G-Series Gateways User Guide

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Telecommunication Systems

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G-Series Gateways User Guide
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Contents

Using this Document

This document is intended as a reference when installing, configuring and changing configurations of the Positron G-Series Gateway PBX units.

It is meant as a reference to the features and settings of the devices.

A “Quick Start Guide,” shipped with the units provides installation information.

A “Fast Start Guide” on the Positron Telecom Systems website at: www.PositronTelecom.com offers up-to-date, and more detailed information about the initial configuration of the systems, and the sequence used to configure the components in the shortest time possible.

This document describes various spreadsheet templates which can be downloaded from the Web Interface itself which will aid in the configuration of certain elements including user extensions and telephone sets for specific installations.

This document in PDF form has hotspots in blue to allow rapid navigation through it. Page numbers within the Table of Contents (but sadly not in the Index) are also “hot,” and will jump to the designated page. Most PDF readers will also allow text searches for terms and can display a document map to the left of the page display. External web links are also provided, allowing navigation to web sites, typically within the Positron Telecom website.

Screen captures have been provided throughout, but as features are continually added to the product line, they are subject to change.

Some features are limited to certain members of the G-Series Gateway product line and are indicated in green similar to:
G124, G122, G1212 Only:

Preparing for Installation

It is also highly recommended to format and install at least 1 GB of removable storage in USB or Compact Flash form, depending on Gateway capabilities, to provide storage of logs, voicemails and recordings. The web interface provides information on using the system to format and configure the storage devices.

Support is available through Positron Telecom's website support section at: www.PositronTelecom.com/html/support.html

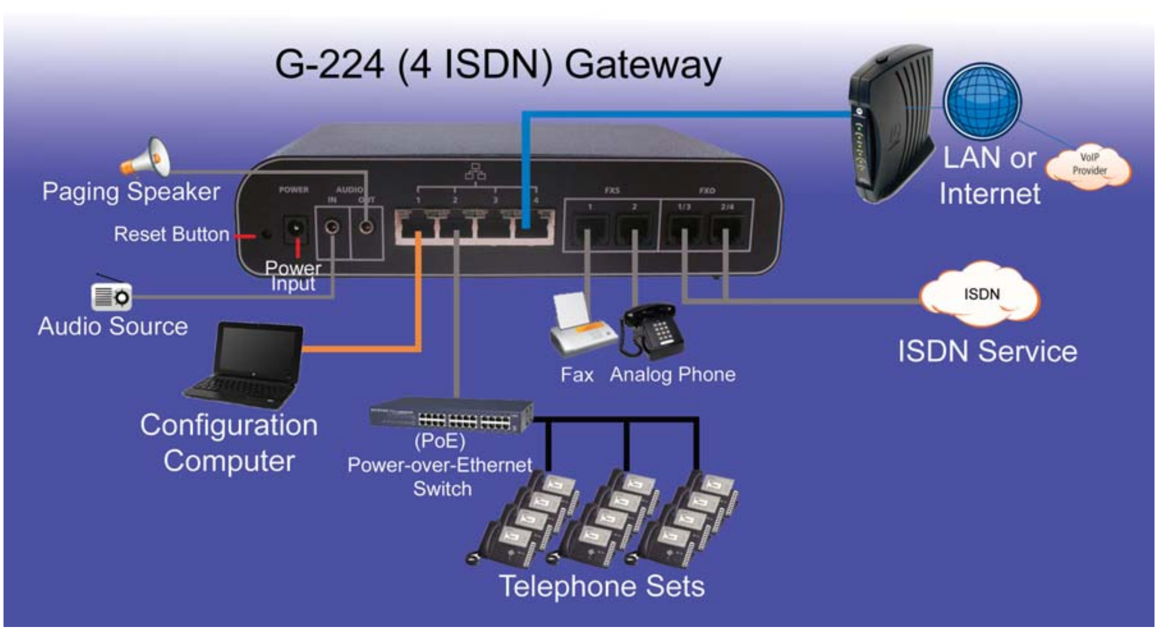
To provide local storage for files and to perform configuration, a configuration computer is required. Throughout this document we will refer to the installer`s configuration computer as the "notebook". Any computer with local file storage, a wired Ethernet jack, and running a compatible browser can be used to configure the system.

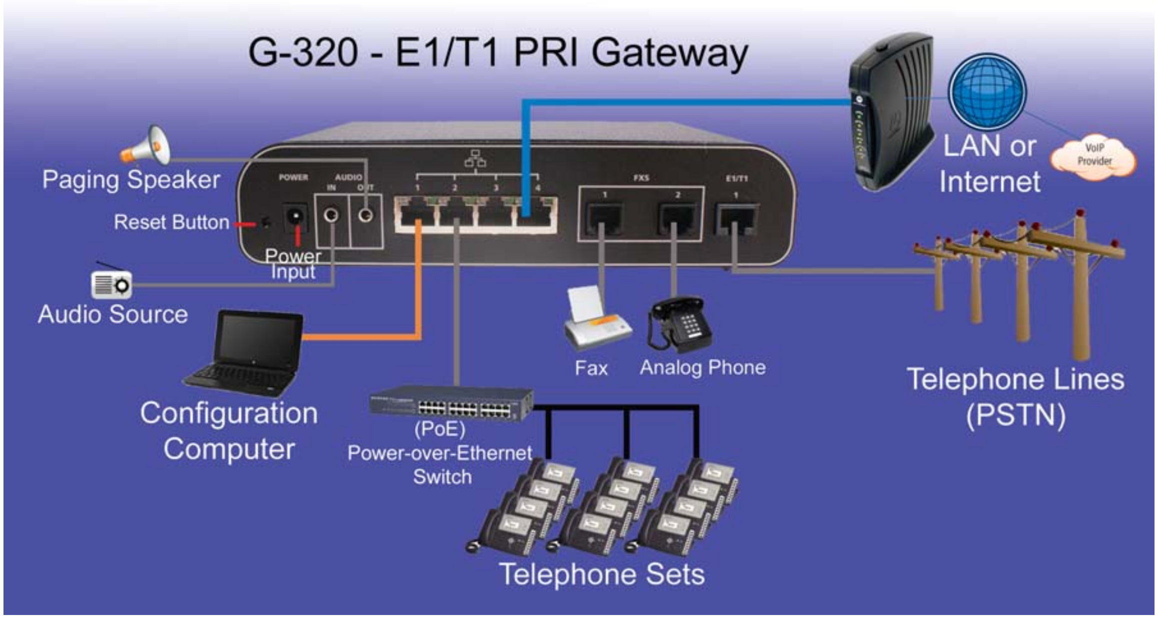
Of course, as with any computer system, it is vital that the administration username and password be changed from the default, and in many cases the IP address of the system be changed from the default. This configuration information should be noted, as it will be required as system configuration and firmware changes are made and PBX users are added and removed.

NOTE: If login information is forgotten or misplaced, contact Positron Support: www.PositronTelecom.com/html/support.html

The notebook is NOT required to run the system, and should be removed once the system is in operation in order to further secure the configuration.

Typical System Configurations




G-320 - E1/T1 PRI Gateway
Paging Speaker
Reset Button
Power Input
Audio Source
Configuration Computer
(PoE) Power-over-Ethernet Switch
Telephone Sets
Fax
Analog Phone
LAN or Internet
VoIP Provider
Telephone Lines (PSTN)


G -1212 - Analog (12 FXO) Gateway
Paging Speaker
Reset Button
Power Input
Audio Source
Configuration Computer
(PoE) Power-over-Ethernet Switch
Telephone Sets
Fax
Analog Phone
LAN or Internet
VoIP Provider
Telephone Lines (PSTN)

The Gateway Web Interface

The Web Interface is the main way to communicated with the G-Series Gateway units.

- **To connect to the Web Interface:**

NOTE: Throughout this document we will refer to the installer`s computer as the notebook. Any computer with local file storage, a wired Ethernet jack, and running a compatible browser can be used to configure the system.

- Verify that the steps outlined in the Quick Start document accompanying the unit have all been performed and that the Circle LED on the G-Series Gateway is glowing steadily red.
- Connect a notebook computer to Ethernet Port 1 on the G-Series Gateway using a standard Ethernet cable. If necessary, Ports 2 or 3 could also be used, or even a free port on an Ethernet switch connected to the same network. In this document we will consider the notebook to be connected to Port 1.

Port 4 has special configuration abilities and should NOT be used for connection to the notebook

- Set the notebook's Static IP address to any address other than `192.168.1.2` . (For example: `192.168.1.200`)
- Use the notebook's browser to access `http://192.168.1.2`.

Login Screen

- The Login screen of the G-Series Gateway Web Interface should appear.

- In the Username field, type `admin`
- In the Password field, type `password`
- From the language pull-down menu, you can select the language used for the Web Interface.
  Default: English

NOTE: It is recommended that the password be changed once configuration is complete as it is necessary to secure the Web Interface against malicious intrusion.

To change the administration account information see:
System -> Admin Account

➢ **To retrieve the IP Address of a G-Series Gateway**

- Dial the IP Address Extension

If an IP Address Extension has not been set, then dial ##** The default IP Address Extension is set in:
PBX -> PBX Settings -> Call Handling tab

- Enter the password. Default password: `1234567890#`
- The Gateway's IP Address will be presented by voice

NOTE: If login information is forgotten or misplaced, contact Positron: www.PositronTelecom.com/html/support.html

Status Screen

The **Status** screen provides system information. It is presented automatically after login or can be accessed by clicking the **Status** link on any screen.

NOTE: The Status screen must be refreshed manually to show updated information.

➢ **To refresh the Status screen**

- Click Refresh in your browser, or type Ctrl + R
- Click **Status** in the upper left corner of the Web Interface

The **Status** screen information includes:

VoIP Section

Provides the status of VoIP trunks or lines, showing which are present and their status.

➢ **To configure VoIP trunks and lines see:**

- [PBX -> Trunks / Lines -> Trunks / Lines.]()

Status messages for the VoIP section include:

- Registered – Indicates a valid connection to a VoIP carrier
- Unregistered – No connection to VoIP carrier
- UNREACHABLE – unable to reach service
- OK (xx ms) – a listed service was reached in xx milliseconds

Phones Section

Provides the status of a phone, showing which user extensions are present and their status.

- **To configure user extensions see:**

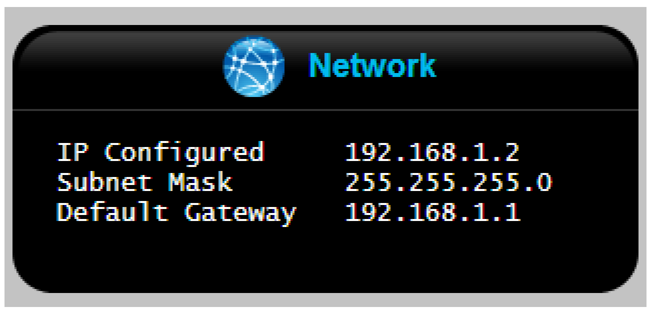
  [PBX -> Users -> Users menu](#)

- **To provision telephone sets see:**

  [PBX -> Users -> Phone Provisioning menu](#)

Network Section

The Network Section provides the gateway's IP address and network configuration information. The **IP Configured** field also represents the current IP address of the Web Interface.



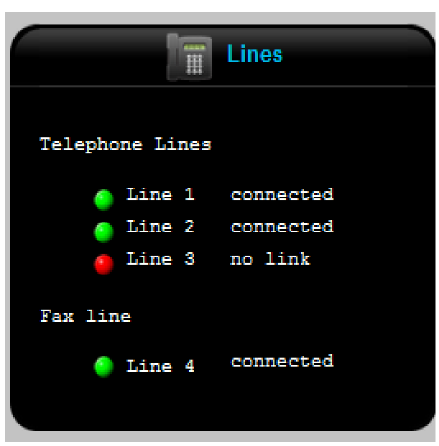
Lines Section

Shows the status of the Gateway's lines

G124, G122, G1212 Only

The status of FXO lines is shown with analog phone and fax shown separately. Possible states include:

- `connected`: Physically connected.
- `no link`: No physical connection.



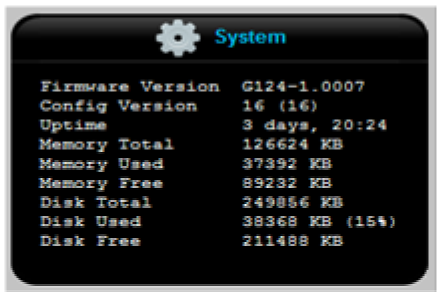
- **To configure lines see:**
  PBX -> Trunks / Lines -> Trunks / Lines menu

System Section

Shows firmware version, uptime measured in days, hours minutes, on-board memory and disk space used.

NOTE: During initial installation, it is recommended to download and install the latest firmware version from Positron: www.PositronTelecom.com/html/support.html

NOTE: New firmware should be installed ONLY AFTER a system backup has been performed and saved on the notebook computer.



Firmware Version:

Lists the current G-Series Gateway firmware version.

- **To check for the latest firmware and to upgrade firmware:**
  - See: System -> Maintenance -> Firmware Upgrade tab

Config (uration) version:

Displays the current system configuration version. Shown in the format xx (yy). Both xx and yy are normally identical. This information is generated by the system firmware. In the event of problems following firmware upgrade, this information will be used by Positron support in restoring the system configuration version.

Uptime:

System uptime since last reboot/restart, measured in days, hours minutes, seconds.

Memory Total, Used and Free:

Provides system main memory (RAM) usage information. In the event of abnormal system activity, the amount of free memory may begin to decrease resulting in degraded or erratic system operation.

When contacting Positron Telecom for support, it is recommended to have available:

- The Positron Telecom product name and model number

- The version of firmware currently installed
- The configuration version – both numbers xx and yy
- The amount of free memory as shown in the latest **Status** screen

Disk Total, Disk Used and Disk Free:

Disk refers to the system storage disk - onboard system storage used for the local file system.

NOTE: The disk memory referred to is fixed in size and located on the system circuit board. It is NOT the CF or USB removable storage.

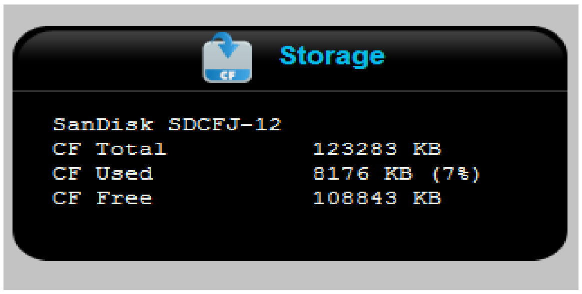
G122 Only:

The G-122 does not support Compact Flash.

Storage Section

Auxiliary flash memory in the form of Compact Flash (CF) card and USB storage is used for voicemails, voice recordings and system logs.

USB memory can be in the form of a keychain thumb drive or hard disk storage.

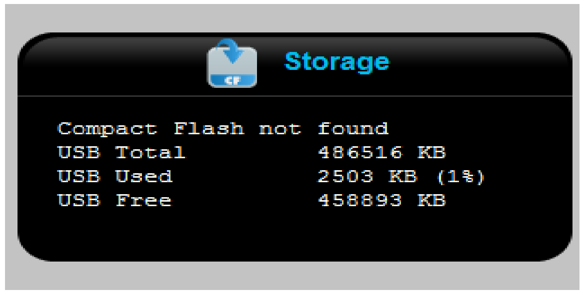


CF Card (only), formatted with available storage

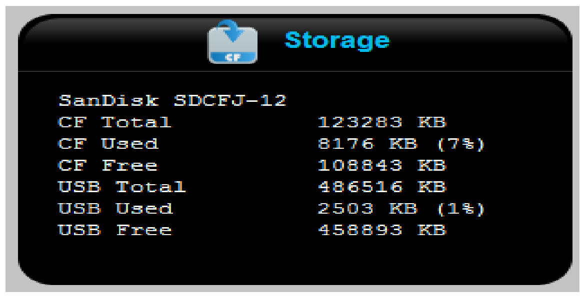
Shown above is the system status with formatted CF memory.

Shown below is a similar dialog box, indicating that CF memory was not found, yet had been configured for storage of user information. Memory used with the system must be formatted.

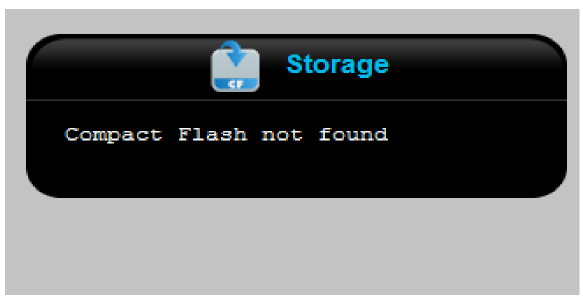
- **To format CF or USB memory see:**
  System -> Maintenance -> Storage tab



USB Storage only



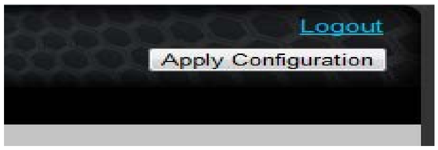
CF Card and USB with available storage



Neither CF nor USB storage found

Logout Link

The **Logout** link, located above the **Apply Configuration** button logs the administrator or user off the system and then presents the **Login** screen. It should be used when the configuration notebook is left alone and there is a possibility that it may encounter unauthorized use. By clicking **Logout**, the username and password will have to be reentered.



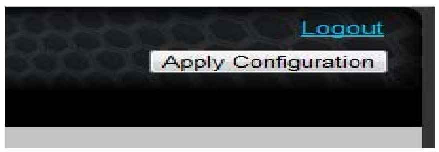
Logout link

There is a system timeout on logins. If there has been no activity for some time, the system will automatically log the notebook off.

Apply Configuration Button

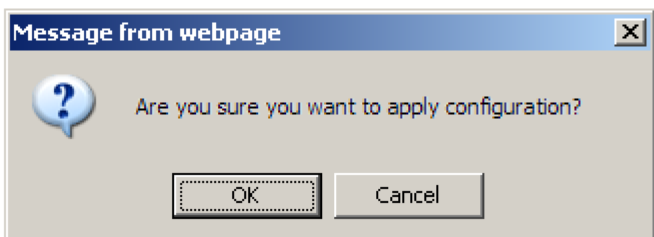
The **Apply Configuration** button applies configuration changes to the system files as specified in the Web Interface. It automatically creates or updates the existing time stamped backup file called 'lkgc' (Last Known Good Configuration). In the event that a system restore is required, this file can be restored.

NOTE: Until **Apply Configuration** is clicked, changes made to configurations will not be applied.

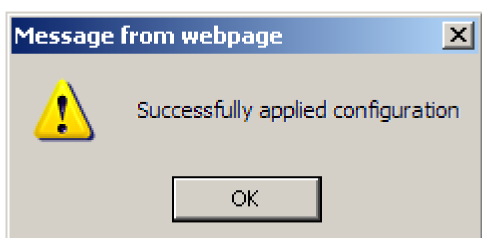


Apply Configuration button

A confirmation dialog box will appear after the button has been clicked: (appearance may differ)

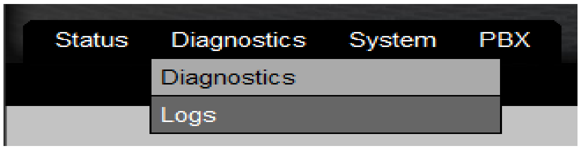


A confirmation dialog will indicate whether the configuration files have been applied, or that updating the configuration has failed.



Diagnostics Menu

The **Diagnostics** Menu provides two submenu choices – **Diagnostics** and **Logs**

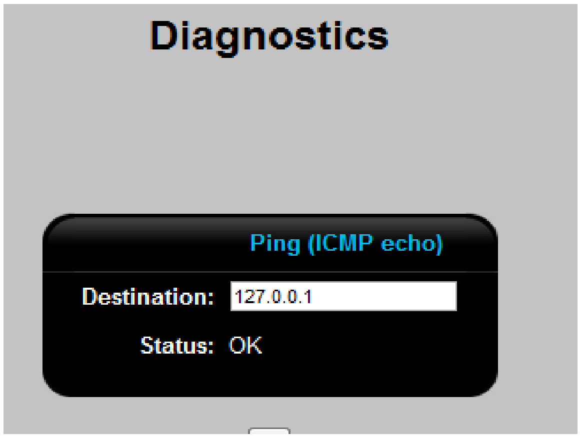


The **Diagnostics** (sub) Menu provides three tabs:

- Ping Used to verify system network connection
- Console Used to verify system operation
- Command Used to execute Asterisk commands

Diagnostics -> Ping Tab

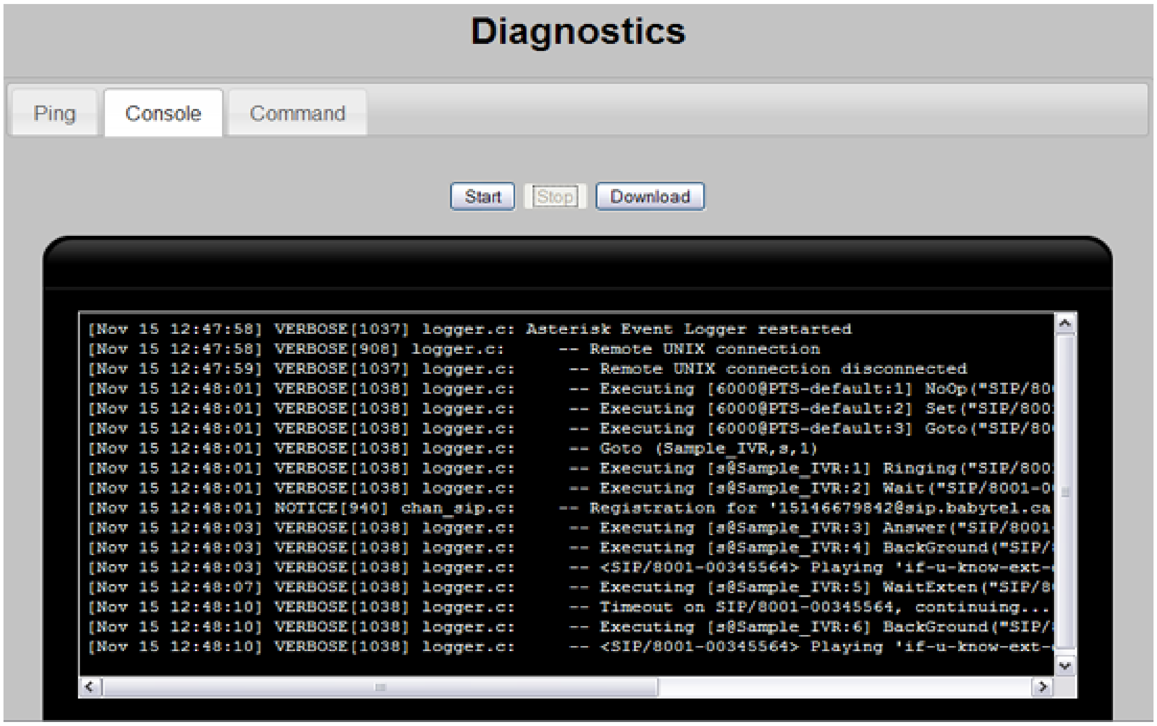
The Ping tab offers a way to verify a connection with the IP network and to determine whether a particular IP phone or VoIP provider is reachable.



- **To verify that the Gateway is connected to a network:**
    - Enter the IP address or URL of a device that is known to be active.
      Example: Google.com
    - Click **Go**. An Internet Control Message Protocol (ICMP) message is sent to the specified address.
    - If the destination address is active the **Status message** will be "OK."
    - If the destination address is not active, the message will be "No response."

Diagnostics -> Console Tab

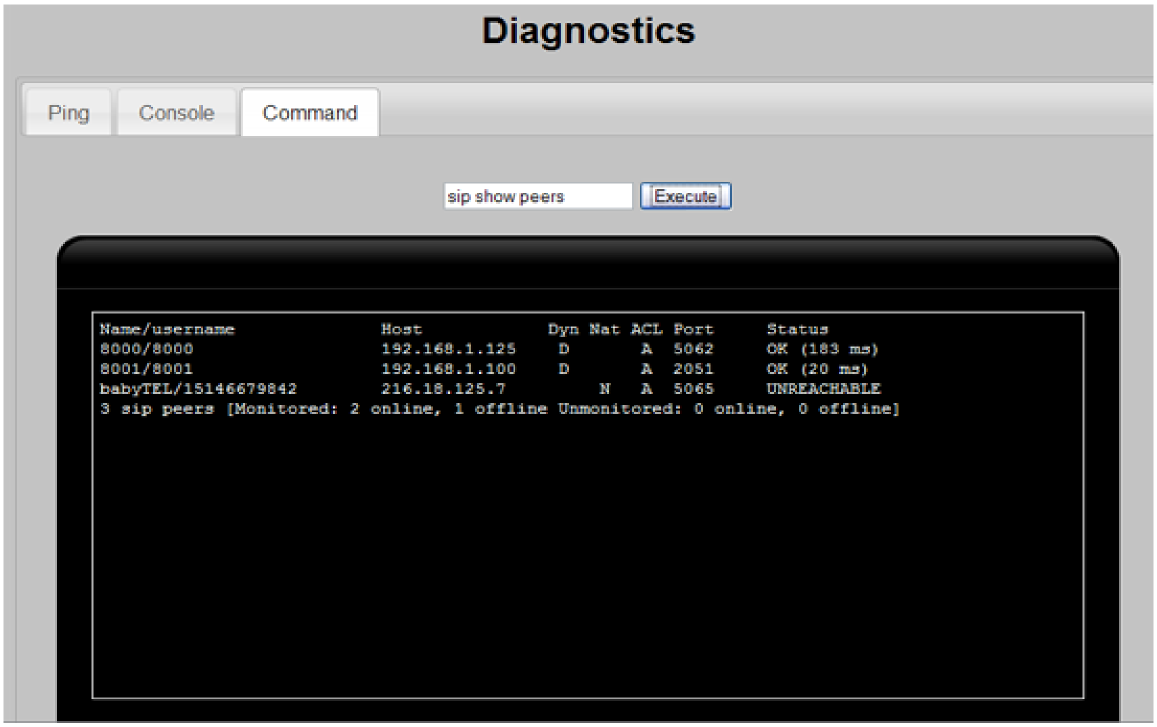
The **Console** tab is available to isolate system messages following a particular activity. Used in diagnosing a problem, the **Console** tab will show that part of the system messages which is the result of a specific operation.



- **To isolate system messages**
  - Click **Start**
  - Perform a system activity which is causing a problem
  - Click **Stop**
  - The **Console** log will present any system messages generated between clicking **Start** and **Stop**
  - This abbreviated log can be downloaded for analysis or sent to Positron

Diagnostics -> Command Tab

The **Execute** button permits the execution of Asterisk commands

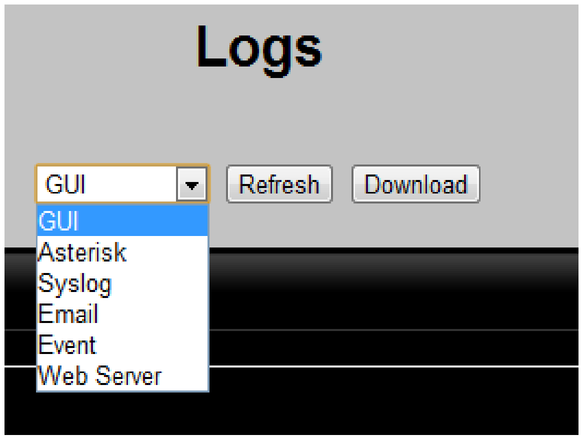


- **To execute an Asterisk command**
  - Enter the command in the text box
  - Click **Execute**

Diagnostics -> Logs Menu

Displays:

The **Logs** Screen displays information generally used in system administration and debugging.



A drop-down menu provides views of the following system logs:

- GUI (Graphical User Interface)
- Asterisk
- Syslog
- Email
- Event
- Web Server

It is good practice to occasionally click **Refresh** to view the most up-to-date log information

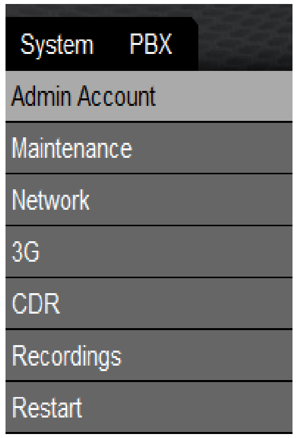
➢ **To view a log**

- Select the appropriate log from the drop-down menu
- Click **Refresh**

➢ **To download the files in tar.gz format to notebook storage:**

- Click **Download**
- Specify the download location in the local file system on the notebook

System Menu

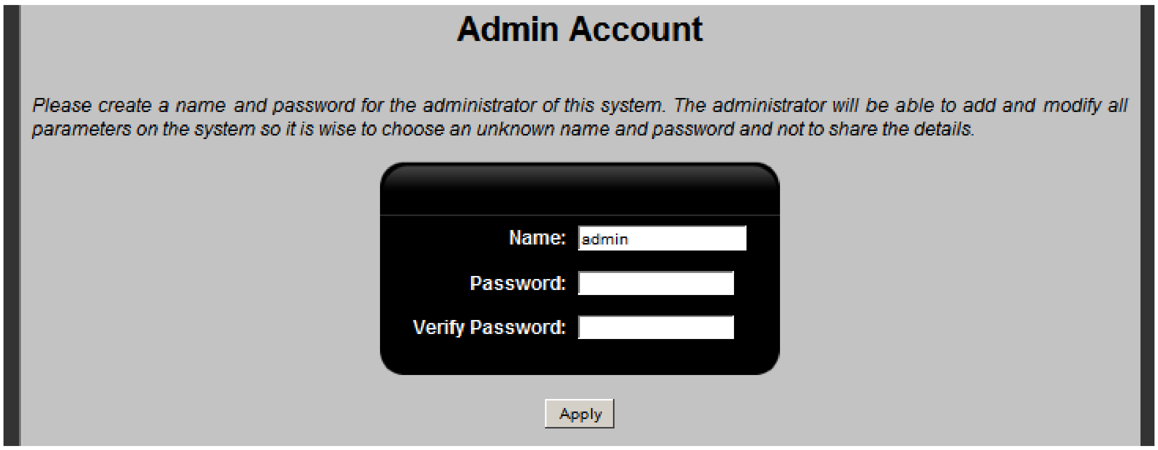


The **System** Tab provides access to:

- Admin Account to set system username and password
- Maintenance to set date and time information, and provides tabs for: firmware upgrades, resetting to default settings, backup/restore functions and formatting CF memory.
- Network to configure time zones, system servers and IP addresses
- 3G to enable and configure cellular-based 3G voice and data services
- CDR to display Call Detail Records
- Recordings to list and control any recorded calls
- Restart to reset the system

System -> Admin Account Menu

This area is for configuring the administrator account. The menu allows for the setting and changes to the username and password which can be used to secure the Web Interface.

