FortiVoice™ v5.0
Administration Guide

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Introduction

This guide

This Admin Guide includes information about configuring and using your FortiVoice telephone system. For information about the initial set-up of the system, consult the *FortiVoice QuickStart Guide*.

Symbols in this guide

The web-based FortiVoice interface uses icon buttons to trigger functions.

	Accept		Settings		Hide information
	Add		History		Reveal information
	Delete		Load		Calendar
	Remove		Play		
	Lookup		Record		

Before you begin

Set password

Setting and frequently changing the system password is recommended to avoid unauthorized access to the system.

It is recommended that PIN codes are changed frequently to avoid unauthorized activity.

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 QuickStart Guide that came with your phone system before reading this guide. The QuickStart Guide contains critical information about setting up your phone system.

Configuration

Introduction

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

Dashboard

System Information

Host Name

Optionally enter the Host name. This should be the company name or a shortened form suitable for use as caller ID during VoIP calls.

Serial Number

This is the serial number of your FortiVoice system.

Firmware Version

This is the current firmware version running on your FortiVoice system. Click on *Update* to update the firmware on your system.

System Time

System Time shows the current date and time programmed into the system. Clicking the *Change* 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.

System Configuration

Backup — Clicking the *Backup* link creates a backup copy of the system configuration.

Restore — Clicking the *Restore* link reloads a selected backup file to the system.

Default — Clicking the *Default* link resets the configuration to default values. The *Network* and *Admin* pages will not be defaulted.

Display Language

You can set the language of the web interface to English, French or Spanish. You can set default languages for different users on the *Admin* page.

Uptime

This is the length of time the system has been running since the last reboot.

Terminal Window (CLI)

The *Terminal Window (CLI)* command displays the *Command Console* window. Use this as directed by customer support.

System Resource

This section provides visual gauges for CPU usage, memory usage and disk usage. High memory usage indicates the system may running a lot of tasks. You should avoid running reports or logs until it returns a normal level.

The disk usage gauge indicates the amount of flash memory your voicemail, greetings and music on hold files are taking up on the system. If the gauge is showing a high percentage go, to *Status* > *Storage* to determine which resource is using up the memory.

The shutdown button will turn off the FortiVoice system. Shut the system down before unplugging it to prevent corruption of voice messages or greetings.

The reboot option will reset the system, and will prompt the user for a reason for this reboot. This information is stored within the system logs.

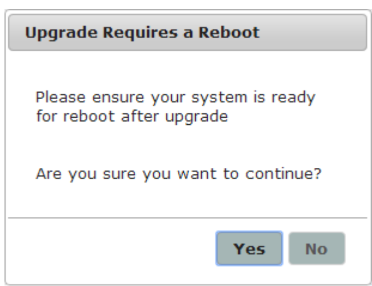
System Alerts

The *System Alerts* area shows details about the operation of the system, such as boot up, restarts and firmware updates. All system-level events are recorded, and the latest 10 events are shown. To review past events, click .

Update Firmware

The *Update* command allows you to update the units with a newer firmware version.

1. Click *Update*. The *Upgrade Requires a Reboot* window appears.



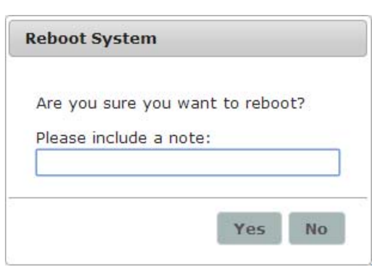
2. Click *Yes*.
3. From the browse window that appears, select the file to update the system to.
4. Click *Open*.

Update progress will be indicated in the popup window, while the lights on the front of the unit will display progress patterns.

Reboot System

The *Reboot System* command reboots the unit.

1. Choose *Tools* > *Reboot System*. The *Reboot System* window appears.

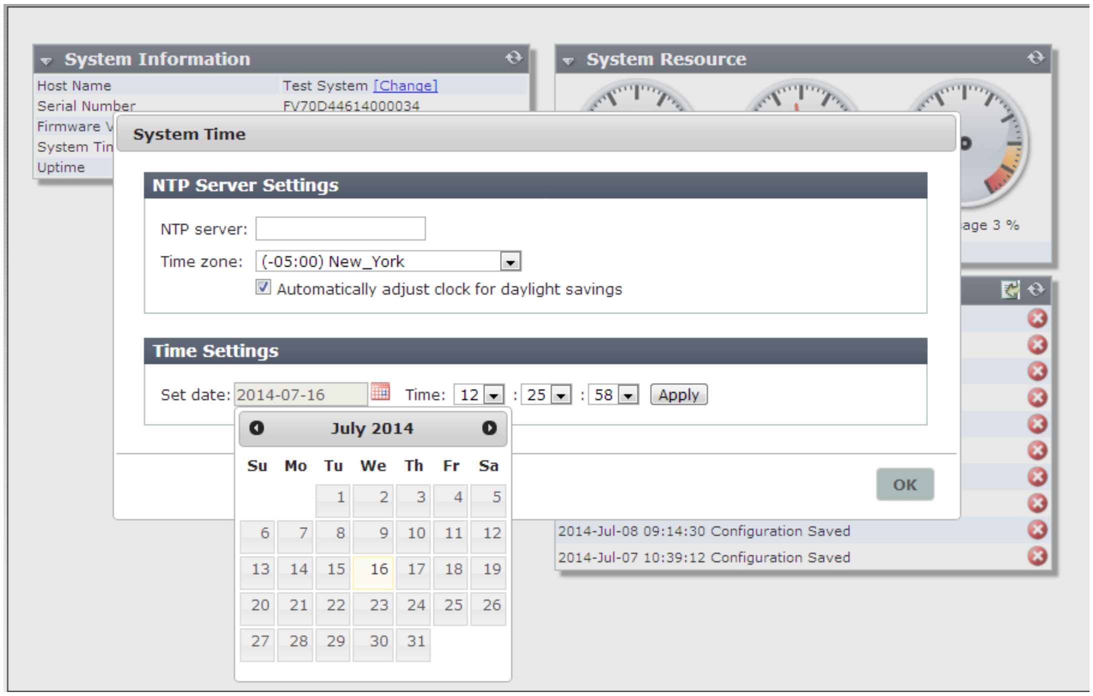


2. Enter a comment into the *Notes* field. This information will stored in the *System Alerts*.
3. Click *Yes*.

NTP Server

The *NTP Server Settings* area allows you to set the Network Time Protocol server and the time for the system.

1. Select the *Change* link beside *System Time*..



2. Enter the *NTP server*.
3. Select your *Time zone*.
4. Click . You can set the date and the correct time.
5. Click *Apply*, then *OK*.

Phone System

IP Extensions

The *IP Extensions* tab includes status, type, MAC address, IP address and firmware for each extension.

Update phones

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.

Trunks

The *Trunks* tab includes status for PRI lines, VoIP numbers, and analog lines.

PRI

The PRI lines status displays the following information:

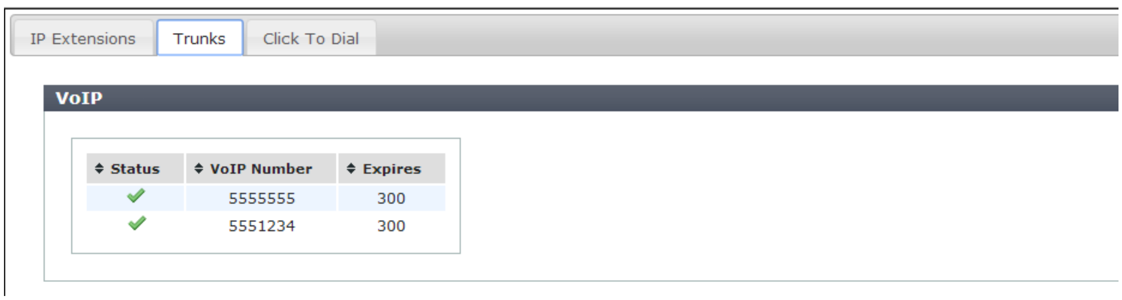
Status — status of the line. A checkmark indicates the line is connected. An exclamation mark indicates it's not functioning.

Line — indicates the port the PRI is connected to.

Details — provides details of the line status.

VoIP

The VoIP trunks status displays the following information:



Status — the registration status. *Registered* indicated by a checkmark or *Not registered* indicated by an exclamation mark.

VoIP Number — lists the VoIP numbers set up within the system.

Expires — the amount of time, in seconds, until the client has to re-register with the SIP server.

Analog Lines

The Analog lines status displays the following information:

Status — status of the line. *Connected* indicated by a checkmark or *Not connected* indicated by an exclamation mark.

Line — lists the line numbers set up within the system.

Details — indicates the status of line.

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.

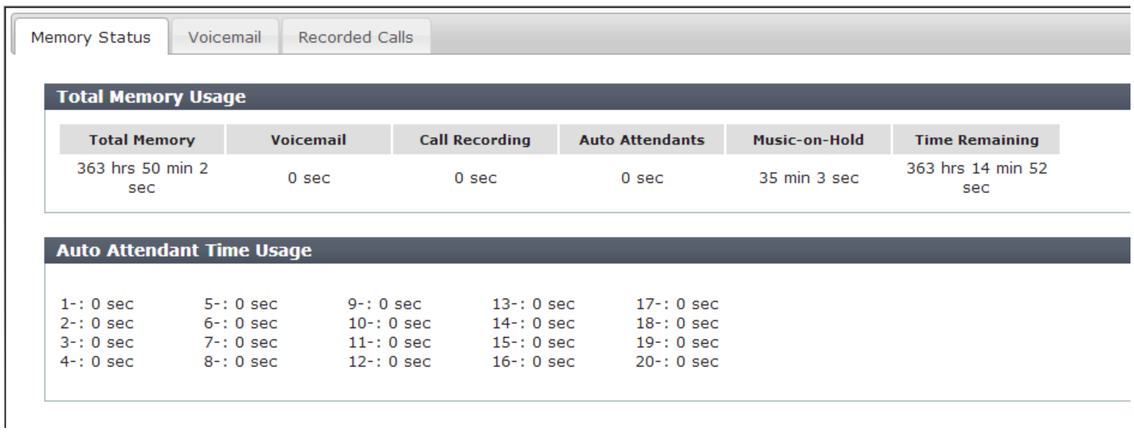
Storage

Memory Status

The *Memory Status* tab shows the amount of time available on the systems, and the amount of time used by voicemail, auto attendant messages and music on hold.

Auto Attendant Time Usage

The *Auto Attendant Time Usage* area displays the time used for each auto attendant in the system.



Voicemail

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

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. Users should delete messages before the mailbox fills up.

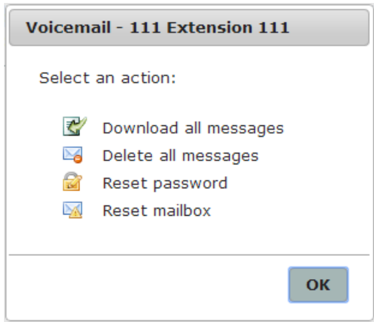
Clicking on the *Mailbox* link will allow you to:

- Download all messages
- Delete all messages
- Reset the mailbox password
- Reset the mailbox — this will delete all messages and recordings in the mailbox.

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 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* tab, or disable the password to listen to them from an extension. See "Delete password" on page 8.

1. Click on the extension. The *Voicemail* window appears listing the actions you can choose.



2. Click *Reset mailbox*.
3. Click *OK*. The greetings, messages, passwords and names for the 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.

You can also download messages from a mailbox without disrupting the password protection.

1. Click on the extension. The *Voicemail* window appears listing the actions you can choose.
2. Click *Delete password*.
3. Click *OK*.

Recorded Calls

The *Recorded Calls* tab shows how much time is taken up by recorded calls. Click next to an extension to download or delete recorded calls.

Call Detail Record

CDR

The *CDR* tab displays the call records for the system. You can select the time frame of the data to display.

Export Call Detail Records

The *Export* button allows you to export the call detail records to a .csv file or a file for the Call Reporting software application. When you click *Export*, you can select the data currently displayed in the web interface or all data. Each entry in the file has the following: Type, Log #, Event, Date, Time, Duration, Connection, Number, Caller ID, Line, Account #, PIN Name.

If you open the .csv file in a text editor, each item will be surrounded by quotes, like this:

“Type”,“Log #”,“Event”,“Date”,“Time”,“Duration”,“Connection”,“Number”,“Caller ID”,“Line”,“Account#”,“PIN Number”

If you open the .csv file in a spreadsheet program, the quotes will be omitted.

Network

Network

System Network Settings

This area is for configuring the system's IP addresses. By default, *Obtain IP and DNS information automatically* is selected and the current IP addresses are shown.

1. Change *Obtain IP and DNS information automatically* to *Use configured IP and DNS information* in order to lock in the IP addresses.
2. If the system IP settings are blank, enter the following:
   a. Enter a static IP address.
   b. Enter the *Subnet mask* for the LAN.
   c. Enter the IP address of the *Default gateway* on your network.The router firewall may act as the default gateway.
   d. Enter the IP address of the *Preferred DNS server*. The gateway may act as the DNS server.
   e. If applicable, enter the IP address of the *Alternate DNS server*.

Audio Server

The audio server handles all audio for the system. Its IP address must be different than the system IP address. The *Audio Server* area allows you to assign an IP address to it.

Public Network 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* supplied by your ISP (Internet Service Provider):
   - *Dynamic public IP address* — This is the default setting.
   - *Static public IP address*

   If *Static public IP address* is selected, enter the *Current public IP address*.
2. If you have a *Dynamic public IP address* and require external IP phones or have multiple locations, a *Public domain name* is required.

   A DDNS (Dynamic Domain Name Service) provider such as www.dyndns.com can create a public domain name.

   Ensure your router supports your DDNS service, and configure it to update the DNS servers.

   Enter the *Public domain name*.

Firewall Settings

If you are setting up external IP extensions, or have multiple locations, port forwarding is required.

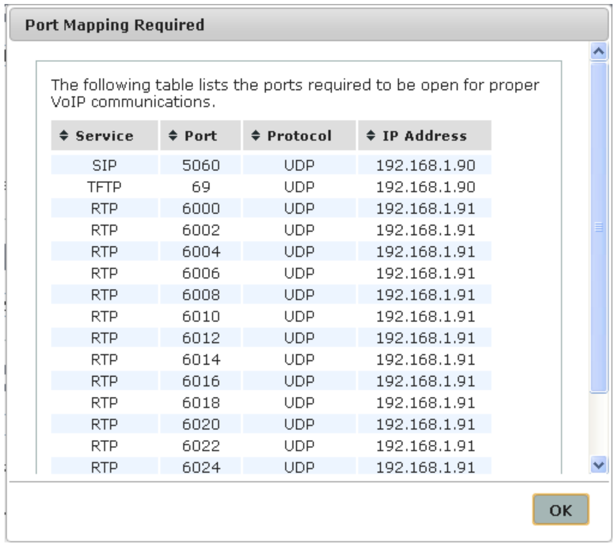
In the *Firewal Settings* area, either *UPnP Enabled* or *Manual port forwarding required* will appear.

UPnP Enabled

1. Click the *UPnP Enabled* link. The port status will show whether all ports were successfully opened.
2. If any ports were not successfully opened, check the configuration of your firewall.

Manual port forwarding required

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


Port Mapping Required

The following table lists the ports required to be open for proper VoIP communications.

Service	Port	Protocol	IP Address
SIP	5060	UDP	192.168.1.90
TFTP	69	UDP	192.168.1.90
RTP	6000	UDP	192.168.1.91
RTP	6002	UDP	192.168.1.91
RTP	6004	UDP	192.168.1.91
RTP	6006	UDP	192.168.1.91
RTP	6008	UDP	192.168.1.91
RTP	6010	UDP	192.168.1.91
RTP	6012	UDP	192.168.1.91
RTP	6014	UDP	192.168.1.91
RTP	6016	UDP	192.168.1.91
RTP	6018	UDP	192.168.1.91
RTP	6020	UDP	192.168.1.91
RTP	6022	UDP	192.168.1.91
RTP	6024	UDP	192.168.1.91

2. Configure your router to forward these ports.
   For information on configuring routers and mapping ports, visit http://www.portforward.com/english/routers/port_forwarding/routerindex.htm.

Admin

Admin Accounts

The *Admin Accounts* tab allows the creation of additional accounts for administering the phone system. Accounts can have full or read-only access to the configuration.

1. Enter a username.
2. Set a password.
3. Enable the permission to provide full access, or leave it disabled to provide read-only access.
4. Select the language that will display in the interface for each administrator. You can select *English*, *French* or *Spanish*.
5. Click to add the account.

