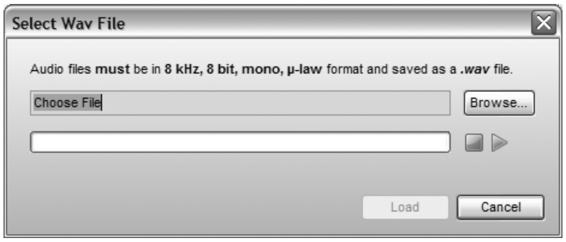
- *Play "double beep" tones* — Plays a "beep beep" sound.
- *Play music whose source is connected to the MUSIC jack* — Plays music from a CD player connected to the MUSIC jack.
- *Play music from file loaded on the system* — Plays the .wav file loaded into the system.

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.



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.

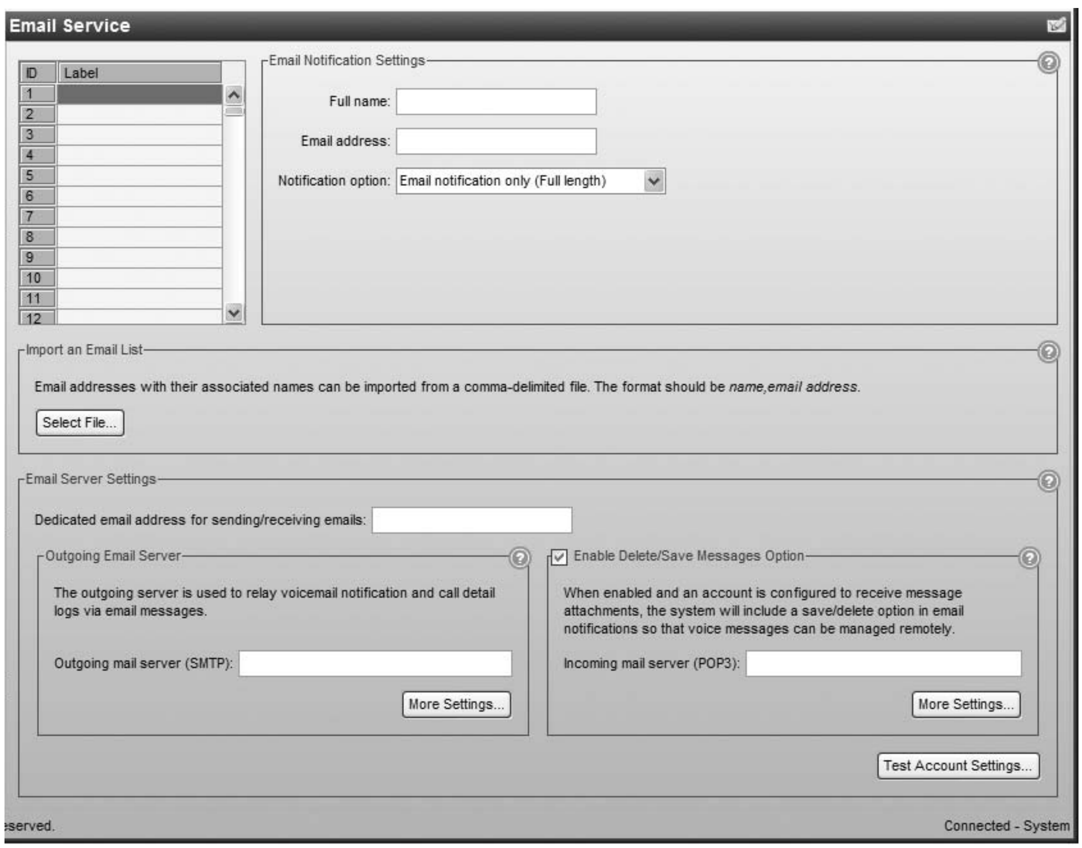
The e-mail can optionally include links to delete or save the voicemail message. If you delete the voicemail message, it will be removed from the system. If you save the voicemail message, the system will change its status from “new” to “saved”. A voicemail message that is saved will no longer activate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

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.

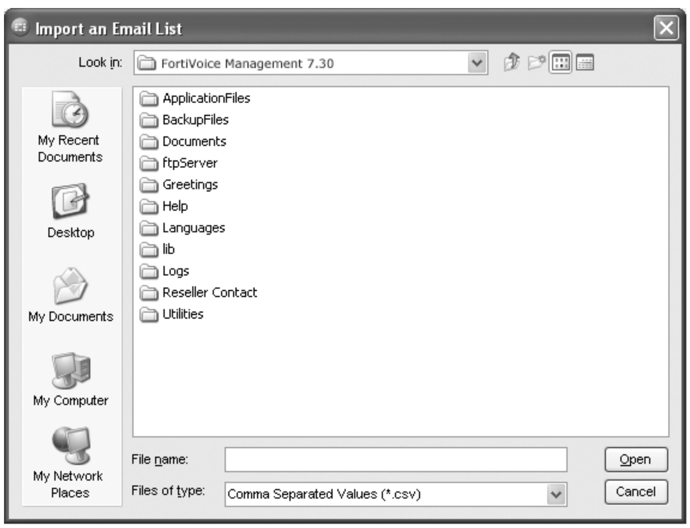
Import an email list

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.

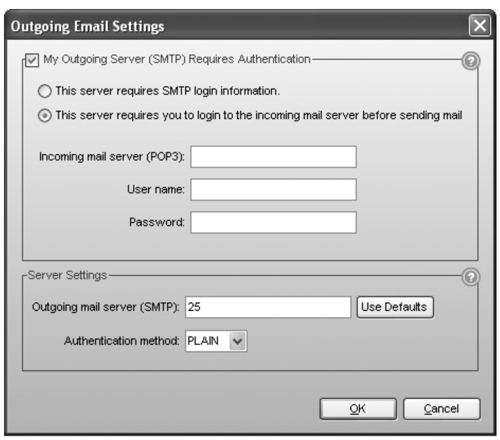
Use e-mail account information from your Internet service provider (ISP), system administrator, or alternatively, obtain one from a compatible online e-mail provider.

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.



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.

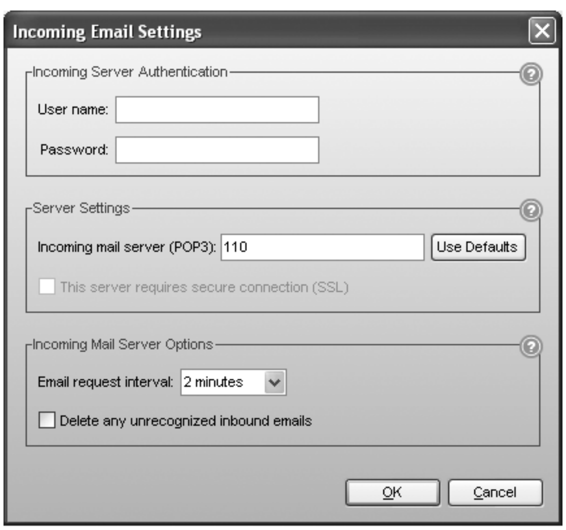
Incoming server authentication

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.



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

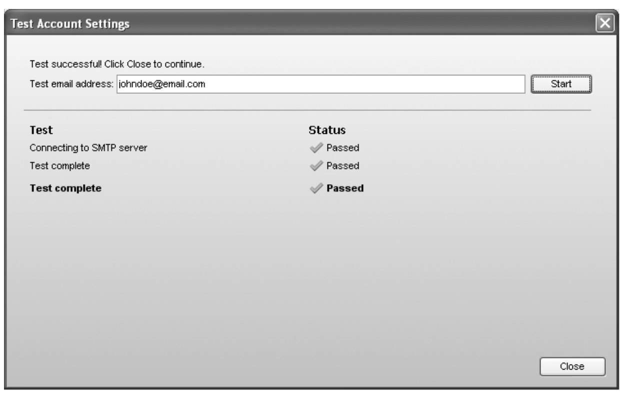
The *Incoming Mail Server Options* area allows you to specify how often the unit should check the incoming server for e-mail messages from users deleting and saving voicemail messages. The incoming server is also referred to as the POP3 server.

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.

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.



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

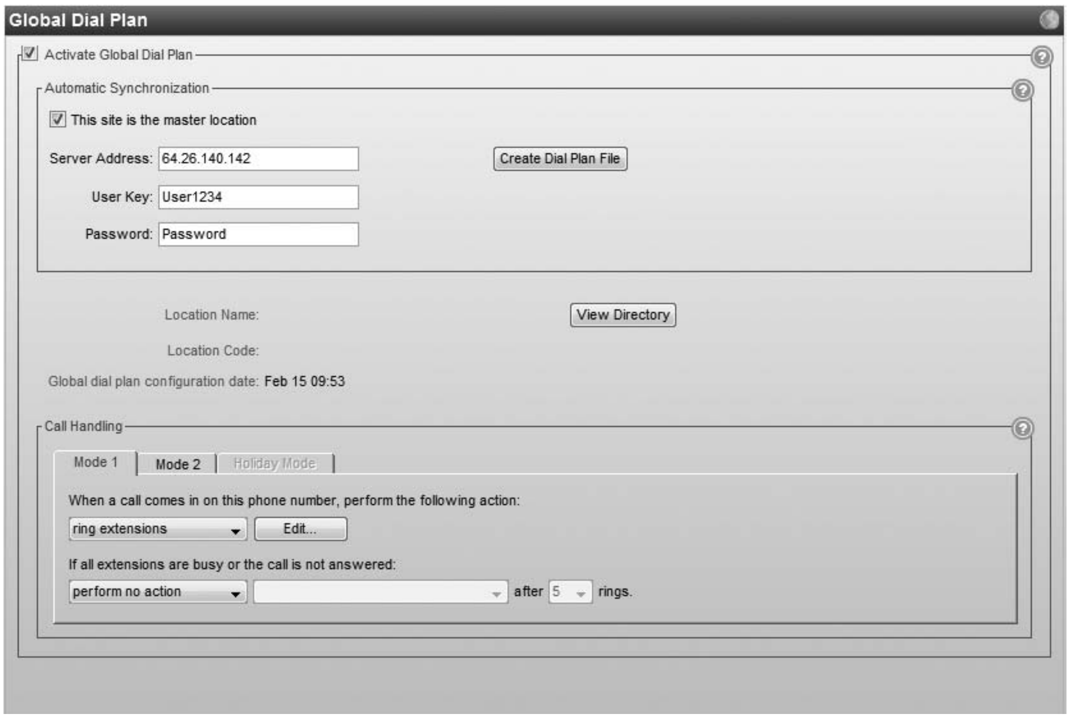
Configuring multiple locations with the global dial plan

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.

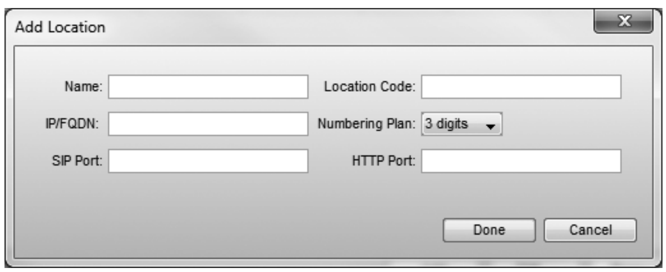


2. Check the *Activate Global Dial Plan* checkbox.
3. Check the *This site is the master location* checkbox. Enter your public IP address or the IP address of the system if you are using a VPN.
4. Create an 8-character *User Key*. The user key will be shared among the locations.
5. Create a *Password*.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

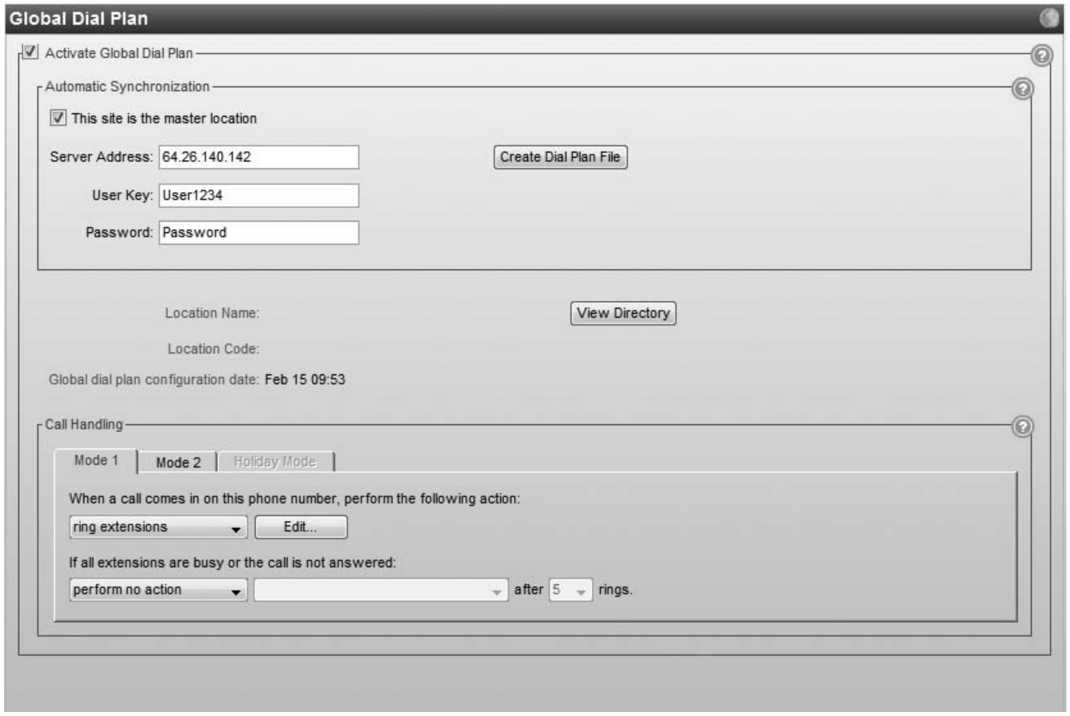
2. Click the *Add* button. The *Add Location* window opens.



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.



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*).

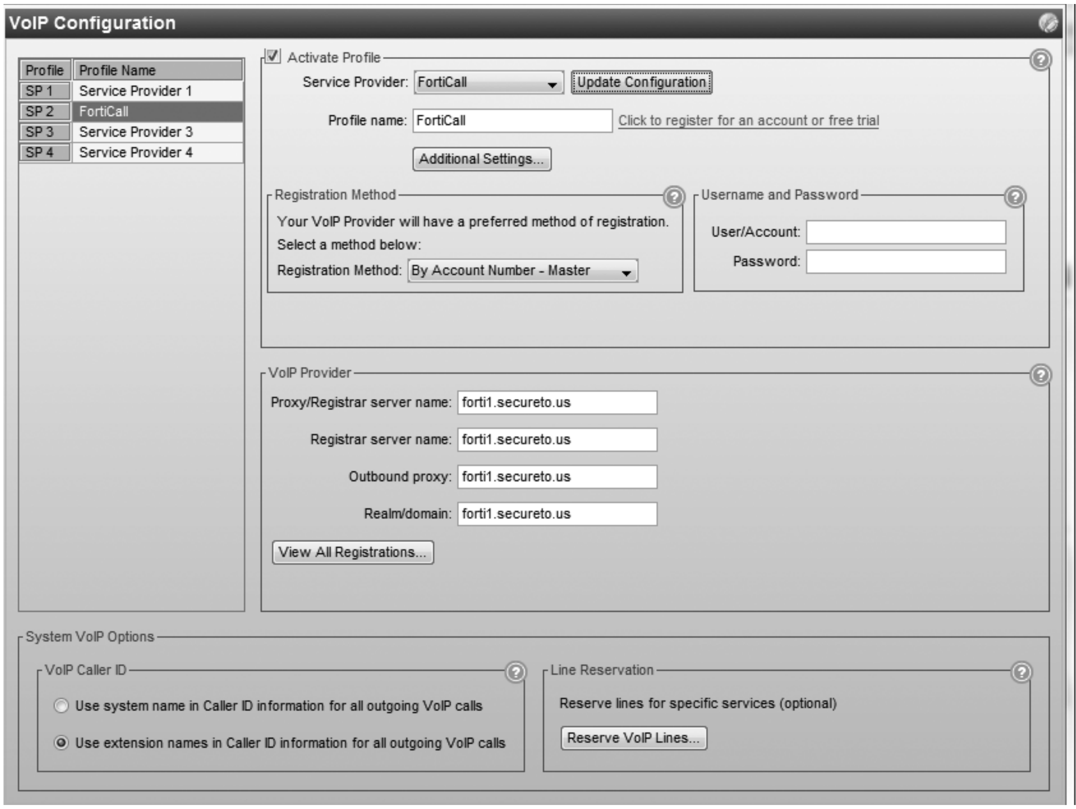
Activate profile

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.



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*

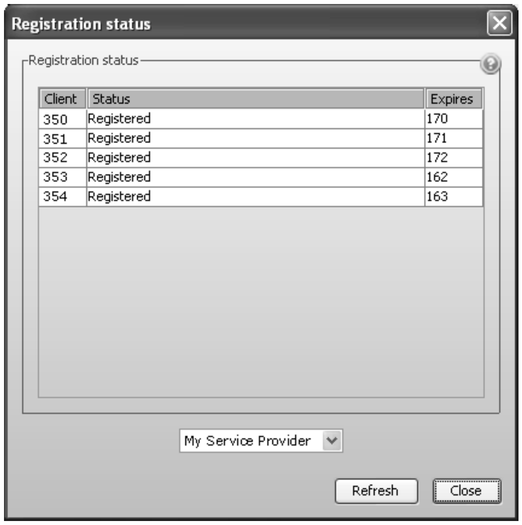
The *View All Registrations* button will allow you to confirm connection to your service provider once you have completed the configuration of your VoIP numbers. See "Viewing registration status" on page 31.

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.

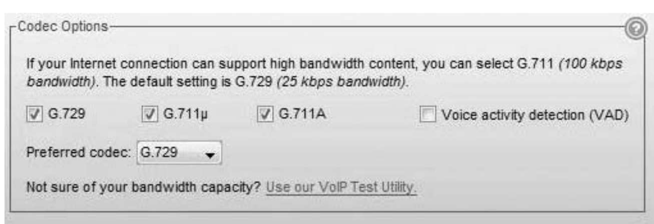


2. Choose *All Registrations* or an active profile.

- *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:



- *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.

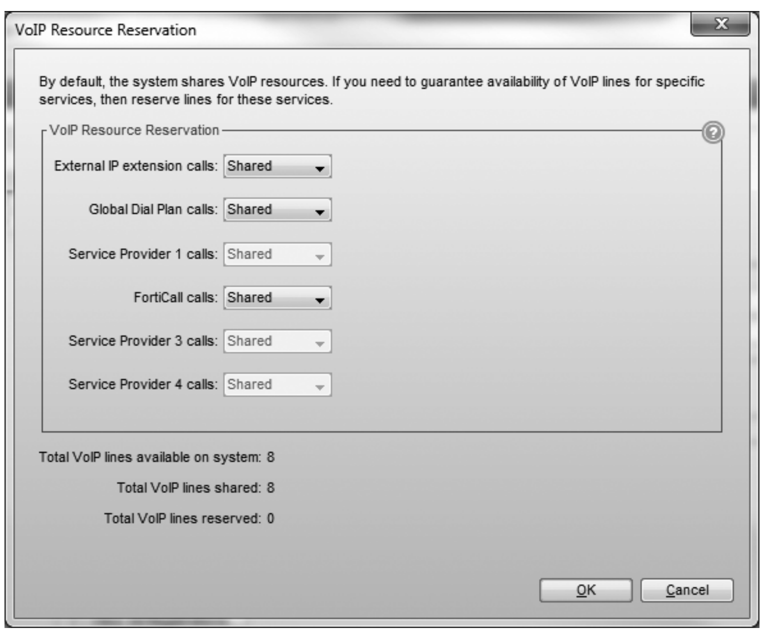
Line reservation

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.



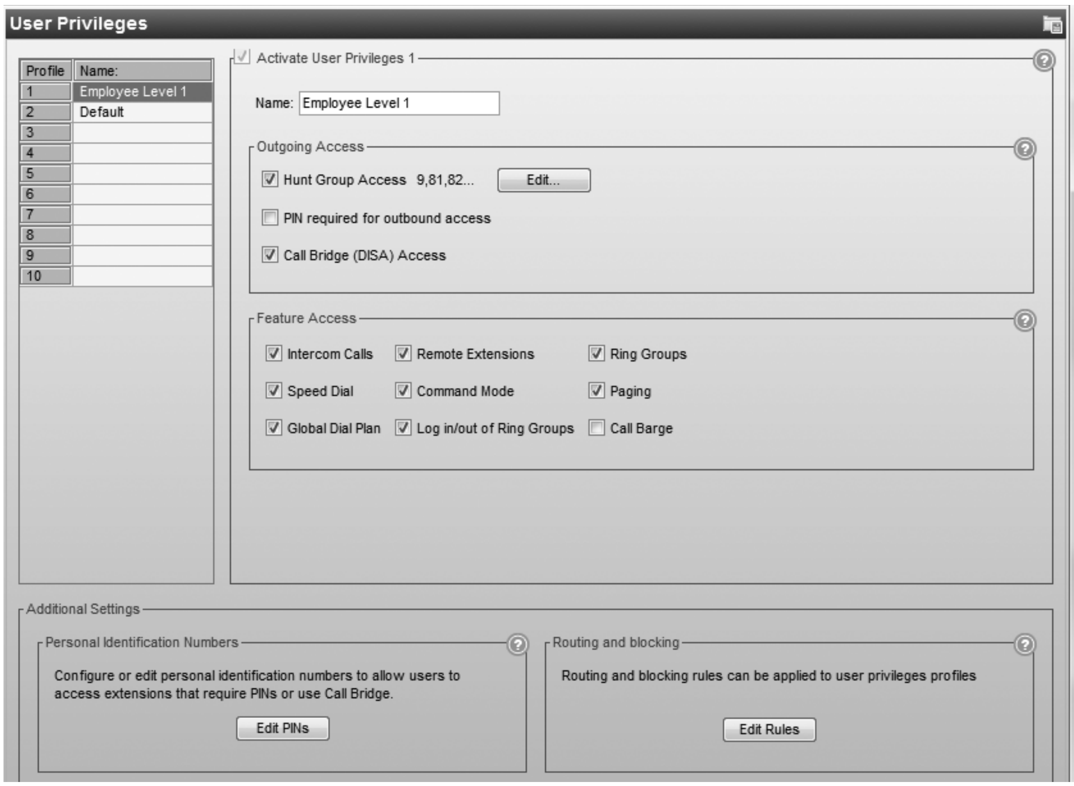
2. Select the number of VoIP lines to reserve for *External IP extension calls* using its pull-down menu.

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.



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.

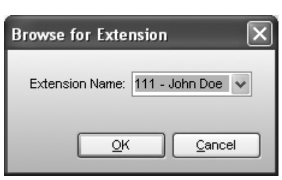
Enable password protection when PINs are used. Change the system password frequently to prevent unauthorized users from making calls or changing the configuration.
See “Administration” on page 6.

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.

   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*.

