Operation/Reference Guide

CSG

SIP Communications Gateway
CSG-500, CSG-544, CSG-580

Control System Accessories

AMX Limited Warranty and Disclaimer

All products returned to AMX require a Return Material Authorization (RMA) number. The RMA number is obtained from the AMX RMA Department. The RMA number must be clearly marked on the outside of each box. The RMA is valid for a 30-day period. After the 30-day period the RMA will be cancelled. Any shipments received not consistent with the RMA, or after the RMA is cancelled, will be refused. AMX is not responsible for products returned without a valid RMA number.

Warranty Repair Policy

- AMX will repair any defect due to material or workmanship issues during the applicable warranty period at no cost to the AMX Authorized Partner., provided that the AMX Authorized Partner is responsible for in-bound freight and AMX is responsible for out-bound ground freight expenses.
- The AMX Authorized Partner must contact AMX Technical Support to validate the failure before pursuing this service.
- AMX will complete the repair and ship the product within five (5) business days after receipt of the product by AMX. The AMX Authorized Partner will be notified if repair cannot be completed within five (5) business days.
- Products repaired will carry a ninety (90) day warranty or the balance of the remaining warranty, whichever is greater.
- Products that are returned and exhibit signs of damage or unauthorized use will be processed under the Non-Warranty Repair Policy.
- AMX will continue to provide Warranty Repair Services for products discontinued or replaced by a Product Discontinuance Notice.

Non-Warranty Repair Policy

- Products that do not qualify to be repaired under the Warranty Repair Policy due to age of the product or Condition of the product may be repaired utilizing this service.
- The AMX Authorized Partner must contact AMX Technical Support to validate the failure before pursuing this service.
- Non-warranty repair is a billable service.
- Products repaired under this policy will carry a ninety (90) day warranty on material and labor.
- AMX will notify the AMX Authorized Partner with the cost of repair, if cost is greater than the Standard Repair Fee, within five (5) days of receipt.
- The AMX Authorized Partner must provide a Purchase Order or credit card number within five (5) days of notification, or the product will be returned to the AMX Authorized Partner.
- The AMX Authorized Partner will be responsible for in-bound and out-bound freight expenses.
- Products will be repaired within ten (10) business days after AMX Authorized Partner approval is obtained.
- Non-repairable products will be returned to the AMX Authorized Partner with an explanation.
- See AMX Non-Warranty Repair Price List for minimum and Standard Repair Fees and policies.

Safety Certification and Agency Approvals

Safety

- US/CSA 60950
- IEC 60950
- AS/NZS 60950
- EN 60950

Other

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

Telecom

- FCC Part 68, ANSI/ITA-968-A, Including Amendment A1 and A2
- AS/ACIF S002
- AS/ACIF S003
- PTC220

EMC

- FCC Part 15 Class A
- EN55022/CISPR22 Class A
- EN55025
- IEC 61000
- AS/NZS CISPR22 Class A

Safety Certification and Agency Approvals

iv   CSG SIP Communications Gateway

Table of Contents

CSG SIP Communications Gateway

Overview

The CSG SIP Communications Gateway is a stand alone Private Branch Exchange (PBX) which runs a version of Asterisk. It is suitable for the desktop, or mounting in a typical network closet or restricted access location. The CSG is ideal for small office environments or as an extension to a central CSG PBX.

The CSG can function not only as a PBX, but also as VoIP ATA, or VoIP gateway. It has eight analog ports that can be configured (via modules) as Foreign Exchange Office (FXO) or Foreign Exchange Station (FXS) ports.

The CSG enables you to transform an intercom-enabled Modero Touch Panel into a full-featured IP phone. With the CSG and an intercom-enabled Modero Touch Panel you can make and receive local, long distance, and international phone calls, and have access to phone features like call waiting, caller ID, call forwarding, call queuing, and voice mail.

The CSG comes in three different models: the CSG-500, CSG-544, and CSG-580. The CSG-500 supports up to 50 users, but offers no analog lines. In addition to supporting up to 50 users, you can integrate the CSG-544 and CSG-580 to outside PSTN or POTS networks. The CSG-544 allows for up to four phones and four PSTN lines. The CSG-580 allows for up to eight PSTN lines. The CSG supports AMX Session Initiated Protocol (SIP)-enabled touch panels, such as the MVP-8400i, NXD-1000Vi, and NXD-700Vi, as well as 3rd party IP phones.

In a typical network, the CSG acts as proxy server, registrar, and gateway. A SIP-enabled touch panel acts as a user agent, which allows the touch panel to act as an IP phone. FIG. 1 displays a high-level diagram of the recommended network infrastructure for using the CSG.
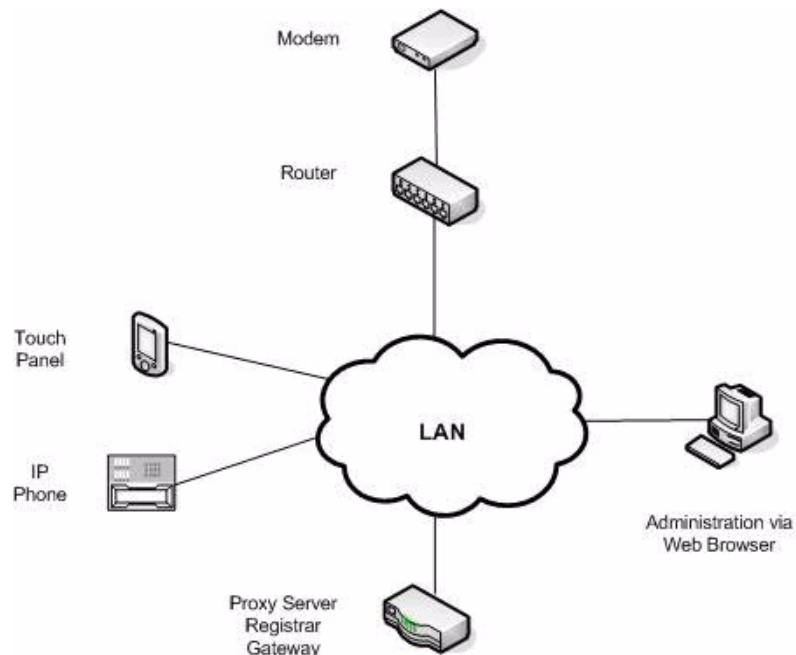

FIG. 1 Recommended network infrastructure

CSG Specifications

The following table lists the specifications for the CSG.

CSG Specifications	
Dimensions (HWD):	1.74" x 11.45" x 5.44" (4.41 cm x 29.09 cm x 13.82 cm)
Weight:	2.25 lbs (1.02 kg)
Enclosure	Metal with black matte finish
Power Requirements	Constant current draw: 2.6 A @ 12 VDC
Memory	• 8MB on board serial Flash memory • 64MB 16-bit parallel SDRAM
Certifications	FCC (Class A), CE, IEC60950, and RoHS
Operating/Storage Environment	• Operating Temperature: 0° C (32° F) to 40° C (104° F) • Operating Humidity: 5% to 85% RH Non-Condensing • Storage Temperature: -20° C (-4° F) to 70° C (158° F) • Storage Humidity: 0% to 85% RH Non-Condensing
Front Panel Components:	
Power LED	Indicates whether the unit is turned on and receiving power.
Network LED	Indicates whether a network cable is connected and the speed of the network.
IP Phones LEDs	Indicates whether the corresponding link is connected and the speed of the link.
Analog LEDs	Indicates whether the analog port is installed and the type of port (FXO/FXS) installed. (Not available on all models.)
Rear Panel Components:	
Power connector	5.1mm OD, 2.1mm pin, 12V 3.0A center positive
Configuration Reset Switch	Resets the current configuration to the factory default when pressed continuously during the boot process.
Console port	1 RJ45 port for console (craft) port serial control
Network port	1 RJ45 10/100baseT port for a general (WAN) connection
IP Phones ports	4 RJ45 10/100baseT ports for IP phone connections
Analog ports	• 8 RJ11 ports: 4 for FXO and 4 for FXS connections (CSG-544 only) • 8 RJ11 ports for FXO connections (CSG-580 only) ***Note***: The CSG-500 does not use analog ports.
Included Accessories	• PSN3.0 Power Supply, 3.0A, 12.0CVD (**FG423-31**) • CSG SIP Communications Gateway Installation Guide (**93-2182-01**) • 2 Surface mounting brackets (**62-2182-04**) • 4 #4-40 X .250 PPH screws (**80-0112**)
Other AMX Equipment	• DIN Rail Mounting Bracket (**FG532-06**)

CSG Interface

The CSG interface enables you to create a PBX solution that rivals the features and functionality of traditional telephony switches. Current PBX solutions are expensive and proprietary. Using the CSG, you can replace an existing small business PBX. Since it runs on Linux, it inherits all of the power and stability of that operating system. The CSG works with most standards-based IP telephone handsets and software. The CSG also supports analog phones and ADSI-screen phones.

Installation

Unpacking the Unit

When you unpack your unit, carefully inspect it for any damage that may have occurred in shipment. If damage is suspected, file a claim with the carrier and contact your reseller from which the unit was purchased or AMX Technical Support.

Keep the original shipping container to use for future shipment or proof of damage during shipment.

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 CSG:

- CSG unit.
- Power supply (FG423-31)
- CSG Installation Guide (93-2182-01)
- 2 Surface mounting brackets (62-2182-04)
- 4 #4-40 X .250 PPH screws (80-0112)

Identifying Communication Ports

The CSG unit consists of up to eight RJ11 analog ports which can be configured as either FXO or FXS ports, depending on the type of modules installed in the CSG model. These ports provide 16ms 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 CSG to act as a router. All the Ethernet ports support auto-MDI/MDX.

An RS-232 console port is also available for additional configuring of the pre-loaded CSG software via direct physical access. The preferred method for configuring the unit is by using the web based interface. You can also configure it remotely using SSH. The CSG is shipped fully configured, but it may be altered for specific applications.

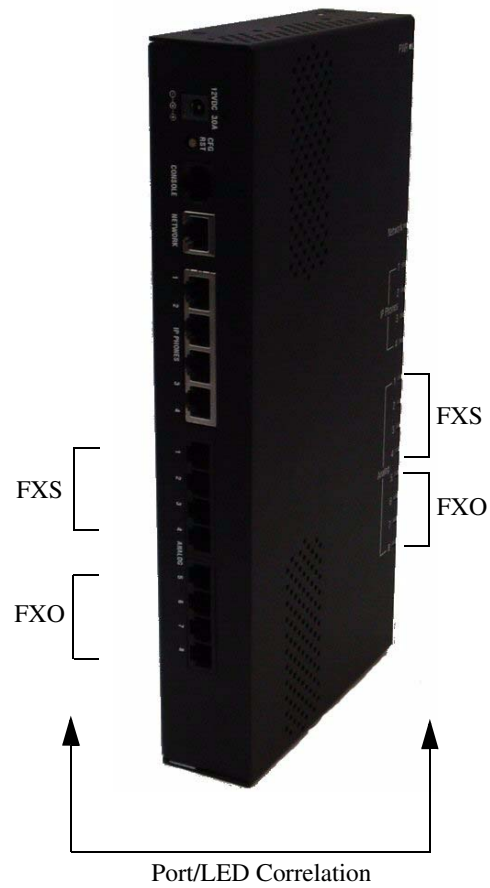FIG. 1 displays the ports and their corresponding LEDs.

FXS

FXO

FXS

FXO

Port/LED Correlation

FIG. 1 CSG - Ports/LEDs

The example shown is configured with four FXO and four FXS ports, model CSG-544.

Understanding the LEDs

There are 14 LEDs on the front panel of the CSG. 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 below.

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 BlackFin processor load.
Network:	• 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.
	• Red (flashing)	Link is up at 10Mbps. LED flashes at 1/10 second intervals as traffic is detected.
IP Phones (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.
	• Red (flashing)	Link is up at 10Mbps. LED flashes at 1/10 second intervals as traffic is detected.
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.

Using the Configuration Reset Switch

The Configuration Reset (CFG RST) switch (rear panel - see FIG. 2) resets the current configuration to the factory defaults when pressed. The switch must be continuously pressed during the boot process. This will force the unit to delete all configuration data.
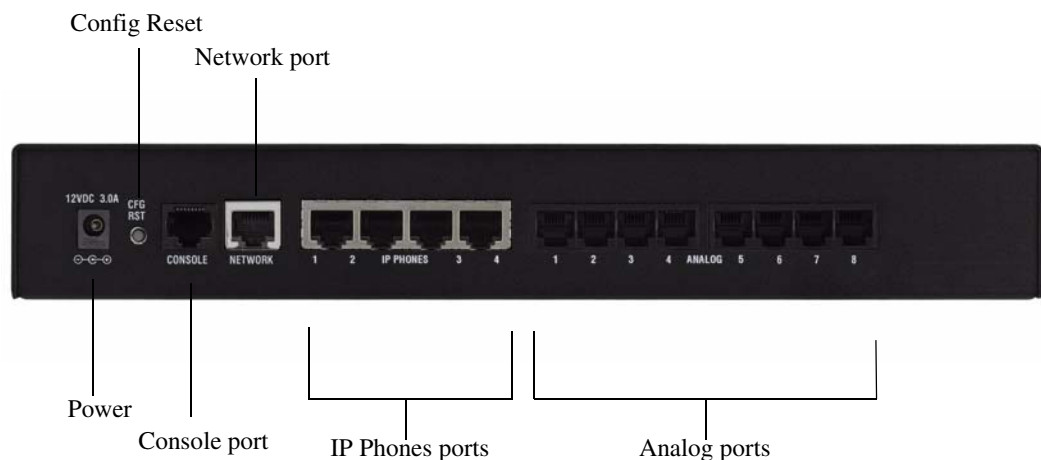

FIG. 2 Rear panel connectors

NOTE

*If you press the RST CFG button, you to lose all configuration settings.*

Pin Assignments

The following tables describe the pin assignments for the different ports on the CSG.

