Asterisk Appliance™ 50
(AA50)

Administrator Manual

601-00005 Rev. C

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Digium, Inc. has made every effort to ensure that the instructions contained in this document are adequate and error free. The manufacturer will, if necessary, explain issues which may not be covered by this documentation. The manufacturer's liability for any errors in the document is limited to the correction of errors and the aforementioned advisory services.

This document has been prepared for use by professional and properly trained personnel, and the customer assumes full responsibility when using it.

Safety Certification and Agency Approvals

Safety:

US/CSA 60950
IEC 60950
EN 60950
AS/NZ 60950

Other:

CE Mark (European Union)
2002/95/EC Restrictions on Hazardous Substances (RoHS), 2005/747/EC lead free exemption (Annex C)

Telecom:

FCC Part 68, TIA-968
TBR-21 1998
Industry Canada IC-CS-03
AS-ACIF S002-2005
AS-ACIF S003-2005

EMC:

FCC Part 15 Class A
EN55022/CISPR22 Class A
EN55025
IEC 61000
CNS13438
VCCI V-32005.04

Federal Communications Commission Part 68 (USA)

This equipment complies with Part 68 of the FCC rules and the requirements adopted by the ACTA. On the back of the Asterisk Appliance 50 enclosure is a label that contains, among other information, a product identifier in the format US:AAAEQ##TXXXX. If requested, this number must be provided to the telephone company.

A plug and jack used to connect this equipment to the premises wiring and telephone network must comply with the applicable FCC Part 68 rules and requirements adopted by the ACTA.

If the Asterisk Appliance 50 causes harm to the telephone network, the telephone company may notify you in advance that temporary discontinuance of service may be required. But if advance notice is not practical, the telephone company will notify you as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.

If you experience problems with the Asterisk Appliance 50, contact Digium, Inc. (+1.256.428.6161) for repair and/or warranty information. If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.

FCC Part 15

This device complies with part 15 of FCC rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) This device must accept any interference received, including interference that may cause undesired operation.

Introduction to Asterisk Appliance 50 Documentation

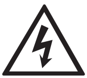
This manual contains product information for the Asterisk Appliance 50. Be sure to refer to any supplementary documents or release notes that were shipped with your equipment. The manual is organized in the following manner:

Chapter/ Appendix	Title	Description
1	Overview	Identifies the features of your unit.
2	Unit Installation	Provides instructions for installing the unit.
3	Asterisk Configuration	Provides instructions on how to configure the Embedded Asterisk Business Edition through the use of the AsteriskGUI.
4	Troubleshooting	Explains resolutions to common problems and frequently asked questions pertaining to the unit.
A	Pin Assignments	Lists the connectors and pin assignments.
B	Specifications	Details unit specifications.
C	License Agreement	Digium End-User Purchase and License Agreement
D	Glossary and Acronyms	Defines terms related to this product.

Symbol Definitions

Caution statements indicate a condition where damage to the unit or its configuration could occur if operational procedures are not followed. To reduce the risk of damage or injury, follow all steps or procedures as instructed.

The ESD symbol indicates electrostatic sensitive devices. Observe precautions for handling devices. Wear a properly grounded electrostatic discharge (ESD) wrist strap while handling the device.



The Electrical Hazard Symbol indicates a possibility of electrical shock when operating this unit in certain situations. To reduce the risk of damage or injury, follow all steps or procedures as instructed.

Important Safety Instructions

Servicing.
Do not attempt to service this unit unless specifically instructed to do so. Do not attempt to remove the unit from your equipment while power is present. Refer servicing to qualified service personnel.

Water and Moisture.
Do not spill liquids on this unit. Do not operate this equipment in a wet environment.

Heat.
Do not operate or store this product near heat sources such as radiators, air ducts, areas subject to direct, intense sunlight, or other products that produce heat.

Warning.
Do not place anything (including paper) on top of the Asterisk Appliance 50. To allow proper cooling, these units must not be stacked.

Caution.
To reduce the risk of fire, use only No. 26 AWG or larger telecommunication wiring for network connections.

Static Electricity.
To reduce the risk of damaging the unit or your equipment, do not attempt to open the enclosure or gain access to areas where you are not instructed to do so. Refer servicing to qualified service personnel.

Emergency 911
The Asterisk Appliance 50 is capable of forwarding arbitrary caller id strings to VoIP service providers, which in multi-office setups could simply be other Asterisk Appliance 50s. Customers of Internet Telephony Service providers to which 911 or Emergency calls are placed should ensure their provider properly forwards the customer's accessible PSTN phone number to the emergency call handling center.

Save these instructions for future reference.

TABLE OF CONTENTS

Chapter 1
Overview

The Digium® Asterisk Appliance 50 (AA50) is a stand alone PBX which runs Embedded Asterisk Business Edition™. It is suitable for the desktop, or mounting in a typical network closet or restricted access location. The Asterisk Appliance 50 is ideal for small office environments or as an extension to a central Asterisk PBX.

The Asterisk Appliance 50 can function not only as a PBX, but also as a voice mail server, IVR server, conferencing server, VoIP ATA, or VoIP gateway. It has up to eight analog ports which are configured as FXO or FXS ports depending upon the product model. Additionally, the built in four port switch and WAN port allow it to also serve as a basic router.

The AsteriskGUI™ is the interface for the Asterisk Appliance 50. It gives you the ability to configure the basic hardware and dial plan elements you need when initially setting up your system, as well as every element needed to customize your setup. You must create trunks, system users, conferencing, voice mail, etc. The AsteriskGUI supports the following browsers:

- Firefox 1.5 through 3.0
- IE 7
- Safari 3.x
- Opera 9.x

Features:

- Embedded Asterisk Business Edition™
- AsteriskGUI™
- Four port 10/100BaseT Ethernet switch with Auto-MDI/MDI-X capability for the four 10/100BaseT LAN ports and one 10/100baseT WAN port (both 802.3/802.3u)
- Up to eight analog ports supporting either FXS or FXO lines depending on product version (available product versions: S800i with VoIP only, S808i with Eight FXO, and S844i with Four FXS and Four FXO)
- SIP and IAX2 VoIP protocols
- CompactFlash interface (Type 1) suitable for standard CompactFlash cards
- Configuration reset switch
- High performance Analog Devices Incorporated (ADI) BlackFin BF537 processor
- uClinux Operating System
- Transcoding provided on the Blackfin processor
- 32ms (Hardware Revision B) or 128ms (Hardware Revision C) of analog port echo cancellation
- 8MB on board serial Flash memory
- 64MB 16 bit parallel SDRAM
- Front panel LEDs

Chapter 2
Unit Installation

This chapter provides the following information:

Figure 1: The Asterisk Appliance 50 (AA50)

Unpacking the Unit

When you unpack your unit, carefully inspect it for any damage that may have occurred during shipment. If damage is suspected, file a claim with the carrier and contact your reseller from which the unit was purchased or Digium Technical Support (+1.256.428.6161). Keep the original shipping container to use for future shipment or proof of damage during shipment.

> ***Note:*** *Only qualified service personnel should install the unit. Users should not attempt to perform this function themselves.*

Inspecting Your Shipment

The following items are included in shipment of the Asterisk Appliance 50:

- Asterisk Appliance 50 (AA50)
- Compact Flash Card
- Power Supply
- Power Cable
- Analog Cables (optional depending on model)
- CD-ROM containing manual and installation files
- Product Registration Card
- Support and Warranty Information

Identifying Communication Ports

The Asterisk Appliance 50 unit consists of up to eight RJ11 analog ports which are configured as FXO or FXS ports depending on the Asterisk Appliance 50 model. These ports provide 32ms (Hardware Revisions B) or 128ms (Hardware Revisions C) of analog port echo cancellation. The unit is rated for a total of 8 REN across all FXS ports. Each individual port is rated for up to 3 REN @ 1500ft (450m).

Four 10/100BaseT LAN ports and one 10/100BaseT WAN port provide the functionality to connect to the local network as well as allowing the Asterisk Appliance 50 to act as a router. All the Ethernet ports support auto-MDI-X.

See Figure 2 on page 22 to locate the ports and their corresponding LEDs.

Understanding the LEDs

There are 15 LEDs on the front panel of the Asterisk Appliance 50. The eight LEDs corresponding to the analog ports on the rear panel indicate the type of interface installed. The definition of each LED and its color representation is explained in Table 1.

Table 1: LED Definitions

LED	Color	Description
Power	Blue (pulsing)	On when the unit boots up, after the bootload process has completed. The LED pulses at a rate which is proportional to the processor load.
Compact Flash	Blue (flashing)	Flashes each time there is read or write activity to or from the CompactFlash card.
WAN	Off	No line is connected or the interface is inactive.
	Green (flashing)	Link is up at 100Mbps. LED flashes at 1/10 second intervals as traffic is detected.
	Orange (flashing)	Link is up at 10Mbps. LED flashes at 1/10 second intervals as traffic is detected.
LAN (4 ports)	Off	No line is connected or the interface is inactive.
	Green (flashing)	Link is up at 100Mbps. LED flashes at 1/10 second intervals as traffic is detected.
	Orange (flashing)	Link is up at 10Mbps. LED flashes at 1/10 second intervals as traffic is detected.

Table 1: LED Definitions

LED	Color	Description
Analog (8 ports)	Off	No analog port is installed in the corresponding port.
	Green (solid)	Port is configured for FXS operation and is enabled. An analog telephone may be connected to this port.
	Green (flashing)	Telephone is ringing.
	Green (slow blinking)	Telephone is in use.
	Red (solid)	Port is configured for FXO operation and is enabled. A telephone line may be connected to this port.
	Red (flashing)	Telephone line is ringing.
	Red (slow blinking)	Telephone line is in use.

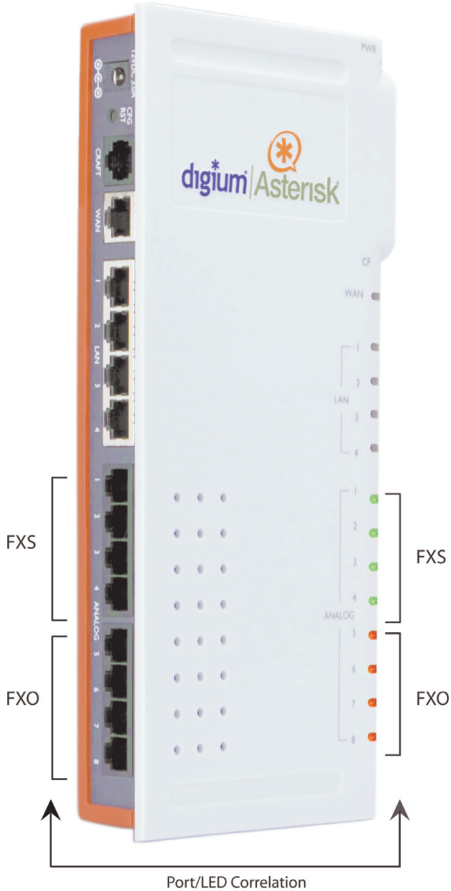


Figure 2: Example Asterisk Appliance 50 Port Identification

Using the Configuration Reset Switch

The Configuration Reset (CFG RST) switch (rear panel) will reset the current Asterisk Appliance 50 configuration to the factory defaults when pressed. The switch must be pressed and held during the boot process. This will force the unit to delete all configuration data. The administrator password will also be reset. See Figure 3 on page 24 to locate the **CFG RST** switch.

Caution.
Pressing the CFG RST switch will cause loss of all configuration settings and reset administration passwords.

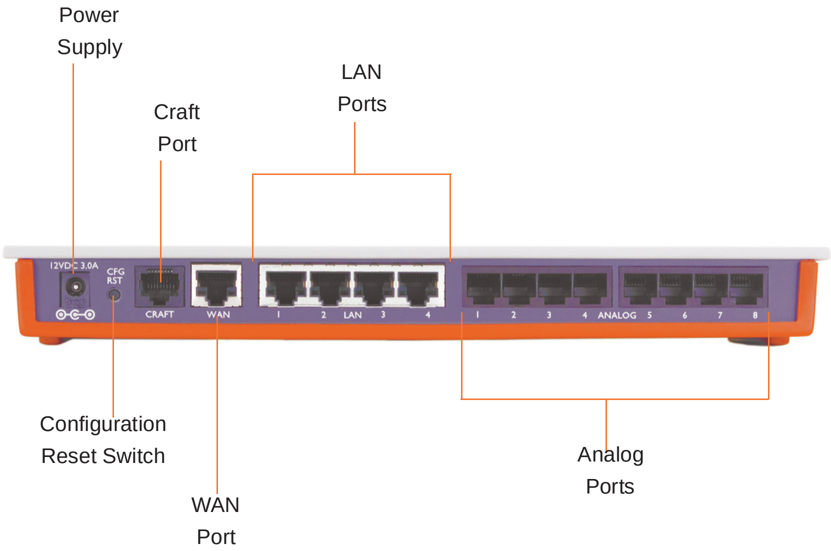


Figure 3: Asterisk Appliance 50 Back View

Installing the Asterisk Appliance 50

1. Remove the Compact Flash cover plate and insert the Compact Flash card before connecting the power supply.

Caution.
The Compact Flash is not hot swappable. The Compact Flash card should be inserted before powering on the unit. Likewise, before removing the Compact Flash card it should be unmounted (using the ***unmount*** *command) and the Asterisk Appliance 50 should be powered off.*

2. Connect one end of an Ethernet cable to an Asterisk Appliance 50 LAN port, and one end to an Ethernet connection on a computer configured to obtain an IP address automatically (DHCP). This step will connect your Asterisk Appliance 50 to your computer so that you may access the Asterisk Appliance 50 GUI from your computer.
3. Connect the provided power cable to the power supply. You can then connect the power supply to the Asterisk Appliance 50's DC power connector. The Asterisk Appliance 50 will immediately power on once connected to a power source.
4. Using an Asterisk Appliance 50 supported web browser, open a browser window and enter the IP address for the Asterisk Appliance 50. The default LAN IP address is 192.168.69.1. The default username is **admin**, and the default password is **password.**

   Note: The first time you log on you will be prompted to change your password from the default. You will then be prompted to log on with the new password. Once the log on process is complete the AsteriskGUI home page will be displayed.
5. You may find it preferable to enable the Asterisk Appliance 50 GUI on the WAN interface for ease of use. Once you have logged on to the Asterisk Appliance 50, click on the **Networking** menu, and then the **WAN** tab.
6. Select the **Enable GUI on WAN interface** checkbox.
7. Click **Save**, and then click **Apply Changes**. Your changes will be applied and Asterisk will reload.
8. Attach the ethernet cable connected to the Asterisk Appliance 50's LAN port to the WAN port. Connect the other end of the cable to the appropriate internet connection (will vary depending on your setup). This will connect the Asterisk Appliance 50 to the internet.

9. Connect telephones to the analog ports that are configured as FXS ports and connect phone lines to the analog ports that are configured as FXO ports.

10. Using an Asterisk Appliance 50 supported web browser, open a browser window and enter the IP address for the Asterisk Appliance 50. The default username is **admin**, and the password is will be the password you chose after first logging into the Asterisk Appliance 50.

11. You are now ready to configure your Asterisk Appliance 50 via the GUI.

Caution.
This unit must be connected to the Telecommunications Network in your country using an approved line cord, e.g.: for Australia use only line cords complying with ACA Technical Standard TS008.

Mounting the Asterisk Appliance 50

Figure 4 below illustrates the proper mounting installation options:

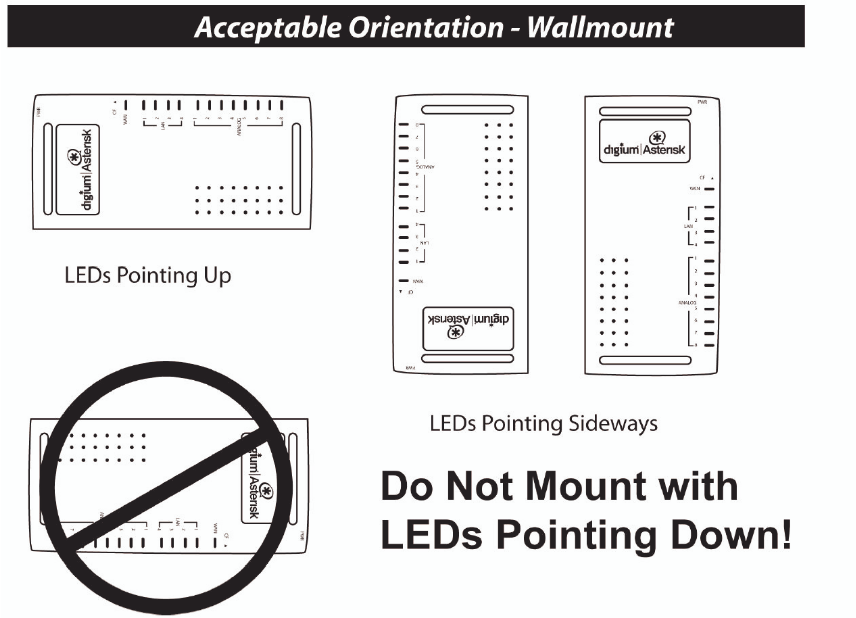


Figure 4: Mounting Instructions

Warning.
Do not place anything (including paper) on top of the Asterisk Appliance 50. To allow proper cooling, these units must not be stacked.

Table 2: Wall Mounting

Step	Instructions for Wall Mounting
1	Select the area to mount the Asterisk Appliance 50 unit (refer to *Figure 4* on page 27). The unit should be mounted at or below eye level to properly view the LEDs.
2	Install two #8 PAN headscrews (1 1/2-inch or longer) into the desired location on the wall. They should be placed approximately 7 1/2-inches, or 19cm, apart horizontally or vertically, which is the distance between the two keyed insets on the back of the Asterisk Appliance 50. Make sure that the two screws are in alignment and level.
3	Leave approximately 1/4-inch of the screw protruding from the wall to allow the head of the screws to slide into the keyed insets, mounting the unit to the wall.

Warning
The Asterisk Appliance 50 should not be mounted with the LEDs pointing downward. Mounting the Asterisk Appliance 50 with the LEDs pointing downward may cause a disruption in air circulation, which could cause the Asterisk Appliance 50 to overheat. Mounting the Asterisk Appliance 50 this way can also expose the LAN, WAN, and analog ports to potential damage.

Chapter 3
Telephone System Configuration

This chapter provides information on how to initially set up your telephone system via the AsteriskGUI™. The following topics are covered:

The Asterisk Appliance 50 comes with Embedded Asterisk Business Edition™. The software includes the AsteriskGUI, a web based configuration interface. The AsteriskGUI gives you the ability to set up your telephone system without the need to use command line configuration. After connecting to the Asterisk Appliance 50, the primary menu is displayed, giving you the ability to configure your system, as well as add features to your call system as your needs change.

Log On to the Asterisk Appliance 50

Your Asterisk Appliance 50 should already be connected to an internet or network connection, as described in **Installing the Asterisk Appliance 50** on page 24. In the address field of an Asterisk Appliance 50 supported web browser, enter the IP address assigned to your Asterisk Appliance 50. The default LAN IP address is 192.168.69.1.

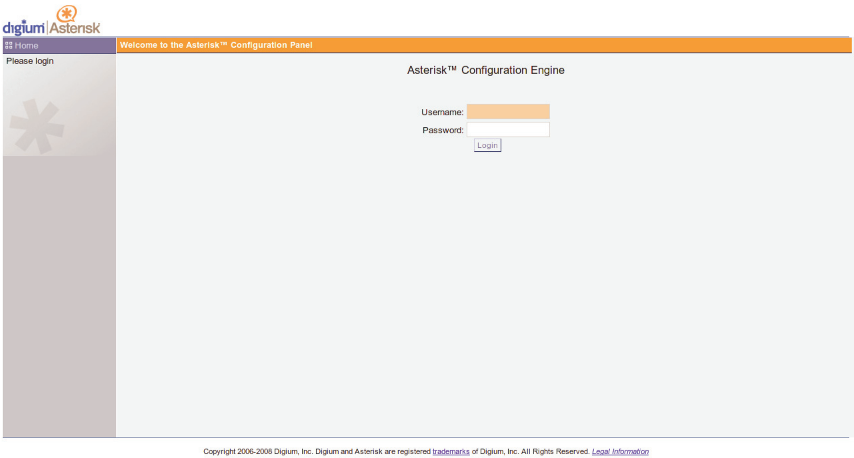


Figure 5: GUI Login

To log on to the system enter the following credentials:

- **Username**: admin
- **Password**: <password>

The first time you log on you will be prompted to change your password from the default. You should have already chosen a new password during the installation process. Once the log on process is complete the AsteriskGUI home page will be displayed.

The Asterisk Appliance 50 Interface

The AsteriskGUI gives you the ability to configure the basic hardware and dial plan elements you need when initially setting up your system. You must create trunks, system users, conferencing, voice mail, etc. After logging into the AsteriskGUI, you're presented with a variety of options on the left side of the page.

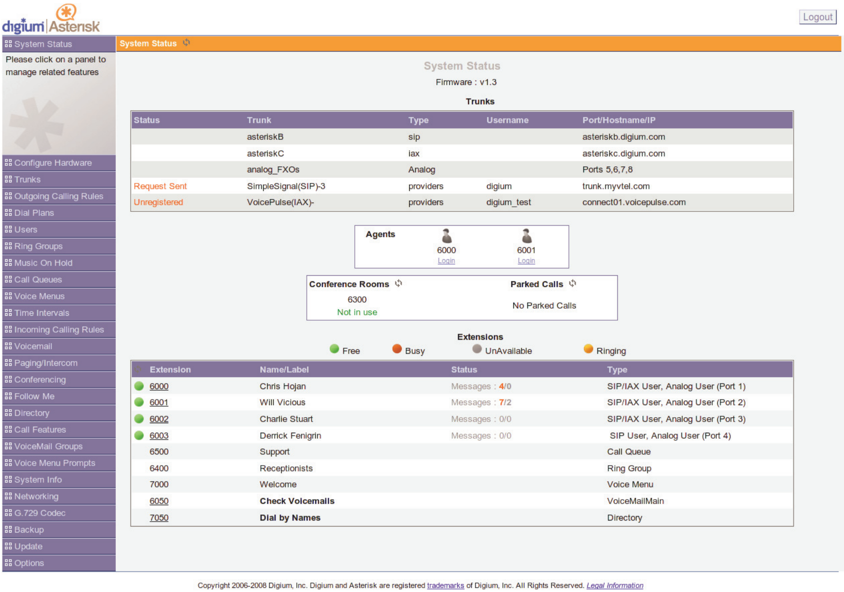


Figure 6: System Status Page

The AsteriskGUI supports the following browsers:

- Firefox 1.5 through 3.0
- IE 7
- Safari 3.x
- Opera 9.x

Every page of the GUI has two columns. The left column identifies all the elements for which you can program the Asterisk Appliance 50. The elements listed begin with **System Status**, which is the first page you see upon logon, and proceed down to **Options**. Clicking any of the tabs on the left of the page opens the corresponding page in the right column. Many pages have additional information. Click on the information symbol, a blue “i” enclosed in a circle, to get more information about a field or page.