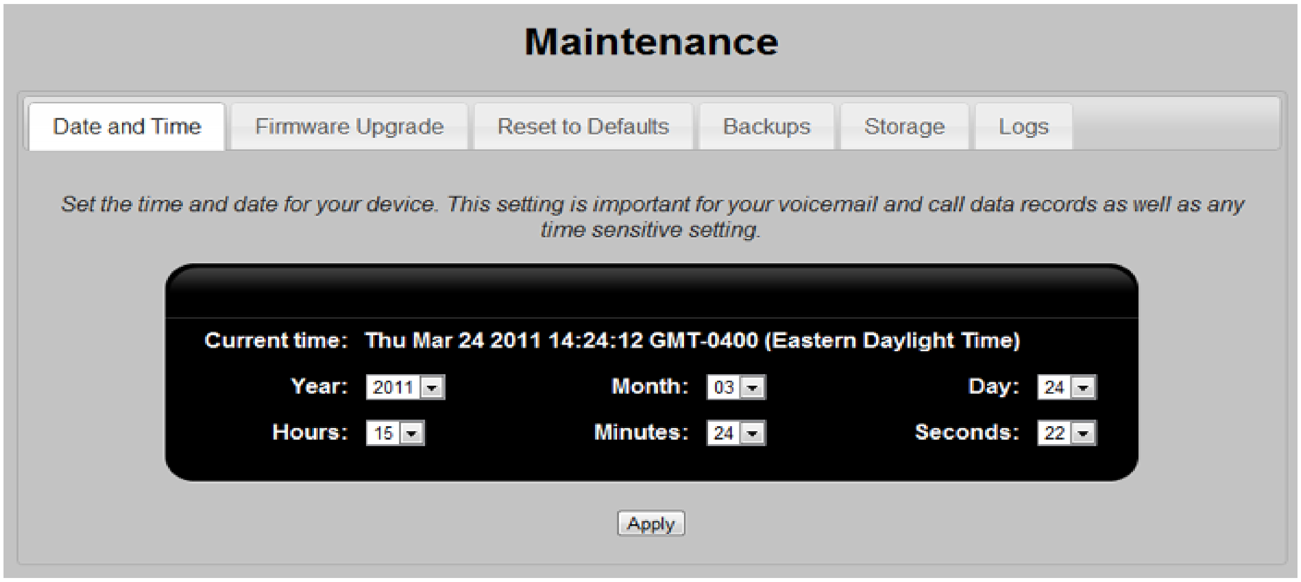


- **To change the administrator account password**
  - Enter the new password in the **Password** and **Verify Password** text fields
  - Click **Apply**
  - Click **Apply Configuration**

NOTE: In the event that the login information is forgotten or misplaced, please contact the Positron Telecommunication System's support team. www.PositronTelecom.com/html/support.html

NOTE: Only one administrator account is available on each system.

System -> Maintenance Menu



The **Maintenance** Menu comprises:

- Date and Time tab To allow for the setting of the time and date.
- Firmware Upgrade tab To allow for upgrading the firmware from the network or local storage.
- Reset to Defaults tab To allow resetting the system through the Web Interface.
- Backups tab To control system backups and restores.
- Storage tab To format and test auxiliary CF and USB storage.
- Logs tab To display and download system logs.

System -> Maintenance -> Date and Time Tab

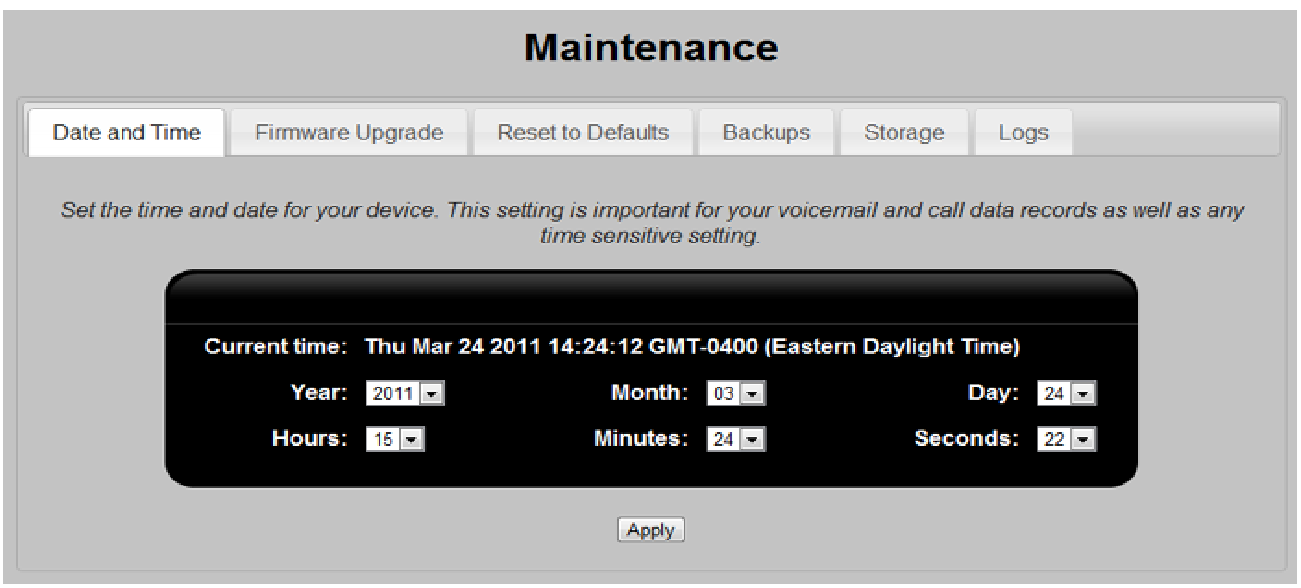
Time information is important as it provides the time stamps for voicemail, recordings, backups and other system operations. Normally, when connected to the Internet, the unit can check with a time server (NTPD) for the correct time, and, assuming the time zone has been set correctly, the system should be set to the correct time. When the NTPD server is enabled AND the NTPD client is UNREACHABLE or UNDEFINED, the real-time clock will be used in its place to serve time to the system, and the system then serves time to the phones.

- **To configure NTPD settings:**
  - See: System -> Network -> Network Tab

G122 Only:
All G-Series Gateways (except the G-122) have a real-time clock.

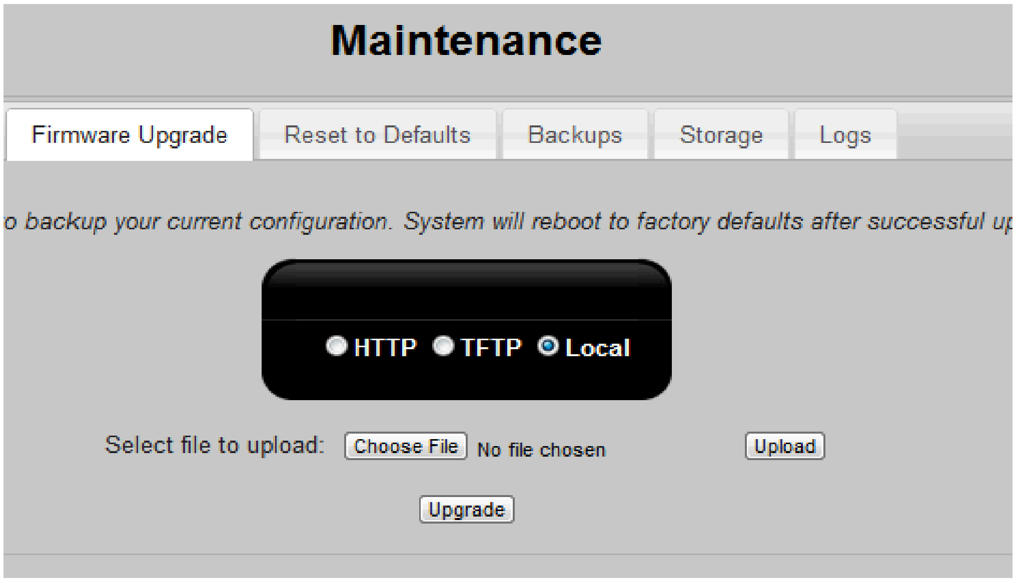
The real-time clock will also maintain correct system time in the event of a system reset/reboot.

If necessary, the system time can be set to correspond to the time in the configuration notebook, using the procedure below:



- **To set system time equal to the notebook computer's time:**
  - Click **Apply** to set the time and time zone equal to the time and date settings of the notebook computer.

System -> Maintenance -> Firmware Upgrade Tab



Firmware upgrades can be loaded from:

- The Internet through an Internet-connected browser
- Through this Web Interface
- Through a TFTP server on the network

➢ **To download and save a firmware upgrade file using an Internet-connected browser:**

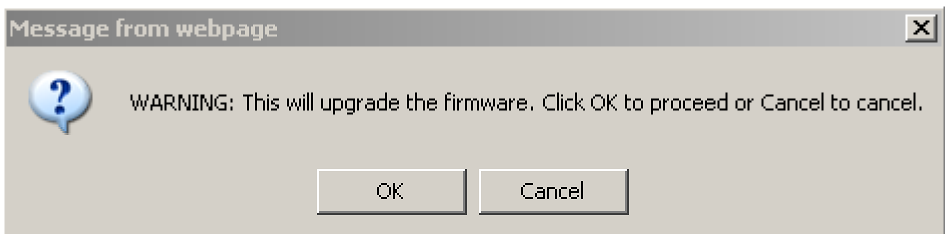
- Locate the **Firmware Downloads** icon on the following page:
  www.PositronTelecom.com/html/support.html
- Select the appropriate category for the model of G-Series Gateway
- Locate and read the **change-log.txt** page for the upgrade to verify its applicability to your situation.
- Right-click the appropriate link and do a Save Target As… in your browser
- Locate a directory for the download on your local file system.
- Do not change the filename, use the exact name specified and click Save.

When using Internet Explorer the File name should be saved as `"uImage-md5"` (with quotes), and the S**ave as type** should be **All Files**.

➢ To upgrade the firmware using the Positron Telecom Web Interface

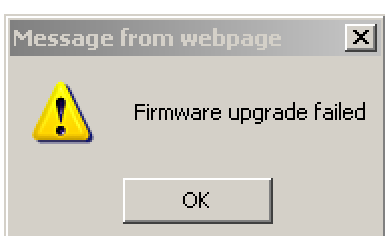
⚠	This operation will disconnect all current calls and should not be done during operating hours.

- Click on the **Local** radio button in the System -> Maintenance -> Firmware Upgrade tab
- Click **Choose File** or **Browse** and locate the downloaded firmware file on your local file system
- Click **Upload** to accept and verify the file
- Click **Upgrade** to apply the firmware upgrade
- A dialog box similar to the one below will be shown

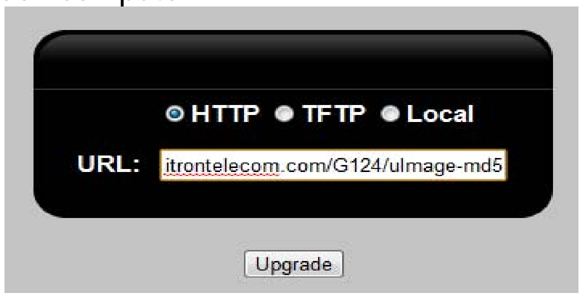


- A dialog box will indicate whether the upgrade succeeded or failed.

or



The firmware can also be upgraded in a single step, bypassing the need to save the file on the notebook computer.

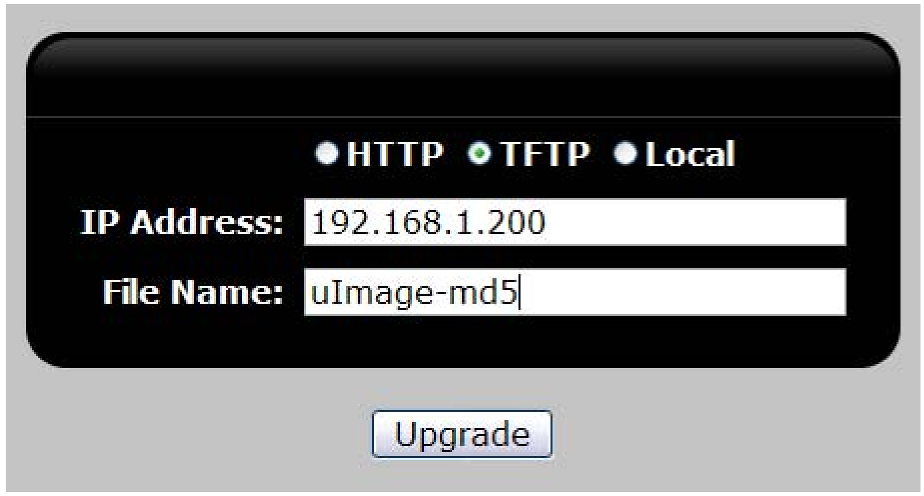


➢ **To upgrade the firmware using HTTP upgrade.**

	This operation will disconnect all current calls and should not be done during operating hours.

NOTE: This method is only possible if the G-Series Gateway has access to the internet.

- Locate the **Firmware Downloads** icon on the following page:
  www.PositronTelecom.com/html/support.html
- Select the appropriate category for the model of G-Series Gateway
- Locate and read the **change-log.txt** page for the upgrade to verify its applicability to your situation.
- Right-click the link for the appropriate upgrade file
- Select “**Copy Link Location**” in your browser
- On the Web Interface, click on the **HTTP** radio button
- Paste the link into the **URL**: text box
- Click **Upgrade**



➢ **To upgrade the firmware using TFTP:**

	This operation will disconnect all current calls and should not be done during operating hours.

NOTE: Before proceeding, be sure the TFTP server is started and has the firmware file available.

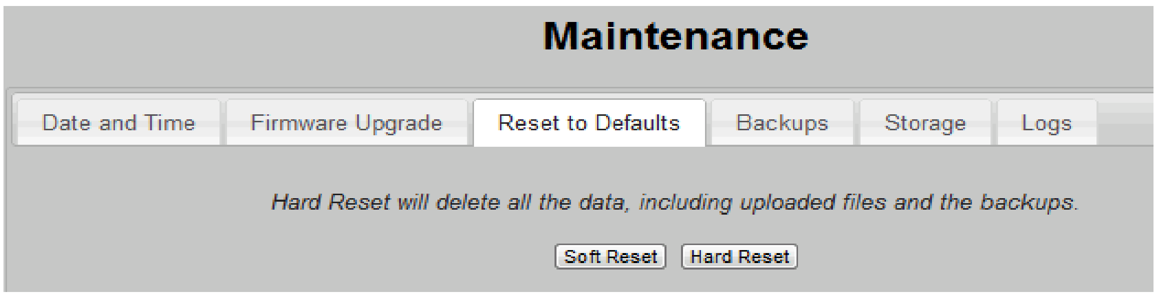
- Click **TFTP**
- Enter the IP address of the TFTP Server
- Enter the filename of the upgrade file
- Click **Upgrade**
- Dialog boxes similar to those above will be shown allowing cancellation and showing whether the operation completed successfully.

System -> Maintenance -> Reset to Defaults Tab

Clicking this tab allows the option of performing either a Hard Reset or Soft Reset.

A hard reset restores the ORIGINAL FACTORY SETTINGS to the Gateway, eliminating ALL user settings entered since configuration began including backups. However, any new firmware installed will be retained.

A Soft Reset will have the same effect, but retains any backups created.

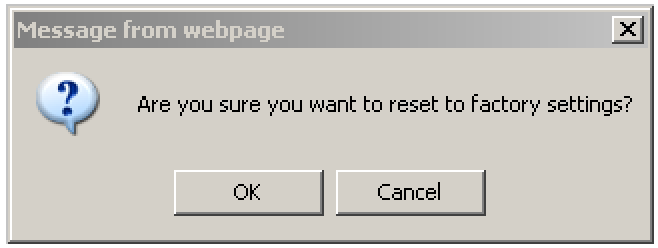


⚠	These operations will disconnect all current calls and should not be done during operating hours.

- NOTE: The Web Interface will not be available until the system has been reset.
  - Its IP address will be reset to `192.168.1.2`
  - The username is `admin`
  - The password is `password`.
- NOTE: Any backup file stored on the notebook's local file system, or the G-Series Gateway may need to be restored.
  - See: System -> Maintenance -> Backups tab
- NOTE: When a backup has been restored, the IP address, username and password will return to their configured values.

➢ **To perform a hard reset of the system**

- Backup the configuration if necessary.
- Click **Hard Reset**
- A confirmation dialog box will allow the reset to be performed or cancelled.



- Click **OK** to reset the system to its default settings.
- The system will display a countdown. The system will be reset when the unit's red LED above the circle icon on the front panel glows continuously.
- Enter the Web Interface default address of `192.168.1.2` in the browser as well as the default username `admin` and system password of `password`.

➢ **To perform a soft reset of the system**

NOTE: Backups of the configuration will remain intact.

- Click **Soft Reset**
- A confirmation dialog box will allow the reset to be performed or cancelled.
- Click **OK** in the dialog box to reset the system to its default settings.
- The system will display a countdown. The system will be reset when the unit's red LED above the circle icon on the front panel glows continuously.
- Enter the Web Interface default address of `192.168.1.2` in the browser as well as the default username `admin` and system password of `password`.

➢ **To Restore a saved backup:**

- o See: System -> Maintenance -> Backups tab

System -> Maintenance -> Backups Tab

Used to perform configuration backup and restore functions. Backups are stored on the Gateway's onboard memory. They can be downloaded as *tar.gz* files onto the notebook`s file system.

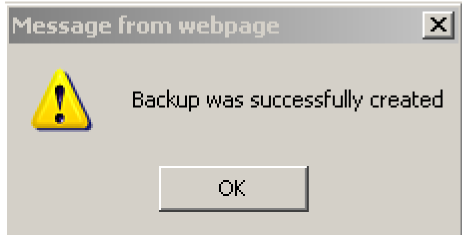
⚠	In the case that a restoration is necessary, before performing a **Firmware Upgrade** it is highly recommended that the configuration is backed up.

Backups can be created during normal system operation however firmware upgrades and system resets will terminate all calls.

➢ **To backup configuration files:**

NOTE: In the event that recent changes have been made, remember to click **Apply Configuration** to ensure that the changes will be saved as part of the backup.

- Enter the name of the backup in the **Name** box. The name cannot contain spaces, use the underscore "_" character if necessary.
- Click **Backup**. A backup file will be created in the Gateway unit's onboard memory.
- A confirmation dialog box will be presented if the backup is successful.



- The new backup will be presented in the list.
- The top menu will show the backup file, firmware version and date and offer **Download**, **Restore** and **Delete** buttons.

➢ **To restore configuration files stored on the G-Series Gateway:**

- Select the file from the list
- Click **Restore**
- During the restore process, the buttons on this page will be gray. When they return to normal color, the restore will be complete.

NOTE: Unlike the backup process, restoring cannot be performed during normal system operation.

➢ **To upload backup files for restoration via TFTP server:**

NOTE: Before proceeding, be sure the TFTP server is started and has the downloaded backup file available.

- Click **TFTP:**
- Enter the IP address of the TFTP server and the name of the restore file in the **File Name** box
- Click **Get**.
- The file can then be restored using **Restore**

➢ **To upload backup files located on the local file system:**

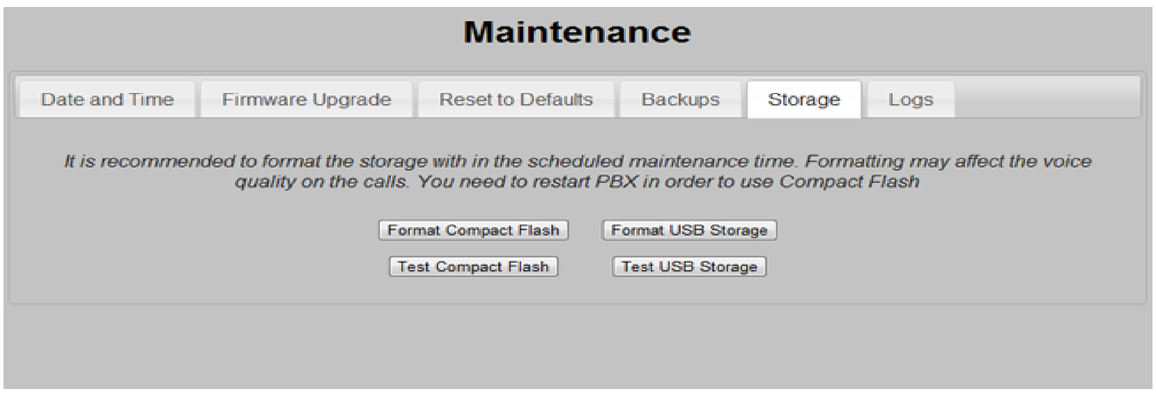
- Click **Local**
- Locate the file on the local file system by clicking **Browse**
- Click **Upload**.
- The file can then be restored using **Restore**

System -> Maintenance -> Storage Tab

Used to format and test the Compact Flash and USB storage used for voicemail, recordings and log data storage.

G122 Only:
The G-122 does not support Compact Flash; therefore USB is the only choice in the Storage section.

NOTE: If a USB or CF storage device is specified, and that storage device is not present, then the associated file will be stored on the system's memory. The information will be lost in the event of a system restart, or if there is insufficient memory available.



Format Compact Flash Button, Test Compact Flash Button
These controls are useful when first installing a Compact Flash card, or when the **Status** screen shows that the Compact Flash card is installed but not recognized.

NOTE: Estimated time for a CF card format is approximately 1.5 minutes/GB but can be dependent on the size of the CF card.

NOTE: All data on the CF card will be lost when it is formatted using this command.

A restart (**System -> Restart**) must be done after formatting in order to recognize the newly-formatted compact flash card. Any telephone calls in progress will be dropped.

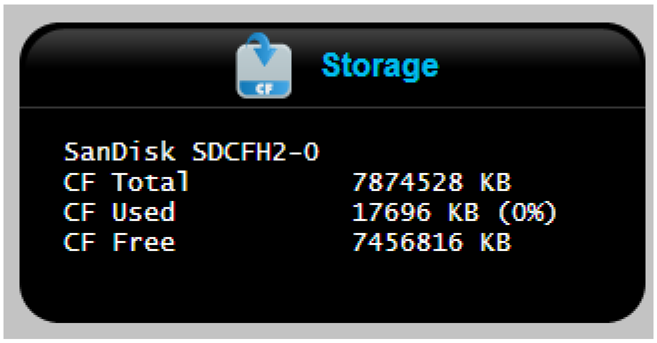
- **To Install and Format Compact Flash (CF) Storage:**
  - Unplug the power to the G-Series Gateway

- Insert the CF card into the CF slot
- Plug the power back into the G-Series Gateway
- Log into the Web Interface and navigate back to the **System -> Maintenance -> Storage** tab.
- Click **Format Compact Flash**
- Warnings may be presented to the effect that any existing data on the device will be erased when formatting
- In the event that error messages are presented during formatting, see: Diagnostics -> Logs -> GUI for details
- If the CF storage is to be put into service after testing, the system must be restarted.

➢ **To restart the system**

- See:System -> Restart Link
- Test the CF storage after formatting by clicking **Test Compact Flash**. A message will appear on the Web Interface to indicate the success or failure of the test.

Once the installed CF memory has been successfully formatted and installed, a message similar to the following will be shown in the **Status** Screen:



Format USB Storage Button, Test USB Storage Button

These controls are useful when first installing a USB drive, or when the **Status** screen shows that the USB drive is installed but not recognized.

- NOTE: It is recommended that a USB drive greater than 100 GB be formatted on computer systems rather than on the Gateway unit due to time constraints.

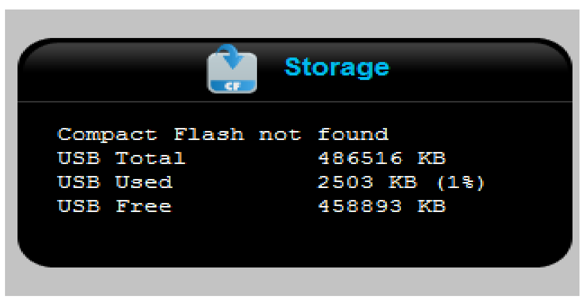
- It is recommended to format large storage devices (e.g. 100 GB+ disk drives) using utilities available on the Internet such as the Gnome Partition Editor - *GParted Live* available for download at:

  http://gparted.sourceforge.net/index.php

- NOTE: Unlike CF storage, a system restart is not necessary when USB storage is added or removed.

- NOTE: If the USB storage device is formatted on the system, once the formatting is complete, a notification will ask for the device to be **removed** and **reinserted** in order for it to be recognized by the system.

- **To Format USB Storage:**
  - Remove any existing USB devices such as a 3G data adapter from the G-Series Gateway unit.
  - Insert the USB storage into an available USB port on the G-Series Gateway unit.
  - Click **Format USB Storage**
  - Warnings may be presented to the effect that any existing data on the device will be erased when formatting
  - In the event that error messages are presented during formatting, see: Diagnostics -> Logs -> GUI for details
  - When formatting is complete, remove and re-insert the USB storage in order to have the system recognize it.
  - Test the USB storage after formatting by clicking **Test USB Storage**. A message will appear on the Web Interface to indicate the success or failure of the test.
  - If any other USB devices such as a 3G data adapter had been removed, they should be replaced
  - Once the USB storage has been successfully formatted, the **Status** screen will present storage information:

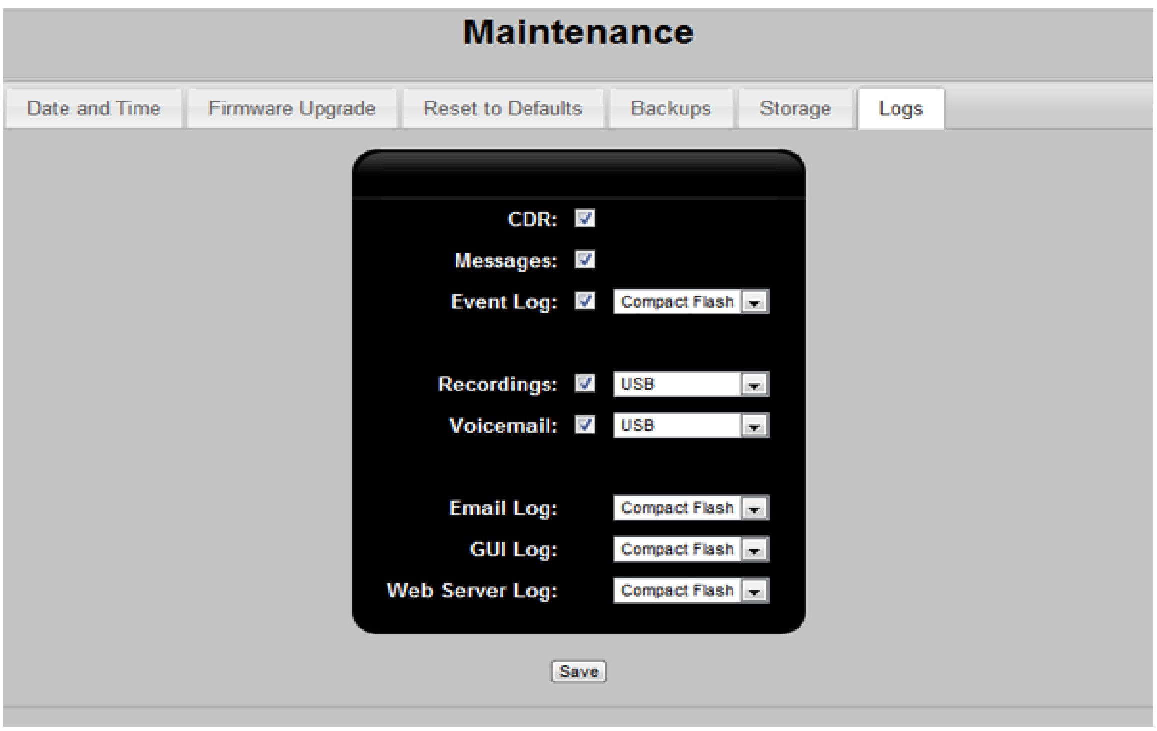


System -> Maintenance -> Logs Tab

This menu allows the specification of which logs will be created and their location on storage devices. Drop-down menus are presented listing possible storage locations, either CF or USB.

G122 Only:
The G-122 does not support Compact Flash; therefore USB is the only choice in the Logs section.

- NOTE: Data will not be captured until the checkboxes have been selected and **Apply Configuration** has been clicked.
- NOTE: If a USB or CF storage device is specified, and that storage device is not present, then the associated file will be stored in the system memory. The information will be lost in the event of a system restart, or if there is insufficient memory available.