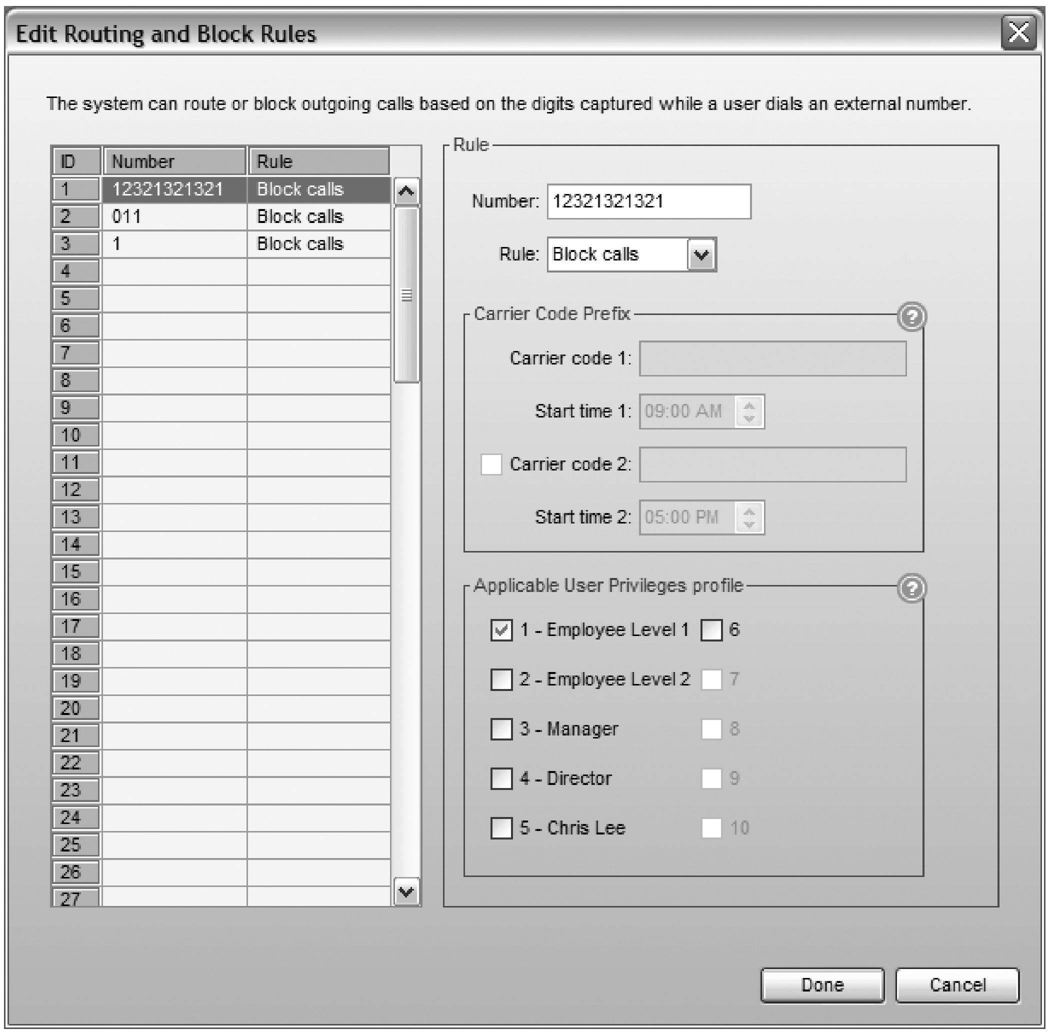
Use of Carrier Selection Prefix and Carrier Codes may require a subscription to a carrier’s discount calling plan. This service may not be available in some countries and on some telephone companies' telephone lines. Carrier Selection Prefix may not be required if a "Pre-Selection" service is provided 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.



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.

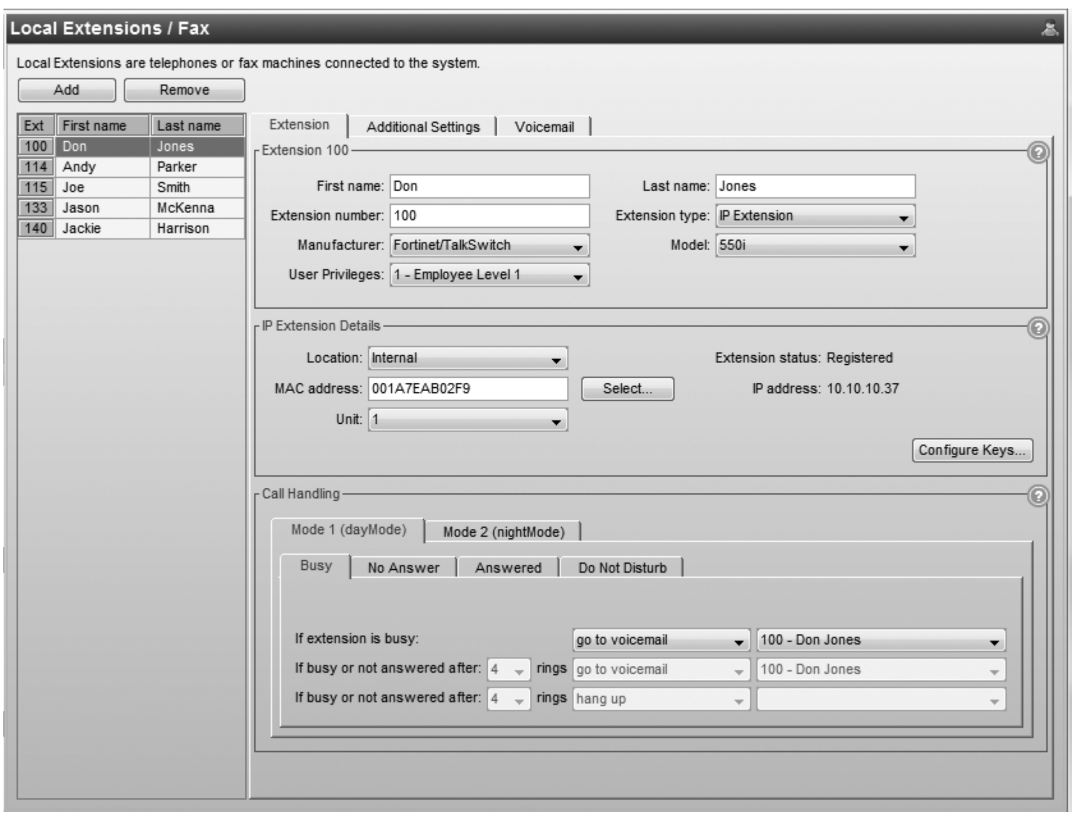
In the list of extensions on the left side of the page, manually configured IP extensions will be shown in bold, and unregistered or disconnected extensions will be shown in red.

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

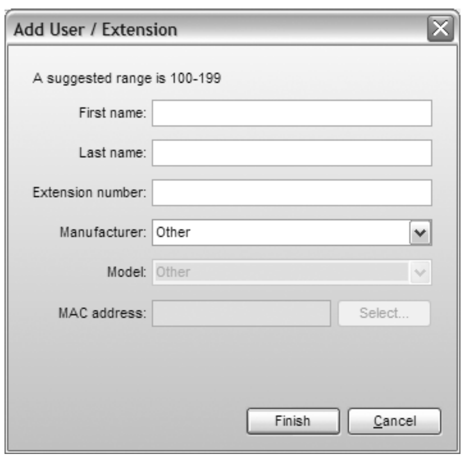
1. Select the *Local Extensions/Fax* page.



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.

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.



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.

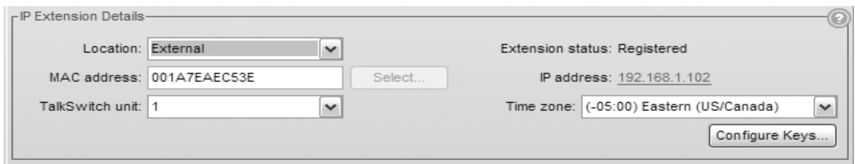
Regular extension details

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.



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.

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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.

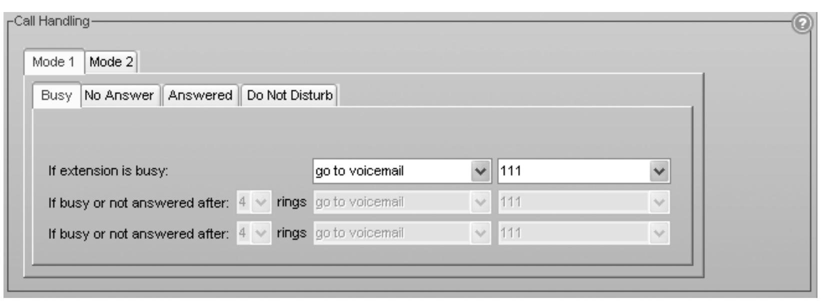
Note that the system can be configured to ignore the call cascade for an unanswered intercom call, or for an unanswered transferred call. See "Transfer Preferences" on page 132.

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.



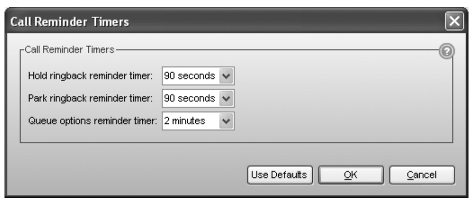
3. Set up the first alternative.
   a. Select the action in the *If extension is busy* list. Choices are:

- *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.



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 *If 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*

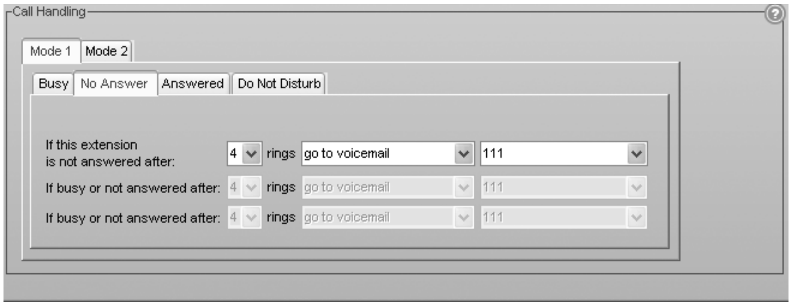
c. Select the resource, if applicable.

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.

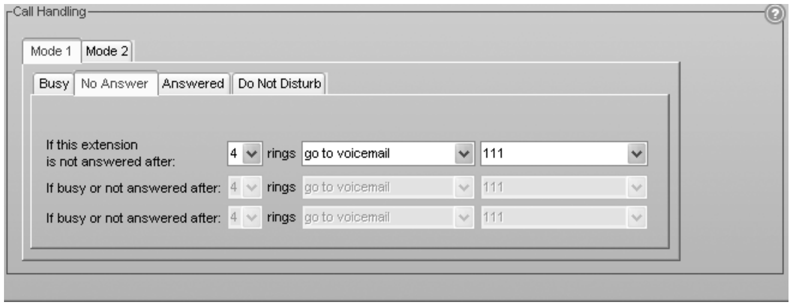


3. Set up the first alternative.
   a. Set the *If this extension is not answered after* list to the number of rings to try the extension. Choices range from 1 to 9.
   b. Select the action. Choices are:
      - *go to voicemail*
      - *go to local extension*
      - *go to remote extension*
      - *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 *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. 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.

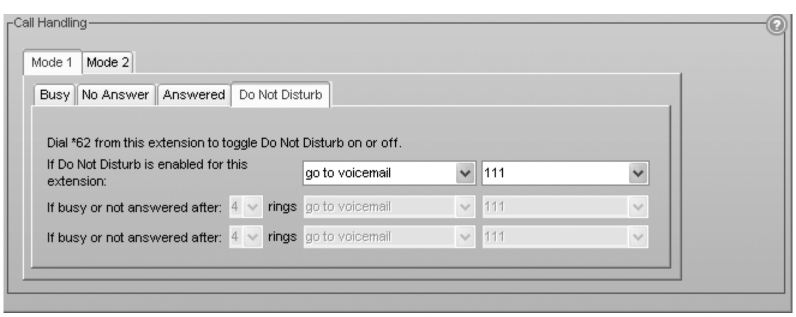


3. Set the *When a call is answered* list. Choices are:
   - *stay connected* — Transfers the call to the extension. This option disables the remaining controls in the tab. This is the default.
   - *play caller's name first* — Performs a screened transfer. The caller is prompted to say his or her name. 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.



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.

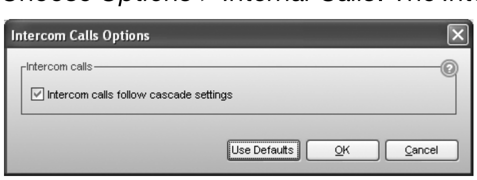
The administrator can enable or disable Do Not Disturb mode for a local extension by phone.

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.

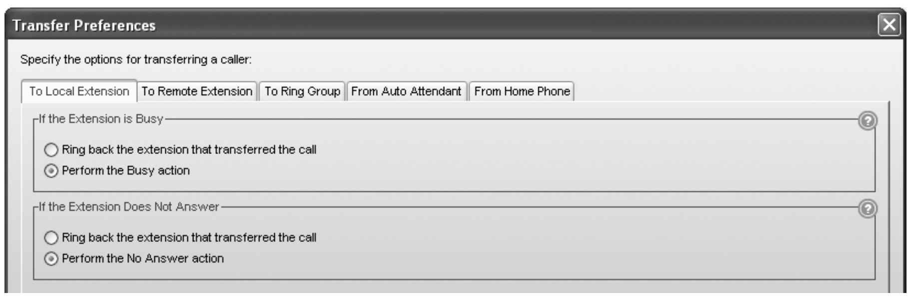


2. To have the system ignore call cascades for unanswered intercom calls, clear the *Intercom calls follow cascade settings* checkbox.
   - If the extension is busy, the caller will hear a busy tone.
   - If the extension is not answered, it will continue to ring.
   - If a local extension has call screening 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.



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.

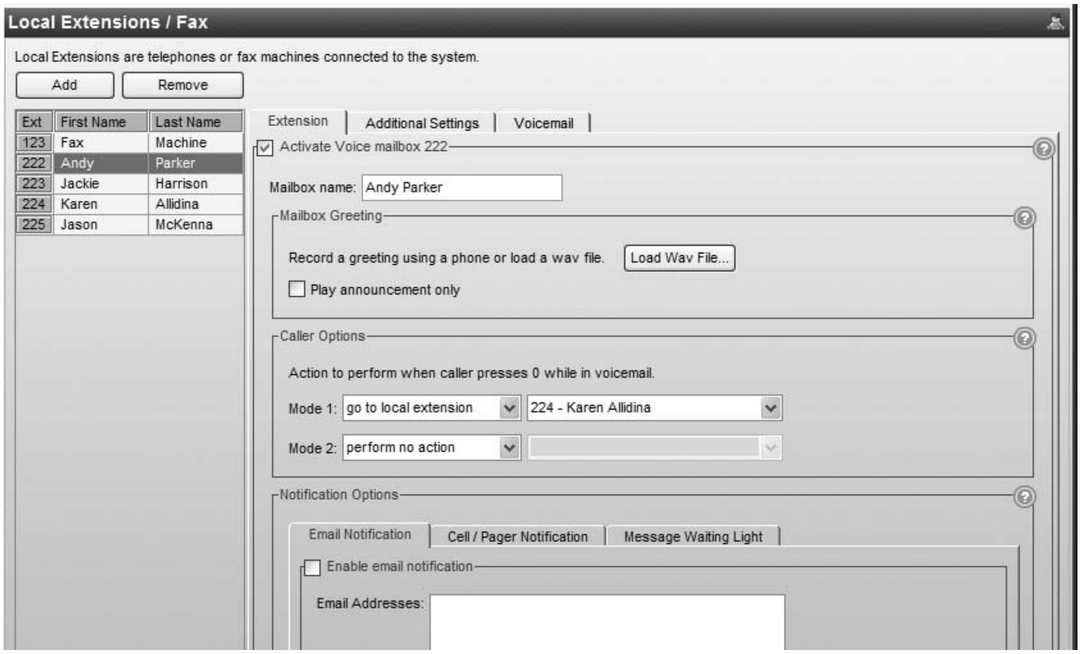
By default, each local extension mailbox is activated.

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.



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.

   - 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

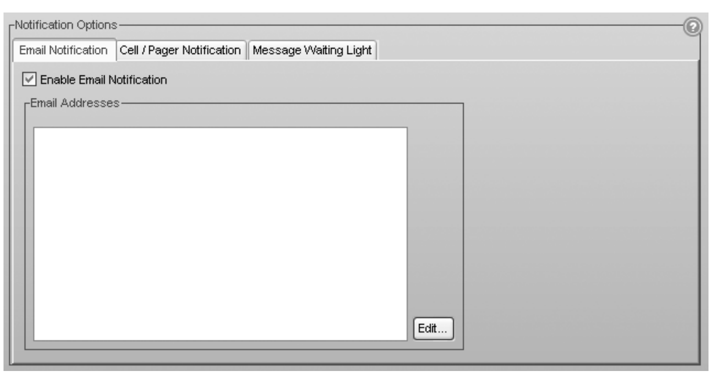
The *Notification Options* area allows you to set up voicemail notification, which tells the user if a caller leaves a message. The system can:

- 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.



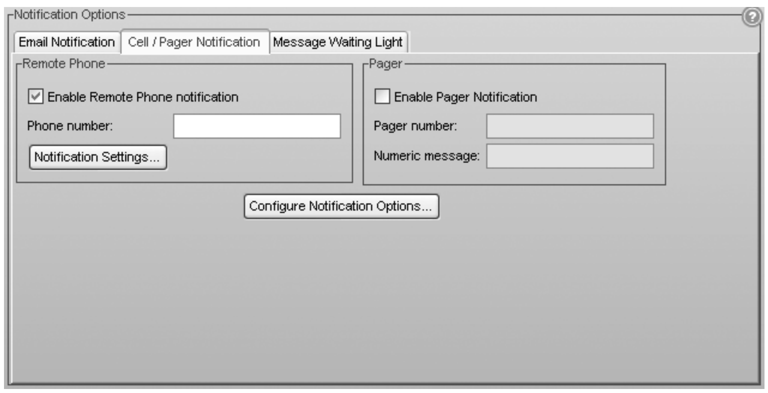
3. Select the *Enable email notification* checkbox.

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

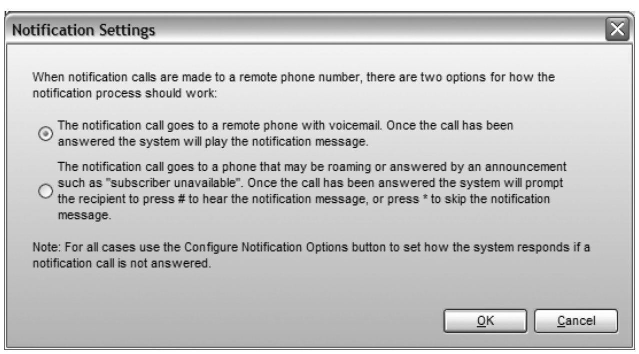
5. Select up to four e-mail addresses.

Setting up notification by phone

1. Select the *Cell/Pager Notification* tab.



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.



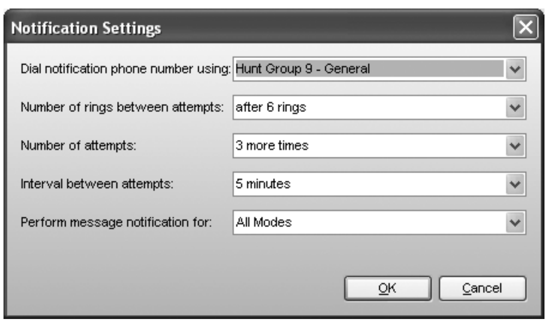
5. Select the notification setting. Choices are:
   - Select the first option to have the system play the notification message once, when the phone is answered. It will then consider notification to be successful. Note that if there is any answer other than a busy tone, (e.g. voicemail, *"subscriber 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.



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*.

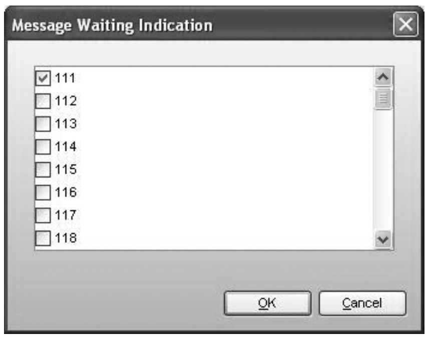
Setting up message waiting light activation

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.



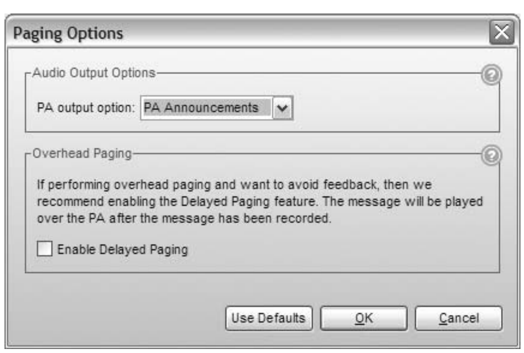
3. Select the extensions.

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.

1. 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.



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

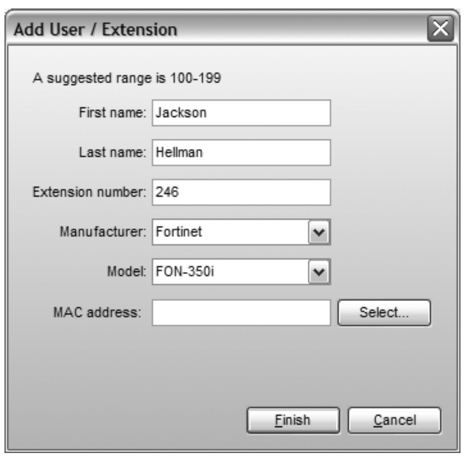
1. Connect a network cable between the LAN port on the phone and your network (i.e. router or LAN connection) and then connect power to the phone. Refer to the phone's quickstart guide for specific instructions.
2. If the phone is connected to the same network as the system, and an extension has been 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.

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.



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*.

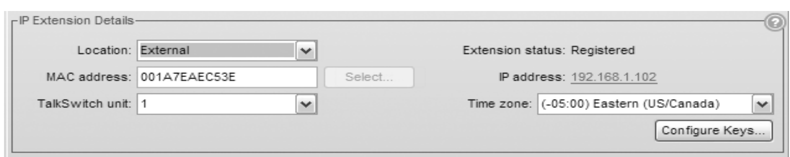
- You can enter the phone/base *MAC Address* manually. This is the required method if you are setting up an external IP extension. The MAC Address is a 12-digit alphanumeric string located in the barcode on the bottom of the phone and the box the phone came in.

  MAC: 001A7EA75DD5

  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



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.

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#*.

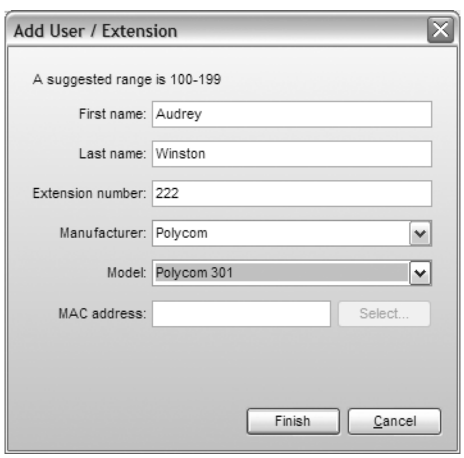
- *Pickup ext* — Do not select a resource. Press the button or softkey, dial an extension, then dial # (or softkey again) to answer a call from an outside number or extension ringing the selected extension. Lights or icons are not used for Pickup keys. Note: You can also pick up a call at a selected 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.

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.



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

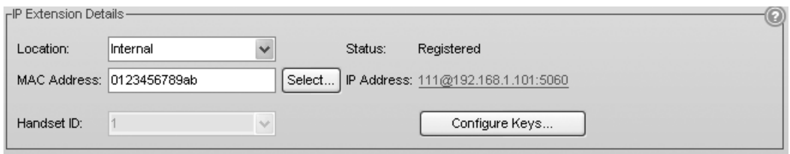
10. Click *Finish*.