The **System Status** page is the default page. This page shows you the current version of firmware you are using, the status of any trunk lines you have configured, the realtime status and additional details of all user extensions, including the new and old voicemail message count for each user extension (*e.g.* Messages: new/old), and the realtime status of all agents, conference rooms, and parked calls. You can click on most extension definitions to get more information. In addition, the **System Status** page gives you the ability to log in, log out, pause, and unpause an agent that is associated with one or more call queues.

> **Note:** A user extension will have the status of “Unavailable” when the VoIP account associated with it is not registered to the Asterisk Appliance 50. The status will not change to “Unavailable” when a user extension has both an analog port and a VoIP account associated with it.

In the upper right corner of each page you will see the **Apply Changes** and **Logout** buttons. Click **Apply Changes** to save and activate any changes you have made on a page so that you can utilize the changes. Click **Logout** on any page to exit the Asterisk Appliance 50 GUI.

Analog Hardware Configuration

You must configure your analog hardware according to the needs of your system as part of your initial Asterisk Appliance 50 configuration. The **Configure Hardware** page gives you the ability to configure both your FXS and FXO ports, as well as your Tone Region, operation mode, message waiting indicator mode (MWI), etc. The number of FXS and FXO ports available for configuration will depend on the Asterisk Appliance 50 model you purchased. Click the **Configure Hardware** tab to configure your analog hardware.

> **Note:** The **Configure Hardware** tab will not be available if you ordered a VoIP only model.

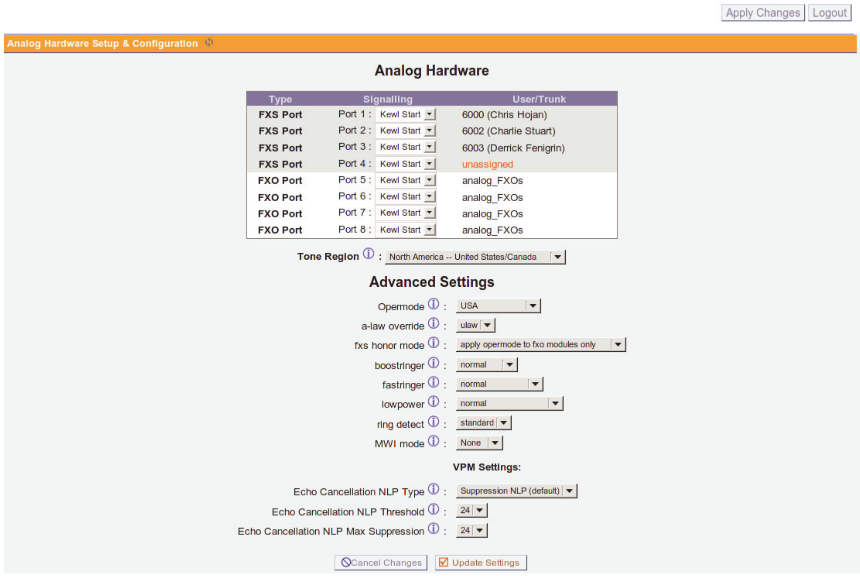


Figure 7: Configure Hardware

FXS and FXO ports provide the ability to receive and send calls through the traditional telephone network, or POTS (Plain Old Telephone System). FXS modules provide both dial tone and ringing voltage to an analog phone. FXO modules accept dial tone and provide an interface to the traditional phone lines. You plug a telephone line into an FXO port, and an analog telphone into an FXS port.

On this page you can specify the signalling type for your FXS and FXO ports. You have two choices; either Kewl Start or Loop Start. The Loop Start method uses a short to request a dial tone. All North American home phone lines use loop start signalling. Kewl Start is the same as Loop Start, but is better able to detect disconnects. Select either **Kewl Start** or **Loop Start** for each FXS and FXO module. Kewl Start is the default and is preferred for analog circuits in Asterisk.

Note: Ground Start signalling is not supported.

You also need to select a tone region, which defines the set of tones (dial tones, ringing tone, busy tone, etc) used in your region. Select your country, or the nearest neighboring country, from the **Tone Region** drop-down list. The default setting is North America (United States/Canada).

Advanced Analog Options

There are also some advanced settings which are applied to your analog hardware. Specify them as needed, or accept the default values.

- **Opermode -** Setting operation mode, or Opermode, sets the On Hook Speed, Ringer Impedance, Ringer Threshold, Current limiting, Tip/Ring voltage adjustment, Minimum Operational Loop current, and and AC Impedance selection as predefined for each countries analog line characteristics. Select the country in which your Asterisk Appliance 50 is operating.
- **A-law Override -** Set the audio compression scheme. The setting you choose is dependent on the country of operation. Ulaw is used in the United States and Canada. A-law is used in most other countries. If possible confirm the scheme which will be best for operation of your Asterisk Appliance 50.
- **FXS Honor Mode -** This setting lets you choose whether you apply the opermode setting to your FXO modules only, or to both FXS and FXO modules.
- **Boostringer -** Set the voltage used for ringing an analog phone. **Normal** will set ring voltage to a normal level, or **Peak** will set the voltage to 89v.
- **Fast Ringer -** The fast ringer tone can be set to normal, or to a 25hz tone.
- **Lowpower -** The low power setting can be set to normal, or to a Fast Ringer peak of 50v.
- **Ring Detect -** Users who are experiencing trouble detecting Caller ID from Analog service providers or whose lines exhibit a polarity reversal before Caller ID is transmitted from the provider should select **Full Wave**. Otherwise, choose **Standard**.

- **MWI Mode -** This option allows the user to specify the type of Message Waiting Indicator detection to be done on trunk (FXO) interfaces. The options are **none**, which performs no detection, **FSK** which performs Frequency Shift Key detection, or **NEON** which perform Neon MWI dection. The default value is none.
- **Echo Cancellation NLP Type -** This option allows you to specify the type of Non Linear Processor you want applied to the post echo-cancelled audio reflections received from analog connections. There are several options:
  - **None -** This setting disables NLP processing and is not a recommended setting. Under most circumstances, choosing **None** will cause some residual echo.
  - **Mute -** This setting causes the NLP to mute inbound audio streams while a user connected to the appliance is speaking. For users in quiet environments, **Mute** may be acceptable.
  - **Random Noise -** This setting causes the NLP to inject random noise to mask the echo reflection. For users in normal environments, **Random Noise** may be acceptable.
  - **Hoth Noise -** This setting causes the NLP to inject a low-end Gaussian noise with a frequency spectrum similar to voice. For users in normal environments, **Hoth Noise** may be acceptable.
  - **Suppression NLP -** This setting causes the NLP to suppress echo reflections by reducing the amplitude of their volume. Suppression may be used in combination with the **Echo cancellation NLP Max**

Suppression option. For users in loud environments, **Suppression NLP** may be the best option. This is the default setting for the **Echo Cancellation NLP Type** option.

- **Echo Cancellation NLP Threshold -** This option allows you to specify the threshold, in dB difference between the received audio (post echo cancellation) and the transmitted audio, for when the NLP will engage. The default setting is **24** dB.
- **Echo Cancellation NLP Max Suppression -** This option, only functional when the **Echo Cancellation NLP Type** option is set to **Suppression NLP**, specifies the maximum amount of dB that the NLP should attenuate the residual echo. Lower numbers mean that the NLP will provide less suppression (the residual echo will sound louder). Higher numbers, especially those approaching or equaling the **Echo Cancellation NLP Threshold** option, will nearly mute the residual echo. The default setting is **24** dB.

  Note: The VPM Settings section will not be visible on older hardware revisions of the Asterisk Appliance 50.

Once you have made the configuration changes to your hardware which you require, click **Save Changes**. A message will display letting you know that in order for these changes to be completed, you must reboot your Asterisk Appliance 50. Click **Options** on the left menu, select the **Reboot** tab, and then click **Reboot Now** to reboot your appliance. Rebooting your Asterisk Appliance 50 will terminate any active calls.

Trunk Configuration

Now that you have configured your analog hardware (assuming your unit had any) you are ready to set up your trunk lines. Trunks are outbound lines used to make calls. Trunks can be either analog or VoIP. Click **Trunks** from the main menu to access the trunk configuration page.

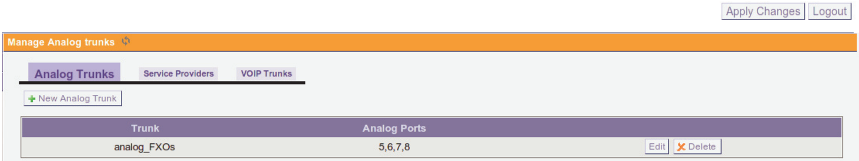


Figure 8: Trunk Configuration Page

Trunk definitions are used in calling rules, dial plans, and call routing, etc. You can use a mixture of both analog and VoIP trunks.

Analog Trunks

Select the **Analog Trunks** tab to access the **Manage Analog Trunks** page. Here you can create an analog trunk definition for each analog port on your Asterisk Appliance 50. Click **New Analog Trunk** to open the New Analog Trunk definition page.

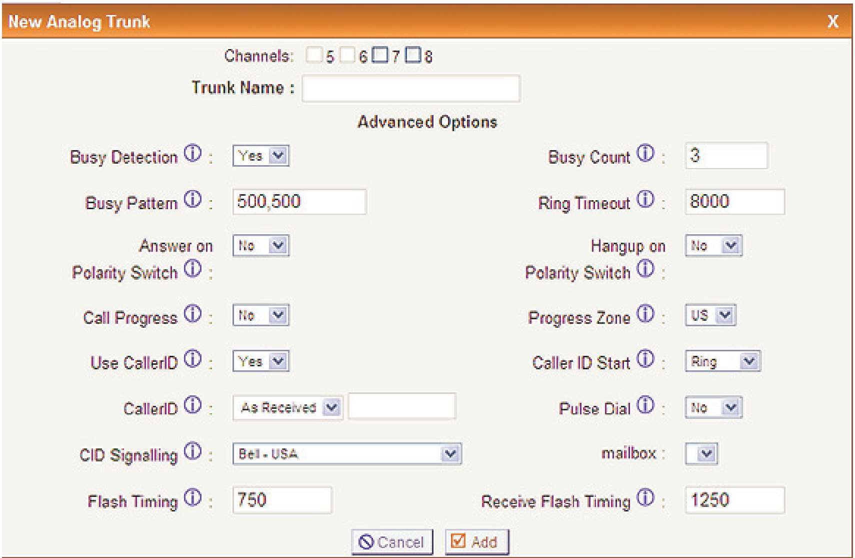


Figure 9: New Analog Trunk Definition

Use the following field definitions as a guide in creating your new analog trunk definition.

- **Channels -** Select one or more analog channel (port) to be associated with this trunk.
- **Trunk Name -** Specify a unique name to help you identify this trunk when it is referred to in other areas such as calling rules.
- **Busy Detection -** This setting is used to detect far end hangup or for detecting busy signal. Select **Yes** to enable this feature.
- **Busy Count -** If Busy Detection is enabled it is also possible to specify how many busy tones to wait for before hanging up. The default is 4, but better results may be achieved by setting to 6 or 8. The higher the number, the longer it will take to hangup a channel. A higher number also lowers the possibility of false detections.
- **Busy Pattern -** If Busy Detection is enabled, it is also possible to specify the cadence of your busy signal. In many countries it is 500 milliseconds on, 500 milliseconds off. Without Busy Pattern specified, the Asterisk Appliance 50 will accept any regular sound-silence pattern that repeats multiple times as a busy signal. If you specify Busy Pattern, then the Asterisk Appliance 50 will check the length of the sound (tone) and silence, which will further reduce the chance of a false positive.
- **Ring Timeout -** Trunk (FXO) devices must have a timeout to determine if there was a hangup before the line was answered. This value can be configured to shorten how long it takes before the Asterisk Appliance 50 considers a non-ringing line to have hung up.
- **Answer on Polarity Switch -** If this option is enabled the reception of a polarity reversal will mark when an outgoing call is answered by the remote party.

- **Hangup on Polarity Switch -** In some countries, a polarity reversal is used to signal the disconnect (or hang up) on a phone line. If the Hangup on Polarity Switch option is enabled, the call will be considered "hung up" on a polarity reversal.
- **Call Progress -** On trunk interfaces it can be useful to follow the progress of a call through Ringing, Busy, and Answering. If turned on, Call Progress attempts to determine answer, busy, and ringing on phone lines. This feature is **highly experimental** and can easily detect false answers and hang-ups. This may cause a hang up during the middle of a call. Few zones are supported, but can be selected with the Progress Zone option.
- **Progress Zone -** This option defines the call progress zone for the trunk interfaces.
- **Use CallerID -** If this option is enabled Caller ID detection is also enabled.
- **Caller ID Start -** This option allows one to define the start of a caller ID signal. Select **Ring** from the drop-down list to start caller ID when a ring is received, or **Polarity**, to start caller ID when a polarity reversal is detected.
- **Caller ID -** This option allows the lines to report the caller ID string as received from the telco, or as a fixed value by using the advanced option.
- **Pulse Dial -** If this option is enabled, pulse dialing, instead of DTMF, will be used.
- **CID Signalling -** This option defines the type of caller ID signalling to use.
  - **bell -** Bell202 as used in the United States

- **v23** - Used in the UK
- **v23_jp** - Used in Japan
- **dtmf** - Used in Denmark, Sweden, and Holland

- **Mailbox -** This setting allows any message waiting indicator received across the associated trunk to be forwarded to a local User, such as a SIP phone.
- **Flash Timing -** Flash Timing defines the duration, in milliseconds, that Asterisk will use if it is sending a flash signal to another system.
- **Receive Flash Timing -** Receive Flash Timing defines the duration, in milliseconds, that Asterisk requires in order to consider a flash operation it receives to be valid.

Once you have completed the Analog Trunk definition, click **Add.** A message will display letting you know that in order for these changes to be completed, you must reboot your Asterisk Appliance 50. Before doing so, you may wish to click the **Edit** button associated with an analog trunk to configure additional options for tuning the audio.

Figure 10: Edit Analog Trunk Definition

The **Audio Tuning** section will allow you to calibrate your analog ports for optimum performance. Please ensure that your analog lines are plugged in before clicking the **Easy Calibrate** button. Your Asterisk Appliance 50 must not have any active calls in order for the calibration process to complete successfully on all analog ports. If you wish to reset the calibration, click the **Reset Calibration** button.

> **Note:** The Easy Calibration feature can take approximately 90 seconds per port to complete.

In addition, an option to configure the gain level for each port will be listed. This option can be used to raise or lower the audio level on your ports. Normally, you should not have to adjust your analog ports beyond the initial calibration. Should you still need to fine tune your audio settings, please select one of the following:

- **Low**
- **Soft**
- **Normal**
- **Loud**
- **Louder**

Once you have completed the Analog Trunk definition, click **Update**. In order for these changes to be completed, you must reboot your Asterisk Appliance 50. Click **Options** on the left menu, select the **Reboot** tab, and then click **Reboot Now** to reboot your appliance. Rebooting your Asterisk Appliance 50 will terminate any active calls.

Adding Service Providers

You must configure a VoIP service provider in order to connect to the Public Switched Telephone Network (PSTN) via a VoIP connection. Access to the PSTN gives you the ability to place calls to telephone numbers no matter how they connect to the PSTN (VoIP or standard analog system). Click the **Service Providers** tab to add a VoIP (SIP or IAX) service provider.

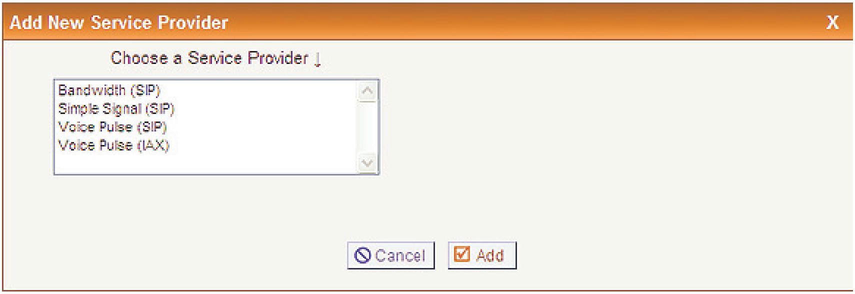


Figure 11: Add New Service Provider

The list of VoIP service providers and corresponding configuration information is pulled dynamically from a secure Digium webservice. If you are already a VoIP provider customer, select the provider from the list, click **Add,** and input your user name and password. Once you have added a service provider it will appear in the Service Providers list. There are **Edit** and **Delete** buttons associated with each Service Provider listing. Click **Edit** to further refine your service provider definition. A detailed definition will be displayed.

Figure 12: Edit VoIP Service Provider

The **Edit Service Provider** page gives you the ability to change your caller ID, as well as select a range of codecs.

- **Username/Password -** You will need to provide your log on credentials in order to update your service provider information.
- **Caller ID -** The caller ID sent to the PSTN will be set to the value specified in this field.
- **Codecs -** Codecs provide the ability for your voice to be converted to a digital signal and transmitted across the Internet. The quality of your call can be affected by the choice you make. The codecs available to you will depend on what is supported by the service provider you choose. You can select the order in which the codecs are used. The codecs commonly available are u-law, a-law, GSM, G.726, G.722, and G.729A. A registered G.729A license is required in order to use the G.729A codec.

Click **Update** when you have completed your changes, or **Cancel** to discard your changes.

Adding VoIP Trunks

If you do not have a subscription with one of the VoIP providers listed above, or you have a special VoIP setup, you can add a custom VoIP trunk. Click the **VoIP Trunks** tab to add a VoIP (SIP or IAX) service provider. The **Create New SIP/IAX Trunk** page will be displayed.

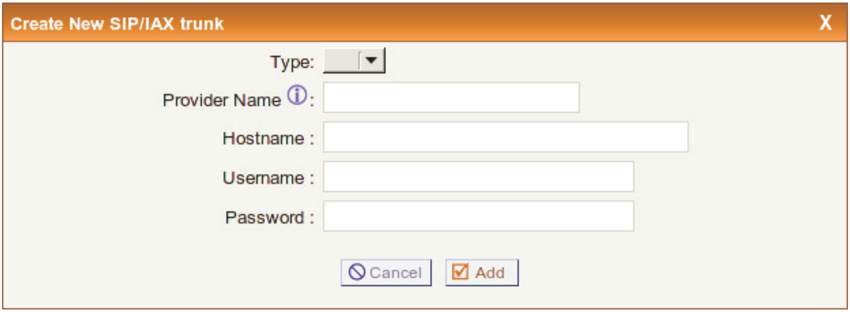


Figure 13: Create New SIP/IAX Trunk Definition

Fill in the initial SIP/IAX trunk definition with the following information:

- **Type -** Select either the SIP or IAX protocol.
  - **SIP -** Identifies that the trunk sends and receives calls using the VoIP protocol SIP.

  - **IAX -** Identifies that the turnk sends and receives calls using the VoIP protocol IAX.
- **Provider Name -** Enter a unique name to help you identify this trunk for use in calling rules, etc.
- **Hostname -** The hostname or IP address assigned to the VoIP provider or server.
- **Username/Password -** You will need to provide your log on credentials to the VoIP trunk server.

  Note: If your VoIP trunk does not require a username, you may leave the username field blank.

Click **Add** once you have completed your definition, or **Cancel** to discard your changes.

Once you have added a VoIP trunk it will appear in the SIP/IAX trunks list. There are **Edit** and **Delete** buttons associated with each VoIP trunk listing. Click **Edit** to further refine your trunk definition.

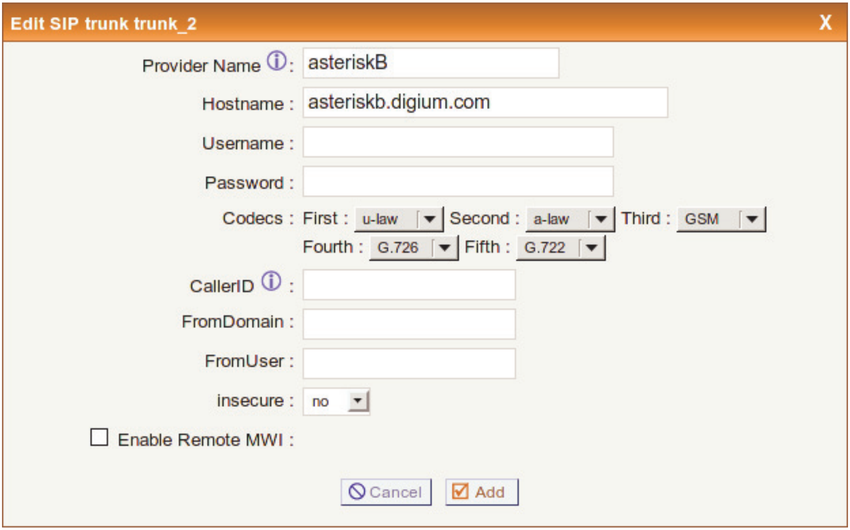


Figure 14: Edit VoIP Trunk

The following options will be available:

- **Provider Name -** Enter a unique name to help you identify this trunk for use in calling rules, etc.
- **Hostname -** The hostname or IP address assigned to the VoIP provider or server.
- **Username/Password -** You will need to provide your log on credentials in order to update your service provider information.
- **Codecs -** Codecs provide the ability for your voice to be converted to a digital signal and transmitted across the Internet. The quality of your call can be affected by the choice you make. The codecs available to you will depend on what is supported by the service provider you

choose. You can select the order in which the codecs are used. The codecs commonly available are u-law, a-law, GSM, G.726, G.722, and G.729A. A registered G.729A license is required in order to use the G.729A codec.