- NOTE: It is highly recommended to store the following logs in the event that they are required for system diagnosis:

- CDR: (Call Detail Records)
- Messages: (var/log) Asterisk messages
- Event Log
- GUI Log
- Web Server Log

Event Log:
Asterisk event logs will be stored.

Recordings:
If enabled they will be stored on the selected device.
To enable Recordings to be made, see:
PBX -> PBX Settings -> Recordings tab.

Voicemail :
If voicemails are enabled on the system, they will be stored on the selected device.

➢ **To avoid storing voicemails on the G-Series Gateway**

- Enable the setting for Send Messages by Email Only: in: PBX -> PBX Settings ->Voicemail tab

Email log:
Log of all messages that have been sent to the organization's email system - usually voicemail messages attached to emails. The log is useful in troubleshooting email problems.

➢ **To disable logging of emails to either CF or USB**

- Select RAM from the drop-down menu.

NOTE: Logs stored in RAM will be lost in the event of a system restart.

GUI log:
Contains a log of Web Interface messages typically includes Web Interface validation or upgrade errors.

➢ **To disable logging of GUI messages to either CF or USB**

- Select RAM from the drop-down menu.

NOTE: Logs stored in RAM will be lost in the event of a system restart.

Web Server log:

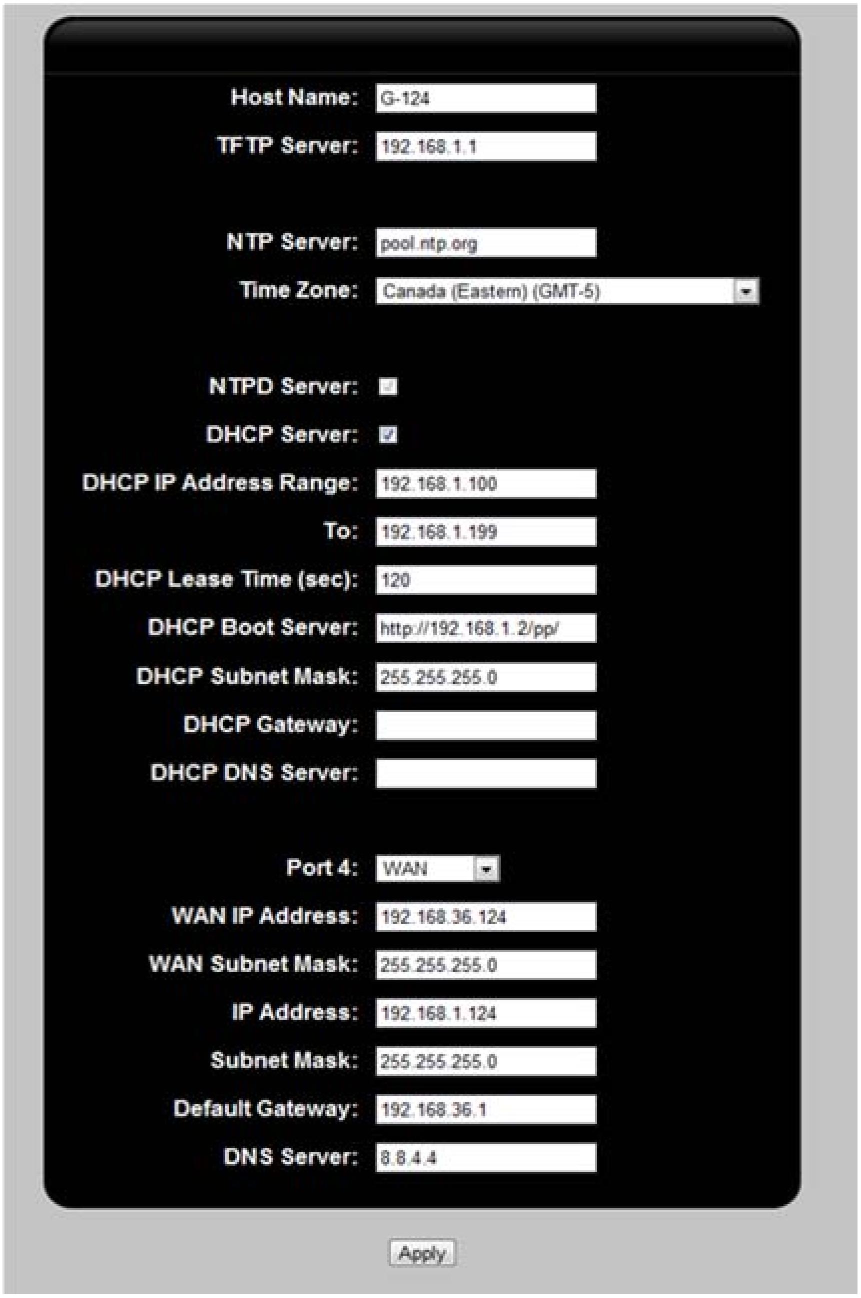
Contains a log of the requested pages and files from the system's Web Server - useful in troubleshooting problems with file serving for Phone Provisioning

- **To disable logging of the Web Server log to either CF or USB**
    - Select RAM from the drop-down menu.

NOTE: Logs stored in RAM will be lost in the event of a system restart.

System -> Network -> Network Tab

Allows for the review and editing of networking parameters.



Host Name:

Sets and displays the name of the host of the system. This name can be anything meaningful to the organization and can be useful in the event that multiple Gateway units have been installed.
Default: G-xxx (where xxx is the model number)

TFTP Server:

Sets and displays the IP address of a TFTP (Trivial File Transfer Protocol) server which could hold firmware upgrades, sound files and system backups.
Default: `192.168.1.1`

NTP Server:

Sets and displays the host name of the time server. The network administrator for the system will determine the appropriate address.
Default: `pool.ntp.org`

Time Zone:

Sets and displays the time-zone to be used by the system from a drop-down menu.
Default: `Canada Eastern (GMT-5)`

NTPD Server:

The ntpd (Network Time Protocol daemon) server is a system daemon that sets and maintains the system time of day in synchronization with time servers.
Default: not enabled

DHCP Server:

Controls whether the unit will act as a DHCP server. The network administrator for the system will generally determine this setting.
Default: not enabled

DHCP IP Address Range (From and To):

If DHCP is enabled, this determines the address range of the devices to be allocated by the Gateway.
Default setting is from 100 to 199.

DHCP Lease Time (sec):

If DHCP is enabled on a dedicated LAN used only for the telephone sets, this determines the time that the IP addresses remain allocated. Renewing the lease is important when wifi phones or laptops with soft-phones connect to the network temporarily throughout the day. In this case the IP address given to a device which quits the network cannot be reused until the lease time expires. In some cases, this would tie up the entire IP address range as defined above, blocking subsequent devices from logging on to the network.
Default: 120 (sec)

DHCP Boot Server:

Provides phone provisioning configuration files.
This field should be set as follows:
http://*G-Series_device_IP_Address*/pp,
Example: `http://192.168.1.2/pp`

NOTE: “http://” MUST be entered

DHCP Subnet Mask:

Default: 255.255.255.0

DHCP Gateway:

Often the PBX network is separate from the organization’s main network. If telephone sets need access to the Internet, this text box should contain the address of the organization’s router. This information is normally supplied by the Network Administrator

DHCP DNS Server:

DNS address of the Internet Service Provider for the organization’s network. Supplied by Network Administrator

Port 4:

Controls the current application of the Gateway’s Ethernet Port 4.
Drop-down menu choices:

- LAN connected to internal network (default)
- WAN phones connected remotely, VoIP carrier
- Monitoring provides connection for packet “sniffing” and traffic monitoring

WAN IP Address:

Static IP Address on the LAN of the Web Interface.

WAN Subnet Mask:

Default: `255.255.255.0`

IP Address:

Sets and displays the IP address of the unit and the Web Interface.
Default: `192.168.1.2`

Subnet Mask:

Sets and displays the subnet mask for the current network. This information will be supplied by the network administrator.
Default: `255.255.255.0`

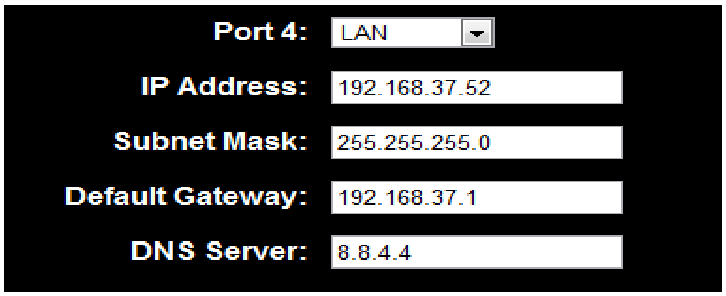
Default Gateway:

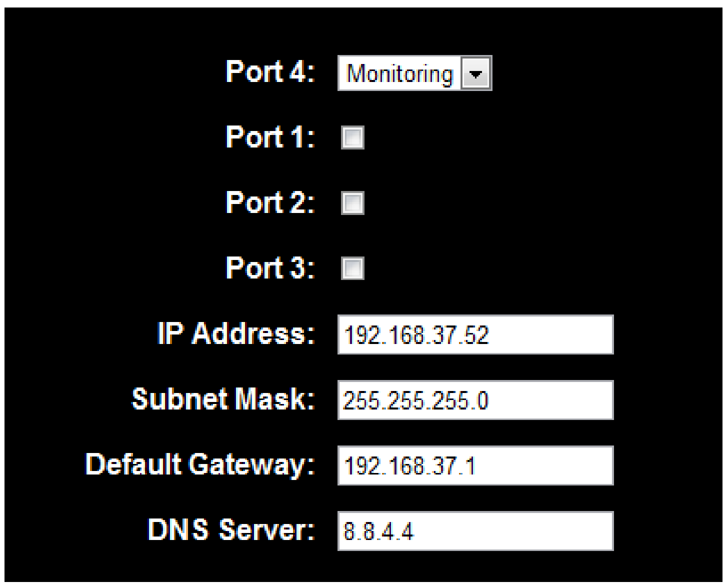
Sets and displays the IP address of the gateway used by the current network. This information will be supplied by the network

administrator.
Default: `192.168.1.2`

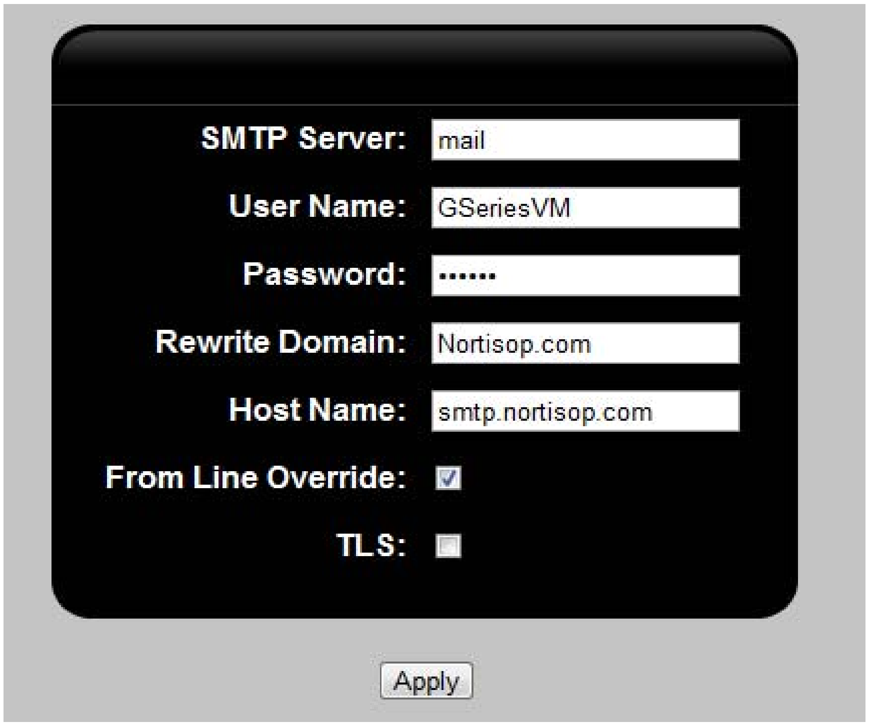
DNS Server:

Sets and displays the IP address of the network's Domain Name System (DNS) server. This information will be supplied by the network administrator.
Default: `192.168.1.2`





System -> Network -> SMTP Tab



SMTP Server:

Sets and displays the address of the organization's mail server. This information is used to allow the sending of voicemail attachments to emails. This address may include a colon followed by a port number.

Username:

Sets and displays the username for the email account used to send voicemail attachments.

Password:

Sets and displays the password for the email account used to send voicemail attachments.

Rewrite Domain:

Allows specification of the domain used in emails containing voicemail as attachments. The domain is the entire part that follows the "@" in an email address.

Host Name:

Specifies the host used for outgoing emails containing voicemail attachments.

From Line Override:

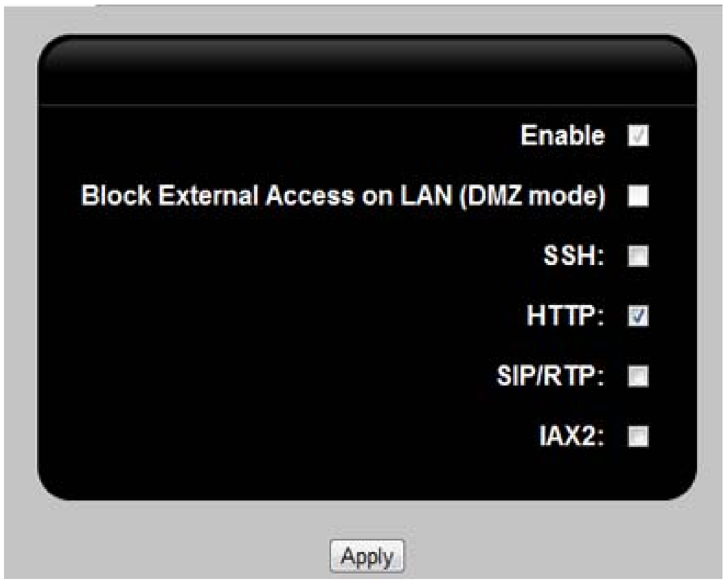
Specifies the information shown the ‘From:’ line in the outgoing emails containing voicemail attachments.

Use TLS (Transport Layer Security):

(Yes/No) – Information for the correct settings can be obtained from the network administrator.

System -> Network -> Firewall Tab

The G-Series Gateway contains an integrated firewall capability. Typically, when installed on a Local Area Network (LAN) the firewall's additional security is not required because security is handled by the organization's firewall. However, when WAN has been selected for Port 4 (see: System -> Network tab), or 3G has been enabled (see: System -> 3G menu), then the Gateway's firewall is automatically enabled to provide additional security.



Enable:

Enables and disables the firewall

Block External Access on LAN (DMZ mode)

When checked allows all traffic from router in DMZ mode to be accepted.

SSH:

When checked allows running secure shell on Gateway with a program such as *putty*

HTTP:

When checked allows external access to the Web Interface.

SIP/RTP:

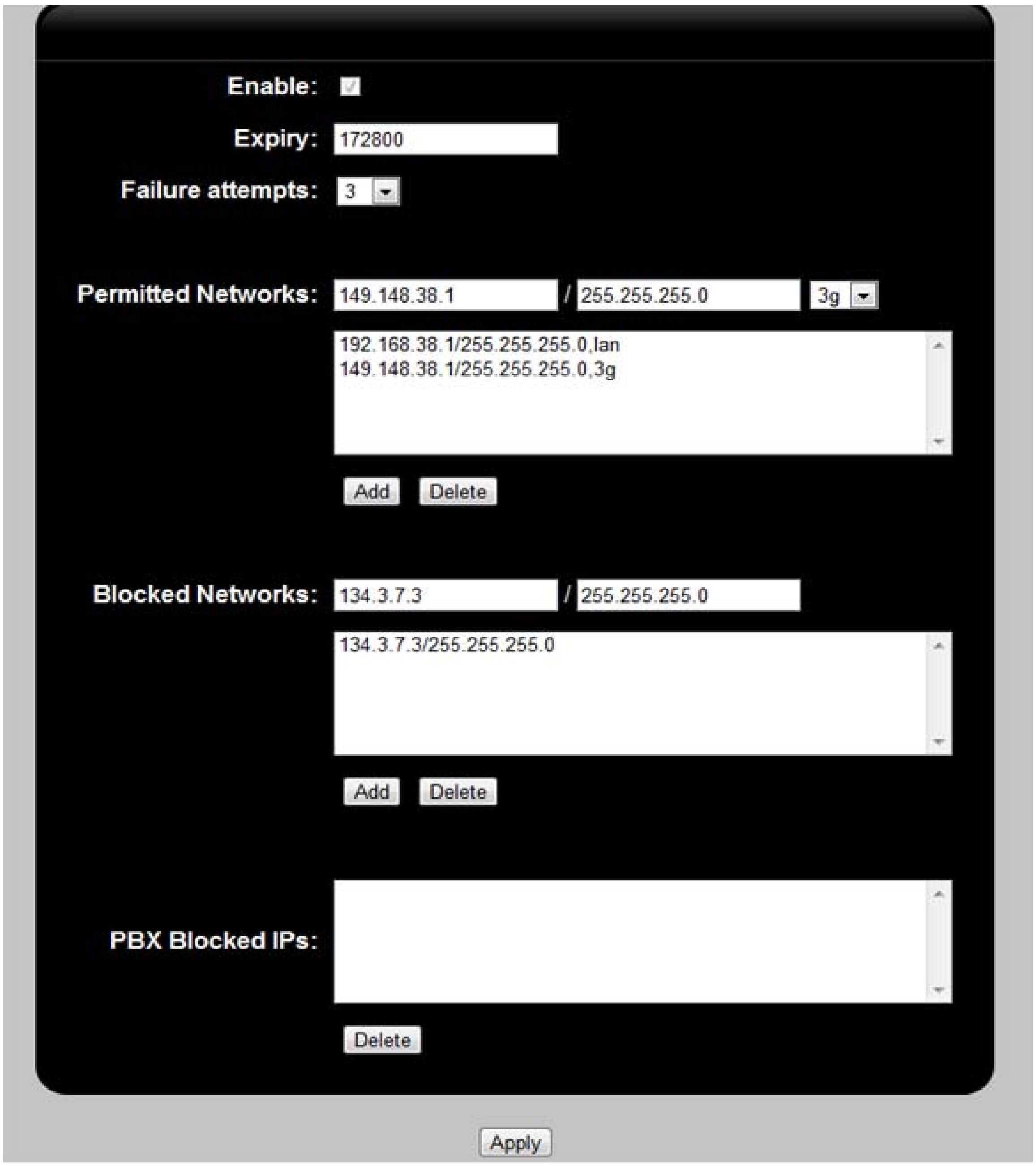
When checked allows SIP/RTP traffic to enter.
Example: When a remote phone is to be registered or when registering to a SIP carrier when Gateway is behind a NAT.

IAX2:

When checked allows external IAX traffic into the Gateway.
Registering to an IAX trunk when Gateway is behind a NAT

System -> Network -> Block IP Tab

As part of the G-Series Gateway`s security features, a **Block IP** screen has been designed to deny access to any IP address that has been identified as a security risk. This tool can be used to block any traffic from coming in to the system if the IP address has performed a specified number of failed attempts at entry.



Enable:

Sets whether IP blocking is enabled.
Default: `enabled`

Expiry:

The blocked IPs are released after the expired time (in seconds) specified in this field has elapsed.

Failure attempts:

Number of attempts before traffic is blocked from that IP address.

Permitted Networks:

Traffic will NOT be blocked from these IP addresses.

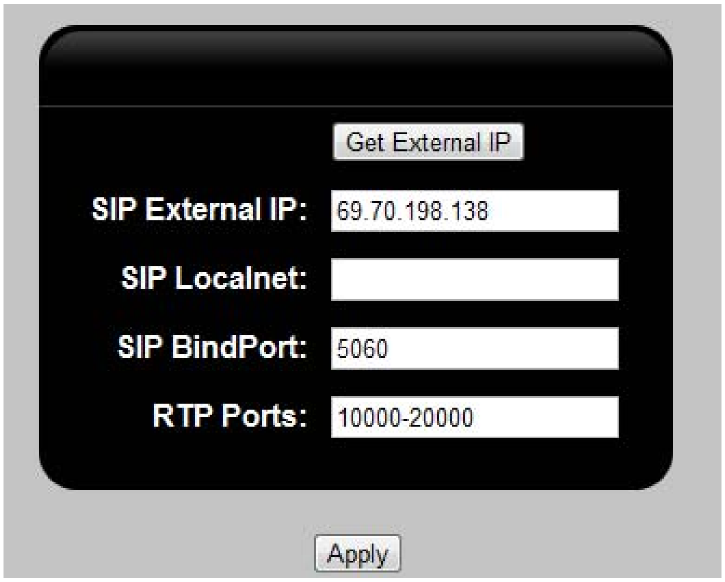
Blocked Networks:

Traffic WILL be blocked from these IP addresses.

PBX Blocked IPs

IP address listed in this table have been blocked due to exceeding the limit on unsuccessful attempts to connect to the device.
If the address is no longer to be blocked, it can be removed from the blocked IPs table.

System -> Network -> SIP / RTP Tab



SIP External IP:

If the Gateway is behind a NAT, and an attempt is made to connect to a carrier, then enable/set **External IP** by clicking the **Get External IP** button.

SIP Localnet:

If another VPN (Virtual Private Network) or LAN (Local Area network), is configured besides the network used with the Gateway, and they are to be connected then enter that network information in the SIP Localnet box.

SIP BindPort:

Allows changing the UDP port for SIP service from 5060 (default) to another port.

RTP Ports:

When the Gateway acts as a SIP server behind the NAT, external clients, or external clients behind a second NAT require port forwarding on the router the Gateway is connected to. The forwarded ports would have to match the port range configured in this field.
Default: **`10000 - 20000`**

System -> Network -> Services Tab

ssh

Secure Shell or SSH is a network protocol that allows data to be exchanged using a secure channel between two networked devices. Secure Shell (ssh) is used to access the command line interface (CLI). The ssh service must be enabled before ssh can be used with the Gateway.
Default: `Running`

tftp

Trivial File Transfer Protocol (tftp) is a file transfer protocol known for its simplicity. When enabled, a G-Series Gateway can act as a tftp server as some models of telephone sets can only be provisioned using tftp.
Default: `Running`

snmpd

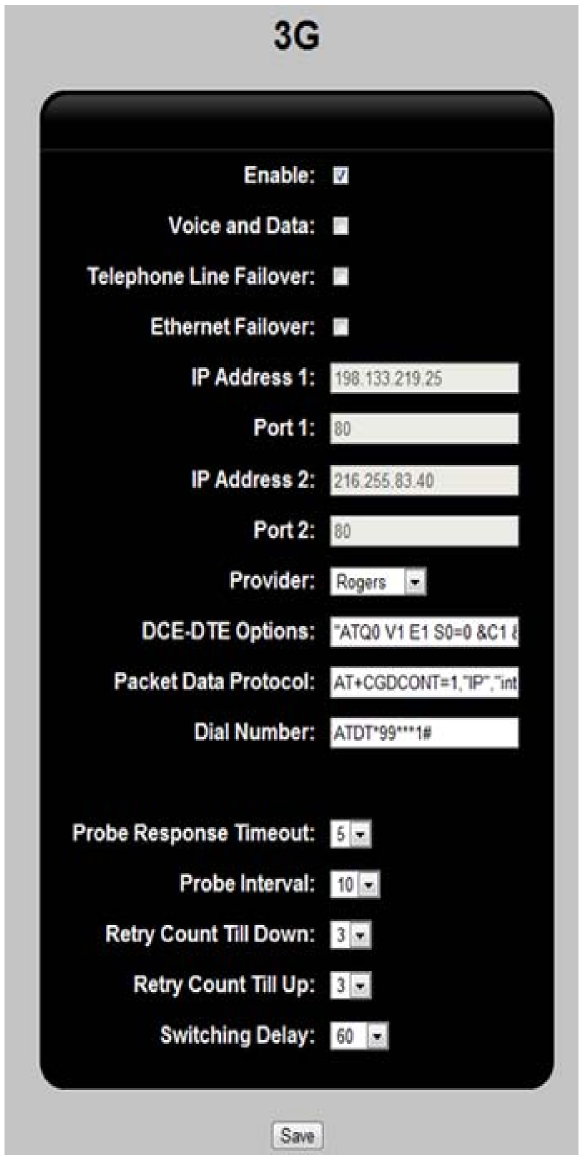
snmpd is an SNMP agent which binds to a port and awaits requests from SNMP management software. It can be used to provide system information to a remote snmp server.
Default: `Stopped`

astmanproxy

astmanproxy is the proxy server for Asterisk's Manager Interface. It allows communication with multiple Asterisk boxes from a single point of contact using a variety of I/O formats. Third party applications that work with Outlook, may use astmanproxy to connect with the G-Series Gateway. Examples of the third party applications would be *OutCALL* and *xtelsio*.
Default: `Stopped`

System -> 3G Menu

A cellular 3G voice and data network can be used to provide a failover connection to the Internet. In the event that the G-Series Gateway detects that the normal wired Internet connection is no longer working, it can use the cellular network for voice and data connection until the wired network connection is restored.



Enable:

Enables the 3G device, which is typically a USB `thumb drive`-type unit plugged into one of the available USB ports on the Gateway.

Default: `not enabled`

Voice and Data:

Enables data to be sent through the G-Series Gateway in addition to voice.
Default: `not enabled`

Telephone Line Failover:

When selected, 3G will be used as a failover in the event of failure of the telephone line.
Default: `not enabled`

Ethernet Failover:

When selected, 3G will be used as a failover for the Ethernet link.
Default: `not enabled`

IP Address 1, Port 1, IP Address 2, Port 2:

These are the IP addresses of popular Internet services and Gateway device ports used to verify the 3G connection. If the Gateway does not get a response for packets sent through the specified address and ports, it will be considered a fail and the system will be unable to connect/reconnect.
Defaults:

IP Address 1:	`198.133.219.25`
Port 1:	`80`
IP Address 2:	`216.255.83.40`
Port 2:	`80`

Provider:

Drop-down box listing preconfigured 3G providers. Selecting the appropriate provider name from the list will automatically configure the next three fields. In the event that the 3G provider is not in the list, the settings can be tried as a starting point.
Default: `standard`

DCE-DTE Options, Packet Data Protocol, Dial Number:

If the specific model cannot be found in the **Provider** drop down box above, then the information for these three fields should be requested from the 3G provider. However, the default values in the fields may prove to be correct.
Defaults: `values have been set`.

Probe Response Timeout:

Time in seconds. If there is no response within the time specified from a packet sent as a probe, then a failure is declared.
Default: `5`

Probe Interval:

Number of seconds between probes from the Gateway system.
Default: `10`

Retry Count Till Down:

How many times the Gateway will try to connect/reconnect its Ethernet connection until the connection is considered down.
Default: `3`

Retry Count Till Up:

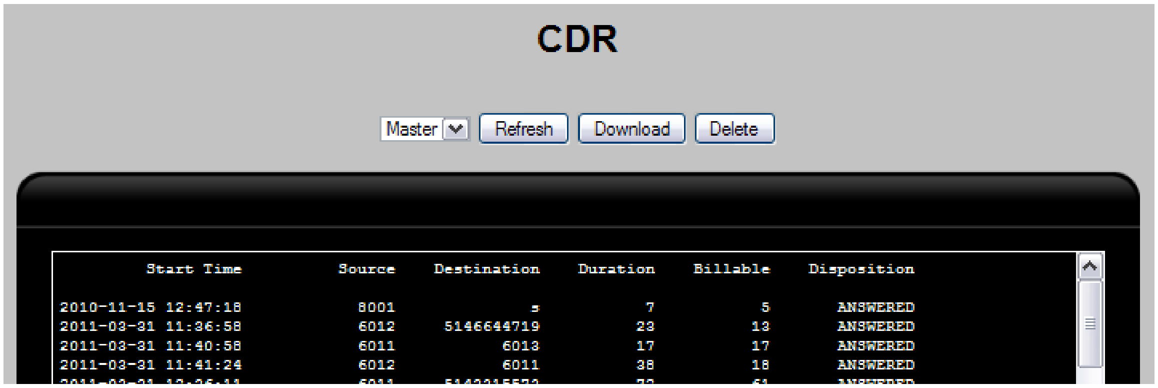
How many times the Gateway will try to reconnect its Ethernet connection until the connection is considered up.
Default: `3`

Switching Delay:

This is the amount of time in seconds, for the system to wait before switching between Ethernet, or telephone lines to 3G and vice versa.
Default: `60`

System -> CDR Menu

Displays information from the Call Detail Record (CDR) system. Buttons allow refreshing the display and Downloading the CDR data as a *tar.gz* file.



Start Time
Start time of the call

Source
Source telephone or extension number

Destination
Number of the called party

Duration
Total duration of the call including ring time

Billable
Duration of the call excluding ring time

Disposition
Call completion information.

Example of a CDR record:

Start Time	Source	Destination	Duration	Billable	Disposition
2010-11-15 12:02:38	6003	6002	9	5	ANSWERED

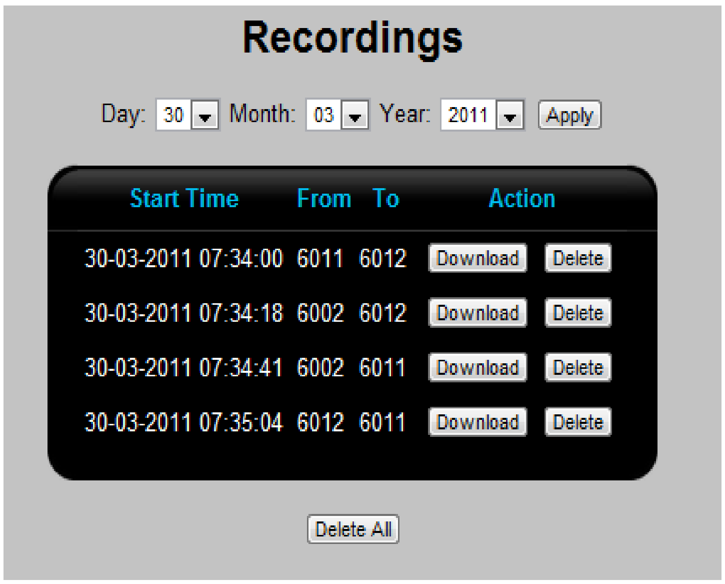
NOTE: The Duration time is the amount of time (in seconds) the phone call lasts, including ring time.