Console Port Pinouts	
Pin	**Description**
1	Ground (Connect to DB9 pin 5)
2	Unused (Leave Open)
3	Primary RxD (To CSG) Connect to DB9 Pin 3
4	Unused
5	Tx (From CSG) Connect to DB9 Pin 2
6	CTS (To CSG) Connect to DB9 Pin 7
7	Open
8	RTS (From CSG) Connect to DB9 Pin 8
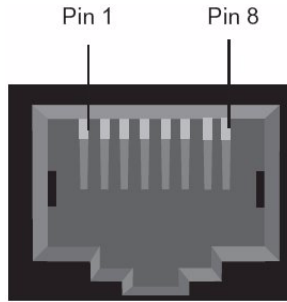
RJ11 Analog Port Connector Pinouts	
Pin	**Description**
1	Unused
2	Unused
3	Tip
4	Ring
5	Unused
6	Unused
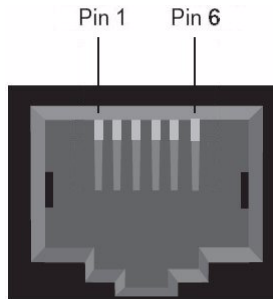
IP Phone and Network Ethernet Port Pinouts	
Pin	**Description**
1	Rx Receive Negative
2	Rx Receive Positive
3	Tx Transmit Negative
4	Unused
5	Unused
6	Tx Transmit Positive
7	Unused
8	Unused

Pin 1   Pin 8
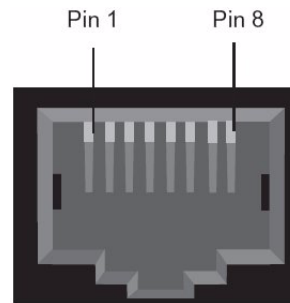
Installing the Hardware

1. Connect an Ethernet cable connected to your network to the Network port on the back of the CSG.

2. Connect an Ethernet cable an IP Phone port. This will be used during the initial configuration of the CSG.

3. Connect the provided power supply to the unit's DC connector. The unit immediately receives power once you connect a power source to it.

4. When the unit completes the boot process, the left-most eight LEDs indicate how the analog ports are configured. A red light indicates the port is FXO, and a green light indicates the port is FXS. If the light for a port is off, the port is not installed.

The analog port configuration is selected when purchasing your CSG unit.

5. 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. If you are using the CSG-500, you can skip this step.

If you are using the CSG-544, do NOT connect analog ports 1-4 to a phone line. Since both the FXS ports and the phone lines supply power, the hardware could sustain damage. This damage is not covered under the AMX standard warranty.

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.

This unit must be connected only to the appropriate Telecommunications Network port (as approved for use in your specific country).

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

Mounting the CSG

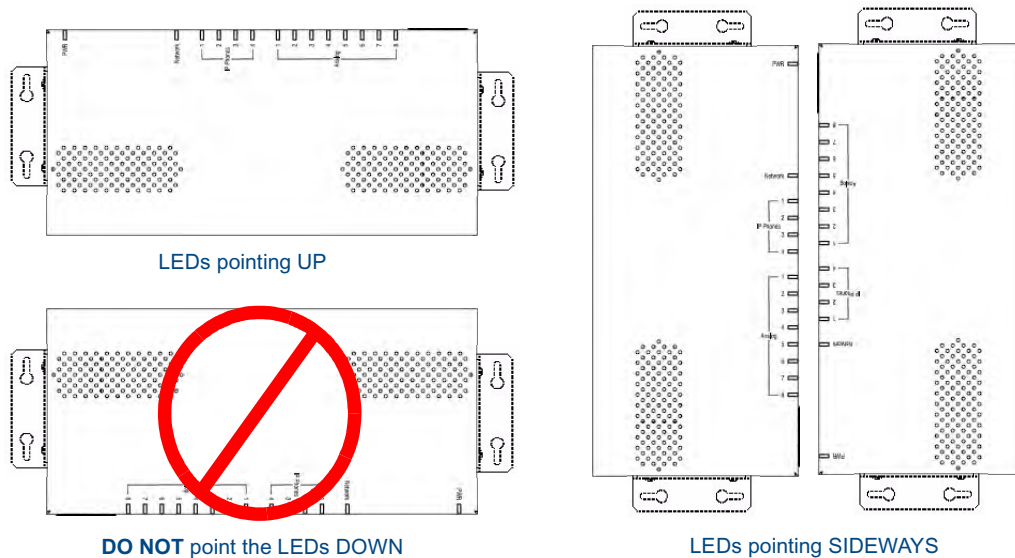FIG. 3 illustrates the proper mounting installation options for wall mounting the unit:

LEDs pointing UP

DO NOT point the LEDs DOWN

LEDs pointing SIDEWAYS

FIG. 3 Acceptable Mounting Orientation options (wall mounting)

Instructions for Wall Mounting

1. Select the area to mount the CSG unit (refer to FIG. 3). The unit should be mounted at or below eye level to properly view the LEDs.

2. Line up one of the surface mounting brackets with the two holes on one side of the CSG. Install two #4-40 X .250 PPH screws into the holes in the bracket to secure the bracket to the CSG.

3. Repeat step 2 to attach the second bracket to the other side of the unit.

4. Affix one screw to the wall. Leave approximately 1/4-inch of the screw protruding from the wall to allow the head of the screw to slide into the settings on the mounting bracket, mounting the unit to the wall.

5. Repeat step 4 to attach the second bracket to the wall.

Instructions for Wall Mounting Using DIN Rail Mounting Brackets

The AC-DIN-CS3 DIN Rail Mounting Brackets (FG532-06) allow you to install the CSG on a standard DIN rail (or "top-hat" rail). The DIN Rail Mounting Bracket comes in a kit that includes everything you need to mount a single device on a standard (35 MM. wide) metal DIN rail. Follow these steps to mount the CSG using the DIN rail mounting brackets:

1. Use the four supplied flat-head screws to secure the DIN mounting clips to the bracket.

2. Use the two supplied pan-head screws to secure the bracket/mounting clip assembly to the CSG. Use the mounting holes on either side panel of the enclosure to mount the bracket/mounting clip assembly.

3. Snap the bracket/clamp onto the DIN rail.

There are indicators on both the bracket and the mounting clips to show which end is the "top" to ensure correct mounting orientation (see FIG. 4).

NOTE

The bracket has a notched side to indicate the top of the piece. The DIN mounting clips also have two holes drilled in the top portion of the clips that indicate the top of the clip.
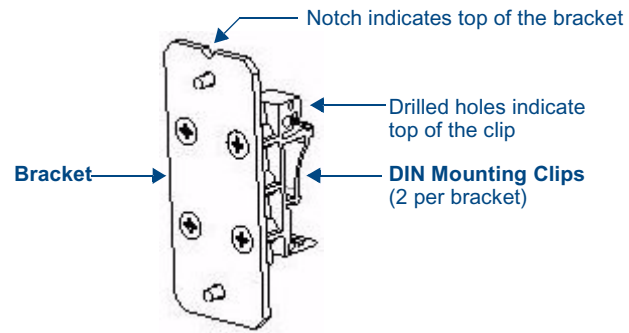
Notch indicates top of the bracket

Drilled holes indicate top of the clip

Bracket

DIN Mounting Clips (2 per bracket)

FIG. 4 AC-DIN-CS3 - Clip/Bracket assembly (attach to CSG)

FIG. 5 illustrates the correct mounting orientation of the DIN Rail Mounting Bracket:

This notch indicates
the top of the bracket

These two holes indicate
the top of the mounting
clips

DIN Rail
(35mm)

Bracket
(mounts to the CSG)

DIN Mounting Clip
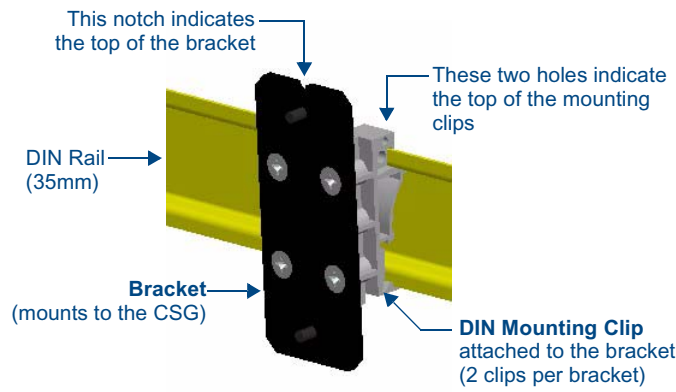attached to the bracket
(2 clips per bracket)

FIG. 5 AC-DIN-CS3 DIN Rail Mounting Bracket - mounted on a DIN rail

FIG. 6 illustrates the correct mounting orientation of the DIN Rail Mounting Bracket and the orientation of the mounted device.
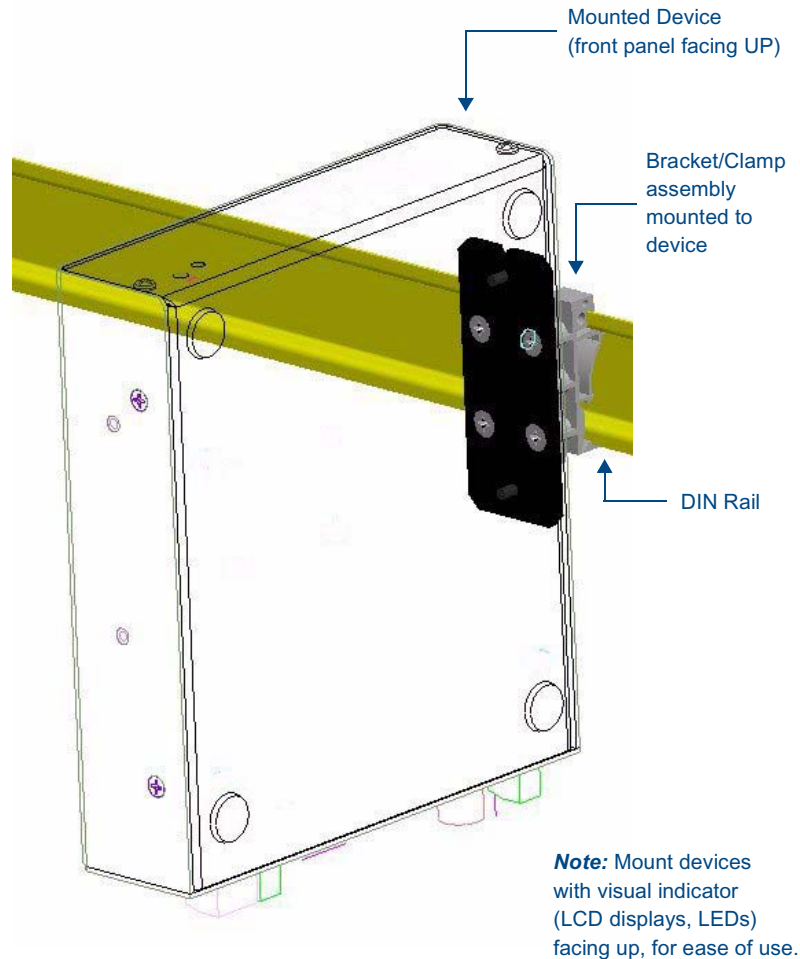

Mounted Device
(front panel facing UP)

Bracket/Clamp
assembly
mounted to
device

DIN Rail

Note: Mount devices
with visual indicator
(LCD displays, LEDs)
facing up, for ease of use.

FIG. 6 Device mounted on a DIN rail using the DIN Rail Mounting Bracket kit

You can also mount the CSG by attaching the mounting brackets to the bottom of the unit. FIG. 7 displays the CSG with mounting brackets attached to its underside. The bracket clamps attach to the DIN rail as shown in FIG. 5.

FIG. 7 DIN Rail Mounting Brackets attached to bottom of CSG unit

Configuration

The CSG comes pre-installed with a web-based configuration tool you can use to configure your unit. This chapter gives you all the information you need to configure your CSG.

For troubleshooting purposes, advanced users may alter the CSG configuration files or perform Linux console commands through the Console port. Advanced configuration changes may also be made remotely using SSH if the unit is connected to a network.

Server Setup

The first step in using the CSG is to get your unit up and running. To get your server up and running, see the *Installation* section on page 3 of this guide.

Setting up Your System

The CSG comes with a user interface framework which gives you the ability to configure it through an easy to use interface. To access the interface, you will need the IP address your CSG server is using. The default IP address is http://192.168.69.1, which you can use to configure your CSG via a web browser.