To change or reset the password of an account:

1. Click to expand the profile.
2. Click *Change Password*.
3. Enter the new password.

To remove an admin account:

1. Click .
2. Click .

Configuration

Email Server

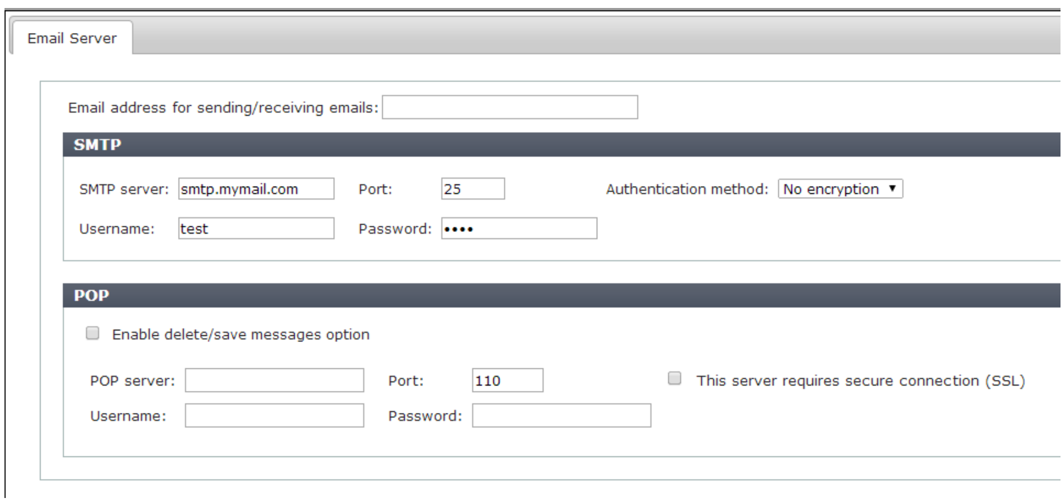
The system can send an e-mail notification of incoming voice messages. The e-mail includes the caller ID, and can include the message as an attachment.

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

For more information, see “Voicemail tab” on page 32.

The *Email Server* page allows you to set up the e-mail server parameters and test the e-mail server.

1. Select the *Email Server* tab.



2. Set up an e-mail account for your system.
3. Enter the address in the *Email address for sending/receiving emails* field. When e-mail notifications arrive from the system, this e-mail address will show up in the *From* field.

In the *SMTP* section:

1. Enter the *SMTP server* for the address.
2. Enter the *Username* of the account.
3. The default *Port* is 25. If required, enter a different port number.
4. Enter the *Password* of the account.
5. Select an authentication method for your server.

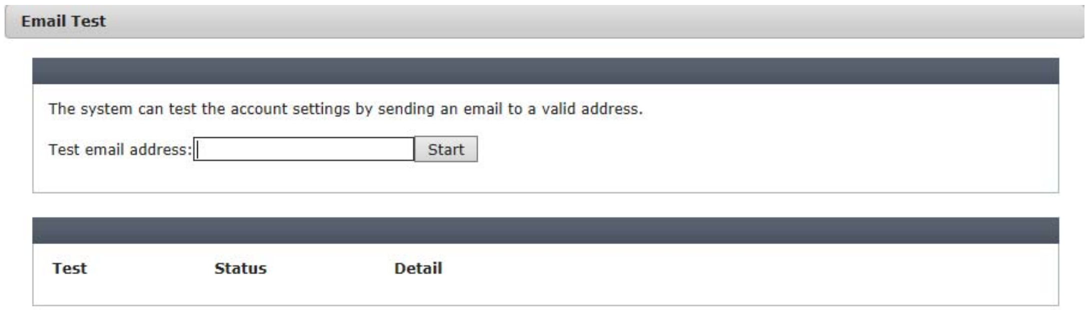
In the *POP* section:

1. Set the POP values in the *Incoming Server (POP)* area. Enter the POP address.
2. Enter the *Username* of the account.
3. The default *Port* is 110. If required, enter a different port number.
4. Enter the *Password* of the account.

5. Check the box if your server requires SSL.

Testing the e-mail server settings

1. Click *Test*.
2. Enter an e-mail address in the *Test email address* box, and then click *Start*.



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

   - If 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.

3. Ensure your e-mail address has received the test e-mail. Note that the e-mail may have been routed to a junk or spam folder.

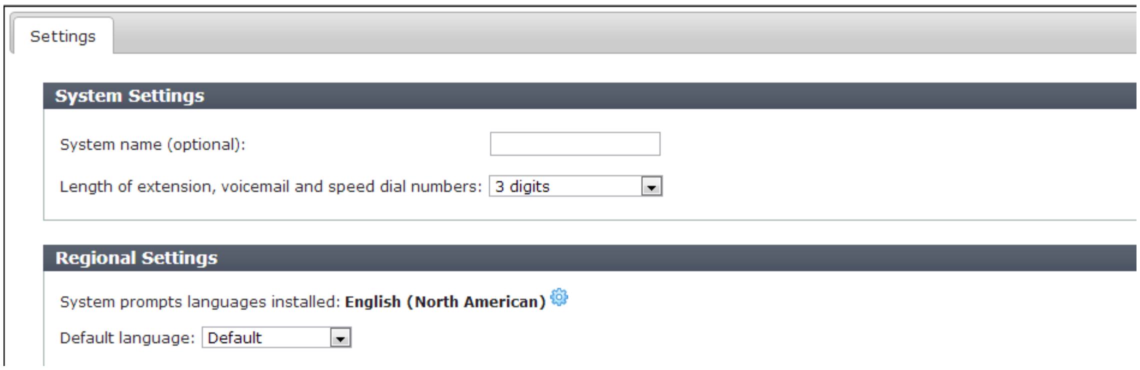
Settings

Settings

System Settings

This area allows you to set up the system name and the numbering plan.

Select the *Settings* tab.



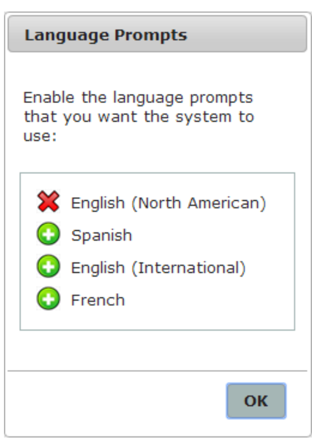
You can use a 3, 4 or 5-digit plan for extensions, ring groups, mailboxes and speed dials in your system. The default is 3 digits. To change it, select the desired number of digits. This selection is system-wide; you can't have a mix, for example, of 3 and 5-digit extension numbers.

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

Regional Settings

This area allows you to select the country where your system will operate to ensure it functions correctly. It also displays the prompt 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 . The *Language Prompts* window appears, listing loaded language files.



   a. To load a language file, click , and select the language file.

   b. To remove a language file, select the language, and click .

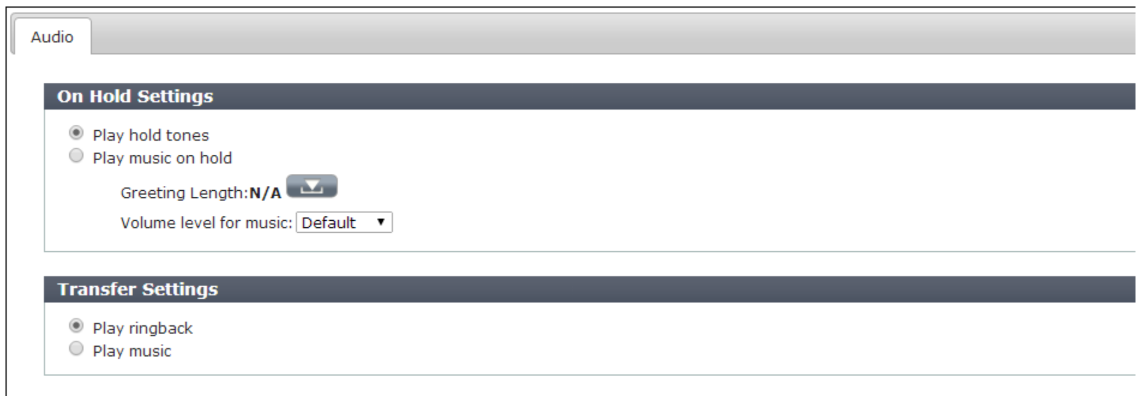
2. Select the default language for prompts heard by callers and users in the Default language for system prompts to callers list.

Audio

Audio

On-Hold Settings

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.



1. Select the sound to play while the caller is on hold. Choices are:
   - *Play hold tones.*
   - *Play music on hold.*
2. If you selected *Play music on hold,* click to upload a wave file.
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.
   c. Click *Save*.

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.

Select the sound to play while the caller is being transferred. Choices are:

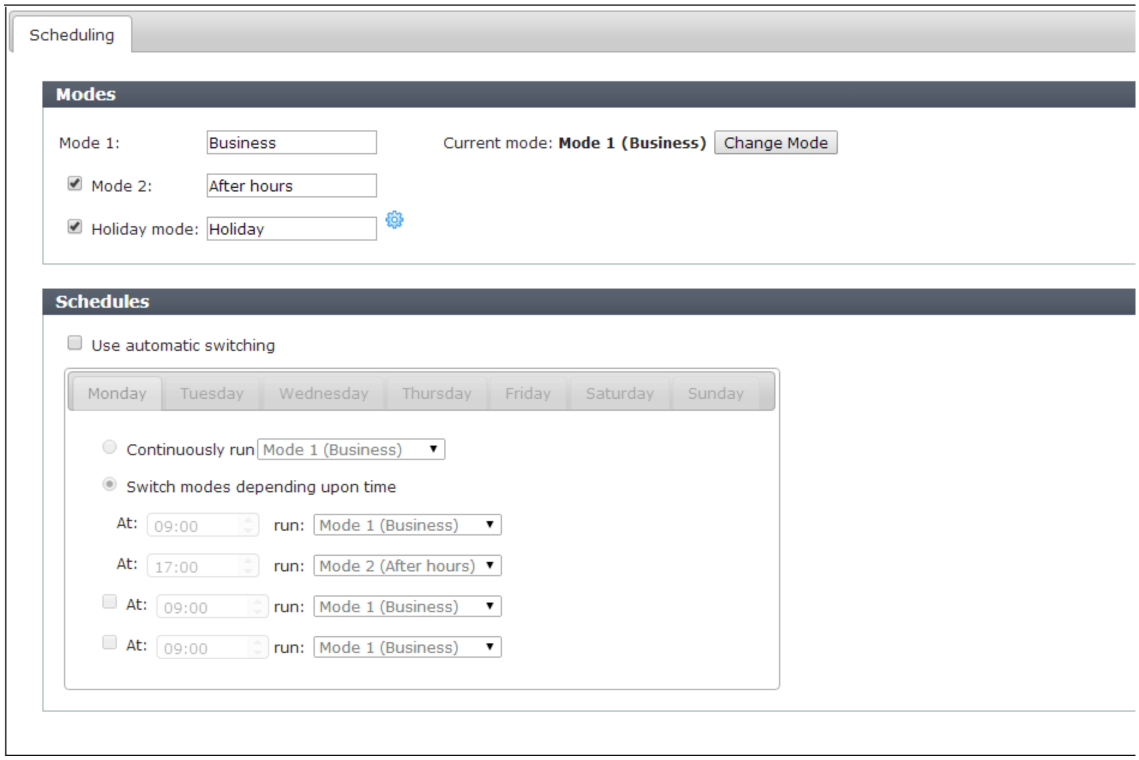
- *Play ringback*— Plays ringing.
- *Play music* — Plays music on hold as configured in the *On-Hold Settings* area.

Scheduling

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.

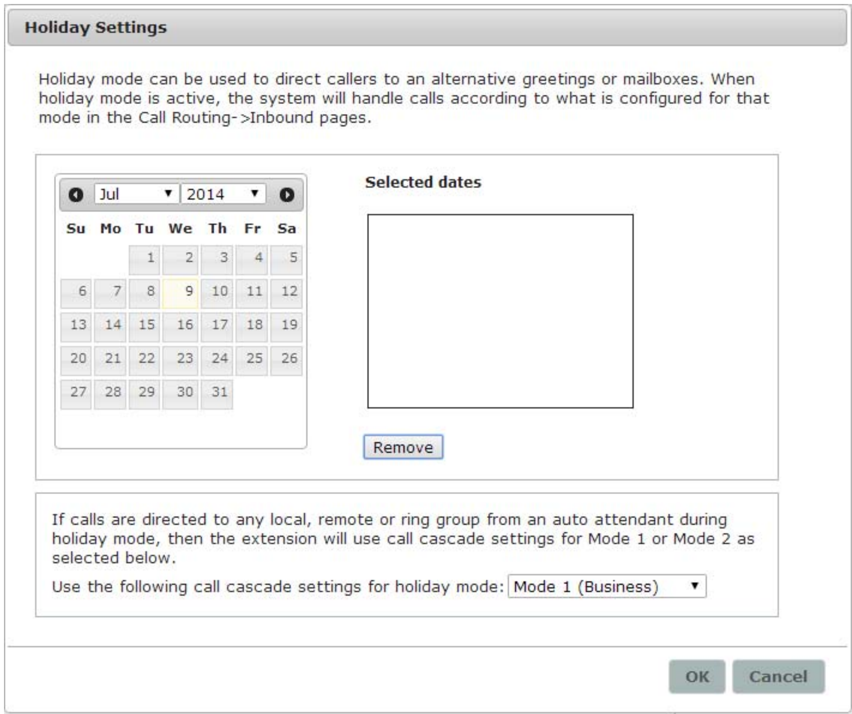
The system can automatically change mode with the time of day, day of week and on holidays.



Modes

1. Enter a label for Mode 1.
2. Activate and enter a label for Mode 2.
3. Set up Holiday Mode.
   a. Select the *Enable holiday mode* checkbox. The window enables the *Settings* button.
   b. Enter the *Holiday Mode label*.

c. Click . The *Holiday Settings* window appears. The calendar shows the current date in green.



d. Click on the holidays in the calendar to add them to the *Selected dates* area, and the calendar will show the date in blue. Click the date again to remove it.

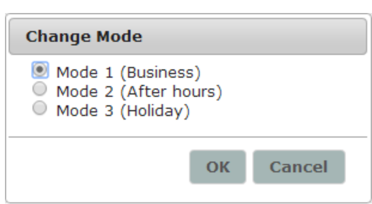
e. Repeat the above step until all required holidays have been added.

f. Select the mode in the *Use the following call cascade settings for holiday mode*. Extensions will use their Mode 1 or Mode 2 call cascades during holiday mode. Select the mode from the *Use the following call cascade settings for holiday mode* list.

Change mode

The *Modes* area displays the current mode.

1. To change mode, click *Change Mode*.



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

Schedules

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

1. Select the *Use automatic switching* checkbox to enable the modes. The window enables the *Schedules* 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 upon 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 upon 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.

Profiles

VoIP

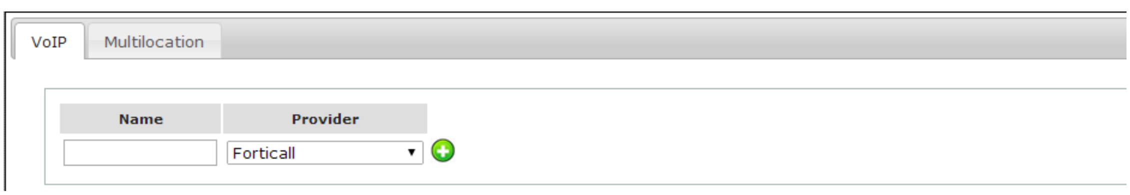
VoIP Configuration

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

The *VoIP* tab area allows you to set up to four service provider profiles. You can add a service provider from a predefined template or manually.

Service configuration guides for certified VoIP service providers are available online.

Select the *VoIP* tab.



Automatic configuration

1. Enter a *Name* for the VoIP profile.
2. Select *FortiCall* as the Provider.
3. Click .
4. If you want to customize any aspects of your VoIP lines, click .

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

Manual configuration

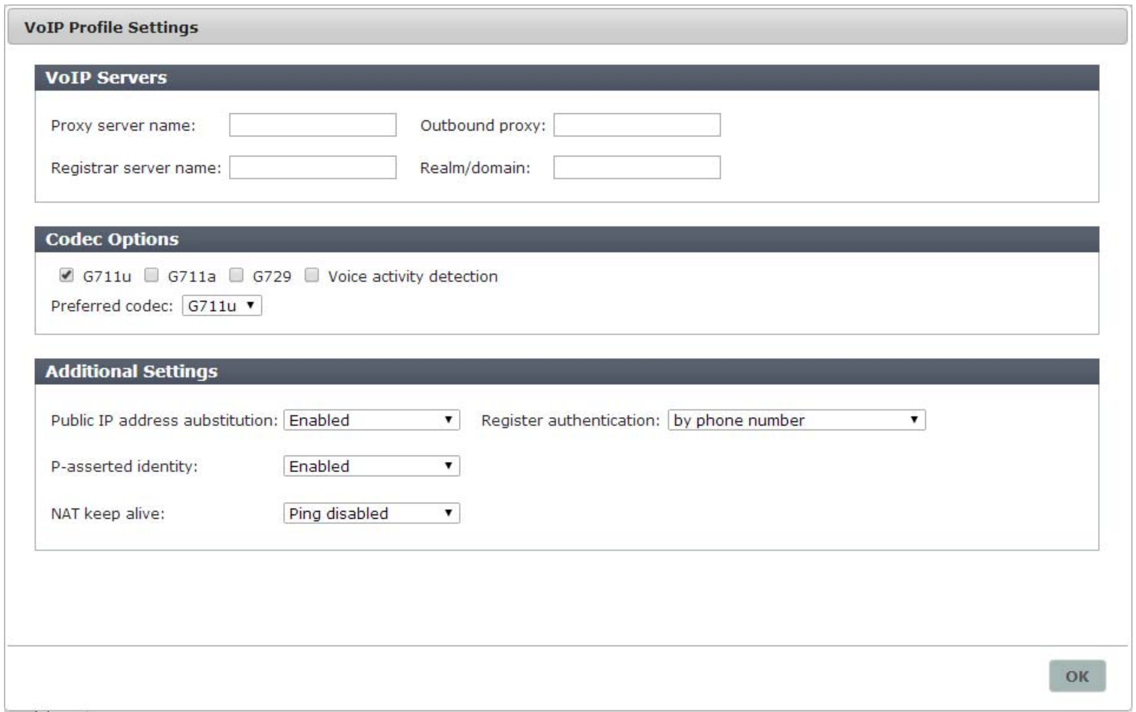
1. Enter a *Name* for the VoIP profile.
2. Click .
3. Enter all service-related information including server, codecs and authentication method.