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



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.
2. 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.
4. All VoIP lines are shared by default. You can also reserve VoIP lines for external IP extensions. See "Line reservation" on page 32.
5. 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.

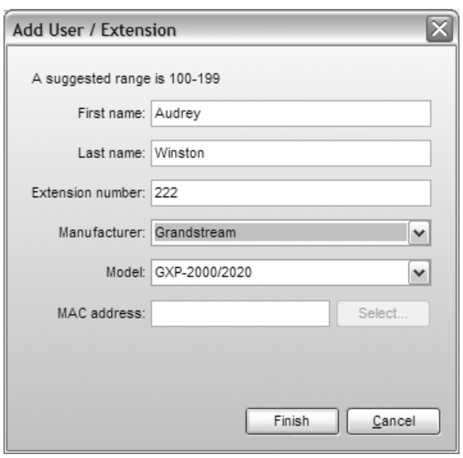
For more information, refer to the "Troubleshooting" section in your phone user guide.

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.

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.



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.

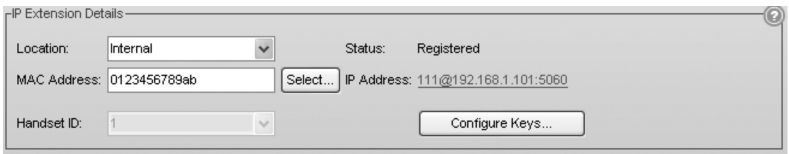
Click *Finish*.

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



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:

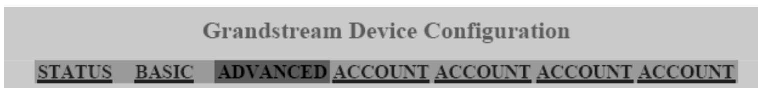
1. Ensure your system is connected to a network. See “Connecting to a network” on page 176.
2. Ensure that you have set up a public IP address for the system. See “Public IP address” on page 14.
3. Ensure the router is configured at the system’s location. There should be no need to make any adjustments 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.
5. 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

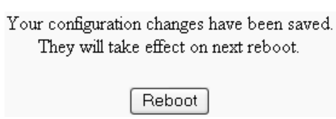
1. Note the IP address that appears on the screen of the Grandstream phone.
2. In a web browser, enter the IP address in the *Address* field.
3. Enter the password. By default this is *admin*. When the unit configures the phone, it changes the password to 23646 (which spells “admin” on the telephone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



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

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.

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



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.

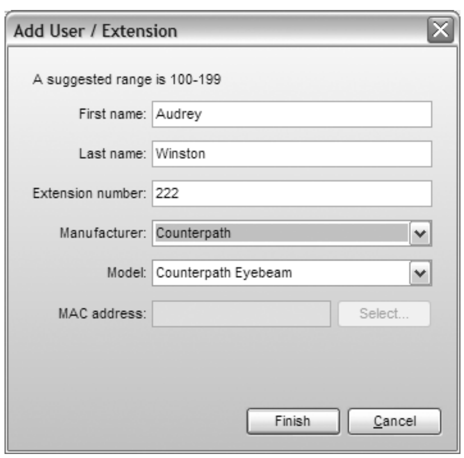
9. Test to ensure the phone is configured properly by dialing another local extension.

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.

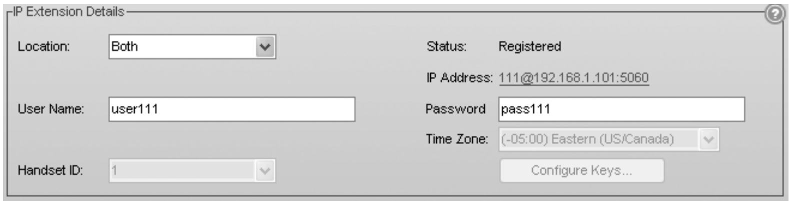


7. Assign an *Extension number.*
8. Select the *Manufacturer* and *Model***.** Click *Finish*.
9. In the *Extension* tab, select the language for prompts heard by the user of the extension in the *System Prompt Language* list. Setting the prompt language will also change the language for text displayed 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



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

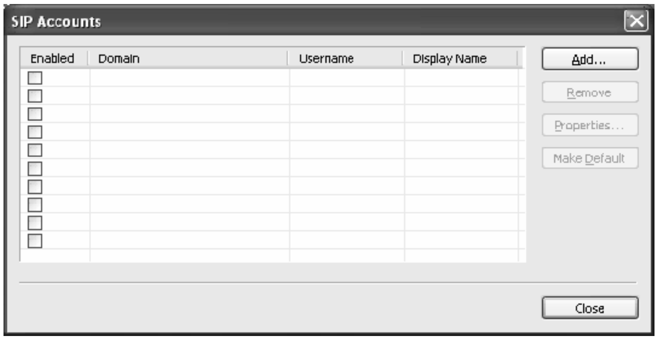
If you are setting up an external IP extension:

1. Ensure your system is connected to a network. See “Connecting to a network” on page 176.
2. 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.
5. 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.

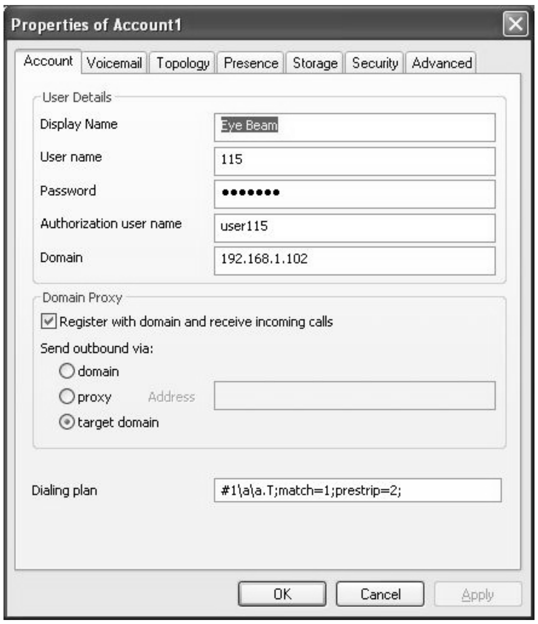
Configuring a new SIP account

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.

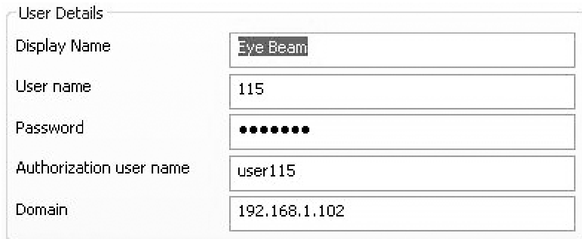


2. Click the *Add* button. The *Properties of Account1* window appears.



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.

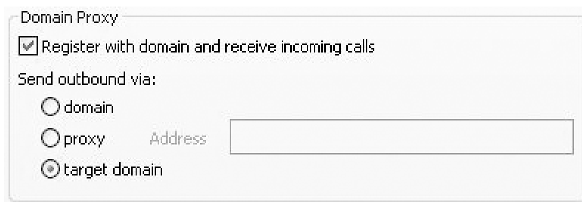
e. If setting up an internal IP extension, set *Domain* to the IP address of the unit acting as local proxy. If setting up an external IP extension, set *Domain* to the public IP address or public domain name of the system.



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.



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].

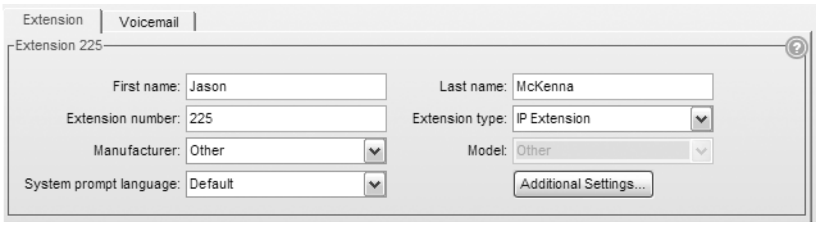
Updating eyeBeam software

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.



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.

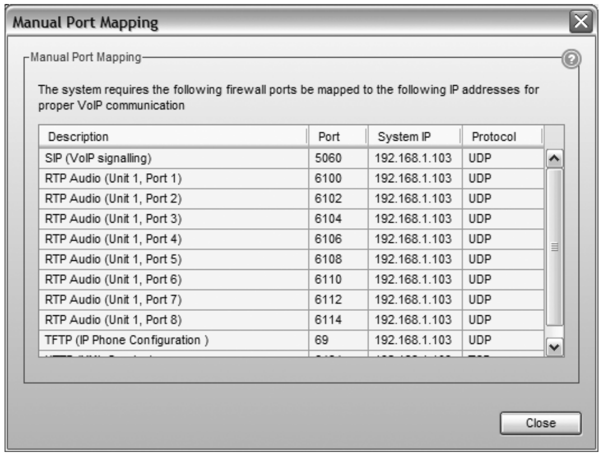
A router is a gateway between the local area network and the Internet. Most routers have a firewall to block unwanted data from the Internet. For voice data to reach the system through the firewall, port forwarding is required. Port forwarding allows the router to map ports to the IP addresses of the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

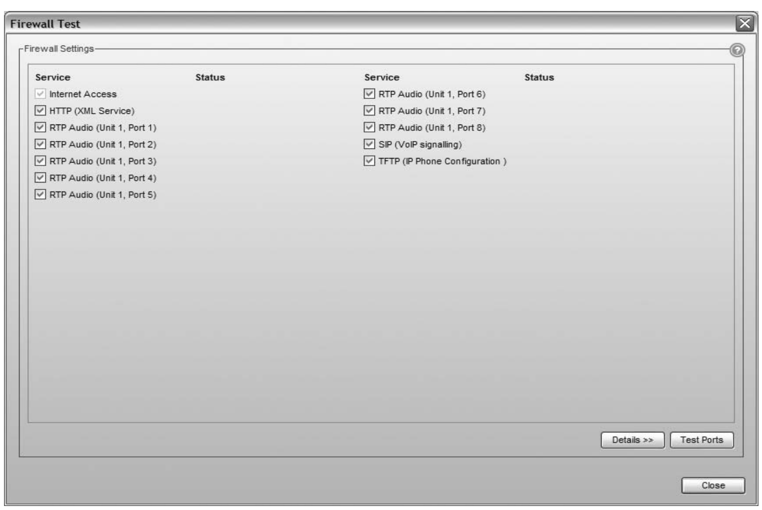
If port forwarding is required, and your router supports uPNP (Universal Plug and Play), ensure uPNP is enabled. The system will use uPNP to automatically set up port forwarding, and the *Automatic (uPNP Enabled)* link will appear. No router configuration is required.

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.



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.



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

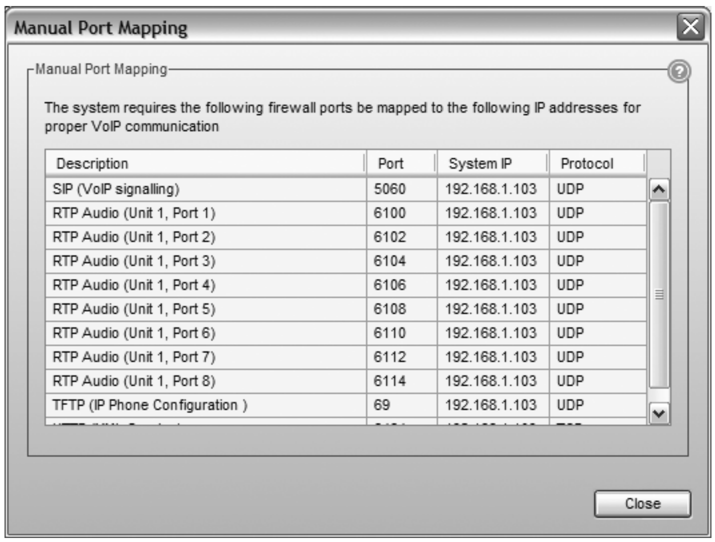
Configuring the router manually

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.



   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.

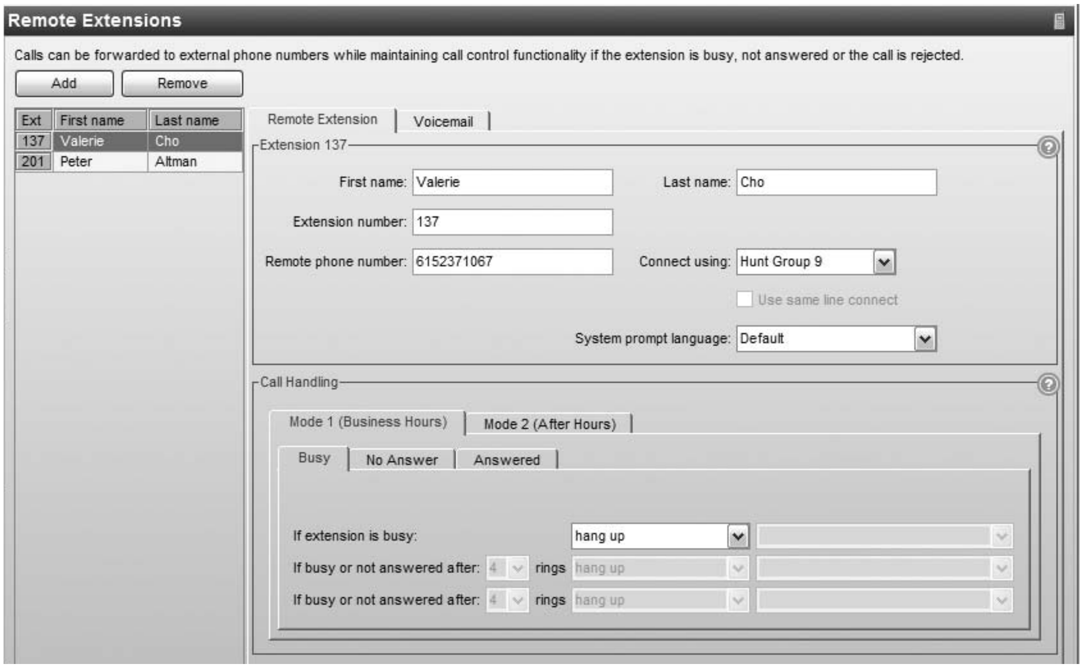
If the system is configured to perform a screened transfer, the user is connected to the other extension and asks whether they want the call. If so, the user dials ** + *4* to complete the transfer. If the other person doesn't want the call, the user dials ** + *5* to cancel the transfer and return to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

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

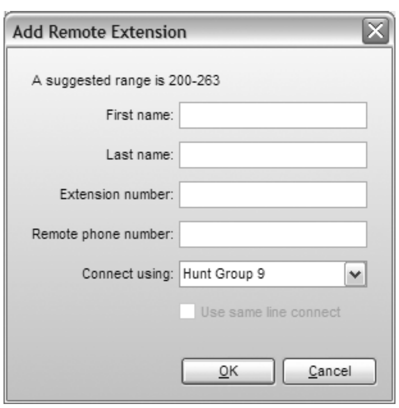
Remote extensions are designed to operate with local major telephone service providers. The feature may not function correctly with some telephone and mobile operator's networks, especially for international phone numbers and mobile phones roaming internationally.

Remote Extension tab

1. Select the *Remote Extensions* page.



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



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.

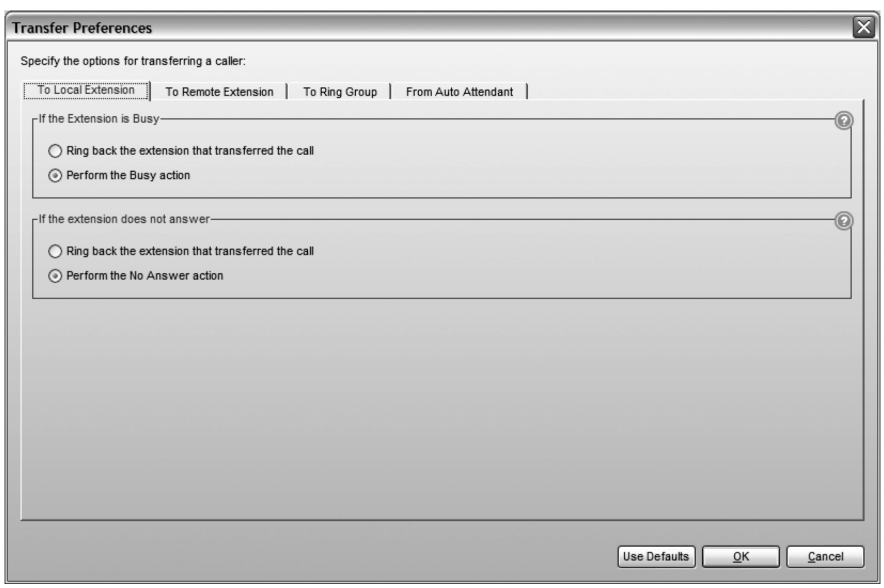
The same line connect feature is not compatible with some telephone company lines.

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.



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

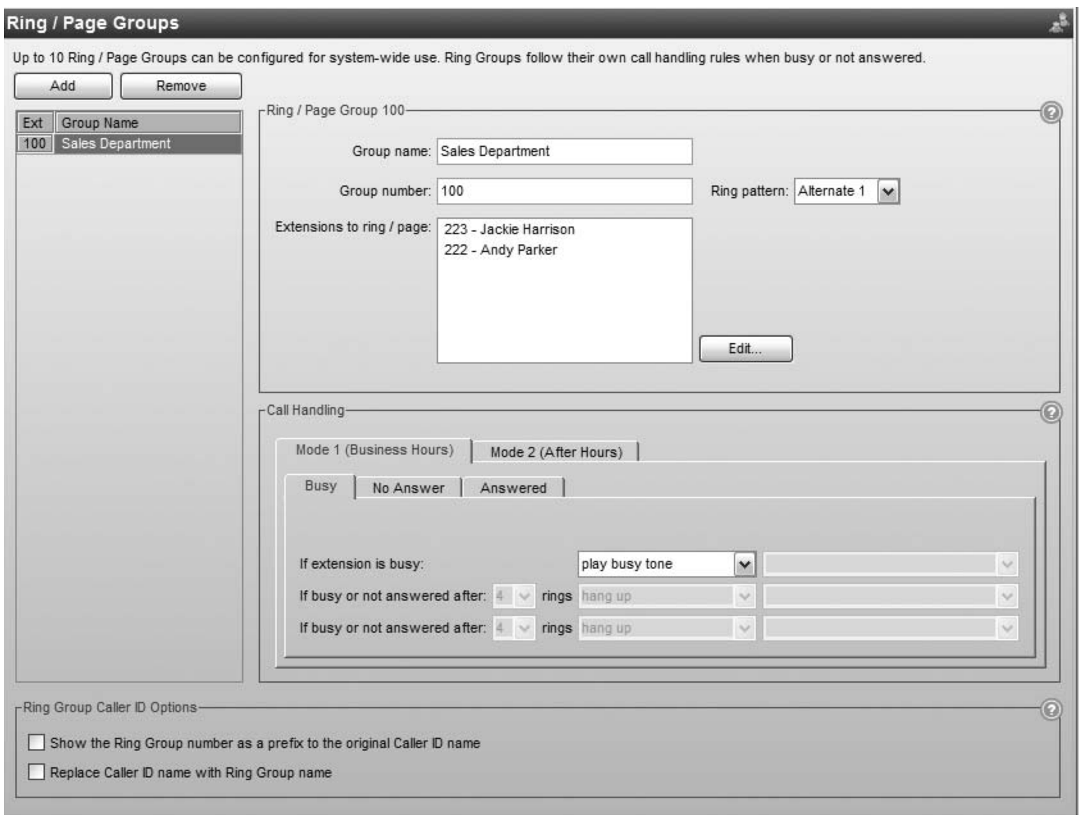
A ring group is a group of local extensions that ring in unison. Local extensions and auto attendants can dial a ring group. A page group is a group of extensions that can be paged in unison. Page groups require telephones that support group paging.

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

Ring/page groups have two main uses:

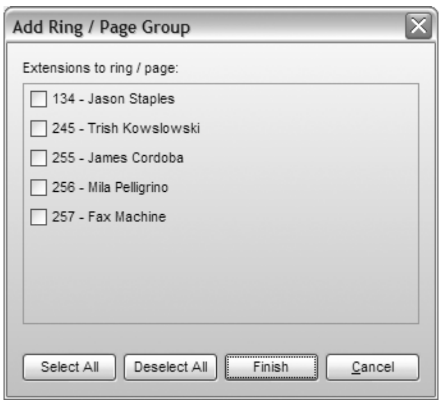
- A ring/page group can reach a group of employees. For example, ring/page group 301 can ring the sales group at extensions 111, 112, 113, and 114. When a customer calls the sales group, the first available salesperson answers for the group.
- A ring/page group with a different ring pattern can differentiate 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.



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

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*.



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:

1. 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”.

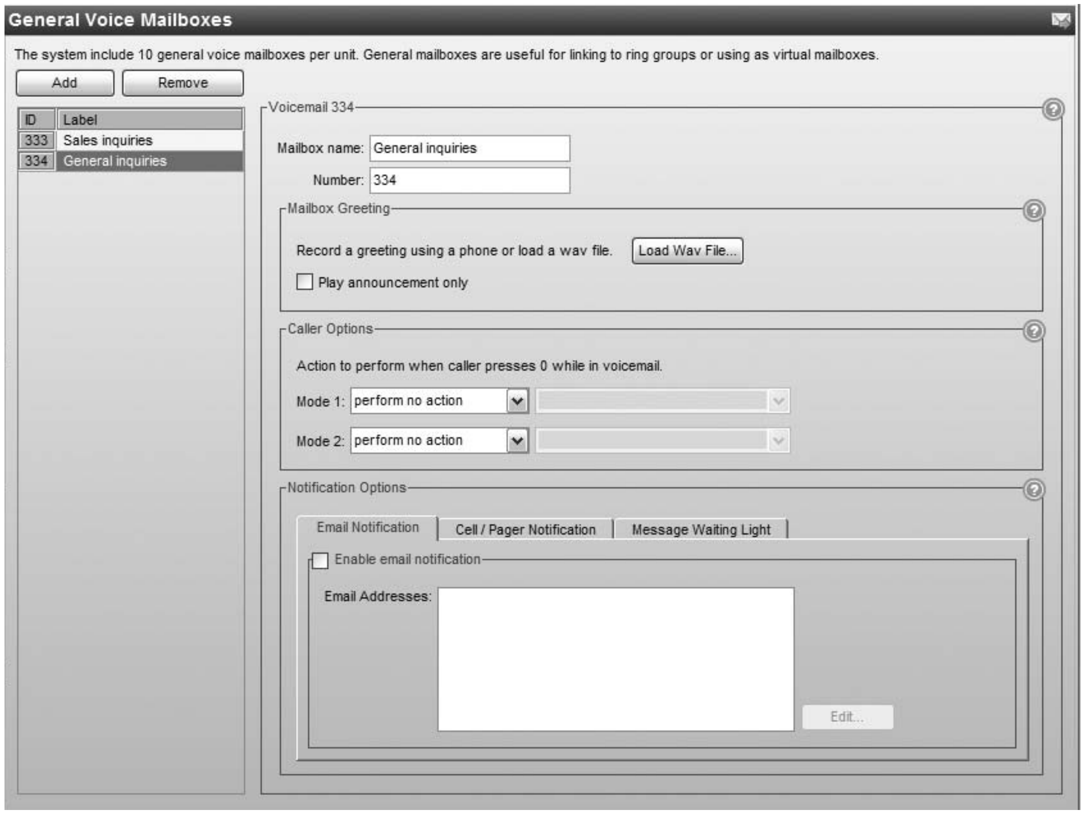
General Voice Mailboxes

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.