The user interface provided with the CSG gives you the ability to set up a telephone system without the need to use command line configuration. The user interface begins with an install wizard, covering the essential pieces required for your phone system. After completing the install wizard, the primary menu displays, which gives you the ability to further modify the existing configurations or add features. The CSG comes with a user interface framework which gives you the ability to configure it through an easy to use interface. To access the interface, you will need the IP address your CSG server is using.

Logging on to the CSG for the First Time

Follow these steps to log on to the CSG:

1. Connect a computer to an IP Phone port on the CSG with an Ethernet cable. The computer must be configured to obtain an IP address automatically.

2. Open a web browser. In the address field of the web browser, enter the default IP address http://192.168.69.1. FIG. 1 displays the initial login page.

FIG. 1 CSG Login Page

3. To log on to the system, enter the following user name and password:

   - Username: **admin**
   - Password: **1988**

Subsequent Logons to the CSG

The method described above will work to log onto the configuration tool of the CSG. In addition, you can log onto the CSG using the CSG's Network port once it has been enabled. In this case, open a web browser on a computer which is located on the same network as the CSG. In the address field of the web browser, enter the IP address of the CSG as defined during the initial installation.

Bonjour (Zero-Configuration) Client

You can also log on to the CSG by using a zero-configuration networking client that allows you to determine the unit's IP address, such as Bonjour or a similar zero-configuration client. Zero-configuration (or Zeroconf, also known as "Bonjour") technology provides a general method to discover services on a local area network. In essence, it allows you to set up a network without any configuration, as described below.

You may need a zero-configuration client to determine the IP address of the CSG. There are many zero-configuration clients available. However, for the purposes of this document, we will refer to *Bonjour for Windows*. It is free, and widely available for download. If you don't already have it installed on your PC, download and install *Bonjour for Windows* before you begin.

Perform these steps to log on to the CSG through Bonjour for Windows:

1. With *Bonjour for Windows* running on a PC that has access to the LAN on which the CSG resides, connect the CSG to the network (see *Installing the Hardware* section on page 7).

2. In Bonjour, you will see the unit join the network at power up (FIG. 2). Double-click the CSG link to access the CSG Login page (FIG. 1).
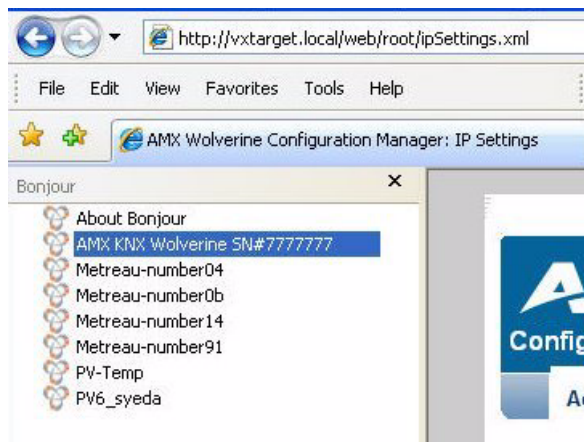

FIG. 2 Bonjour for Windows - example screen

3. To log on to the system, enter the following user name and password:

   - Username: **admin**
   - Password: **1988**

As shown in FIG. 2, Bonjour for Windows operates as a plug-in to Internet Explorer (version 7 shown), and is displayed in the IE Explorer Bar. If you have installed Bonjour for Windows, but don't see the Bonjour toolbar icon, you may need to "unlock" and expand the toolbars to see it.

Administrative Options

The CSG provides you with options to modify the existing elements of or add features to your system. After logging into the user interface, a variety of options displays along the top of the page.

The main window for the user interface has three tabs: Admin, System, and Telephony. Clicking any of the tabs opens a menu of options for the selected tab. Selecting an option makes a series of associated options appear in the center section of the page.

The Admin tab enables you to perform administrative functions such as changing passwords, backing up the system, and upgrading firmware.

Local Extension Settings

The Local Extensions page enables you to modify the boilerplate settings for the local extension, agent login, and default new user. You can access this page by selecting **Admin. Options | Local Ext. Settings** on the Admin tab. FIG. 3 displays the Local Extensions page.

FIG. 3 Local Extensions page

The following table lists the options on the Local Extensions page:

Local Extensions Page Options	
Option	**Description**
Local Extension Settings	
Local Extensions are	This combo box enables you to indicate how many digits comprise an extension.
First Extension Number	This field enables you to indicate the first extension number to be created when you start adding user extensions. Each new user extension increments by one from the previously-created extension unless you specify otherwise.
Operator Extension	This combo box enables you to designate an extension for the operator.
Allow analog phones to be assigned to multiple extensions	This option enables you to indicate whether you want to allow analog phones to be assigned to multiple extensions.
Allow extensions to be AlphaNumeric (SIP/IAX users)	This option enables you to indicate whether you want to allow extensions to be alphanumeric. This option only applies to SIP users.
Agent Login Settings	
Agent Login Extension	This field enables you to enter an extension for agents to log into.
Agent Callback Login Extension	This field enables you to enter an extension for agent callbacks.
Default Settings for a New User	
Is Agent	A call queue is a bank of agents who receive calls. An extension listed as Is Agent can be added to a call queue.
In Directory	Indicates whether the extension is listed in a directory.
SIP	Indicates whether the extension can receive SIP calls.
Call Waiting	If call waiting is not enabled, the extension accepts only call before it is defined as busy
Voicemail	Indicates whether an extension has access to voice mail.
CTI	Indicates whether the user can connect applications to the CSG.
IAX	Indicates whether the extension can receive IAX calls.
3-Way Calling	Indicates whether an extension has access to 3-way calling.
VoiceMail Password	This field enables you to enter a default password for all new voice mail boxes.

Change Password

The Change Password page enables you to change the administrator password to the CSG user interface. You can access this page by selecting **Admin. Options | Change Password** on the Admin tab. FIG. 4 displays the Change Password page.

FIG. 4 Change Password page

To change the administrator password, enter the new password in the Enter New Password field. Confirm the password by entering it again in the Retype New Password field. Click **Accept** to save the new password.

Restore Factory Defaults

The Restore Factory Defaults page enables you to restore the original settings to the CSG user interface as they were when you first received your CSG unit. Restoring the original settings deletes any new configuration settings you previously created. You can access this page by selecting **Admin. Options | Factory Defaults** on the Admin tab. FIG. 5 displays the Restore Factory Defaults page.

FIG. 5 Restore Factory Defaults page

Click **Restore Factory Defaults** to restore the CSG's original settings.

Backup/Restore

Backup is a housekeeping option that enables you to back up your CSG settings to preserve your changes. You can access this page by selecting **Back-up** on the Admin tab. FIG. 6 displays the Backup / Restore Configuration page.
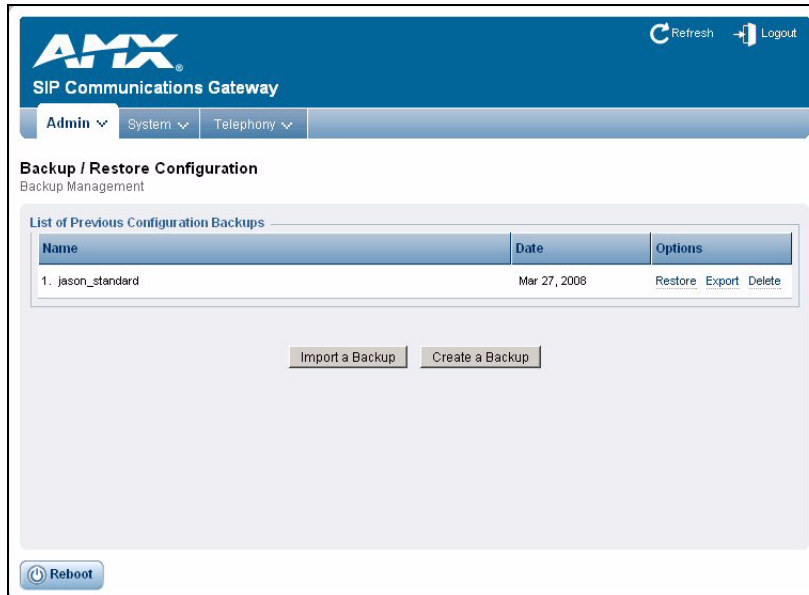

FIG. 6 Backup / Restore Configuration page

Perform the following steps to create a backup of your system:

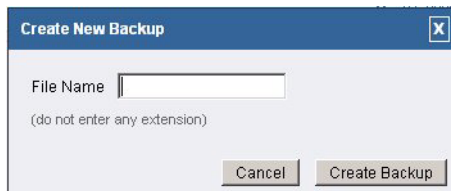**1.** Click **Create a Backup**. The Create New Backup dialog box opens (FIG. 7).

FIG. 7 Create New Backup dialog box

2. In the **File Name** field, enter a name for the backup file.

3. Click **Create Backup**. The backup file appears on the Backup / Restore Configuration page.

To restore a backup file, click **Restore** under Options on the same line as the file you want to restore. A confirmation dialog box appears confirming that you want to proceed with restoring the backup file. If you click OK, the CSG reboots after restoring the file.

Upgrading Firmware

When firmware upgrades are available, you can download and install them on the Upgrade Firmware page (see FIG. 8). You can access this page by selecting **Upgrade Firmware** on the Admin tab. You can upgrade firmware by HTTP URL or TFTP server.

FIG. 8 Upgrade Firmware page

System Options

The System tab enables you to view and edit information about your system such as network configurations and SIP and IAX global configurations.

System Information

The general system information of the CSG displays on the System Information page. This page also includes tabs for your ifconfig, disk partitioning resources, and product configuration for your CSG model.

Networking Configuration

The Networking Configuration page enables you to view and edit your network configuration. This page contains five tabs: Main, WAN Network, IP Phones, Timezones, and Operation Mode. FIG. 9 displays the Main tab of the Networking Configuration page. You can access this page by selecting **Network | Main** from the System tab.
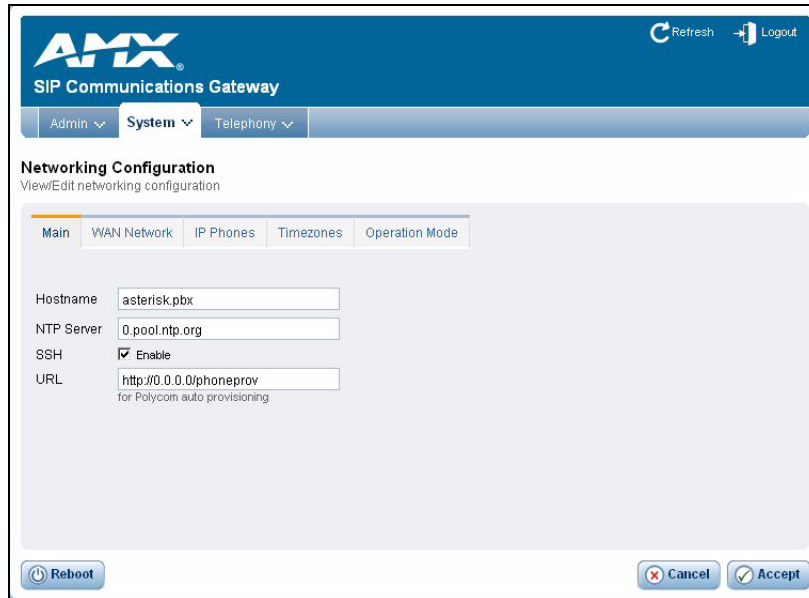
FIG. 9 Networking Configuration page - Main tab

The following table lists the options on the Main tab of the Networking Configuration page:

Networking Configuration - Main Tab Options	
Option	**Description**
Hostname	This field enables you to view or change the host name of your CSG. This name is only used to identify the CSG on your network.
NTP Server	This field enables you to specify the URL of an NTP server. This is useful if you want to regularly synchronize the CSG time setting with the time setting on an NTP server.
SSH	This option enables you to indicate whether you want to enable the Secure Shell (SSH) server on your CSG. Enabling this option causes your unit to provide SSH access on both the WAN and LAN interfaces, which can pose a security risk.
URL	This field enables you to enter a URL that enables auto-provisioning for Polycom phones.

FIG. 10 displays the WAN Network tab of the Networking Configuration page. You can access this page by selecting **Network | WAN Network** from the System tab. You can use the options on this tab to establish the settings that enable you to connect to the Internet, or to an internal, private network.

FIG. 10 Networking Configuration page - WAN tab

The following table lists the options on the WAN Network tab of the Networking Configuration page:

Networking Configuration - WAN Network Tab Options	
Option	**Description**
DHCP	This option enables you to indicate whether you want to enable the Dynamic Host Configuration Protocol (DHCP) on your WAN. This option is enabled by default.
IP	This field enables you to view or change the IP address for your WAN.
SUBNET	This field enables you to view or change the subnet address for your WAN.
GATEWAY	This field enables you to view or change the gateway address for your WAN.
DNS	This field enables you to view or change the DNS address for your WAN.
GUI	This option enables you to indicate whether you want to enable access to the CSG's Graphical User Interface (GUI) via the WAN interface.
Polycom Provisioning	This option enables you to indicate whether you want to allow provisioning of Polycom phones connected through the WAN.