VoIP Settings

VoIP Profile Settings

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

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



Setting codec options

A codec is a method of compressing and decompressing audio signals for communication across a network. The system supports the G729 and G711 (u-law or A-law) codecs for VoIP calls. If your VoIP 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:

- *G711u* — This codec provides high quality and supports Fax over IP. It requires the most bandwidth and accommodates the fewest number of concurrent calls. G711u is used in North America and Japan.
- *G711a* — This codec provides high quality and supports Fax over IP. It requires the most bandwidth and accommodates the fewest number of concurrent calls. G711a is used worldwide outside North America and Japan.
- *G729* — This codec provides good quality. It requires the least bandwidth and accommodates the highest number of concurrent calls.
- *Voice activity detection* — Enabling this reduces voice bandwidth when no speech is detected, and reduces transmission of background noise. We recommend disabling 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*.

Additional settings

Public IP

If your VoIP provider requires you to register using your private IP address, select *Disabled* in the *Public IP address substitution* pulldown menu. Check with your VoIP provider.

Preferred ID

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

NAT

If your VoIP provider requires keep alive messages, and if your router does not support uPNP, select *Simple Ping Enabled* under the *NAT keep alive* pulldown menu.

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.

Multilocation

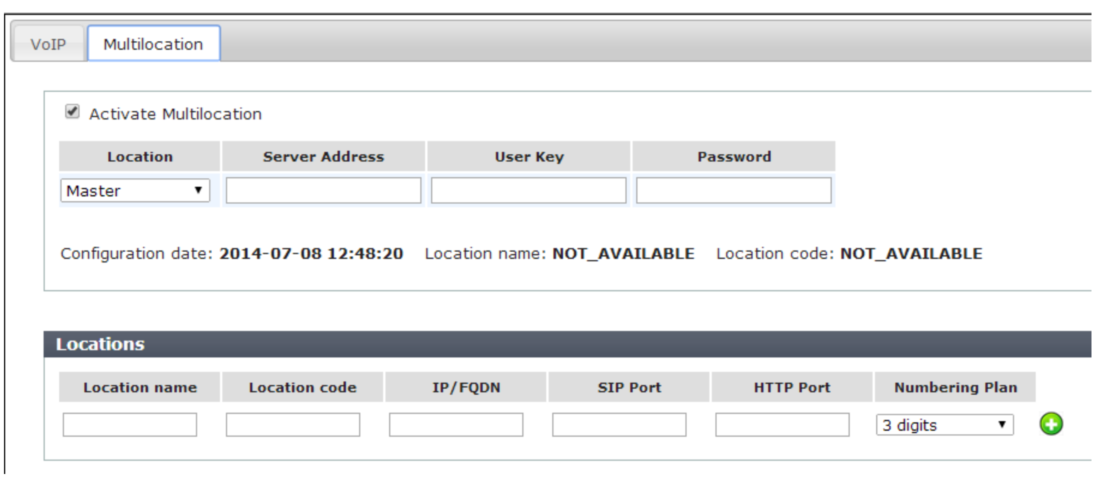
Configuring multiple locations

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 multilocation directory is set by the central administrator and instantly broadcast to all other locations.

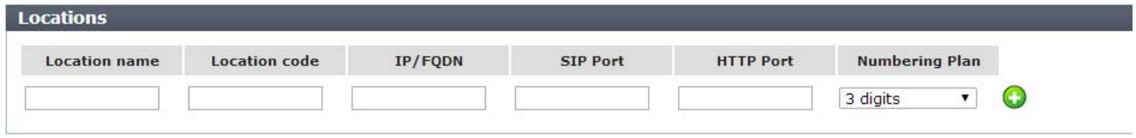
Setting up multilocations: master system

1. Select the *Multilocation* tab.



2. Select *Master* as the *Location*.
3. Enter your public IP address as the *Server 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.

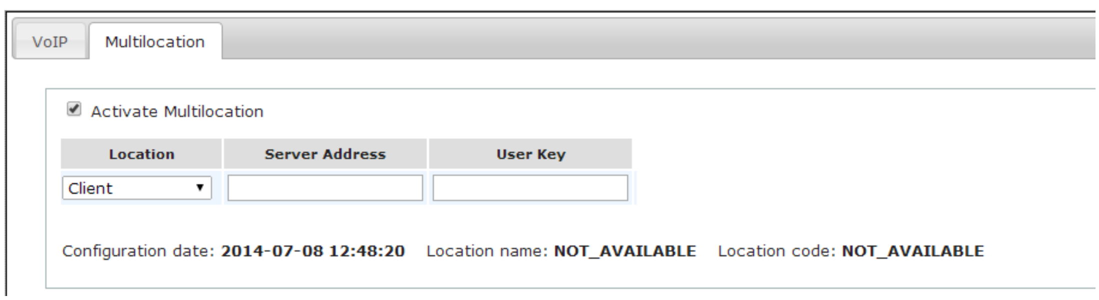
Add the locations



1. Enter the *Location name*.
2. Assign a *Location code*. Location codes can be 2 or 3 digits. Each location must have the same number of digits.
3. Enter the IP address or Fully Qualified Domain Name of the location.
4. Enter the *SIP Port*.
5. Enter the *HTTP Port*.
6. Select the number of digits in the location's numbering plan. For optimum usability, each location should use the same number of digits in extensions, but the plan will work if they differ.
7. Click ⊕ to add another location.
8. Repeat steps 1 to 6 for each location, including the master location.

Setting up multilocation: all other systems

1. Select the *Multilocation* tab.



2. Select *Client* as the *Location*.
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*.

Security

DoS Blacklist

The DoS Blacklist table displays the IP addresses that have been banned by the system.

To release an IP address from the system:

1. Click on the IP address.
2. In the popup window, click *Release*.

Extensions

IP Extensions

A local extension is an IP extension.

An IP extension is a phone connected through a network to the system. An internal IP extension is a phone connected on the LAN of the system. An external IP extension is connected outside the LAN.

You can import an extension list created with a text editor or a spreadsheet program. Each entry in the file requires the first name, last name, extension number, phone type, location and MAC address (if applicable). If you're using a text editor, the values must be separated by commas. The file must be saved as a .csv file.

Adding IP phones

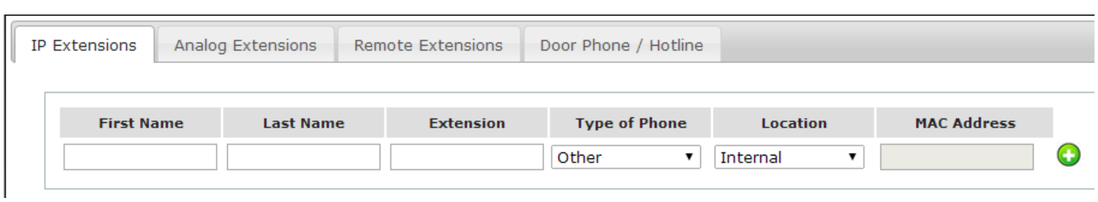
This section describes how to configure an IP phone as a local extension. An IP phone can be internal (located in the office) or external (located outside the office).

Connecting the IP phone to the network

1. Connect a network cable between the LAN port on the phone (marked) and your network (i.e. router or LAN connection). The phones also have a PC port that can be used to connect the PC to the network.
2. Connect power to the phone, either using the optional power adapter or an 802.3af Power-over-Ethernet (PoE) source.

Adding the extension to the system

1. Select the *IP Extensions* tab.



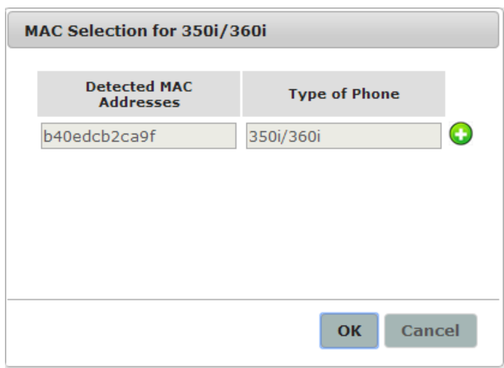
2. Enter the user's *First Name* and *Last Name*. The names are used for caller ID and the name directory.
3. Assign an *Extension* number.
4. Select *Type of Phone*.

5. 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. See your phone's QuickStart 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. See "Emergency Zones" on page 49 for configuring calls to emergency numbers.

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

6. Enter the *MAC Address* of the phone:
   - You can select the *MAC Address* from a list of automatically-detected phones connected to your LAN. To use this method:
     - **i.** Click the *Lookup* link. 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 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.

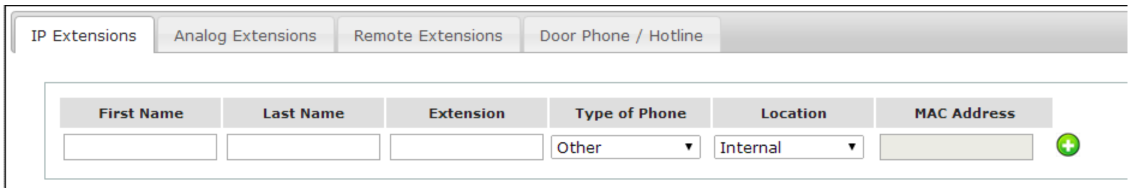
   MAC: 001A7EA75DD5

   Click *Finish*.
7. Click .
8. Reboot the phone. Once complete, the phone will display the extension name and number.
9. Click beside the extension if you wish to change user privilege or set a username and password.

10. Click to open the *Additional Settings* page where you update preferences, call handling, voicemail settings and key appearance (not available on 860i or 870i IP phones). For further details, see "Additional Settings" on page 26.

Adding other 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.

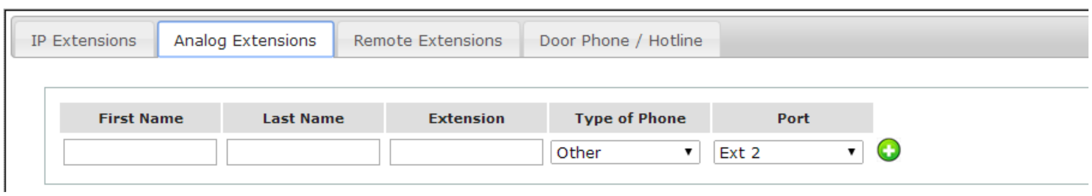


Refer to the specific IP phone user guide's details for information on configuration.

Analog Extensions

Adding analog phones

1. Select the *Analog Extensions* tab.



2. Enter the user's *First name* and *Last name*. The names are used in the name directory and caller ID.
3. Enter the *Extension number*.
4. Select the *Type of Phone*.
5. Select the jack the analog phone is plugged into.
6. Click .
7. Click beside the extension if you wish to change the user privilege.
8. Click to open the *Additional Settings* page where you update preferences, call handling and voicemail settings. For further details, see "Additional Settings" on page 26.

Remote Extensions

You can import a remote extension list created with a text editor or a spreadsheet program. Each entry in the file requires the first name, last name, extension number 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.

Adding remote extensions

The remote extension feature integrates outside phones with your system. A remote extension can be a mobile phone, home phone or any phone anywhere.

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.

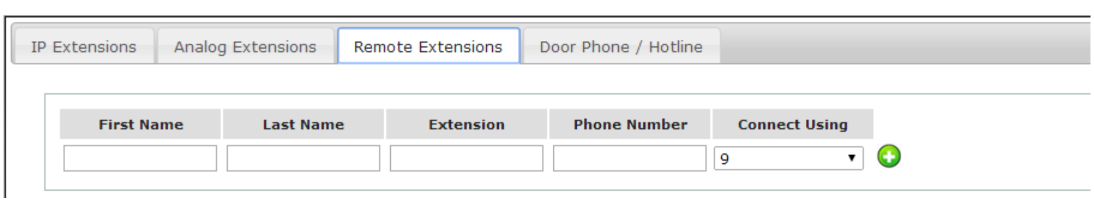
The remote extensions can be configured to perform a blind transfer or a screened transfer. However a transfer to another remote extension is always a blind transfer to avoid tying up a third line.

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.

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.

1. Select the *Remote Extensions* tab.



2. Enter the user's *First Name* and *Last Name*. The names are used for caller ID and the name directory.
3. Assign an *Extension* number.

4. 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.
5. Select the outgoing access in the *Connect Using* list. The unit will use a line from this group to connect with the remote extension. We recommend the default *9* (or *0* in certain regions) unless you have set up a different access code for calling remote extensions.
6. Click .

   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 group setting from Step 6 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 group selected in Step 6.

   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.

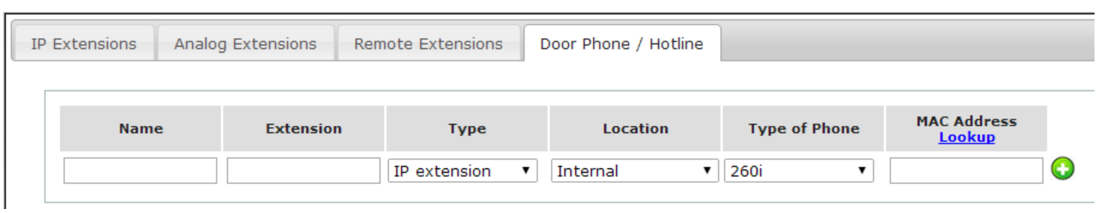
   Set the type of transfer performed by users at remote extensions. This affects transfers to local extensions and ring groups. Transfers to other remote extensions are always blind transfers.

Door Phone/Hotline

Adding door phones/hotline

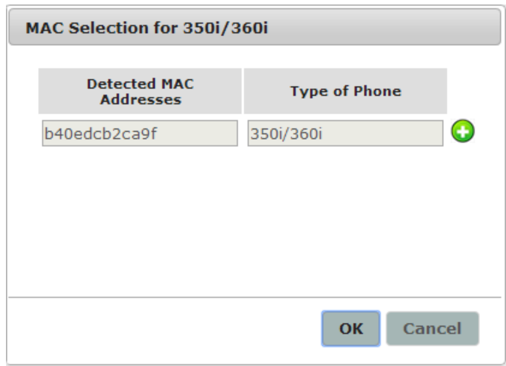
An analog phone, or a 3x0i, 4x0i or 5x0i IP phone will automatically connect to the resource when you lift the handset, or press the speaker or headset button. Other IP phones do not support hotline access.

1. Select the *Door Phone/Hotline* tab.



2. Enter a *Name* for this resource.
3. Assign an *Extension* number.
4. Select *Type*.
5. Select 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 your phone's QuickStart Guide for further details.
6. Select *Type of Phone*.

7. Enter the *MAC Address* of the phone if required.
   - You can select the *MAC Address* from a list of automatically-detected phones connected to your LAN. To use this method:
     i. Click the *Lookup* link. 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 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*.
8. Click .

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.

1. Click .
2. Select the action in the *Connect to the following resource* list. Choices are:
   - *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 extension group* — Connects to the selected extension 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.

Additional Settings

The *Additional Settings* window enables you to set up or customize access to lines or outgoing access code, designate the extension as a hotline and select Caller ID display options.

Click ⚙ to open the *Additional Settings* window.

Preference tab

Prompt language

Select the language for prompts heard by the user of the extension in the language list.

Direct line access

Direct Line Access allows you to select the outgoing access codes that the 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 outgoing access codes. Enable direct line access and select the access code. As soon as the fax goes off-hook, it finds an available line within the group.

Use the dropdown to select an outgoing access code to enable this feature.

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
- Speed dial numbers
- Calling the receptionist
- Attaching an account code
- Intercom paging
- Group paging
- Overhead paging
- Stutter dial tone for new voicemail
- Voicemail retrieval/access
- 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 dial the number or function.

Time Zone

Select the time zone that matches the location of the IP phone.

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 *Preferences* 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 first 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.

Call handling

About call cascades

The *Call Handling* tab 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.

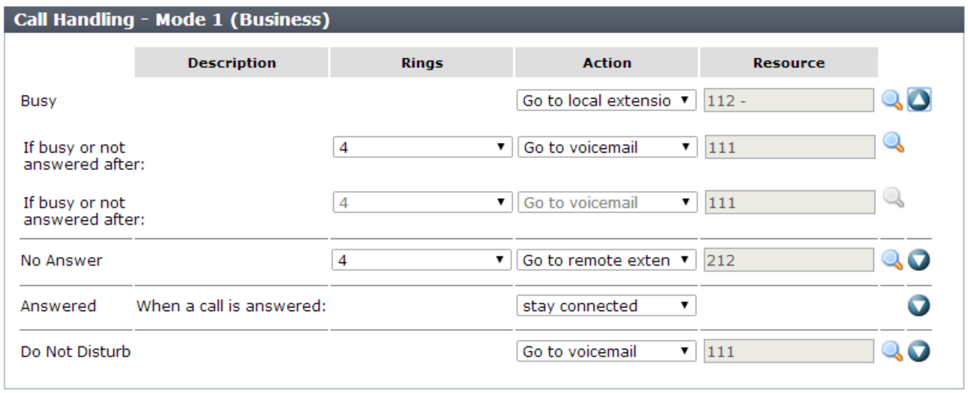
For example, if a call reaches your desk when you're away, it can be sent to another local extension. If that extension isn't answered, the call can be routed to your cell phone. If you don't answer your cell phone, the call can go to your voicemail.