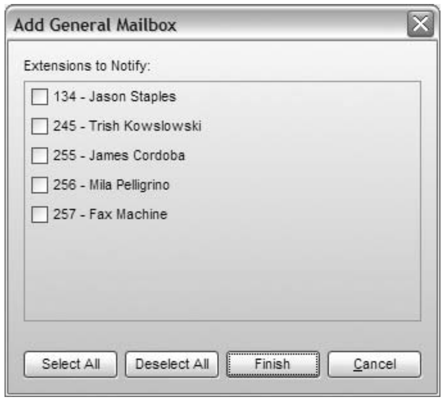


Note: In the *Caller Options* area, the screen may refer to when a caller presses *9* depending on the region.

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

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*.



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.

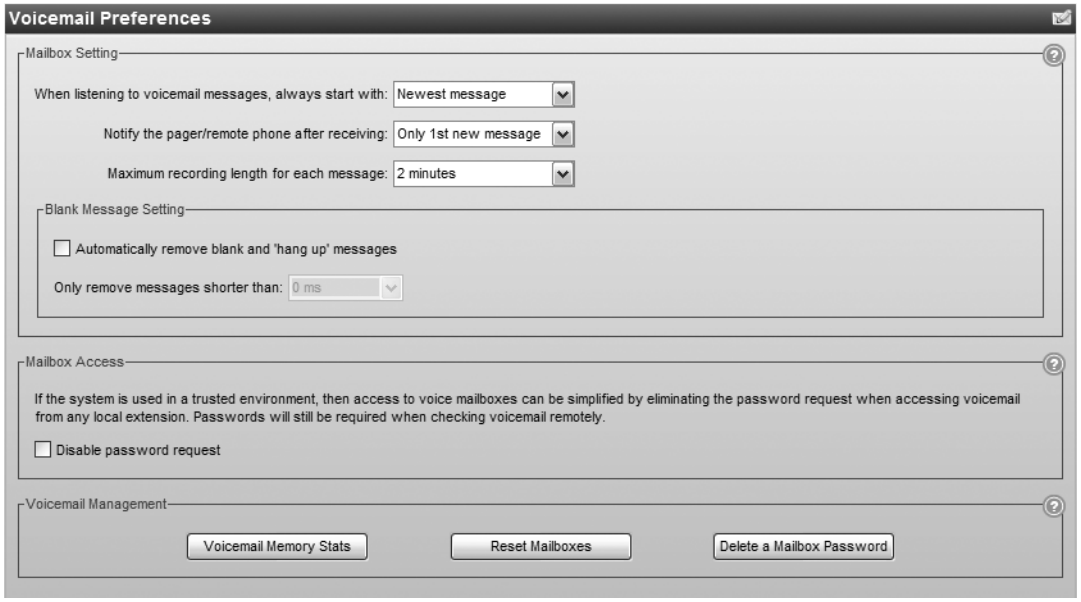
Notification options

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.



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

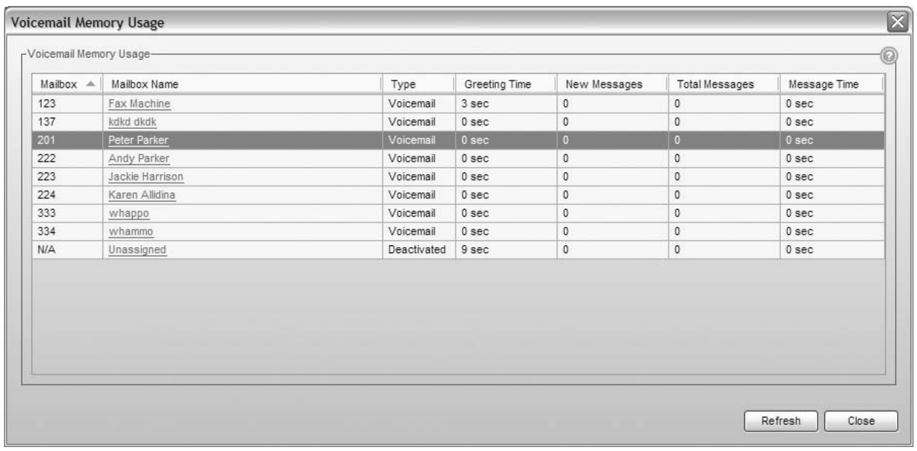
The *Voicemail Memory Usage* window shows how much recording time is taken up by voicemail greetings and messages.

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*.



Unassigned messages

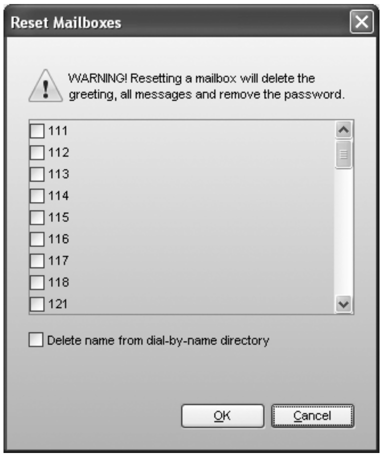
Due to deactivation of mailboxes or errors, the system may display messages currently not assigned to a mailbox. Clicking on the *Unassigned* link will allow you to either download these files to your PC as .wav files, or delete them. The files will be stored in a folder in the Management software program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



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*.

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

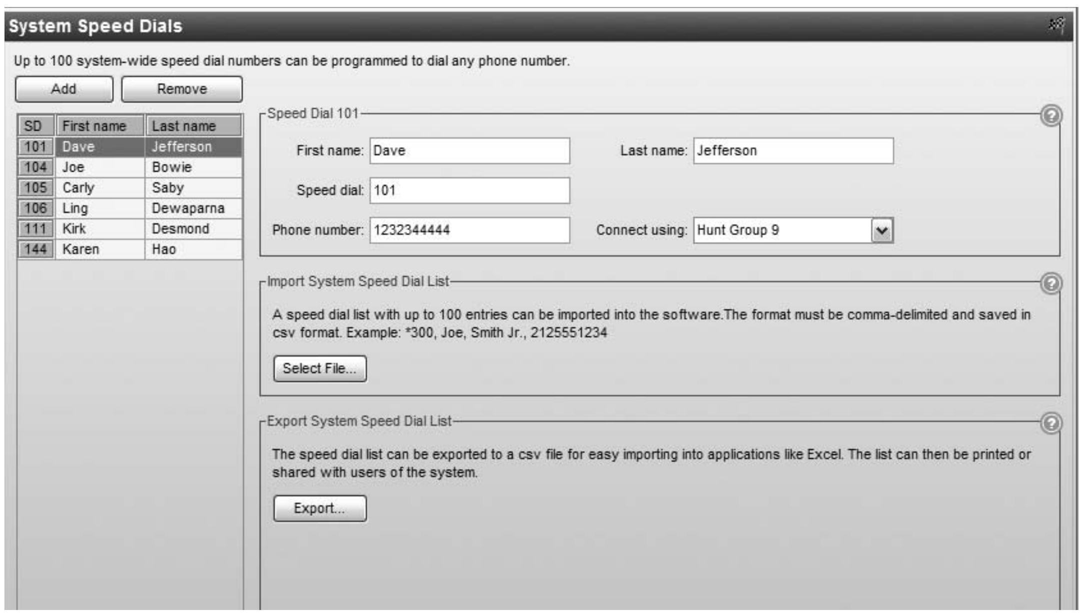
System speed dials allow the user to quickly dial an outside phone number from any local extension by dialing a speed dial number. Because the speed dial numbers are maintained within the system, they don't need to be programmed into each individual phone. If a new client is added or an existing client's 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.



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.

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

4. Assign a *Speed dial* number.
5. 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

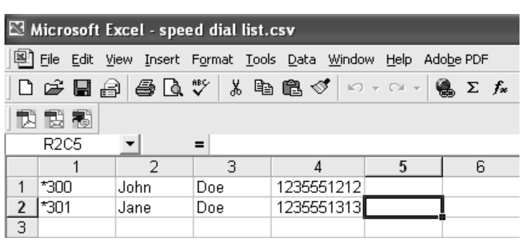
You can import a speed dial list created with a text editor like Microsoft Notepad or a spreadsheet program like Microsoft Excel. Each entry in the file requires the speed dial number, first name, last name, and phone number. If you're using a text editor, The values must be separated by commas. The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

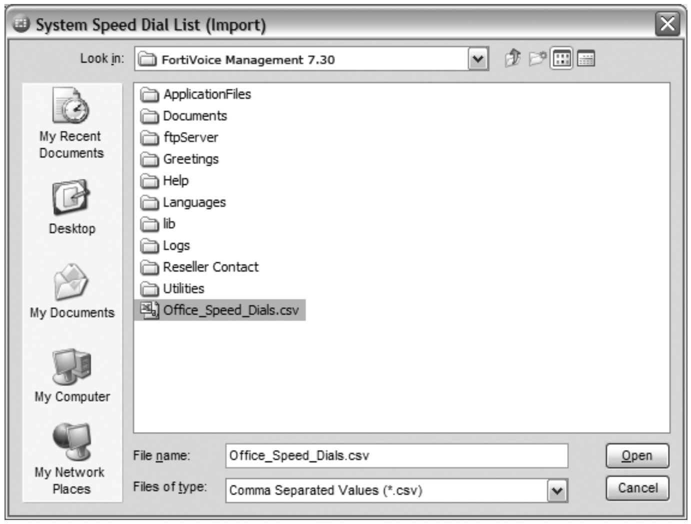
If you're using a spreadsheet program, put the speed dial number, first name, last name and phone number each in their own column.



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.



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).

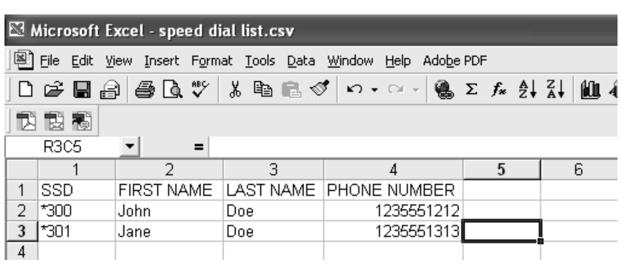
Export system speed dial list

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:

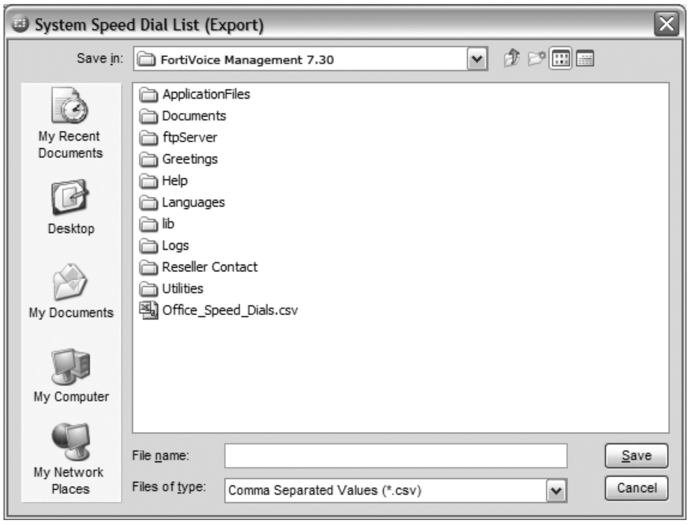
```
“INDEX”,“SSD”,“FIRST NAME”,“LAST NAME”,“PHONE NUMBER”
“1”,“300”,“John”,“Doe”,“1235551212”
“2”,“301”,“Jane”,“Doe”,“1235551313”
```

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



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.



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.

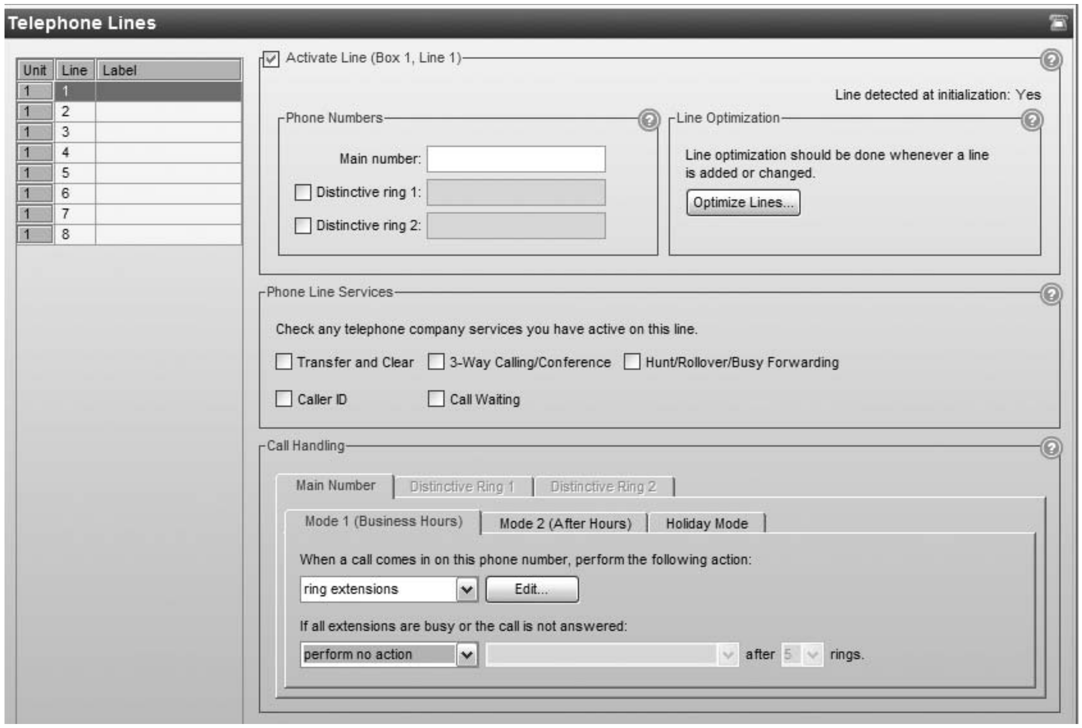
For more information, see "Caller ID (or CLID) Based Routing" on page 103.

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.

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.

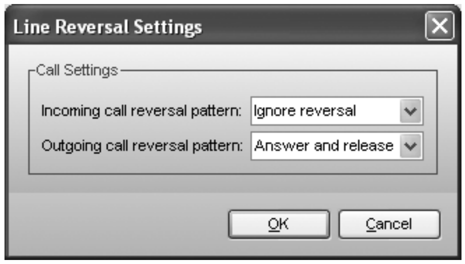
Line reversal settings

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.



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

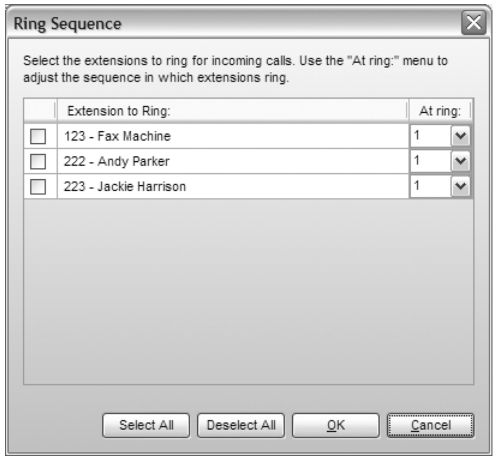
The *Call Handling* area allows you to set up what happens when a call comes in from this line. Calls can be handled differently in each of the three modes. Calls can be sent directly to an extension, or they can ring extensions, go to an auto attendant, ring group, voice mailbox, announcement or VoIP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



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*.

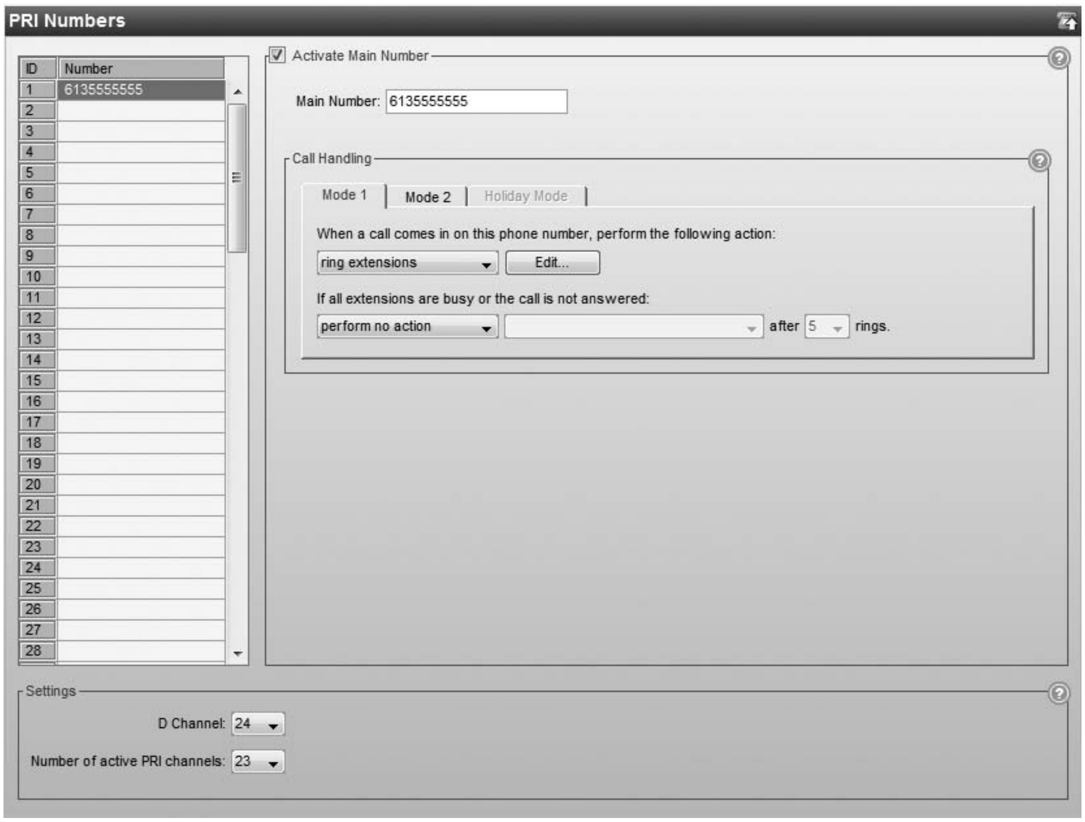
PRI Numbers

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.



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

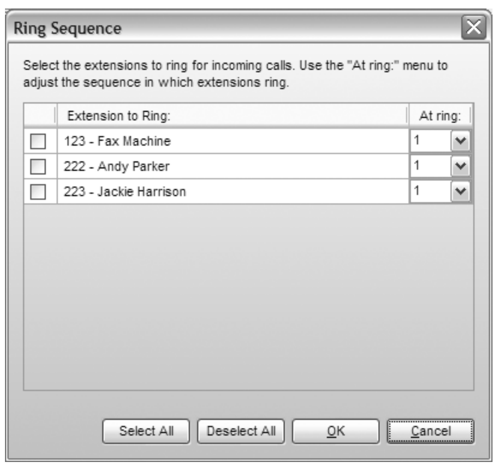
The *Call Handling* area allows you to set up what happens when a call comes in from this line. Calls can be handled differently in each of the three modes. Calls can be sent directly to an extension, or they can ring extensions, go to an auto attendant, ring group, voice mailbox, announcement or VoIP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



     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

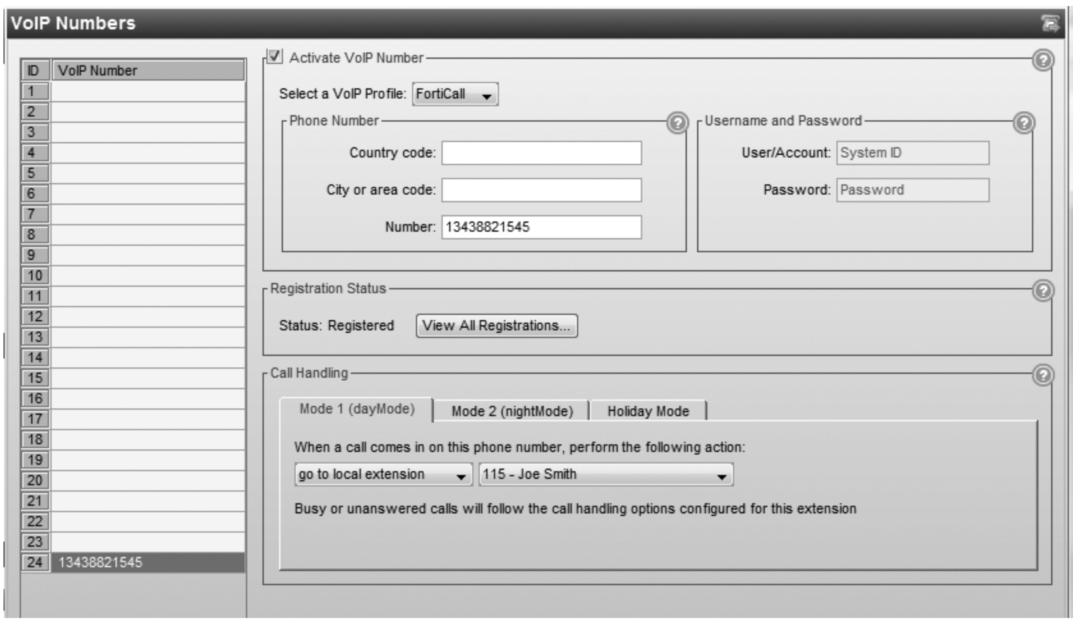
A VoIP number is like a telephone number. It allows a caller to dial the system through a VoIP service provider.

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.



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

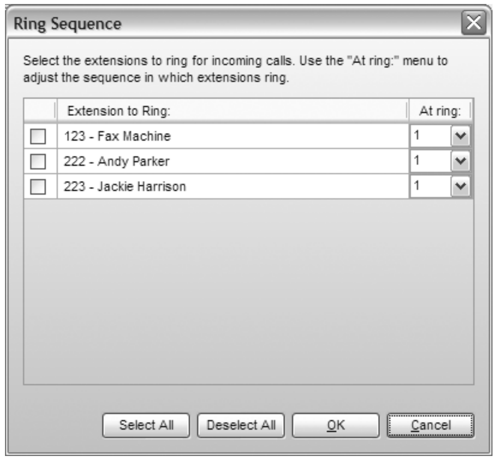
The *Call Handling* area allows you to set up what happens when a call comes in from this line. Calls can be handled differently in each of the three modes. Calls can be sent directly to an extension, or they can ring extensions, go to an auto attendant, ring group, voice mailbox, announcement or VoIP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



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.