If you have difficulty obtaining an IP address dynamically, clear the DHCP check box and specify the IP address, subnet, gateway, and DNS settings. This information should be available from your network administrator or Internet Service Provider (ISP).

FIG. 11 displays the IP Phones tab of the Networking Configuration page. You can access this page by selecting **Network | IP Phones** from the System tab. You can use the options on this tab to establish settings to access your CSG.
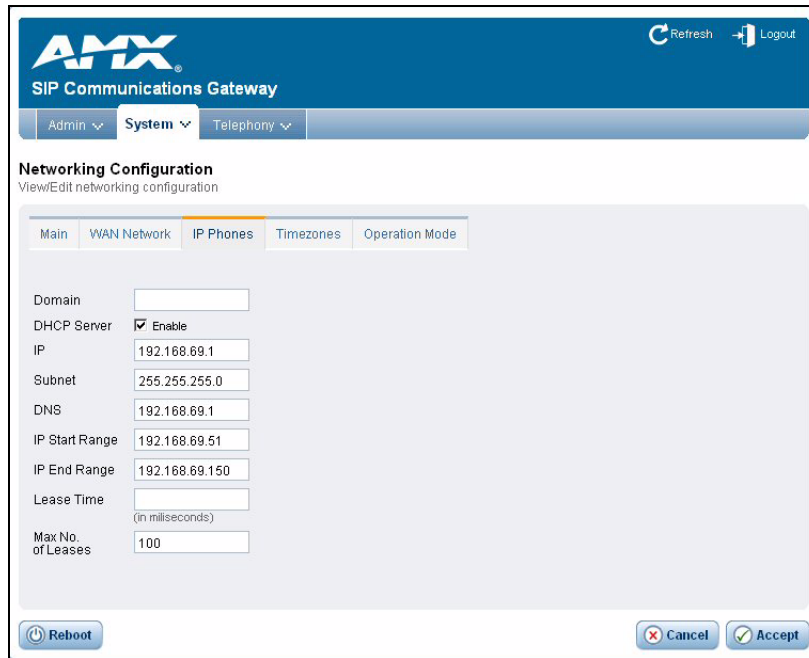

FIG. 11 Networking Configuration page - IP Phones tab

The following table lists the options on the IP Phones tab of the Networking Configuration page:

Networking Configuration - IP Phones Tab Options	
Option	**Description**
Domain	This field enables you to view or change the domain address for your IP phone.
DHCP Server	This option enables you to indicate whether you want to enable the DHCP on your IP phone. This option is enabled by default.
IP	This field enables you to view or change the IP address for your IP phone. The default value is 192.168.69.1.
SUBNET	This field enables you to view or change the subnet address for your IP phone. The default value is 255.255.255.0.
DNS	This field enables you to view or change the DNS address for your IP phone. The default value is 192.168.69.1.
IP Start Range	This field enables you to view or change the starting IP address for a range of IP addresses on your IP phone. The default value is 192.168.69.51.
IP End Range	This field enables you to view or change the ending IP address for a range of IP addresses on your IP phone. The default value is 192.168.69.150.
Lease Time	This field enables you to view or change the amount of time, in milliseconds, you want your IP phone to lease an IP address.
Max No. of Leases	This field enables you to view or change the maximum number of IP addresses you want your IP phone to lease at one time. The default value is

You should not change the default values assigned to each field on the IP Phones tab unless it is absolutely necessary.

FIG. 12 displays the Timezones tab of the Networking Configuration page. You can access this page by selecting **Network | Timezones** from the System tab.
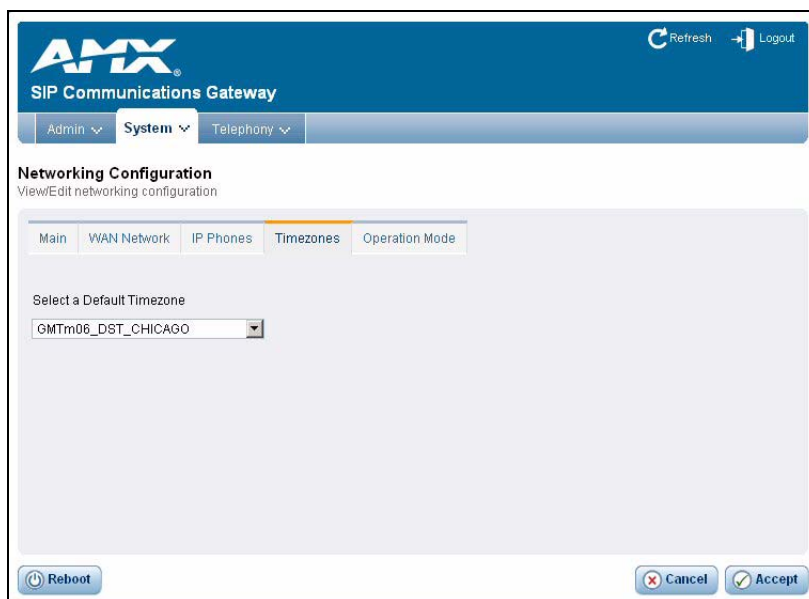

FIG. 12 Networking Configuration page - Timezones tab

You can select a time zone file by using the Select a Default Timezone combo box to select the appropriate time zone. Once you select the time zone, you can click **Accept** to set it.

FIG. 13 displays the Operation Mode tab of the Networking Configuration page. You can access this page by selecting **Network | Operation Mode** from the System tab.
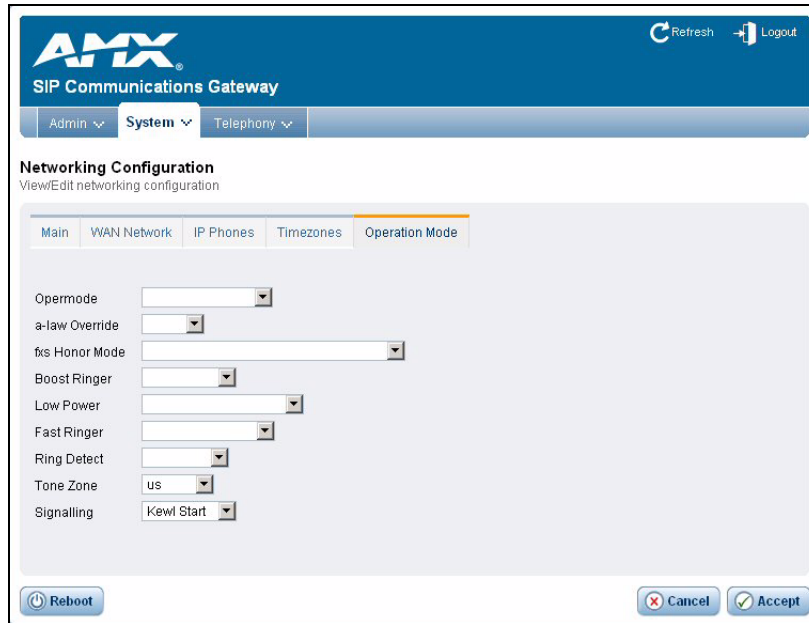
FIG. 13 Networking Configuration page - Operation Mode tab

The options on the Operation Mode tab specify operational mode settings for your CSG which can vary between countries. The following table lists the options on the Operation Mode tab of the Networking Configuration page:

Networking Configuration - Operation Mode Tab Options	
Option	**Description**
Opermode	This option enables you to select the country in which your CSG is operating.
a-law Override	This option enables you to set the audio compression scheme. The setting you choose is dependent on the country of operation. U-law is used in the United States and Canada. A-law is used in most other countries. If possible, confirm the scheme which is best for operation of your CSG.
fxs Honor Mode	This option enables you to choose whether you apply the opermode setting to your FXO modules only, or to both FXS and FXO modules.
Boost Ringer	This option enables you to set the voltage used for ringing an analog phone. You can choose from **normal** and **peek(89V)**. Normal sets the ring voltage to a normal level. Peek sets the voltage to 89v.
Low Power	This option enables you to set the low power to normal or to a fast ringer peak of 50v.
Fast Ringer	This option enables you to set the fast ringer tone to normal or to a 25Hz tone.
Ring Detect	This option enables you to select the ring detect mode. Users who are experiencing trouble detecting caller ID from analog service providers or whose lines exhibit a polarity reversal before the provider transmits caller ID should select **Full Wave**. Otherwise, choose **Standard**.
Tone Zone	This option enables you to select the touch tones, as defined by a two-letter country code, which you will hear when pressing the keys on your analog phone.
Signalling	This option enables you to select the type of signalling. You can choose from **Kewl Start** and **Loop Start**. Kewl Start is the default setting.

Global Settings

The Global Settings options enable you to view and edit your global settings configuration. You can use the corresponding pages to manage your SIP or IAX configurations. FIG. 14 displays the SIP Global Configuration page. You can access this page by selecting **Global Settings | SIP** from the System tab.
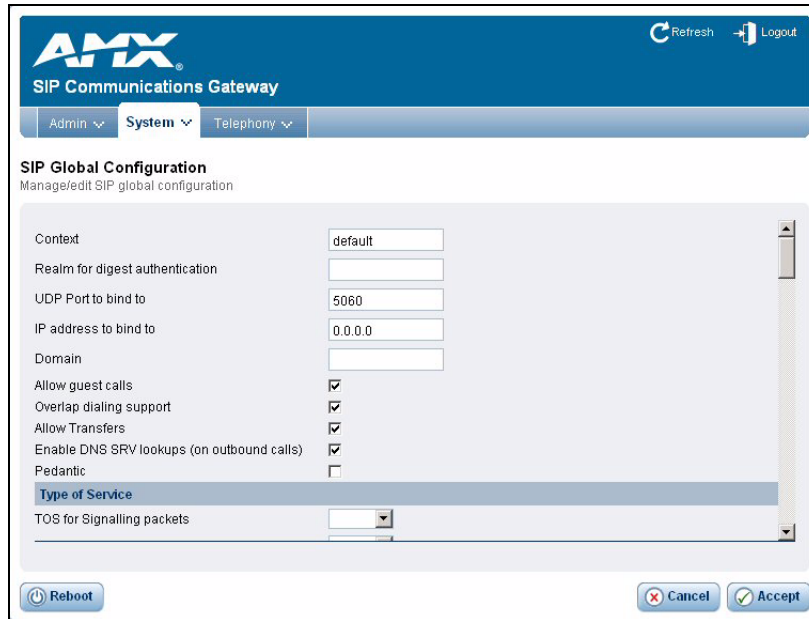
FIG. 14 SIP Global Configuration page

After configuring the numerous options as appropriate, you can click **Accept** to save the settings.

FIG. 15 displays the Inter Asterisk Exchange Protocol (IAX) Global Configuration page. You can access this page by selecting **Global Settings | IAX** from the System tab.
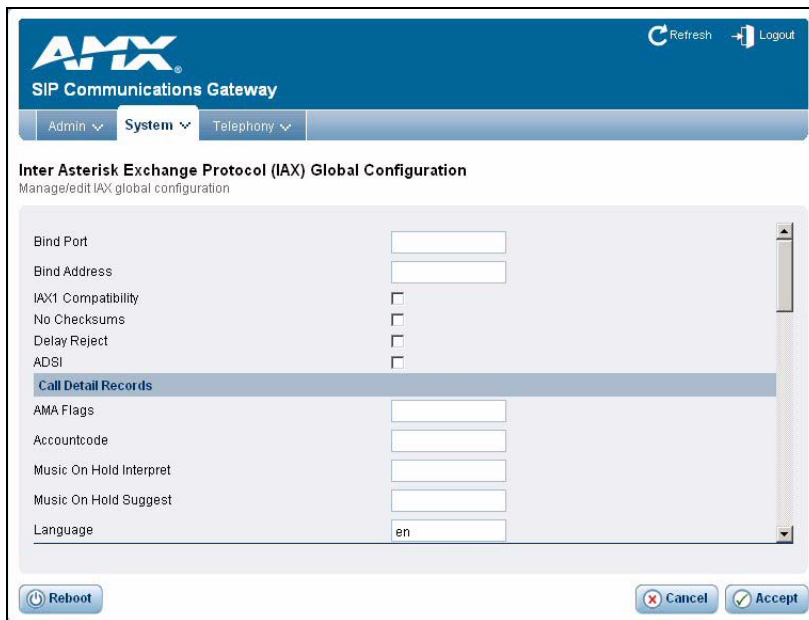

FIG. 15 Inter Asterisk Exchange Protocol (IAX) Global Configuration page

After configuring the numerous options as appropriate, you can click **Accept** to save the settings.

Telephony Options

The Telephony tab enables you establish all of your telephone settings such as user extensions, conference bridges, voice mail, service providers, and call queues. The following sections describe the various options on the Telephony tab.

User Configuration

The Users Configuration page establishes the connections from the listed extension to other systems within the CSG. Select **Users** from the Telephony tab to display a series of options for individual user extensions. FIG. 16 displays the Users Configuration page.