Setting up call handling

Setting busy call cascade

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

1. Select the *Call Handling* tab.



2. Set up the first alternative.

  a. Select the action in the *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 extension group* — Attempts to transfer the call to the selected extension group.
  - *go to 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. Click to find resources.

3. If permitted, set up the second alternative by clicking .

  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.

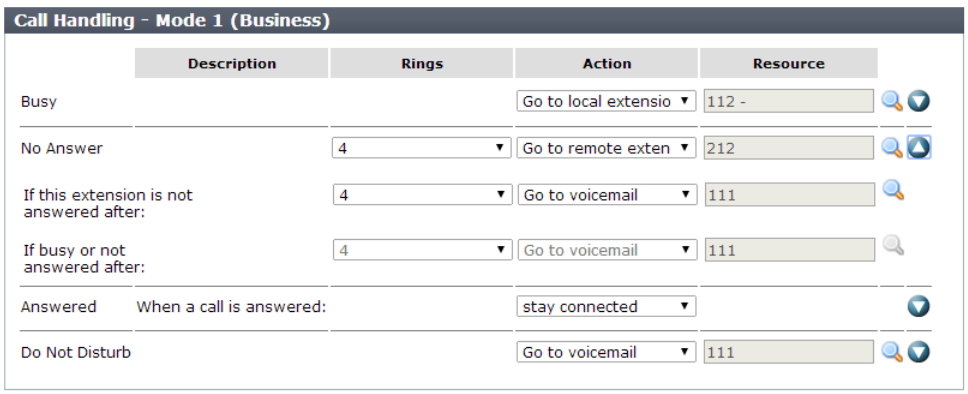
  c. Select the *Resource*, if applicable.

4. 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*
      - *go to announcement*
      - *hang up*
   c. Select the *Resource*, if applicable.

Setting no answer call cascade

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

1. Select the *Call Handling* tab.



2. Set up the first alternative.
   a. Set the *No Answer* 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 extension group*
      - *go to announcement*
      - *go to auto attendant*
      - *go to extended ringing*
      - *hang up*
   c. Select the *Resource*, if applicable.
3. If permitted, set up the second alternative by clicking .
   a. Set the *If this extension is 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 extension group*
      - *go to announcement*

- *go to auto attendant*
- *hang up*

c. Select the *Resource*, if applicable.

4. 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:

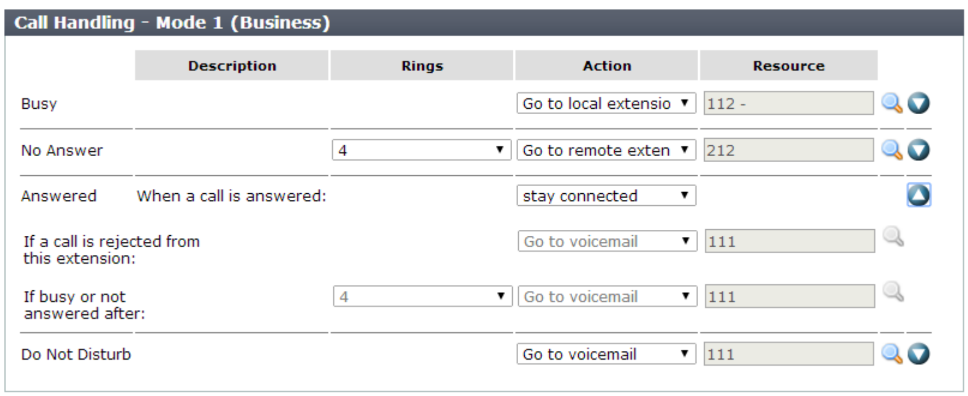
- *go to voicemail*
- *go to auto attendant*
- *go to announcement*
- *hang up*

c. Select the *Resource*, if applicable.

Setting answered call cascade

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

1. Select the *Call Handling* tab.



2. 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.

3. Set up the first alternative.

a. Set the *If a call is rejected from this extension* action.

b. Select the *Resource*, if applicable.

4. If permitted, set up the second alternative by clicking .

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.

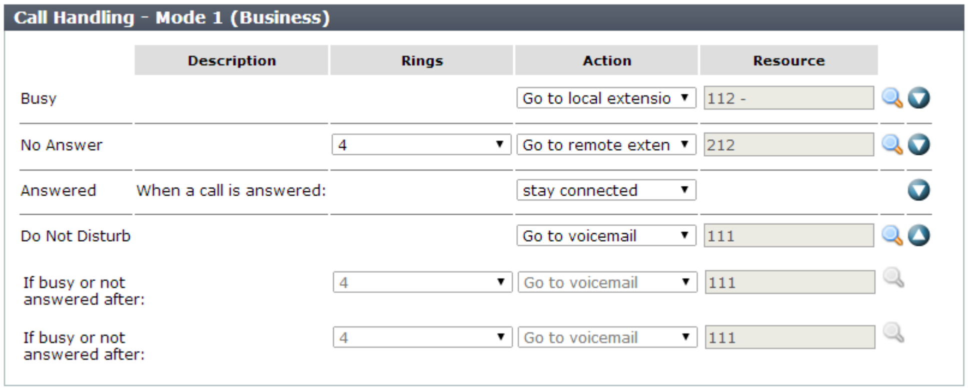
c. Select the *Resource*, if applicable.

5. If permitted, set up the third alternative.
   a. Follow the same steps as the second alternative.

Setting do not disturb (DND) cascade

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

1. Select the *Call Handling* tab.



2. Set up the first alternative.
   a. Set the *Do Not Disturb* list. Choices are:
      - *go to voicemail*
      - *go to local extension*
      - *go to remote extension*
      - *go to extension group*
      - *go to announcement*
      - *go to auto attendant*
      - *hang up*
   b. Select the *Resource*, if applicable.
3. If permitted, set up the second alternative by clicking .
   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.
4. 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 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.
3. Press * to exit command mode.

Voicemail

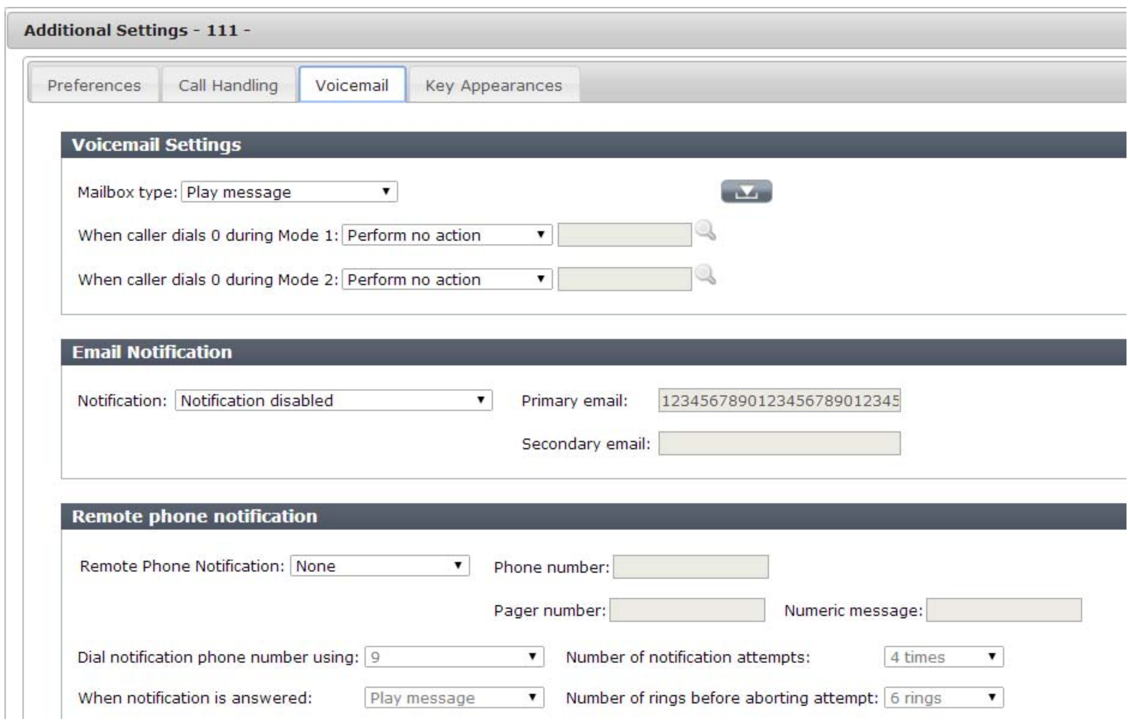
Voicemail tab

The *Voicemail* tab allows you to set 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.

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.

Click . This will open the *Additional Settings* page.



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

Voicemail Settings

The *Voicemail Settings* 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, 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 .

c. Browse to select a .wav file. Click to load the .wav file into the system.

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 extension group*
   - *go to voicemail*
   - *go to announcement*
   - *name directory*
   - *perform no action*
2. Depending on the *Action*, enter the *Resource*.

Email Notification

The *Email Notification* area allows you to add an e-mail address to receive e-mail notifications when voicemails arrive.

1. 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.

     This notification option allows the recipient to play, save or delete the voicemail message.

     i. To play the voicemail message, the recipient double-clicks the attachment. The default .wav player opens the voicemail message.

     ii. 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.

     iii. 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.
   - *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.
2. Enter the recipient’s *Primary email* address.

3. Enter the *Secondary email* address (optional).

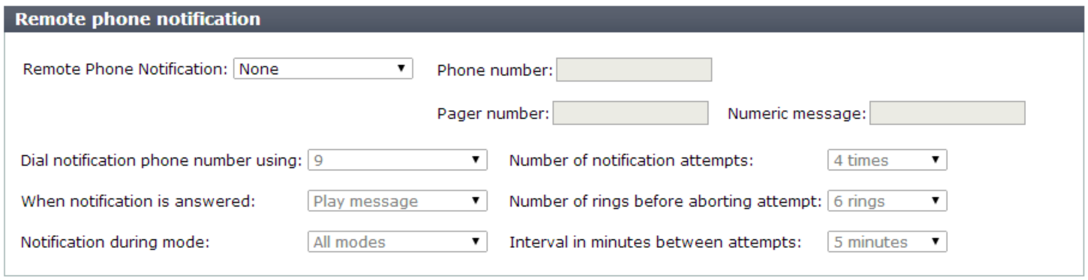
Remote Phone Notification

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

- Notify a user by phone and/or pager.

Setting up notification by phone

1. Select *Cell or Cell and Pager options* from the *Remote Phone Notification* dropdown menu.



2. Enter the *Phone number*. Enter the number as you would normally dial it (i.e. without the outgoing access code). You can enter digits 0–9, space, dash, comma, # and *. A comma pauses dialing for two seconds.

Setting up notification by pager

1. Select *Pager or Cell and Pager options* from the *Remote Phone Notification* dropdown menu.
2. 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.
3. Enter the *Numeric message* that will appear on the user's pager.

Setting up notification options

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

6. Select the notification options when the call is answered. Use the *When notification is answered* dropdown to select one of these choices:
   - Select *Play message* 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 *Play prompt* 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.

Message waiting indicator

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. Go to the *Message Waiting Indicator* area.

2. Click *Edit*.
3. Select the extension you want to notify.

Key Appearance

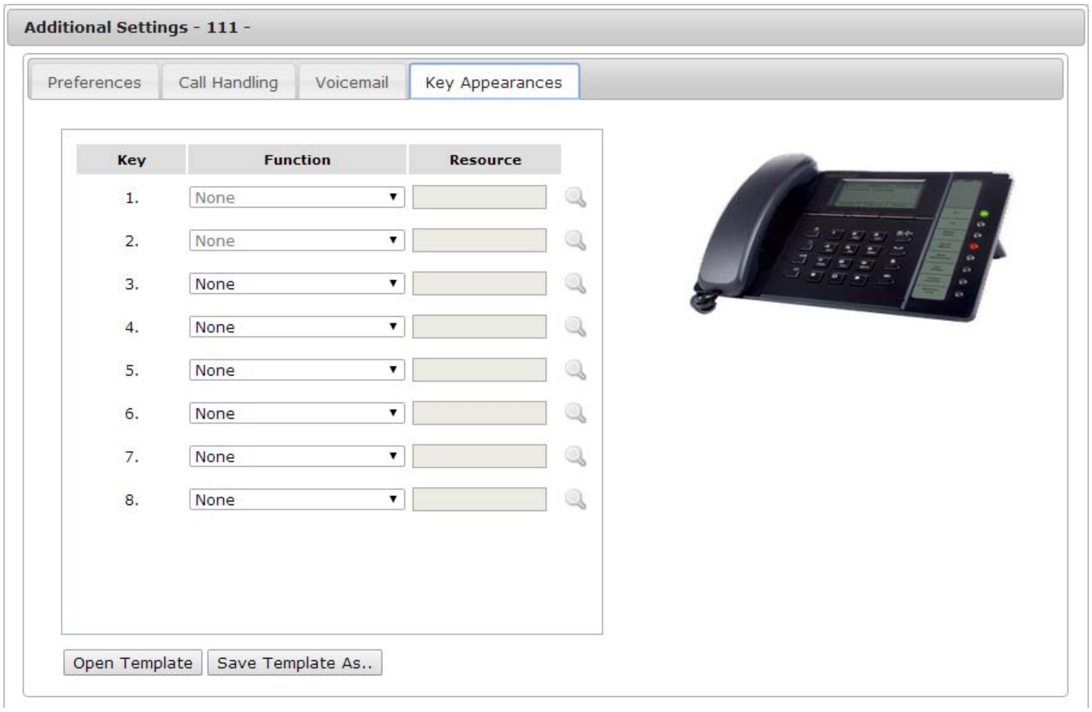
About programmable function keys

The 350i/360i has 6 programmable functions keys, the 450i/460i has 10 programmable keys, and the 550i/560i has 22 programmable keys. The keys allow the user to access features, and to monitor and engage lines, extensions and queued calls (i.e. line appearance).

Note that 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.)

Programming function keys

1. Click the *Key Appearances* tab.



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, Phone Book configuration, or User Defined (phone). For further details, see “Phone programmable key functions” on page 37.

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 36.

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 37.

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*.

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*.

Phone programmable key functions

Supported functions for a phone model typically include most of the items listed below.

- *Line appearance* — The button will allow you to access the line. It will light up when the line is in use, flash if the line is ringing, or be off when the line is available. (not available on the FON-260i)
- *Extension appearance* — Press the button to call the extension. 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. (not available on the FON-260i)
- *Queue appearance* — Press the button to pick up the oldest queued call. The button will flash if calls are queued for the extension, or be off when there are no queued calls.
- *Voicemail* — Press the button to access the voicemail of the extension.
- *DND* — Press the button to toggle Do Not Disturb on or off.
- *Park* — Press the button to put the call on hold in the next available park orbit. The system will respond with the park orbit n4umber.
- *Un-park* — Press the button, select the park orbit number, then press the button to retrieve the call.
- *Pickup any* — Press the button to answer an inbound call ringing any extension.
- *Pickup ext* — Press the button, dial an extension and press # to answer a call ringing the dialed extension.
- *Intercom* — Press the button, dial an extension, and press # to page the extension in Intercom mode. The intercom page function can only page selected FortiFones.
- *Phone book* — Press the button 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. (not available on the FON-260i)
- *User Defined* — The button is assigned using the phone.
- *Speed dial* — Press the button to call the resource.
- *Group page* — Press the button to page the phones in the group. Not all phones can receive pages.
- *Call recording* — Press the button to record a call. The button will flash while recording is on. Press the button again to stop recording. The recording will be stored in the user’s voice mailbox.

Groups

Extension Groups

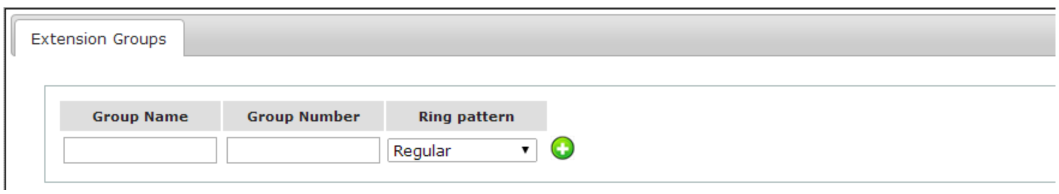
An extension group is a group of local extensions that ring in unison.

There are 10 extension groups available in the system.

Extension groups have two main uses:

- They can reach a group of employees. For example, extension 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.
- An extension 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 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 *Extension Groups* tab.