NOTE: The Billable time is the amount of time the call lasts from answer to hang-up.

System -> Recordings Menu

The **Recordings** page allows selecting recordings by date for downloading and deleting. It displays call time and the parties involved.

See: PBX -> PBX Settings ->Recordings tab for control over whether recordings are permitted by the system.



- **To Download a recording**
  - Select the date of the recording to be downloaded using the text fields and drop-down buttons
  - Click **Apply**
  - Click **Download** beside the recording to be downloaded
  - Locate the destination for the recording on the notebook's local file system and save it.

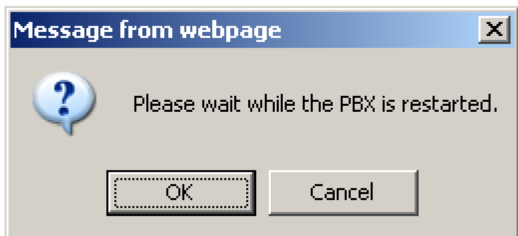
System -> Restart Link

Restarts the system. Existing configuration and backup files will remain intact, however any files stored in RAM will be deleted.

	When a restart is performed, any telephone calls in progress will be dropped.

➢**To perform a system restart:**

- Click **OK** to begin restart, or **Cancel** to return without restarting.



- A countdown timer will be displayed after which the Web Interface should be available again. The unit's red LED above the circle icon will glow steadily when reset is complete.

- The system's **Login** screen will be presented, the IP address and Password to be used will be the ones stored in the current system configuration.

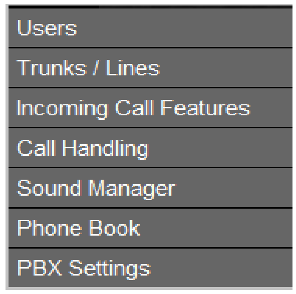
NOTE: Resetting the system will clear any logs stored in on-board random-access memory (RAM).

➢ **To configure log storage see:**

- System -> Maintenance -> Logs tab

NOTE: Resetting the system should not affect Gateway systems with battery backup clocks. In the unlikely event of a problem with the time setting, the system time can be set using the: System -> Maintenance -> Date and Time tab

PBX Menu



Users

Offers a series of menus to configure user extensions. Displays a list of extensions and the names of the users associated with them, allows for the editing (configuration) and deletion of VoIP and analog extensions and real and virtual voicemail boxes.

Allows the establishment of user templates containing common information to be automatically applied to new extensions.

Provides for the individual provisioning of telephone sets allowing the provisioning of specific features and configuration.

Allows the management of ring groups, i.e., of extensions belonging to a common group of users which can be rung simultaneously or in sequence.

Trunks/Lines

Provides control over the configuration of Analog and VoIP trunks.

Provider templates can be created containing VoIP provider specified information, in the event that several trunks from a specific provider is required.

Allows management of outgoing line groups.

Incoming Call Features

Controls features such as IVR, time frames which allow or prohibit services based on time or date, conference bridges, call parking, paging and DISA.

Call Handling

Rules for incoming and outgoing calls as well as call accounting.

Sound Manager

Management of sound files used for music-on-hold, IVR and language selection.

Phone Book

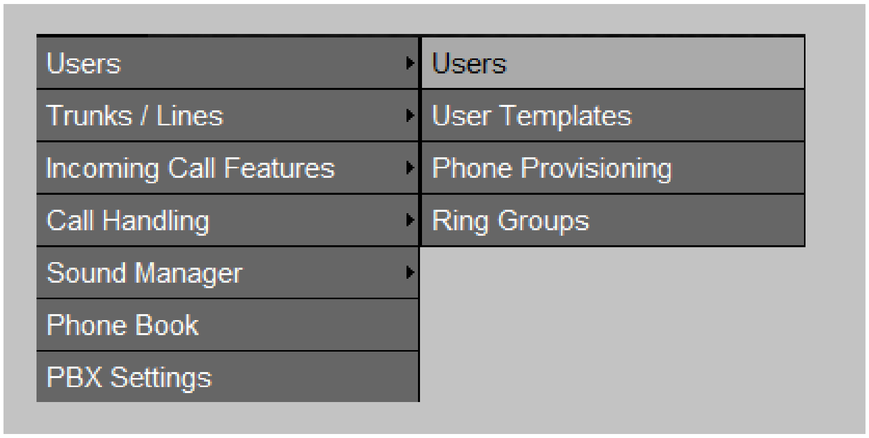
Establishment and control of common phone book, containing a list of external telephone numbers which can be called as though they were extensions.

NOTE: If callers are allowed to dial extensions through the IVR system, this may pose a privacy risk.

PBX Settings

Overall configuration of PBX settings and services shared among all users.

PBX -> Users Menu



The Users menu permits access to the:

Users menu

Allows the creation and initial configuration of all extensions, including hardware and software telephone sets and virtual extensions for voicemail boxes.

User Templates

Allows the creation and management of templates used to configure common fields for similar users.

Phone Provisioning

Provides a quick and simply way of registering a phone set, while also having the ability to manage its features and services.

Currently two types of phone provisioning are supported – auto and file-based phone provisioning.

Auto Phone Provisioning

Allows the complete configuration of the phone set to be done through the Web Interface, after which, once the phone set has been rebooted, all changes saved in the Web Interface will be uploaded to the phone via the DHCP Boot Server.

- **To set the DHCP Boot Server**
  - See: System -> Network -> Network Tab

File Based Phone Provisioning

Telephone set files can be configured and uploaded to the G-Series device if the phones do not have Auto Phone Provisioning support.

NOTE: If the phone is using a static IP address, then within the phone set's user interface, the server address can be manually set: http://*G-Series_device_IP_Address*/pp

Ring Groups configuration

Allows the creation and management of ring groups.

PBX -> Users -> Users Menu

Displays a list of extensions and the names of the users associated with them, allows for the editing (configuration) and deletion of users and voicemail boxes.

The **Add** button allows for the creation of new users. By selecting the appropriate user extension type from the **Type**: drop-down menu relevant parameter information is to be entered.

NOTE: After adding users, be sure to click **Apply Configuration** to make the new user active.

NOTE: Some changes to user information may require the associated telephone set to be re-initialized using the telephone's menu or interrupting/restoring telephone power.

Extensions can be created for:

- Analog phone sets
- SIP telephone sets
- SIP software phones
- Virtual Extensions (voicemail boxes)

Bulk Import:

Allows user information to be imported from a Comma Separated Values (CSV) -formatted spreadsheet when setting up the system.

NOTE: It is only to be used in the initial setup. Once user information has been entered into the system, this procedure WILL NOT update current information. Changes to users thereafter will have to be entered through the Web Interface.

➢ **To create an import spreadsheet**

- Click **Template**
- The system will download a file called `template_users.csv` to the local file system.
- Follow any instructions in the top rows of the spreadsheet.
- Note that row 5 of the spreadsheet contains a sample record which will be disregarded by the system.

➢ **To import user information:**

- Click **Choose File**
- Locate the file on the notebook.
- Click **Open**
- The file will appear next to the **Import** button.

NOTE: Do not use row 5 for actual data, the system will only interpret records from row 6 of the spreadsheet onwards.

- Fill in and save the spreadsheet in .CSV format with a meaningful name that does not contain spaces, on the local file system.

NOTE: Any data that has been imported can later be changed using the Web Interface, however the spreadsheet cannot be re-imported to replace existing data already on the system.

Extension:

A unique extension number, typically four digits, with a minimum of two digits. Extensions can be of mixed length – for example, three and four digits. A timer on the system can differentiate between a user dialing extension 364 and 3647.

An error message will result if the entered extension number is already in use.

Type:

A drop-down menu which provides the following selection of extension types: **SIP, Analog** and **Virtual**.

One of these three must be specified. This field cannot be left blank.

SIP:

Selected when using a SIP telephone set or SIP software phone.

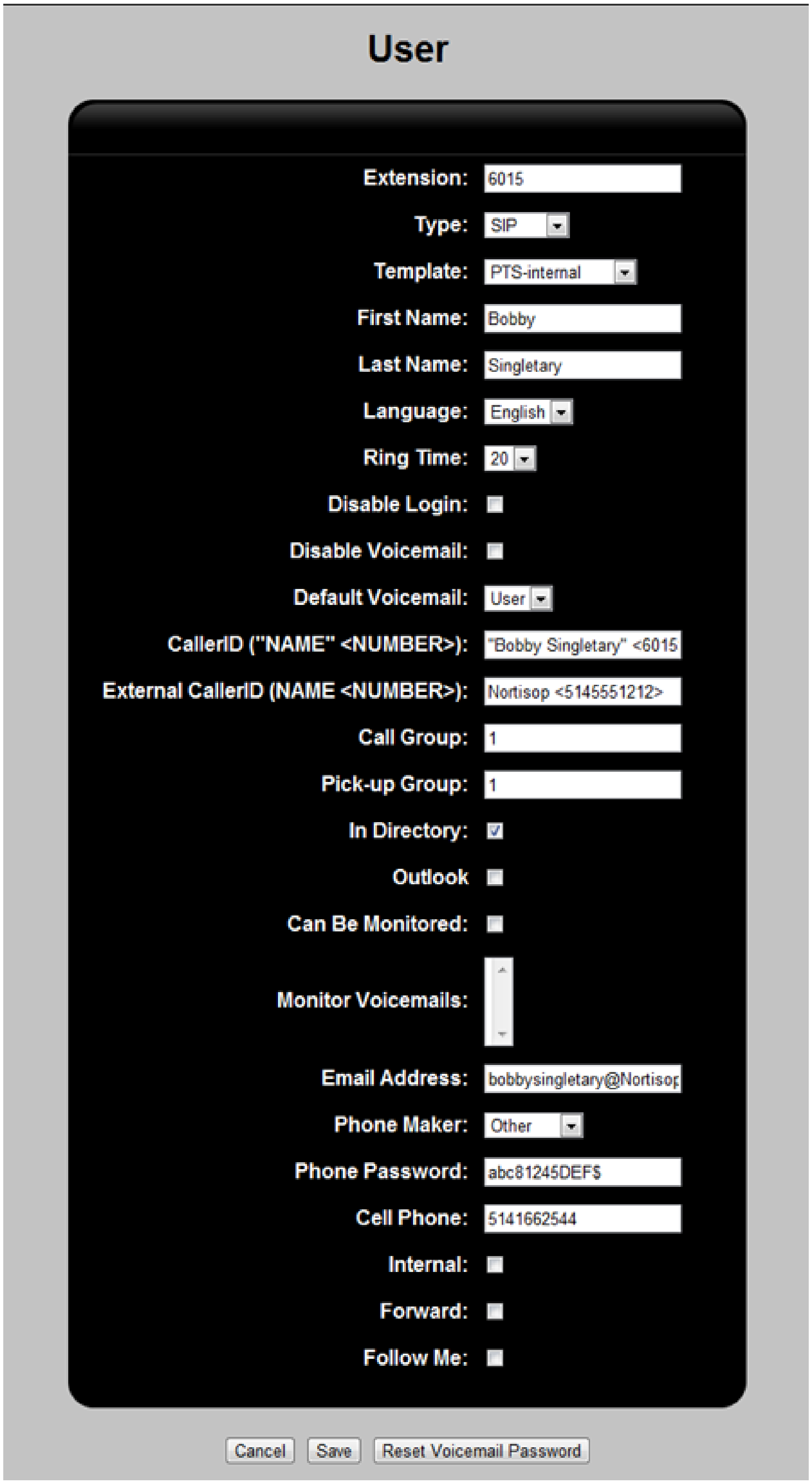
Analog

Selected when using an analog phone set or analog fax machine.

Virtual

A virtual extension is used for shared or departmental Voicemail boxes.

Establishes a virtual voicemail box which is not associated with a specific user.
Example: The sales department can use a virtual voicemail box to store customer requests, and a number of sales staff can access this box in order to process the messages. This kind of mailbox can also serve to catch messages left by callers to a ring group which was not answered.


User
Extension:
6015
Type:
SIP
Template:
PTS-internal
First Name:
Bobby
Last Name:
Singletary
Language:
English
Ring Time:
20
Disable Login:
Disable Voicemail:
Default Voicemail:
User
CallerID ("NAME" <NUMBER>):
"Bobby Singletary" <6015
External CallerID (NAME <NUMBER>):
Nortisop <5145551212>
Call Group:
1
Pick-up Group:
1
In Directory:
Outlook
Can Be Monitored:
Monitor Voicemails:
Email Address:
bobbysingletary@Nortisop
Phone Maker:
Other
Phone Password:
abc81245DEF$
Cell Phone:
5141662544
Internal:
Forward:
Follow Me:
Cancel
Save
Reset Voicemail Password

Template:

Drop-down list offers the selection of created user templates to use for configuration of the user.
For user template creation and configuration, see
PBX -> Users -> User Templates menu.

G124, G122, G1212 Only:

FXS Port:

Choice of analog Gateway device telephone (FXS) port 1 or 2

Hotline:

Used for emergencies, elevator or building entrance telephones etc. Picking up handset will automatically dial a stored telephone number specified below.

Before selecting Hotline, configure Outgoing Call Rules selection below. Selection will be locked once Hotline is selected.

Outgoing Line Groups: (Hotline checkbox selected only)

Drop-down list offers line groups available for the hotline:

- To configure Outgoing Line Groups see:
  PBX -> Trunks / Lines -> Outgoing Line Groups menu

Phone Number: (Hotline checkbox selected only)

Phone number to be dialed by hotline extension.

Ring Time: (Hotline checkbox selected only)

Drop-down menu offers time in seconds for hotline call response before it hangs up.

Outgoing Call Rules:

Drop-down list of available outgoing call rules.
To configure Outgoing Line Groups see:
PBX -> Trunks / Lines -> Outgoing Line Groups menu

Music-on-hold:

Drop-down list of available music-on-hold options. These are presented to external callers to this user.

- DefaultMoH – Default Music-on-hold file
- DefaultAudioIn – Mono mini-jack for audio input in selected Gateway units

G122, G1000 Only:
The G-122 and G-1000 do not support Audio-In.

First Name, Last Name:

The user's name information. This is used in the automated directory which will permit callers to spell the first three digits of either first or last name.

NOTE: Choice of searching on the first or last name for directory is done in the **GoToDirectory** action in the main IVR menu. See: PBX -> Incoming Call Features -> IVR Menus

Language:

Language used in providing user instructions for the voicemail feature.

RingTime:

Time in seconds for call response before it hangs up, or before voicemail picks up.

Disable Login:

Disables access to the **User Web Interface**.

Disable Voicemail:

Disables the user's voicemail box.

Default Voicemail:

A drop-down menu offers a choice between User and Virtual voicemail boxes.

CallerID ("NAME" <NUMBER>):

Caller ID as displayed on outgoing internal calls.
Enter the CallerID as numeric or alphanumeric text as applicable.

External CallerID (NAME <NUMBER>):

Caller ID as displayed on outgoing external calls.
Certain carriers will permit the setting of the callerID displayed to the called phone.

Call Group:

Defines the user's call group.

Pickup Group:

Defines the user's pickup group.
Example: If a nearby extension is ringing and is a member of the same pickup group, it can be answered by dialing #7, the default pick-up setting.
The default pickup setting is defined in
PBX -> PBX Settings -> Call Handling tab.

Note: A user can belong to multiple pickup groups e.g.: pickup group: 1,2,3. This user would then be able to answer calls going to pickup groups 1, 2 or 3 by dialing #7.

In Directory:
Determines whether the name information entered above will be placed in the extension directory system for use by incoming callers.

Outlook:
Determines whether the user has access to the function of making and receiving calls through Microsoft Outlook.

Send Messages by Email Only:
Voicemails will not be stored, only attached to emails.

Can be Monitored:
Whether this extension can be monitored.

Once the monitoring session begins, it remains open for all subsequent calls until the extension doing the monitoring hangs up.

Monitor Voicemails:
Whether voicemail box can be monitored.

Email Address:
The full email address of the user to which voicemail attachments will be sent.

Phone Maker:
Drop-down list of available phone makers. Provides configurations for certain IP telephone set manufacturers.

Phone Model:
List of phone models is presented depending on **Phone Maker** selection. This automatically provides device parameters to the system.

Phone Serial Number:
The MAC address of the telephone set.

NOTE: Any letters in the MAC address must be entered in lower-case – not capitals.

Phone Password:
Password for the phone to be used on phone registration

Cell Phone:
If the user can be called on their cell phone, then the number of their cell phone can be entered here.

Internal:

When enabled, this extension can be accessed from within, but not from the outside. This can be useful for virtual voicemail extensions.

Forward:

Designates a call forwarding number.
If selected, offers the following parameters:

- External number
- Internal users
- Ring groups
- Virtual voicemails

Follow Me:

Allows unanswered calls to be forwarded to other extensions, useful when the user frequently uses another office. When the checkbox is clicked, the menu will be extended, and additional drop-down menus and input boxes will be displayed:

Announce Callers:

The caller will be asked to record their name. This recording will be announced to the called party.

Ring Time 1:

After the specified number of seconds, the call will be forwarded to the number listed in Phone Number 1.

Ring Time 2:

After the specified number of seconds, the call will be forwarded to the number listed in Phone Number 2.

If not Answered:

This drop-down box offers the following choices:

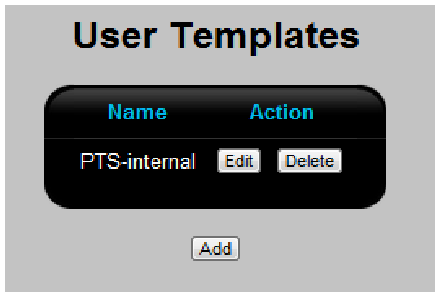
- **Goto Voicemail Box**
  Causes a further drop-down menu to be displayed showing available Voicemail boxes.
- **Goto IVR**
  Displays available IVR systems
- **Hangup**
  Hangs up the call if it has not been answered in the specified times by the specified **Follow Me** numbers.

Reset Voicemail Password

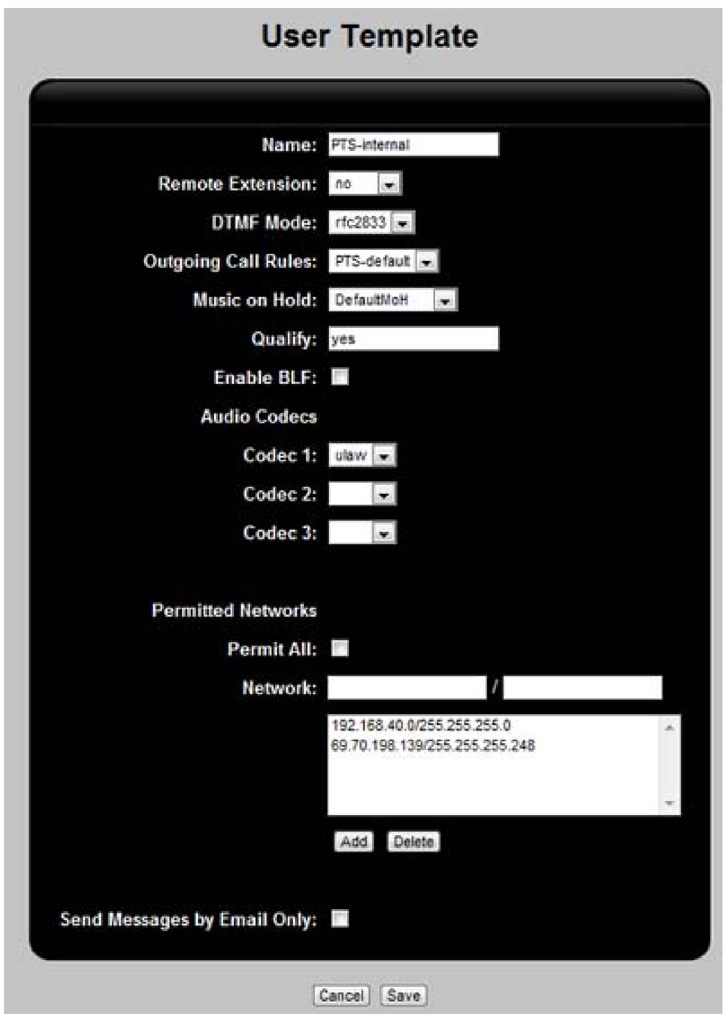
Allows for the resetting of the voicemail password to the default of 1234.

PBX -> Users -> User Templates Menu

A user template is used to allow for quicker configuration of a user extension. In many cases, there will be only one template for the organization. However, in larger organizations, it may be efficient to develop specific templates for departments, for example.



Main User Templates Menu



Name:

Name of user template

Remote Extension:

Is the extension to be a remote extension? A drop-down menu presents the following choices:

- No
- Never
- Yes
- Route

DTMF Mode:

Drop-down menu which allows for selection among different DTMF signaling protocols – typically country-dependant.

- Auto
- Rfc2833
- Inband
- Info

Outgoing Call Rules:

A drop-down menu presents choices among the available rules. For example, these two rules are typically pre-configured with Gateway devices:

- FXS lines
- PTS-default (PTS = Positron Telecom System)

Music-on-hold:

Lists the sound files which can be played for music-on-hold. Also allows the selection of the audio-in connection on selected G-Series Gateway units as a source of audio for music-on-hold. A drop-down menu presents the following:

- DefaultMOH
- DefaultAudioIn

Qualify:

If "qualify=yes", Asterisk will send an option packet every 60 seconds to see if the destination is still online. If the destination does not respond within two seconds for 7 tries in a row, it will be marked as unreachable.

If qualify=xxx (xxx being a value in milliseconds), Asterisk will send an option packet every xxx milliseconds, to see if the destination is still online. If the destination does not respond within two seconds for 7 tries in a row, it will be marked as unreachable

Enable BLF:

Whether the Busy Lamp Field (BLF) can be configured

Audio Codecs

Codec 1:
Codec 2:
Codec 3:

A drop-down menu presents choices similar to the following:

- uLaw
- aLaw
- g729

Permitted Networks

Permit All:

Determines whether any network can be used.

Network:

Lists permitted networks and sub-net masks which can be used.

Add button

Allows the entry of new network addresses

Delete button

Allows deletion of the selected network

Send Messages by Email Only:

Determines whether voicemails should be sent by email instead of being stored in voicemail boxes.

Default User Template - PTS-Internal:

Name:	`PTS-Internal`
Remote Extension:	`no`
DTMF Mode:	`rfc2833`
Outgoing Call Rules:	`PTS-default`
Music on Hold:	`DefaultMoH`
Qualify:	`yes`
Enable BLF:	`not enabled`
Audio Codecs	
Codec 1:	`uLaw`
Codec 2:	
Codec 3:	
Permitted Networks	`none`
Permit All:	`not enabled`
Network:	`/`
Send Messages by Email Only:	`Not enabled`

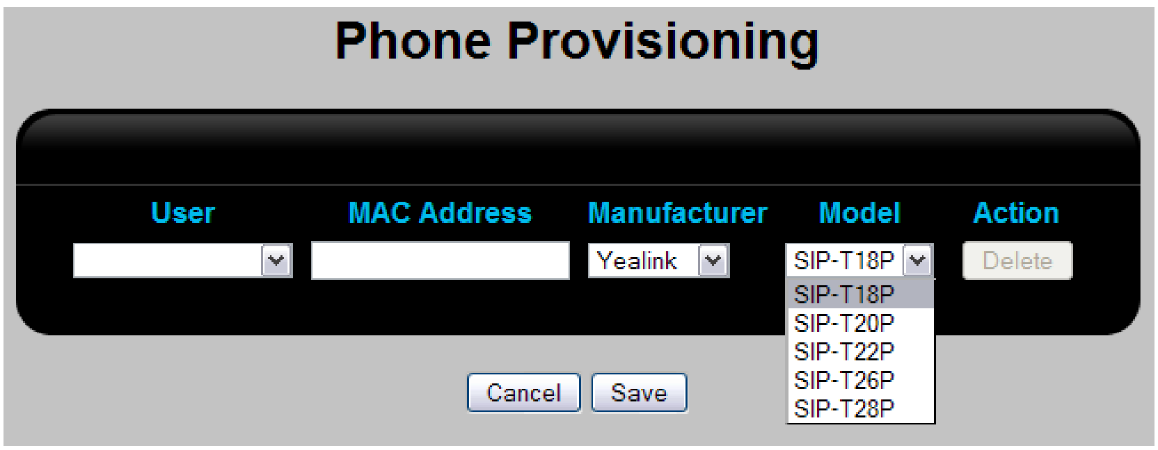
PBX -> Users -> Phone Provisioning Menu

The phone provisioning menu allows the entry of specific phone set (hardware) information, the management of features and services and the ability to assign those phone sets to specific Users.

Add button

Allows the addition of a new telephone set to be used with provisioning, and for that telephone set to be assigned to a user.

Requires the User extension number and name of the individual, the MAC address of the specific set, Manufacturer and Model number.



User:

Provides a drop-down list of registered extensions. Contact Positron if manufacturer is not listed.

MAC Address:

MAC Address of specific telephone set

NOTE: The MAC address is typically printed on a sticker on the underside of the telephone set. By entering the MAC address, the system can keep track of the set even if it is relocated in the office.

NOTE: Any letters in the MAC address on a Yealink phone set must be entered in lower-case – not capitals.

Manufacturer:

Drop down list of available telephone set manufacturers.

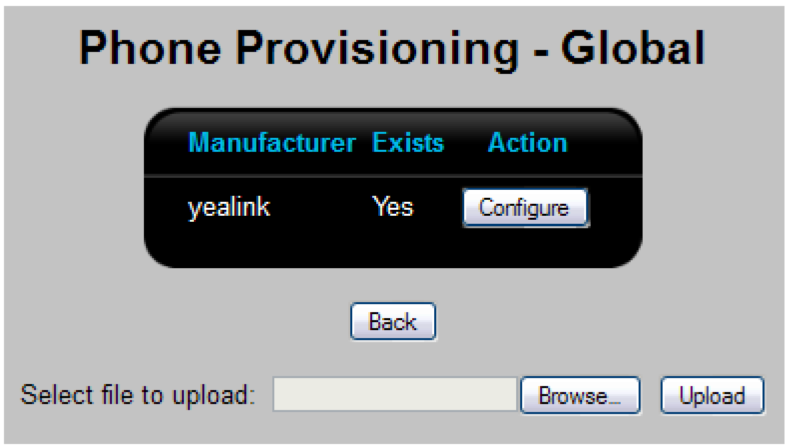
Model

Drop down list of available preconfigured telephone set models from the selected manufacturer.

Global button

Allows the ability to set the User and Administrative Passwords, at a global level (for all phone sets), i.e., the phone set's user interface passwords are changed across the board. Configuration files for unsupported phones sets can be uploaded as well.

NOTE: Only Yealink phone sets are currently supported.



Reboot All button

This button will reboot ALL phone sets which are registered to the system.

⚠	When a reboot is performed, any telephone calls in progress will NOT be dropped, but once the call has ended, the phone set will immediately be rebooted.

Edit button

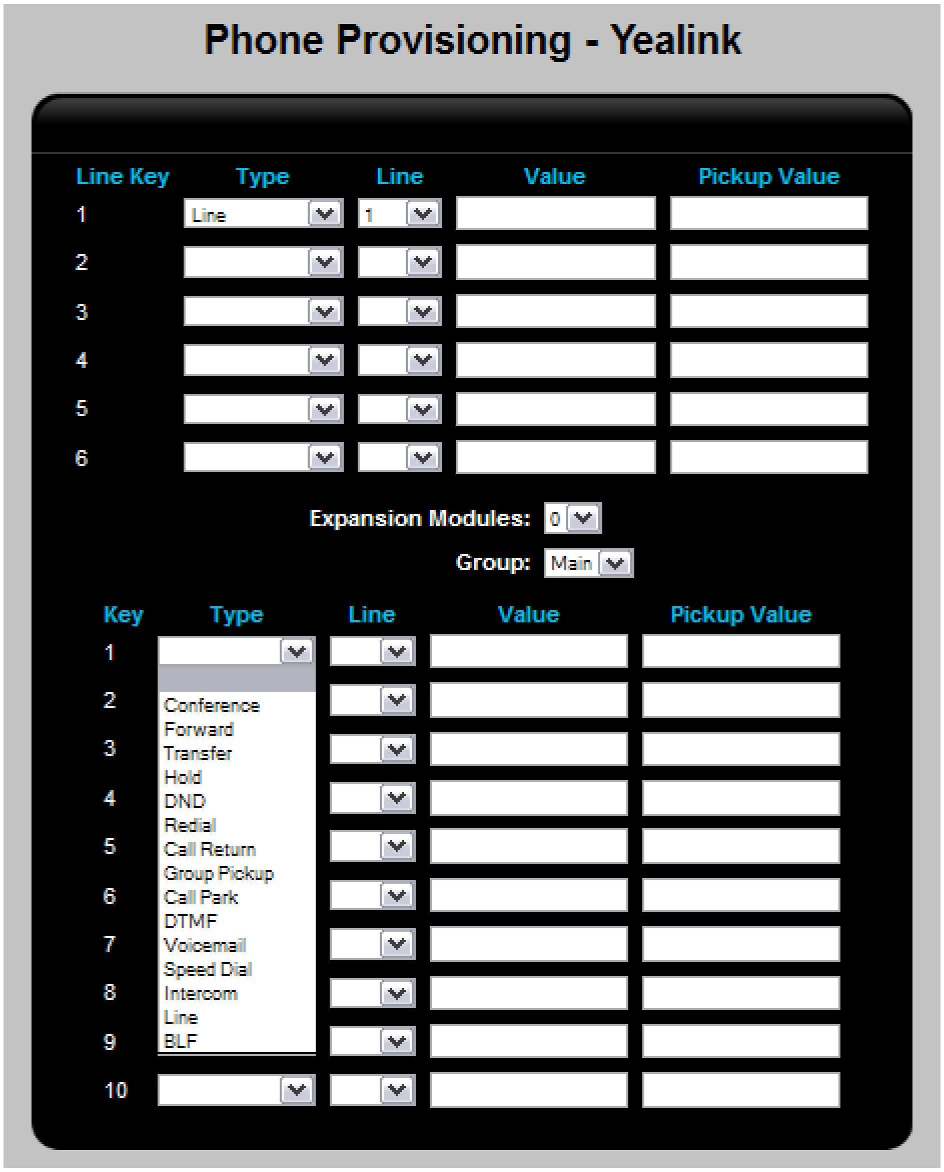
Allows editing of specific telephone set hardware, along with the IP address it's registering to and the user information for the extension.