FIG. 16 Users Configuration page

To create a new user extension, click **New**. The next available extension in the sequence automatically populates the Extension field. The system defaults to four-digit extensions, beginning with the number 6000. Then, configure the extension using the available options on the page. You can also edit an existing user extension by selecting it from the available list. When you are finished, click **Accept** to save the user extension. The following table lists the options on the Users Configuration page:

User Configuration Options	
Option	**Description**
Extension	This field displays the number of the selected extension. You can assign a different extension number by entering a new number.
Name	This field enables you to enter the name of the user associated with the extension.
Password	This field enables you to enter a numeric password for access to the extension.
VM Password	This field enables you to enter a numeric password for access to the extension's voice mail.

User Configuration Options (Continued)	
Option	**Description**
E-mail	This field enables you to enter an e-mail address for the extension. You must enter a valid e-mail address here if the owner of the extension wants to receive e-mail with a recording of each voice mail attached to it. See the *Voice Mail* section on page 31 for more information about attaching voice mail recording to an e-mail.
Caller ID	This field enables you to enter the name that appears on a call recipient's caller ID when the user at this extension calls.
Analog Phone	This option enables you to select an analog phone for the extension. If there are no analog phones available, select **None**.
Dial Plan	This option enables you to select a dial plan for the extension. See the *Call Rules* section on page 39 for more information on creating a dial plan.
Phone Serial	This field enables you to specify the MAC address of the SIP phone connected to the specified port. The MAC address associates the phone with this extension and enables the synchronization of provisioning information.
Extended Options:	
Voicemail	Builds a voice mail box for the extension that you can reach by dialing the Check Voicemail extension.
In Directory	CSG establishes a directory of all extensions so that inbound callers can reach someone in your office by dialing the first few digits of the person's first or last name. The company directory includes only the name of the extension if you check this option.
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.
CTI	Computer Telephony Integration. Selecting this option enables the user to connect applications to the CSG.
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 on the Queue Extension Configuration page. See the *Calling Queues* section on page 35 for more information.
Call Waiting	If call waiting is not enabled, the extension accepts only one call before it is defined as busy.
3-Way Calling	Enables 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.
Can Reinvite	Enables the CSG to send SIP reinvite messages to establish direct media streams between endpoints.
NAT	Network Address Translation. This is a method that rewrites the IP addresses and ports of IP packets that pass through a router. Checking this box indicates the user is on a network with NAT enabled.
DTMFMode	Enables you to set the default mode for sending Dual-Tone Multi-Frequency (DTMF) tones. 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** - Uses rfc2833 if offered, inband otherwise.

User Configuration Options (Continued)	
Insecure	Insecure in a SIP parameter used to determine peer matching. The setting determines whether an insecure connection is allowed or if authentication is required. The valid options are: • **port** - Select this option to match against only an IP address. This setting is useful if you have multiple endpoints behind a NAT device. • **invite** - Select this option to match against both the IP address and port number provided in the Contact field of a SIP header. A call will be allowed without authentication if a match is found. • **very** - Select this option if you do not want to require authentication upon an initial value. • **no** - Select this option if you do not want to allow an insecure connection.
Flash	Flash timing defines the duration, in milliseconds, the CSG uses if it is sending a flash signal to another system.
RXFlash	Receiver Flashtime defines the duration, in milliseconds, the CSG requires to consider a flash operation that it receives to be valid.

Setting Codec Preferences

For each user extension, you can allow or disallow specific codecs to be used. Perform the following steps to set the codec preferences for a user extension:

1. Select **Users** from the Telephony tab to display the Users Configuration page (FIG. 16).

2. Select an existing user extension or click **New** to create a new extension.

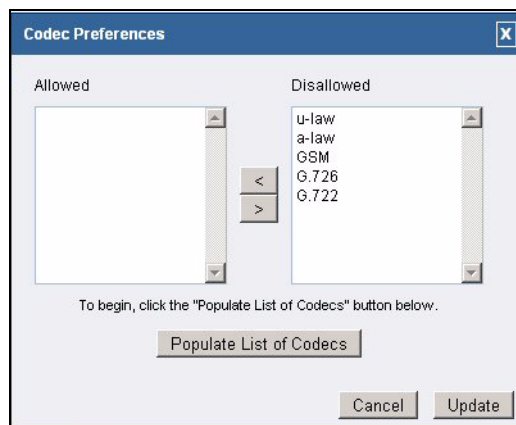3. Click **Allow/Disallow Codecs**. The Codec Preferences dialog box opens (FIG. 17).

FIG. 17 Codec Preferences dialog box

4. Click **Populate List of Codecs**. You only need to click this button when there are new codecs to add to the list.

5. Select a codec from the **Allowed** or **Disallowed** areas, and click the appropriate directional button to move the codec to its appropriate area. Repeat this for each codec you want to move.

6. Click **Update**. The dialog box closes and the Users Configuration page displays.

Conferencing

Every company reaches the point of needing more people on a phone call than it can effectively include through three-way calling. CSG conference bridges enable you to include more people as well as project a professional image. A conference bridge enables you to provide music while a caller is on hold as well as give you the capability to record your conferences. FIG. 18 displays the Conference Bridge Extensions Configuration page.
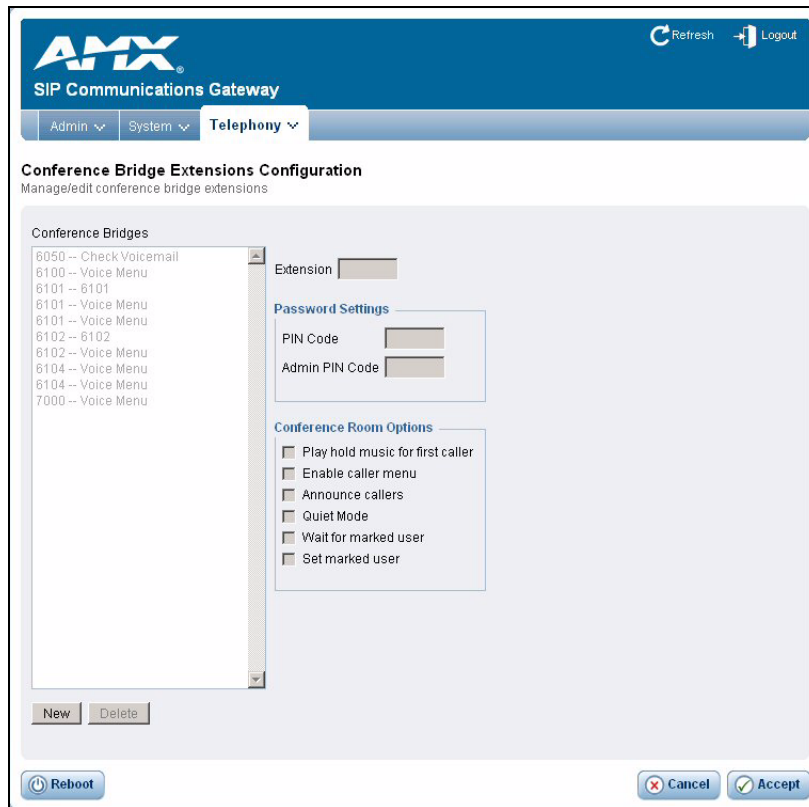

FIG. 18 Conference Bridge Extensions Configuration page

To create a new conference bridge, click **New**. The next available extension is automatically populated in the sequence. The system defaults to four-digit extensions, beginning with the number 6000. Then, configure the bridge using the available options on the page. You can also edit an existing conference bridge by selecting it from the available list. When you are finished, click **Accept** to save the conference bridge. The following table lists the options on the Conference Bridge Extensions Configuration page:

Conference Room Options	
Option	**Description**
Extension	This field displays the number of the selected conference bridge. You can assign a different extension number by entering a new number.
Password Settings	
PIN Code	This field enables you to assign the PIN code used by participants to enter the conference.
Admin PIN Code	This field enables you to assign the Administrative PIN code used by the moderator of the conference to open the conference bridge. This option also enables the administrative menu when you press the asterisk (*) key on the phone.

Conference Room Options (Continued)	
Conference Room Options	
Play hold music for first caller	This option enables you to indicate whether you want music to play for the first caller entering the conference until another caller joins.
Enable caller menu	This option enables you to indicate whether you want callers to be able to access the Conference Bridge Menu by accessing the asterisk (*) key.
Announce callers	This option enables you to indicate whether you want all new callers to be identified when they arrive to the conference.
Quiet Mode	You may choose this feature for a conference bridge because it enables all users on the bridge to have listen-only access to the conference. Establishing two access points, with one group of people using the main extension while the other remains in quiet mode, allows a controlled environment in which one group can deliver information while the second group listens.
Wait for Marked User	This is a "Big Brother" feature that keeps all participants in quiet mode until a certain 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.
Set Marked User	This option works in conjunction with the Wait for Marked User feature. Checking Set Marked User makes the individual arriving from this extension the Marked User. If the CEO of the company does not want anyone chatting in the conference bridge until he or she arrives, this option is set to keep everything quiet.

Voice Mail

Voice mail is an available option for every extension in CSG. The relationship between the extension and the voice mail is established in the User Extension section of the user interface. That section covers only the relationship between the extension and the voice mail, but it does not identify the parameters of the voice mail service itself. You can access the Voice Mail Configuration page by clicking the **Telephony** tab, then selecting **Voicemail | Main**. FIG. 19 displays the Voice Mail Configuration page.

FIG. 19 Voice Mail Configuration page

To configure the voice mail for an extension, select the extension from the available list on the left side of the page. Establish the voice mail settings for the extension, as appropriate. When you are finished, click **Accept** to save the Voice Mail configuration. The following table lists the options on the Conference Voice Mail Configuration page:

Voice Mail Configuration Options	
Option	**Description**
Extension for checking messages	This field displays the number of the selected extension. You can assign a different extension number by entering a new number.
Attach recordings to e-mail	This option enables you to toggle whether the voice mail message can be attached as a sound file to an e-mail notification.
Max greeting (seconds)	With this option, you stipulate the maximum amount of time available for your employees to record their voice mail greeting.
Dial "0" for Operator	Callers who are sent to voice mail can press "0" for the operator and be transferred either during the voice mail salutation or after recording the message. Without this option, if a caller presses "0", it will be ignored.
Message Options	
Message Format	This option enables you to choose the default format for recording all voice mails. You can choose from WAV (GSM), WAV (16-bit), and RAW GSM.
Maximum messages (per folder)	This option enables you to set the maximum number of messages per voice mail box.
Max message time	This option enables you to set the maximum duration of a message left by a caller.
Min message time	This option enables you to set the minimum duration of a message left by a caller. Any message left that is under the listed duration is discarded and is not processed or retrievable.
Playback Options	
Send messages by e-mail only	This option enables you to choose to have voice mail sent to your e-mail address.
Say message Caller-ID	This option reads the caller ID before playing the voice mail message.
Say message duration	This option identifies exactly how long the message lasted.
Play envelope	The envelope provides the date, time, and caller ID related to a voice mail.
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 voice mail extension. Standard options are presented to you, enabling you to discard the message or re-record it if you are not happy with it.

The Voice Mail Configuration page also enables you to establish or edit your STMP settings. Click **SMTP Settings** to open the Voice Menus Configuration page (see FIG. 20).

FIG. 20 Voice Menus Configuration page

On this page, enter your SMTP server and port in the appropriate fields and click **Accept** to save the settings. Click **Back to Voice Mail Configuration** to return to the previous page.

Voice Mail E-mail Configuration

On the Voice Mail Configuration page, if you indicate you want an extension to receive an e-mail with an attached voice mail recording whenever someone leaves a voice mail on the extension, you can create a template for the voice mail e-mail so the recipient receives the necessary information in the subject and body of the e-mail. You can set up this template on the Voice Mail E-mail Configuration page. You can access this page by clicking the **Telephony** tab, then selecting **Voicemail | E-mail Template** (see FIG. 21).

FIG. 21 Voice Mail E-mail Configuration page

Perform the following steps to create a template for voice mail e-mails:

1. In the **From** field. enter a sender. All voice mail e-mails a user receives will be received from this sender.

2. In the **Subject** field, enter a subject for the e-mail. All voice mail e-mails a user receives will contain the subject line you enter.

3. Use the **Message** area to create a standard e-mail that will be sent to the recipient. You can type text in the area, as well as use any of the template variables listed at the bottom of the page.