- **Caller ID -** This is the number the trunk will try to use when making outbound calls. For some providers it is not possible to set the CallerID with this option. Thus this option may be ignored. When making outbound calls the following rules are used to determine which Caller ID is used, if they exist:
  - The first Caller ID used is the Global CID defined in the **Options** tab.
  - The Caller ID set in the **VoIP Trunks** configuration, if defined, takes precedence over the Global CID.
  - The Caller ID set for the user making the call as defined in the **Users** page will take precedence over the Global CID and the CID set in VoIP trunks.
- **From Domain -** If required by your provider, specify your primary domain identity to show in the domain field of the From header for outgoing SIP invites. Otherwise, only your IP address will be sent in the From header.
- **From User -** If required by your provider, specify the user to show in the user field of the From header for outgoing SIP invites. Otherwise, only your IP address will be sent in the From header.
- **Insecure -** This is a SIP parameter used to determine peer matching. The setting determines whether or not an insecure connection will be allowed, or if authentication is required. The valid options are:
  - **port -** Enter this value to match against only an IP address. This setting is useful if you have multiple endpoints behind a NAT device.

  - **very -** Specify this value if you do not want to require authentication upon an initial invite.
  - **no -** Specify this value if you do not want to allow an insecure connection.
- **Enable Remote MWI -** When you select this option, you enable voicemail from your remote provider. Typically a user's voicemail is stored locally on the Asterisk Appliance 50. The notification of new voice mail is provided by the same local Asterisk Appliance 50. If you would like to receive voicemail notifications from a remote provider, this option is available. To enable this option, click the check box, and in the Remote Mail Box field, specify the remote mail box number or identity to which you wish to subscribe, *e.g*. 6001. Select the local user who should receive this MWI notification. Please note: enabling this option for a local user will disable the local user's Asterisk Appliance 50 voice mail. It is not possible to provide local voice mail and remote MWI simultaneously.

Click **Add** when you have completed your changes, or **Cancel** to discard your changes.

Outgoing Calling Rules

An outgoing calling rule pairs an extension pattern with a trunk used to dial the pattern. This allows different patterns to be dialed through different trunks (*e.g.* "local" 7-digit dials through an analog line but "long distance" 10-digit dials through a low-cost SIP trunk). You can optionally set a failover trunk to use when the primary trunk fails. The Outgoing Calling Rules give you the ability to use basic pattern matching to differentiate outbound calls and route them accordingly. The tab displays each outgoing calling rule established and the service providers assigned.

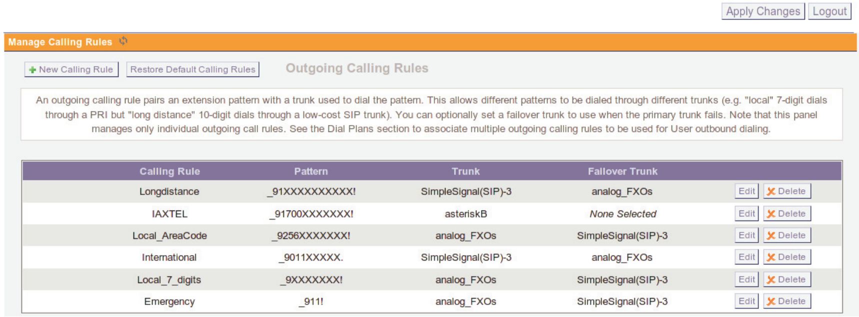


Figure 15: Outbound Calling Rules

> **Note: Outbound Calling Rules** manages only individual outgoing call rules. See the **Dial Plans** section to associate multiple outgoing calling rules to be used for User outbound dialing.

The Calling Rules menu shows every rule name established, the pattern the rule will match against, the trunk used to complete the call, and the failover trunk to be used. of call. Several default calling rules will be available when you initially set up your Asterisk Appliance 50. Click on

Add a Calling Rule to define a new calling rule. The following dialog will be displayed.

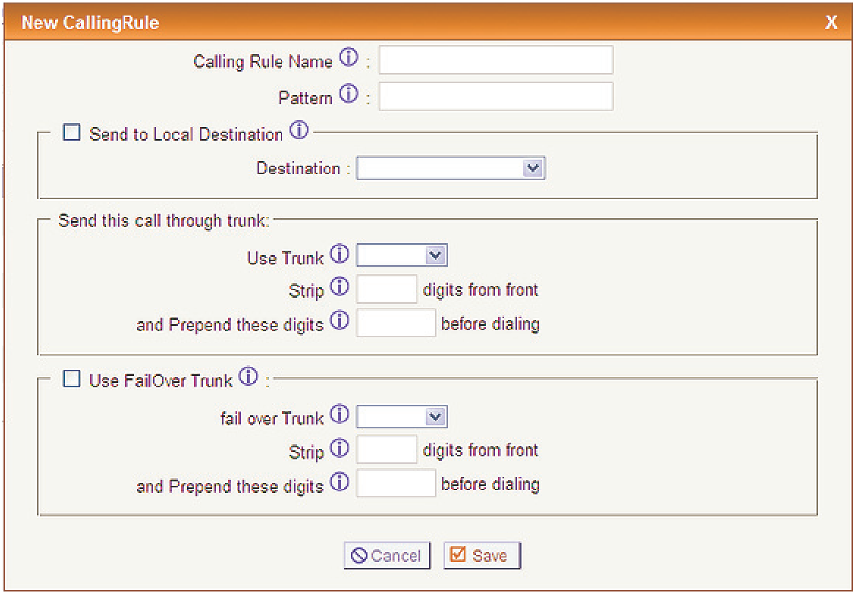


Figure 16: New Calling Rule

A calling rule is comprised of the following items:

- **Calling Rule Name -** Choose a name that describes the type of rule you are creating, *e.g.* "Local" or "Long Distance".
- **Pattern -** The Pattern field gives you the ability to use basic pattern matching to differentiate calls and route them accordingly. For instance, if a number begins with _9256, and is followed by 7 or more digits, that would define a call within the state of Alabama. If a call began with _9 followed by 7 digits, it would be a local call that proba-

bly doesn’t require a long distance charge. Instead of adding a rule for every extension or phone number you call, specify the pattern in this rule similar to the example. All patterns begin with the underscore “_” character. There are special characters which can be used in patterns:

- **X -** Any digit from 0-9
- **Z** - Any digit from 1-9
- **N** - Any digit from 2-9
- **[1,2,3,6-9] -** Any digit within the brackets, in this instance 1, 2, 3, 6, 7, 8, 9.
- **. -** The period is the wildcard which will match anything remaining. For example, _9011. matches anything starting with 9011.
- **! -** The exclamation point is a wildcard which causes the matching process to complete as soon as it can determine that no other matches are possible.

- **Send to Local Destination -** Calls matching the pattern specified will be routed to the destination specified in **Destination** if this checkbox is selected.
- **Destination -** Specify a destination, such as voicemail or main menu, for calls to be routed to when the **Send to Local Destination** checkbox is selected.
- **Use Trunk -** Specify the trunk through which calls, matching the specified pattern, will be placed.
- **Strip -** This option gives you the ability to remove specified number of digits from the front of the call string before the call is dialed and placed through the trunk specified in **Use Trunk**.

- **Prepend These Digits -** This option gives you the opportunity to add digits to the front of the call string before the call is dialed and placed through the fail over trunk. For example, a 3 digit area code could be prepended to a 7 digit string for calls to a service provider which requires 10 digit dialing.

  Note: You may also prepend the ‘w’ character for analog trunks to provide a 500ms delay before dialing. This is useful if your telecommunications provider does not immediately provide dial tone after going off hook.
- **Use Failover Trunk -** Failover trunks can be used to ensure that a call goes through if the primary trunk is busy or down. If the **Use Failover Trunk** checkbox is selected and **Fail Over Trunk** is specified, then calls that can not be placed through the primary trunk will be placed through this alternate route. If your primary trunk is a VoIP trunk, but you want calls to be placed through the PSTN when the VoIP trunk isn’t available, then this option will suit your needs.

Once you have completed the calling rule definition click **Save** to accept the rule or **Cancel** to abandon your changes. Click **Apply Changes** in the upper right corner of the page to make your changes immediately available. Click **Edit** next to a rule on the calling rule list to edit a previously defined rule, or **Delete** to delete the rule.

Dial Plans

A **Dial Plan** is a collection of Outgoing Calling Rules. Dial Plans are assigned to user extensions to specify the dialing permissions associated with that extension. For example, you might have one Dial Plan for local calling that only permits extensions associated with that Dial Plan to dial local numbers, via the "local" outgoing calling rule. Another extension may be permitted to dial long distance numbers, and so would have a Dial Plan that includes both the "local" and "longdistance" outgoing calling rules.

Click **New** at the top of the **Calling Rules** page and create a new dial plan name. You can then add calling rules for that dial plan definition.

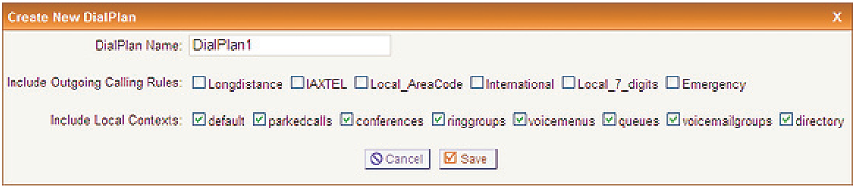


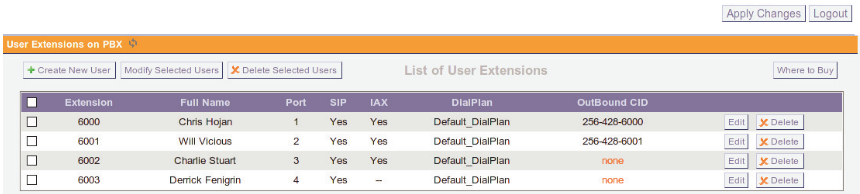
Figure 17: Create New Dial Plan

The default dial plan, the collection of your calling rules, is **Default_Dialplan**. You can create more than one dial plan, especially if you want to have different dial plans for different user extensions. Change the **DialPlanName**, and then select the checkbox for each **Outgoing Calling Rule** associated with this plan. You can also select which local contexts, such as conferences, voicemenu, and queues should be part of the dial plan.

Once you have completed the dial plan definition click **Save** to accept the plan, or **Cancel** to abandon your changes. Click **Apply Changes** in the upper right corner of the page to make your changes immediately available. Click **Edit** next to a dial plan on the list list to edit a previously defined plan, or **Delete** to delete a dial plan.

User Extensions

The User Extensions page is used to create individual user accounts on the system. Each user definition includes an extension, name, password, etc. User extension definitions are the basic components of your phone system. They are needed for voicemail, conferencing, call queues, dial plans, etc. Click the **Users** tab to view the main User Extensions page.



	Extension	Full Name	Port	SIP	IAX	DialPlan	OutBound CID		
☐	6000	Chris Hojan	1	Yes	Yes	Default_DialPlan	256-428-6000	Edit	Delete
☐	6001	Will Vicious	2	Yes	Yes	Default_DialPlan	256-428-6001	Edit	Delete
☐	6002	Charlie Stuart	3	Yes	Yes	Default_DialPlan	none	Edit	Delete
☐	6003	Derrick Fenigrin	4	Yes	--	Default_DialPlan	none	Edit	Delete

Figure 18: User Extensions

The main page lists all previously created user extensions. You can edit individual users as well as change attributes of several users at the same time. Your first step when setting up a new system will be to create one or more users. Click **Create New User** to create a new user extension.

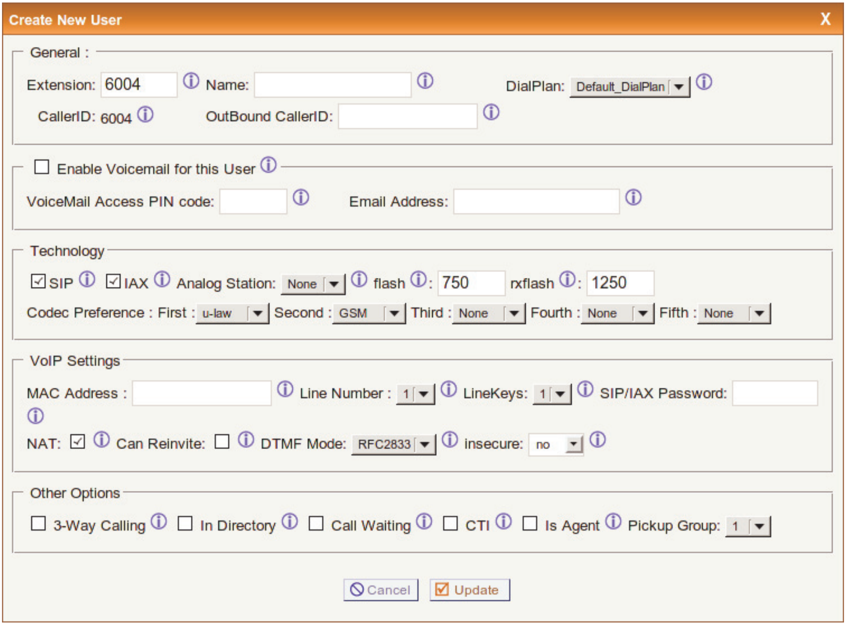


Figure 19: Create New User

- **Extension -** The numbered extension, *e.g.* 6000, assigned to the defined user. The extension must be a number within the range specified in **Extension Preferences** on the **Options** page.
- **Name -** The first and last name of the individual assigned to this extension. The name can also be that of a department, such as Sales or Support, for example. This is important because the Dial By Name Directory function of Asterisk uses this information to route calls.

- **Dial Plan -** This option references the Dial Plans option on the left tool bar. Based on the calling rules you've created, you can restrict the outbound dialing of this extension to local calls, emergency calls, and standard long-distance calls for North America. This option also possibly allows blocking or allowing international (011 prefix dialed) calls.
- **Caller ID -** Identifies the Caller ID presented when the listed extension dials an internal extension.
- **Outbound Caller ID -** Identifies the Caller ID presented when the listed extension dials an extenal number. Your ability to manipulate your outbound CID may be limited by your VoIP provider. Manipulation of CID across analog trunks is not possible.
- **Enable Voicemail -** Builds a voice mail box for the extension that can be reached by dialing the Check Voicemail extension. The Voicemail extension can be configured. The current default is 6050.
- **Voice Mail Access Pin Code -** The password used to access voicemail for the specified extension.
- **E-mail Address -** Voice mails received by this extension can be sent as audio file attachments e-mailed to a specific address.
- **SIP -** Identifies whether the extension sends and receives calls using the VoIP protocol SIP.
- **IAX -** Identifies whether the extension sends and receives calls using the VoIP protocol IAX.
- **Analog Station -** A drop-down menu is available to identify the analog phone port which this extension will access. If more than one phone is connected to your Asterisk Appliance 50 you may need to confirm the port number listed on the back of the Asterisk Appliance 50.
- **Flash -** Flash Timing defines the duration, in milliseconds, that Asterisk will use if it is sending a flash signal to another system.

- **RXFlash -** Receive Flash Time defines the duration, in milliseconds, that Asterisk requires in order to consider a flash operation that it receives to be valid.
- **Codec Preference -** Codecs provide the ability for your voice to be converted to a digital signal and transmitted across the Internet. The quality of your call can be affected by the choice you make. The codecs available to you will depend on what is supported by the service provider you choose. You can select the order in which the codecs are used. The codecs commonly available are u-law, a-law, GSM, G.726, G.722, and G.729A. A registered G.729A license is required in order to use the G.729A codec.
- **MAC Address -** The MAC Address field is used to specify the MAC address of a PolyCom® phone connected to the Asterisk Appliance 50. The MAC address associates the phone with this extension and enables the auto-synchronization of provisioning information.
- **Line Number -** Polycom brand VoIP phones are capable of servicing 1 to 6 separate VoIP phone lines, depending on the model of the phone. If you are using the Polycom Auto-provisioning feature of the Asterisk Appliance 50, this option can be used to define which line of your phone will be used by the user. No more than one user can be assigned to a line on a phone.

  Note: Each phone must be configured with a user that has Line Number set to 1. Additionally, assigned line numbers must be in a contiguous range.
- **Line Keys -** Polycom brand VoIP phones include multiple line keys. The number of line keys available will depend on the model of the phone. If you are using the Polycom Auto-provisioning feature of the Asterisk Appliance 50, this option can be used to define how many line keys on the phone should be associated with this user (*e.g.* Let's

says you configure a single Polycom phone with two users. User 6000 with Line Number set to 1 and Line Key set to 2 will display user 6000 on line keys 1 and 2 on the phone. User 6001 with the same MAC, Line Number set to 2, and Line Key set to 4 will display user 6001 on line keys 3, 4, 5 and 6 on the phone.). Be sure not to select more line keys than your phone supports.

- **SIP/IAX Password -** The password used if the user has a SIP/IAX account.
- **NAT -** Try this setting when your Asterisk Appliance 50 is on a public IP, communicating with devices behind a NAT device (broadband router). If you have one-way audio problems, you usually have problems with your NAT configuration or your firewall's configuration of SIP and RTP ports.
- **Can Reinvite -** By default, the Asterisk Appliance 50 will route the media streams from SIP endpoints through itself. Enabling this option causes the Asterisk Appliance 50 to attempt to negotiate the endpoints to route the media stream directly. It is not always possible for the Asterisk Appliance 50 to negotiate endpoint-to-endpoint media routing. This option can be used to tell the Asterisk Appliance 50 whether or not to issue a reinvite to the client.
- **DTMF Mode -** Set the default DTMF mode for sending DTMF (touch tone). The default setting is rfc2833. Other options include:
  - **info** - Used to display SIP Info messages
  - **inband** - Inband audio (requires 64 kbit codec - alaw, ulaw)
  - **auto** - Use rfc2833 if offered, inband otherwise.
- **Insecure -** Insecure is a SIP parameter used to determine peer matching. The setting determines whether or not an insecure connection will be allowed, or if authentication is required. The valid options are:

  - **port -** Enter this value to match against only an IP address. This setting is useful if you have multiple endpoints behind a NAT device.
  - **invite -** Enter this value to match against both the IP address and port number provided in the Contact field of the SIP header. A call will be allowed without authentication if a match is found.
  - **very -** Specify this value if you do not want to require authentication upon an initial invite.
  - **no -** Specify this value if you do not want to allow an insecure connection.
- **3-Way Calling -** Allows the extension to receive a call and then dial out to another phone number to conference with the inbound call and the recipient of the outbound call.
- **In Directory -** Check this option if you want a user to be searchable using the system telephone directory.
- **Call Waiting -** If call waiting is not enabled, the extension accepts only one call before it is identified as busy.
- **CTI -** Selecting this option (Computer Telephony Integration) allows the user to connect applications to the Asterisk Management Interface.
- **Is Agent -** Call queuing is made up of a bank of agents who receive calls. An extension listed as Is Agent can be added to queues from the Call Queues option.
- **Pickup Group -** A **Pickup Group** is a group of user extensions. Each member of a pickup group can answer another member’s phone by dialing *8. Select the pickup group to associate with the user extension.

Once you have completed the user extension definition click **Save** to accept the definition, or **Cancel** to abandon your changes. Click **Apply Changes** in the upper right corner of the page to make your changes immediately available. Click **Edit** next to a user extension on the list to edit a previously defined extension, or **Delete** to delete the user definition.

Editing Multiple User Definitions

You can edit multiple user definitions by selecting one or more user's checkboxes and then click **Modify Selected Users**. You will be able to edit the definition attributes common to all users such as Dial Plan, voicemail PIN, or Pickup Group setting. Click **Update** to update the selected users, or **Cancel** to abandon your changes. You can also delete multiple users by selecting one or more users from the displayed list and clicking **Delete Selected Users**. Click **OK** to complete the deletion, or **Cancel**.

Ring Groups

Ring groups allow a group of phones, or devices, to ring simultaneously or in sequence (ring order). This provides the opportunity for multiple people to answer a call (ring all) or one person can answer a call from any phone. The Asterisk Appliance 50 does not come with a default ring group. To create a new ring group click **New Ring Group** at the top of the **Ring Groups** page.

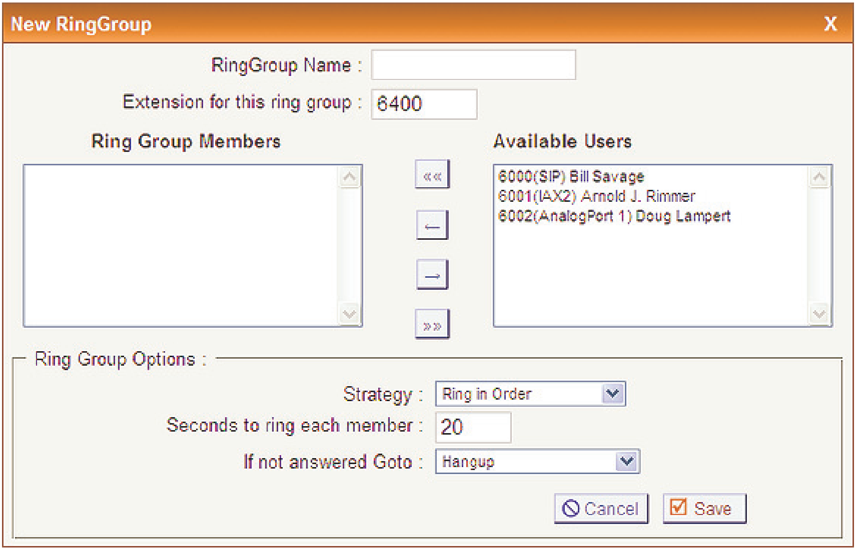


Figure 20: New Ring Group

Note: You need at least one member to be able to define a ring group. You will not be able to define a ring group without any user extensions.

To create a ring group, use the following procedure.

1. Define the **Name** of the group. The name can be any mnemonic such as **Sales** or **Technical Support.**
2. Specify an extension to associate with the ring group. This is the extension that can be dialed to ring all members of the group simultaneously or in order of listing.

   Note: Go to **Options, General Preferences** to see which range of numbers have been specified for ring groups.
3. Choose the members of the ring group from the **Available Users** list. Click on a user extension or trunk, and then click the arrow pointed at the **Ring Group Members** list to transfer. Select a user extension or trunk in the **Ring Group Members** list and then click the arrow pointing toward **Available Users** to transfer the selected item back to the list. Click the double arrow symbol to transfer all group members back to the **Available Users** list.
4. Choose a ring group strategy from the **Strategy** drop-down list. You can choose either **Ring All** which will ring all phones in the defined group simultaneously, or **Ring Order** which will ring phones in sequence determined by the order of the users or trunks in the group.
5. Specify the number of seconds that each phone (or all phones) should ring before ringing the next phone in order.
6. Lastly, determine which action you want the system to take if no one answers the call. You can either direct the call to the voicemail of a user, go to an IVR menu, or end the call.