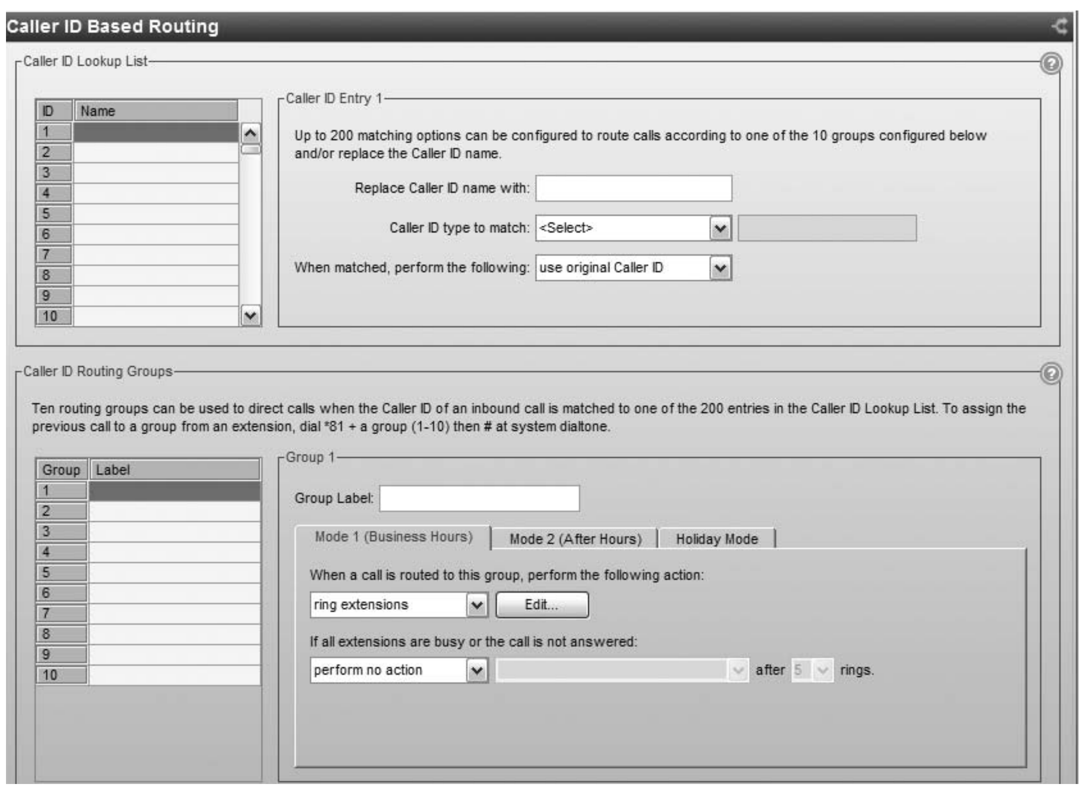
For example, if the call is from an important client, the call can be routed directly to the president's extension. If no caller ID is present, the call can be routed to voicemail.

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 *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 *Caller ID Based Routing* page.



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

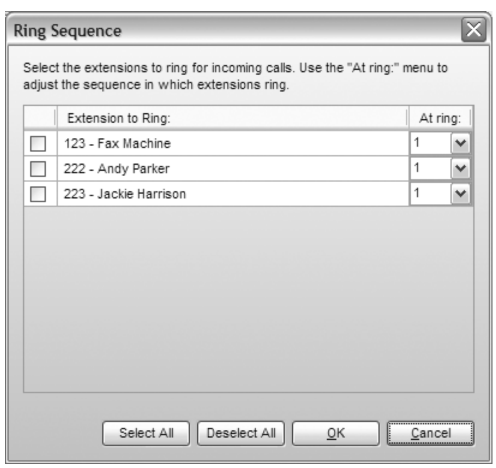
The *Caller ID Routing Groups* area (or *CLID Routing Groups* in some regions), allows you to set up call handling for up to 10 groups for Mode 1, Mode 2 and Holiday Mode. Call handling for a group is identical to call handling for telephone numbers, except you can set the ring pattern for each group. 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.

     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.

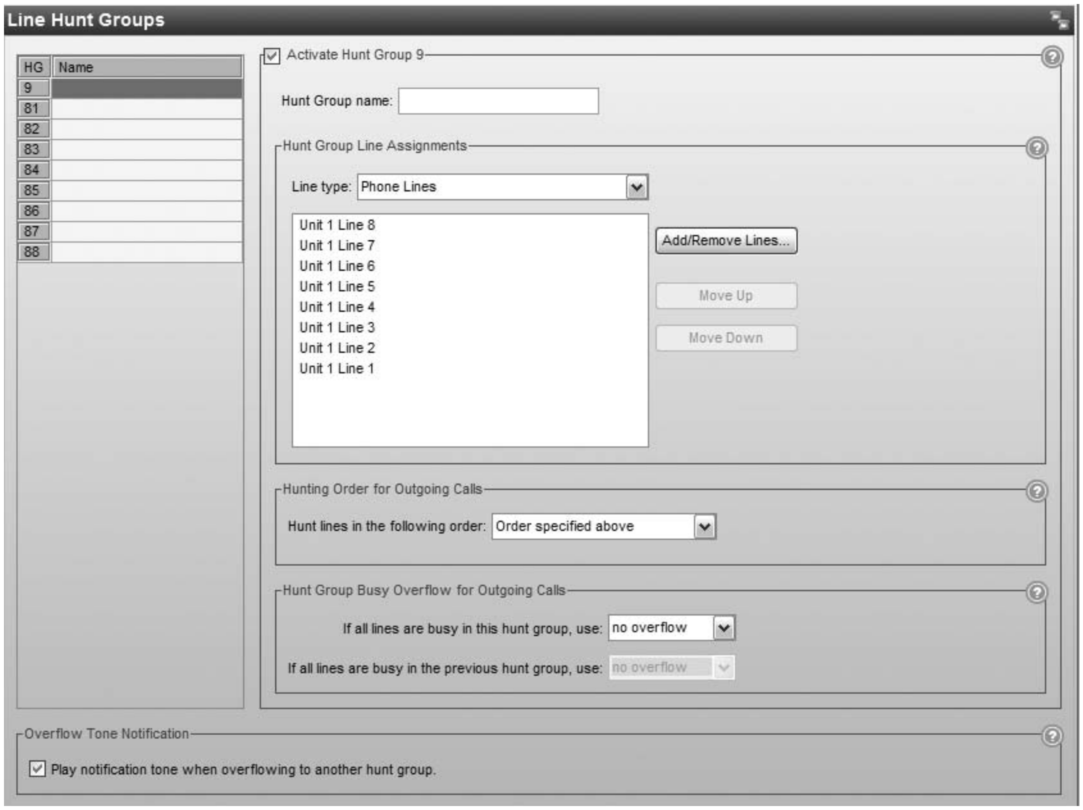
The unit uses a hunt group when placing a call from a local extension, to a remote extension, or with the call bridge (DISA) feature. The hunt groups do not affect incoming calls.

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.



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.

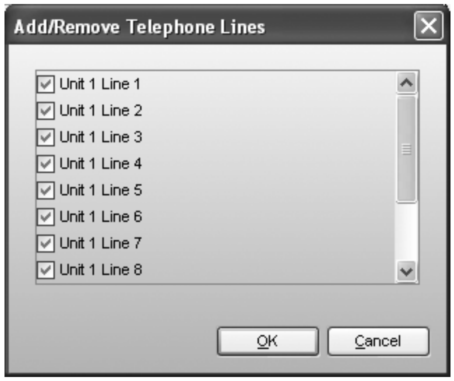
Hunt group line assignments

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.



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 #.

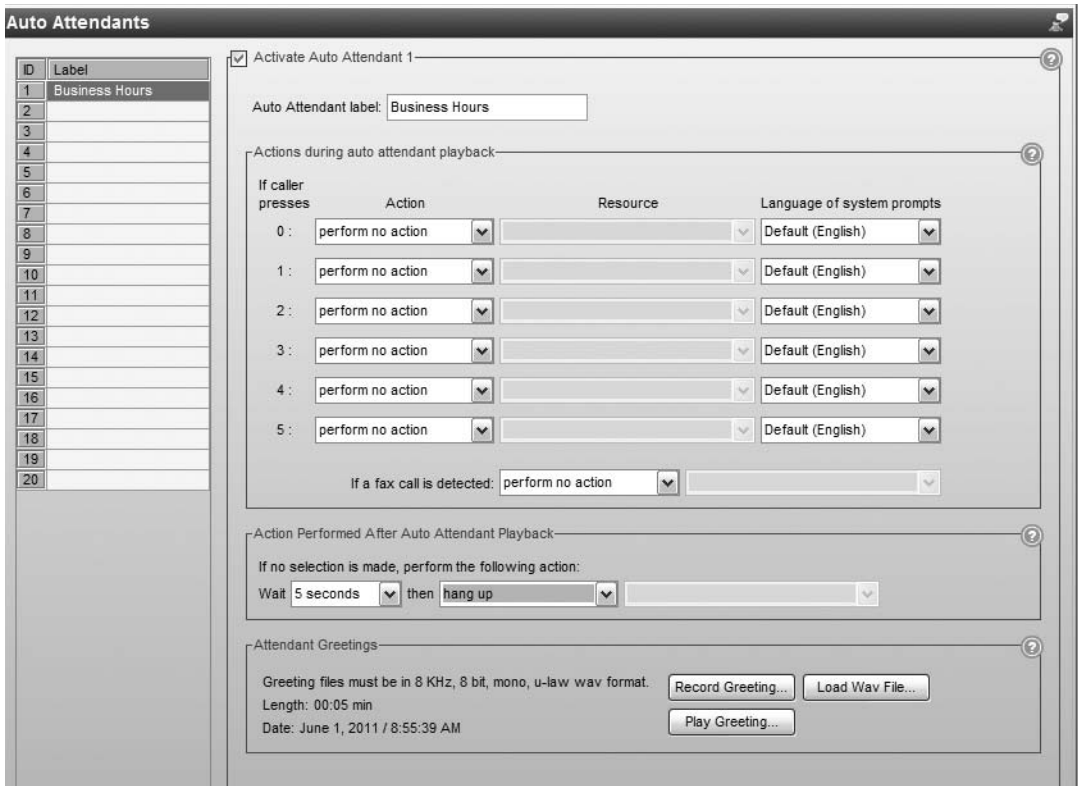
The caller is able to dial an extension, even if the first number of the extension is the same as an auto attendant option. For example, the caller can dial *111* even though the first "1" is the same as auto attendant option 1. This is because the system waits after the caller dials a digit, before 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.



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.

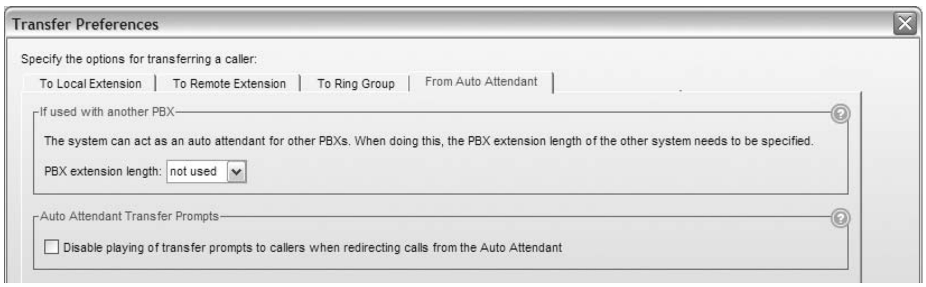
2. Select an auto attendant.
3. If necessary, select the *Activate Auto Attendant* checkbox.
4. Enter the *Auto Attendant label*. The name will identify the auto attendant elsewhere in the Management software.

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.



    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.

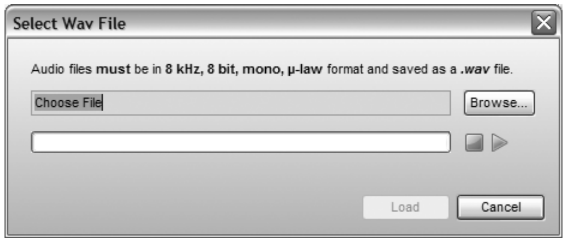
Loading a message

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.

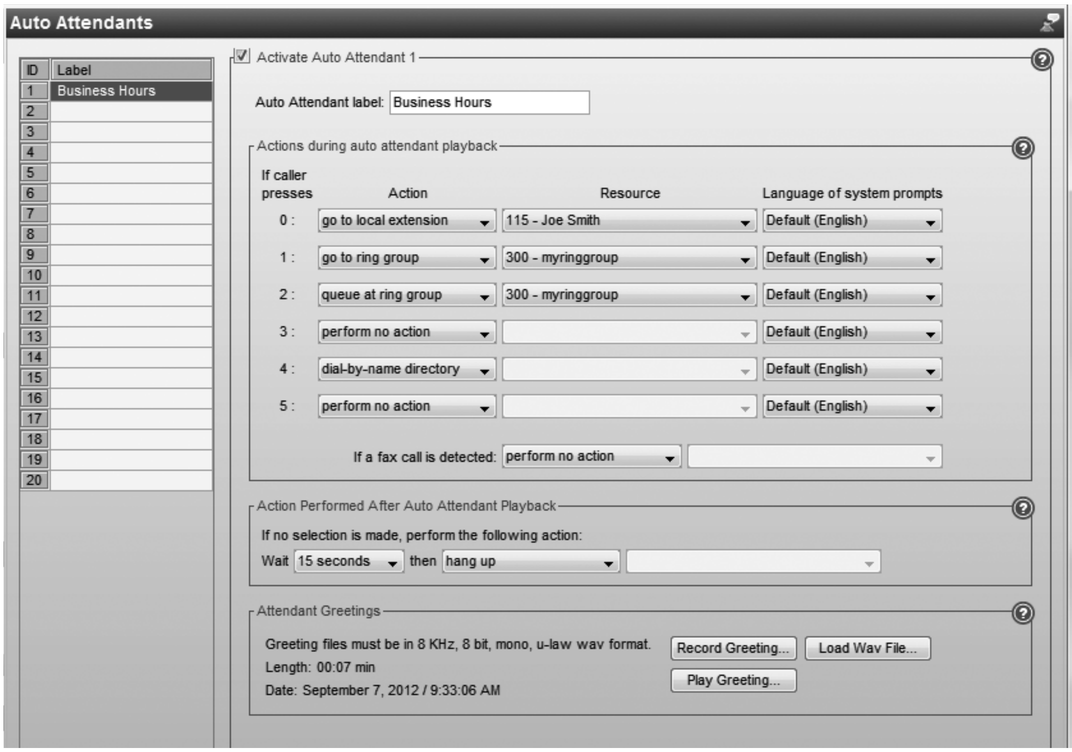
1. Pick up a local extension, or dial into the system from a remote phone. If you pick up a local extension, you will hear the dial tone. If you dial in from a remote phone, the auto attendant will answer.
2. Enter command mode by either pressing # on an analog extension phone, or **55#* on a proprietary 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.



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.

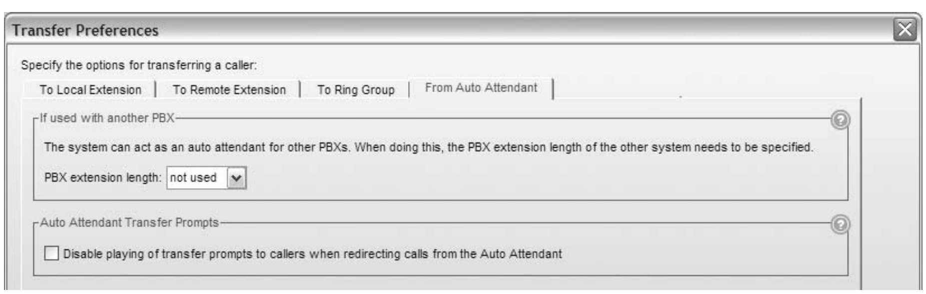
You can delete the names recorded for the dial-by-name directory by resetting the voice mailbox. See “Reset mailboxes” on page 86.

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.



3. Set the *PBX extension length* to the number of digits in an extension number on the other PBX.

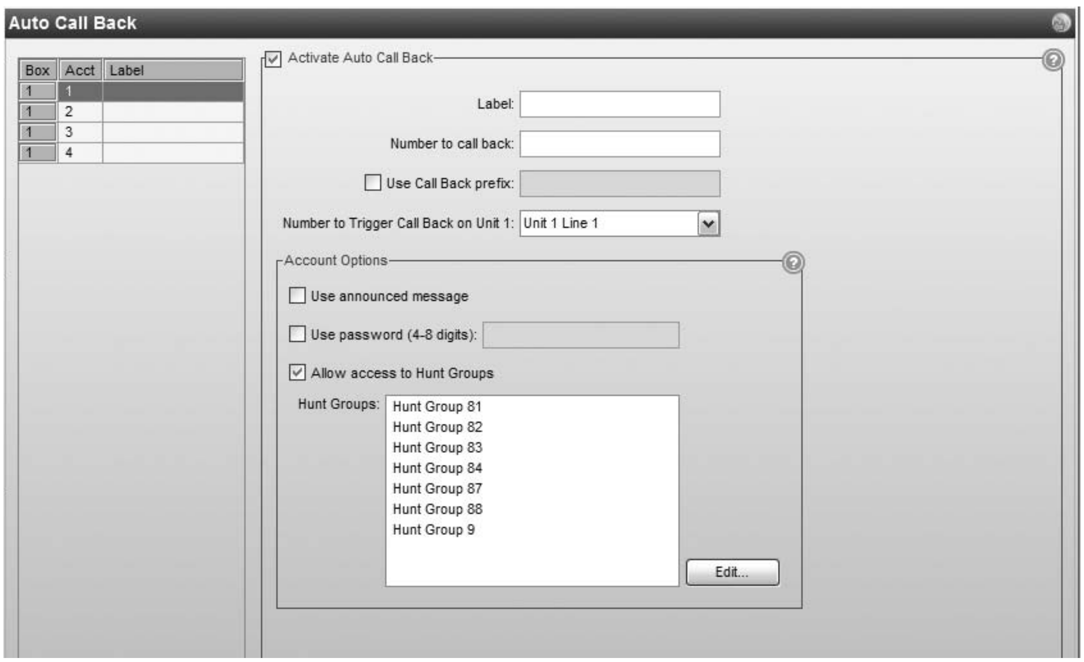
Auto Call Back

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.

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.

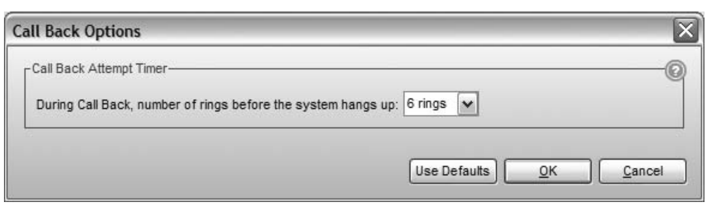
When the system returns your call, it can play a default message that tells the recipient to press # to accept the auto call back. You can record and enable an announced message that will be played instead of the default message. For example, the announced message can tell the front desk at your hotel 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.

   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.

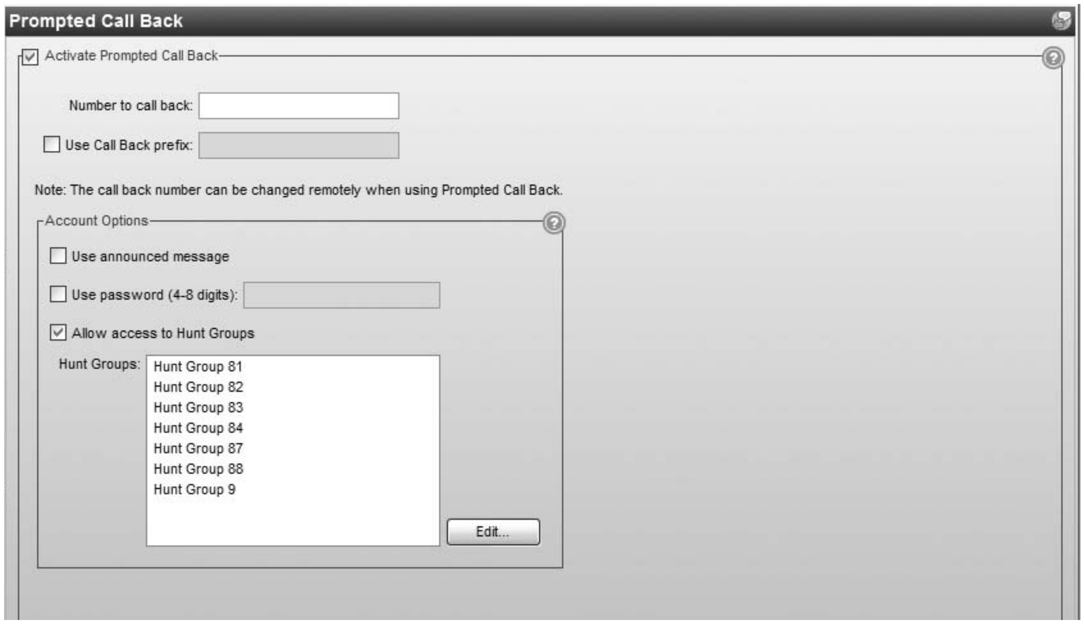
Prompted Call Back

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.



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.

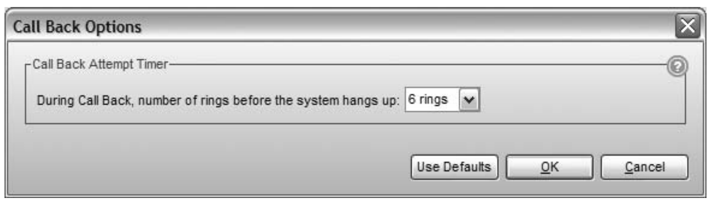
To configure prompted call back, call the system, and let the auto attendant answer. Dial *6*, and then follow the prompts. You can change the call back number, change the announced message, and enable/disable the announced message.

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.



   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.

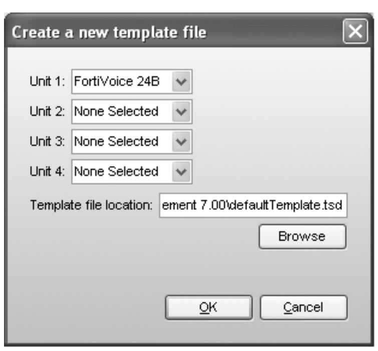
For more information about the *Configuration Selection* page, see "Starting the Management software" on page 2.

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.

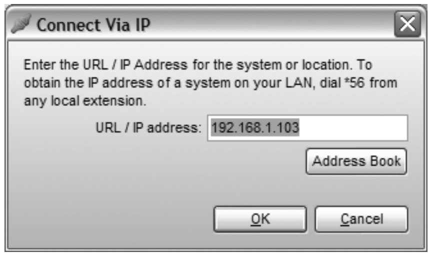
2. Select the types of units in your system.
3. Click the *Browse* button to select the location and name of the configuration file.
4. Click *OK*. The system creates the file, and opens it in the Management software.

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.

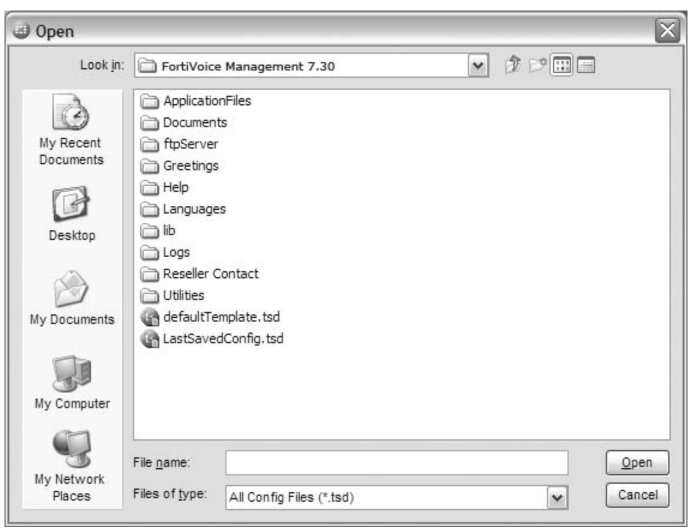
2. Enter the *URL/IP Address* of the unit, and then click *OK*. You can also choose the system from the address book.

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.