4. Click **Accept** to save the changes.

Calling Queues

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

The CSG identifies which extensions are capable of belonging to a call queue by identifying whether the Is Agent option is selected on the Users Configuration page. If no check mark appears next to Is Agent, that extension will not appear in the Agents area on this page. FIG. 22 displays the Queue Extension Configuration page.
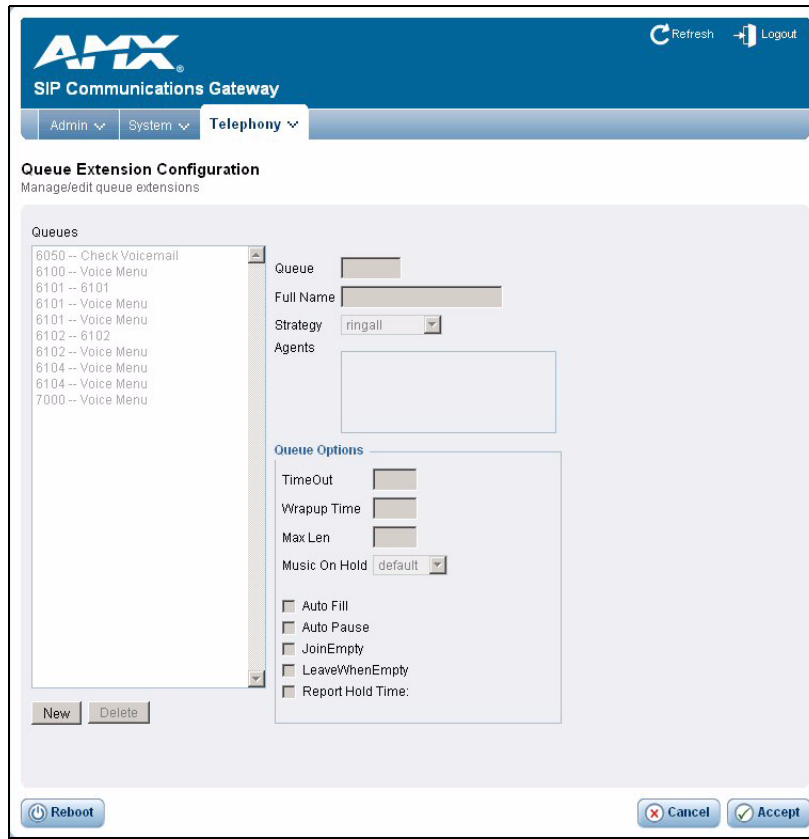
FIG. 22 Queue Extension Configuration page

To create a new calling queue, click **New**. The next available extension is automatically populated in the sequence. The system defaults to four-digit extensions, beginning with the number 6000. Then, configure the queue using the available options on the page. You can also edit an existing calling queue by selecting it from the available list. When you are finished, click **Accept** to save the calling queue. The following table lists the options on the Queue Extension Configuration page:

Queue Extension Configuration Options	
Option	**Description**
Queue	This field enables you to view or change the four-digit extension assigned to the calling queue.
Full Name	This field enables you to create a name for the calling queue.

Queue Extension Configuration Options (Continued)	
Option	**Description**
Strategy	This option enables you to choose one of the following options for transmitting calls throughout the queue: • **Ring All** - Rings every agent who is not 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 is 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.
Agents	This option enables you to select which users you want to assign to the calling queue. You can assign users by clicking the check box next to the user's extension.
Queue Options	
TimeOut	The number of seconds an agent's phone will ring before the call is forwarded to another agent. The default setting for this option is 15 seconds.
Wrapup Time	This is a buffer of time that allows 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 the queue before the calls are sent elsewhere.
Music On Hold	This option enables you to select the music to play when an agent puts a caller on hold.
AutoFill	This option is enabled by default and 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.
Join Empty	This option enables callers to enter a queue even if no agents are logged into it. If this option is not checked, callers cannot enter a queue until at least one agent is present.
Leave When Empty	This option mirrors the Join Empty option, but it represents a queue in which agents had been logged in but are now gone. For example, 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 the queue are forced to exit, and no new callers are granted access to the queue.
Report Hold Time	The Report Hold Time option tells the agent how long the call was holding in the 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 is not blindsided.

Service Providers

The CSG generically identifies the carriers used to reach your local phone carrier, long distance carrier, or VoIP provider as *service providers*. The service providers may use analog lines for your carrier or IP connections for your VoIP carriers. You can assign service providers on the Service Providers page (see FIG. 23).
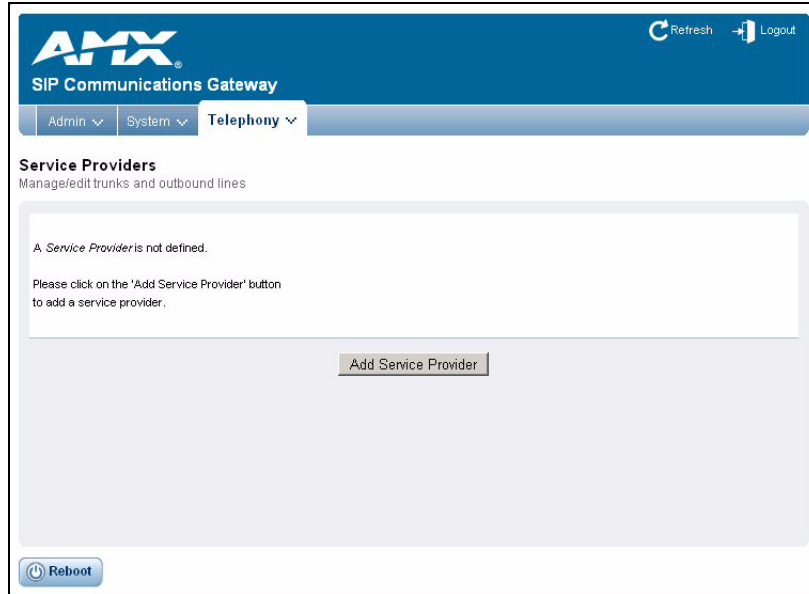

FIG. 23 Service Providers page

Service providers exist in three categories, two of which are VoIP. No selections are present on this page until you manually add your connections. To add a service provider, click **Add Service Provider**. The Add Service Provider dialog box opens (see FIG. 24, FIG. 25, and FIG. 26).
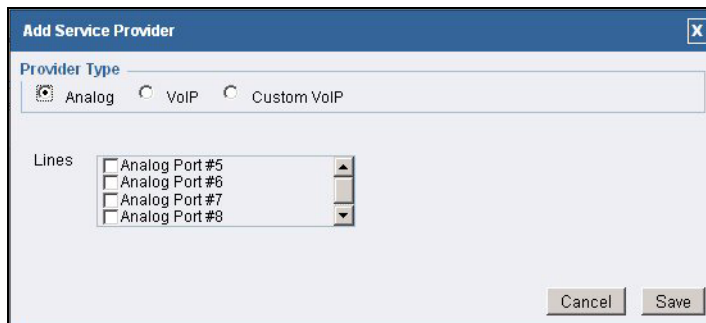

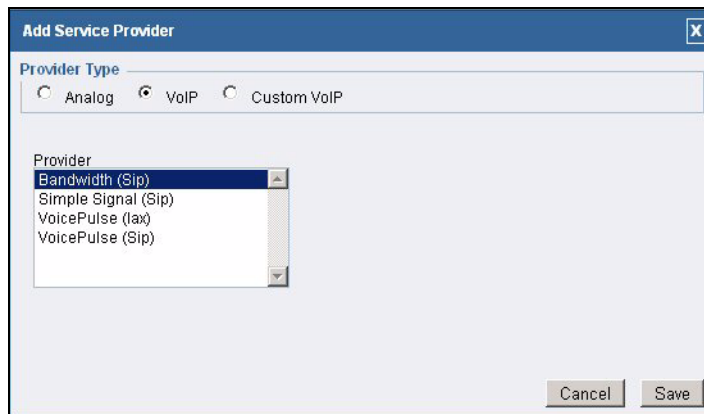**FIG. 24** Add Service Provider dialog box - Analog

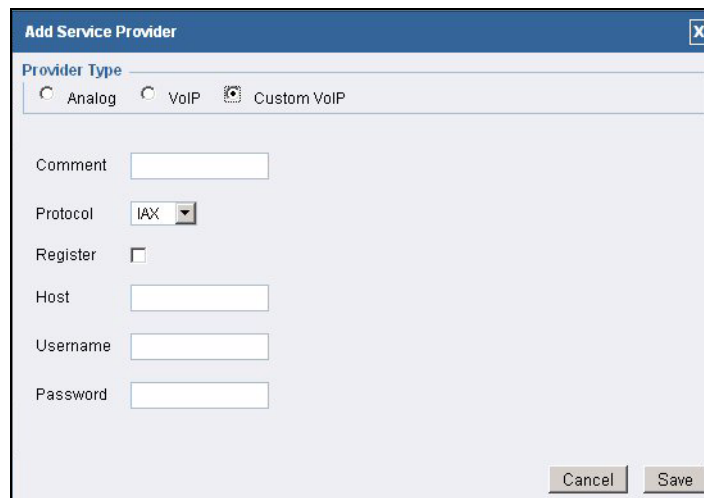**FIG. 25** Add Service Provider dialog box - VoIP

FIG. 26 Add Service Provider dialog box - Custom VoIP

Selecting the Analog, VoIP, or Custom VoIP radio button changes the options listed here to accommodate the telephony being used. Once you select a provider type and configure the provider type as necessary, click **Save** to return to the Service Providers page. The service provider now appears on the page. You can set advanced options for the service provider and allow or disallow certain codecs by using the Options menu.

If you configure an analog service provider, you must restart the CSG after clicking **Save**. *If you calibrate your phone lines before saving, you must first log off the CSG, log on again, then restart the unit.*

Call Rules

The Call Rules page enables you to use basic pattern matching to differentiate calls and route them accordingly. The page displays a rundown of every calling rule you establish and the service providers you assign to it. FIG. 27 displays the Call Rules page.
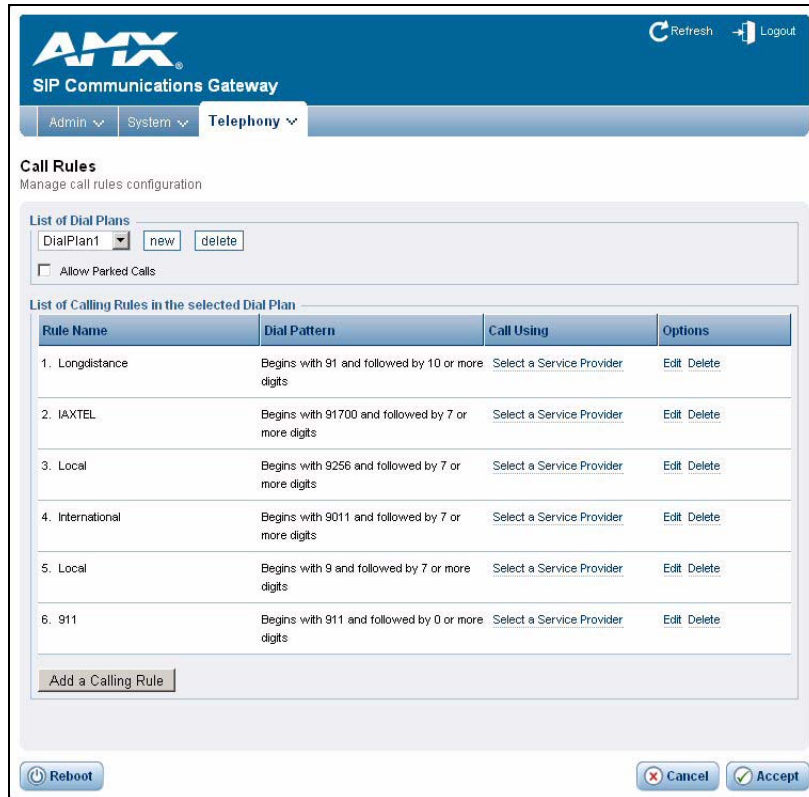
FIG. 27 Call Rules page

The CSG provides six default call types. You can edit or delete each default call types by using the hypertext next to the calling rule. The table provided also identifies whether a Service Provider (trunk) has been assigned to process the calls. The trunk assigned to handle the calls can also be changed when the Call Rules are edited. The default call types for North America are listed in the following table.

Call Types	
Call type	**Description**
Long distance	This set of call rules is based on traditional dialing and may include calls that are not truly long distance. An overlay is where two area codes function within the same geographic area. In this case, a call requiring ten digits may be local, but it is routed by the first set of call rules.
IAXTEL	This set of call rules identifies calls sent over your IAXTel.com VoIP account.
Local	This set of call rules identifies calls to your local area code and are routed as you specify.
International	This set of call rules defines how calls with a 011 prefix are routed.
Local	This second set of call rules processes outbound calls dialed with only seven digits.
911	This set of call rules define the trunk group to which your emergency 911 calls are routed. Your VoIP carrier may not yet provide 911 service so you would have to send those calls over your analog carrier.

The routing defaults for the CSG are a basic feature for the jurisdiction of calls. Calls to Canada, Guam, and the Caribbean are international, but you can reach them by dialing 1 + the area code and a seven-digit phone number, exactly like calling to another state. Similarly, calling a phone number within your area code, but to a location more than 13 miles away, is technically not a local call and will probably be billed by your carrier as a local toll, or possible even a long distance call. Call rules for locations outside of North America are not currently implemented in the user interface.

Creating a Call Rule

You probably will not need to hone your calling rules more than these defaults, but if you do, you can establish your own logic in by creating a new call rule. Follow these steps to create a new call rule:

1.   Click the **Telephony** tab on the main page and select **Calling Rules**. The Call Rules page opens (see FIG. 27).

2.   Click **Add a Calling Rule**. The Add Calling Rule dialog box opens (see FIG. 28).
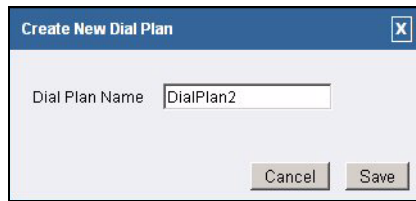

FIG. 28   Add Calling Rule dialog box

3.   In the **Rule Name** field, enter a name for the call rule.

4.   Using the **Place this call through** drop-down combo box, select a service provider. No options appear if you did not set up at least one service provider on the Service Providers page (see the *Service Providers* section on page 38 for more information).