2. Enter the *Group Name.*

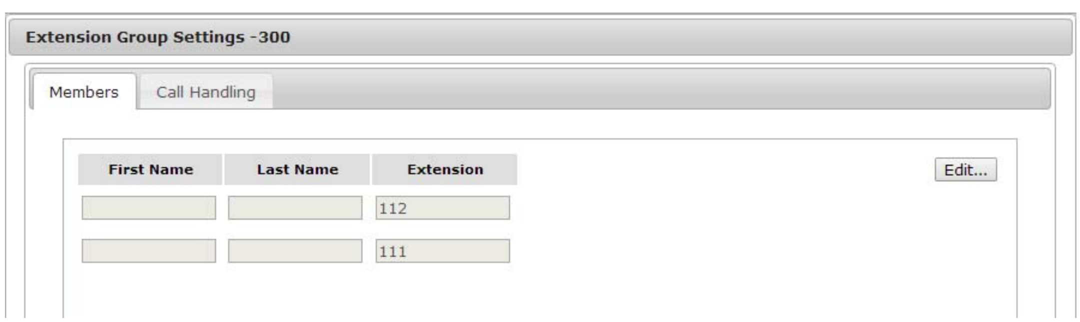
3. Assign a *Group Number.*

4. Select a *Ring Pattern* to indicate the call is for the extension group. Click *Next*.

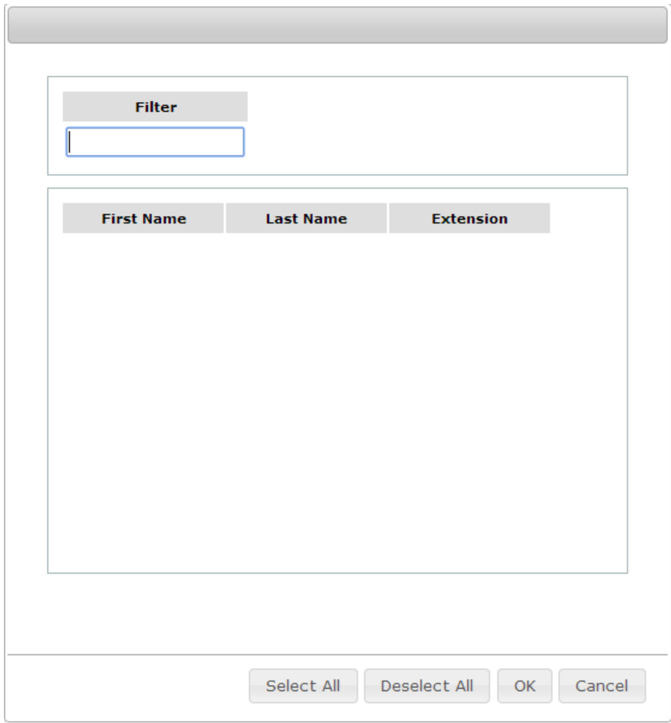
5. Click .

Add member

1. Click to open the *Extension Group Settings* page.



2. Click *Edit* to search a resource.



3. Click .

About call handling

See “Call handling” on page 28.

Setting up call handling

The procedure for setting up a call cascade is similar to that described in *IP Extension* with the differences described below. See “Call handling” on page 28.

- The *Extension 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 ext group* option replaces *queue call*, and places the call in the 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 extension group, and not them personally.

Two options can be selected or combined:

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

General Voicemail

General Voicemail

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.

1. Select the *General Voicemail* tab.

2. Enter a mailbox *Name.*
3. Assign a *Number.* Click *Next*.
4. Click . The *Add General Mailbox* window appears.
5. Click to open the *Voicemail Settings* page.

Voicemail settings

The procedure for Voicemail settings is identical to that described in *IP Extensions*. See “Voicemail Settings” on page 32.

Notification options

The procedure for setting up the notification settings is identical to that described in *IP Extensions*. See “Remote Phone Notification” on page 34.

Trunks

VoIP

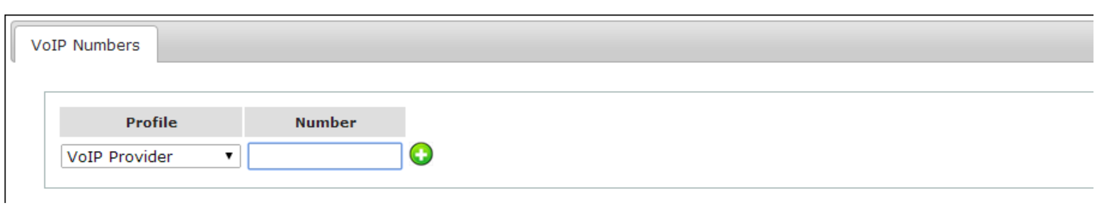
VoIP Numbers

A VoIP number is a telephone number that allows a caller to dial the system through a VoIP provider.

You can set up unique call handling for each VoIP number. 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 an extension, to an auto attendant, extension group, voice mailbox or announcement.

Add VoIP number



1. Select the VoIP *Profile*.
2. Enter the *Number*.

3. Click .
4. Click .
5. Enter the *Username and Password* required to access the service provider’s SIP server.

   If the *Register Authentication for a VoIP Profile* is set to *Master Account*, the username and password will automatically be populated.

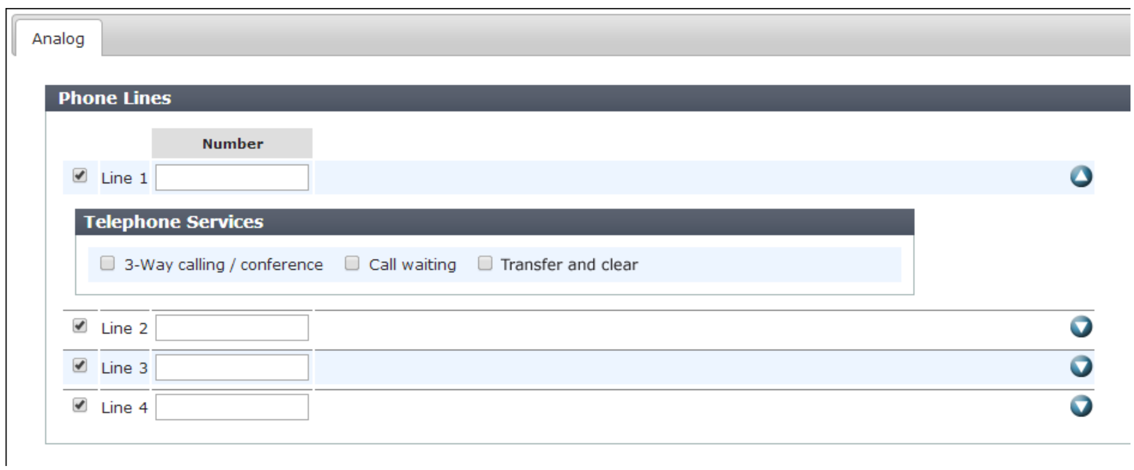
Registration status

See “VoIP” on page 6.

Analog

Phone Lines

The *Phone Lines* area allows you to activate a telephone line, set up the telephone numbers, telephone company services, and call handling for each telephone line.



Telephone Services

The *Telephone Services* area allows you to select the telephone company services that are active on the telephone line. Click to view these options:

- *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.
- *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.

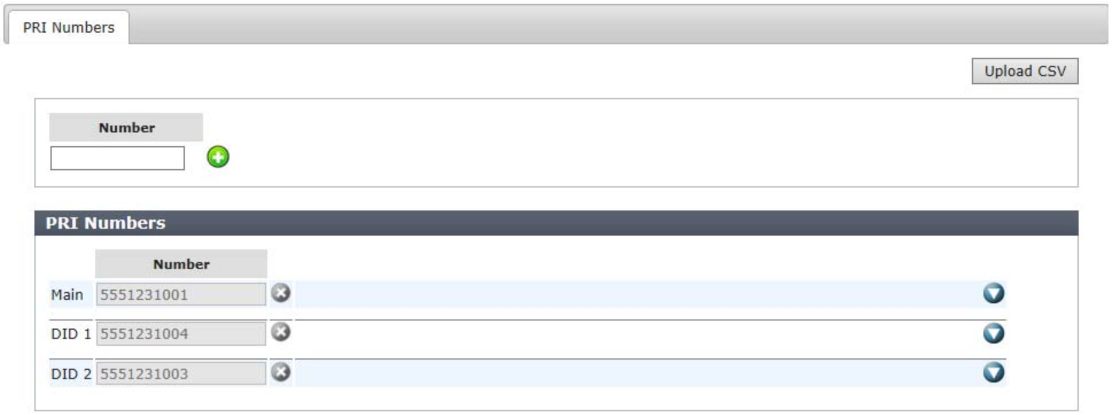
PRI

PRI Numbers

A PRI number is a telephone number that allows a caller to dial the system through a 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 an extension, to an auto attendant, extension group, voice mailbox or announcement.



Add number

1. Enter the *Number*.
2. Click .

Call Routing Inbound

VoIP Numbers

The *Call Handling* area allows you to set up what happens when a call comes in from this number. Calls can be handled differently in each mode. Calls can be sent directly to an extension, go to an auto attendant, extension group, voice mailbox, or announcement.

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 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. Click beside the number.
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 extension group* — goes to the specified extension group. The call will follow the call cascade for that extension group.
   - *go to voicemail* — Accesses the selected voice mailbox.

PRI Numbers

The *Call Handling* area allows you to set up what happens when a call comes in from this number. Calls can be handled differently in each mode. Calls can be sent directly to an extension, go to an auto attendant, extension group, voice mailbox, or announcement.

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 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. Click beside the number.
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 extension group* — goes to the specified extension group. The call will follow the call cascade for that extension group.
   - *go to voicemail* — Accesses the selected voice mailbox.

Analog Lines

The *Call Handling* area allows you to set up what happens when a call comes in the line. Calls can be handled differently in each mode. Calls can be sent directly to an extension, go to an auto attendant, extension group, voice mailbox, or announcement.

1. Click beside the number.
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 extension group* — goes to the specified extension group. The call will follow the call cascade for that extension group.
   - *go to voicemail* — Accesses the selected voice mailbox.

Routing Groups

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 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.

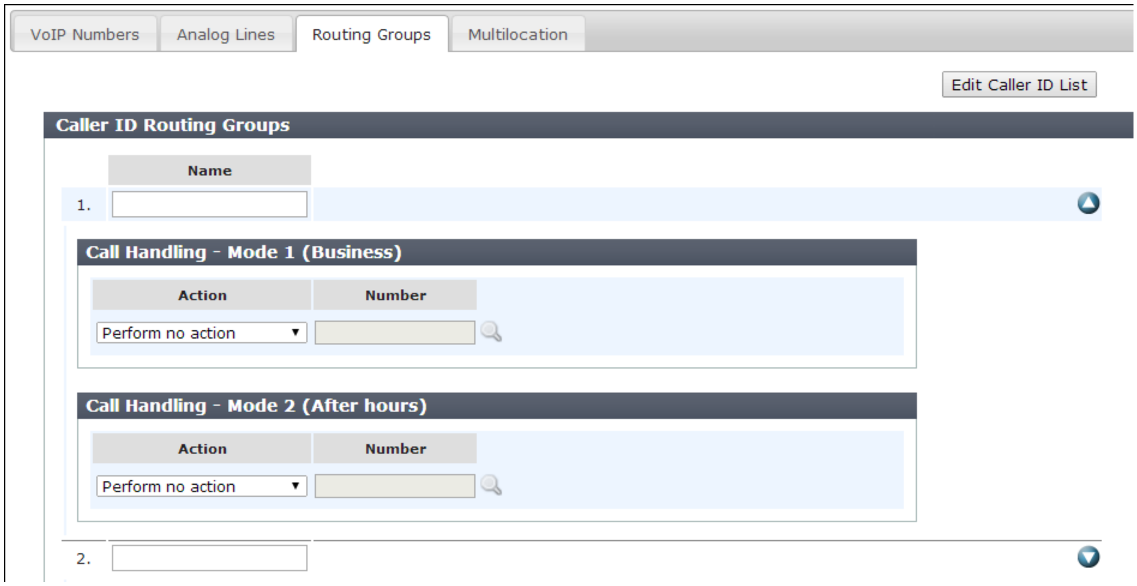
You can define up to 200 caller ID entries. Each caller ID entry has an optional name, a phone number and a routing assignment. The routing assignment can use call handling set up for a group, or can display an alternate name on the user's extension, instead of the name from the caller ID.

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 analog lines, except you can set the ring pattern for each group. Depending on the caller ID, calls can be sent directly to an extension, go to an auto attendant, extension group, voice mailbox, or announcement.

Select the *Routing Groups* tab.



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

Caller ID List

Caller ID Routing (or *CLID routing* in some regions), allows you to define up to 200 caller ID entries. Each entry has a name, a pattern 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. Select the *Routing Groups* tab. Click *Edit Caller ID List*.
2. Select the *Function 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.

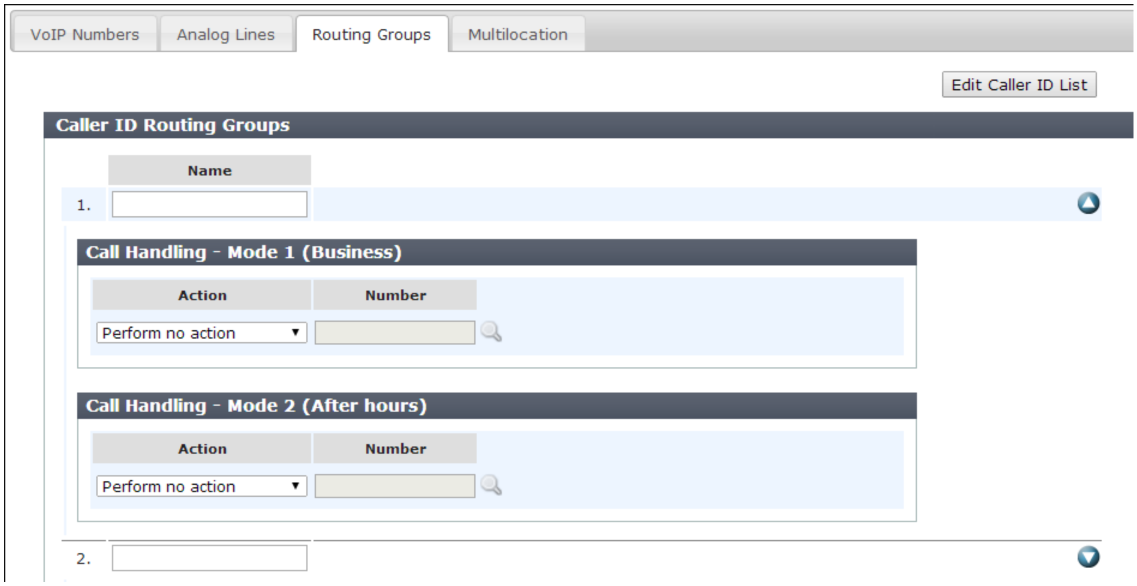
3. If you selected *Phone number ends with* or *Phone number starts with*, enter the digits in the *Pattern* box.
4. Enter the name in the *Caller ID name* box. The extension will display this alternate name, instead of the name from the caller ID.
5. Select the routing in the *Route Using* list. Choices 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.
6. Click .

Caller ID Routing Groups

The *Caller ID Routing Groups* area allows you to set up call handling for up to 10 groups. Depending on the caller ID, an incoming call can be sent directly to an extension, go to an auto attendant, extension 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. Select the *Routing Groups* tab.



2. Enter a *Name* for the group.
3. Click beside the group name.
4. Select the *Action* you wish to apply:
   - *go to auto attendant* — plays the selected auto attendant.
   - *go to extension* — goes to the specified extension. Busy or unanswered calls will follow the call cascade for that extension.
   - *go to extension group* — goes to the specified extension group. The call will follow the call cascade for that extension group.
   - *go to voicemail* — accesses the selected voice mailbox.
   - *go to announcement* — plays the selected announcement.

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.

Multilocation

If a user dials a location code without an associated extension number, the call will go through to that location. The *Call Handling* area allows you to set up what happens when a call comes in from this location. Calls can be handled differently in each mode. Calls can be sent directly to an extension, go to an auto attendant, extension group, voice mailbox, or announcement.

Call Routing Outbound

Outbound

Outgoing Access Code

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

An outgoing access 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 outgoing access code. The system selects an available line from the group. However, the user does not have to dial an outgoing access code before remote extension numbers or system speed dials. These automatically use the outgoing access code configured for the respective pages.

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 outgoing access codes. If you are using multiple service provider VoIP networks, set up an outgoing access code for each service provider.

A local extension can be restricted to a set of outgoing access codes, in order to reserve telephone lines for high-priority users, or to control access to VoIP networks. See “User Privileges” on page 55.

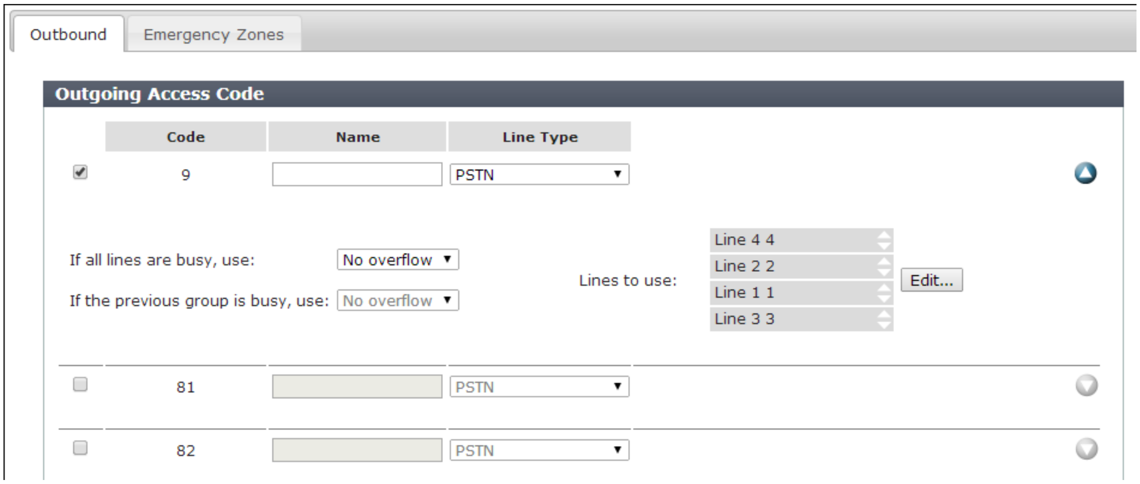
The unit uses an outgoing access code when placing a call from a local extension, to a remote extension, or with the call bridge (DISA) feature. The outgoing access codes do not affect incoming calls.