NOTE: Some phone sets have multiple line keys and therefore can support multiple users.

Configure button

Allows the selection of specific phone set features. In the case where Auto Phone Provisioning is unsupported, phone set configuration files can be uploaded as well.

NOTE: Only Yealink phone sets are currently supported with Auto Phone Provisioning.



Phone Provisioning Commands

Conference

Allows configuring the DSS key to be used as a conference key while remaining in the current call. This key allows a user on a call to conference another party while remaining in the conference.

Forward

If the key is configured as Forward key, pressing it will display the telephone's forward page. Setting the Forward To: number, will cause the phone to forward the call to that number automatically.

Transfer

The key can be configured as a transfer key to perform the Blind/Attended Transfer.

Hold

The key can be configured as a hold key. Calls can be put on hold and retrieved during the conversation.

DND

If the key is configured as DND (Do Not Disturb) key, pressing it will activate the DND function. Pressed a second time and DND will be deactivated.

Redial

If the key is configured as Redial key, pressing it will enter the Dialed Calls interface, and a call can be made by pressing the line keys.

Call Return

When the key is configured as Call Return key it will dial the number of the last call received.

Group Pickup

When configured as a Group Pick Up key with an extension group, any call received by any member of the monitored group can be picked up. If the group receives multiple calls simultaneously, the server will assign the call to be picked up.

Call Park

Call Park is a feature that allows a call to be put on hold at one telephone set and be retrieved at another telephone set. Pressing the "call park" button will place the call on hold, and provide the number of the "park room" or extension where the call can be retrieved.

DTMF

Allows sending of a pre-programmed set of DTMF tones during a conversation.

Voicemail

When the key is configured as Voicemail key, pressing it accesses voicemail directly.

Speed Dial

This key allows accessing the most frequently dialed numbers with one or two key presses.

Intercom

When configured for Intercom mode allows direct connection to a specified extension.

Line

Line keys can activate up to the six user accounts.

BLF

Busy Lamp Field (BLF) allows monitoring the status (idle, ringing, or busy) of other SIP extensions. The user can also dial out on a BLF configured key.

Delete button

Deletes configuration of a specific telephone set.

Reboot button

This button will reboot the specific phone set registered to the system.

When an entry is deleted, any telephone calls in progress will NOT be dropped, but once the call has ended, the phone set will immediately be rebooted

Bulk Import

By downloading and populating the provided template file, Bulk Import is an easy way of creating all of the Phone Provisioning entries required.

Choose File button

Allows selection of import file.

Import button

Begins import of selected file.

Template button

Downloads a template file which can be populated with information about specific telephone set characteristics listed by individual users.

Ring Groups Menu

Incoming calls can be directed to an IVR, specific extensions, or to a ring group – a set of extensions any one of which can respond to the call.

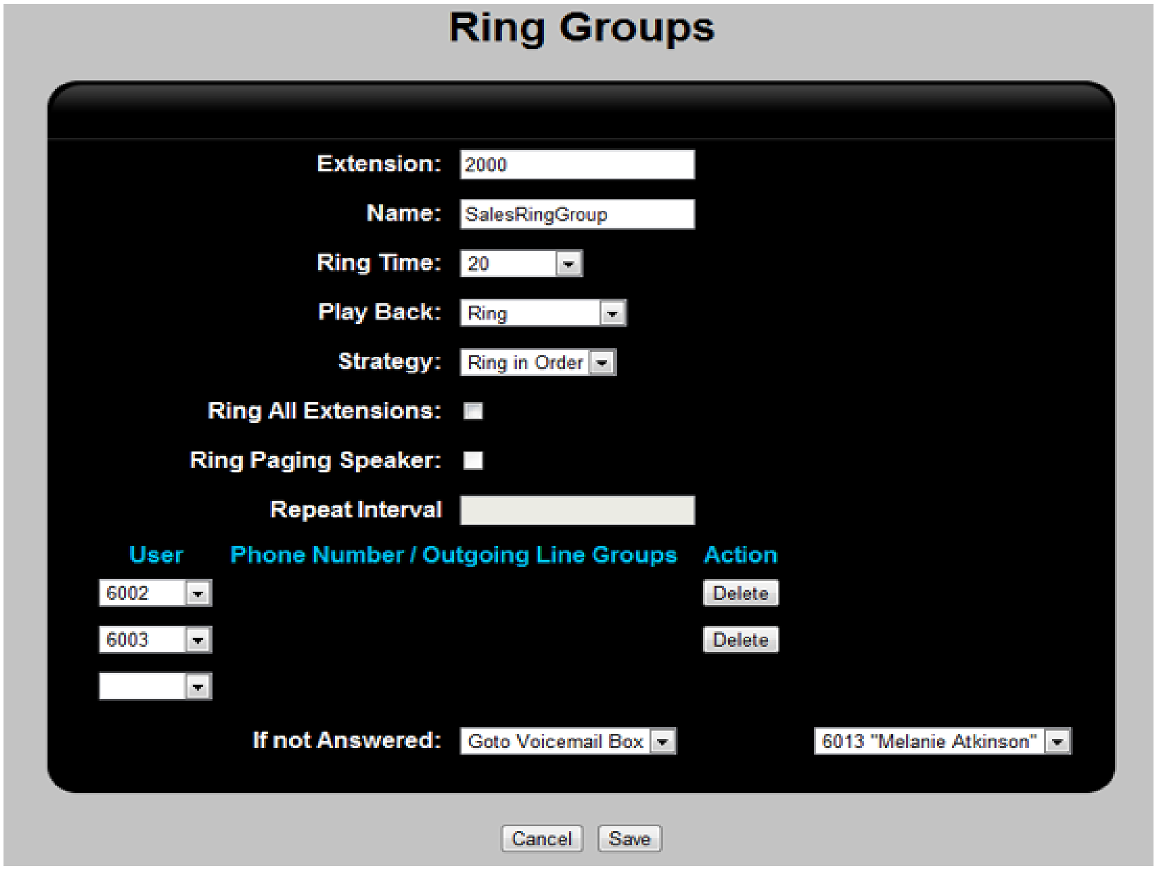
Add button

Creates a new ring group

Edit button

Allows the making of changes to the name and extension number an existing ring group

NOTE: After making changes to ring groups, click **Apply Configuration** to make the changes active.



- **To Create a ring group**
  - Click **Add**
  - Enter the following information:

Extension:

Extension number assigned to the ring group.

NOTE: The extension number cannot be the extension of an individual user.

NOTE: A ring group extension can begin with a letter if it is to be called only from within an IVR. In this way, the ring group cannot be called from a telephone keypad.

Name:

The name of the ring group which will be shown in listings.

Ring Time:

The time measured in seconds during which ringing is allowed before control moves to the next stage in the strategy.

Play Back:

The audio heard by the calling party during the Ring Time until the call is picked up. Choices are:

- Ringing (default)
- The default music-on-hold
- The Default Audio In source.

Strategy:

Choices are to ring all extensions, or ring group members in order or ring all (simultaneously)

- Ring All Extensions
- Ring Paging Speaker: Yes/No
- Selecting the **Ring All Extensions** check box will ring ALL user extensions.
- The **Ring All Strategy** will ring all user extensions selected in the ring group (only).
- The **Ring in Order Strategy** will ring the extensions in the order selected with the drop down boxes.

Repeat Interval

Interval in seconds between notifications on the paging speaker.

If not Answered:

Determines the routing of the call if the call is not answered. Choices are:

- Hang-up
- Go to selected Voicemail box
- Go to selected IVR Menu

PBX -> Trunks / Lines -> Trunks / Lines Menu

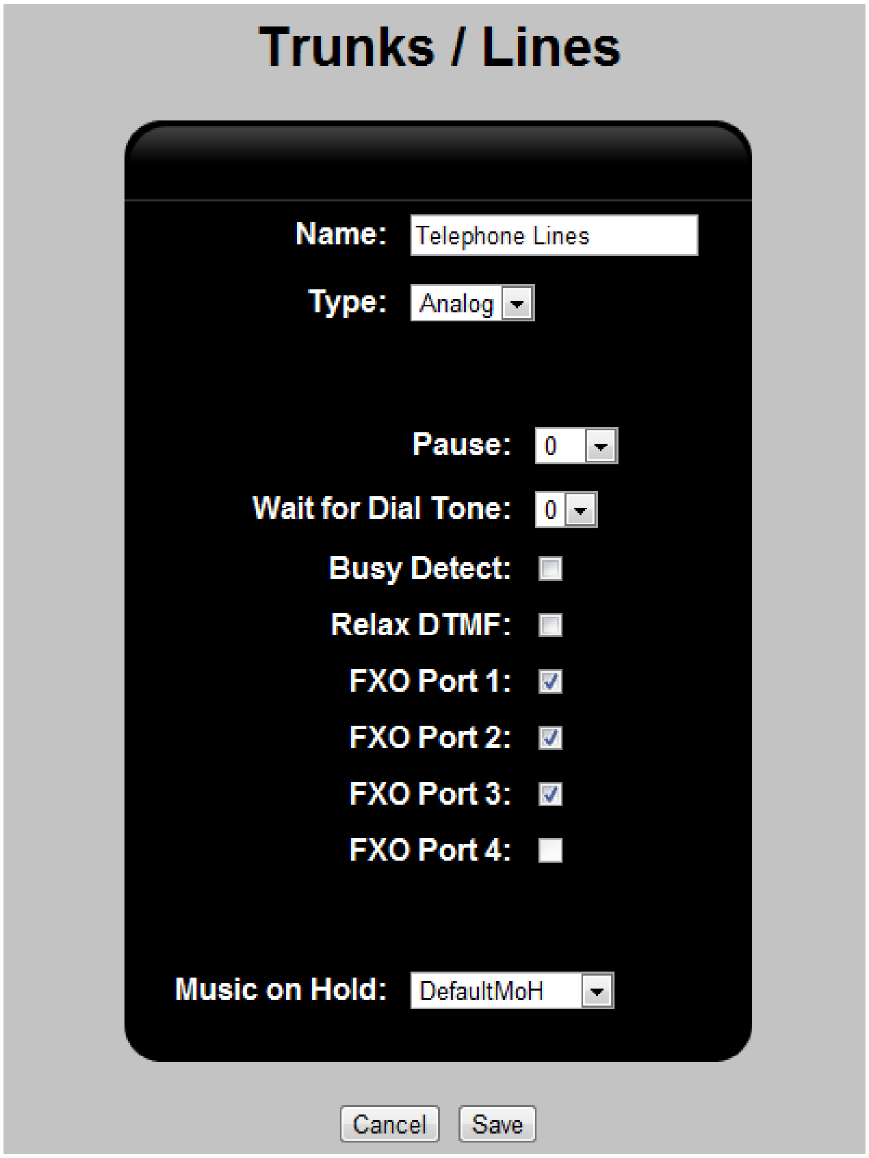
Trunks and lines are used to allow the system to make and receive calls. The calls connect using VoIP, or traditional telephony lines. When connecting to these service providers, the connection made is typically referred to as a “trunk.” A trunk can use regular analog lines or a VoIP carrier.

Analog lines are used to connect to the traditional PSTN (Public Switched Telephone Network). G-Series Gateway units with FXO ports offer Kewl Start or Loop Start. They can accommodate up to four analog lines. A single analog phone set can be connected to each available FXS port.

NOTE: The FXO auto pass-through feature will be activated In the event of a power failure:

G124 Only:
An analog phone on FXS 2 is connected to the analog line on FXO1.

G1212 Only:
An analog phone on FXS 2 is connected to the analog line on FXO9.



Name:

Name to identify the line or lines being configured.

Type:

Drop-down menu allowing selection whether line is analog or VoIP

Group Number:

A group is one or more lines with common characteristics – for example, if four lines are available, but only three should be used for outgoing calls, those three should be in one group and the remaining line should be assigned another group. Group numbers are chosen from the **Group Number** drop-down menu.

Pause:

Number of seconds to pause before dialing number on analog lines (FXO)

Wait for Dial Tone:

Number of seconds to wait for a dial tone. To disable dial tone detection, enter 0.

NOTE: It is recommended NOT to enable dial tone detection.

Busy Detect:

Detects far-end hang-up and busy signals. A drop-down menu offers Yes and No choices.

Relax DTMF:

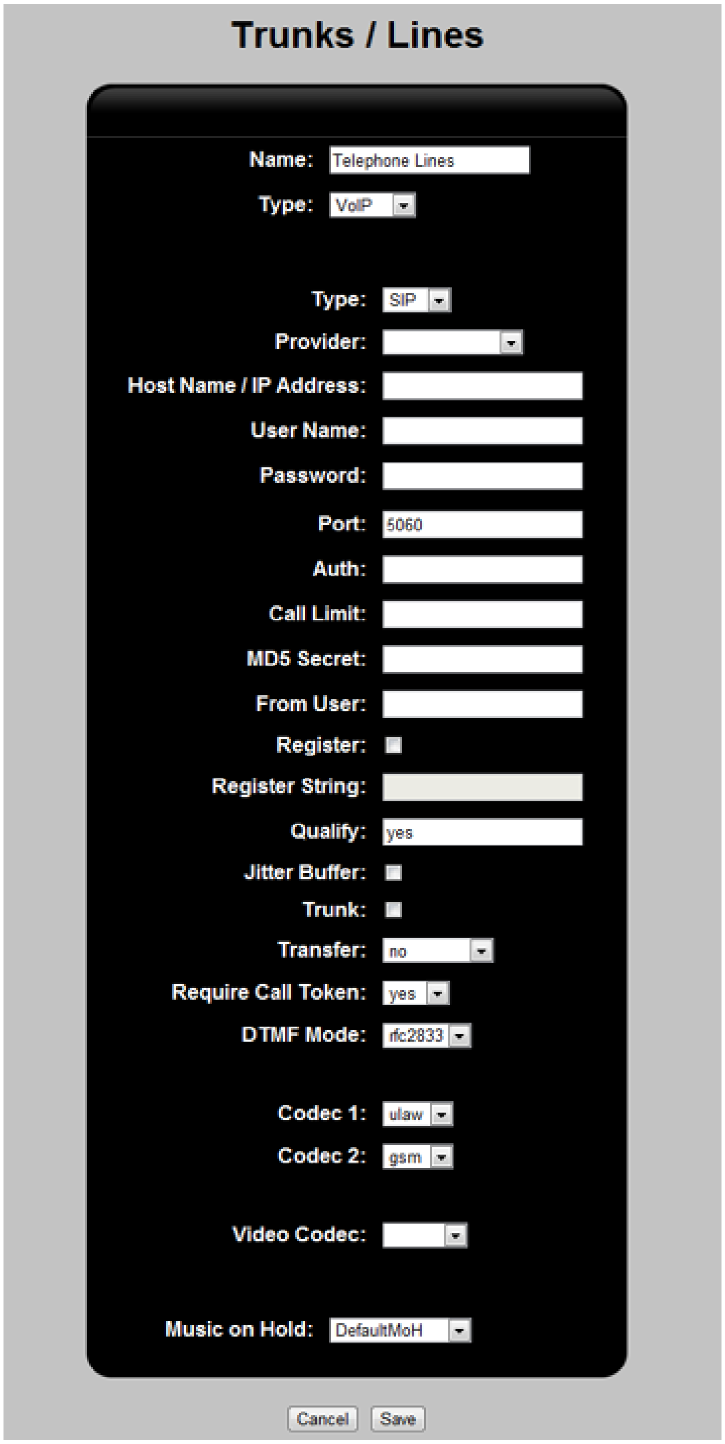
Used to apply loose or strict DTMF tone detection to the line.

NOTE: The default is Yes, which will allow incoming calling users to press keys for varying length of time when selecting menu items. If users have difficulty selecting items correctly from the menu system, choose No to reduce errors.

G124, G122, G1212 Only:

FXO Port 1 - 4:

Selects which port should be a member of the group



Name:

A text name to identify the line or lines being configured.

Type:

Select **VoIP** from the drop-down menu.

NOTE: Up to 4 VoIP lines can be configured.

Provider:
The name of the VoIP carrier.

Host Name / IP Address:
The URL or IP address as provided by the VoIP carrier.

Username:
Username as provided by the VoIP carrier.

Password:
Password as provided by the VoIP carrier.

Port:
Default port is 5060. Some carriers will provide service on a different port.

Auth:
MD5 – information supplied by VoIP carrier

Call Limit: (SIP Only)
Information supplied by VoIP carrier

MD5 Secret: (SIP Only)
Information supplied by VoIP carrier

From User: (SIP Only)
Information supplied by VoIP carrier

Register:
Dependant on VoIP carrier. It is recommended to have it set to “Yes”, unless trunks are created within a LAN.

Register String:
Information supplied by VoIP carrier

Qualify:
If "qualify=yes", Asterisk will send an option packet every 60 seconds to see if the destination is still online. If the destination does not respond within two seconds for 7 tries in a row, it will be marked as unreachable.

If qualify=XXX (XXX being a value in milliseconds), Asterisk will send an option packet every XXX milliseconds, to see if the destination is still online. If the destination does not respond within two seconds for 7 tries in a row, it will be marked as unreachable

Jitter Buffer: (IAX2 Only)
Information supplied by VoIP carrier

Trunk: (IAX2 Only)

Information supplied by VoIP carrier

Transfer: (IAX2 Only)

Drop-down menu which allows for selection among different native transfer options. The default values should be correct unless the provider specifies otherwise.

- yes
- no
- media only

Require Call Token: (IAX2 Only)

Drop-down menu which allows for specification of telephone validation options. The default values should be correct unless the provider specifies otherwise.

- yes
- no
- auto

DTMF Mode: (SIP Only)

Drop-down menu which allows for selection among different DTMF signaling protocols – typically country-dependant.

- auto
- rfc2833
- inband
- info

Codec 1 – 2:

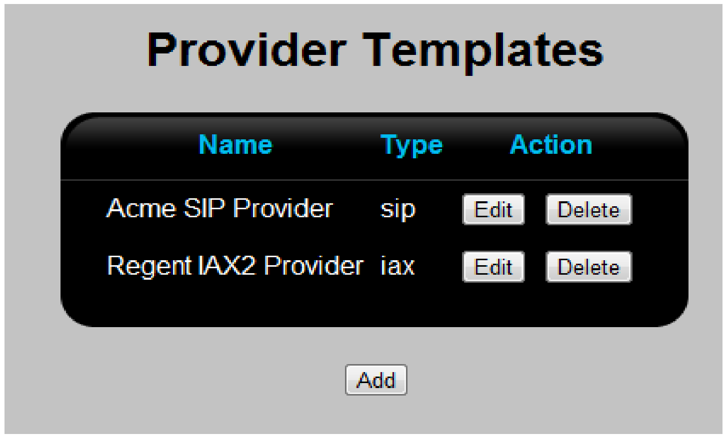
Drop-down menus offering selections of codecs – typically ISP-dependant. The documentation that comes from your SIP carrier should specify the preferred codec used by that network.

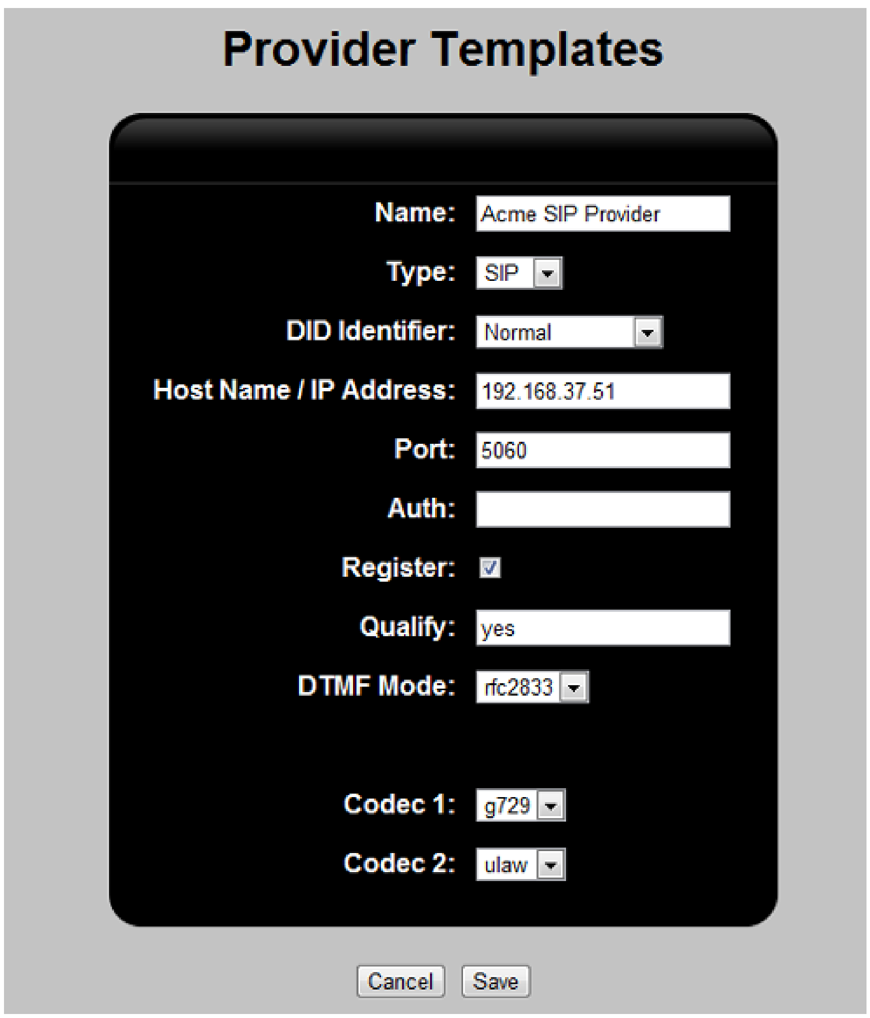
Music on Hold:

Drop-down menu for source of music on hold:

- DefaultMoH
- DefaultAudioIn

PBX -> Trunks / Lines -> Provider Templates Menu





Name:

Name of the provider which will be shown in the drop-down box.

Type:

Specify the appropriate setting according to the documentation

- SIP
- IAX2

DID Identifier:

Specify the appropriate setting according to the device manufacturer's documentation

- Normal
- RDNIS
- To-SIP-Header

Host Name / IP Address:

Information for the SIP carrier either the name e.g.: sip.babytel.ca or its IP address

Port:

Default to `5060`, can be changed if carrier needs different port.

Auth:

Information to come

Register:

To enable registering of the trunk

Qualify:

If "qualify=yes", Asterisk will send an option packet every 60 seconds to see if the destination is still online. If the destination does not respond within two seconds for 7 tries in a row, it will be marked as unreachable.

If qualify=XXX (XXX being a value in milliseconds), Asterisk will send an option packet every XXX milliseconds, to see if the destination is still online. If the destination does not respond within two seconds for 7 tries in a row, it will be marked as unreachable

DTMF Mode: (SIP Only)

Drop-down menu which allows for selection among different DTMF signaling protocols – typically country-dependant. Specify the appropriate setting according to the device manufacturer's documentation:

- Auto
- Rfc2833
- Inband

- Info

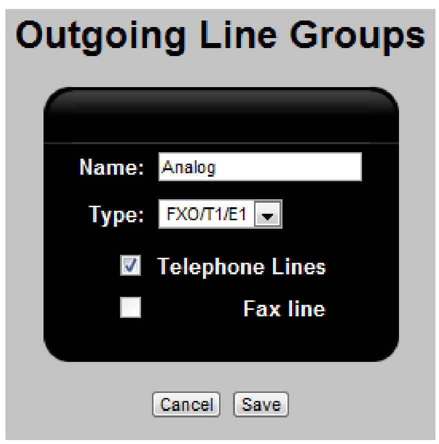
Codec 1:, Codec 2:

Drop-down menu listing installed codecs. If a required codec is not listed, contact Positron Telecom to find out if the appropriate codec is available.

PBX -> Trunks / Lines -> Outgoing Line Groups Menu

Configures outgoing line groups. In the case of analog Gateway units, these can include analog telephone lines, a fax line and a VoIP line.

Main Menu



Analog line group configuration

Name:

Text name for line group

Type:

Drop-down menu offers choice of:

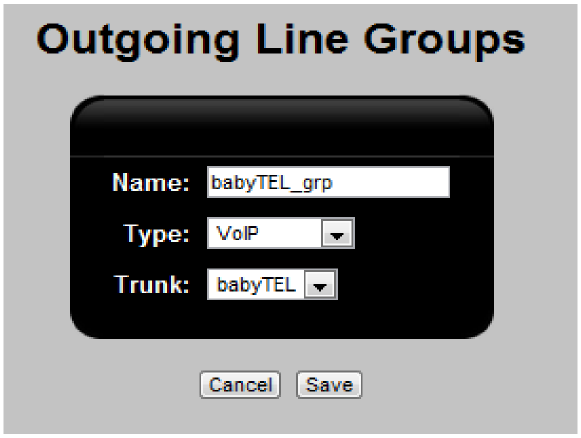
- VoIP
- FXO / T1 / E1

Telephone Lines (FXO / T1 / E1 Only)

Checked if group is analog telephone line(s)

Fax line (FXO / T1 / E1 Only)

Checked if group is to be a fax line



Trunk: (VoIP Only)

The drop-down menu provides choice of preconfigured VoIP carriers. Contact Positron if a local carrier should be added to the menu.

PBX -> Incoming Call Features Menu

The **PBX -> Incoming Call Features Menu** offers access to:

- IVR Menus to configure the Interactive Voice Response (IVR) system.
- Time Frames which are used to create time-based incoming (routing) rules.
- Conference Bridges.
- Park and Page used to park calls into the parking lot and announce the calls over the (audio-out port) paging speaker.
- DISA (Direct Inward System Access) can allow calls from outside your PBX to be interpreted as if they are inside the PBX.

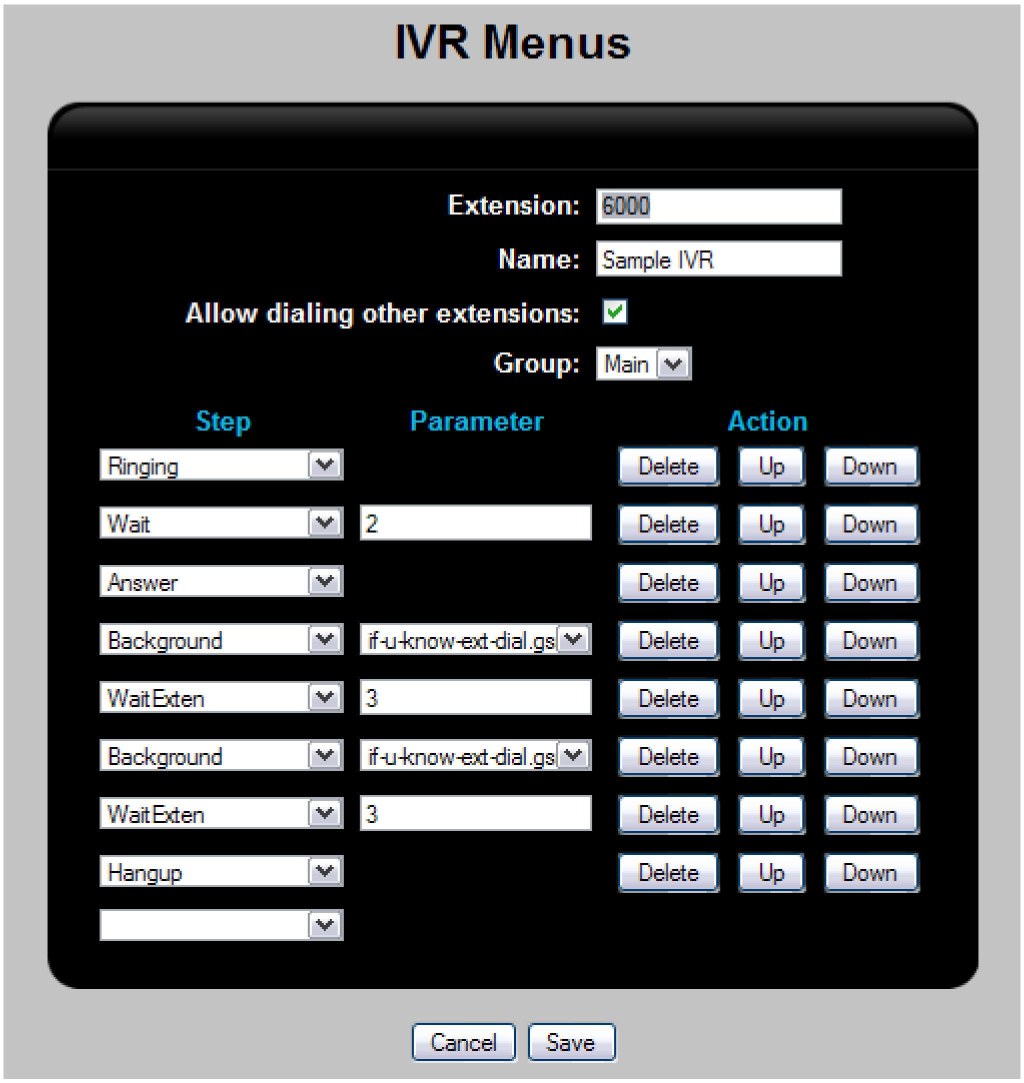
NOTE: For security purposes, the DISA application will verify that the incoming CallerID has been configured in the DISA table. If the number has not been configured, the caller will not be provided with a dial tone.