Music on Hold

Music on hold is the music played to individuals on hold or during conference calls while conference members are waiting for the call to begin. The Asterisk Appliance 50 comes with a default group, or class, of sound files which can be used for music on hold. Click **Music on Hold** and then select the default class to see the list of default sound files.

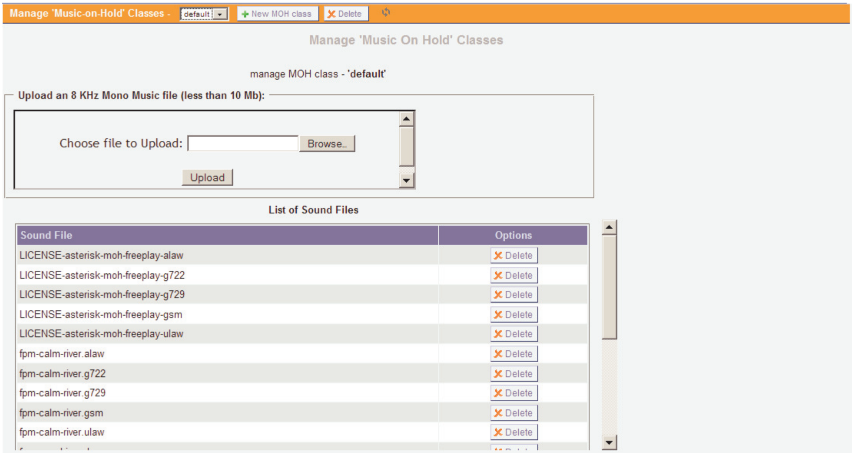


Figure 21: Music on Hold

If you think the default music is acceptable, but you'd like to give your system a more customized feel, you can also upload your own music or sound files. Each file uploaded must be less than 10 megabytes, in 8KHz mono, and in ulaw, alaw, g722, or gsm format. Not sure how to convert your audio to an acceptable format? Linux users should try the Sox utility, and Windows users should look into Audacity. Any conversion program is acceptable as long as the file meets the upload criteria.

Click **New MOH Class** to create a new label for a new group of music on hold files. Select the newly created class from the Music on Hold list, and then use the upload form to upload new music on hold files to the list. Once you have uploaded your files, click **Apply Changes** to make the files available. You can now use them for call queues, parked calls, conferences, etc.

Call Queues

A call queue lines up callers and allows them to wait to speak to any group of employees taking a high volume of calls. The feature allows you to speak to more people rather than send callers back to voice mail to leave a message and receive a call back when time permits.

Asterisk identifies which extensions under the **Users** tab are capable of belonging to a call queue by whether the **Is Agent** option is selected. The Is Agent option indicates that the user is available to answer customer calls. If a check mark does not appear next to Is Agent, that extension won't appear in the list of agents in the configuration for this option.

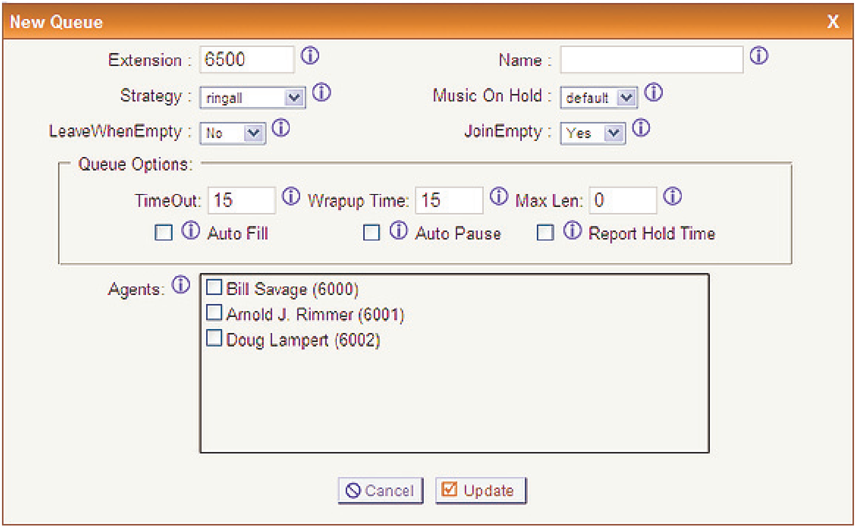


Figure 22: New Call Queue

The **Call Queues** page, with the **Queues** tab selected, lists the existing queues. None will be listed if you have not yet created a queue. To create a new queue, click **Create New Queue.** Use the following steps to create a queue. Keep in mind the purpose of the queue and how it should operate.

Creating a Queue

1. The extension for the queue will automatically populate in the **Queue** field with the next available extension. If you want the number to be something other than the automatically chosen one, enter it in the **Queue** field.

   Note: Go to **Options, General Preferences** to see which range of numbers have been specified for ring groups.
2. Next, give the queue a name that will be meaningful. The queue will be referenced by this name, so be sure to make it sufficiently descriptive as well. For example, “Technical Support” for the technical support queue, “Sales”, and so on.
3. You now should choose the strategy used in your queue call logic. Using the **Strategy** drop-down list, choose one of the following options for routing calls:
   – **Ring All -** Rings every agent who isn’t on an active call when a new call arrives. The first agent to answer the call receives it.
   – **Round Robin -** Every available agent receives a call in turn, akin to how cards are dealt in a poker game.
   – **Least Recent -** The agent who has been without a call the longest receives the next call.

  - **Fewest Calls -** The agent who has handled the fewest calls receives the next incoming call.
  - **Random -** Goes by the luck of the draw; any agent can receive the next incoming call.
  - **RrMemory -** This option is Round Robin with Memory. It's similar to Round Robin, but smarter — it remembers over the course of days, weeks, or years which agent received the last call so that it can commence with the next agent in sequence when calls begin again.

4. The **Agents** box lists all Users that are designated as an agent that can receive calls as part of a call queue. All users listed have the **Is Agent** checkbox selected on their user profile. Many Users may be listed as potential agents, but some may be assigned to a sales queue and some for a service queue. This box lists all agents and enables you to choose which users you assign to the queue.

You have now filled in the basic information necessary to create a call queue. There are additional queue options available to control the timing and managing of the calls, as well as the agents. You may not want to work with these finer points of call queuing until after your call queue has been working for a while, and you have an idea of call volume and the turnover of calls by each agent.

- **Music on Hold -** Select the music on hold class to associate with this call queue. Music on hold can be managed on the **Music on Hold** page.
- **Join Empty -** This option allows callers to enter a queue even if no agents are logged into it. There are three options available:
  - **Yes -** Callers can join a queue with no agents or only unavailable agents.

  - **No -** Callers can not join a queue with no agents. This is the default option.
  - **Strict -** Callers can not join a queue with no agents or if all agents are unavailable.
- **Leave When Empty -** This option mirrors the Join Empty, but it represents a queue in which agents had been logged in but are now gone. At 5:00 pm, when your employees go home, you can program the queue to shut down when the agents log out. The existing callers in queue are forced to exit, and no new callers are granted access to the queue. There are three options available:
  - **Yes -** Callers are forced out of a queue when no agents are logged in.
  - **No -** Callers will remain in a queue with no agents.

  - **Strict -** Callers are forced out of a queue with no agents logged in, or if all agents logged in are unavailable. This is the default option.
- **Timeout -** The default for this option is 15, representing 15 seconds that an agent's phone will ring before the call is forwarded on to another agent.
- **Wrapup Time -** This is a buffer of time allowing your agents to finish work on one call and remain unavailable in the queue. The default on this option is 0 seconds, providing no buffer time for your agents and allowing the next call to ring through immediately after a call is complete.
- **Max Len -** This option sets the maximum number of callers allowed in the queue before they are sent to voice mail or receive a busy signal. The default is "0," which allows for an unlimited number of calls in queue before they are sent elsewhere.
- **Auto Fill -** This option allows multiple calls that arrive at the same time to be immediately forwarded on to agents.
- **Auto Pause -** If an agent fails to answer a call, this option temporarily postpones sending calls to that agent.
- **Report Hold Time -** The Report Hold Time tells the agent how long the call was holding in queue before it was sent to the agent. If the hold time was short, the agent will probably be happy to accept the call. If the hold time was 10, 15, or 20 minutes, the agent might want to brace for a frustrated customer, but at least the agent isn't overwhelmed.

Click **Update** to add the new queue, or **Cancel** to abandon your changes. Once saved the new queue will be displayed on the **Manage Queues** page. You can edit or delete any previously created queue from the **Manage Queues** page.

Agent Login Settings

The **Agent Login Settings** tab, accessible from the **Manage Queues** page, lets you specify the extensions for agents to log into their queue, as well as callback login.

- **Agent Login Extension -** Use this field to specify the extension which all agents can dial to log into the queue(s) associated with their extension.
- **Agent Callback Login Extension -** Use this field to specify the extension which all agents can dial to log into the queue(s) associated with their extension. This is the same as **Agent Login**, but the agent does not have to remain on the line.
- **Agent Logout -** To logout of **Agent Login** just hang up your phone. To logout of **Agent Callback Login**, dial the same extension used to login, specify your extension and password when prompted, and press # when asked for your callback extension. This will successfully log you out of all queues.

Click **Save** to retain the agent login settings.

Voice Menus

A valuable feature of Asterisk is the ability to create Interactive Voice Response (IVR) or voice menus. Voice menus are designed to allow for more efficient call routing. The menus provide a caller with specific instructions, receive responses from the caller, and process those responses into an action.

Each Asterisk Appliance 50 ships with a default voice menu already created. To better understand the creation and operation of these menus, we will examine the default one.

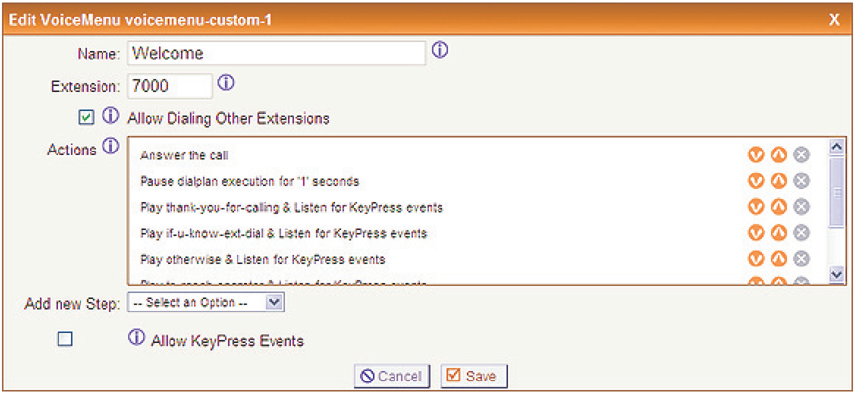


Figure 23: Default Voice Menu

Voice menus are constructed depending on your needs. Just like your business you need to create the solution best suited to your customers. The best way to understand how a voice menu is constructed is to examine the default “Welcome” menu provided with your Asterisk Appliance 50. Click **Voice Menus - Welcome** in the **Voice Menus** list. The options for the welcome menu are displayed similar to the example shown in the above illustration. The Welcome menu consists of the following steps:

- Answer the Call
- Wait ‘1’ Sec
- Play ‘thank-you-for-calling & Listen for KeyPress
- Play ‘if-u-know-ext-dial’ & Listen for KeyPress
- Play ‘otherwise’ & Listen for KeyPress
- Play ‘to-reach-operator’ & Listen for KeyPress
- Play ‘pls-hold-while-try’ & Listen for KeyPress
- WaitExten ‘6’ Sec

In the example, when a caller dials your company number ending in 7000, the call is answered, and after a pause of one second the caller is greeted in the following manner: “Thank you for calling. If you know your party’s extension, please dial it now. Otherwise to reach an operator please dial 0.” If the caller tries an extension, the menu will respond with “Please wait while I try that extension.” If no action is taken by the caller, the menu will repeat after 6 seconds.

This is an example of a basic voice menu. In the example, each action is a step chosen from the list of available menu options. The available menu options are as follows:

- **Answer -** This step is automatically added when creating a new menu. This step answers the incoming call.
- **Authenticate -** The Authenticate step is used to restrict access to one or more areas of your system. This is useful when one wants users to have to enter a PIN code in order to proceed to a particular part of the current menu, to a different menu, or to ring an extension.

- **Background -** This step is used to play an audio file in the background while waiting for the caller to enter an extension or number. Playback is terminated once the user begins to enter an extension. To select a file to play, click and hold in the field next to the **Background** choice to scroll through a list of pre-recorded sound files. In the example above, “Play ‘otherwise’ & Listen for KeyPress” is an example of using the **Background** option.
- **Busytone -** The Busytone option should be selected if there is a step in the process in which you want to play a busy signal to the caller. You would play the busyone to the caller, for instance, if the call is over.
- **Congestion -** The **Congestion** option is similar to the **Busytone** option. The Congestion option should be selected if there is a step in the process in which you want to play a congestion signal to the caller. You would play the congestion signal to the caller, for instance, if you want to indicate the line is not available.
- **Digit Timeout -** The Digit Timeout option is used to set the maximum amount of time allowed between key presses. If a full extension is not entered in the specified time, the entry will be considered invalid. A field for entering the number of seconds before timeout appears next to the option.
- **DISA - DISA** (Direct Inward System Access) is an application which allows callers from outside the system to get access to an internal dial tone and place calls from within your internal system. A passcode is required. If the passcode entered is correct, the user is given a system dial tone on which a call may be placed.

  Note: Use caution when choosing this option. This option can pose a security risk.

- **Response Timeout -** If a caller does not enter a response with the time specified in this field, the call will terminate. This step could be put at the end of a series of menu choices.
- **Playback -** The **Playback** option is similar to the **Background** option because it will play a sound file you select. However, this option does not allow interruption from a KeyPress event, and will move on to the next step in your list.
- **Set Music on Hold Class -** Set the group of music on hold files to be associated with this voice menu.
- **Wait -** The **Wait** option pauses the execution of steps in the voice menu list for the number of seconds you specify.
- **WaitExten -** The **WaitExten** option is specified to give a caller a specified amount of time to enter an extension.
- **Goto Destination -** The **Goto Destination** option lets a caller choose to go to either a voice menu, a specific extension, voicemail box, or a ring group from a list of destinations.
- **Set Language -** This option gives you the ability to set the language for voice prompts in your voice menu. This option is especially useful if you want to begin with the default language, and then give the option of setting a different language for the rest of the menu. For

example, voice prompts will begin in English, but if a user is given a choice, and presses 2 for Spanish, all further voice menu prompts will be in Spanish (provided that language module is loaded).

- **Goto Directory -** The **Goto Directory** option sends a caller to the system phone directory. This gives the user the chance to select a user name from the directory if the extension is unknown.
- **Dial a Number via Trunk -** This option allows you to specify an external number to dial, including the trunk that should be used for the call.
- **User Event -** This option gives you the ability to send an arbitrary event to the manager interface, with an optional body representing additional arguments. Specify the eventname in the User Event field. If necessary, specify additional arguments in the Body field.
- **Hangup -** The Hangup option terminates the call.
- **Custom App -** This option allows you to specify an Asterisk application, along with the application's corresponding parameters, which is not already listed in the **Add new Step** drop-down menu (*e.g.* 'NoOp(hello world)' to echo "hello world" on the Asterisk CLI).

  Note: The **Custom App** option is only visible when **Advanced Options** are enabled under the **Options** menu item. This option should only be configured by experienced Asterisk administrators. Refer to section titled **Advanced Options** on page 138 for further details.

Creating a Voice Menu

Use the following procedure as a guide to creating a voice menu.

1. On the **Voice Menu** page, click **New** to create a new voice menu.
2. Specify a **Name** and an **Extension**. The extension will be the direct dial to the voice menu.
3. Specify the **Steps** of your voice menu using the welcome menu example and step descriptions as guides.
4. Select the **Dial Other Extensions** checkbox if you want to give a user the ability to break out of the menu selections and dial an extension within your system.

   Warning: The Dial Other Extensions option is important. This option allows an inbound caller to break out of the listed Keypress Events and dial another extension. A malicious person may be able to hack through your Asterisk implementation to find an outside dial tone and use it for fraud. Any extensions that are known to the public should be completely handled by the Keypress Events; callers should not be allowed to dial other extensions. Sticking to this policy protects your Asterisk system from being compromised.
5. Specify the **Keypress Event** actions for digits 0-9 as well as *, #, t, and i. The options available for a Keypress Event are:
   – **None -** The associated key is not enabled.
   – Goto Menu - Pressing a key with this option will send the caller to a specified menu.
   – **Goto Extension -** Pressing a key with this option will send the caller to a specified extension.
   – **Goto Queue -** Pressing a key with this option will send the caller to the specified queue.

- **Operator** - This option will send the caller to the designated operator.
- **Hangup** - Pressing a key with this option will terminate the call.
- **Congestion** - Pressing a key with this option will play a busy signal.
- Both the **t** key and **i** key should be used for specific actions. The action associated with the **t** key should be the desired action if a user response has timed-out. The action associated with the **i** key should be the desired action if a user makes an invalid entry.

6. Once you have constructed your voice menu, click **Save**. You can then click **Apply Changes** to add the voice menu to your current configuration.

Record a Voice Menu

In the event that one wants to record custom menu prompts for Asterisk which can be used in a voice menu, the **Voice Menu Prompts** tab may be used.

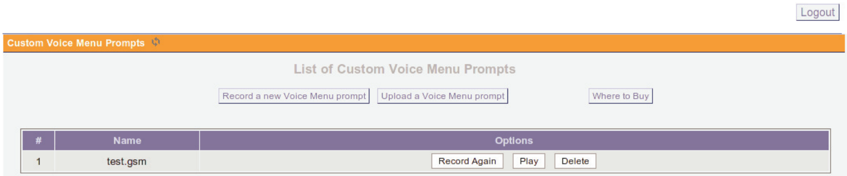


Figure 24: Custom Voice Menu Prompts Page

A list of previously recorded menus is displayed on the **Custom Voice Menu Prompts** page. Here, the user may modify several options:

- **Record Again -** Clicking this button allows the user to make another attempt at recording and replacing an existing custom sound file.
- **Play -** Clicking this button brings up a dialog entry box to allow the input of an extension that Asterisk will dial and play the prompt.
- **Delete -** Clicking this button will delete the selected prompt.

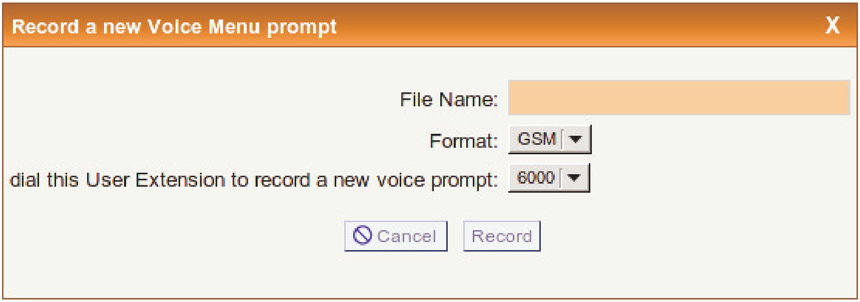


Figure 25: Record Menu Prompts

Click **Record a new Voice Menu Prompt** to record a custom voice menu prompt. The following options will be available:

- **File Name -** This text entry box specifies the saved name of the file that is to be recorded.
- **Format -** Select whether the recording will be in GSM or WAV format.
- **Extension Used for Recording -** This drop-down select box allows the user to choose which extension Asterisk will dial to wait for the user to speak the prompt.
- **Record -** Clicking this button causes Asterisk to launch the call that will record a file.

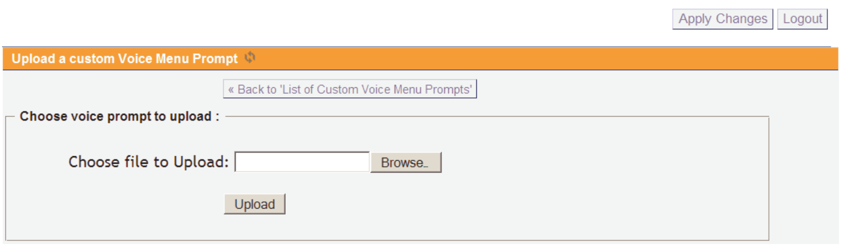


Figure 26: Upload Menu Prompts

Click **Upload a Voice Menu prompt** to upload a custom voice menu prompt. You will be prompted to specify the path to the audio file that you wish to upload. Each file uploaded must be less than 10 megabytes, in 8KHz mono, and in GSM or WAV format.

Once your recording or upload of a custom voice menu prompt is finished, it will be listed on the **Custom Voice Menu Prompts** page. You will be able to play back the prompt, re-record the prompt, or delete the prompt. The prompts can now be included when creating voice menus.

Time Intervals

Time intervals are definitions of a period of time during a day, week, month, etc. which are used to route calls. Time interval definitions are utilized in the **Incoming Calling Rules** section. To define a time interval, select **Time Intervals** from the left menu, and then **New Time Interval** from the Time Intervals page.

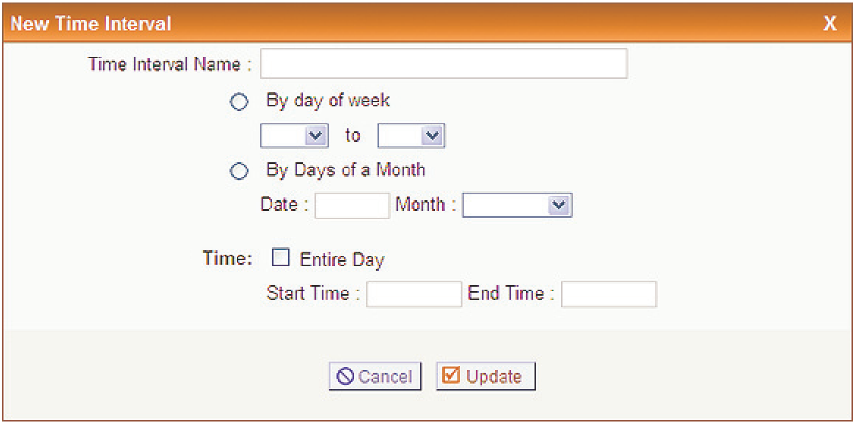


Figure 27: New Time Interval

Creating a Time Interval definition is fairly simple. You just need to define a range of time in which you expect to receive calls. The following fields are used to create the definition:

- **Time Interval Name -** Specify a unique name to help you identify this time interval when it is referred to in the creation of calling rules. A name can be anything such as BusinessHours, OffHours, or Holiday.
- **By Day of Week -** Select this radial button if you wish to specify one or more days of any week. Select the range of days using the drop-down lists. For example, if you were creating the time interval "Business Hours" you would specify Monday in the first drop-down list and Friday in the second drop-down list. For time intervals that occur on a single day, select that day in both drop-down lists.
- **By Days of a Month -** Select this radial button if you wish to specify a day of a specific month instead of a day of a week. Enter the day of the month, and then select the month from the drop-down list. For example, if you were creating a time interval named Christmas, you would enter "25" and then select "December" from the drop-down list.
- **Time -** You need to specify a time during which this interval should be applied. Select either the **Entire Day** checkbox, or a **Start Time** and **End Time.** In the Business Hours example, which is from Monday to Friday, you would specify a start time of 8:00 AM and an end time of 5:00 PM. In the "Christmas" example you would select the **Entire Day** checkbox.