The system has the following outgoing access codes 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 outgoing access codes as required.

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

The *Outbound* tab allows you to set up the name, line type, set of telephone lines, hunting order, overflow outgoing access code and overflow notification for each outgoing access code.

Select the *Outbound* tab.



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

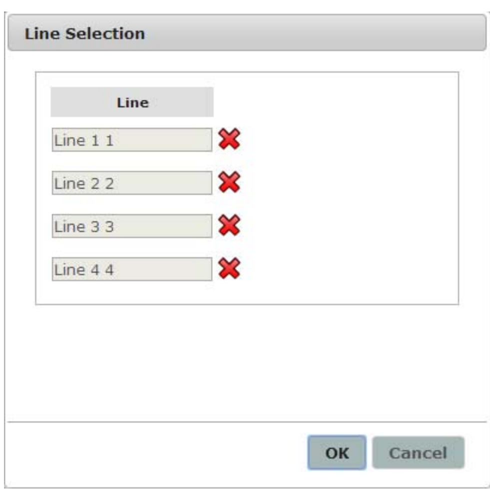
Activate outgoing access code

1. Select an outgoing access code.
2. If necessary, select the *Enable* checkbox.
3. Enter a name in the *Outgoing Access Code* box.
4. 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 outgoing access code.
   - *PRI Lines* — use the PRI service in the outgoing access code.

Outgoing access code line assignments

The *Outgoing Access Code* area allows you to set up an outgoing access code for telephone lines or VoIP lines.

1. Click beside the name.
2. If you set *Line Type* to *Phone lines*:
   a. Click *Edit*. The *Line Selection* window appears.



   b. Select the telephone lines for the outgoing access code, and click *OK*.

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

Outgoing access code busy overflow for outgoing calls

The *Outgoing Access Code Busy Overflow for Outgoing Calls* area allows you to select overflow outgoing access codes. If the user dials an outgoing access code, but there are no lines available, the system will play a warning tone while hunting for a free line in the first overflow 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 group.

If the dialed outgoing access code and the overflow outgoing access code are both busy, the user will hear the busy tone.

An outgoing access code that contains telephone lines can have overflow outgoing access codes that contain telephone lines or VoIP lines. Similarly, an outgoing access code that contains VoIP lines can have overflow outgoing access codes that contain telephone lines or VoIP lines.

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

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

1. Select the overflow outgoing access code in the *If all lines are busy in this outgoing access code* list. Choices are no overflow and the activated access groups.
2. Select the overflow outgoing access code in the *If all lines are busy in the previous outgoing access code use* list. Choices are no overflow and the activated access groups.

Emergency Zones

The system can be configured to send emergency calls over a specific line set with the correct location address for the user.

The admin must ensure that the phone numbers configured on this page have the correct address registered with the phone provider for emergency services. Contact your provider to ensure they have all the correct addresses on file.

To configure the emergency zones:

1. Enter a name for the zone, eg. `First Floor`.
2. Configure an email address of the party to be notified; typically this would be the front desk or someone who could guide emergency services to the correct location within the building.
3. Select the line type.
4. Select the phone number to be used to contact emergency services.

To add members to an emergency zone:

1. Click .
2. In the popup window, click *Edit*.
3. Search for the extensions you wish to add, and click .

4. Enter the descriptive location field. This can be used to provide more specific details within the zone if required.

Auto Attendants

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 extension groups.

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 extension group. The call then follows the extension's call cascade.
- Transfer the call to the call queue of an extension group. The call is placed on hold. The system will ring the next available local extension in the extension 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 name directory so the caller can find a user's extension number. The 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 name directory" on page 54.
- 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:

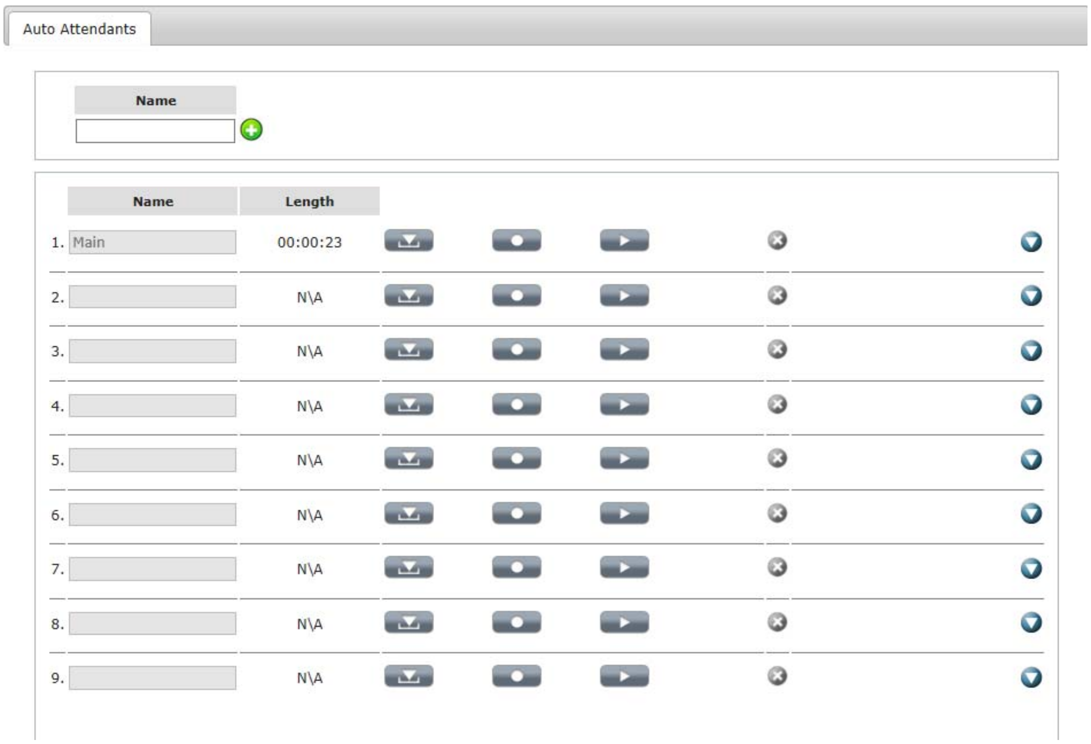
- Reach an extension by dialing an extension number.
- Access voicemail by pressing ****** then the mailbox number.
- Access call bridge/DISA by dialing a hunt group number.

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 *General Preferences* > *Timers/Prompts*. The *Auto Attendants* window will appear. Select the desired time from the pull-down menu *Single digit fall through time*.

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

Add an Auto Attendant

1. Select the *Auto Attendants* tab.



2. Enter the *Name*. The name will identify the auto attendant elsewhere in the web-based interface.
3. Click .

Actions during auto attendant playback

The *Action* 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. Click .
2. Select the language in the *Language* list. If the caller selects this option, they will hear all subsequent prompts in the selected language.
3. Click the row beside the name and 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 extension group* — Transfers the call to the selected extension group.
    - *go to announcement* — Plays the selected announcement.
    - *go to auto attendant* — Routes the call to the selected auto attendant.
    - *queue at extension group* — Transfers the call to the call queue of the selected extension group.
    - *name directory* — Accesses the name directory. See “Setting up the name directory” on page 54.
    - *perform no action* — The option is unused.

4. Select the *Resource* by clicking . Depending on the action selected in Step 3, resources are voice mailboxes, extensions, announcements or auto attendants.
5. 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*
6. Select the *Resource* by clicking .
7. 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 "Timers/Prompts" on page 67.

Action performed after auto attendant playback

The *If no selection is made in* field 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 *Immediately* to *30 seconds*.
2. Select the action. Choices include:
   - *go to voicemail*
   - *go to local extension*
   - *go to remote extension*
   - *go to extension group*
   - *go to announcement*
   - *go to auto attendant*
   - *queue at extension group*
   - *name directory* — See "Setting up the name directory" on page 54.
   - *hang up*
3. Enter the resources.

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 Web interface or from a local extension or remote phone.

1. Open the *Auto Attendants* tab. Select the auto attendant that you wish to record and click .
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. Dial *4 [auto attendant number]* # to record the message. For example, dial *41#* to record the message for auto attendant 1.
4. Press # when you have completed saying the message.
5. Dial *5 [auto attendant number]* # to listen to the message. For example, dial *51#* to listen to the message for auto attendant 1.
6. Repeat Steps 4 to 6 to re-record the message, or hang up to keep the message.

Listening to the recorded message

1. Select the auto attendant that you wish to listen to and click .
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 and click beside .
2. Click .
3. Browse to select the .wav file. Click to load the .wav file into the system.

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.

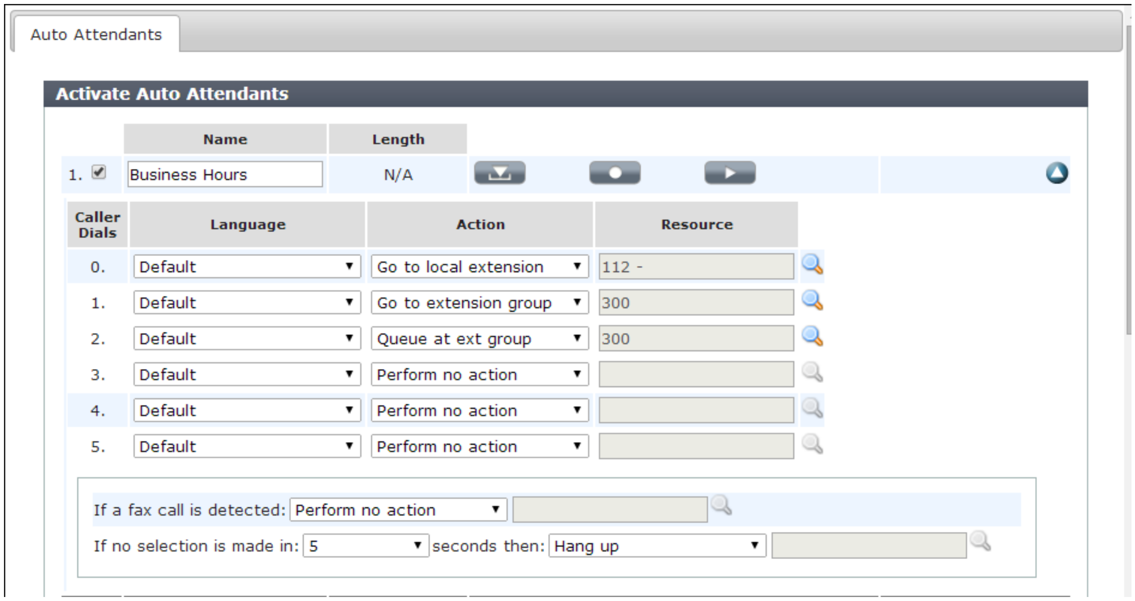
1. Pick up a local extension, or dial into the system from a remote phone. If you pick up a local extension, you will hear the dial tone. If you dial in from a remote phone, the auto attendant will answer.
2. Enter command mode by either pressing # on an analog extension phone, or **55#* on a proprietary IP phone (note: other brands may use **55 Send* or **55 Dial*).
3. 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 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: Depending on the region, callers can press either *0* or *9*.

Setting up the name directory

The 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 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.

The directory can search by last name only, first name only or first and last name. Under the *General Preferences* menu, select the *Timers/Prompts* tab and select *Dial by name search* to specify the search method.

Have each user set up their voice mailbox by dialing **# and then following the prompts. They will record their names for the 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.

You can delete the names recorded for the name directory by resetting the voice mailbox. See "Reset mailboxes" on page 7.

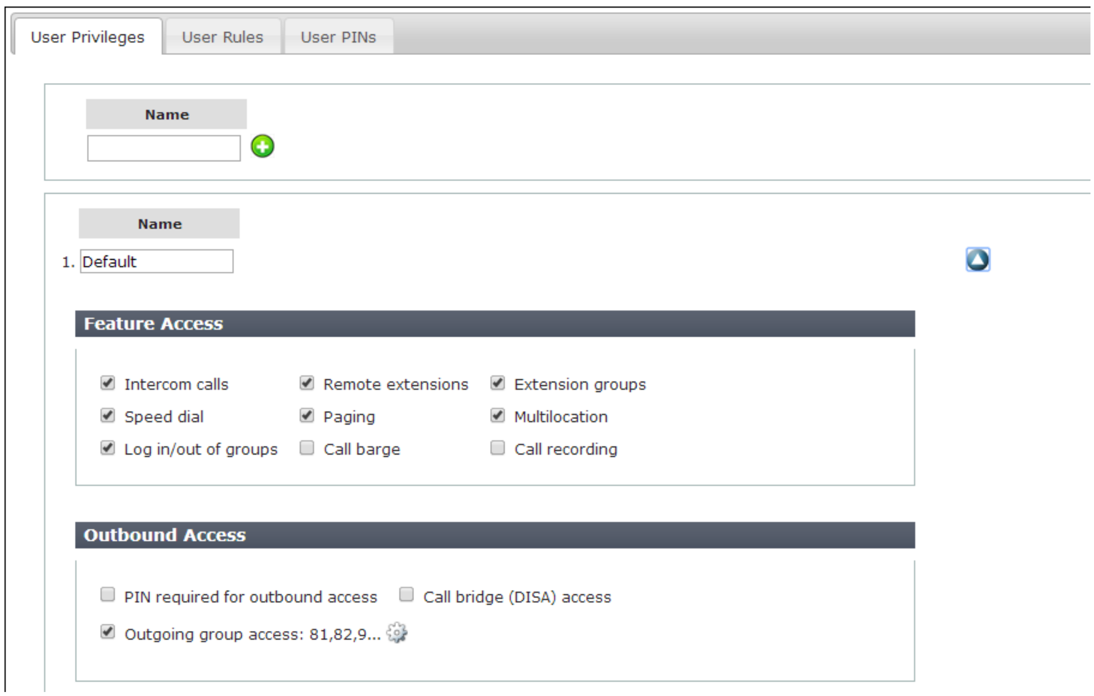
User Privileges

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.

To add a user privilege

1. Enter a name.
2. Click .



Feature Access

Check the features that you wish this profile to have access to. Click beside the name to see the following options:

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.

Extension groups — Allows users to call or transfer calls to extension groups.

Speed dial — Allows users to dial system speed dials

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

Multilocation— Allows users to call or transfer calls to other locations.

Log in/out of groups — Allows users to join or leave extension groups.

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

Call recording — allows a user to record their phone calls.

Outbound Access

The *Outbound Access* area controls the access codes an extension can use. If a user dials an access code without access, a PIN code will be required to gain access.

To assign outgoing access

1. Click beside the name of the user privilege.
2. Check the *Outgoing group access* checkbox in the *Outbound Access* area.

Blocking outbound access

If you wish all users in a profile to use a PIN code in order to use an outgoing access code, 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.

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.

1. Dial into a telephone line.
2. When the auto attendant answers, select an access code — *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 a local or remote extension or an extension 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.

User Rules

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 “Admin” on page 10.

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 an outgoing access code, routing and blocking rules can override the selection and use a different access code 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 outgoing access code 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 an outgoing access code 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, *, and -.

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.

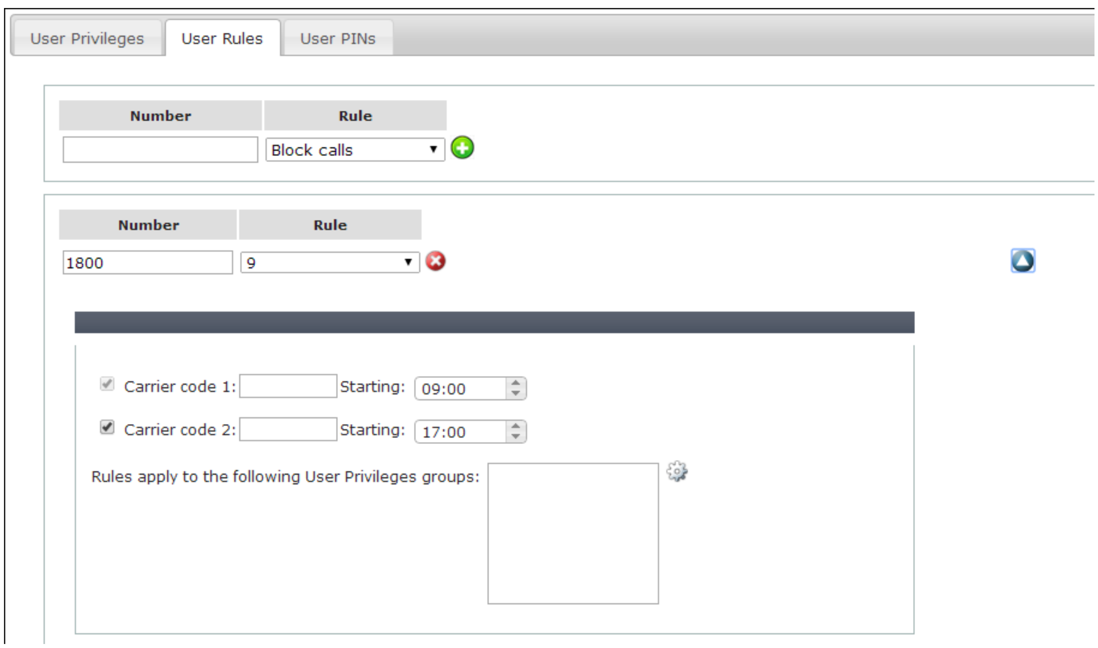
1. If the route requires one carrier code for the whole day:
   a. Enter the carrier code in *Carrier code 1*.
   b. Clear the *Carrier code 2* checkbox.
2. If the route requires two carrier codes, one for each part of the day:
   a. Enter the first carrier code in *Carrier code 1*.
   b. Enter starting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

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 user rules

1. Click the *User Privileges* page. Select the *User Rules* tab.



2. Enter the leading digits in the *Number* box. Omit the outgoing access code from the leading digits. The system will only block or redirect calls where there is a matching leading digit match.
3. Select the *Rule*. Choices are:
   - *Access Code (n)* — Routes the call to the access lines within this access code.
   - *Block calls* — Blocks the call.
4. Click .
5. To assign the rule to user privileges, click and select applicable user privilege profiles.