The principle is the following: when the caller dials into the PBX and invokes the DISA application, the DISA application in turn requires the user to enter his/her password. The call is then disconnected. If the password is correct, the caller will get a call back from the system and immediately hear dial tone, on which a call may be placed.

When using **DISA-NoPassword**: a password is not required; the DISA application will provide dial tone immediately, so long as the incoming CallerID matches the **Phone Number** field configured in the DISA table.

PBX -> Incoming Call Features -> IVR Menus

Allows the creation and editing of Interactive Voice Response (IVR) menus.



NOTE: The IVR extension must first be created

➢ **To create an IVR menu:**

- Open **PBX -> Incoming Calls -> IVR Menus**
- Click **Add** to open the new IVR. The IVR screen will appear
- Enter the new IVR's extension and name.

NOTE: If an IVR is to be directed to another IVR, it can have a letter in the first position of its **Extension** text box. This way, the IVR cannot be dialed from a telephone set, and has to be initiated by another IVR.

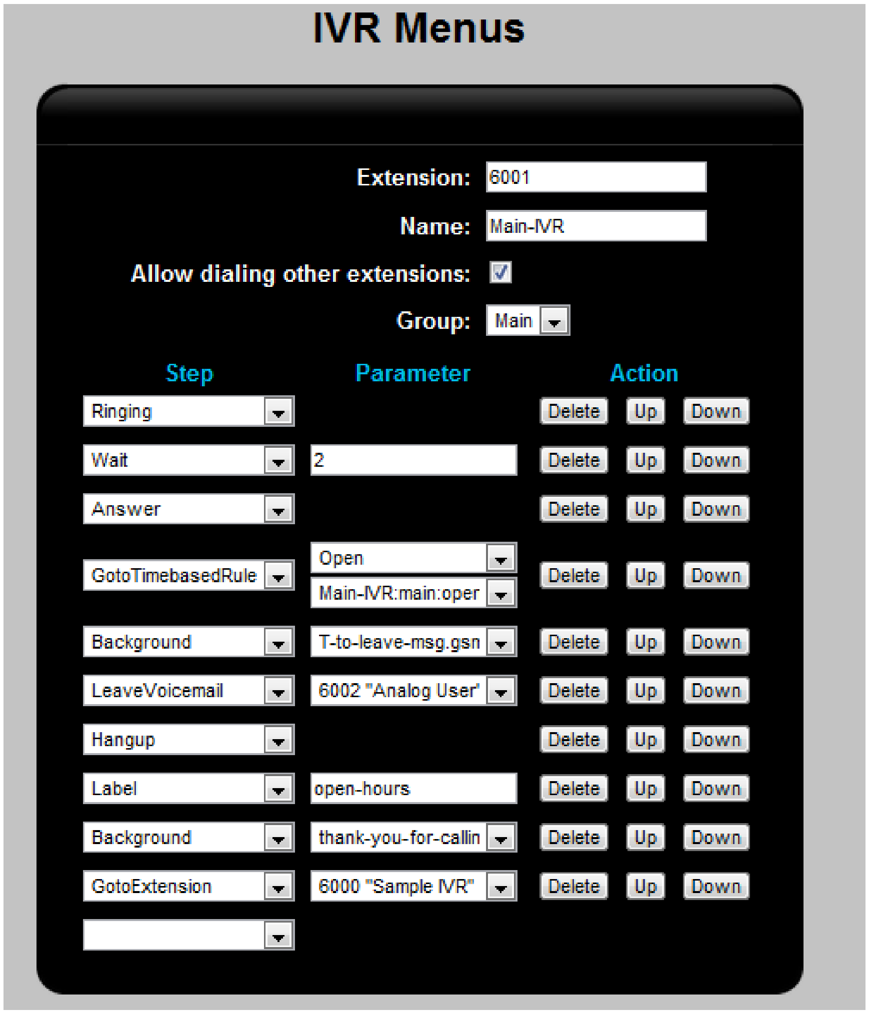
- Select the **Allow dialing other extensions** checkbox to allow users to dial extensions at any time in the IVR process.
- In the **Step** column, select **Answer** from the drop-down menu. (Answer is the step that begins all IVR applications.) A new drop-down box will appear below the first box.
- Select the next command from the new drop-down box.
- If a parameter is required, it will be shown beside the associated **Step** box. Some parameters will be chosen from the drop-down menu, others will require being filled in.
- Buttons in the **Action** column allow for moving step up or down or deleting it.
- Blank steps will be ignored
- Click **Save**
- Click **Apply Configuration**
- Any other extensions that will be used within the IVR menu should also be created at this time. These extensions can be for people, departments, ring groups and departmental voicemail boxes.

NOTE: Before navigating away from this page for any reason, remember to click **Save** to save your work to that point. If you do not click **Save**, the interface will not retain any work to that point.

NOTE: After verifying the operation of the IVR menu, be sure to click **Apply Configuration** to make the changes active.

➢ **To create extensions**

- See: PBX -> Users -> Users menu



IVR Menu Commands

Answer

Answers the call. Usually this is the first step in an IVR menu.
Parameter: none

Authenticate

Authenticate the caller with a numeric password defined in the parameter field. The caller has to enter the right password in order to continue to the next step.
Parameter: Numeric password
Example: `1234`

Background

Plays a sound file in background while waiting for user input. Playback of the sound file ends when user begins key presses.
Parameter: Available sound files listed in a drop-down menu.
Example: `welcome-greet.gsm`

- **To create or upload a sound file:**
  - See: PBX -> Sound Manager -> Sound Manager menu

Busy

Plays a busy tone indefinitely to the caller.
Parameter: none

Check Voicemail

Allows caller to enter a voicemail box extension to retrieve messages. Caller will then have to enter password.
Parameter: none

Congestion

Plays a fast busy tone indefinitely to the caller.
Parameter: none

Dial

Dial the phone number defined in the parameter field using the outgoing line group.
Parameter 1: Outgoing line group
Parameter 2: Phone number
Example:
```
Analog
5143452221
```

- **To create or modify an outgoing dial group:**
  - See: PBX -> Trunks/Lines -> Outgoing Dial Groups menu

DigitTimeout

Sets the maximum amount of time between digits when the user is typing an extension.
Parameter: time in seconds
Example: 3

DISA

Direct Inward System Access (DISA). The application will ask for a password and then validate it with the DISA table which lists allowable phone numbers and passwords.
Parameter: none

- **To create or modify the DISA table**
  - See: PBX -> Incoming Call Features -> DISA menu.

DISA-NoPassword

Direct Inward System Access (DISA). The application will NOT ask for a password but validates the user based on the CallerID received. The CallerID is validated with the DISA table defined.
Parameter: none

GotoDirectory

The caller is directed to the name directory where they can enter the first three characters of the name using the telephone keypad. The system will then play the matched choice(s) to the caller and, based on the selection made, the caller is sent to the appropriate extension.
Parameter: first name or last name (depending on directory setting)
Example : `7,6,4` will result in system looking for user with last name beginning with `Smi`.

GotoExtension

Sends the caller to the specified extension.
Parameter: Drop-down menu list of user extensions
Example: `6005 "Operator"`

GotoMenu

Sends caller to the top of the specified IVR menu file.
Parameter: IVR menu file
Example: `SalesMenu-IVR`

GotoTimebasedRule

Transfers control to another IVR program or IVR-label if a selected time frame is currently in effect.
Parameter 1: Name of time frame to be applied
Parameter 2: IVR to be transferred to if this time frame is in effect
Example: `open-hours`
`sales -IVR`

- **To create an IVR program or IVR-label**
  - See: PBX -> Incoming Call Features -> IVR Menus

- **To create a time frame:**
  - See: PBX -> Incoming Call Features -> Time Frames menu

Hangup

Terminate call.
Parameter: none

Label

Tags previous step with a label name. Used to allow entry into the IVR such that execution begins at previous step. Useful in skipping a number of preliminary steps in an IVR application.
Parameter: label name
Example: `NoReply`

➢ **To create an IVR program or IVR-label**

- See: PBX -> Incoming Call Features -> IVR Menus

Language

Sets the language for the voice channel. Any system level prompts played to the callers on this specific channel will be in the language chosen.
Parameter: Available languages chosen from drop-down menu
Example: `English`

➢ **To install or upload language packs:**

- See: PBX -> Sound Manager -> Language tab

LeaveVoicemail

Sends user to specified voice mailbox.
Parameter: Extension chosen from drop-down menu
Example: `6011 "Bill Smith"`

Playback

Plays a sound file to completion without waiting for user input. When playback of the file is complete, control moves to the next step in the sequence.
Parameter: Name of sound file chosen from drop-down menu
Example: `welcome-greet.gsm`

➢ **To record or upload a sound file:**

- See: PBX -> Sound Manager -> IVR tab

PlayInvalid

Plays system sound file to advise caller that they have made an invalid entry.
Parameter: none

ResponseTimeout

The maximum amount of time permitted to key in a selection or an extension number.
Parameter: Time in seconds
Example: 5

Ringing

Plays ringing sound. Caller will hear ringing sound X seconds before system advances to next step. Generally used in combination with Wait.
Parameter: Time in seconds
Example: 3

SetCallerID

The caller ID is modified to prefix or suffix with the parameter supplied. If the parameter is numeric, the CallerID number is modified. Otherwise the CallerID name is modified as defined in parameter two.
Parameter 1: Number or a string
Parameter 2: prefix / suffix
Example: `7 prefix` , `ABC prefix`

SetMusicOnHold

Sets the music-on-hold for the channel to the sound file specified in the parameter field. MusicOnHold is defined under PBX sound manager.
Parameter: Music-on-hold sound file or DefaultAudioIn chosen from drop-down menu
Example: `DefaultMOH`

➢ **To set or change the music-on-hold file**

- See: PBX -> Sound Manager -> Music on Hold tab

Wait

Pauses execution for the specified number of seconds before moving to the next step. During the wait, the system will not listen for any key presses. When used in combination with ringing to the caller will hear ringing for X seconds
Parameter: time in seconds
Example: 3

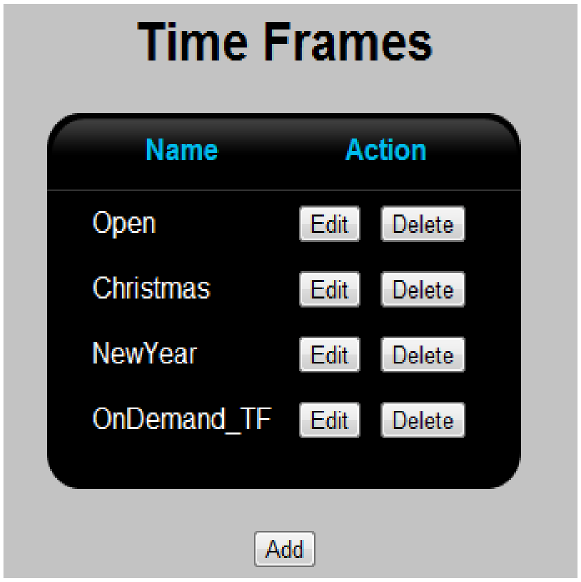
WaitExten

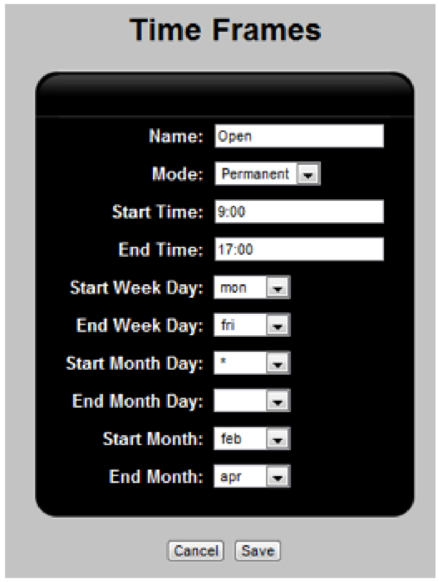
Listens for key presses in silence.
Parameter: time in seconds
Example: 3

PBX -> Incoming Call Features -> Time Frames Menu

Determine the schedule for implementing certain incoming call rules in IVR menus. For example, the organization may program its IVR to use time frames to handle incoming calls one way during open hours, and another way the rest of the time. Holiday periods, for example, national and local holidays can be defined which may override normal operating hours. Each time frame is given a name which will be available as a parameter in the IVR menus.

"Permanent " Time Frame rules are activated automatically depending on time-of-day, day-of-week, day-of-month etc. "On Demand" rules are activated/deactivated by dialing an extension and giving a correct password. The implementation of OnDemand rules is still identical to Permanent rules, by incorporating them into the IVR menu as a branch.





Name:

The name of the Time Frame.

Mode:

- Permanent– applied automatically while system is running
- OnDemand – applied only when requested

Start Time:

Start time or left blank to allow starting at midnight.

End Time:

End time or left blank to allow ending at midnight.

Start Week Day:

The day-of-week to start, or “” to choose any day.

Start Month Day

The day-of-month to start or “” to choose any day.

End Month Day

The day-of -month to end or “” to choose any day.

Start Month:

The month to start the Time Frame or “” to choose any month.

End Month:

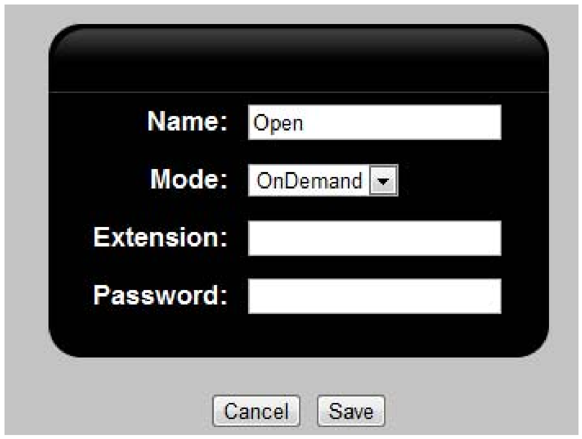
The month to end the Time Frame, or “” to choose any month.

Extension:

Used by OnDemand Time Frame rules coupled with the password to initiate an on demand rule.

Password:

Password (digits only) to enable/disable time frame.



➢ **To enable/disable an OnDemand Time Frame rule:**

- Dial the extension of the rule from a telephone
- Enter the password
- The system will provide the current status of the OnDemand time frame and will offer a menu to Enable or Disable the time frame.
- Any subsequent calls using the appropriate IVR menu will be branched by the status of the rule.

Example:

The Time Frame called “Open” above is in effect from 9AM to 5PM (using 24-hour clock), Monday to Friday of any month, starting immediately.

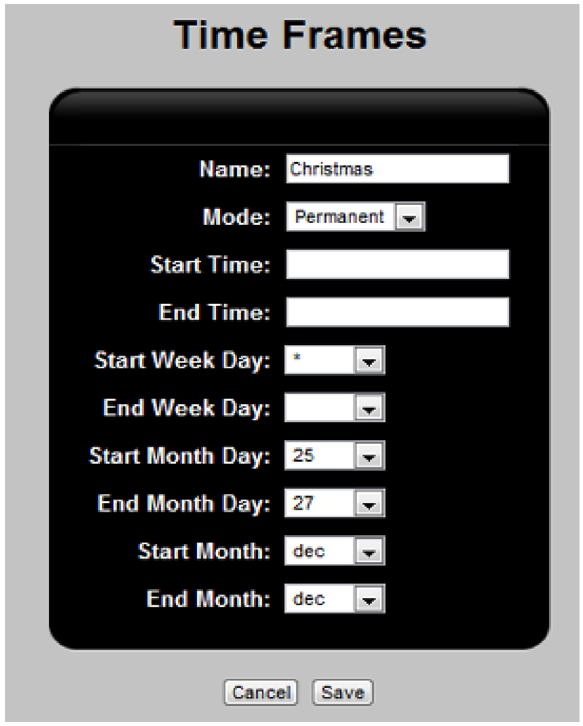
NOTE: “*” is used to indicate “don’t care.”

The organization may choose to allow use of outgoing analog lines during all periods while the Open time frame is in effect.

Example:

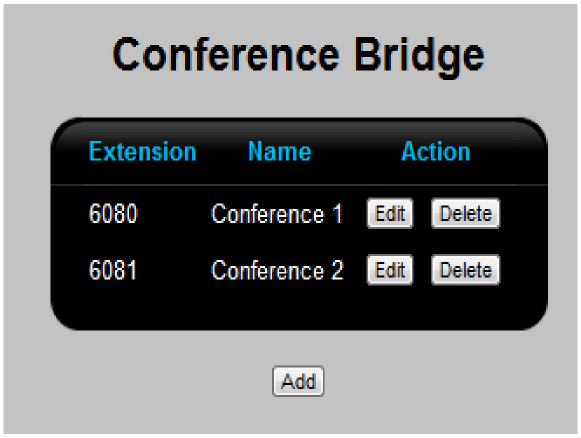
The Time Frame called “Christmas” below is in effect from the 25th of December to the 27th of December inclusive.

NOTE: “*” is used to indicate “don’t care.”


Time Frames
Name:
Christmas
Mode:
Permanent
Start Time:
End Time:
Start Week Day:
*
End Week Day:
Start Month Day:
25
End Month Day:
27
Start Month:
dec
End Month:
dec
Cancel
Save

PBX -> Incoming Call Features -> Conference Bridge Menu

The Conference Bridge feature allows quick, ad-hoc conferences, also known as Meet Me conferences, with or without PIN security. The Conference Bridge Menu allows the creation and editing of Conference Bridges. The default configuration provides two conference bridges, at extension 6080 and 6081.

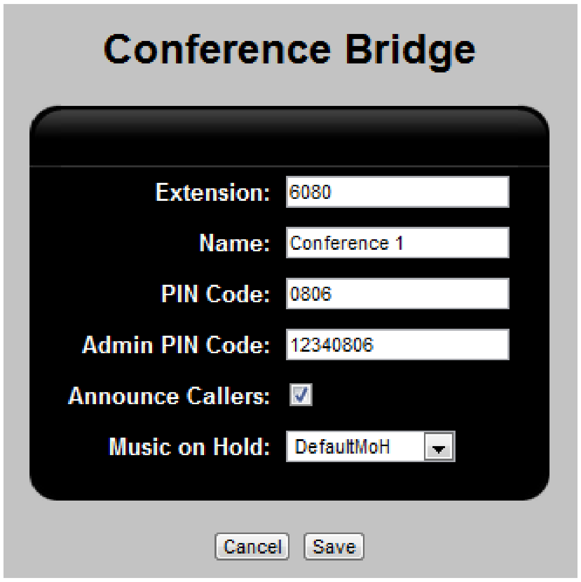


NOTE: The number of conference bridges available on the system is set to a default of 2.

➢ **To change the number of conference bridges available on the system:**

- See: PBX -> PBX Settings -> General tab

NOTE: After adding extensions, be sure to click **Apply Configuration** to make the new extensions active.



Extension:

The extension for the current conference bridge

PIN Code:

This code is given to participants in the conference. Conferees must enter the PIN code to access the conference.

Admin PIN Code:

Used by the conference administrator. The conference administrator must enter the Admin PIN code to access the conference.

When the conference administrator logs on:

- Participants will be taken off music-on-hold if it has been configured
- Participants may then be able to communicate with the conference administrator or other conferees.

Announce callers:

Check box which determines whether a system message will be played to existing conference members when a new member joins the conference.

Music-on-hold:

Name of the sound file played to participants while waiting for the conference to start.

➢ **To set or change the music-on-hold file**

- See: PBX -> Sound Manager -> Music on Hold tab

Two default Conference Bridges are included with the system:

6080

Extension:	`6080`
Name:	`Conference 1`
PIN Code:	`0806`
Admin PIN Code:	`12340806`
Announce Callers:	`Enabled`
Music on Hold:	`DefaultMoH`

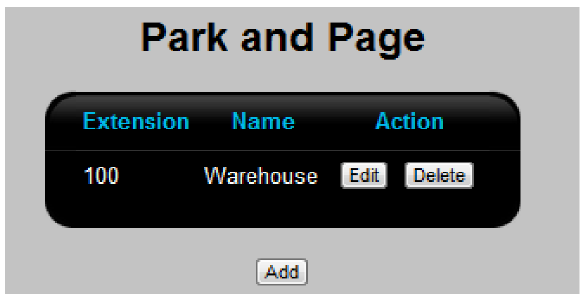
6081

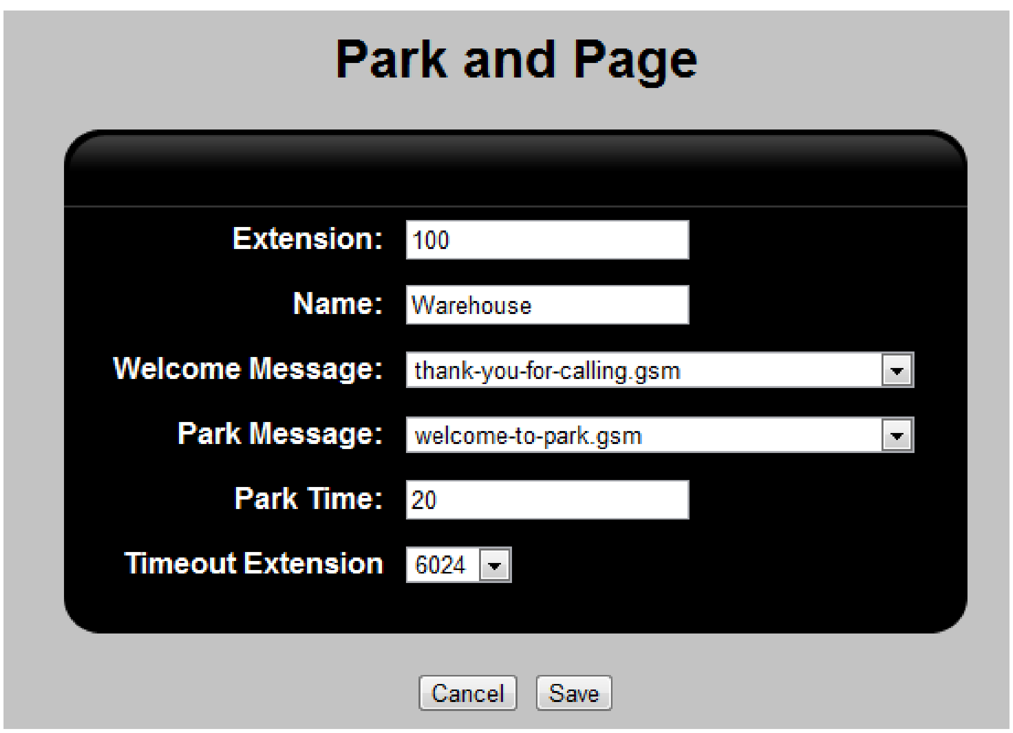
Extension:	`6081`
Name:	`Conference 2`
PIN Code:	`1806`
Admin PIN Code:	`12341806`
Callers announced:	`Enabled`
Music on Hold:	`DefaultMoH`

PBX -> Incoming Call Features -> Park and Page Menu

An incoming call can be forwarded to a park extension where it will wait to be answered by the called person. While the call is in park, a pre-recorded announcement can be made through the paging system to instructing the called person to dial the park extension and talk to the waiting caller. The caller will also be presented a recorded message to assure them that their call is being dealt with. In the event that the caller is not attended to within the "park time," the call is transferred to a voicemail box.

G122, G1000 Only:
The G-122 and G-1000 do not support Audio-Out, and therefore do not support the Park and Page feature.





Extension:

Extension to be dialed to park the caller and to speak to the parked caller.

Name:

Text name for this particular park room to appear in the Park and Page menu

Welcome Message:

Message in the form of a sound file played to the waiting caller.

Park Message:

Message in the form of a sound file played to the called party usually over the paging speaker.

➢ **To record or upload a sound file:**

- See: PBX -> Sound Manager -> IVR tab

Park Time:

Measured in seconds, this determines when the call is considered "timed out"

Timeout Extension

When a timeout occurs, the call will be sent to this extension.

PBX -> Incoming Call Features -> DISA Menu

Direct Inward System Access (DISA) allows users of the PBX to access their communications functions from outside.

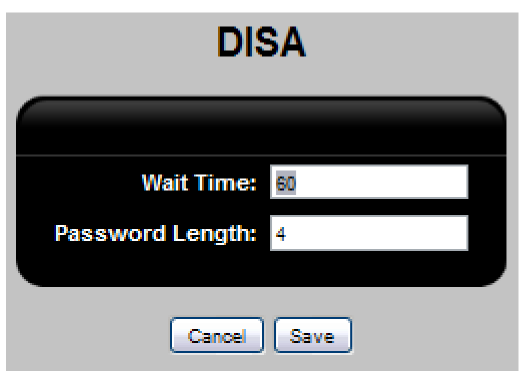
Example: A system-accredited salesperson on the road can place a call to the PBX, and be given access to analog lines or SIP trunks to make a toll-call to a customer. The call will be charged to, and bear the custom caller ID specified in the DISA table.

DISA Procedure

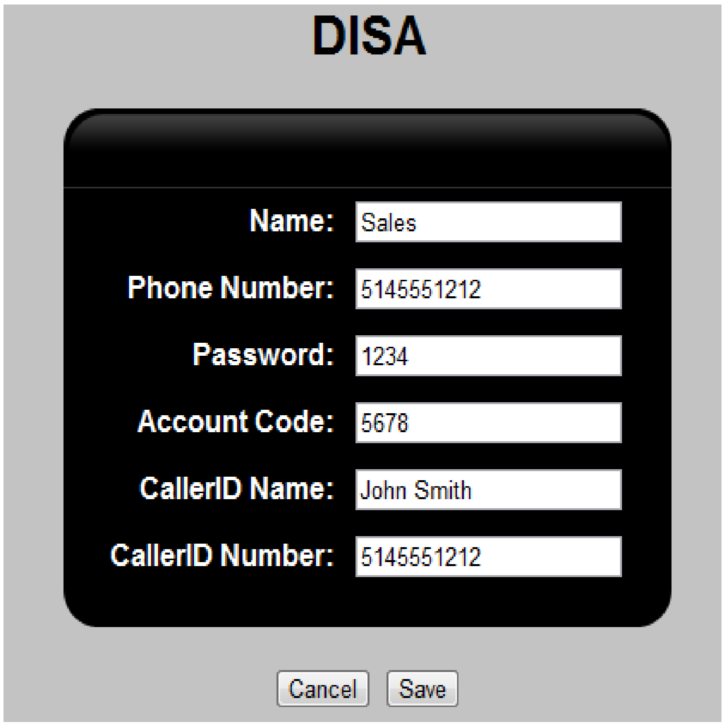
To maintain security, the procedure for using DISA is as follows:

- A DISA user calls the system, and is directed to the DISA application through the incoming call rules, or through the IVR.
- User provides the password at the prompt.
- System hangs up.
- In one minute, the system calls the telephone number identified in the **Phone Number** field of the DISA menu. The call will have the callerID of the **CallerID** field of the DISA menu.

NOTE: The **Wait Time** and **Password Length** can be configured in the Options section.



- User provides the password at the prompt and is given a dial tone.
- User can then place a call using the network's normal calling rules.
- The called party will see the callerID as entered in the **CallerID** field.



Adding Records to the DISA table

Name:

Name of the DISA user.

Phone Number:

Phone number of the (typically) cell phone used to make calls to the DISA system.

Password:

A numbers-only password which the user must present to the system.

Account Code:

This is a user-supplied billing code which will be displayed in the Call Detail Record for each call.

CallerID:

The callerID which will be shown to the user of the system when the system calls back during initial call setup, and will be shown to the called party when the call is made.

PBX -> Call Handling -> Outgoing Call Rules Menu

Outgoing call rules use dialing patterns to determine whether an external call can be completed. The dialing patterns determine the number of digits required to place a call and whether specific digits (such as an area code) must be dialed.

Dialing patterns can also govern whether the system should add or remove digits before the call is sent to the carrier. For example: if the caller must add “9” to an external number, the system can remove it before passing the number to the PSTN.

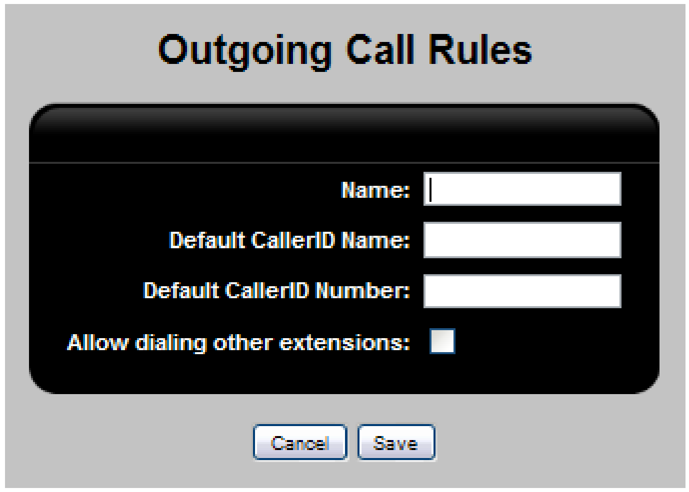
Outgoing call rules contain one or more dialing patterns. If the number dialed matches any of the dialing patterns, the call will be routed accordingly. For example, an organization may choose to route local calls through the analog trunk, and long-distance calls through the VoIP trunk.