Click **Update** to save your time interval definition, or **Cancel** to discard your changes. Click **Apply Changes** to make the new time interval active.

Once a time interval definition is created, you can either **Edit** or **Delete** the definition from the **Time Interval** page.

Incoming Calling Rules

Incoming Calling Rules give you the ability to use basic pattern matching to route incoming calls based on time intervals for each analog or VoIP trunk with which you receive inbound calls. Click **Incoming Calling Rules** to access the Incoming Calling Rules page.

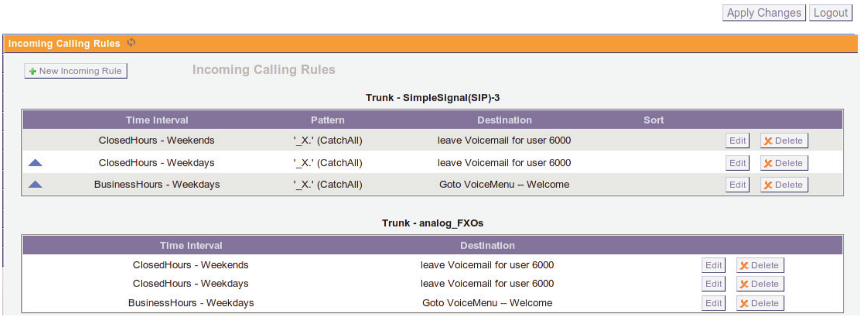


Figure 28: Incoming Calling Rules

The main page displays the incoming calling rules created for each trunk. No rules are displayed if you have just setup your Asterisk Appliance 50. Click **New Incoming Rule** to create a new incoming calling rule. The new incoming rule form will be displayed.

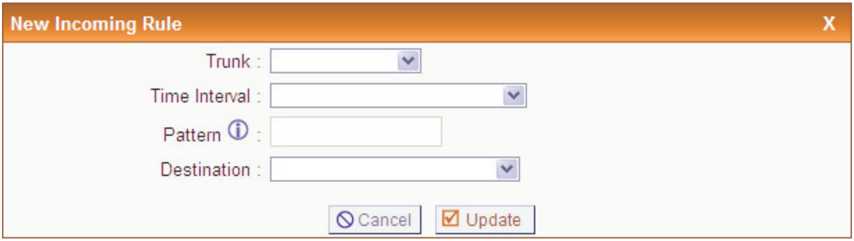


Figure 29: Incoming Calling Rules

There are only a few options you will need to define to create a new rule.

- **Trunk** - Select the trunk which the incoming rule should apply to from the drop-down list. The trunk can be either an analog or VoIP trunk.
- **Time Interval** - Select the time interval from the list available in the drop-down list. You may have created time intervals for business hours, weekend hours, holiday time, etc. You can also select **None** if you want to bypass any time intervals or patterns.
- **Pattern** - The Pattern field gives you the ability to use basic pattern matching to differentiate calls and route them accordingly. For instance, if a number begins with _9256, and is followed by 7 or more digits, that would define a call within the state of Alabama. If a call began with _9 followed by 7 digits, it would be a local call that probably doesn't require a long distance charge. Instead of adding a rule for every extension or phone number you call, specify the pattern in this rule similar to the example. All patterns begin with the underscore "_" character. There are special characters which can be used in patterns:
  - **X** - Any digit from 0-9

  - **Z -** Any digit from 1-9
  - **N -** Any digit from 2-9
  - **[1,2,3,6-9] -** Any digit within the brackets, in this instance 1, 2, 3, 6, 7, 8, 9.
  - **. -** The period is the wildcard which will match anything remaining. For example, _9011. matches anything starting with 9011.
  - **! -** The exclamation point is a wildcard which causes the matching process to complete as soon as it can determine that no other matches are possible.

  Note: If you have selected an analog trunk, this field will be grayed and populate with an s. This is not a pattern, but an indication that the analog phone should proceed to the destination.

- **Destination** - Select the **Destination** for the incoming call. You can choose to send the call to to either a voice menu, a specific extension, voicemail box, ring group, the operator, or even hang up the call.
  - The **Local Extension by DID** destination setting allows you to route the incoming call to a local extension based on the DID (Direct Inward Dialing) number that is sent to you by your telecommunications provider. Upon selecting **Local Extension by DID**, you will notice the **Local Extension by DID Pattern** option appear. This option gives you the ability to remove a specified number of digits from the front of the DID number string before routing the call to a local extension.

  Note: The **Local Extension by DID** destination setting is not applicable for analog trunks.

The rules you need to create are dependent on your needs. If you are configuring your system for a business, for example, you’ll probably want to set up rules for business hours, off hours, weekend hours, etc. In any case, you should also create a calling rule which utilizes the time interval and uses a catch all pattern to route any calls that don’t fit the other rules you’ve created. This will insure that you don’t miss any calls.

Once you have completed the definition of each incoming calling rule, click **Update.** Click **Apply Changes** in the upper right corner of the page to make your changes immediately available. Each rule you create will be listed on the **Incoming Calling Rules** page, organized by trunk. From the main page you can either **Edit** or **Delete** the rule.

Voicemail

Voicemail is an option available for every extension. The relationship between the extension and voicemail is established in **Users**. In that section you can specify whether voicemail is enabled for an extension, as well as the PIN for retrieving voicemail. The **Voicemail** page lets you specify voicemail parameters, as well as settings for sending voicemail notices to e-mail.

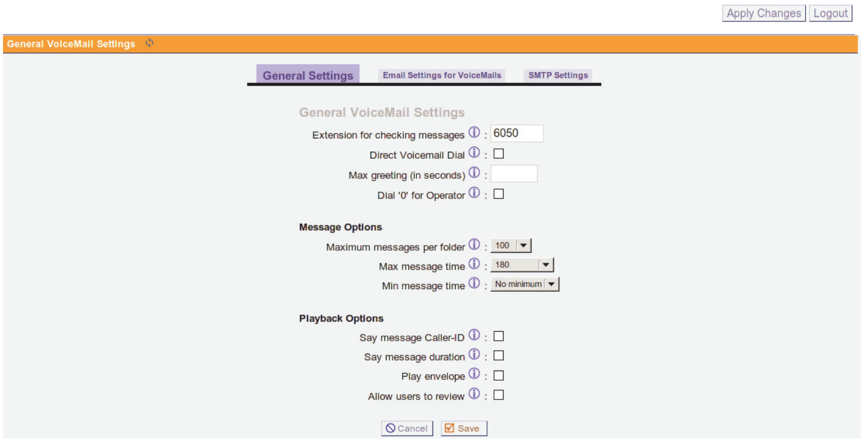


Figure 30: Voicemail

There are three tabs on the **Voicemail** page used for configuration: **General Settings**, **Email Settings**, and **SMTP Settings**.

General Settings

The General Settings page is the primary page used to configure Asterisk Appliance 50 voicemail. Standard configuration information is present, allowing you to confirm the extension used to check messages, as well as general parameters such as the following:

- **Extension for Checking Messages -** This option defines the extension which Users call in order to access their voicemail account.
- **Direct Voicemail Dial -** Select this checkbox to enable direct voicemail dialing. For example, someone would be able to dial *6001 to directly dial the voicemail box and leave a message for the person at extension 6001 if this checkbox is selected.
- **Max Greeting (Seconds) -** With this option, you specify the maximum amount of time available to record your voicemail greeting.
- **Dial “0” for Operator -** Callers who are sent to voicemail can press “0” for the operator and be transferred either during the voicemail salutation, or after recording the message. If this option is not enabled, a caller’s pressing “0” will be ignored.

There are several options which can define the characteristics of the voicemail messages in the system.

- **Maximum Messages per Folder -** This field sets the maximum number of messages that a user can have in any over their voice mail box folders.
- **Maximum Message Time -** The maximum duration of a message left by a caller. Time is specified in seconds.
- **Minimum Message Time -** The minimum duration of a message specified in seconds. Any message left that’s under the listed duration is discarded and isn’t processed or retrievable.

There are also several message playback options which can be specified.

- **Say Message Caller-ID -** The Say Message Caller ID option reads the caller ID before the voice mail message is played.
- **Say Message Duration -** If this option is enabled the duration of the message, in minutes, will be played back before the voicemail message is played.
- **Play Envelope -** Turn on/off playing introductions about each message when accessing them from the voicemail application.
- **Allow Users to Review -** This option provides incoming callers the option to review their message before it is saved and can be played back by the owner of the voicemail extension. Standard options are presented to the caller, allowing them to discard the message or re-record it.

E-mail Settings

The E-mail Settings page is used to set e-mail options for voicemail, as well as the format of the e-mails sent.

Note: SMTP settings must be specified in order to send e-mail.

- **Send Messages by E-mail Only -** If this option is set, voicemail messages will only be accessible by e-mail.
- **Attach Recordings to E-Mail -** This option is used to choose whether voicemail is sent to a users e-mail address as an attachment. Click the check box to enable this option. Messages will be sent in the .wav format.
- **Template for E-mails -** The e-mail template gives you the ability to specify the general content for each e-mail sent with a voicemail alert. To load a sample template, click the **Load Defaults** button. Be sure to change the **From** address to a valid e-mail address before saving.

SMTP Settings

The SMTP Settings page is used to enable sending voicemail alerts through e-mail.

- **SMTP Sever -** The IP address or a hostname of an SMTP server which the Asterisk Appliance 50 can connect to, without authentication, to send voicemail notifications to an e-mail address.
- **Port -** The port number on which the SMTP server is running. The default port is 25.
- **Use SMTP Authentication -** Click this checkbox if the SMTP server requires a username and password for authentication.
- **Auth User -** The username used for authentication to the SMTP server.
- **Auth Password -** The password used for authentication to the SMTP server.

Once you have completed specifying the settings on a tab, click **Save** to keep your settings, or **Cancel** to discard your settings. Click **Apply Changes** in the upper right corner of the page to make your changes immediately available.

Paging/Intercom

The Paging/Intercom tab allows you to set up 1-way paging or 2-way intercom for calling an individual or a group of extensions. This can be used to make an announcement over the speakerphone of a group of phones. Phones which are part of a page/intercom group will not ring, but will immediately answer into speakerphone mode.

> **Note:** This functionality is dependent on a compatible and correctly configured handset. For a user to be able to dial a page/intercom group, the 'pagegroups' local context must be included in the user's dialplan.

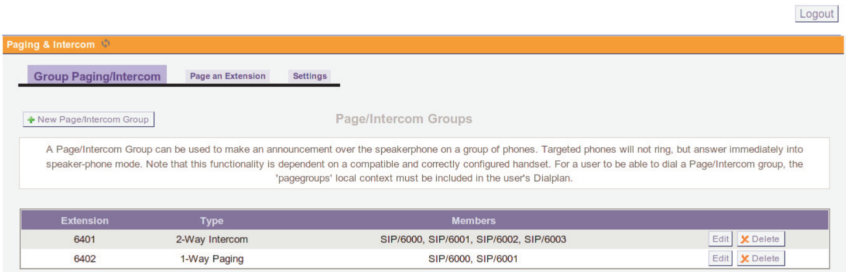


Figure 31: Paging/Intercom

Click **New Page/Intercom Group** to define which available users will be part of a page/intercom group.

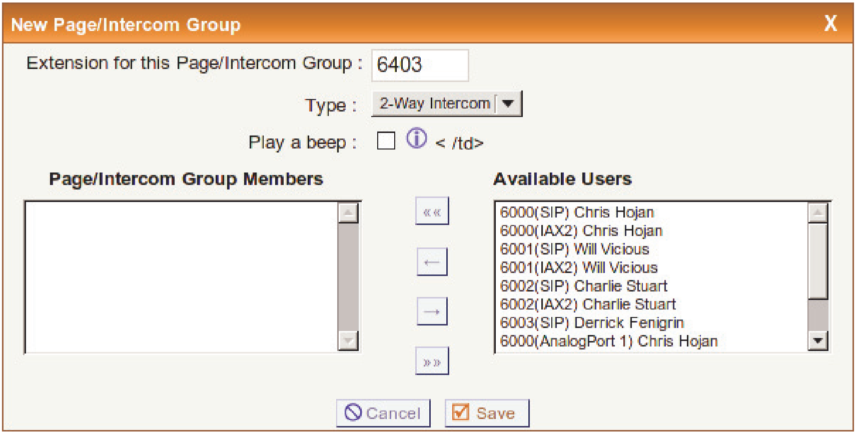


Figure 32: New Page/Intercom Group

The following options are available when defining a new page/intercom group:

- **Extension for this Page/Intercom Group -** Specify the extension associated with this page/intercom group.
- **Type -** Specify the type of group for this extension.
  - **2-Way Intercom -** The person initiating the call and all members of the intercom group will be able to speak to each other during the call.

   – **1-Way Page -** Only the person initiating the call will be able to speak during the call. All members of the paging group will be muted.

- **Play a beep -** If this option is checked, a beep sound will be played when the intercom call is connected to inform users that they can begin talking.
- **Page/Intercom Group Members -** This is the list of available users which are part of this page/intercom group.
- **Available users -** This is the list of users which are available to be assigned to this page/intercom group.

The double left arrows will move all available users to this page/intercom group. The double right arrows will remove all page/intercom group members. The single left arrow will be move an individual available user to the page/intercom group. The single right arrow will remove an individual page/intercom group member.

Click **Save** to retain your page/intercom group, or **Cancel** to abandon your changes. From the **Paging & Intercom** page, you can either **Edit** or **Delete** a page/intercom group.

Click **Page an Extension** along the top to configure a key sequence which initiates a page or intercom call to a specific extension.

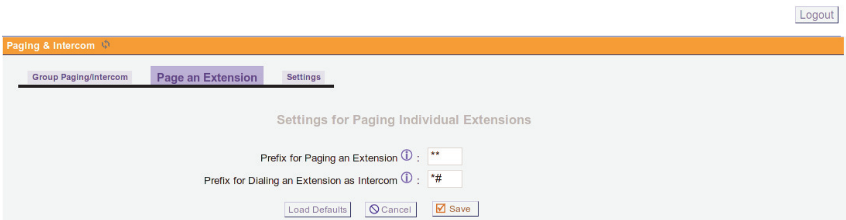


Figure 33: Settings for Paging Individual Extensions

The following settings are available:

- **Prefix for Paging an Extension** - Specify the key sequence used to prefix a page call to a specific extension. For example, setting this value to ** would allow you to initiate a page call to extension 6000 by dialing **6000**.
- **Prefix for Dialing an Extension as intercom** - Specify the key sequence used to prefix an intercom call to a specific extension, For example, setting this value to *# would allow you to initiate an intercom call to extension 6000 by dialing ***#6000**.

Click **Save** to retain your changes, or **Cancel** to abandon them.

Then click **Settings** along the top to specify additional settings for paging and intercom.

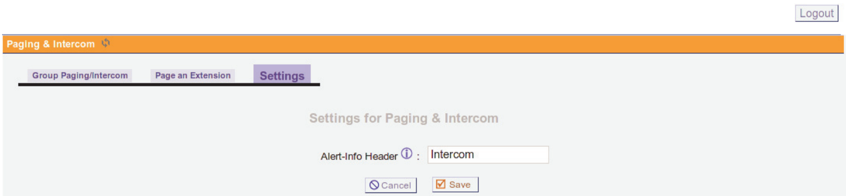


Figure 34: Paging & Intercom Settings

The following setting is available:

- **Alert-Info Header** - This is the value that is sent in the alert info header to the phone for an intercom call. It is not recommended that this value be changed from the default of **Intercom**.

Click **Save** to retain your changes, or **Cancel** to abandon them. Once you have completed making changes to the **Paging & Intercom** sections, click **Apply Changes** to make them immediately available.

Conferencing

Every company reaches the point of needing more people on a phone call than it can effectively include through three-way calling. Conference bridges allow you to include more people as well as project a professional image. The configuration of the conference bridge and standard features is very straightforward. Click **New Conference Bridge** on the **Conferencing** page to design a conference bridge.

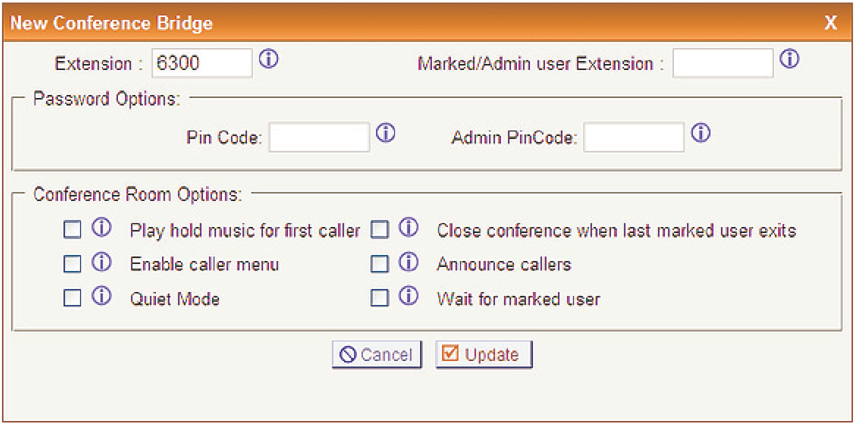


Figure 35: New Conference Bridge

The GUI auto-populates the extension with the next available extension in sequence, but can be changed to any extension number that is available. After establishing the extension for the bridge, you need to specify the password settings for the conference. Assign the **PIN Code** used by participants to enter the conference as well as the **Administrator PIN Code** used by the moderator of the conference to open the conference bridge.

Now that you have established the conference bridge extension and password codes, you can set your conference room options.

- **Marked/Admin User Extension -** This option works in conjunction with the Wait for Marked User feature. If the conference bridge is to have marked users or admin users, those users should enter the conference from a separate extension. Admin users can lock and unlock the conference, and can kick the most recent conference participant. Marked users are special users whose entrance and exit, if the **Wait for Marked User** or **Close Conference When Last Marked User Exits** are selected, can either begin or end the conference. If the CEO of the company, for example, doesn't want anyone chatting in the conference bridge until he or she arrives, these options are set to keep everything quiet. The main conference extension of 6003 is configured with Wait for Marked User selected. Everyone in the conference arriving from extension 6003 remains silent until the CEO arrives.
- **Play hold music for first caller -** Checking this option makes music play for the first caller entering a conference until another caller joins. Some people don't like sitting in a quiet room — even a virtual room — alone, and this feature prevents anyone from being in that position.
- **Enable caller menu -** This feature allows callers to access the Conference Bridge Menu by pressing the asterisk (*) key.
- **Quiet Mode -** Do not play enter/leave sounds when callers join or leave the conference.
- **Close Conference When Last Marked User Exits -** When this option is selected, the conference call will be closed when the last marked user exits the call.
- **Announce callers -** All new callers to a conference are identified when they arrive when this feature is selected.

- **Wait for Marked User -** This is a feature that keeps all participants in quiet mode until a special participant, using a unique extension, arrives. Only after the marked user arrives is the audio activated so that all of the participants can speak to each other.

Click **Update** to retain your conference bridge definition, or **Cancel** to abandon your changes. From the **Conferencing** page you can either **Edit** or **Delete** a bridge definition. Once you have saved a conference bridge definition click **Apply Changes** to make the bridge immediately available.

Follow Me

Follow Me is a very useful feature which allows a caller to reach you wherever you may be by forwarding your calls to a list of predefined numbers until you are reached. If you cannot be reached, Follow Me will transfer the caller to your voicemail box. The Follow Me feature may also be referred to as Find Me.

> **Note:** The Follow Me feature will only function for user extensions which have voicemail enabled.

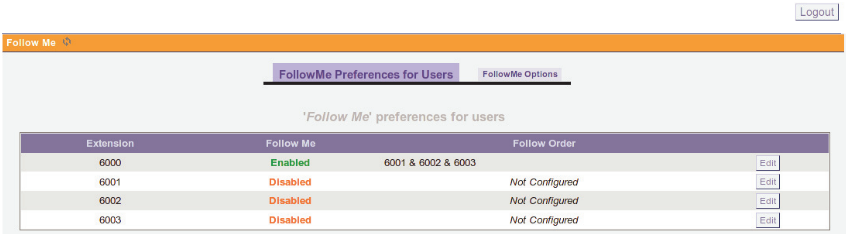


Figure 36: Follow Me

The following is an example scenario of using the Follow Me feature:

1. Derrick dials extension 6000 from his mobile phone to call Chuck.
2. Chuck’s office phone rings several times, but is not answered.
3. Derrick hears, “After the tone, say your name, and then press the pound key.”, followed by a beep tone.
4. Derrick says his name, and then presses the pound key.
   - If Derrick had not said his name and/or pressed the pound key, the call would have continued on to the next step as normal.
5. Derrick hears, “Thank you. Please hold while I try to locate the person you are calling.”

6. Then Chuck's mobile phone and home phone begin to ring simultaneously.
7. Chuck's android at home answers the phone and hears, "Incoming call from". Then it hears Derrick state his name. Then it hears, "Press 1 to accept this call, or 2 to reject it."
   - This occurs while Chuck's mobile phone continues to ring.
8. Chuck's android quickly hangs up the phone instead of pressing 1 or 2.
   - If Chuck's android had pressed 1, it would have begun speaking with Derrick, and Chuck's mobile phone would have stopped ringing.
   - If Chuck's android had pressed 2, the call would have been rejected, Chuck's mobile phone would have stopped ringing, and Derrick would have been transferred to Chuck's voicemail box.
   - If Chuck's android had not hung up the phone and not pressed anything, the message would have looped itself until either the ring timeout for the Follow Me number had been met or Chuck had answered his mobile phone, whichever would have come first.
   - If the ring timeout for the Follow Me number had been met while Chuck's android was listening to the accept/reject message, it would have been disconnected from the call, Chuck's mobile phone would have stopped ringing, and Derrick would have been transferred to Chuck's voicemail box.
   - If Chuck would have answered and accepted the call from his mobile phone while his android was listening to the accept/reject message, it would have been disconnected from the call, and Chuck would have begun speaking with Derrick.
9. Chuck answers his mobile phone and hears, "Incoming call from". Then he hears Derrick state his name. Then he hears, "Press 1 to accept this call, or 2 to reject it."