5.   In the **Dialing Rules** section, enter a number pattern in the first available field and the number of digits to follow this number pattern in the second field. Any number dialed that meets the criteria you set in this step will have the rule you establish in the next step appended to it.

6.   Enter the number of digits to strip from the beginning of the dialed number in the next available field.

7.   Click **Save** to save the call rule and return to the Call Rule page.

Creating a Dial Plan

A dial plan holds several different types of call rules. Follow these steps to create a new dial plan:

1.   Click the **Telephony** tab on the main page and select **Calling Rules**. The Call Rules page opens (see FIG. 27).

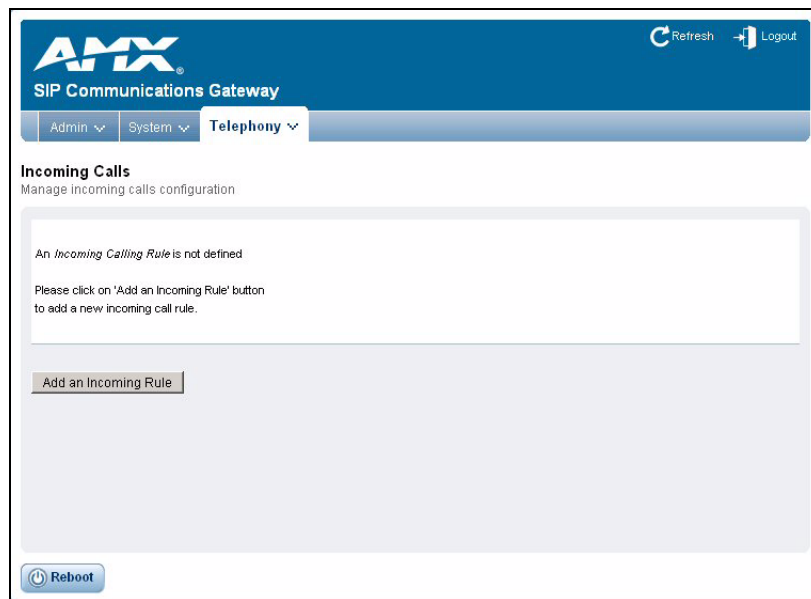2.   In the **List of Dial Plans** area, click **New**. The Create New Dial Plan dialog box opens (see FIG. 29).

FIG. 29 Create New Dial Plan dialog box

3. In the **Dial Plan Name** field, enter a name for the dial plan.

4. Click **Save** to save the dial plan and return to the Call Rule page.

Incoming Calls

The same pattern-making logic used for processing outbound calls can also be used for inbound calls. Incoming call routing is based on whether the call matches or does not match a pattern you prescribe. FIG. 30 displays the Incoming Calls page.

FIG. 30 Incoming Calls page

Creating an Incoming Call Rule

Follow these steps to create an incoming call rule:

1. Click the **Telephony** tab on the main page and select **Incoming Calls**. The Incoming Calls page opens (see FIG. 30).

2. Click **Add Incoming Rule**. The Set Rule dialog box opens (see FIG. 31).
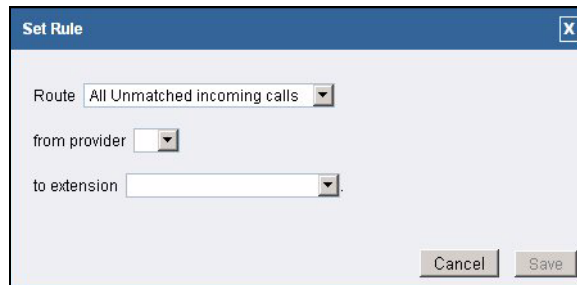
FIG. 31 Set Rule dialog box

3. Select an option to route from the Route drop-down combo box. You can choose from **All Unmatched incoming calls** or **Incoming calls that match**.

4. Select a service provider from the **From provider** drop-down combo box. No options appear if you did not set up at least one service provider on the Service Providers page (see the *Service Providers* section on page 38 for more information).

5. Select an extension to which you want these calls routed from the **To extension** drop-down combo box.

6. Click **Save** to save the incoming call rule and return to the Incoming Calls page.

Voice Menus

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

Voice menus are constructed depending on your needs. 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 the CSG.
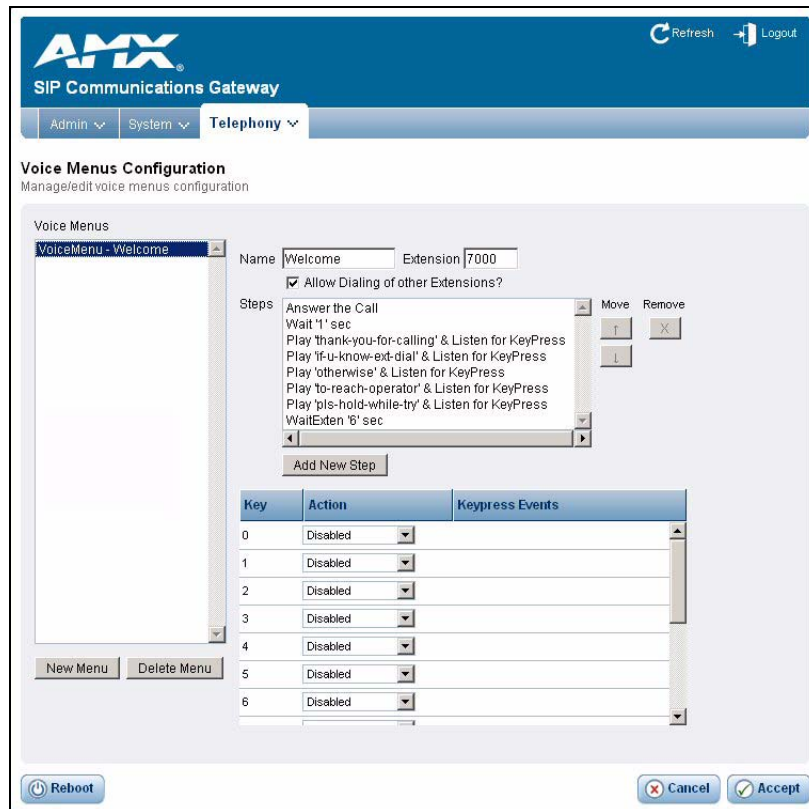
FIG. 32 Voice Menus Configuration page

Click **Telephony | Voice Menus** to open the Voice Menus Configuration page (FIG. 32), and select **Voicemenu - Welcome**. 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 responds with, "Please wait while I try that extension." If no action is taken by the caller, the menu repeats after six 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 accessed from the Add New Step dialog box (FIG. 33), which you can access by clicking **Add New Step**.
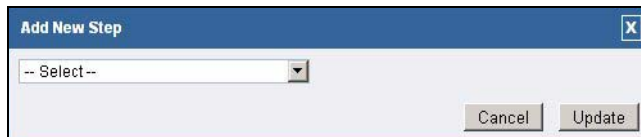
FIG. 33 Add New Step dialog box

The following table lists the available voice menu options:

Voice Menu Options	
Option	**Description**
Answer	This step is automatically added when creating a new menu. This step answers the incoming call.
Authenticate	This step is used to restrict access to one or more areas of your system. This is useful when you want users to have to enter a PIN code to proceed to a specific 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	You should select this option if there is a step in the process in which you want to play a busy signal to the caller. For instance, you might play it when the call is over.
Congestion	This option is similar to the Busytone option. You should select this option if there is a step in the process in which you want to play a congestion tone for the caller.
Digit Timeout	This option sets the maximum amount of time allowed between key presses. If a full extension is not entered in the specified time, the entry is considered invalid. A field for entering the number of seconds before the timeout appears next to the option.
DISA	This option allows callers from outside the system to obtain access to an internal dial tone and place calls from within your internal system. A passcode is required. *Note:* Use caution when selecting this option. This option can pose a security risk.
Response Timeout	If a caller does not enter a response within the time specified in this field, the call terminates. This step is often put at the end of a series of menu choices.
Playback	This option is similar to the Background option because it plays a sound file you select. However, this option does not listen for a KeyPress event, and moves on to the next step in your list.
Wait	This option pauses the execution of steps in the voice menu list for the number of seconds you specify.
WaitExten	This option gives the caller a specified amount of time to enter an extension.
Goto Menu	This option sends a caller to one of the voice menus that you specify.
Goto Directory	This 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.
Goto Extension	This option sends a caller to a specified extension. Select the extension from the available list.

Voice Menu Options (Continued)	
Option	**Description**
Dial RingGroup	This option dials a specified ring group. For example, if your menu says "Press 1 for Technical Support", the Technical Support ring group is dialed.
Check Voicemail for Own Extension	This option, as part of your voice menu, allows you to check your voicemail remotely. This option is normally preceded with a keypress event.
Hangup	This option terminates the call.

To create a new configuration for a specific extension, click **New**. The following table lists the options on the Voice Menus Configuration page:

Voice Menus Configuration Options	
Option	**Description**
Name	This field enables you to enter the name of the voice menu.
Extension	This field displays the number of the selected extension. You can assign a different extension number by entering a new number.
Allow Dialing of other Extensions	This option enables you to allow an inbound caller to break out of the listed Keypress Events and dial another extension.
Steps	This area lists the current procedural order for the handling an incoming call. You can move an individual command up or down in the list by highlighting the command and clicking the Up and Down buttons.
Add New Step	This option enables you to select assign a command for an incoming call. Many of the options require extra information. For example, the Wait command requires you to enter number of seconds for the length of the wait period. After making a selection and entering any necessary information, you can click **Update** to add the command to the bottom of the Steps list.
Action	This option enables you to assign various actions which occur when a caller presses a specific key on his phone. You can choose from **Disabled**, **Goto Menu**, **Goto Extension**, **Custom**, **Hangup**, and **Play Invalid**.
Keypress Events	This option enables you to select a specific event for certain actions you set.

The Allow Dialing of other Extensions option is important. This option enables 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 CSG 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 CSG from being compromised.

Perform the following steps as a guide to creating a voice menu:

1. On the **Voice Menu Configuration** page, click **New Menu** to create a new voice menu.

2. Enter a name and extension in the respective fields. The extension will be the direct dial number to the voice menu.

3. Click the **Allow Dialing of other Extensions** check box if you want to give a user the ability to break out of the menu selections and dial an extension within your system.

4. Click **Add New Step** to open the Add New Step dialog box and specify the steps of your voice menu using the welcome menu example and step descriptions as guides.

5. Specify the actions for digits 0-9 as well as *, #, t, and i. The options available are:

- **Disabled** - The associated key is not enabled.

- **Goto Menu** - Pressing a key with this option sends the caller to a specified menu.

- **Goto Extension** - Pressing a key with this option sends the caller to a specified extension.

- **Hangup** - Pressing a key with this option terminates the call.

- **Play Invalid** - Press a key with this option tells the caller that he has made an invalid entry.

*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 **Accept**.

Record a Menu

If you want to record custom menu prompts for the CSG, which you can use in a voice menu, you can use the Record a Menu option. Here, you can modify several options:

- **Record Again** - Clicking this button enables you to make another attempt at recording and replacing an existing custom sound file.

- **Play** - Clicking this button opens a dialog box enabling you to enter an extension that the CSG dials and over which plays the prompt.

- **Delete** - Clicking this button deletes the selected prompt.

FIG. 34 displays the Record Custom Menus Configuration page.

FIG. 34 Record Custom Menus Configuration page

You can record a custom menu by clicking **Record a new voice menu**. Clicking the button opens the Record a new voice menu dialog box (see FIG. 35).

FIG. 35 Record a new voice menu dialog box

The following table displays the available options on this page:

Record a new voice menu Options	
Option	**Description**
File Name	This field enables you to specify the saved name of the file to be recorded.
Extension used for recording	This drop-down combo box enables you to choose which extension the CSG will dial to wait for the user to speak the prompt.
Record	Clicking this button causes the CSG to launch the call that will record a file.