Any dial attempt that does not match a dialing pattern within the given rule, will be checked against internal extensions or features, otherwise the caller will be given an “If Failed“ message.

NOTE: Every call made from a phone connected to the PBX is processed by these rules.

The **Add** button allows the addition of new outgoing call rule entries.

The **Outgoing Call Rules menu** defines the outgoing call entry.



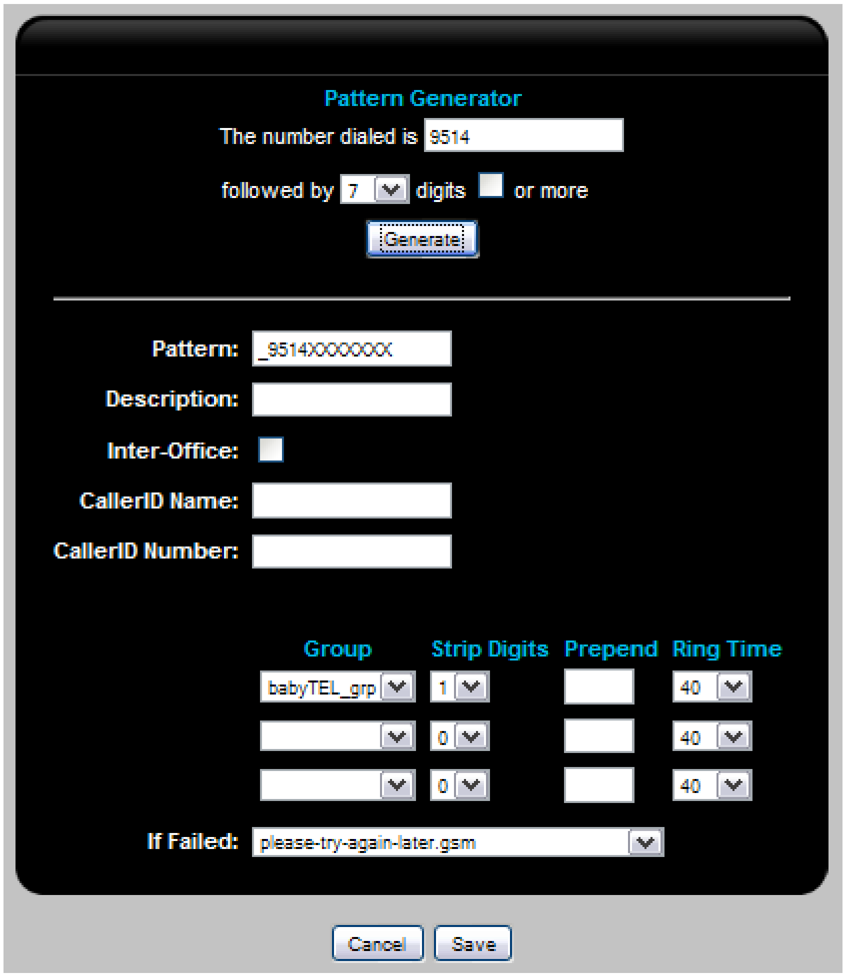
Name:
The name of the outgoing call entry

Default CallerID Name:
The CallerID name shown for the outgoing call entry.

Default CallerID Number:
The CallerID number shown for the outgoing call entry.

Allow dialing other extensions:
Allows the ability to dial extensions without any restrictions, i.e., extensions can be dialed freely without having to follow specific rules within the entry.

Rules Button:
This menu allows the creation of rules for the selected entry.



The number dialed is:
A portion of the dial pattern which will always be dialed.

Followed by:
The number of digits dialed after the first portion of the dial pattern.

Or more:
This option can be used as a wildcard to indicate any number of digits to follow is accepted.

Description:
A description used for the specific dial pattern.

Inter-Office:
Phone number provided by the service provider.

CallerID Name:
The CallerID name shown for the dial pattern.

CallerID Number:

The CallerID number shown for the dial pattern.

Group :

Select from the drop-down menu, the Outgoing Line Groups which will use the created pattern.

Strip Digits:

Select from the drop-down menu, the number of digits you wish to strip from the dialed number. In the example above, the user is asked to dial 9 to make a call; however, the 9 should be stripped before sending the call out.

Prepend:

This field allows the option to prepend digit(s) onto the dialed number.

Ring Time:

Drop-down menu offers time (in seconds), for call response before it hangs up.

If Failed:

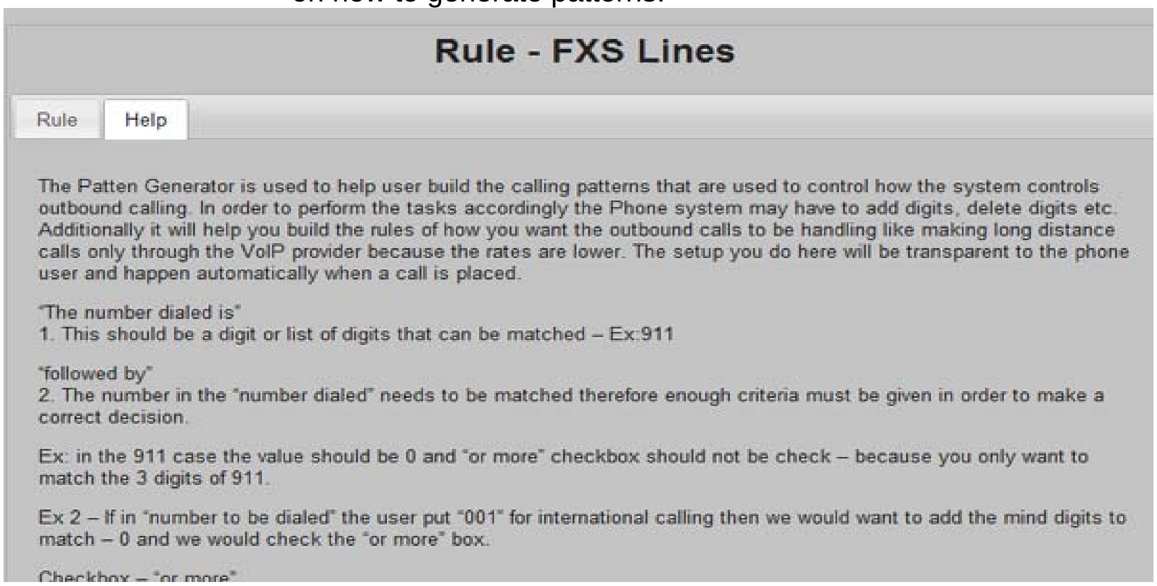
Drop-down menu offers a selection of messages to play back to the caller if the call has failed.

Patterns are executed in sequence, from top to bottom as they are listed in the Web Interface. For this reason, the rules will generally list the long patterns at the top, and shorter ones toward the bottom.

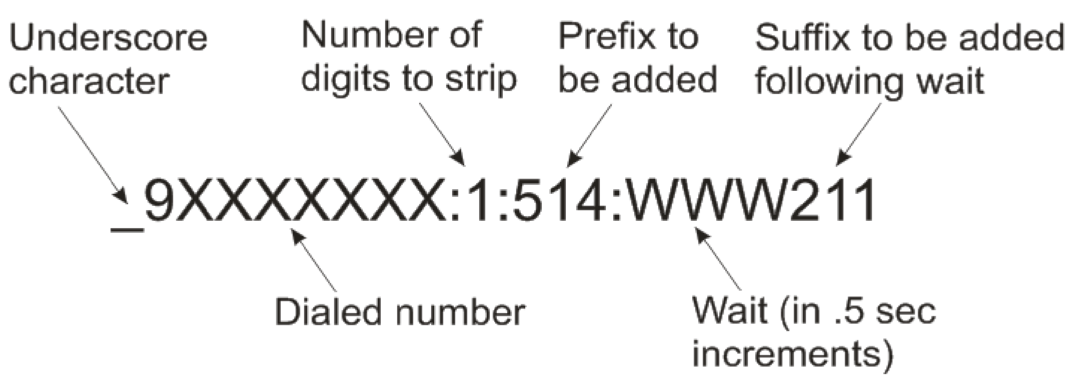
NOTE: Rules are applied in order of rank, with the uppermost rules taking precedence over the ones below. This can be controlled with the Up/Down buttons.

Help tab:

This menu explains the Pattern Generator and provides examples on how to generate patterns.



The following information can be use when creating a pattern without the use of the pattern generator:



_ (underscore) begins any dial plan that is NOT made up exclusively of digits. characters which can include:

X – representing any digit 0-9

Z – representing any digit 1-9 ("0" is not allowed)

N – representing any digit 2-9 (often used to begin any North American area code which cannot begin with "1" or "0".

[1,2,3, 7-9] – for any digits within brackets, in this example, 1,2,3,7,8,9 are all permitted.

. (period) – wildcard matching any number of digits which follow, primarily used for international calls with varying numbers of digits.

! – wildcard which causes pattern matching to stop once no other matches exist

: - (first colon) lists the number of digits to remove from the beginning of the dialed number prior to sending dialed number to outside line.

: - (second colon - optional) followed by prefix which will be prefixed to dialed number.

W - (wait .5 sec) Used only with suffixes, denotes number of .5 sec delays prior to dialing suffix

suffix – digits to be added to the end of the dialed number

Examples:

NOTE: The examples are presented in order of increasing complexity. However, the system will execute the dial plans in sequence, and allow the first rule matched to be executed.

_9NXXXXXXXXX:1 (North American ten-digit calls)
In this example, the :1 will cause the leading 9 to be stripped off, and the remaining 10 digits to be passed to the outside line. Note that in this rule, the first digit after the 9 cannot be "0" or "1". Because N and X's are used in the rule, the leading "_" (underscore) character is required.

_91NXXXXXXXXX:1 (North American toll call where "1" is required before the area code.)

_98XXXXXXXXX:1 1 (North American toll-free call using an area-code beginning with "8")
A variant would be: _918XXXXXXXXX:1 if "1" must be dialed before the "8".

_9NXXXXXX:1 (7-digit local calls)

NOTE: If local calls are permitted, then this rule must appear before the rule shown in the first example, otherwise the system will wait for three more digits to be dialed.

_91900XXXXXXX:1 (Calls to North American premium services with "900" area-code)
A variant would be: _9900XXXXXXX:1 if "1" must be dialed before the "9".

_9976XXXX:1 (Calls to “976” premium services)
Call is being made to a local “976” number.

_9011.:1 (Overseas from North America)
Strips the “9” from the dialed number, and forwards an overseas call comprising any number of digits.

95143452220:1 (Isolate unique phone number)
The phone number 514 345-2220 is isolated for special processing. A use for this might be to route calls to branch offices over VoIP.

NOTE: There is NO underscore at the beginning of the last pattern above, because no special characters found in the Pattern list above were used.

_91NXXXXXXXX:1:1010123 (“10-10” toll-call numbers)
Demonstrates prefixing a dialed number. The “9” will be stripped then 1010123 will be prefixed to the called number.

95143452220:11:5145551212 (replace entire dialed number)
Replaces the dialed number 5143452220 with 5145551212 instead.

NOTE: There is NO underscore at the beginning of the last pattern above, because no special characters found in the Pattern list above were used.

Other patterns which do not require the underscore character:
The following are actual phone numbers, there is no pattern involved, so they do not contain the underscore character

- 90:1 will strip the “9” from the dialed number, and allow contact with the PSTN operator
- 9411:1 will strip the “9” from the dialed number, and allow calls to 411 directory information in North America
- 9911:1 will strip the “9” from the dialed number, and allow calls to 911 emergency number in North America
- 911 will allow calls to the 911 emergency number in North America

Wait Time (Pause)

- In certain applications, for example, in calling automated voice response systems, or making long distance calls using passwords, it may be necessary to send a dial string, then wait for a specified time, then resume sending digits. To accomplish this, “W” is a special character defining .5 sec of wait time.
- Example: Dial 9 (outside line) then 345-2220, wait 1.5 seconds, then dial 211.
- Dialing rule: 93452220:1:3452220WWW211

PBX -> Call Handling -> Incoming Call Rules Menu

The Incoming Call Rules menu controls the routing of incoming calls. Typically, incoming calls are routed to the main IVR system, which then directs calls. However, specific incoming calls can also be routed to a conference bridge, a ring group, a virtual voicemail box, or even a particular extension. For example: Calls received on a particular line can be sent directly to a specified IVR system, while all other calls will be sent to a ring group.

The Add button on this menu allows the addition of new incoming call rule entries.

This menu defines which trunk the incoming rule applies to.

Name:
The name of the incoming call entry

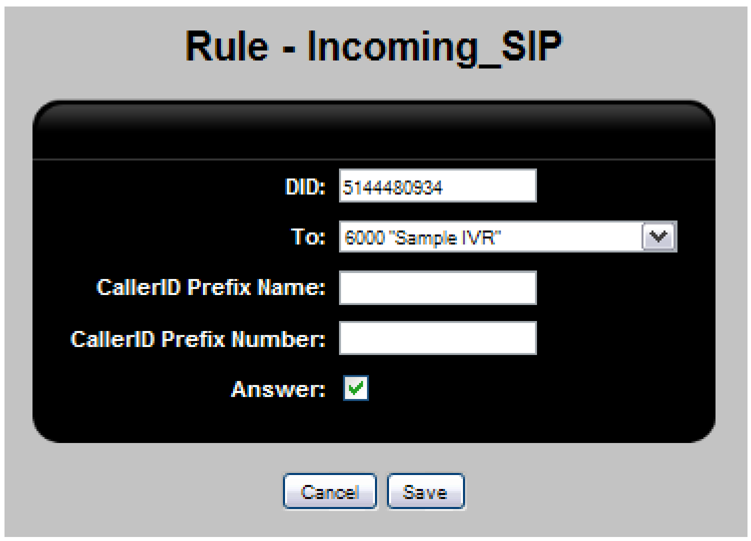
List of possible line trunks

Below the Name field will be a list of trunk lines as defined in PBX -> Trunks / Lines -> Trunks / Lines menu .

The checkbox accompanying each will identify the trunk associated with the incoming call entry.

Rules Button:

This menu allows the creation of rules for the selected entry.



DID:

Phone number provided by the service provider.

To:

Select from the drop-down menu, the ivr, ring group, virtual voicemail box… etc. to receive the call.

CallerID Prefix Name:

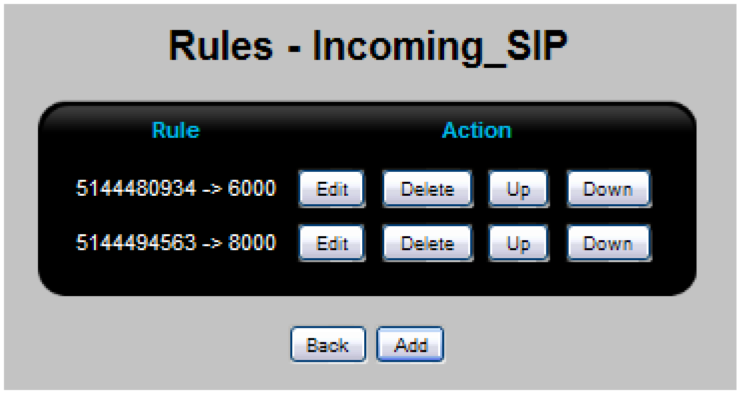
A name can be prefixed to the CallerID name, to allow the called party to determine which DID the call came in on. This information is sometimes useful in establishing which greeting to give the caller.

CallerID Prefix Number:

A number can be prefixed to the CallerID number, to allow the called party to determine which DID the call came in on. This information is sometimes useful in establishing which greeting to give the caller.

Answer:

Used by analog lines, enabling this option confirms to the PSTN that the call has been answered.



NOTE: Rules are applied in order of rank, with the uppermost rules taking precedence over the ones below. This can be controlled with the Up/Down buttons.

➢ **To add an incoming call rule:**

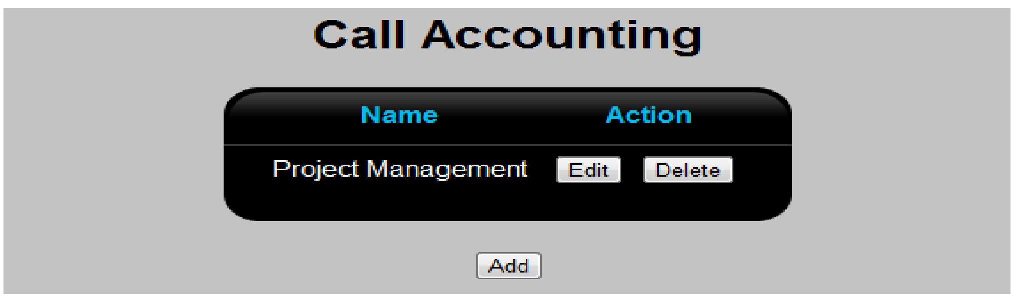
- Click **Add**
- Give the entry a descriptive name
- Select the truck from the choices presented
- Click **Save**
- From the Incoming **Call Rules Menu** click the **Rules** button
- In the Rules <name of rule> menu, click **Add**
- If the rule is to apply to a specific DID on a VoIP trunk, enter the number in the **DID field:** If the rule is to apply to calls over an analog trunk, leave it blank.
- Choose the destination for the call using the drop-down menu in the **To:** field
- If a policy has been developed to code the incoming trunks with a specific prefix, enter it in the **CallerID Prefix Name/Number:** field
- If the trunk is analog, the default is to enable the **Answer** checkbox
- Click **Save**

PBX -> Call Handling -> Call Accounting Menu

Some organizations require that the time spent on outgoing calls be tracked for accounting purposes. Often they require coding the calls by person, project type and project number.

The Call Accounting menu allows for gathering up to three levels of codes in response to voice prompts when outgoing calls are attempted. Examples of codes are: employee number, project number, work code. At each level, a voice prompt asks for a specific type of accounting code. The caller must enter a valid code, followed by the "#" key in order to move to the next step.

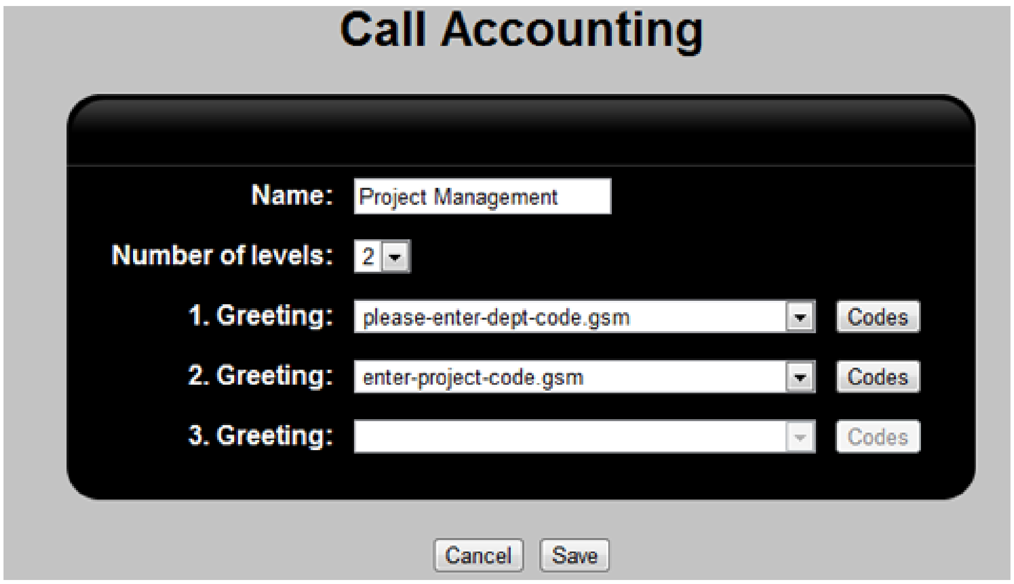
Call Accounting is enabled using the PBX -> PBX Settings -> Call Handling tab.



The Call Accounting menu allows the control of call accounting categories.

- **To add Call Accounting**
  - In the PBX -> PBX Settings -> Call Handling tab enable Call Accounting
  - Record the voice prompts required for the levels of codes using the process outlined for IVR menu recordings in the "**To record a new IVR sound file:"** section of PBX -> Sound Manager -> IVR Tab
  - In the PBX -> Call Handling -> Call Accounting menu, click Add
  - Type the name of the call accounting category in the Name field.
  - Select the number of levels of codes needed from the drop-down menu.
  - Select the voice prompt from the list of available recordings
  - Click **Code** beside the prompt.
  - Enter the name of the code category in the subsequent menu.
  - Enter the first valid code within that category.

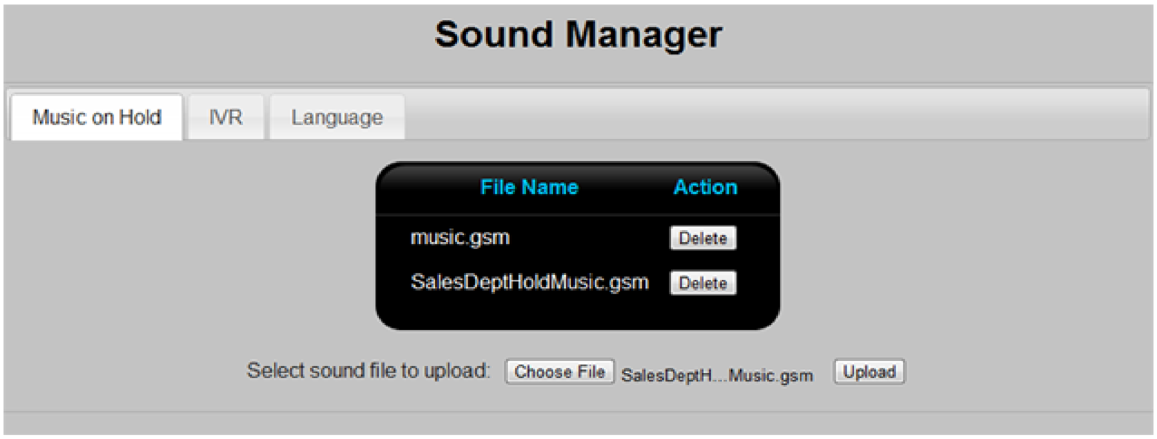
- Repeat the process of creating codes OR use a template to enter the codes valid for that level.
- Add as many levels of codes and as many codes as are needed for each level.
- Create a new Outgoing Call Rule, or modify an existing one specifying that call accounting information is to be gathered, and to present the message to be played to the caller if the codes are not correct.



VoIPon www.voipon.co.uk sales@voipon.co.uk Tel: +44 (0)1245 808195 Fax: +44 (0)1245 808299

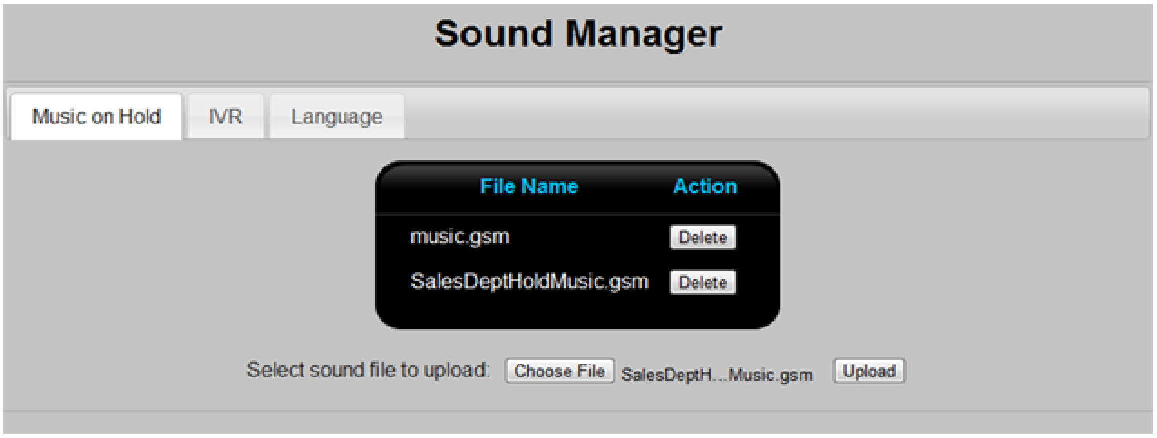
PBX -> Sound Manager -> Sound Manager Menu

Used to load or delete sound files used in the music-on-hold, IVR systems and main language systems.



This procedure uploads the files from the host file system to the G-Series Gateway unit's memory, and identifies them as music-on-hold, IVR or Language files, making them available to the system.

PBX -> Sound Manager -> Music on Hold Tab



➢ **To upload a new music-on-hold sound file into the system:**

- Click **Browse** and locate the file to be uploaded.
- Click **Upload**

NOTE: Uploading the file makes it available as a choice, but you must use the Music-on-hold menu to make it active as a music-on-hold file.

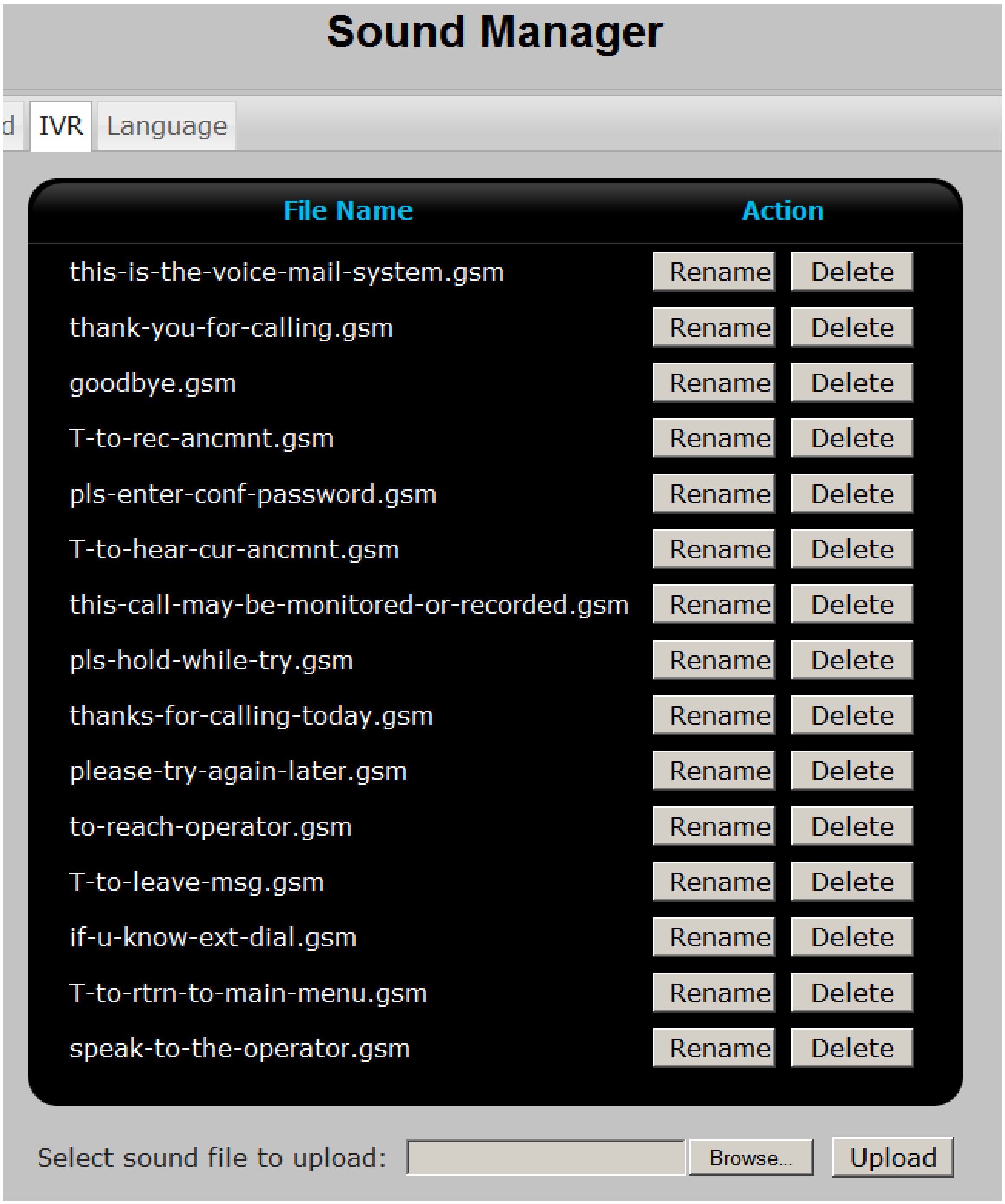
Uploading an IVR sound file makes it available to the IVR system and lists it in the drop-down menu.

➢ **To remove a music-on-hold sound file from the system:**

- Click **Delete** beside the filename.

PBX -> Sound Manager -> IVR Tab

The **IVR** Tab allows control over the sound files used in the Interactive Voice Response (IVR) system. Sound files can be uploaded from the notebook's local file system, renamed, deleted and downloaded.



File Name:

Available sound files will be listed in the drop-down menu. It is good practice to name the sound files after their contents.

Action:

Rename, download or delete selected sound file.

- **To record a new IVR sound file:**
    - Using any extension, dial the IVR recording code.
    - Default IVR recording code: *95.

NOTE: The default IVR recording code is set in:
PBX Settings -> Call Handling tab.