10. Chuck presses 1 to accept the call.
11. Lastly, Chuck begins speaking with Derrick.

    Note: If no one had answered and accepted the call, Derrick would have been transferred to Chuck's voicemail box.

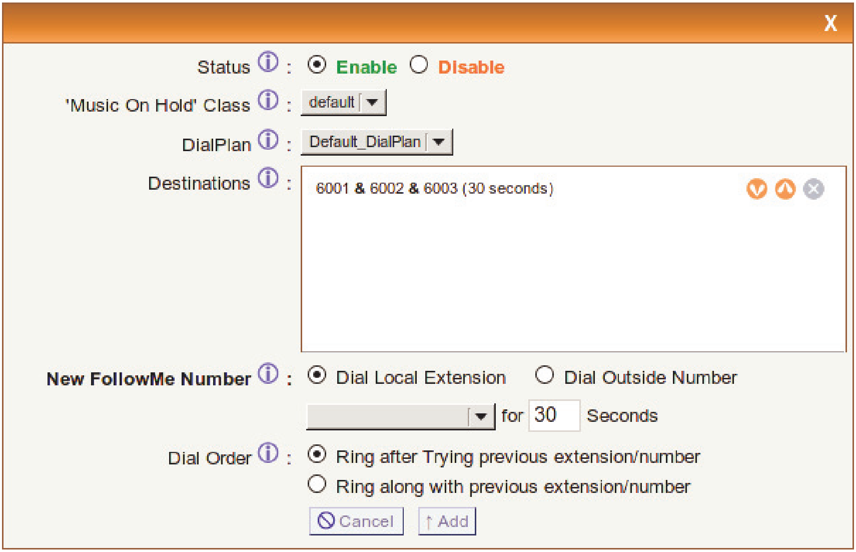


Figure 37: New Follow Me Definition

Use the following procedure as a guide to configure Follow Me for an user extension.

1. Click **Edit** for the user extension which you wish to configure. The edit box for the Follow Me definition will appear.
2. In order to enable the Follow Me feature, select **Enable** for the **Status** option.

3. Select the **'Music On Hold' Class** which you would like for the caller to hear while Follow Me attempts to reach you.
4. Select the **DialPlan** that should be used for dialing the Follow Me numbers. The dial plan associated with the user extension will be selected by default.
5. Click the **Add Follow Me Number** button in order to create a list of Follow Me numbers which will be dialed to reach the user. Upon doing so, additional options will appear near the bottom of the edit box.
6. Select **Dial Local Extension** if you would like to specify a local extension on the system to be dialed, or **Dial Outside Number** if you would like to specify an outside number to be dialed.

   Note: In order to properly match one of the patterns in your **Outgoing Calling Rules**, be sure to prepend the necessary digits when specyfing the outside number to be dialed.
7. Specify the number of seconds before the ring timeout occurs for the new Follow Me number. The ring timeout for the new Follow Me number is the total amount of time from when the Follow Me feature is initiated to when the call is accepted.

   Note: The ring timeout is not reset or cancelled when the prompt is played to allow someone to accept or reject the call. If the ring timeout is met while that prompt is being played, the call will be rejected and sent to voicemail.
8. Select the **Dial Order** in which this Follow Me number should be dialed to reach the user.

– Selecting **Ring after trying previous extension/number** will cause the defined Follow Me number to be called after the last entry listed in the **Destinations** box.

  Note: You must select **Ring after trying previous extension/ number** if no other Follow Me number exists in the **Destinations** box. Otherwise, you will be unable to save the Follow Me definition.

– Selecting **Ring along with previous extension/number** will cause the defined Follow Me number to be called simultaneously along with the last entry listed in the **Destinations** box.

9. Click **Add** to add this Follow Me number to the **Destinations** box, or **Cancel** to discard it.

You may reorder the entries in the **Destinations** box by using the up and down arrows located to the far right of each entry. If you wish to delete an entry, simply click the **X** located next to the up arrow.

Click **Save** to retain your changes or **Cancel** to discard them.

Then click **FollowMe Options** along the top to configure additional options for Follow Me.

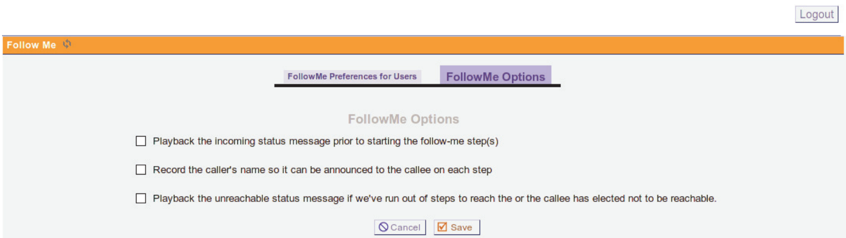


Figure 38: Follow Me Options

The following self-explanatory options can be enabled or disabled:

- Playback the incoming status message prior to starting the follow-me step(s).
- Record the caller's name so it can be announced to the callee on each step.
- Playback the unreachable status message if we've run out of steps to reach the callee, or if the callee has elected not to be reachable.

Click **Save** to retain your changes or **Cancel** to discard them. Then click **Apply Changes** to make the changes available.

Directory

The **Directory** settings page gives you the ability to set your preferences for the Dial by Names Directory. Dialing the directory extension gives callers the opportunity to search the telephone directory by first or last name.

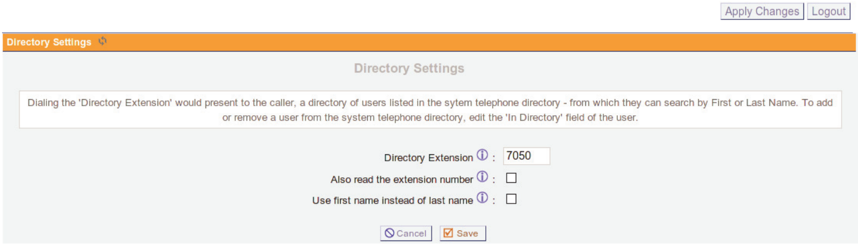


Figure 39: Directory Settings

On this page you specify the extension for dialing the system directory, as well as announcement and search preferences.

- **Directory Extension -** The extension to dial to access the names directory.
- **Also read the extension number -** Select this checkbox if you would like the extension number as well as user name to be read before presenting dialing options to the caller.
- **Use first name instead of last name -** Select this checkbox if you want to give callers the ability to search on first name instead of last name.

Click **Save** to retain your changes or **Cancel** to discard them. Click **Apply Changes** to make the changes available. To add or remove a user from the system telephone directory, edit the **In Directory** field of user's extension accessible from the **Users** page.

Call Features

The **Call Features** tab gives you the ability to configure feature codes, call parking, application maps, and dial options. These are explained in the following sections.

Feature Codes

The **Feature Codes** tab gives you the ability to define a keypress sequence which will initiate a blind transfer, attended transfer, call park, or call disconnect.

> **Note:** Feature codes will only function when two channels are answered and bridged together. They cannot be used while the remote party is ringing or in progress.

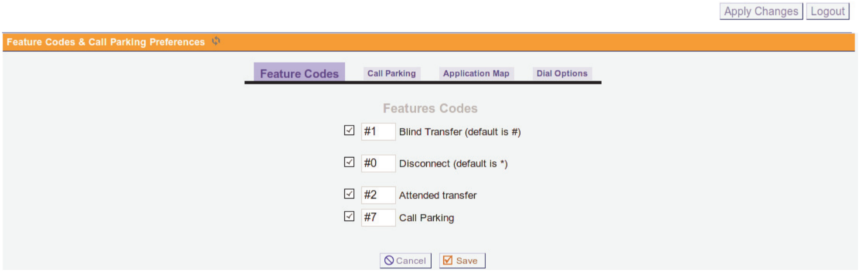


Figure 40: Feature Codes

The checkbox must be selected for any feature for which you wish to define a custom key sequence. The feature code options are described below.

> **Note:** Take care when specifying the key sequence for each feature code. The key sequence detection will stop as soon as it finds a possible match (*e.g.* If you have the key sequence for Blind Transfer set to ‘#’ and Attended Transfer set to ‘#2’, pressing ‘#2’ during a call will initiate a blind transfer instead of an attended transfer because DTMF detection will stop after pressing ‘#’. An example of properly configuring two feature codes starting with ‘#’ would be to set the key sequence for Blind Transfer to ‘#1’ and Attended Transfer to ‘#2’.)

Blind Transfer Feature Code

Specify the key sequence to initiate the **Blind Transfer** feature during an active call. The default key sequence is ‘#’. Blind Transfer may also be referred to as an unannounced, unsupervised, or cold transfer.

When initiated, this feature will prompt you to enter the destination extension for the blind transfer. You must then enter the destination extension within a few seconds, otherwise the blind transfer will be cancelled. After entering the destination extension within the alloted time, the calling party will be transferred to the destination extension without prior notification and the initiator of the transfer will be disconnected. The calling party’s Caller ID will be preserved when the call is transferred to the destination extension.

> **Note:** The **T Option** and/or **t Option** must be enabled under **Dial Options** in order for this to function.

Disconnect Feature Code

Specify the key sequence to initiate the **Disconnect** feature during an active call. The default key sequence is ‘*’.

When initiated, this feature will disconnect the active call.

> **Note:** The **H Option** and/or **h Option** must be enabled under the **Dial Options** tab in order for this to function.

Attended Transfer Feature Code

Specify the key sequence to initiate the **Attended Transfer** feature during an active call. A default key sequence is not defined for this feature. Attended Transfer may also be referred to as an announced, supervised, consult, full-consult, or warm transfer.

Initiating this feature will prompt you to enter the destination extension for the attended transfer. You must enter the destination extension within a few seconds, otherwise the attended transfer will be cancelled. After entering the destination extension within the alloted time, you will hear ringback if the destination extension is available. If the destination extension answers, you will be given the opportunity to announce the call transfer. Simply hang up the phone to complete the call transfer. If you hang up before the destination extension answers, the calling party will be transferred to the destination extension without prior notification (*i.e.* similar to blind transfer, but without CallerID preservation). If the destination extension does not answer and you do not hang up the phone, the attended transfer will be cancelled after 15 seconds. The calling party's Caller ID will not be preserved when the call is transferred to the destination extension.

> **Note:** The **T Option** and/or **t Option** must be enabled under the **Dial Options** tab in order for this to function.

Call Parking Feature Code

Specify the key sequence to initiate the **Call Parking** feature during an active call. A default key sequence is not defined for this feature.

Initiating this feature will prompt you with the first available parking extension. This is the number that can be dialed to retrieve the call from the parking lot. The caller will be immediately transferred to the specified parking extension, and the initiator of the call park will be disconnected. In order to retrieve the call, dial the parking extension that was specified by the Asterisk Appliance 50 prompt.

The amount of time that the call remains parked is determined by the number of seconds specified in the **Number of seconds a call can be parked for** field on the **Call Parking** tab. If the call is not retrieved within this time, the call will be redirected to the user that originally parked the call.

> **Note:** The **K Option** and/or **k Option** must be enabled under the **Dial Options** tab in order for this to function. This method of Call Parking may also be referred to as one step parking.

Click **Save** when you are done configuring this section. Then click **Apply Changes** to make these changes immediately available for new calls.

Call Parking

Call Parking is an Asterisk feature which allows a user to place a call on hold so that it can be taken off hold from another extension. Click the **Call Parking** tab from the **Call Features** page to configure this feature. The **Call Parking** page gives you the ability to define the call parking options which will enable use of this feature.

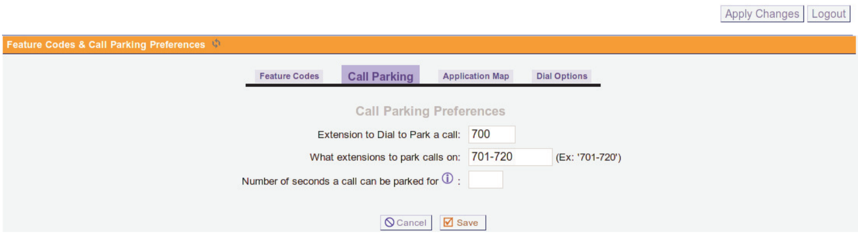


Figure 41: Call Parking Preferences

The following options must be configured to enable call parking.

- **Extension to Dial to Park a Call -** Specify the extension to call when transferring a call to hold or the "parking lot".
- **What Extensions to Park Calls On -** The extensions specified here will be the "parking lot" designations for the calls you place on hold. The call on hold will be retrieved by dialing one of these extensions.
- **Number of Seconds a Call Can Be Parked -** The number of seconds a call can be placed on hold. After the time has elapsed the call will ring the originating extension.

Parking a Call

You can park a call using either an analog or VoIP phone. To use an analog phone, hit the flash button, or quickly press the hook switch, wait for a dial tone, then dial the extension (700). With a VoIP phone, initiate the transfer, dial the call parking extension (*e.g.* 700), then complete the transfer (such as by pressing send). The method using a VoIP phone will vary depending on the phone.

At this point, the Asterisk Appliance 50 will prompt you with a number. The number it prompts you with is the number from the pool specified. This is the number that can be entered to retrieve the call. To retrieve the call, pickup a phone, and dial the parking number that was previously specified by the Asterisk Appliance 50 prompt. The amount of time that the call remains parked is determined by the number of seconds specified. If the call is not retrieved in this time, the call will be redirected to the user that originally parked the call.

> **Note:** In order to properly park a call, you must use attended transfer functions. Using a blind transfer function will not provide the parking number to the person parking the call. This makes recovery of the call impossible, except for the fall through timeout.

Application Map

The Application Map tab gives you the ability to define a keypress sequence which will execute a specific application, along with the application’s arguments. One example of using this feature would be to allow the caller or callee to playback a specific sound file on demand when pressing a predefined key sequence. Click **New Application Map** to define a new application map.

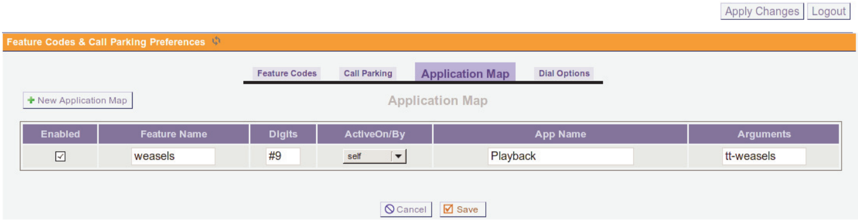


Figure 42: Application Map

The options associated with an application map are described below.

- **Enabled** - Select whether or not this application map is enabled.
- **Feature Name** - Specify a unique name to be associated with this application map.
- **Digits** - Specify the key sequence used to activate this feature.
- **Activate On / By** - Select which channel of the call that the application will be executed on, and which channel is allowed to activate this feature. The available settings are describe below.
  - **self** - Run the application on the same channel that activated this feature. This feature will be accessible by both the caller and callee.
  - **peer** - Run the application on the opposite channel from the one that has activated this feature. This feature will be accessible by both the caller and callee.
  - **self / caller** - Run the application on the same channel that activated this feature. This feature will be accessible by the caller only.
  - **peer / caller** - Run the application on the opposite channel from the one that has activated this feature. This feature will be accessible by the caller only.

- **self / callee -** Run the application on the same channel that activated this feature. This feature will be accessible by the callee only.
- **peer / callee -** Run the application on the opposite channel from the one that has activated this feature. This feature will be accessible by the callee only.
- **self / both -** Refer to the description for **self**.
- **peer / both -** Refer to the description for **peer**.

- **App Name -** Select the application to execute once the defined key sequence is detected.

  Note: The application map feature is not intended to be used for all Asterisk applications. Applications which are statically defined in the extensions.conf are executed by the PBX core. In contrast, applications which are dynamically called from an application map are executed outside of the PBX core. It is not appropriate to use any application which has any concept of dialplan flow when using an application map. Examples of this would be applications such as Macro, Goto, Background, and WaitExten.

- **Arguments -** Specify the arguments to be passed to the application defined in **App Name**.

  Note: Enabling the application map feature will cause the Asterisk Appliance 50 to remain in the media stream during all calls. This will occur regardless of whether two endpoints are configured to redirect their media stream from the Asterisk Appliance 50 to each other after the call setup has completed (*e.g.* two SIP phones with reinvite enabled).

Click **Save** when you are done configuring this section. Then click **Apply Changes** to make these changes immediately available for new calls.

Dial Options

The **Dial Options** tab gives you the ability to configure feature code permissions for the called party and the calling party. Either party can be allowed or restricted access to the transfer, hang up, and call parking feature codes.

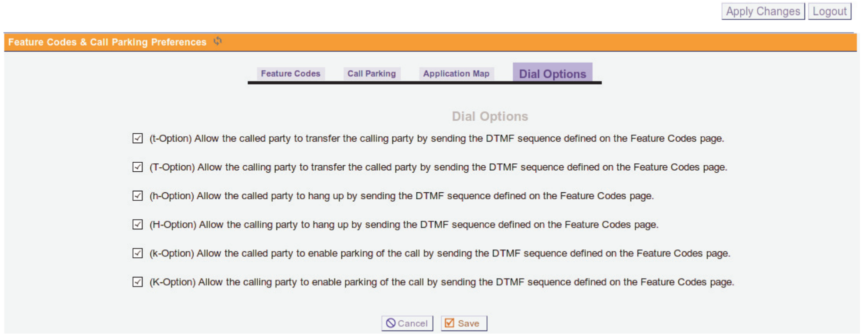


Figure 43: Dial Options

These permission options are explained below.

- **t Option -** Allows the called party to transfer the calling party by sending the DTMF sequence defined on the Feature Codes page.
- **T Option -** Allows the calling party to transfer the called party by sending the DTMF sequence defined on the Feature Codes page.
- **h Option -** Allows the called party to hang up by sending the DTMF sequence defined on the Feature Codes page.
- **H Option -** Allows the calling party to hang up by sending the DTMF sequence defined on the Feature Codes page.
- **k Option -** Allows the called party to enable parking of the call by sending the DTMF sequence defined on the Feature Codes page.
- **K Option -** Allows the calling party to enable parking of the call by sending the DTMF sequence defined on the Feature Codes page.

Click **Save** when you are done configuring this section. Then click **Apply Changes** to make these changes immediately available for new calls.

Voicemail Groups

A voicemail group gives you the ability to create a voicemail box that can be shared by any of the users on an Asterisk Appliance 50 system. A group message can thus be sent by dialing one extension and leaving a message. Click **Voicemail Group** to access the Voicemail Group page.

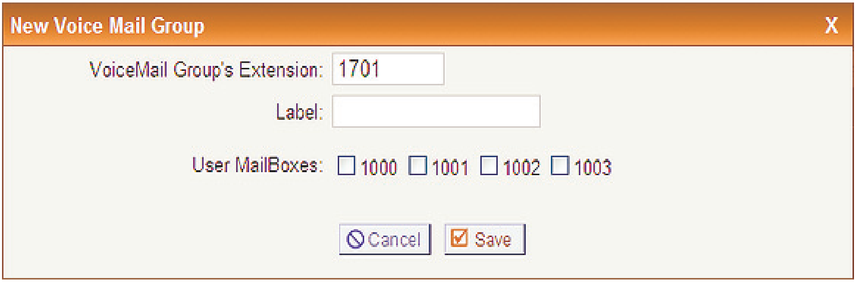


Figure 44: New Voicemail Group

Click **New Voicemail Group** to create a voicemail group.

- **Voicemail Group Extension -** Specify the group voicemail extension.
- **Label -** Specify a unique name for the voicemail group which can be referred to in the configuration of your Asterisk Appliance 50.
- **User Mailboxes -** Click the checkbox of each user voicemail box which should be part of the group voicemail box.

Click **Save** when you have completed the voicemail group definition, and then **Apply Changes** to make the voicemail box immediately available. You can either **Edit** or **Delete** the voicemail group from the main **Voicemail Group** page.

System Info

The general system information of the Asterisk Appliance 50 is displayed from this tab, as well as tabs for your network interfaces, disk usage, memory usage, and DHCP leases allocated on the LAN side by the Asterisk Appliance 50's DHCP server. If you do not have an NTP server specified, you can set your default time zone from within the General tab. The **S800i Config** tab describes the exact model information. This information will be useful for technical support.

Figure 45: System Information

Networking

The **Networking** page is used to configure your general network settings, as well as your Wide Area Network (WAN) settings, Local Area Network (LAN) settings, and Timezone settings.

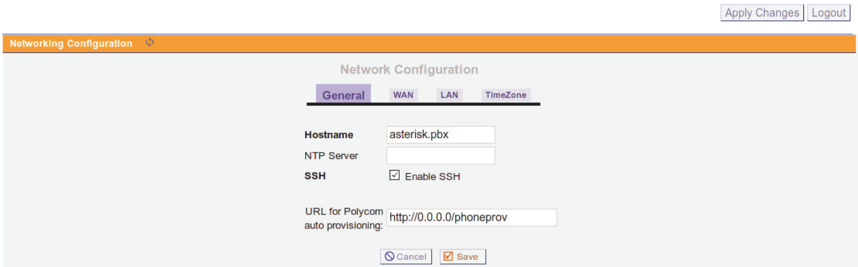


Figure 46: Networking

The General tab, which is the default selection on the Networking page, is used to specify the following settings:

- **Hostname -** The hostname assigned to the Asterisk Appliance 50. This name will be used to identify the Asterisk Appliance 50 on your network.
- **NTP Server -** This field gives you the ability to specify the URL or IP address of an NTP server. This is useful if you wish to regularly synchronize the Asterisk Appliance 50 time setting with that of an NTP server.

- **SSH -** Select the SSH checkbox to activate the SSH server on the Asterisk Appliance 50. The default root password is digium. Enabling this option will cause your unit to provide SSH access on both WAN and LAN interfaces, which can pose a security risk.

  Note: It is suggested that you change the default root password the first time you SSH into the Asterisk Appliance 50. Use the “passwd” utility from the shell to change the password. Changing the default password will increase security.
- **URL for Auto-Provisioning -** The URL specified in this field is used to enable auto-provisioning for Polycom phones. The default for this field is http://0.0.0.0/phoneprov. The 0.0.0.0 will resolve to the IP of the Asterisk Appliance 50 with the LAN IP for requests from the LAN ports, and the WAN IP for requests over the WAN port.