2. Choose the configuration file, and then click *Open*.

Save

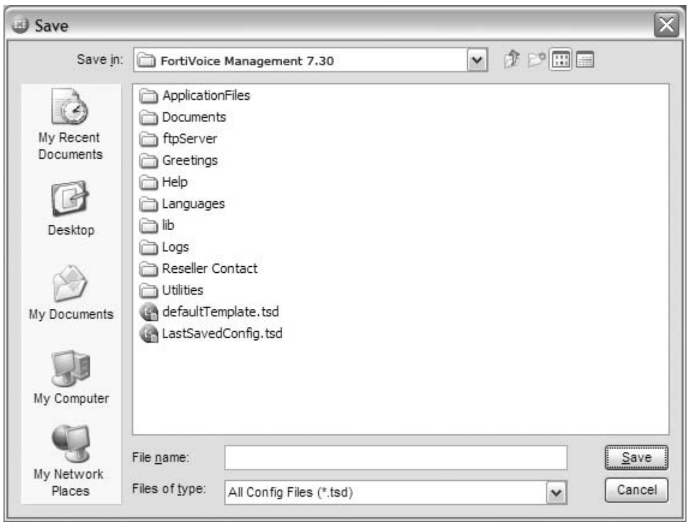
The *Save* command saves the configuration to the open configuration file or to the connected unit. The unit can be local or at a remote location.

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.

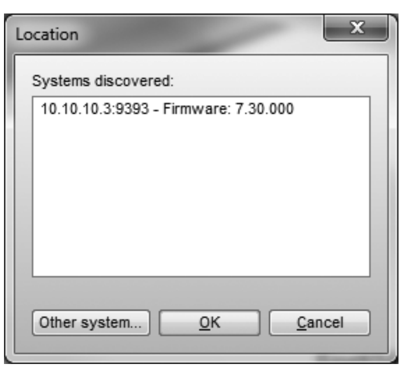


2. Enter the name of the configuration file, and then click *Save*.

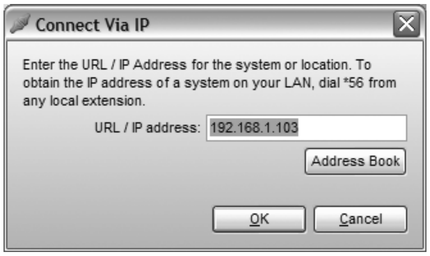
Save To > Location

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.



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.



   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.

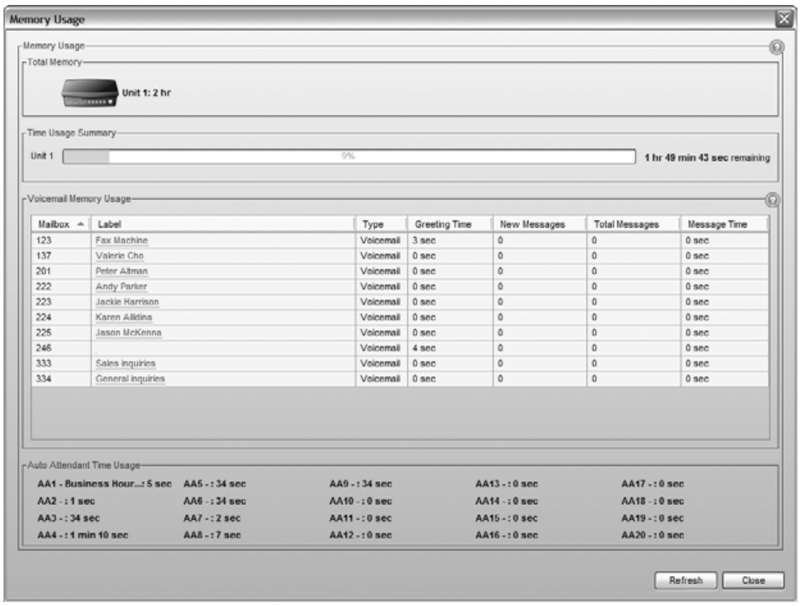
Exit

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.



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.

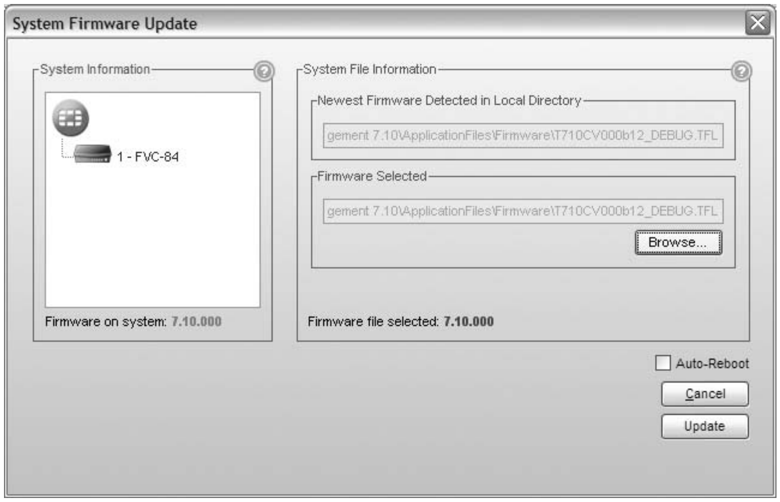
Update Phone

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.



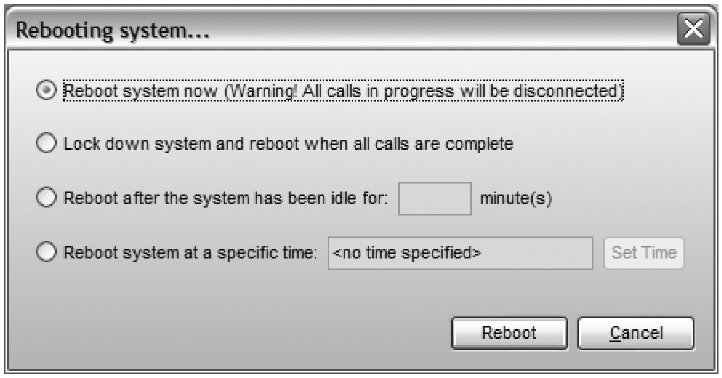
2. If the *Firmware detected* is lower than the *Firmware file*:
   - **a.** Set the *Auto-Reboot* checkbox.
     - Select *Auto-Reboot* to automatically reboot the units once the firmware has been updated. This will cancel any calls in progress.
     - Clear *Auto-Reboot* to prevent the units from rebooting 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.

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.



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.

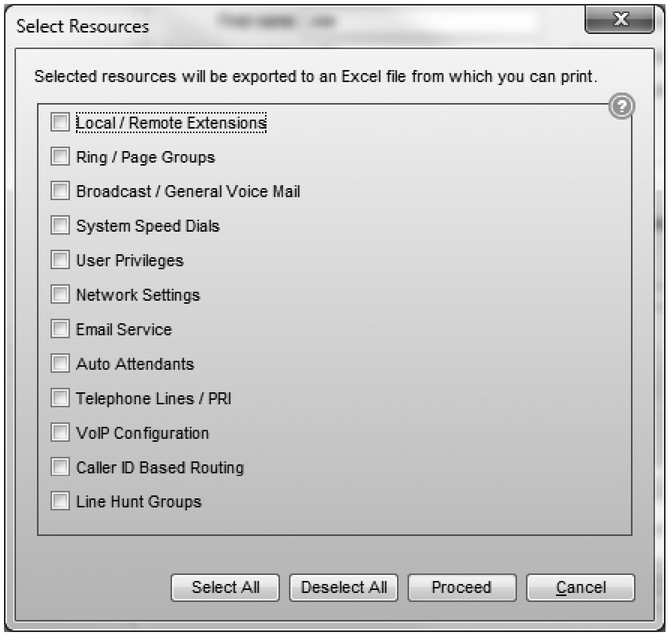
1. Choose *Tools* > *Print* > *Print Labels*. The *Select Extensions* window appears.

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.



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.

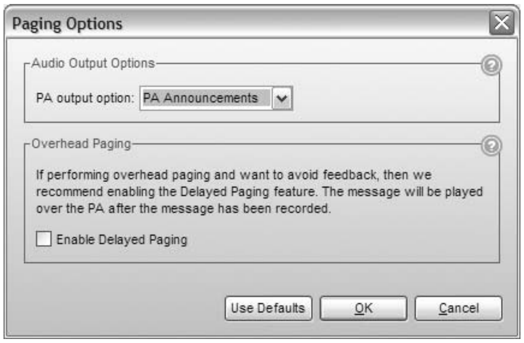
Options menu

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.



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

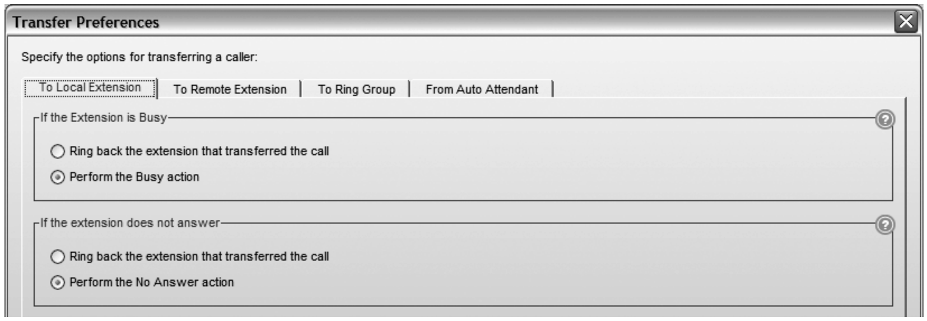
The *Transfer Preferences* command allows you to set options for transferring callers. You can set options for calls transferred to local extensions, remote extensions and ring groups, and from auto attendants and home phones.

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.

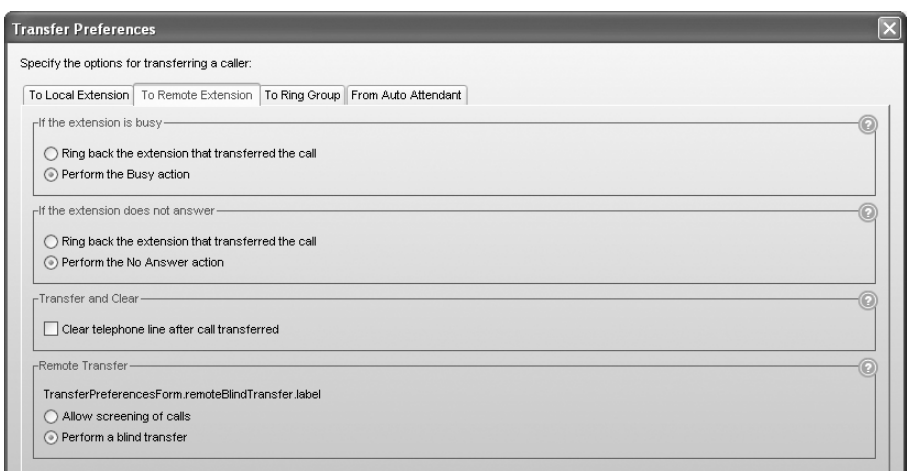


2. Select the option if the extension is busy:
   - *Ring back the extension that transferred the call* — If the extension is busy, transfer the call back to the source extension.
   - *Perform the Busy action* — If the extension is busy, perform the target extension's Busy call cascade.
3. Select the 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.



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.

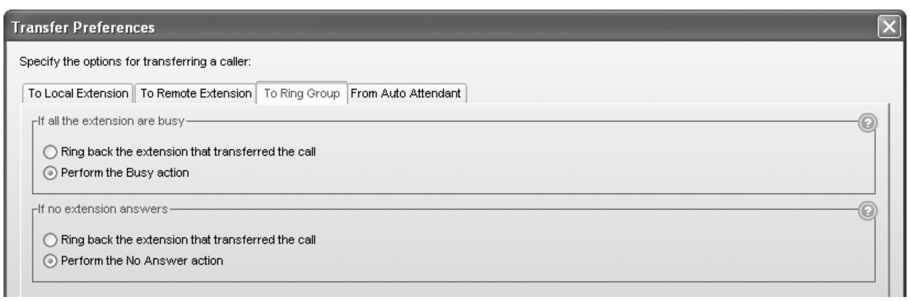
Transferring a call from a remote extension requires telephone lines without the 3-Way Calling/Conference service or the Transfer and Clear service.

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.

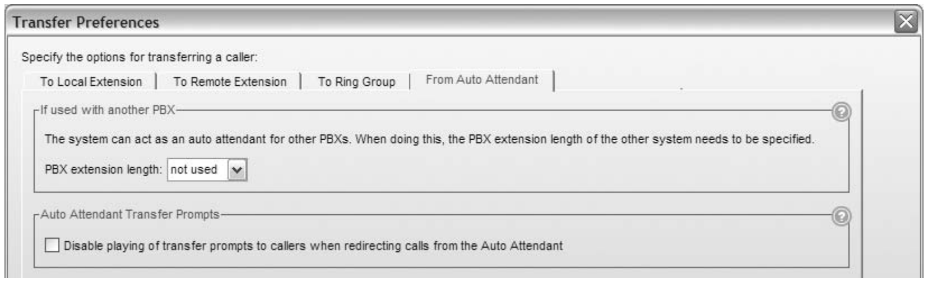


2. Select the option if all the extensions are busy:
   - *Ring back the extension that transferred the call* — If all the extensions are busy, transfer the call back to the source extension.
   - *Perform the Busy action* — If all the extensions are busy, perform the ring group's Busy call cascade.
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.



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.

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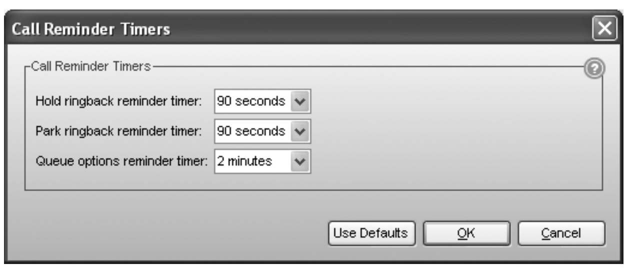
3. Select the *Disable playing of transfer prompts* checkbox to disable the *"one moment please"* prompt when transferring a call.

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.

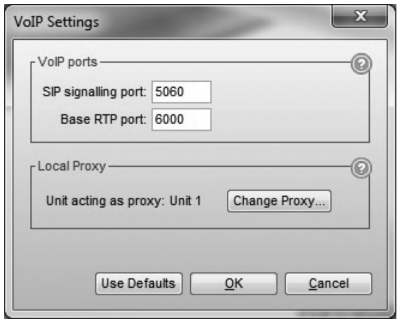


2. Set the *Hold ringback reminder timer* to how often the phone should ring when a caller is on hold.
3. Set the *Park ringback reminder timer* to how often the phone should ring when a caller is parked.
4. Set the *Queue options reminder timer* to how often a queued caller should hear a prompt.

VoIP Trunking

The *VoIP Trunking* command allows you to set the VoIP ports and choose the local Proxy.

1. Choose *Options* > *VoIP Trunking*. The *VoIP Settings* window appears.



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.

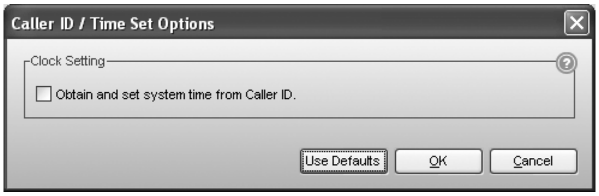
The router must map the SIP, TFTP and HTTP ports to the local proxy. See “Firewall settings” on page 14.

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.

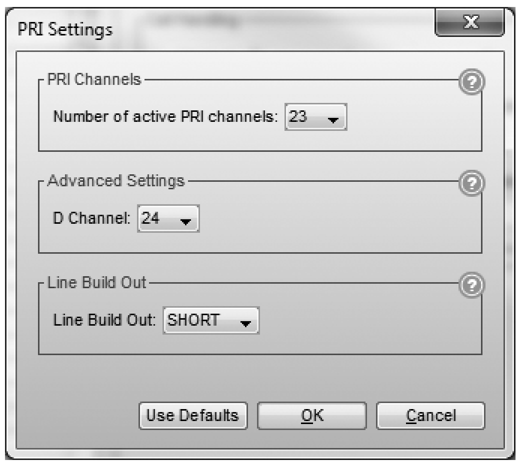


2. Select the *Obtain and set system time from Caller ID* checkbox to have the unit update its internal clock using the timestamp within caller ID. If it is not selected, set the time using the *Date and Time Properties* window. See "System time" on page 11.

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 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.

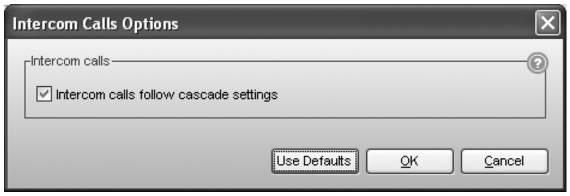
Line Build Out

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.

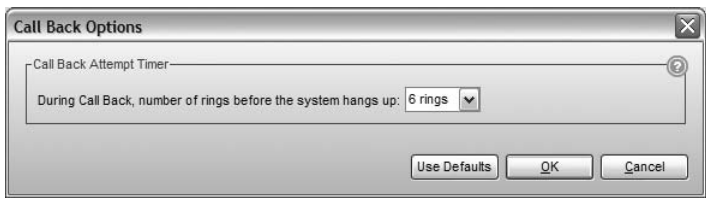


2. Select the *Intercom calls follow cascade settings* checkbox to have an intercom call use the call cascade of the dialed extension. Otherwise the intercom call will ring continuously.

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.



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

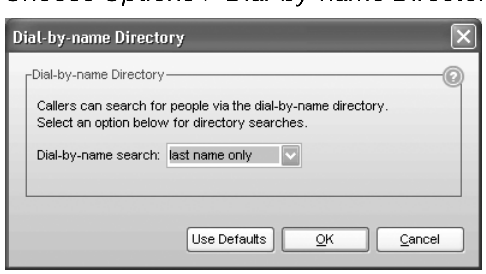
You can turn off the missed calls indicators here. In offices with busy ring groups, the missed call indicators can be an annoyance. To disable missed call indication, uncheck the box and press *OK*.

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.



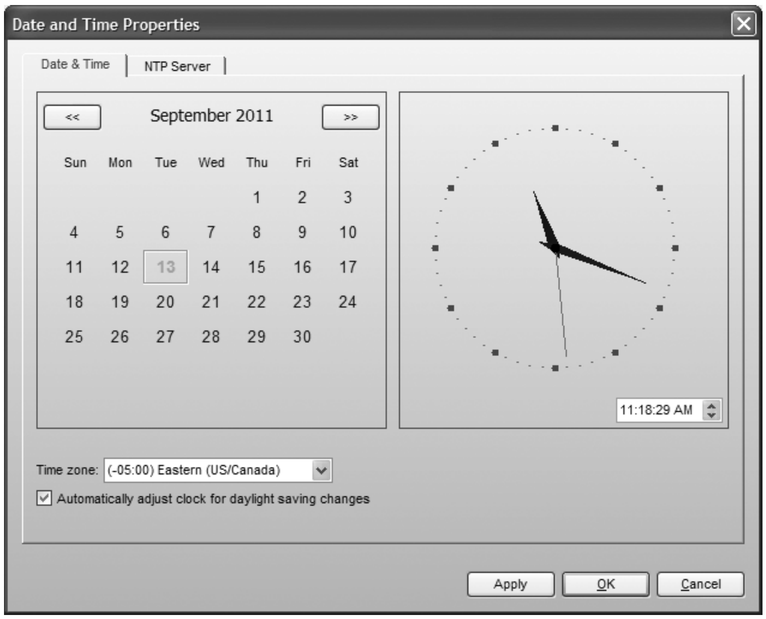
2. Select the desired search option. Choices include first name, last name, or both names.

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.



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.

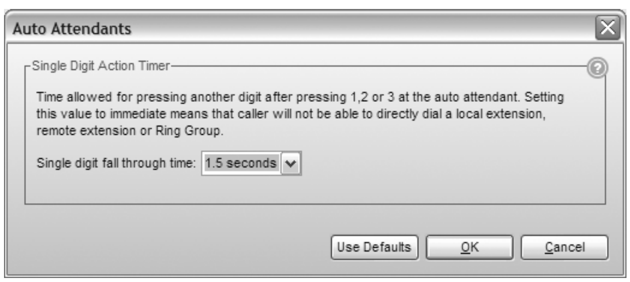
Auto Attendants

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.



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.

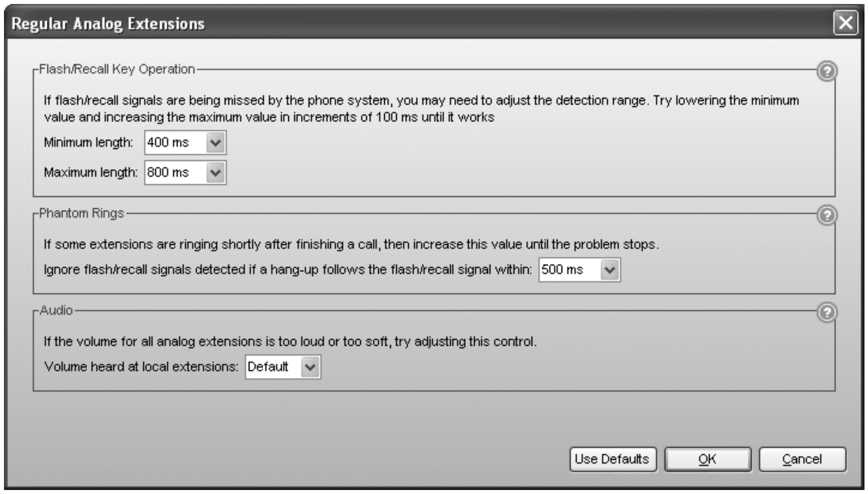
Extensions > Regular Analog Extensions

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.

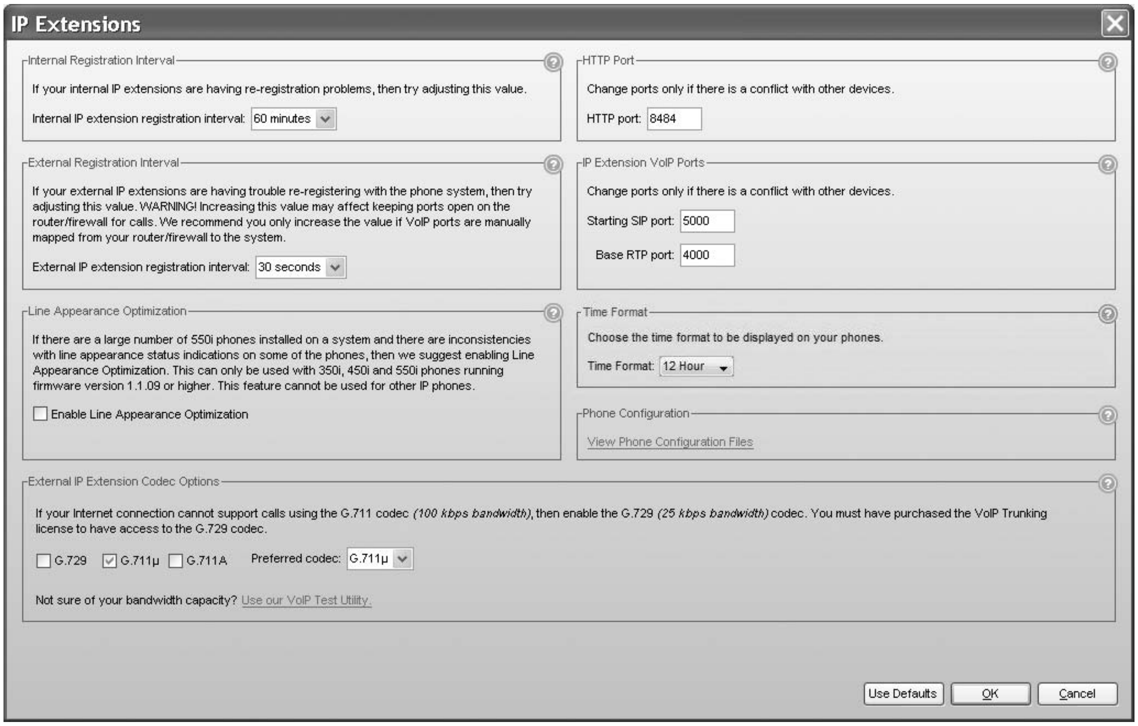


2. The *Minimum length* sets the minimum accepted duration of a valid Flash/Recall signal from a local extension. If there are false ring backs after a call, try increasing the *Minimum length*. If the system ignores genuine Flash/Recall signals, try reducing the *Minimum length*.
3. The *Maximum length* 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.