Call Parking

Call parking enables you to put a call on hold and continue the conversation from another phone. When you park a call, you assign a call to one extension within a series of extensions for a set period of time. You can establish call parking settings on the Call Parking page (see FIG. 36).
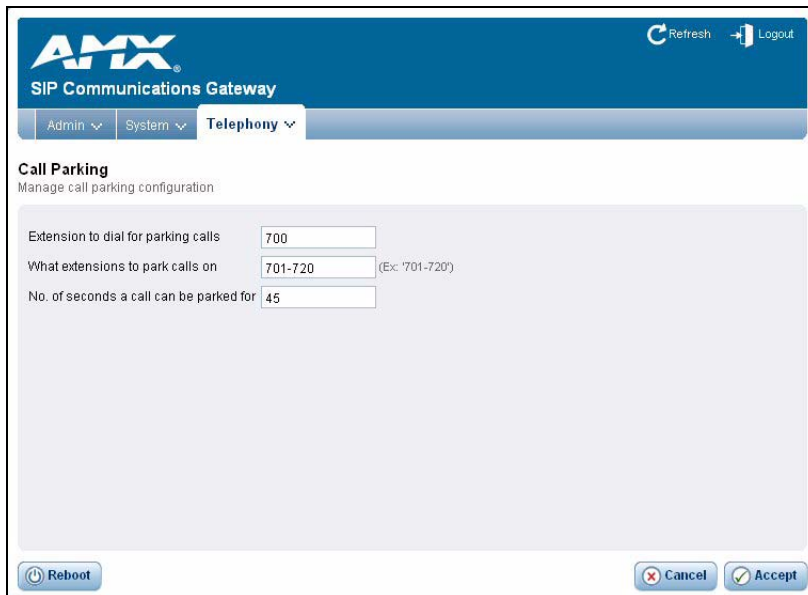

FIG. 36 Call Parking page

The following table displays the available options on this page:

Call Parking Options	
Option	**Description**
Extension to Dial for Parking Calls	This field enables you to set the extension to dial when you want to park a call.
What extensions to park calls on	This field enables you to set an extension or range of extensions to be used for parking calls.
No. of seconds a call can be parked for	This field enables you to indicate the number of seconds a call can be parked on one of the designated extensions.

You can park a call using either an analog or VoIP phone. To use an analog phone, hit the flash button, or quickly click 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 (i.e. 700), then complete the transfer (such as by click send). The method using a VoIP phone varies depending on the phone.

At this point, the CSG prompts you with a number. The number it prompts you with is the number from the pool specified. This is the number that you can enter to retrieve the call. To retrieve the call, pick up a phone and dial the parking number specified by the CSG. 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.

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.

Ring Groups

A ring group is a set of extensions you can create that ring simultaneously when a call is made to a designated extension. You must define a ring group if you plan to route inbound calls to multiple extensions. When one of the extensions in the ring group answers the call, all other extensions stop ringing. You can create a ring group on the Ring Groups Configuration page (FIG. 37).
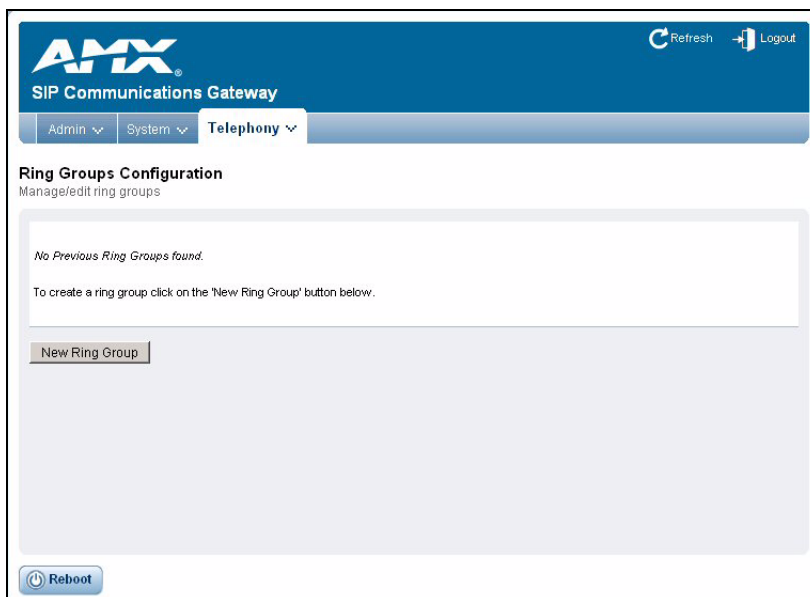

FIG. 37 Ring Groups Configuration page

Follow these steps to create a ring group:

1. Click the **Telephony** tab on the main page and select **Ring Groups**. The Ring Groups Configuration page opens (see FIG. 37).

2. Click **New Ring Group**. The Edit Ring Groups dialog box opens (see FIG. 38).

FIG. 38 Edit Ring Groups dialog box

3. In the **Name** field, enter a name for the ring group.

4. Using the **Strategy** drop-down combo box, select a ringing option. You can choose from Ring All or Ring in Order. Ring All indicates all extensions in the ring group will ring when a call arrives. Ring in Order indicates the extensions will ring in the order listed in the Ring Group Members area. Each extension will ring for the amount of time you set in step 7 before moving on to the next extension.

5. In the **Available Channels** list, select an extension you want to include in the ring group and click the left-arrow button to move the extension into the Ring Group Members list. Repeat this step until all necessary extensions are assigned to the ring group.

6. In the **Extension for this ring group** field, enter the extension for the ring group. This step is optional.

7. In the **Ring (each/all) for these many seconds** field, enter the amount of time, in seconds, you want the call to ring before other options proceed.

8. Select the **Goto Voicemail of this user**, **Goto an IVR menu**, or **Hangup** option button to indicate which option you want to occur if no one is available to answer a call. With the first two options, a combo box appears with further options for you to select.

9. Click **Save** to save the ring group and return to the Ring Group page.

Active Channels

The Active Channels Configuration page provides you with a remote view of the active calls and devices. It displays a snapshot of the activity of the server and can be refreshed to view the progression of calls. You can access this page by selecting **Active Channels** from the Telephony tab. FIG. 39 displays the Active Channels Configuration page.
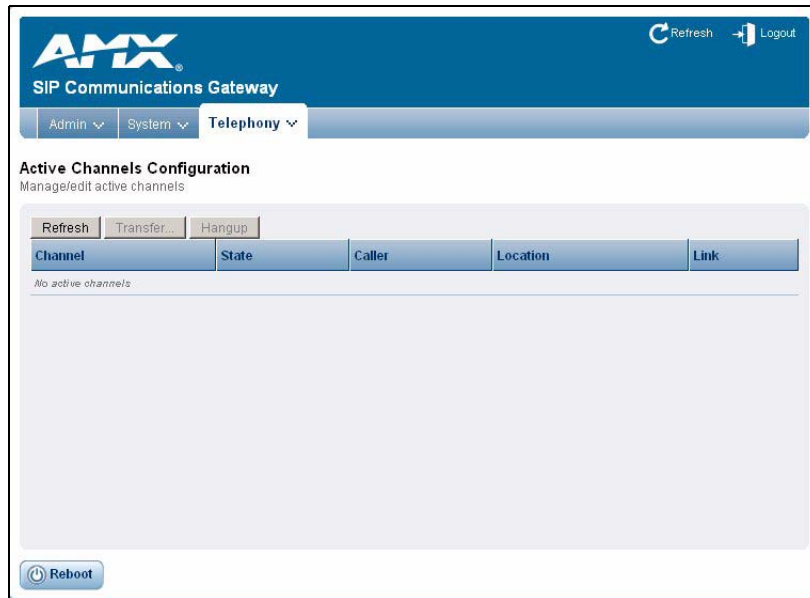

FIG. 39 Active Channels Configuration page

Touch Panel Configuration

AMX Modero panels feature on-board Setup pages. You can use the options in the Setup pages to access panel information and make various configuration changes. The MVP-8400i, NXD-700Vi, and NXD-1000Vi panels support SIP calls, so you can use these models with your CSG. This chapter only covers the pages necessary to help you configure your touch panel for use with the CSG. For more information about other features on these setup pages, consult the instruction manual for your specific touch panel at www.amx.com.

Setup Pages

On the main Setup page on your touch panel, press Protected Setup (FIG. 1) to access the Protected Setup page.

FIG. 1 Protected Setup button

The Protected Setup page provides secure access to advanced panel configuration options, including communication and security settings. Enter your password or the factory default password (*1988)* into the password keypad to access this page.

The Protected Setup Navigation buttons appear on the left of the panel screen when the Protected Setup page is active.

Other Settings Slide Out Menu

The Other Settings button (FIG. 2) provides a slide out menu with the option to select the SIP Settings page. Select **SIP** to access the SIP Settings page.

FIG. 2 Other Settings menu

The options on the SIP Settings page (FIG. 3) enable you to establish network settings for using your touch panel as an IP phone. You may need to load a Duet module to enable the touch panel to receive SIP calls. The Duet module translates between the standard interface and the device protocol. It parses the buffer for responses from the device, sends strings to control the device, and receives commands from the UI module or telnet sessions. Refer to the documentation supplied with the Duet Module for more details.

A sample UI module is provided in the module package. It is not intended to cover every possible application, but can be expanded as needed by a dealer to meet the requirements of a particular installation.
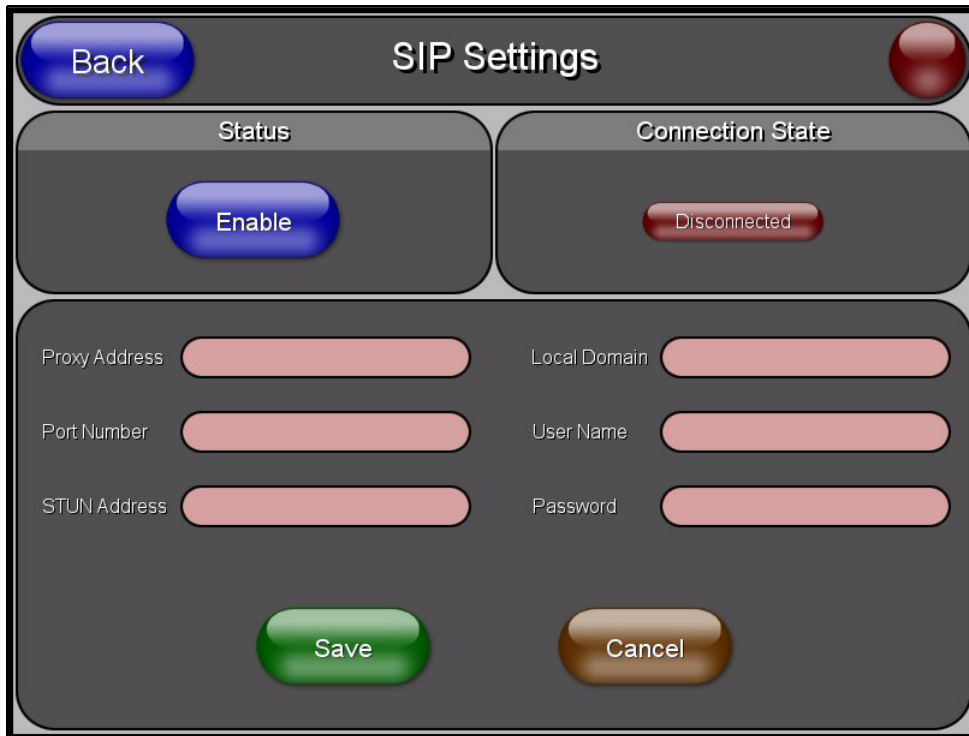
FIG. 3 SIP Settings page

The following table describes the features on this page:

SIP Settings Options	
Option	**Description**
Back	Returns you to the previous page. If you make a change on this page, the Back button appears dimmed and is unavailable until you press either Save or Cancel.
Connection Status icon	The icon in the upper-right corner of each Setup page shows online/offline state of the panel to the master. • Bright red - disconnected • Bright green - connected. Blinks when a blink message is received to dark green every 5 seconds for half a second then go back to bright green. • Bright yellow - panel missed a blink message from the master. It will remain yellow for 3 missed blink messages and then turn red. It will return to green when a blink message is received. *Note: A lock appears on the icon if the panel is connected to a secured NetLinx Master.*
Status	This option enables the SIP Stack on startup. If you disable this option, the panel will not attempt to read the rest of the configuration and will not register with a proxy server. However, point-to-point SIP will still be enabled allowing for existing intercom functionality.
Connection State	This option displays whether you are connected to the proxy server.
Proxy Address	This option enables you to enter the IP address or DNS name of the proxy server that you want to use to register. The IP address depends on whether you connect the touch panel to one of the IP Phone ports or through the LAN with the CSG.

SIP Settings Options (Continued)	
Option	**Description**
Port Number	The option displays UDP the port you use to connect to the proxy server. The standard SIP port is 5060, but some providers use different ports.
STUN Address	This option enables you to enter the IP address or DNS name of the Simple Traversal of UDP through NATs (STUN) server. STUN is used for NAT tunneling of SIP messages. This field is optional.
Local Domain	This is the authentication domain or realm used for authentication. This field is optional.
User Name	This option enables you to enter the user name used for authentication to the proxy server. Normally, the user name is the same as the phone number assigned to the extension you are using.
Password	This option enables you to enter the password for the user at the proxy server.

Setting Up Your Touch Panel to Work with Your CSG

Follow these steps to configure your touch panel to work with your CSG:

1. From the main Setup page on your touch panel, press **Protected Setup**. The Protected Setup page opens.
2. Press **Other Settings**. A slide out menu appears.
3. Press **SIP**. The SIP Settings page opens (**FIG. 3**).
4. In the **Proxy Address** field, enter the IP address or DNS name of the server you want to access.
5. In the **Port Number** field, enter the port number of the server you want to access.
6. In the **User Name** field, enter a user name you can use for authentication to the server.
7. In the **Password** field, enter the password for the user name.
8. Press **Enable** to have your panel read the configuration and attempt to connect to the proxy server.
9. Press **Save**.
10. Press **Back** to return to the previous page.

AMX

It's Your World - Take Control™