The **WAN** tab is used to specify the settings which will enable connection to the Internet, or to an internal, private network.

- **DHCP -** The DHCP setting enables the automatic assignment of an IP address to the Asterisk Appliance 50. This checkbox is selected by default.
- **Enable GUI on WAN Interface -** Select this checkbox only if you are certain you want to enable access to the Asterisk Appliance 50 GUI via the WAN interface.
- **Enable WAN Side Provisioning -** Select this option to enable the provisioning of Polycom phones connected through the WAN.

If you have difficulty obtaining an IP address dynamically, deselect the DHCP checkbox and specify the IP address, Subnet, Gateway, and DNS settings. This information should be available from your company network administrator or Internet Service Provider (ISP).

The **LAN** tab is used to specify the settings for your local network. A local network is usually a smaller network which is part of a WAN. The information specified here is used to access your Asterisk Appliance 50. The default IP address specified, 192.168.69.1, is used to access the AsteriskGUI. You can change this address to an IP address specified in the IP start and end ranges. In most cases, the default should be used.

The **Timezone** tab is used to specify the default timezone for your Asterisk Appliance 50. The time zone information is used to set the date and time on the Asterisk Appliance 50. The time zone files are located on the flash card which comes with your Asterisk Appliance 50. Select the appropriate timezone from the list. Click **Set as Default** to set the corresponding time zone as your default time zone. Clicking the **Update Timezones** button will download and install the latest timezone files from Digium’s website. You will need to restart your Asterisk Appliance 50 in order to complete setting the time zone as your default. To reboot your appliance, go to **Options**, **Reboot**, and click **Reboot Now.**

> **Note:** The time zone files are named after cities that adhere to the time zone you need.

G.729 Codec

The **G.729 Codec** page allows you to register and manage your G.729 codec license keys. The G.729 Codec is an industry standard algorithm that compresses and decompresses a digital audio stream. Applied to Voice over IP (VoIP) calls, G.729 compresses the audio data to use significantly less network bandwidth than a standard or uncompressed VoIP call. This compression allows for more calls to be carried without increasing network capacity and allows voice to travel on limited-bandwidth connections that would otherwise not support VoIP.

> **Note:** The Asterisk Appliance 50 can support a maximum of 8 simultaneous calls being transcoded with G.729. Calls using G.729 which are not being transcoded (pass-thru) do not count towards that total.

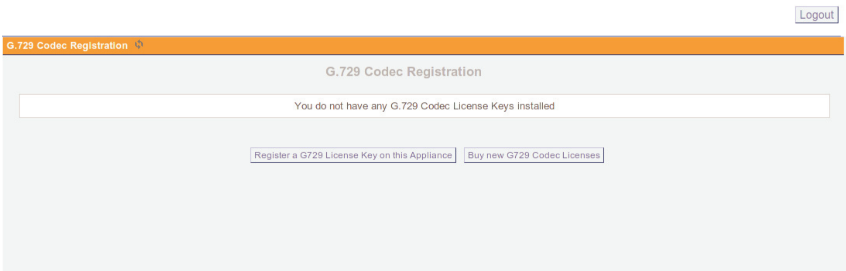


Figure 47: G.729 Codec Registration

Click **Register a G729 License key on this Appliance** to cause the Asterisk Appliance 50 to download the End User License Agreement (EULA) from a server at Digium. Read the EULA carefully. If you agree to its terms, click **I agree to the above License**.

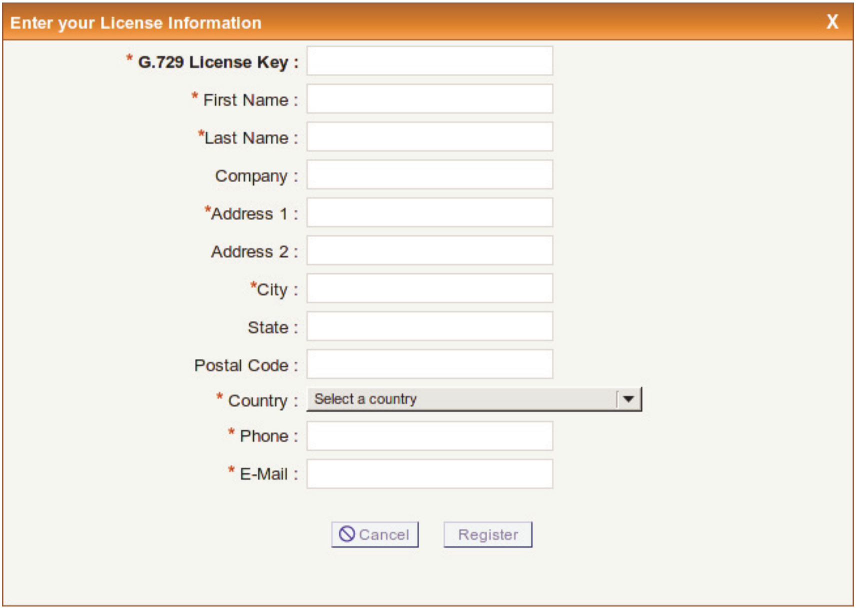


Figure 48: G.729 Codec License Information

Complete the G.729 Codec License Information form in full. The **G.729 License Key** field should begin with “G729-”. Then click **Register**. Your registration information will be securely sent to Digium’s registration server.

The registration process may take a few minutes to complete. A message box will appear to let you know once the firmware has finished updating. You must reboot your Asterisk Appliance 50 in order for these changes to take effect. Click **Options** on the left menu, select the **Reboot** tab, and then click **Reboot Now** to reboot your appliance. Rebooting your Asterisk Appliance 50 will terminate any active calls.

If the registration process fails, please confirm that you have entered the G.729 license key correctly, and that a firewall is not blocking the Asterisk Appliance 50 from communicating with Digium’s registration server on TCP port 443.

If you do not currently own a G.729 codec license, click **Buy new G729 Codec Licenses** to be directed to a page where you can purchase G.729 codec licenses.

Backup

This is a housekeeping tab which allows you to back up your Asterisk Appliance 50 configuration to preserve your changes. To create a backup, click **Create New Backup**, specify a file name (*e.g.* the backup date), and select whether you want to also backup voicemails and custom prompts. You can then download a previously created backup, restore from the backup, delete the backup file, or upload a backup from another machine.

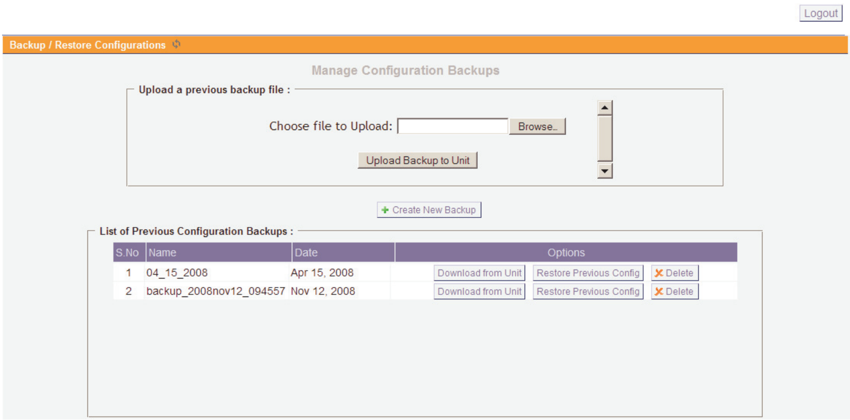


Figure 49: Backup Page

Update

The Updates tab provides an interface for downloading or uploading newer Asterisk Appliance 50 firmware images, and for downloading newer Polycom firmware and bootrom images to the Asterisk Appliance 50. Customers with an active Service Subscription may visit the Digium.com website to register their Asterisk Appliance 50, activate their Service Subscription, and download software updates for the Asterisk Appliance 50.

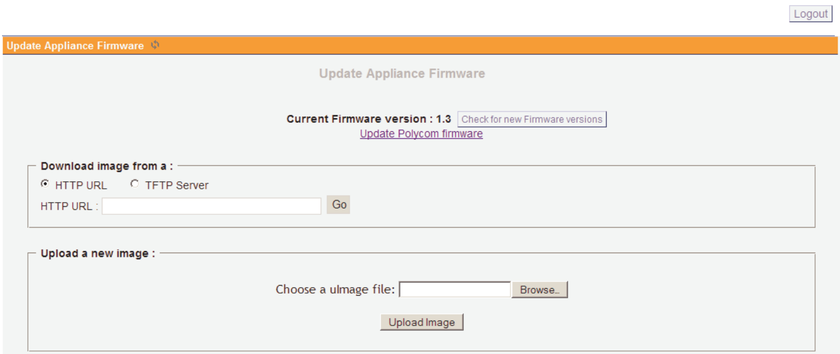


Figure 50: Asterisk Appliance 50 Update

There are two interfaces for putting a new Asterisk Appliance 50 firmware image on the Asterisk Appliance 50. The first section provides the user the ability to specify a location from which the Asterisk Appliance 50 will connect and download the updated software. The user may specify an absolute HTTP location such as: http://company.com/downloads/software.img, or the address and filename on an accessible TFTP server. The second section provides a web-based interface for uploading software updates. Here, the user may click **Browse**, select a local copy of the new software, and click **Upload**.

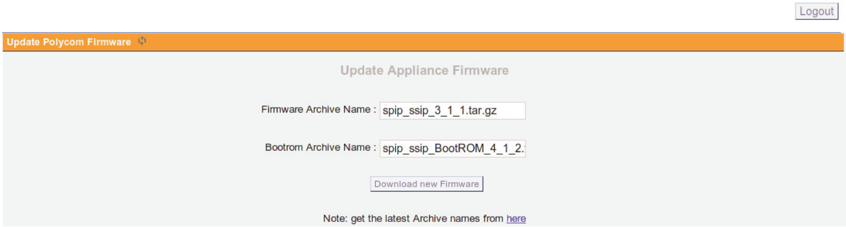


Figure 51: Polycom Update

Click **Update Polycom firmware** to update the Polycom firmware or bootrom images on the Asterisk Appliance 50. A link is provided under the **Download new Firmware** button to get the latest archive names. The firmware and bootrom archive names must be specified exactly as they are at that link in order for the update to complete successfully. Click **Download new Firmware** after you have specified the latest archive names in the firmware and bootrom fields.

The download process may take a few minutes to complete, and even longer if your network bandwidth is limited. A message box will appear once the firmware has finished downloading.

The Polycom firmware and bootrom images will be installed during the next reboot cycle of the Asterisk Appliance 50.

Click **Options** on the left menu, select the **Reboot** tab, and then click **Reboot Now** to reboot your appliance. Rebooting your Asterisk Appliance 50 will terminate any active calls.

> **Note:** The next reboot cycle will be increased by approximately 5 minutes during the installation process. In addition, you must reboot your Polycom phones in order for them to download the new firmware and bootrom images from the Asterisk Appliance 50.

Options

The options tab provides several options which allow you to change the password for your AsteriskGUI logon, modify local extension and agent settings, as well as reboot the Asterisk Appliance 50. The Advanced tab allows you to enable or disable advanced options. The basic options are displayed by default. Please refer to the **Advanced Options** section for a description of the advanced options.

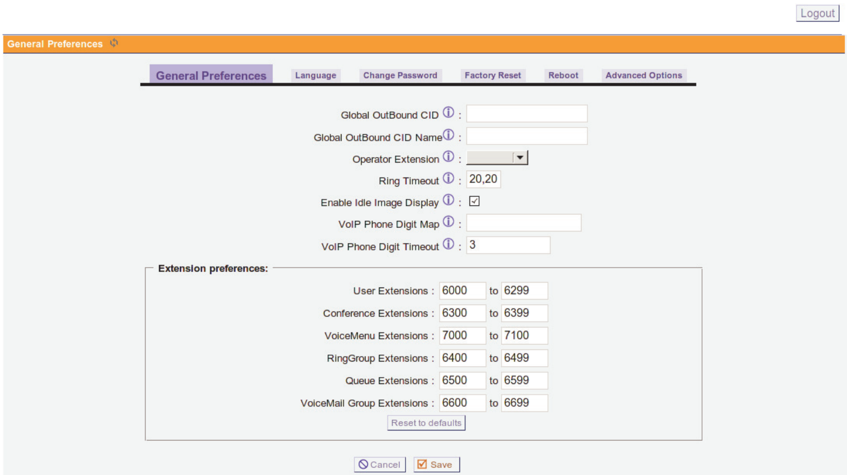


Figure 52: Asterisk Appliance 50 Options

General Preferences

The **General Preferences** tab gives you several useful global settings for your Asterisk Appliance 50.

- **Global Outbound CID -** This setting specifies the default global CallerID that is used for all outgoing calls when no other CallerID of a higher priority is specified.
- **Global Outbound CID Name -** This setting specifies the default CallerID name that is used for all outgoing calls. You may wish to set this to your company's name. Leave this value blank if you want the user's CallerID name to appear on outbound calls.
- **Operator Extension -** Select the user extension from the drop-down list which will be dialed when a user or caller presses "0" to exit voicemail. It is also available as a Voice Menu option.
- **Ring Timeout -** Specify the number of seconds to ring a device before sending a call to a user's voicemail box or Follow Me numbers.
- **Enable Idle Image Display -** Select this option to enable the display of an image on a phones LCD display when the phone is idle.
- **VoIP Phone Digit Map -** This option gives you the ability to define a global digit mapping string compatible with RFC 3435, section 2.1.5, to be used with VoIP phones provisioned by this system. There is no default setting, and this option does not sync with the dialplan assigned to an individual user. The following examples should assist in writing an acceptable digit mapping string.
  - **[2-9]11 -** Calls beginning with digits 2-9, followed by digits 11, are dialed immediately.
  - **0T -** Calls beginning with digit 0 are dialed after the Digit Timeout is reached.

- **+011xxx.T -** Calls beginning with the + character, followed by 011 digits, and then at least three more digits before any arbitrary number is matched, are dialed after the Digit Timeout is reached.
- **0[2-9]xxxxxxxxx -** Calls beginning with 0, followed by any digit from 2-9, followed by 9 more digits, are dialed immediately.
- **+1[2-9]xxxxxxxx -** Calls beginning with the + character, followed by 1, followed by any digit from 2-9, followed by 8 more digits, are dialed immediately.
- **[2-9]xxxxxxxxx -** Calls beginning with any digit from 2-9, followed by 9 more digits, are dialed immediately.
- **[2-9]xxxT -** Calls beginning with any digit from 2-9, followed by three more digits, are dialed after the Digit Timeout is reached.
- These examples would be represented in this option entry box as: **[2-9]11|0T|+011xxx.T|0[2-9]xxxxxxxxx|+1[2-9]xxxxxxxx|[2-9]xxxxxxxxx|[2-9]xxxT**
  Each entry is separated by the | character. For more information, please refer to RFC 3435.

- **VoIP Phone Digit Timeout -** This option specifies the number of seconds the phone will wait after a digit is dialed before trying to establish a call when a digit map pattern has not been matched. This must be defined as an integer.
- **Extension Preferences -** This section gives you the ability to define the numerical range for all extension types.

Language

The **Language** tab gives you the ability to specify the default language for all prompts for phone to phone, inbound, and outbound calls. If a soundpack selection is made, but the soundpack is not already installed, the soundpack will be downloaded from the Digium website. English, Spanish, and French prompts are loaded by default.

Change Password

The **Change Password** tab gives you the ability to change your administrator password.

Factory Reset

The **Factory Reset** tab gives you the ability to reset your Asterisk Appliance 50 to the factory defaults. **Warning:** If you reset your Asterisk Appliance 50 to factory defaults, you will lose all configuration changes. Be sure to make a backup of your configuration before resetting your Asterisk Appliance 50.

Reboot

The **Reboot** tab gives you the ability to reboot your Asterisk Appliance 50. Some configuration changes you make may require a system reboot. **Warning:** Rebooting the appliance will terminate all active calls.

Advanced Options

There are several advanced options which can be made accessible from the **Options** page. This gives advanced users with a background in Asterisk the ability to refine the Asterisk Appliance 50 configuration. Clicking **Show Advanced Options** provides additional advanced options under certain existing menu pages, and also activates several advanced menu items on the left hand sidebar.

> **Note:** Digium does not provide support for the options which are made accessible by selecting **Show Advanced Options** nor for bugs discovered in the options which are made accessible by selecting **Show Advanced Options**. If your unit becomes inoperable due to the editing of an option which is made accessible by selecting **Show Advanced Options**, Digium Technical Support will request that you reset your unit to Factory Default configuration.

> **Note:** Any changes made on the advanced options pages must be activated by clicking **Apply Changes** at the top of the GUI.

The following is a list of the advanced menu items which will be made accessible from the left hand sidebar after selecting **Show Advanced Options**:

- **Active Channels -** This page displays the active channels on the PBX, along with options to transfer a call to a user on the system or to hang up a call.
- **Bulk Add -** This page gives you the ability to add multiple users to the system in one easy step **--** import from a CSV (comma-separated values) file or create a range of extensions.
- **File Editor -** This page lets you edit any Asterisk configuration file within the GUI, as well as create a new configuration file.
- **Asterisk CLI -** The Asterisk CLI is a command line interface which can be used for issuing any Asterisk command or series of commands. The results of the commands are displayed in the pane below the command line field. Enter **Help** in the command line field for a list of commands.
- **SIP Settings -** The advanced SIP settings can be configured from this page.
- **IAX Settings -** The advanced IAX settings can be configured from this page.

Chapter 4
Troubleshooting

Where can I find answers to additional questions?

There are several places to inquire for more information:

1. Digium Technical Support (+1.256.428.6161), or Toll Free in the U.S. (1.877.344.4861), is available 7am-8pm Central Time (GMT -6), Monday - Friday.
2. Asterisk users mailing list (asterisk.org/lists.digium.com).
3. IRC channel **#asterisk** on (irc.freenode.net).

Subscription Services Program

Digium is dedicated to supporting your Asterisk system by offering full technical support through our Subscription Services Program. The Subscription service covers the Asterisk Appliance 50. Through this program, you can be at ease knowing that your business will always have access to the Asterisk experts. Pricing on Subscription Services may be obtained from your nearest reseller or you may call Digium Sales for referral to your nearest reseller at +1.256.428.6000. You can also visit http://www.digium.com/sales.

Appendix A
Pin Assignments

This appendix provides pin assignments for the Asterisk Appliance 50 unit.

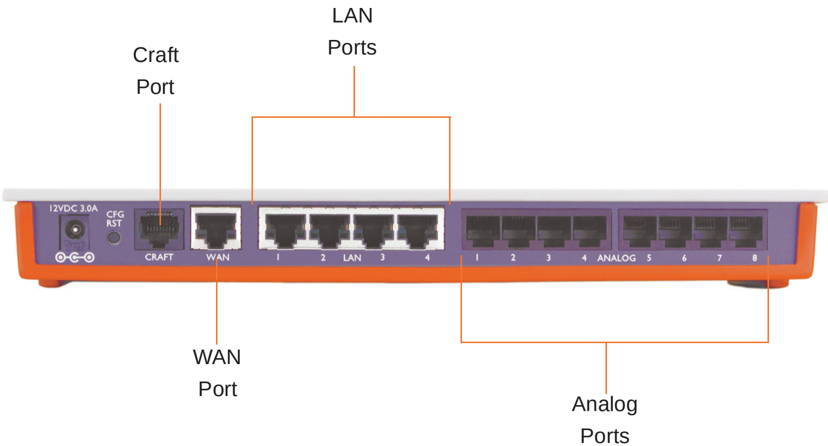
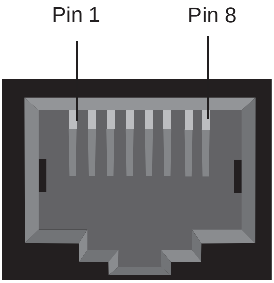


Figure A-1: Back Panel Ports

All eight analog ports on the Asterisk Appliance 50 are 8-pin RJ11 ports. The pin assignments are identified in Table A-2.

Table A-1: CRAFT Port Pinout

Diagram	Pin	Description
	1	Ground (Connect to DB9 pin 5)
	2	Unused (Leave Open)
	3	Primary RxD (To AA50) (Connect to DB9 Pin 3)
	4	Open
	5	Tx (From AA50) (Connect to DB 9 Pin 2)
	6	CTS (To AA50) (Connect to DB9 Pin 7)
	7	Open
	8	RTS (From AA50) (Connect to DB9 Pin 8)

The CRAFT port serial parameters are 57600 8N1 (57600 bits per second, 8 data bits, no parity, 1 stop bit), with hardware flow control = no, software flow control = yes.

Table A-2: RJ11 Analog Port Connector

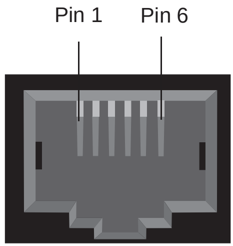
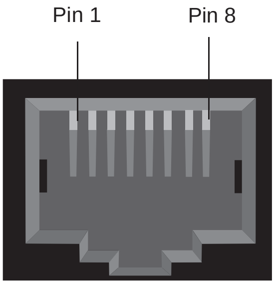
Diagram	Pin	Description
	1	Unused
	2	Unused
	3	Tip
	4	Ring
	5	Unused
	6	Unused

Table A-3: LAN & WAN Ethernet Port Pinouts

Diagram	Pin	Description
	1	Rx Receive Negative
	2	Rx Receive Positive
	3	Tx Transmit Negative
	4	Unused
	5	Unused
	6	Tx Transmit Positive
	7	Unused
	8	Unused

Appendix B
Specifications

This appendix provides specifications, required environmental conditions, and maximum power consumption for the Asterisk Appliance 50 unit.

Physical.

Size: 11.7” × 7.5” × 1.72” (29.72 x 19.05 x 4.37 cm)
Weight: Full Assembly 1.4 lbs (635g)

Interfaces.

LAN Ports - Quad RJ45 10/100baseT
WAN Port - RJ 45 10/100baseT
Analog Ports - Octal RJ11
Craft Port - RJ45, 57600 8N1
DC Power - 6.3mm O.D., 2mm pin; 12V 3A center positive
CompactFlash - Type 1

Environment.

Temperature: 0 to 40° C (32 to 104° F) operation
-20 to 70° C (4 to 158° F) storage
Humidity: Up to 90% non-condensing

Table B-4: Maximum 12V Power Consumption

Item	Power
Total Each FXS port in use with 3REN load	36 Watts 1.5 Watts

Note: Power consumption is determined by the number of analog phones connected to the FXS ports and the REN rating of the phones.