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.

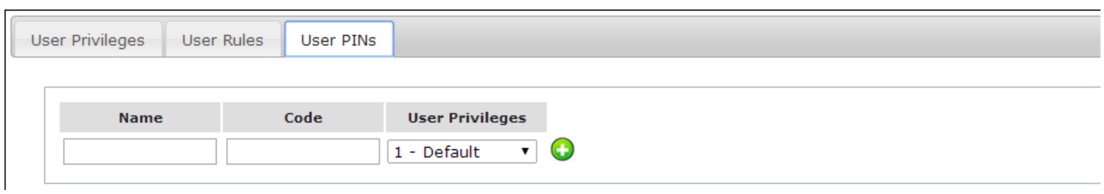
User PINs

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 “Admin” on page 10.

Adding a PIN

1. Enter a *Name.* This is typically the name of the user, or the name of the office where the extension is located.



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.

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

Removing a PIN

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

Speed Dials

Speed dials

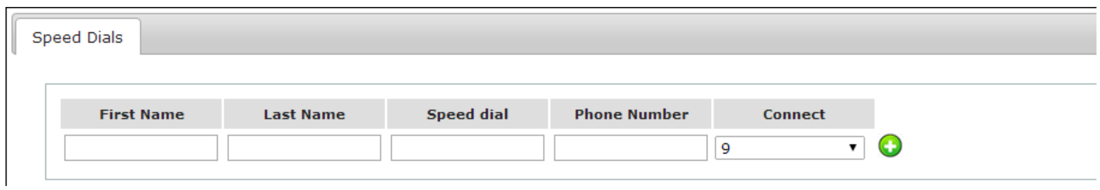
System speed dials allow the user to quickly dial an outside phone number from any local extension by dialing a speed dial number. Because the speed dial numbers are maintained within the system, they don’t need to be programmed into each individual phone. If a new client is added or an existing client’s 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 Settings” on page 13 for more information.

The *Speed Dials* tab allows you to define up to 100 speed dial numbers. Each speed dial number includes the first name, last name, phone number and outgoing access code. 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 *Speed Dials* tab.



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

2. Enter a *First Name* and *Last Name*.

3. Assign a *Speed Dial* number.

4. Enter the *Phone Number*. Valid characters are *0–9*, **-**. The phone number can be up to 19 characters long.

If the routing rules block the phone number, the phone number will not be blocked if a user dials the speed dial number.

5. Select the outgoing access codes.

If the routing rules or user privileges block the outgoing access codes, the outgoing access codes will not be blocked if a user dials the speed dial number.

6. Click .

Import system speed dial list

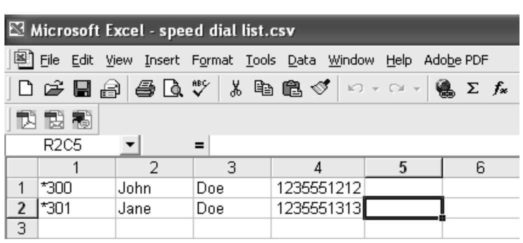
You can import a speed dial list created with a text editor or a spreadsheet program. Each entry in the file requires the speed dial number, first name, last name, and phone number. If you’re using a text editor, the values must be separated by commas. The file must be saved as a .csv file.

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, entries 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.



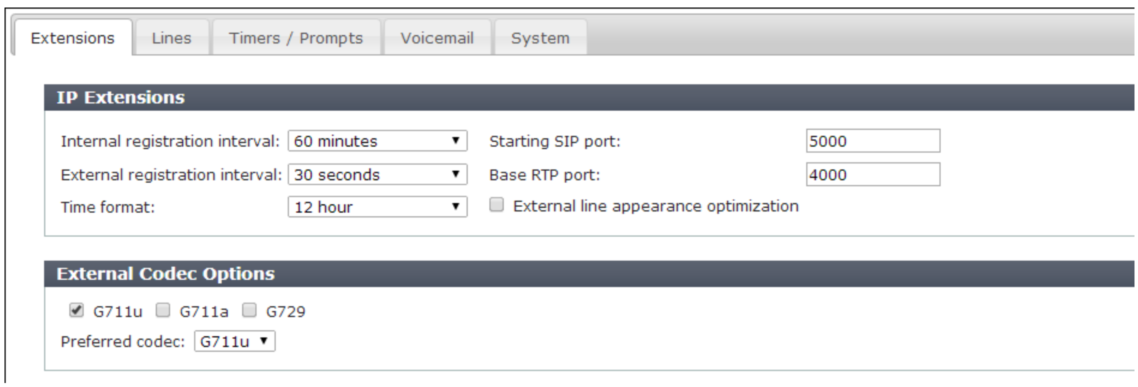
General Preferences

Extensions

IP Extensions

The *IP Extensions* window allows you to set the amount of time between registration messages from IP extensions. You can also change the Starting SIP and Base RTP.

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.

Time format — Choose the time format to be displayed on your phones, either *12 Hour* (default) or *24 Hour*.

Starting SIP port — The default Starting SIP port is 5000. If required, enter a different port ranging from 1024 to 65535.

Base RTP port — The default Base RTP port is 4000. If required, enter a different port number ranging from 1024 to 65535.

External line appearance optimization — Enabling this feature allows the system to send fewer messages for notifications to external IP extensions by using larger IP packets.

External Codec Options

Control the audio quality of external IP extensions. For more information on codecs, see “Setting codec options” on page 18.

Analog Extensions

The Extensions > *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.

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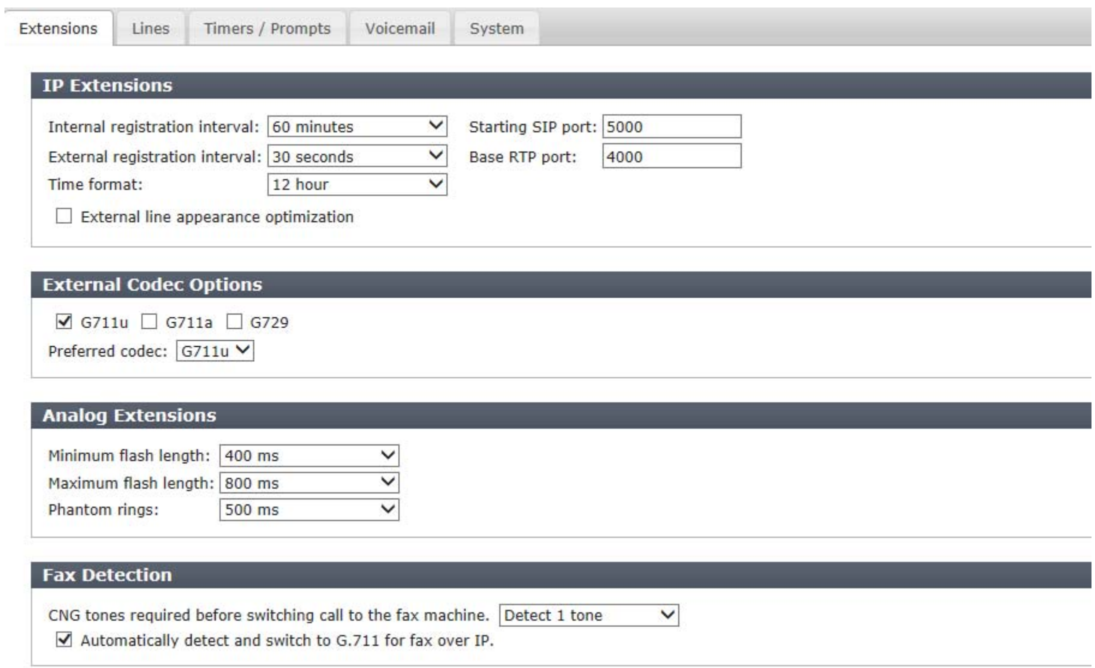
1. Select the *General Preferences* page. The *Analog Extensions* area will appear on the *Extensions* tab.



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.

Fax Detection

The *Fax Detection* area 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.

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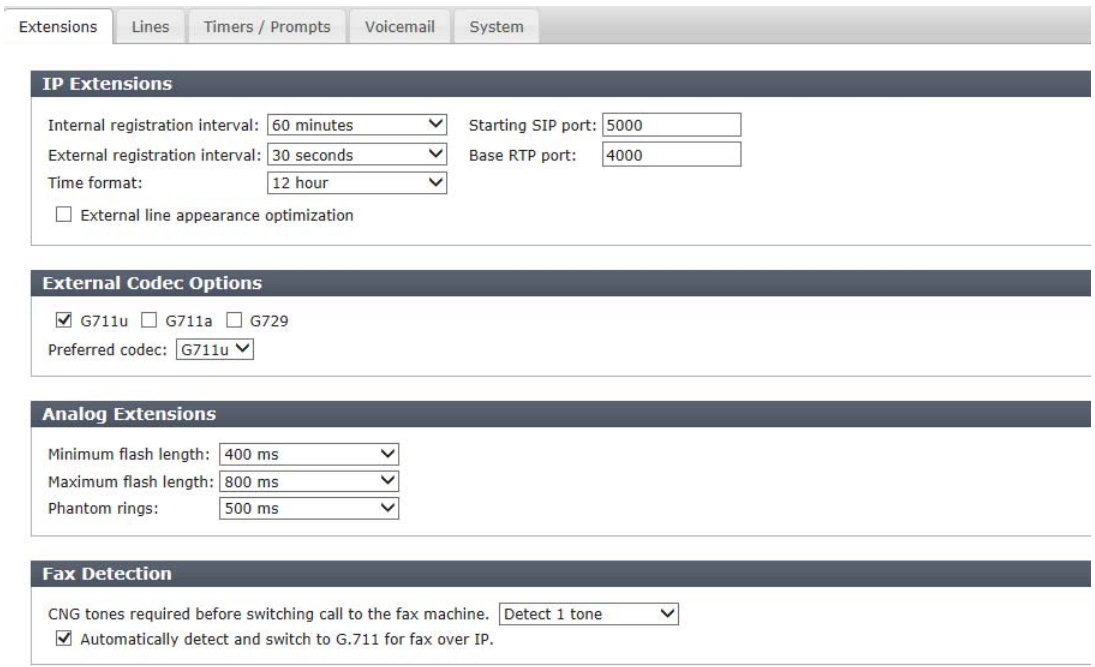
1. Select the *General Preferences* page. The *Fax Detection* area will appear on the *Extensions* tab.



2. 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.
3. Select the *Automatically detect and switch to G711 for fax over IP* checkbox to have the unit detect fax over IP and then switch to G711.

Lines

Telephone lines

The *Telephone Lines* area allows you to set telephone line timers.

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 *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. Select the *General Preferences* page followed by the *Lines* tab.
2. Select the Line disconnect timer for your provider. The default is *250 ms*.
3. Select the Call waiting timer used by your provider. The default is *450 ms*.
4. Select 3-Way calling / Centre Transfer timer used by your provider. The default *2.5 seconds*.

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.

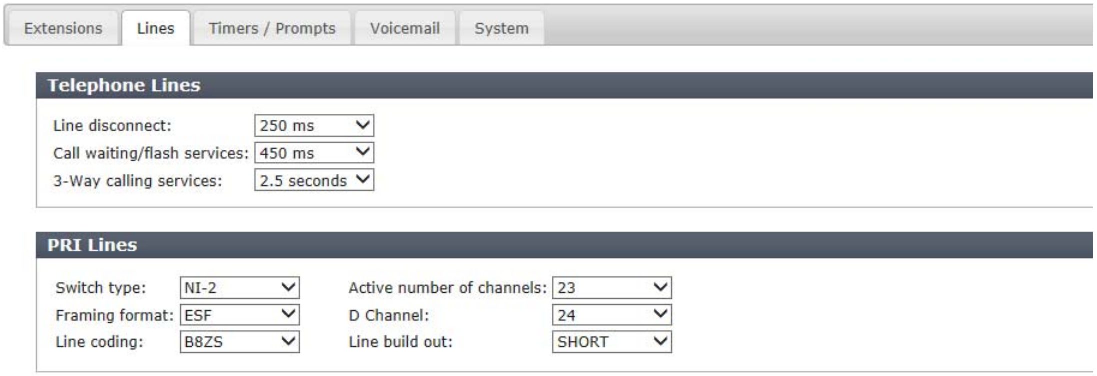
The *Call waiting and other Flash/Recall Based Services* timer sets the time required for a Flash/Recall signal to activate calling features on a telephone line. If a Flash/Recall signal is not successfully engaging telephone company features, try adjusting the timer.

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.

PRI Lines

The *PRI Lines* area allows you to set the active number of channels, the D channel and the Line Build Out level.

1. Select the *General Preferences* page followed by the *Lines* tab.

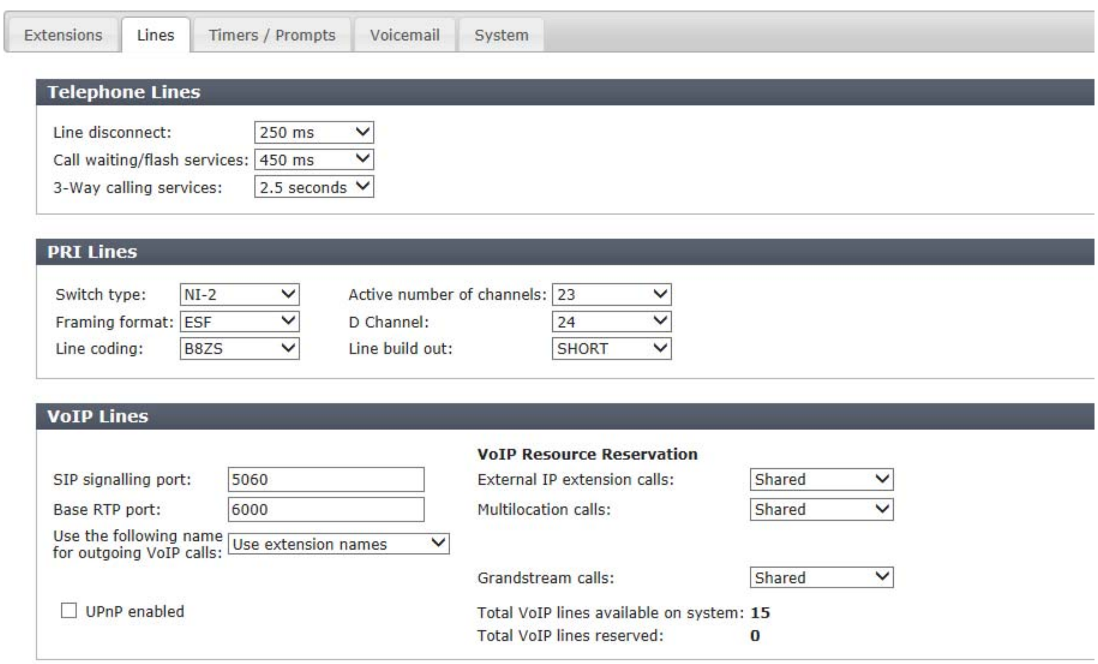


2. Select the Switch type for your provider. The default is *NI-2*.
3. Select the Framing format used by your provider. The default is *ESF (Extended Superframe)*.
4. Select the Line coding used by your provider. The default is *B8ZS* in North America and *HDB3* elsewhere.
5. Select the signalling channel used by your PRI service provider. The default D Channel is *24* for North America and *16* elsewhere.
6. Select the Line build out. 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*.

VoIP Lines

The *VoIP Lines* area allows you to set the SIP server port, and the starting RTP port.

1. Select the *General Preferences* page followed by the *Lines* tab.



2. 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.

3. The default *Base RTP port* is 6000. If required, enter a different port number ranging from 1024 to 65535.

   The system requires 8 RTP ports. The RTP ports are evenly numbered from the starting port as follows:

   - 6000 to 6014

   These are the RTP ports the system listens to for audio. If these are not opened correctly, users will not hear their callers.

Ensure your router is set up to perform port forwarding for the SIP signalling and RTP ports. See "Firewall Settings" on page 9.

The outgoing VoIP calls option allows you to set up the source for the caller ID name for outbound VoIP calls. The same setting is used for 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 *Settings* page, select *Use system name* from the *Use the following name for outgoing VoIP calls* dropdown list.
   - 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* dropdown list.

See "Caller ID settings" on page 27 to select the phone number that will appear on the other phone.

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 9.