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*.

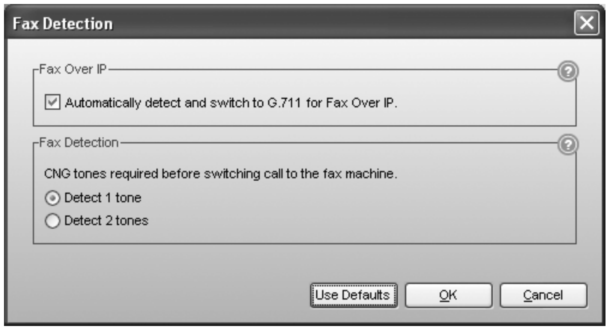
Phone Configuration
Click to open a browser with detailed configuration files for each IP extension.

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.



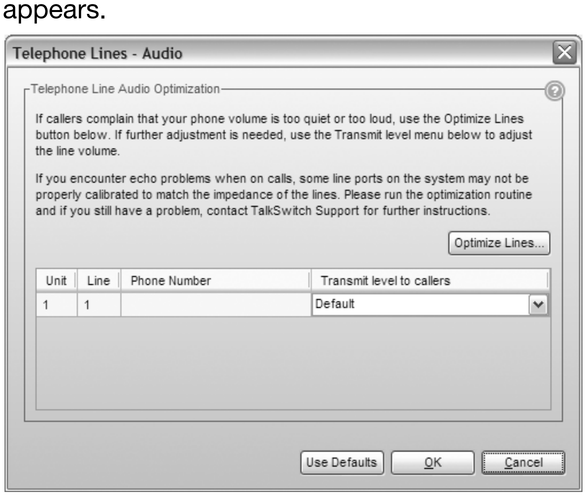
2. Select the *Automatically detect and switch to G.711 for fax over IP* checkbox to have the unit detect fax over IP and then switch to G.711.
3. Set the number of CNG tones the unit must detect before the auto attendant routes the call to the fax machine. Choices are:
   - *Detect 1 tone* — The auto 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.



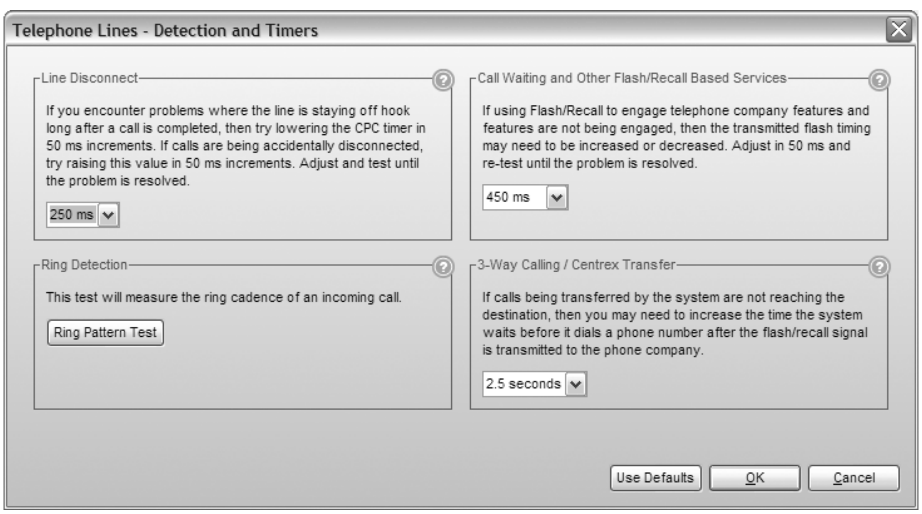
2. Optimize the unit to match the far end impedance of the telephone lines. Click *Optimize Lines*. The *Optimize Lines* window appears. Click *Run Optimization*.

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.

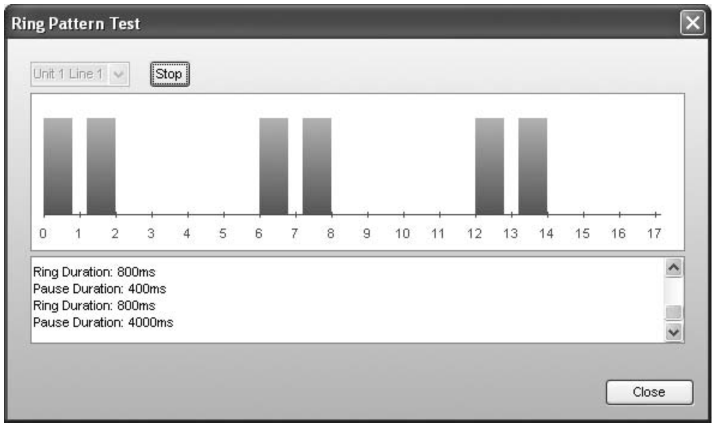
If your system does not appear to be functioning properly, first ensure the country where the system operates is set correctly in the *Region Selection* field of the *About* page, then click the *Use Defaults* button on this window. If this does not resolve the problem, please contact your local dealer or technical support centre for assistance.



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.

VoIP

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.

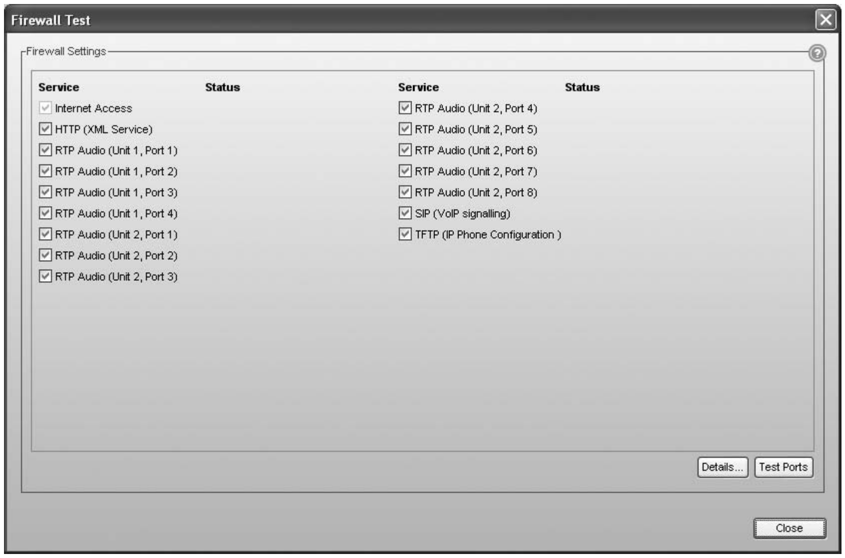
Global Dial Plan Codec Options

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.

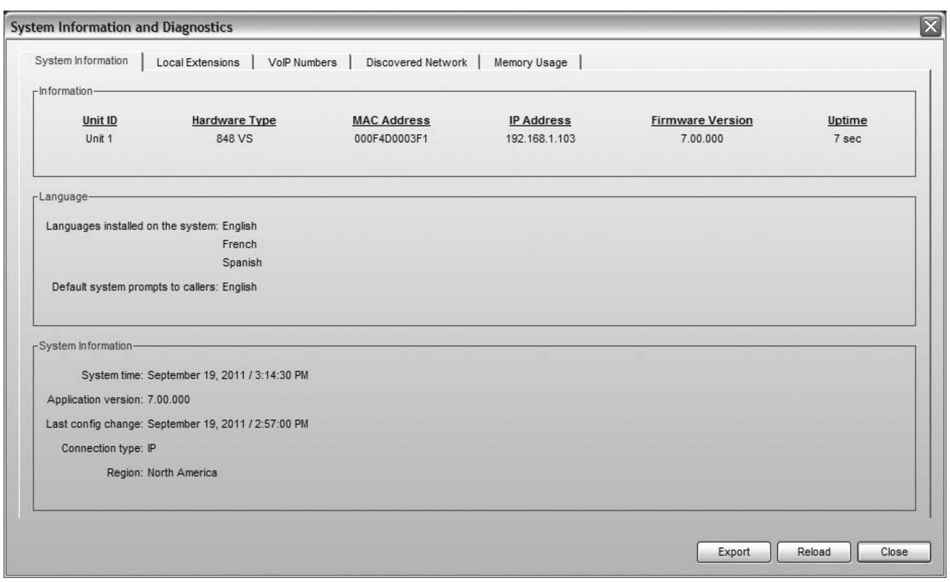
1. Choose *Troubleshooting > Firewall Test*. The *Firewall Test* window appears.



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.



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.

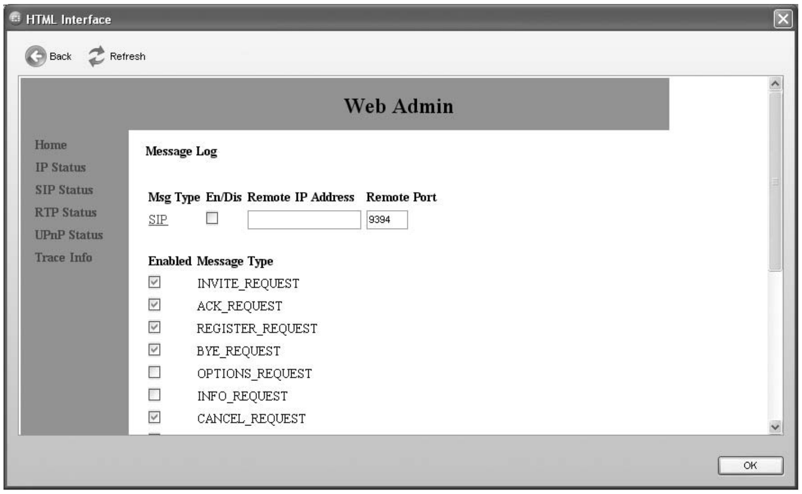
Support Tools > Call Logging Options

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.

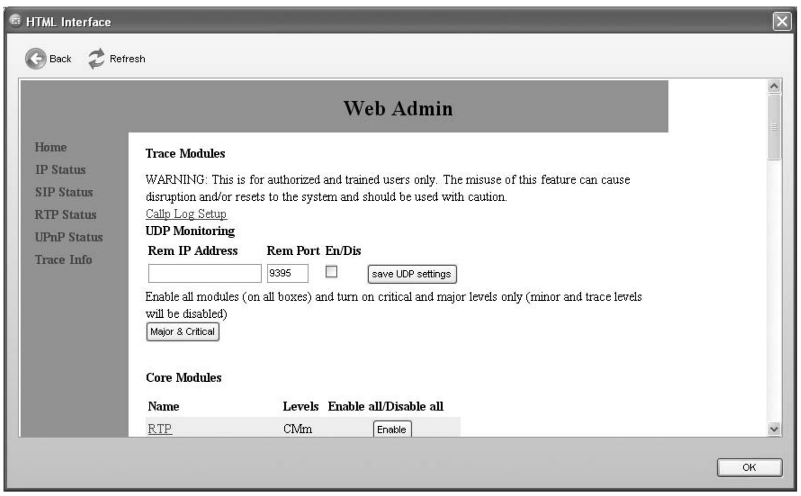
The *Troubleshooting* > *Support Tools* > *Call Logging Options* window provides diagnostic logging tools to capture call data and phone system events. These tools are intended for the exclusive use of technical support personnel.

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.



3. Select the *CP Logging* url to open a browser interface to permit detailed logging of phone system events.



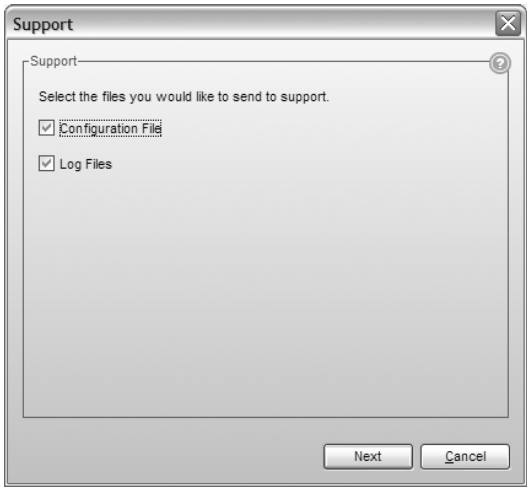
4. Click *OK* to exit the *Logging* window.

Support Tools > Email Software Configuration Logs

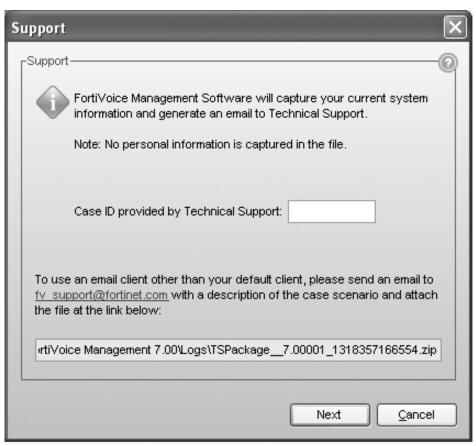
The *Troubleshooting* > *Support Tools* > *Email Software Configuration Logs* window allows you to automatically e-mail log files to Customer Support. Only use this feature as directed, after receiving a Case ID from Customer Support.

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.



2. Select the files you would like to send to support. Click *Next*.
3. Enter the *Case ID* supplied by Technical Support.



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.

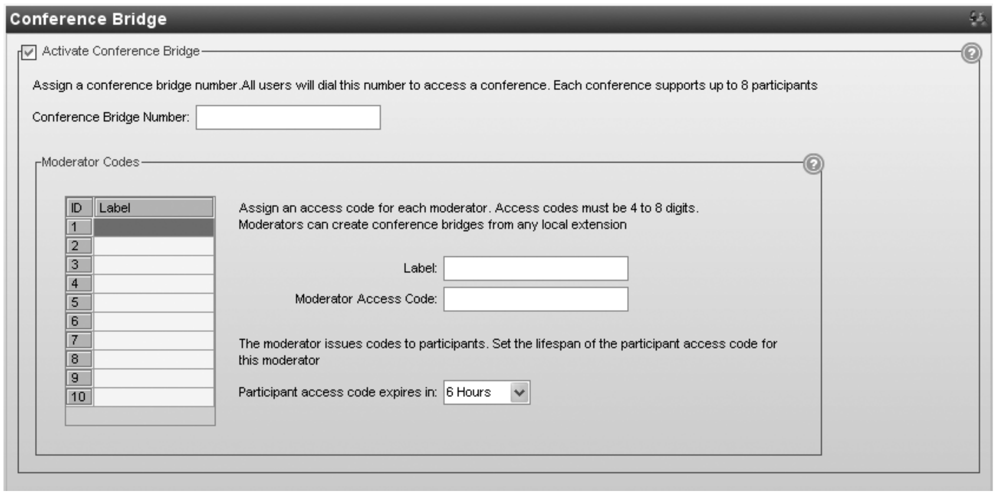
Using Conference Bridge

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.



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

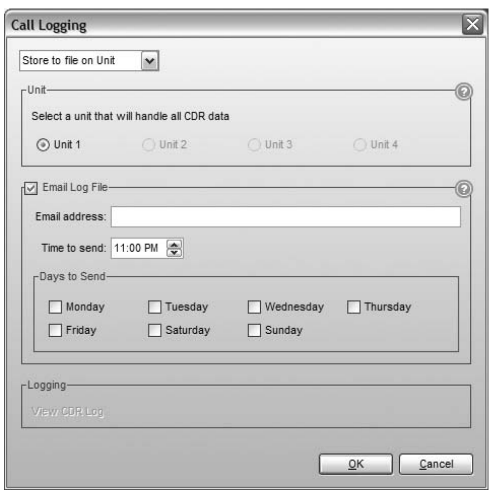
This chapter covers the setup and retrieval of call detail records (CDRs), which are records generated by calls through the phone system.

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.



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

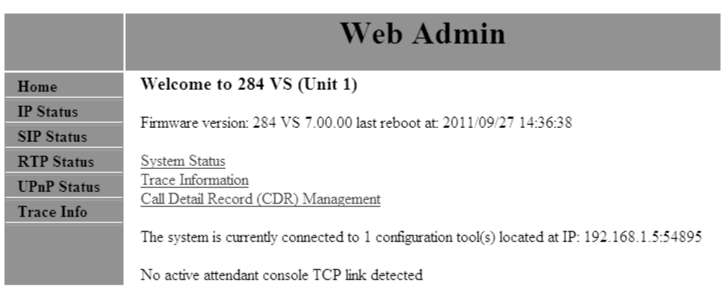
The Call Reporting licensed add-on application automatically connects to the unit, downloads the CDRs, and builds detailed reports of call activity. The application requires the *Store to file on unit* setting in the *Call Logging* window.

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.

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.



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. Type	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

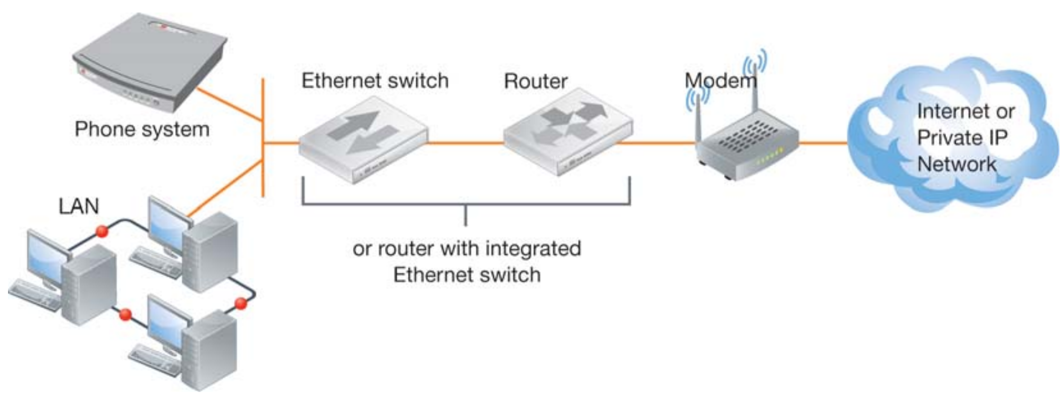
Connecting to a network

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



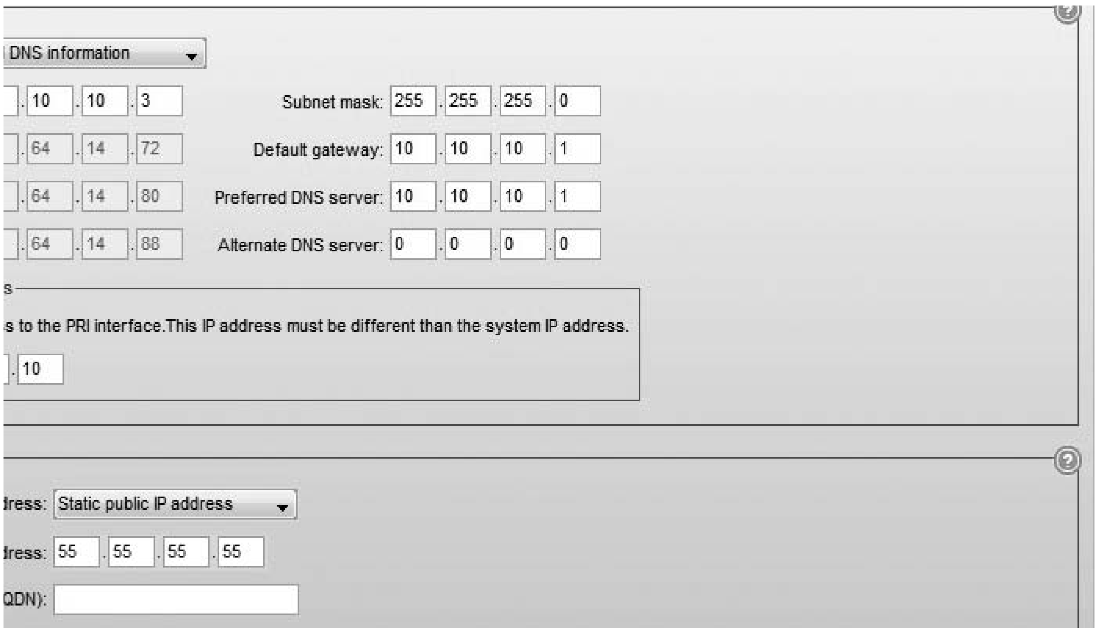
Confirming network capacity

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

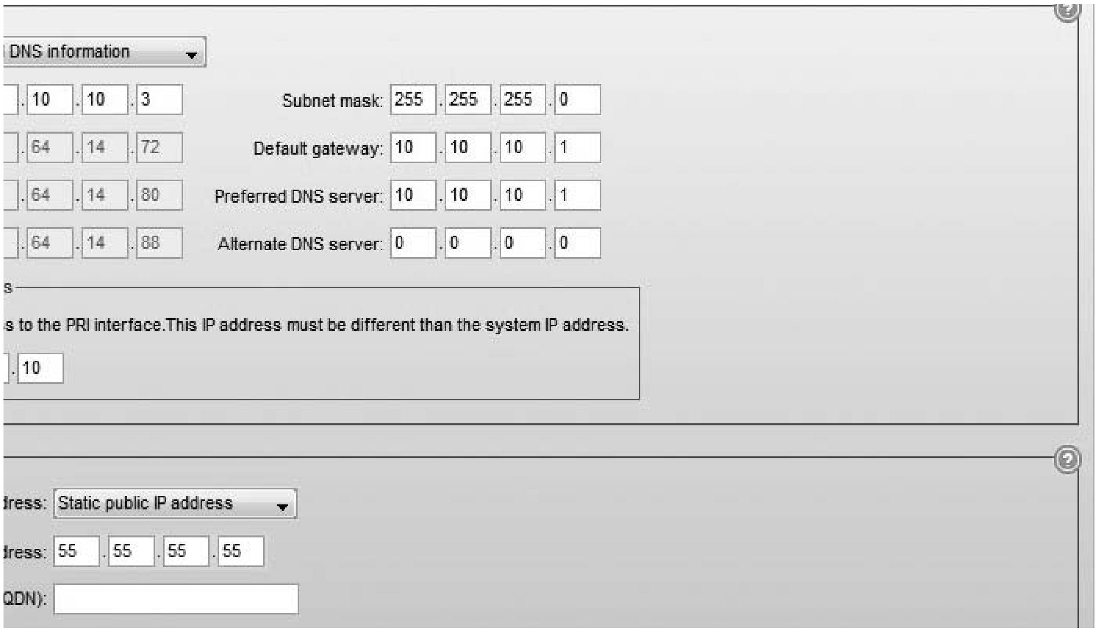
Each system must be configured with a local IP address. These addresses are used to direct VoIP calls to the appropriate location.



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.
3. 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.



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.