Appendix C
Asterisk Appliance 50 (AA50) License Agreement

DIGIUM END-USER PURCHASE AND LICENSE AGREEMENT FOR THE ASTERISK APPLIANCE 50
December, 2008

IMPORTANT – PLEASE READ CAREFULLY

1. This Digium End-User Purchase and License Agreement (the "Agreement") is a legal agreement between Digium and the licensee, purchaser and end user respectively (hereinafter, “you”, “You” or “your”) of the Digium distribution media, embedded software and related documentation (the "Software"), Digium services (“Services”), and any Digium computer electronics (“Hardware”), entitlements granted pursuant to a Subscription Agreement, and related manuals (collectively the "Products"). By downloading or installing the Software or installing the Hardware, you agree to and accept the terms and conditions of this Agreement. If you do not accept, or are not authorized to accept the terms and conditions of this Agreement, then you should not install the Software or Hardware and should remove any installed Software and Hardware from your computer.

2. **GRANT OF LICENSE.** Subject to the terms and conditions of this Agreement and Exhibit B, Digium grants you a non-exclusive, non-sublicenseable, non-transferable license to use the Software for internal business purposes and not for resale, sublicense, leasing, or providing hosted services to third-parties. “Software” shall include

any upgrades, updates, bug fixes or modified versions (“Upgrades”) or backup copies of the Software supplied to you by Digium or an authorized reseller, provided you hold a valid license to the original Software and have paid any applicable fee for Upgrades. Notwithstanding the foregoing, Digium acknowledges that certain components of the Software are licensed for distribution under so-called “open source” software licenses (“Open Source Components”). Digium provides a list of these Open Source Components for the initially distributed version of the Software in Exhibit B, along with their respective licenses. To the extent the terms of the license applicable to an Open Source Component prohibit any of the restrictions in this Agreement with respect to such Open Source Component, such restrictions will not apply to such Open Source Component.

3. **RESERVATION OF RIGHTS.** Except for the limited license rights expressly granted in this Agreement, Digium reserves all rights in and to the Software and any modifications thereto, including title, ownership, trademark, copyright, patent, trade secret and any other rights and interests. You will own only the Hardware (exclusive of Software embedded in the Hardware) and the physical media on which the Software and associated documentation are reproduced and distributed. If you allow an existing Services or Subscription Agreement plan to lapse, you may be required to pay retroactively annual fees for all Products from the date of the lapse in order to reinstate such Services. Products that are provided or sold as demo or evaluation units, or are marked with the legend “For Evaluation Only” or a similar notation, are not authorized for commercial business use in production or deployment, but are made available only for demonstration or evaluation purposes, and may not be resold or

transferred to any third party without prior written permission from Digium.

4. **EMERGENCY CALLS.** You understand and acknowledge that the Products may be used to implement, supplement, or replace telephone systems and telecommunications services, and that in some cases, certain government regulations may apply to their implementation or use; and compliance with such regulations is your sole responsibility. You understand and acknowledge that users of the system on which you install the Products may attempt to use that system to place emergency calls. You acknowledge and agree that: the Products must be properly configured for your system or application; that the nature of the Products and any networks they may operate upon allow many possible configurations; that such configuration may be beyond the scope of the documentation supplied with the Products; and that specialized experience and training may be required to properly configure the Products. You acknowledge and agree that it is your sole responsibility to ensure that the Products and associated networks and systems are implemented and configured such that emergency calls are properly handled, and that any system or application based on the Products complies with all applicable laws and regulations. You acknowledge and agree that telephone and telecommunications systems can be complex and must be installed, implemented, and configured by the appropriate technically qualified personnel, and that you or your authorized agents have the qualifications necessary to properly implement and configure the Products to handle emergency calls, if applicable. You further acknowledge and agree that it is your sole and ongoing responsibility to ensure the proper operation of any emergency calling system based on the Products, including, but not limited to: initially and regularly testing the operation of the Products,

including testing the operation with emergency services; notifying and training all users of any system on which the Products are installed how to use the system for emergency calls; and notifying such users of any and all limitations of your configuration and implementations of the Products and any network or system the Products are used on or with. By using the Products under this Agreement, you explicitly release Digium from any warranty, duty, liability, or obligation to train you or any users of your system regarding the proper configuration, operation, or use of the Products or any system or network they are used in conjunction with on which it is installed; to ensure that your configuration, implementation, or use of the Products provides for the proper handling or routing of emergency calls; or to ensure that your use of the Products is in compliance with any applicable laws and regulations.

5. LIMITED WARRANTY

5.1 Limited Warranty - Software. Digium warrants that the Software will substantially conform to Digium’s specifications as set forth in the applicable documentation, at and from the date of original retail purchase of the Software for a period of ninety days provided that the Software is properly installed and used as contemplated in its documentation. Digium’s sole obligation and your sole remedy for a breach of the foregoing warranty shall be to replace the non-conforming Software with software substantially conforming to Digium’s specifications or to refund fees as described below. Digium further warrants that the media on which Digium delivers the Software will be free of physical defects for a period of ninety days or will be replaced by Digium. Except as otherwise agreed by Digium in writing, the replacement Software is subject to the terms and conditions of this Agreement. If Digium determines in its sole discretion that a material non-conformance cannot be corrected, or that it is not practical to replace the nonconforming Software, the price paid by the original licensee for the Software will be refunded by Digium to you and the license grant automatically terminated, in which case you will promptly return to Digium or destroy the Software and any copies in your possession.

5.2 Limited Warranty - Hardware. Digium warrants that the Hardware will be free from material defects in workmanship and materials from the date of original purchase by you for a period of two years. Digium's sole obligation shall be to repair or replace the defective Hardware during the warranty period at no charge to you or to refund as described below. Such repair or replacement will be performed by Digium or its authorized representative. Digium may in its sole discretion replace the defective Hardware (or any part thereof) with a reconditioned product determined by Digium as materially equivalent or better than the defective Hardware. Repaired or replacement Hardware continues tolling the original warranty period from the date of original retail purchase.

5.3 Limited Warranty – Services AND SUBSCRIPTION AGREEMENTS. Digium warrants it will perform Services and/or deliver entitlements (which shall be defined in a Subscription Agreement) to you in a manner consistent with reasonably applicable industry standards. As your sole remedy and Digium's sole liability for breach of the foregoing warranty, Digium will re-perform or remedy at no charge to you any non-conforming performance reported by you within ten days of Digium's receipt of notification of breach of the foregoing warranty. Digium shall include technical support and/or deliver the Subscription Agreement entitlements for the Products in accordance with the policies, terms and conditions set forth on the Digium web site for the Products you purchase. You may elect to renew your support or Subscription Agreement from Digium at Digium’s then current rates unless such Products and associated Services are discontinued in Digium’s sole discretion. You will be informed of the subscription entitlements, and terms and conditions (Subscription Agreement) during the Product registration process. A current copy of the entitlements may be viewed on Digium's website and is incorporated in this Agreement by reference. Digium may amend the Subscription Agreement at any time. You are responsible for reviewing Digium's website for changes in the terms and conditions of the Subscription Agreement. In the case where Product registration is required in order to activate a Subscription Agreement, that Subscription Agreement shall be considered activated and legally binding beginning upon the date of Product registration. However, should you fail to register the product within thirty (30) days of your receipt of Product, the Subscription Agreement shall be considered activated and legally binding beginning on the thirty first (31st) day following your receipt of the Product.

5.4 Warranty Disclaimers. The limited warranties provided by Digium do not cover (i) Products that have been subjected to misuse, tampering, modification, experimentation, alteration, negligence, faulty installation; (ii) Products with the model or serial number altered, tampered with, or removed; (iii) Initial installation, installation and removal of the Product for repair, and shipping costs; (iv) Configuration of the Product; (v) Products that are provided or sold as demo or evaluation units, or are marked with the legend “For Evaluation Only” or a similar notation; (vi) Damage that occurs in shipment due to act of God, failures due to power surge, and cosmetic damage; (vii) Any hardware, software, firmware or other materials or services provided by anyone other than Digium; or (viii) Repair by anyone other than Digium or Digium's authorized representative. Unless under a current Services contract with Digium, you shall not upgrade or update Product using software of any origin except commercial versions of the Software released to you by Digium or its authorized distributors from time to time. the product is provided without any implied warranties of any kind whatsoever, including, without limitation, any implied warranty of merchantability, fitness for a particular purpose and non-infringement. Except as expressly covered under the limited warranty provided herein, digium disclaims all laibility as to the quality, selection and performance.

6. **Return Material Authorization.** If returning a Product under warranty, you must include with the Product a written description of the Hardware defect or Software nonconformance in sufficient detail to allow Digium to confirm such determination. Prior to returning a Product under warranty to Digium, the original purchaser must obtain a Return Material Authorization ("RMA") in accordance with the Digium policy in effect at the time as published on the Digium web site. You must package the warranted Product in the original shipping package to ensure that it will not be damaged in transit, with the RMA number prominently marked on the outside of the package. No manuals or accessories should be returned to Digium, and Digium will only refund or replace the defective portion of the Product, but will not ship back any other material. Digium will ship repaired or replaced Products to you via UPS Ground or any common carrier selected by Digium, with shipping charges prepaid. You may request and prepay expedited shipping. Digium has no responsibility for packages that

are lost in transit to Digium. Digium will not be obligated to accept physical return of any Software.

7. **LIMITATION OF LIABILITY.** To the maximum extent permitted by law, Digium is not liable under any contract, negligence, strict liability or other legal or equitable theory for any loss of use of the Products, inconvenience or indirect damages of any character, whether special, incidental or consequential (including, but not limited to, loss of revenue or profit, work stoppage, computer failure or malfunction, failure of connected equipment or programs, loss of information or data or loss of goodwill) resulting from the use of the Products, relating to warranty service, or arising out of any breach of this Agreement, even if Digium has been advised of the possibility of such damages. The sole remedy for a breach of the foregoing limited warranty is repair, replacement or refund of the defective or non-conforming product. The maximum liability of Digium under this Agreement is limited to the purchase price of the applicable Products. The foregoing express written warranties and remedies are exclusive and in lieu of any other warranties or remedies, express, implied or statutory.

8. **PROPRIETARY WORKS.** The Products contain trademarks, trade secrets and/or copyrighted materials of Digium or its suppliers.

   8.1 You agree not to reverse engineer, decompile, or disassemble the Software, except to the extent such restriction is expressly prohibited by applicable law. You shall not disclose or make available such trade secrets or copyrighted material in any form to any third party nor remove any trademark notices, copyright notices, or licensing terms from the Software or any components therein.

8.2 You will not, without Digium written consent, use the name, trademarks, trade names or logos of Digium, or the name of any product or service of Digium, in any manner. If Digium grants you a right to use the aforementioned, you will do so only in strict compliance with Digium trademark policies.

8.3 You agree not to fork any version of the components of Software licensed to you by Digium or promote or contribute to any fork of the components of Software licensed to you by Digium. For purposes of this Agreement, "fork" means a royalty free GNU General Purpose License (“GPL”) or GPL-compatible distribution of the Software that is independent of the Digium-maintained Software code base.

9. **TERMINATION.** This Agreement shall terminate upon either destruction of the Products or return of the Products by you to Digium. In the event of a breach of the scope of use permitted by the grant in Section 2, or if you do not comply with other materials terms and conditions of this Agreement, Digium shall have the right to immediately terminate this Agreement, in which case you must promptly destroy or return all Products to Digium. Notwithstanding the foregoing, the provisions of Sections 5.4, 7, 8, 9, 10, 11, 12, 13 and 14 shall survive termination of this Agreement.

10. **EXPORT RESTRICTION.** You acknowledge that the Software, with the possible exception of certain third-party components, is of U.S. origin. You agree to comply with applicable domestic and international laws that apply to the Software, including the U.S. Export Administration Regulations and other limitations issued by

U.S. and foreign governments.

11. **TRANSFER AND ASSIGNMENT.** This Agreement and the rights and obligations under it are not assignable by you without the prior written approval of Digium, voluntarily or by operation of law. Any attempt by you to assign this Agreement without such approval shall be void. This Agreement shall inure to the benefit of the successors and assigns of Digium. Notwithstanding the foregoing, you may move the Software to different internal computers to the extent consistent with the scope of license you have purchased to the Software.

12. **U.S. GOVERNMENT USERS.** The Software and documentation qualify as “commercial items” as defined at 48 C.F.R. 2.101 and 48 C.F.R. 12.212. All Government users acquire the Software and documentation with only those rights herein that apply to non-governmental customers of Digium.

13. **GOVERNING LAW AND JURISDICTION.** This Agreement is to be construed in accordance with and governed by laws of the State of Alabama, excluding its conflict of law provisions. Digium and you agree to submit to the personal and exclusive jurisdiction of, and agree that venue is proper in, the Alabama State or Federal Courts located in the County of Madison, Alabama, for any such legal action or proceeding. The United Nations Convention on International Sale of Goods, the application of which is expressly excluded, does not govern this Agreement.

14. **ENTIRE AGREEMENT.** This Agreement and Exhibit B constitute the entire understanding between the parties relating to the subject

matter hereof and supersede all prior writings, negotiations or understandings with respect thereto. The provisions of this Agreement shall take precedence over any conflicting terms in any subsequent purchase order, documentation or collateral. The parties agree that this Agreement may be executed electronically and that electronic copies of this Agreement shall be binding upon the parties.

EXHIBIT B – OPEN SOURCE COMPONENTS

1. **OPEN SOURCE LICENSES.** Digium acknowledges that the components of Software listed in this Exhibit are licensed for distribution under so-called "open source" distribution licenses, and includes the licenses for those components in compliance with the license terms.

   You may obtain distribution media containing the source code for these components from Digium by contacting Digium at:

   Postal Service:

   Digium, Inc.
   Sales Department
   445 Jan Davis Drive NW
   Huntsville, AL 35806

   Electronic mail:
   http://www.digium.com/sales

   Via either means, you may order product code 8AA50GPLCD, which has a unit cost of US$1.00, and will be shipped to you after payment of shipping handling charges (US$20.00 for United States destinations and US$50.00 for all other destinations). Orders placed via electronic mail must be paid via a Digium-accepted credit card; orders via postal service can be paid either via credit card or a check drawn on a United States bank.

2. **GNU GENERAL PUBLIC LICENSE VERSION 2 COMPONENTS.** The following components are licensed under version 2 of the GNU General Public License (included as Exhibit B.1).

Linux® kernel	libnetold
ext2fs	uuid
libipq	libiptc
busybox	gdbserver
e2fsck	mke2fs
iwconfig	inetd
ssmtp	dnsmasq
dhcpcd	version
expand	

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uClibc	gettext
libiconv	gmime
glib	

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ncurses	Exhibit B.2
dropbear	Exhibit B.3
OpenSSL	Exhibit B.4
ftp	Exhibit B.5
ntp and ntpdate	Exhibit B.6
zlib	Exhibit B.7
ss and et libraries	Exhibit B.8
libpcap	Exhibit B.10
tcpdump	Exhibit B.11

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loginrec.h
atomicio.h
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Jean-loup Gailly
jloup@gzip.org
Mark Adler
madler@alumni.caltech.edu

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Appendix D
Glossary and Acronyms

ANSI *American National Standards Institute*

An organization which proposes and establishes standards for international communications.

asynchronous

Not synchronized; not timed to an outside clock source. Transmission is controlled by start bits at the beginning and stop bits at the end of each character. Asynchronous communications are often found in internet access and remote office applications.

attenuation

The dissipation of a transmitted signal's power as it travels over a wire.

bandwidth

The capacity to carry traffic. Higher bandwidth indicates the ability to transfer more data in a given time period.

bit

The smallest element of information in a digital system. A bit can be either a zero or a one.

bps *bits per second*

A measurement of transmission speed across a data connection.

broadband

Broadband transmission shares the bandwidth of a particular medium (copper or fiber optic) to integrate multiple signals. The channels take up different frequencies on the cable, integrating voice, data, and video over one line.

channel

A generic term for an individual data stream. Service providers can use multiplexing techniques to transmit multiple channels over a common medium.

Cat5

Category of Performance for wiring and cabling. Cat 5 cabling support applications up to 100 MHz.

Cat5E

Category of Performance for wiring and cabling. Category 5 Enhanced wiring supports signal rates up to 100 MHz but adheres to stricter quality specifications.

CLEC *competitive local exchange carrier*

A term for telephone companies established after the Telecommunications Act of 1996 deregulated the LECs. CLECs compete with ILECs to offer local service. See also *LEC* and *ILEC*.

CO *central office*

The CO houses local switching equipment. All local access lines in a particular geographic area terminate at this facility (which is usually owned and operated by an ILEC).

CPE *customer premises equipment*

Terminal equipment which is connected to the telecommunications network and which resides within the home or office of the customer. This includes telephones, modems, terminals, routers, and television set-top boxes.

DS0 *Digital Signal, Level 0*

A voice grade channel of 64 Kbps. The worldwide standard speed for digitizing voice conversation using PCM (Pulse Code Modulation).

DS1 *Digital Signal, Level 1*

1.544 Mbps in North America (T1) and Japan (J1) -up to 24 voice channels (DS0s), 2.048 Mbps in Europe (E1) - up to 32 voice channels (DS0s). DS1/T1/E1 lines are part of the PSTN.

DS3 *Digital Signal, Level 3*

T3 in North America and Japan, E3 in Europe. Up to 672 voice channels (DS0s). DS3/T3/E3 lines are not part of the PSTN

DTMF *Dual Tone Multi-Frequency*

Push-button or touch tone dialing.

E1

The European equivalent of North American T1, transmits data at 2.048 Mbps, up to 32 voice channels (DS0s).

E3

The European equivalent of North American T3, transmits data at 34.368 Mbps, up to 512 voice channels (DS0s). Equivalent to 16 E1 lines.

EMI *Electromagnetic Interference*

Unwanted electrical noise present on a power line

full duplex

Data transmission in two directions simultaneously.

FXO *Foreign Exchange Office*

Receives the ringing voltage from an FXS device. Outside lines are connected to the FXO port on your Asterisk Appliance 50 unit.

FXS *Foreign Exchange Station*

Initiates and sends ringing voltage. Phones are connected to the FXS ports on the Asterisk Appliance 50 unit.

G.711

A recommendation by the Telecommunication Standardization Sector (ITU-T) for an algorithm designed to transmit and receive mulaw PCM voice and A-law at a digital bit rate of 64 Kbps. This algorithm is used for digital telephone sets on digital PBX.

G.723.1

A recommendation by the Telecommunication Standardization Sector (ITU-T) for an algorithm designed to transmit and receive audio over telephone lines at 6.3 Kbps or 5.3 Kbps.

G.729a

A recommendation by the Telecommunication Standardization Sector (ITU-T) for an algorithm designed to transmit and receive audio over telephone lines at 8 Kbps.

H.323

A recommendation by the Telecommunication Standardization Sector (ITU-T) for multimedia communications over packet-based networks.

IAX *Inter-Asterisk eXchange*

A VoIP protocol used by Asterisk. It is used to enable VoIP connections between Asterisk servers, and between servers and clients that also use the IAX protocol.

iLBC *internet Low Bitrate Codec*

A free speech codec used for voice over IP. It is designed for narrow band speech with a payload bitrate of 13.33 kbps (frame length = 30ms) and 15.2 kbps (frame length = 20 ms).

ILEC *incumbent local exchange carrier*

The LECs that were the original carriers in the market prior to the entry of competition and therefore have the dominant position in the market.

interface

A point of contact between two systems, networks, or devices.

ISO *International Standards Organization*

LED *light-emitting diode*

Linux

A robust, feature-packed open source operating system based on Unix that remains freely available on the internet. It boasts dependability and offers a wide range of compatibility with hardware and software. Asterisk is supported exclusively on Linux.

loopback

A state in which the transmit signal is reversed back as the receive signal, typically by a far end network element.

MGCP *Media Gateway Control Protocol*

multiplexing

Transmitting multiple signals over a single line or channel. FDM (frequency division multiplexing) and TDM (time division multiplexing) are the two most common methods. FDM separates signals by dividing the data onto different carrier frequencies, and TDM separates signals by interleaving bits one after the other.

MUX *multiplexer*

A device which transmits multiple signals over a single communications line or channel. See multiplexing.

PBX *private branch exchange*

A smaller version of a phone company's large central switching office. Example: Asterisk.

PCI *peripheral component interconnect*

A standard bus used in most computers to connect peripheral devices.

POP *point of presence*

The physical connection point between a network and a telephone network. A POP is usually a network node serving as the equivalent of a CO to a network service provider or an interexchange carrier.

POTS *plain old telephone service*

Standard phone service over the public switched telephone network (PSTN). This service provides analog bandwidth of less than 4 kHz.

PPP *point-to-point protocol*

Type of communications link that connects a single device to another single device, such as a remote terminal to a host computer.

PSTN *public switched telephone network*

The public switched telephone network (PSTN) is the network of the world's public circuit-switched telephone networks. Originally a network of fixed-line analog telephone systems, the PSTN is now almost entirely digital, and now includes mobile as well as fixed telephones.

QoS *quality of service*

A measure of telephone service, as specified by the Public Service Commission.

RJ11

A six-pin jack typically used for connecting telephones, modems, and fax machines in residential and business settings to PBX or the local telephone CO.

SIP *Session Initiation Protocol*

An IETF standard for setting up sessions between one or more clients. It is currently the leading signaling protocol for Voice over IP, gradually replacing H.323.

T1

A dedicated digital carrier facility which transmits up to 24 voice channels (DS0s) and transmits data at 1.544 Mbps. Commonly used to carry traffic to and from private business networks and ISPs.

T3

A dedicated digital carrier facility which consists of 28 T1 lines and transmits data at 44.736 Mbps. Equivalent to 672 voice channels (DS0s).

TDM *time division multiplexer*

A device that supports simultaneous transmission of multiple data streams into a single high-speed data stream. TDM separates signals by interleaving bits one after the other.

telco

A generic name which refers to the telephone companies throughout the world, including RBOCs, LECs, and PTTs.

tip and ring

The standard termination on the two conductors of a telephone circuit; named after the physical appearance of the contact areas on the jack plug.

twisted pair

Two copper wires commonly used for telephony and data communications. The wires are wrapped loosely around each other to minimize radio frequency interference or interference from other pairs in the same bundle.

V *volts*

VoIP *Voice over IP*

Technology used for transmitting voice traffic over a data network using the Internet Protocol.

Zaptel (Zap)

Zapata Telephony Project dedicated to implementing a reasonable and affordable Computer Telephony platform into the world marketplace.