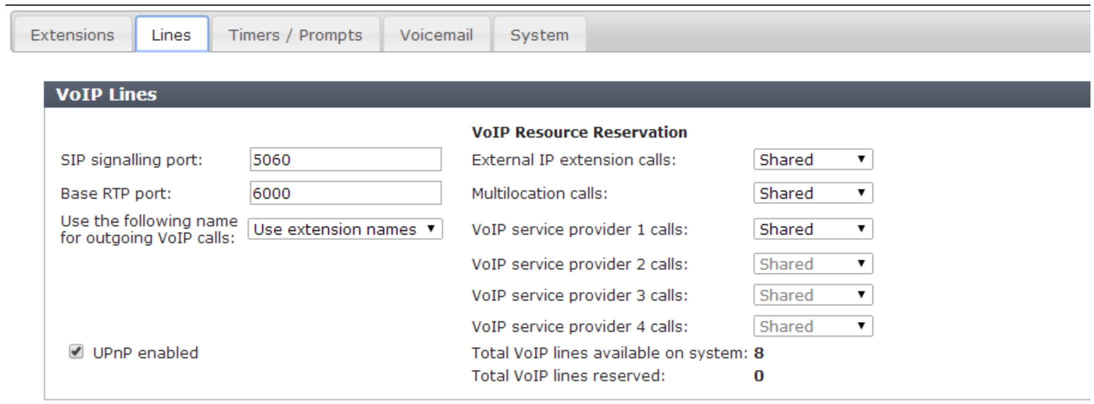
VoIP Resource Reservation

By default, all VoIP lines are available for external IP extensions, calls to other locations in a multilocation 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 the *Lines* tab.



2. Select the number of VoIP lines to reserve for *External IP extension calls* using its pulldown menu.
3. Select the number of VoIP lines to reserve for *Multilocation calls* using its pulldown menu.
4. Select the number of VoIP lines to reserve for VoIP service provider calls for each provider using its pulldown menu.

Multilocation Codec Options

Control the audio quality for your multiple locations. For more information on codecs, see “Setting codec options” on page 18.

Overflow Tone Notification

The *Overflow Tone Notification* area allows you to disable the warning tone that sounds when an overflow outgoing access code is being used. The warning tone sounds by default.

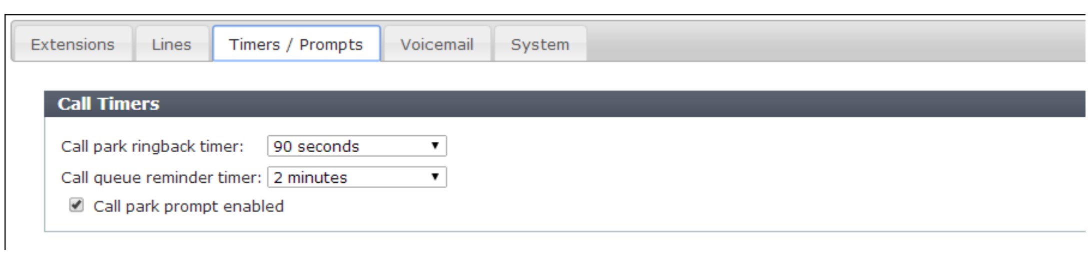
1. To disable the warning tone, clear the Play notification tone when overflowing to another outgoing access code checkbox.

Timers/Prompts

Call Timers

The *Call Timers* area 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 *Call queue reminder timer* expires.

1. Select the *General Preferences* page followed by the *Timer/Prompts* tab.



2. Set the *Call park ringback timer* to how often the phone should ring when a caller is parked.

3. Set the *Call queue reminder timer* to how often a queued caller should hear a prompt.

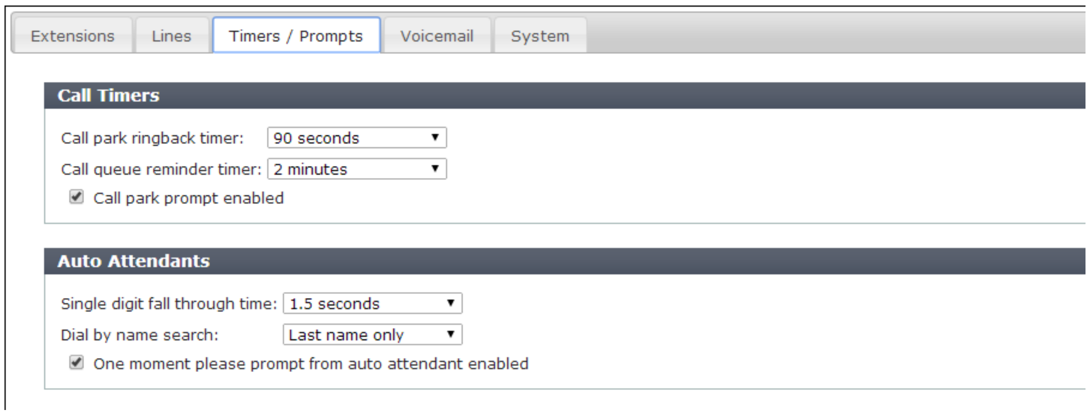
Auto Attendants

The *Auto Attendants* area allows you to set auto attendant timers.

Do not adjust these settings unless directed by your local dealer or TAC. The default settings should provide correct operation.

---

1. Select the *General Preferences* page followed by the *Timer/Prompts* tab.

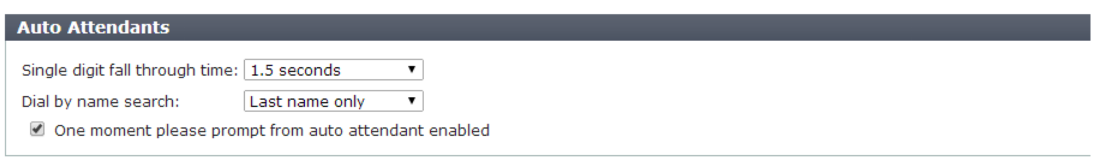


2. The *Single digit fall through time* sets a time-out period at the auto attendant for detection of additional digits after the first digit — e.g. an extension number. Once the wait time elapses, the auto attendant processes the received digits. Selecting *immediately* limits each entry to a single 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.
3. Select the *One moment please prompt from auto attendant enabled* checkbox to enable the "one moment please" prompt when transferring a call.

The *Dial by name search* option allows you to specify whether the name directory search will provide results based on first, last, or both names.

Callers can connect to an extension using their telephone dialpad to spell the name of the party they wish to speak with.

1. Select the *General Preferences* page followed by the *Timer/Prompts* tab.



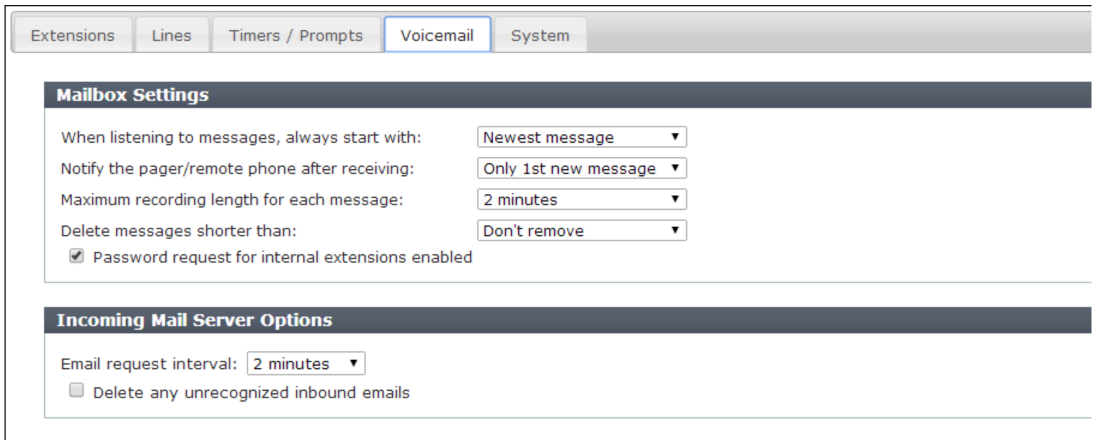
2. From within the *Auto Attendants* area, select the desired search option. Choices include: *Last name only, First name only* and *First and last name.*

Voicemail

Mailbox Settings

The *Mailbox Settings* area allows you to set the voice mailbox preferences. Any changes entered in this area affect all mailboxes.

1. Select the *General Preferences* page followed by the *Voicemail* tab.



2. Set the *When listening to messages, always start with* list to which message should be played first. Choices are *Oldest message* or *Newest message*.
3. 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.*
4. Set the *Maximum recording length for each message* list to how long a message can be. Choices range from *1 minute* to *8 minutes*.
5. Set the *Delete messages shorter than* list. The system will automatically remove messages that are shorter than the selected length. Choices range from *Don't remove* or *1 second* to *5 seconds* in half second increments.
6. Local extensions can access voicemail without a password if you deselect the *Password request for internal extensions enabled* checkbox. Passwords will still be required when checking voicemail remotely.

Incoming Mail Server Options

The *Incoming Mail Server Options* area allows you to specify how often the unit checks the incoming server for deletions and saves of voicemail messages.

1. Select the *Email request interval*.
2. To delete, select the *Delete any unrecognized inbound emails* checkbox.

System

Speed Dials

The *Speed Dials* area allows you to enable the substitution of the caller ID name with the name from the *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 *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 *Speed Dials* page. These names will appear on the extension instead of the caller ID name.

1. To enable caller ID name tagging of the entries within the *Speed Dials* area, choose one of the options from the *Replace inbound caller ID speed dial name* list.
2. Select the order for the names. Choices are *First name, last name*, or *Last name, first name*.

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 extension group* — Connects to the selected extension group.
   - *go to announcement* — Plays the selected announcement.
   - *go to auto attendant* — Connects to the selected auto attendant.
   - *queue at extension group* — Connects to the call queue of the selected extension group.
   - *name directory* — Accesses the name directory.
2. Select the resource. Depending on the action, resources are voice mailboxes, extensions, announcements or auto attendants.

Conferencing

Conferencing

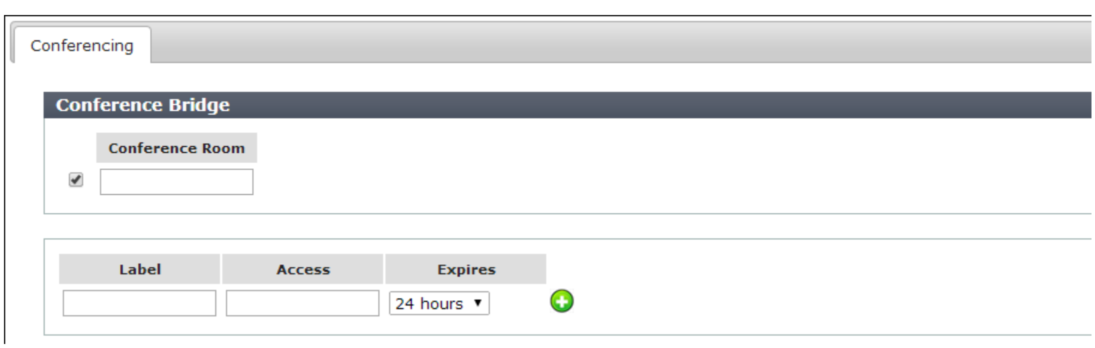
Conference Bridge

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.

Create a conference bridge number

1. Go to the *Conferencing* tab.



2. Check the *Conference Room* checkbox.

3. Assign a conference room number. This is a unique 3-, 4- or 5-digit number, depending on your dial plan.

Creating moderators

1. Assign *Name/Label*.
2. Assign *Code*.
3. Assign a lifespan to participant access codes in the *Expiry* list. 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.

Logs and Reports

Log Settings

The *Log Settings* page provides diagnostic logging tools to capture call data and phone system events. These tools are intended for the use of technical support personnel.

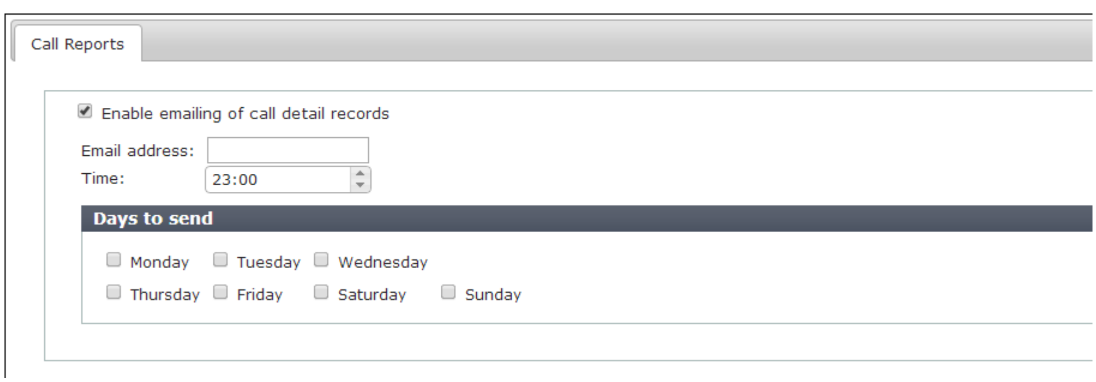
1. Select *Logs & Reports* > *Log Settings* page.
2. Select the appropriate logging level.
3. Select the events / messages to log in the system.
4. Click *Download Logs* to save the logs from the system.

Call Reports

The *Email Log File* area allows you to have the system regularly e-mail a log file containing the call detail records.

Note that if you want the system to e-mail a file, you must configure the outgoing mail server (SMTP). See "Email Server" on page 11.

1. Select the *Call Reports* tab.



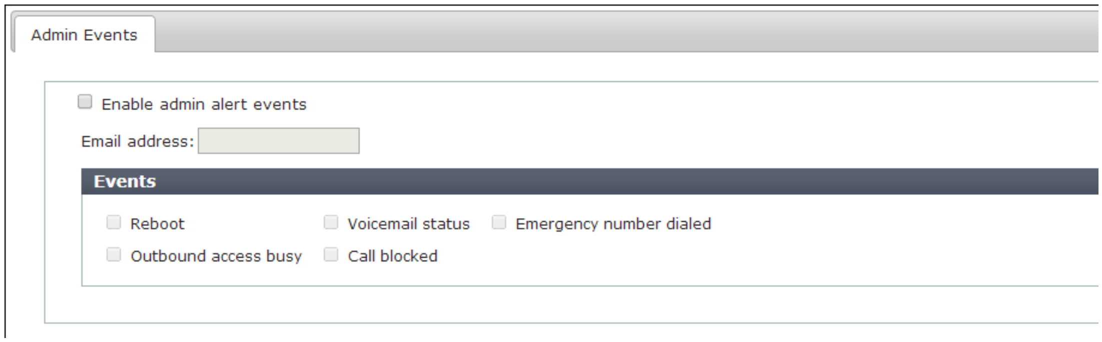
2. To have the system e-mail the log file, select the *Enable emailing of call detail records* checkbox.
3. Enter the *Email address*.

4. Select the *Time* to send.
5. Select the *Days to send*.

Alerts

Admin Events

1. Select the *Admin Events* tab.



2. Select the *Enable admin alert events* checkbox to activate notifications.
3. Enter the administrator's email.

Admin Events sends notifications to the administrator when certain events occur.

The notification options are:

- *Reboot*
- *Voicemail status* (system nears maximum capacity)
- *Emergency number dialed* (emergency services was dialed)
- *Outbound access busy* (if all lines are busy in an outbound group)
- *Call blocked* (attempted call to a blocked number)

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 from the web browser.

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 Server" on page 11.

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 is one method for retrieving CDRs:

- Browser

Web interface

Status > Call Detail Record

Click on the *Call Detail Record* page. Within the *CDR* tab, select the time frame you wish to view. Click *Export* to download the records to your PC.

Analyzing the data

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: 1001 — PSTN line, line 1
    2003 — PRI, channel 3
    3010 — VoIP, 10th channel
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 comma 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	

Troubleshooting and Support

Troubleshooting

- Auto attendant
- Music on hold
- Call routing
- Answering and fax machines
- Local extensions
- VoIP

Some problems might be due to physical connections such as loose cables.

Auto attendant

The auto attendant does not play when calls come in

1. Make sure you have an auto attendant message recorded.
2. Ensure the system is running the correct mode. Open the Web Management. Select *Scheduling*. Check the *Current mode* in the *Modes* area.
3. Ensure the telephone line has the correct auto attendant setup. Select *Call Routing* > *Inbound* > *Analog 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 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 Web Management. Expand *Auto Attendants*. In the *Actions During Auto Attendant Playback* area, check if the actions and resources point to the correct extensions.

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 local extension, they can transfer you to your voicemail box by transferring your call by pressing **, followed by your extension number.

Music on hold

Callers hear only silence when put on hold at an extension

1. 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 *Trunks* > *Analog Lines*.

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

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 *Extensions* 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. Make sure the telephone cord you are using between the phone and the system is working properly.
2. 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 *Extensions* page in the Web Management. 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.

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 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.

Support

If you are having problems with the configuration or operation of your system:

1. Contact your authorized reseller or point of purchase.
2. 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.	
* and the speed dial number	Dials the speed dial number.
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.
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*.
**56*	Reads back system IP

Keys	Function performed
*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.
On an IP extension, press *Dial, Send* or # after your entry for the following options:	
*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 extension group number	Page a phone at an extension or the phones of an extension group.
*88 and account code	Attach account code to call detail record (CDR) for last call.
*9	Call pick-up of any ringing line.

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).
Extension group	Rings all the extensions in the group (e.g. 301). There are 10 extension groups for the entire system.
9 (*0* in some regions) and *81–88*	Access code for call bridge and is protected with the PIN 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.

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.

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