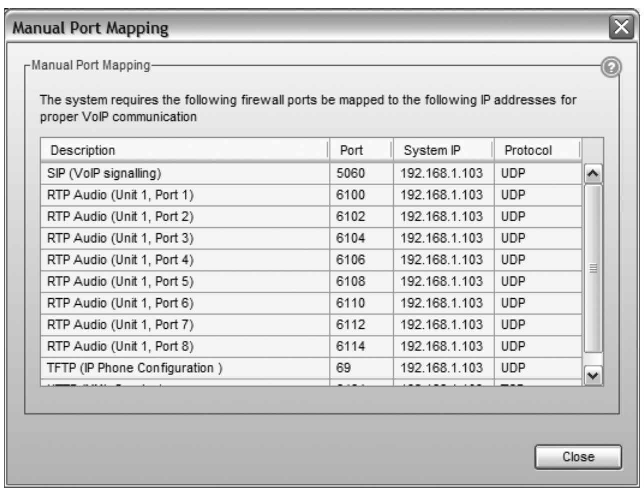
A router is a gateway between the local area network and the Internet. Most routers have a firewall to block unwanted data from the Internet. For voice data to reach the phone system through the firewall, port forwarding is required.

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.

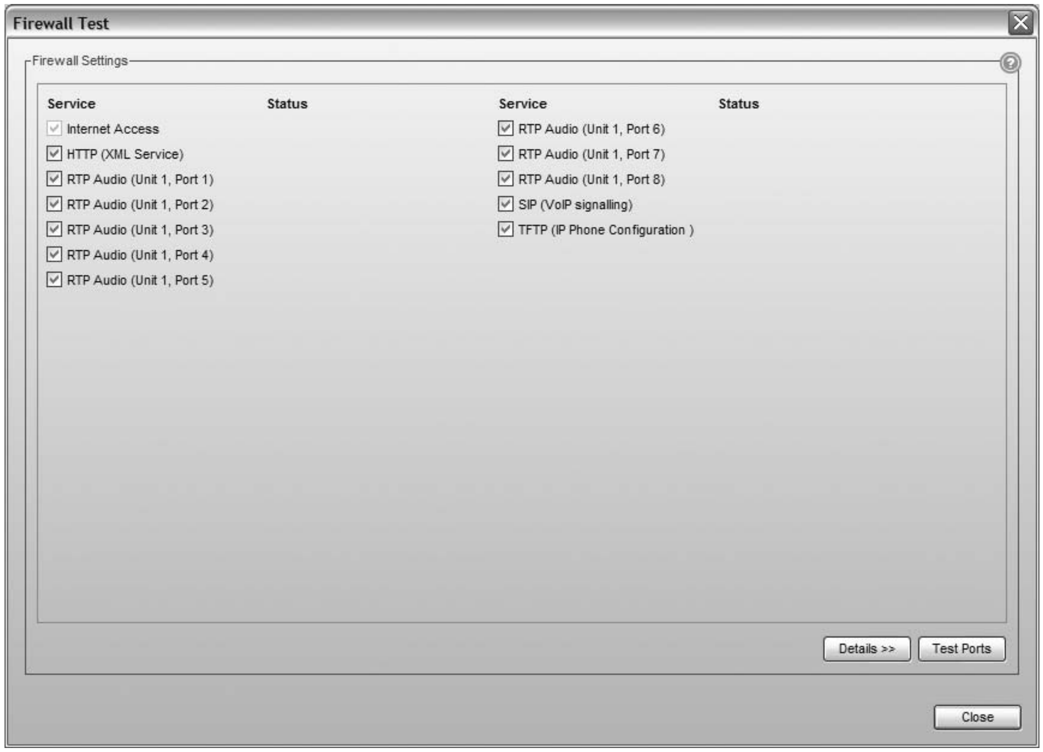
If port forwarding is required but your router doesn't support uPNP, or automatic port forwarding doesn't work, the *Manual Port Mapping Required* link will appear. You will need to configure the router as described below.

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.



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.



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

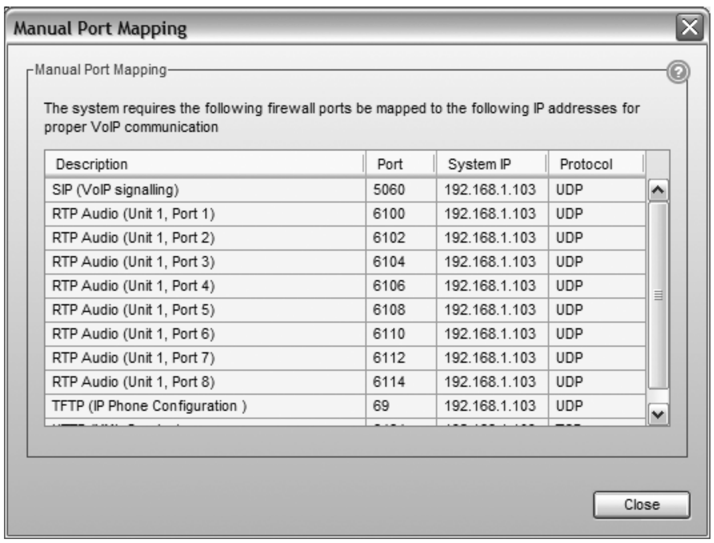
Configuring the router manually

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.



   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

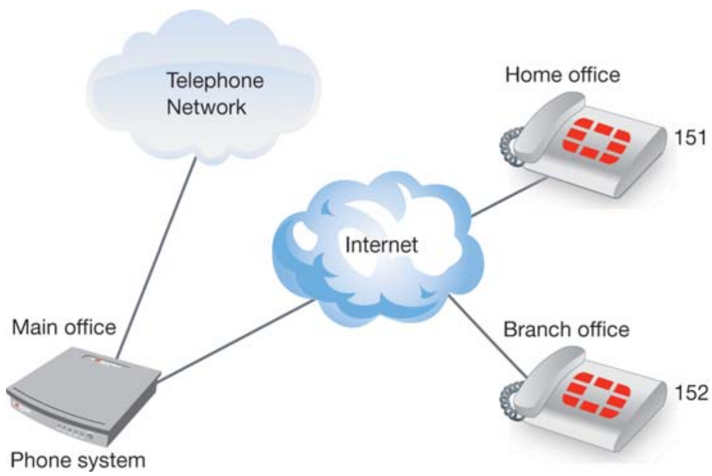
Using an external IP extension to call an emergency service number will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP extension stating:

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

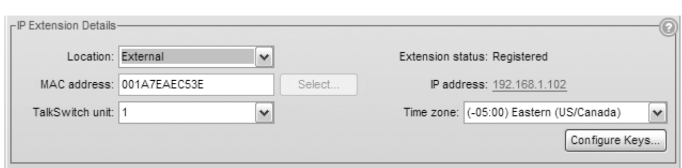


Before proceeding, ensure you have:

- 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.



3. Optionally, set the *Time zone* of the extension.

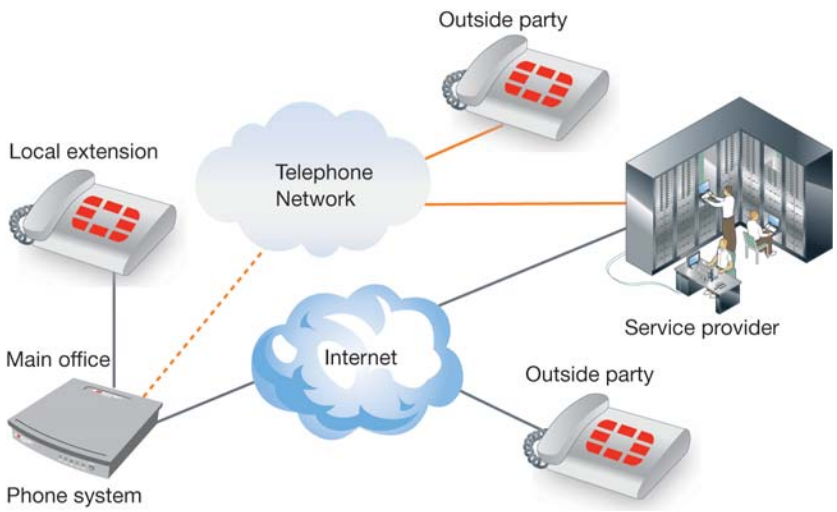
4. Optionally, perform advanced VoIP configuration:
   a. All VoIP lines are shared by default. You can reserve VoIP lines for the external IP extensions. See “Reserving VoIP lines” on page 190.
   b. External IP extensions use G.711μ as their preferred codec. This can be changed under *Troubleshooting* > *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).

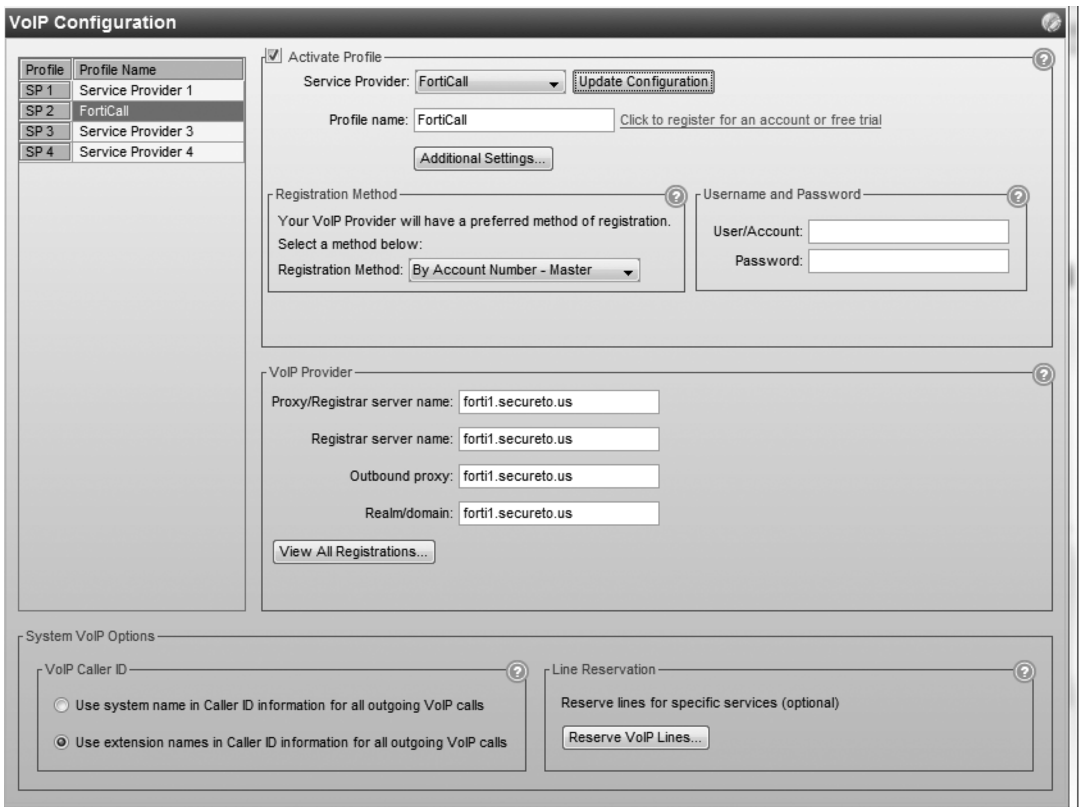
Setting up a service provider profile

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*).



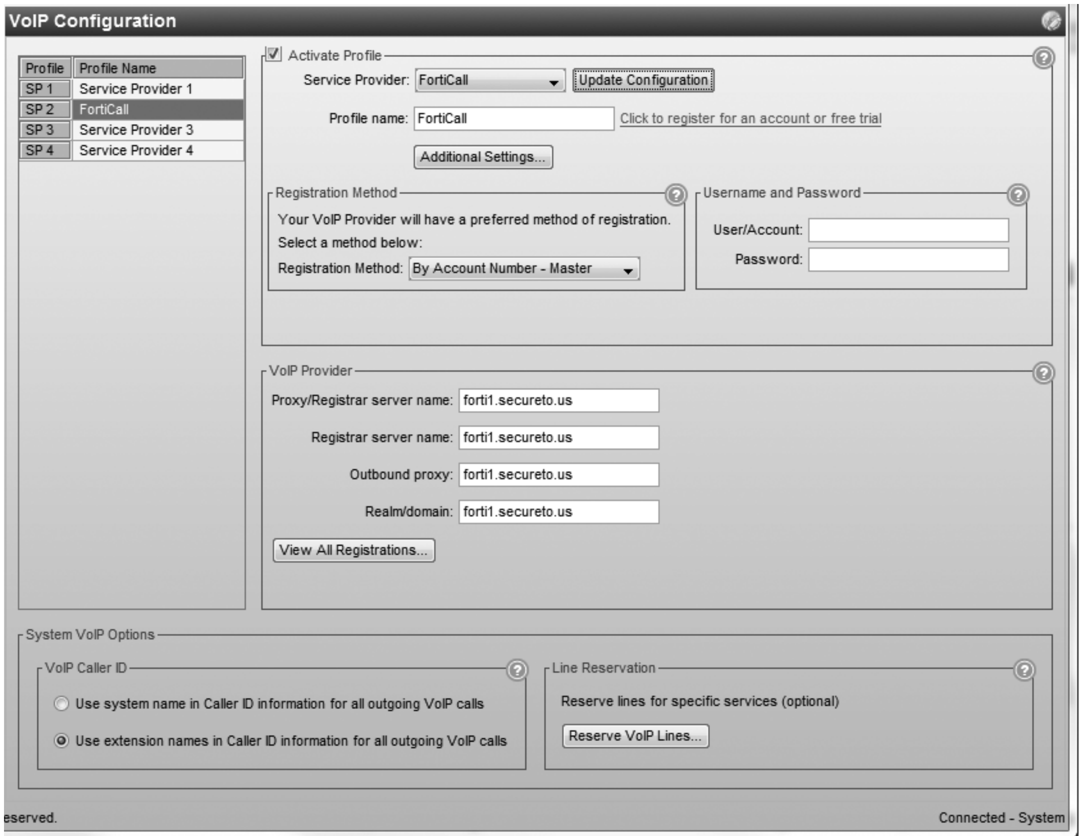
Activate profile

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.



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*

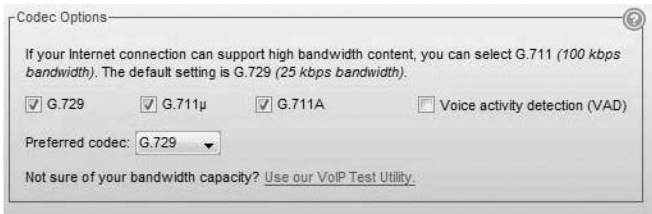
The *View All Registration* button will allow you to confirm connection to your service provider once you have completed the configuration of your VoIP numbers.

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:



- *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.

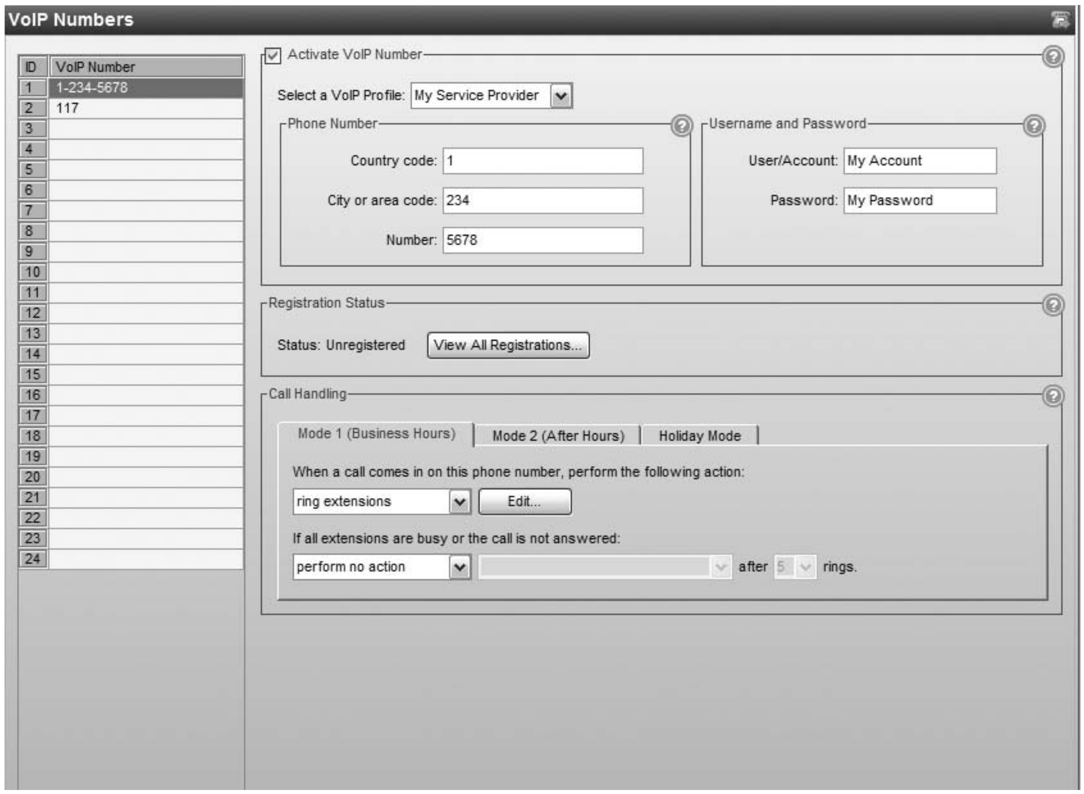
You may select which codecs to use or clear the unsupported codecs as well as select a *Preferred codec*.

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.



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

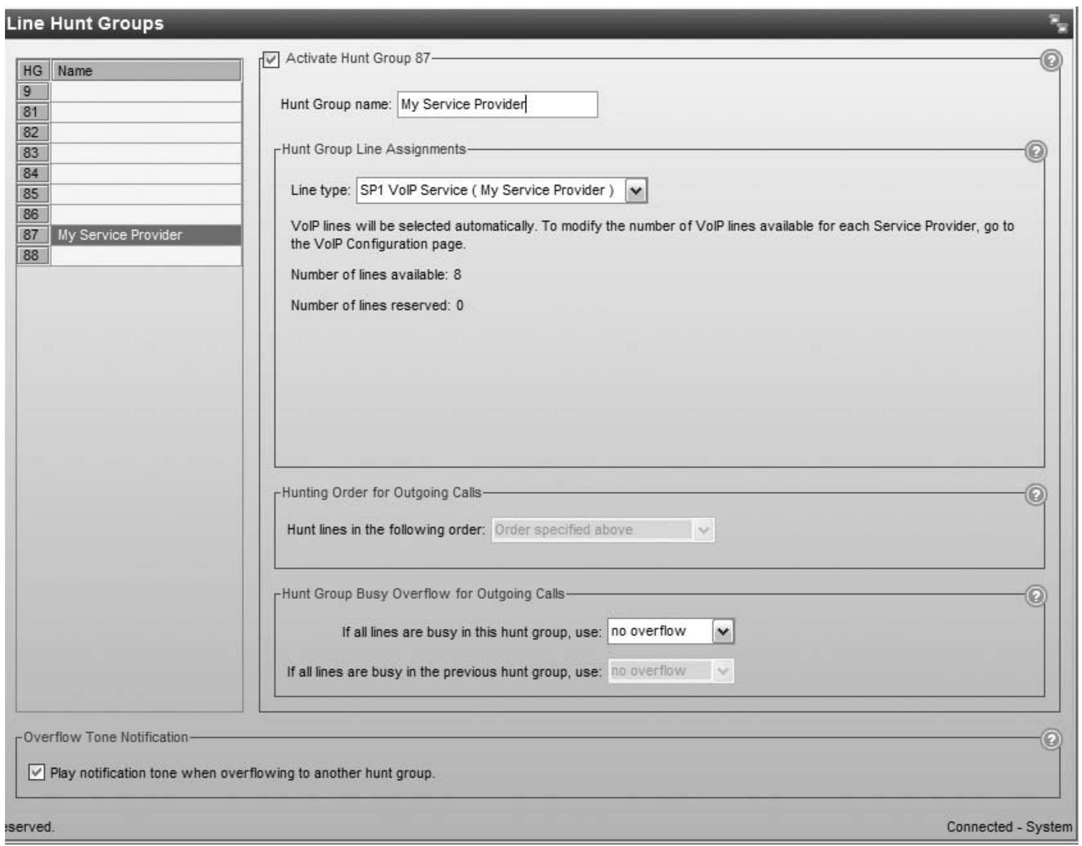
Ensure that hunt group 9 or 0 is assigned to the group of telephone lines or VoIP trunks used for calls to emergency services. Failure to properly configure hunt groups could prevent emergency calls.

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.



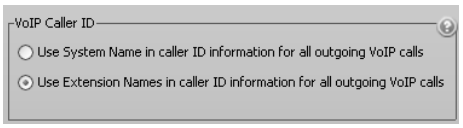
2. Select a hunt group.
3. If necessary, select *Activate Hunt Group*. All hunt groups are active by default.
4. Enter a *Hunt Group name*. The default name is *Hunt Group*.
5. Set *Line type* to *SPn VoIP Service*.
6. Select the overflow hunt group. Choices are *no overflow* and the other activated 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*.

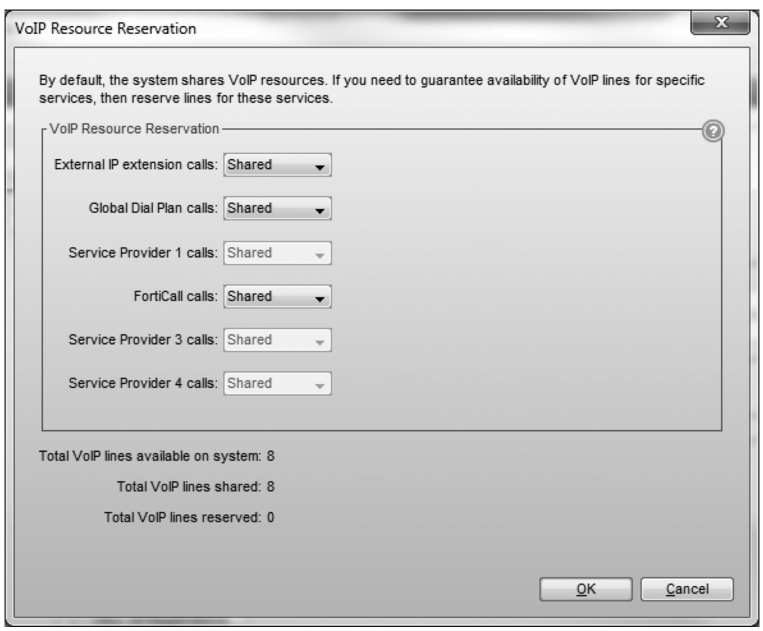


Reserving VoIP lines

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.



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
- VoIP

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

1. 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.

VoIP

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, *Flash* or *Recall/Hold* puts the call on hold and you hear the internal dial tone.
Flash or *Recall/Hold*	Pressing *Flash* or *Recall/Hold* again at the internal dial tone retrieves the most recent call on hold.
Flash or *Recall/Hold* at external dial tone	When you are connected to an outside line, pressing *Flash* or *Recall/Hold* 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 *Flash 4* or *Recall 4*. On an IP extension, press *Hold 4* followed by *Dial, Send* or # to complete the transfer.	
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 *Flash 5* or *Recall 5*.
Flash or *Recall/Hold *500–509*	Press *Flash* or *Recall/Hold* 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 then press **510* to place the call in the next available park orbit, starting at 500.
***500–509*	Retrieves a parked call.

Keys	Function performed
Flash or *Recall/Hold 6*	Conference call for 3 parties. Press *Flash* or *Recall* to put the first caller on hold, dial another extension or external number and press *Flash 6* or *Recall 6*.
**050	Plays the music on hold
**60* **61* **62*	Disables Do Not Disturb at the current extension. Enables Do Not Disturb at the current extension. 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 *Flash 7* or *Recall 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 *Dial, Send* 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 Send* or **55 Dial*).	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.

Index

Numerics

A

B

C

H

I

K

L

Q

R

S

T

U

V

W