- After the beep signal, record the new IVR message.
- Press “#” when finished.
- Hang up the extension to end the recording.
- Go to the PBX -> Sound Manager -> IVR tab.
- Locate the new recording, identified by an extension number and timestamp.
- Click **Rename**, select the existing text and give the sound file a meaningful name.

NOTE: Do not leave spaces in the filename.

NOTE: The filetype should remain `.gsm`

PBX -> Sound Manager -> Language Tab

This tab allows the uploading and deleting of language files. These files are used to instruct users of the voicemail system.

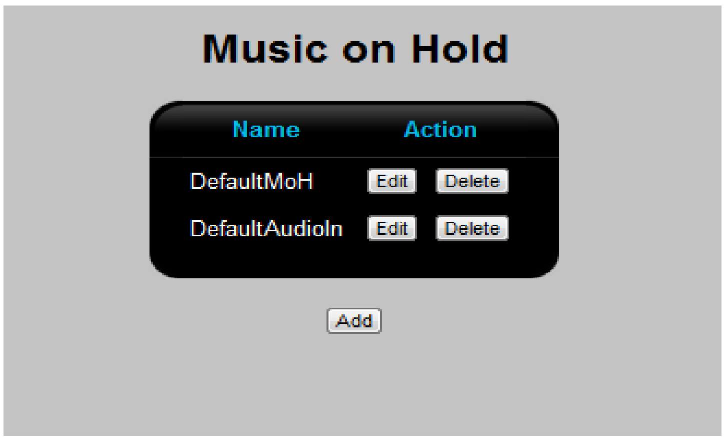
Once a new language pack has been added to the system, users can choose it from a drop-down menu when setting preferences for their voice mailbox.

- NOTE: Language sound files are selected using the drop-down menu in the PBX -> PBX Settings -> General tab.
- NOTE: Language sound files are unaffected by the **Languages**: drop-down menu on the Web Interface. That menu only affects the text of the Web Interface screens.

PBX -> Sound Manager -> Music on Hold Menu

Music-on-hold is played to callers who are on hold and to users of a conference room while they are waiting for the conference to begin.

The **Music-on-Hold** Menu allows for the management of the active Music-on-hold sound file. The files can be deleted, and uploaded from the notebook's local file system.



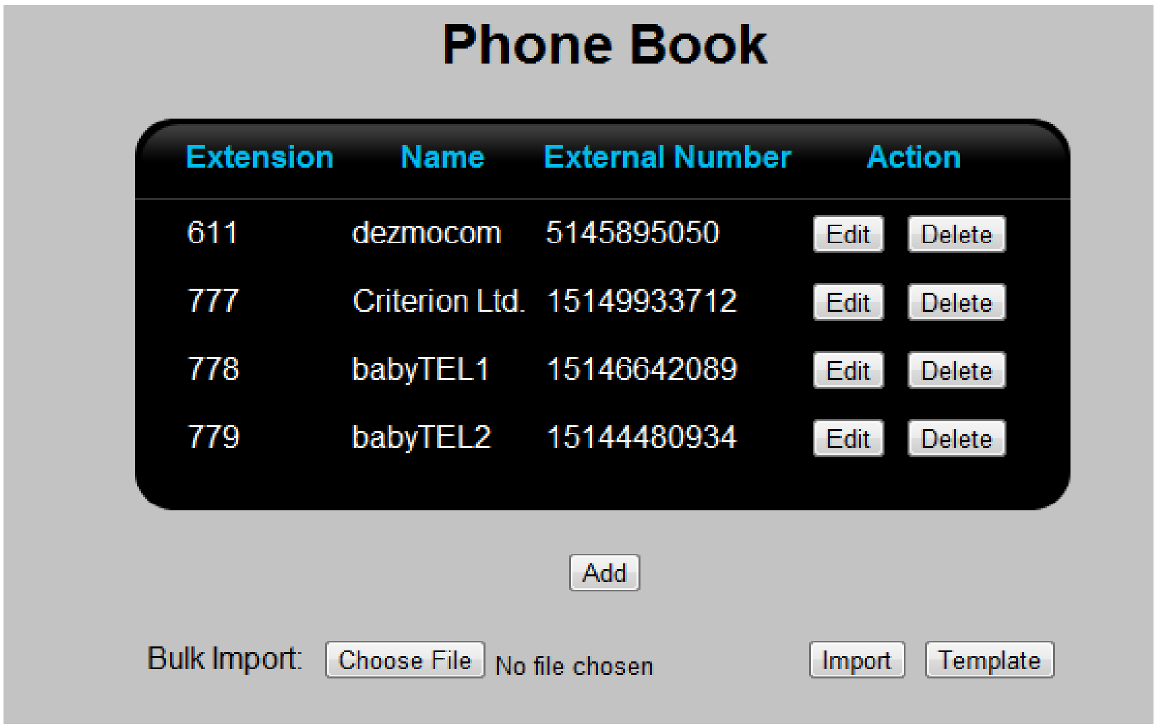
G122, G1000 Only:
The G-122 and G-1000 do not support Audio-In, and therefore will not show Audio-In as a possible music on hold source.

➢ **To upload a sound file**

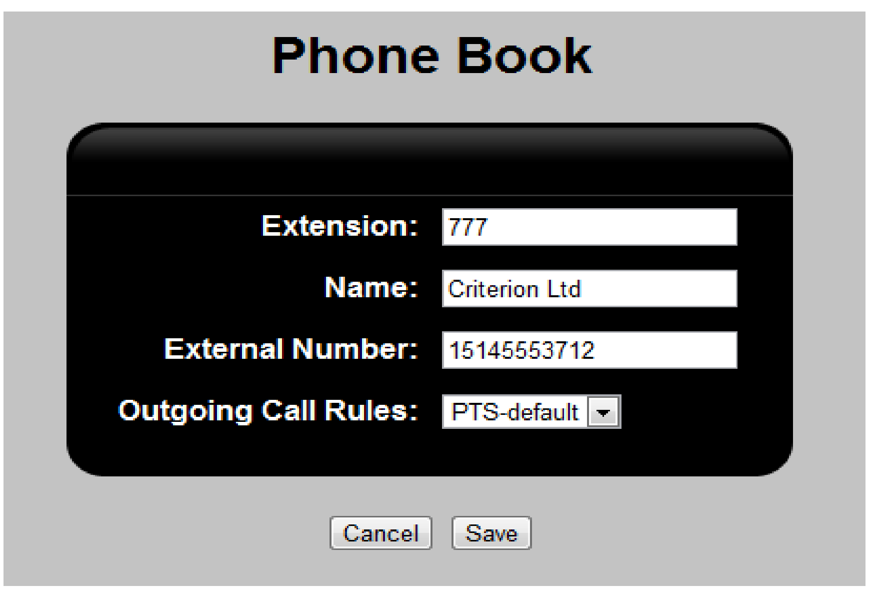
- Click **Choose File**
- Locate sound file on notebook's local file system
- Click **Upload**

PBX -> Phone Book Menu

This menu allows for the creation and management of external telephone numbers in a local phone book system. Phone numbers can be displayed, edited and deleted in the main screen. Numbers in the phone book can then be dialed as though they are extensions to the PBX system.



Entries can be added by clicking **Add**. This calls up the data window shown below.



Extension:

Extension number to be dialed to cause the external number to be dialed.

Name:

Name of individual or organization to be dialed.

External Number:

Full external number, generally with area code

Outgoing Call Rules:

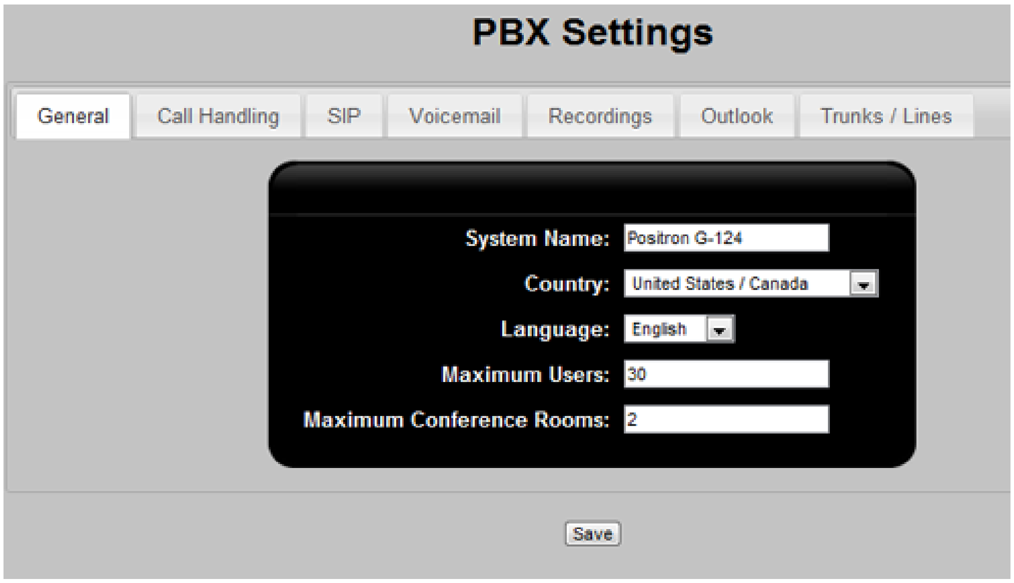
Which outgoing call rule is to be used when calling this number.

Template

By clicking **Template**, the system will download a spreadsheet in Comma Separated Values (CSV) format to allow the bulk uploading of phonebook entries.

PBX -> PBX Settings -> General Tab

Provides for general information describing the PBX system.



System Name:

Text name given to the G-Series Gateway. This is most useful when several Gateways are used in the same system.

Country:

Drop-down menu to select country of operation for the PBX. By selecting the appropriate country, certain telephony parameters will be automatically applied.

Language:

Drop-down menu to select language for the available sound files. English is the default language for sound files shipped with the system. Contact Positron for other language files.

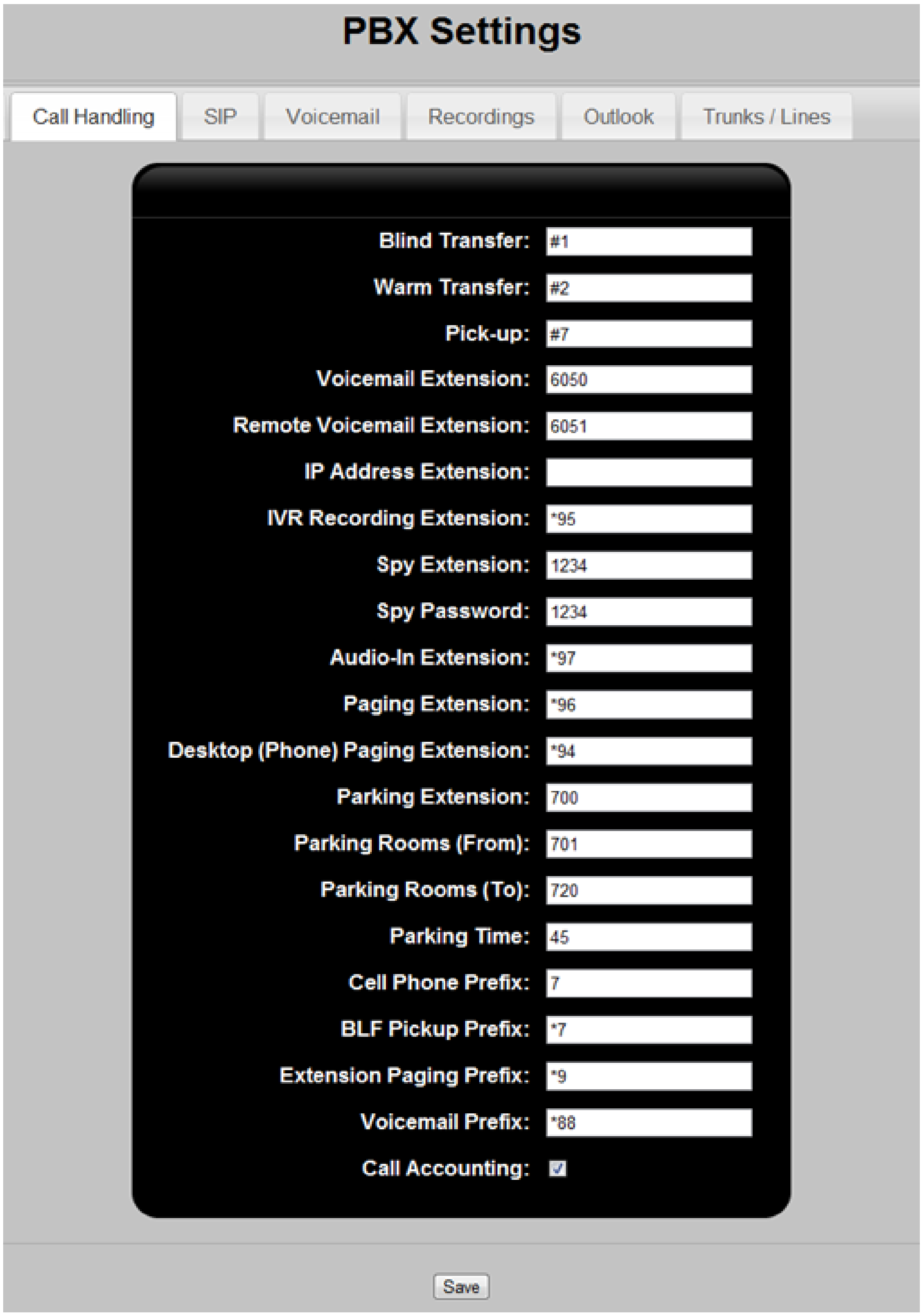
Maximum Users:

Specify the number of simultaneous users of the PBX system. Allows the system to more accurately allocate resources.

Maximum Conference Rooms:

Specify the number of conference bridges. Allows the system to more accurately allocate resources.

PBX -> PBX Settings -> Call Handling Tab



Blind Transfer:

When the transfer is done through the PBX, user enters “#1” and dials extension, then hangs up. This is sometimes called cold transfer or unattended transfer. The default value is #1

Warm Transfer:

When the transfer is done through the PBX, user enters “#2” and dials extension, talks to the called extension, then hangs up. Sometimes called attended transfer.
Default: #2

Pick-up:

Identification of the pickup groups available on the system – comma separated.
Default: #7

Voicemail Extension:

Extension to be used when system users call in for messages.
Default: 6050.

Remote Voicemail Extension:

Extension to be used when system users call in for messages from outside the network.

'0' Extension (Voicemail):

Extension to be dialed when a user of an IVR system dials 0.
Default: 6005

'*' Extension (Voicemail):

Extension to be dialed when a user of an IVR system dials *.
Default: 6051 (voicemail menu).

IP Address Extension:

Extension to be dialed when an administrator needs to find the network IP address of the Gateway.

➢ **To retrieve the IP Address of a G-Series Gateway**

- Dial the IP Address Extension

If an IP Address Extension has not been set, then dial ##**

- Enter the password
  Default password: 1234567890#
- The Gateway’s IP Address will be presented

IVR Recording Extension:

Local digits to dial to access the built-in voice recorder for IVR menus. The default value is *95

Spy Extension:

Extension to dial to initiate a call monitor on a selected extension.

Once the extension is dialed and the monitoring session begins, it remains open for all subsequent calls until the extension doing the monitoring hangs up.
Default: 1234

Spy Password:

Password to enter to validate a call monitoring user.
Default: 1234

Audio-In Extension:

Allows a PBX user to monitor the Audio Input of a G-Series Gateway.
Default: *97.

Paging Extension
(Snom and Linksys phones with multicast support):

Extension to dial to activate the page system. Pages will be typically announced over loudspeakers, though some sets have local speakers.
Default: *96

Desktop (Phone) Paging Extension:

Extension to dial to activate the page system though sets which have local speakers.
Default: *94

Parking Extension:

The park extension assigns a parking room to a call that is taken from one extension, is then put on "hold" only to be taken from another extension (usually by the called party).
Default: 700

Parking Rooms:

Range of extensions that can be assigned to parking rooms.
Default: 701-720

Parking Time:

Number of seconds of park time allowed. After this time the calls are routed to the Main IVR.
Default: 45 (seconds)

- **To park a call then resume the call from another extension:**

NOTE: In this example, the number for a blind transfer is #1, and the code for the parking extension is 700. The range 701 – 720 is available for parking rooms.

- While on a call, dial `#1` followed by `700`. The automated attendant will assign a parking room (in this example) of `705`.
- Hang up
- Pick up any other another extension
- Dial `705` (the assigned parking room)
- Resume the call.

If the call was not answered within the number of seconds specified in **Parking Time** above, the call will be routed to the main IVR program.

Cell Phone Prefix:

Allows contacting the cell phone of a PBX user by dialing the **Cell Phone Prefix** followed by the user's extension. The cell phone number is configured in PBX -> Users -> Users menu.
Default: 7
Example:
To reach the cell phone of the user of extension 6011, dial 76011

BLF Pickup Prefix:

Prefix which allows for the picking up of a call to an extension which is flashing in the Busy Lamp Field (BLF). The BLF is provisioned in PBX -> Users -> Phone Provisioning menu
Default: `*7`.

Example:
The user on extension 6012 has a BLF which shows extension 6011 is flashing and has not been answered. She picks up and dials `*76011` to answer the call.

If *76011 is put into the pickup value field under BLF in the provisioning field of extension 6012, that extension can be picked up by pressing the button associated with the flashing lamp.

Extension Paging Prefix:

Prefix which allows paging a particular extension's speaker.
Default: `*9`.
Example:
To page the person at extension 6005, dial `*96005` and speak the page.

Voicemail Prefix:

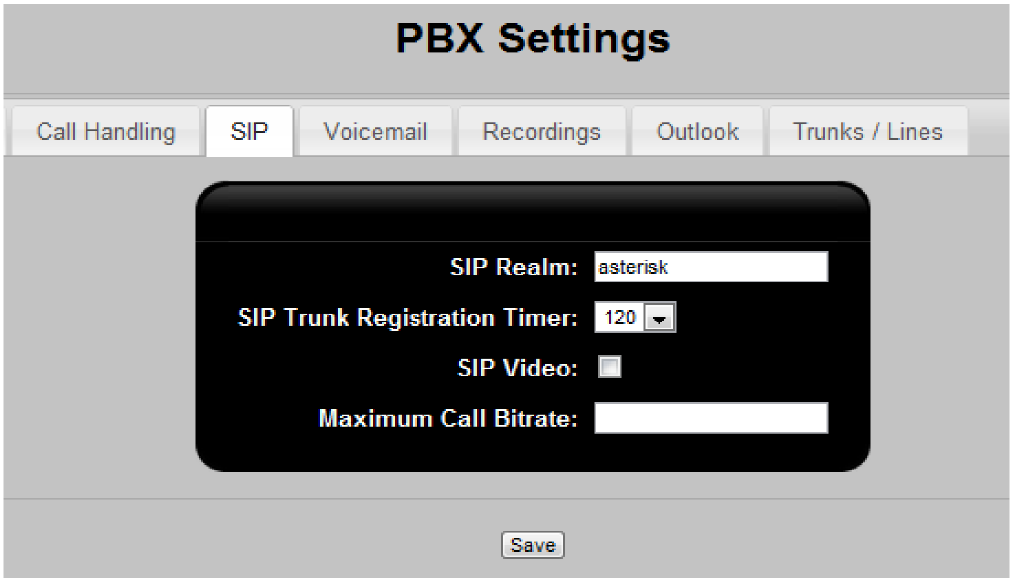
Allows a message to be left in the extension's voicemail box without ringing the extension. If the value of `*6` is entered then a call to `*66011` would leave a message in the voicemail box of extension 6011 without ringing the extension.

NOTE: `*6` is only an example, there is no default value.

Call Accounting:

Enables call accounting.
See PBX -> Call Handling -> Call Accounting menu

PBX -> PBX Settings -> SIP Tab



SIP Realm:

Used in the calculation of MD5 digits. Used by VoIP carriers.
Default: `asterisk`

SIP Trunk Registration Timer:

SIP trunks are registered every X seconds. The default is 120, but some carriers require the value to be set a little lower. Dezmocom for example, requires it to be at 90 seconds.

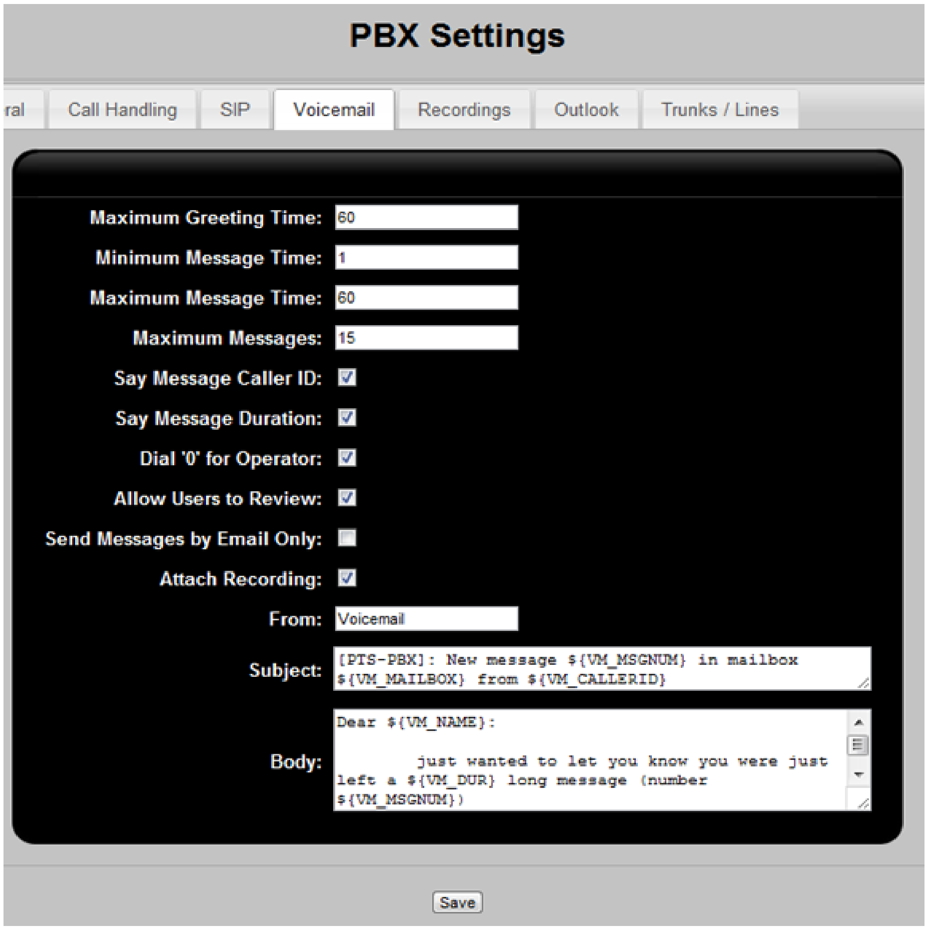
SIP Video:

If SIP video is enabled, then video codecs are provided in user templates which would enable interoffice video calls.
Default: `not enabled`

Maximum Call Bitrate:

Maximum call bit rate is the bandwidth used for the video calls.
Default : `384kbps`.

PBX -> PBX Settings -> Voicemail Tab



Maximum Greeting Time:

Time in seconds allowed for users to create their voicemail greetings.
Default: `60`

Minimum Message Time:

Time in seconds allowed for incoming voicemail box messages.
Default: `1`

Maximum Message Time:

Time in seconds allowed for incoming voicemail box messages.
Default: `60`

Maximum Messages:

Maximum number of messages allowed in a voicemail box.
Default: `15`

Say Message Caller ID:

Checkbox to enable the system to provide spoken CallerID for the voicemail message.
Default: `enabled`

Say Message Duration:

Checkbox to enable the system to provide spoken message duration time.
Default: `enabled`

Dial '0' for Operator:

Checkbox to enable callers to access the operator.
Default: `enabled`

Allow Users to Review:

Checkbox to enable the callers to review their voicemail messages.
Default: `enabled`

Send Messages by Email Only:

Checkbox to enable the system to send voicemail messages by email. Messages will NOT be saved on the PBX.
Default: `not enabled`

Attach recording:

Checkbox to enable the system to attach voicemail messages to emails.
Default: `enabled`

From:

Specifies the From: field to be used when sending voicemail messages.
Default: `Voicemail`

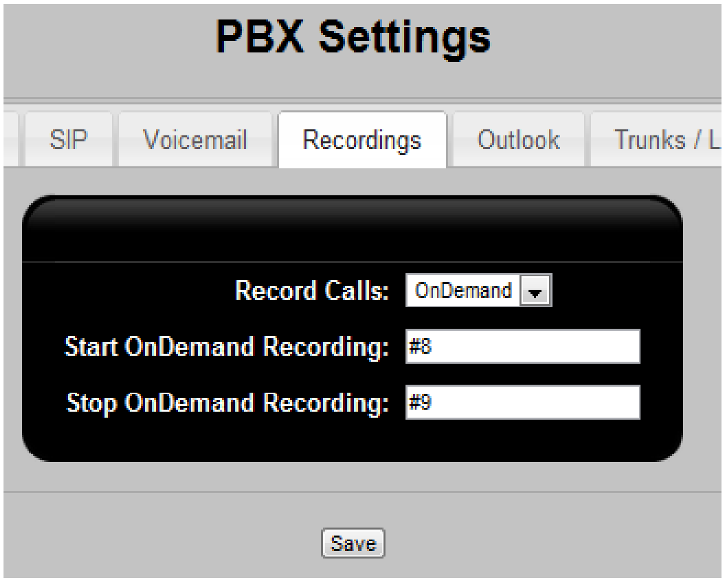
Subject:

Specifies format for the Subject field in email messages providing notification of missed calls.
Default: `message exists`

Body:

Allows specification of the body of the email sent to users with the voice mailbox message attachment.
Default: `message exists`

PBX -> PBX Settings -> Recordings Tab



Controls when and if calls are to be recorded.

Record Calls:

Choices shown in the drop-down menu are:

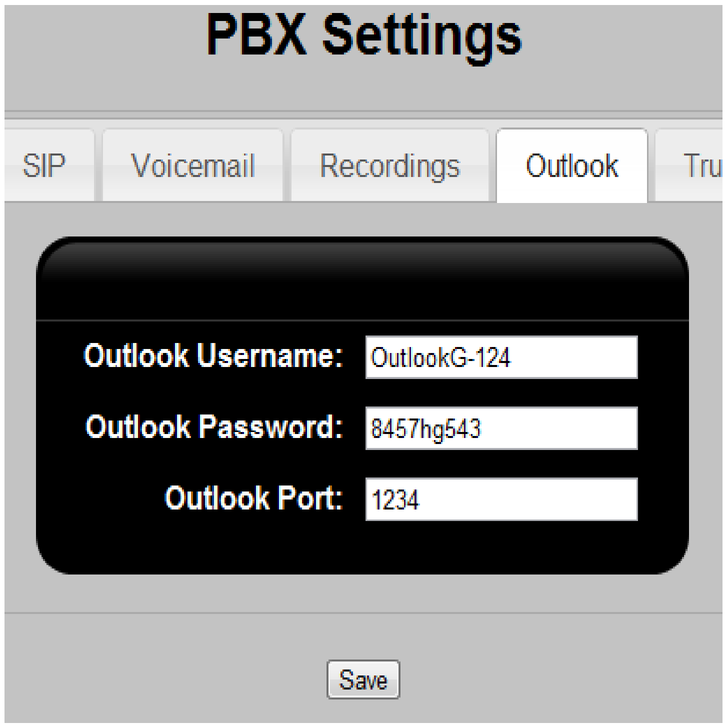
- Always
- On Demand
- Never (default)

Start OnDemand Recording, Stop OnDemand Recording:

Determines the keypress sequence to begin and end recording.
Start on demand recording Default: #8
Stop on demand recording Default: #9

PBX -> PBX Settings -> Outlook Tab

If appropriate software is installed on the organization's network, the G-Series Gateways can allow users to make calls from the organization's Microsoft Outlook contact list. The Gateway can interface with the software to allow pop-ups to appear on the user's computer with incoming call information from Outlook.

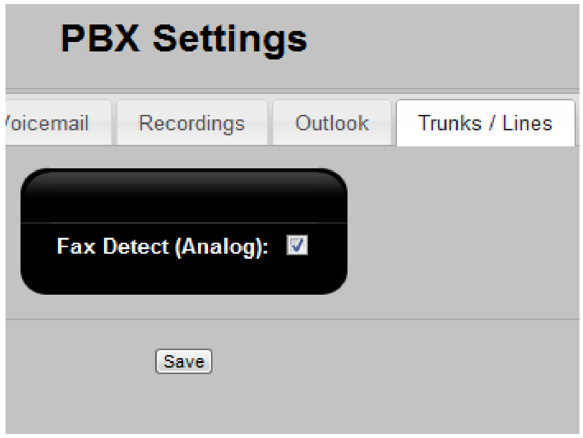


Outlook Username:
The username for the third-party software which interfaces with the organization's Microsoft Outlook system.

Outlook Password:
The password for the third-party software which interfaces with the organization's Microsoft Outlook system.

Outlook Port:
The port used by the third-party software to access the Microsoft Outlook server.
Default: 1234

PBX -> PBX Settings -> Trunks/Lines Tab



Fax Detect (Analog):

If enabled, will add “Fax” to IVR menus. Will forward incoming fax calls from the IVR to the default analog fax extension: 6003.
Default: `not enabled`

International Prefix:

Default: `00`

National Prefix:

Default: `0`

Music on Hold:

Default: `none`

Appendix 1

Acronyms

ADSI – Analog Display Services Interface

AMA – Automated Message Accounting

ANI – Automatic Number Identification

CDR - Call Detail Record

CID - Caller ID

CTI - Computer Telephony Integration

DID - Direct Inward Dialing

DNS - Domain Name System

DTMF - Dual-tone multi-frequency

FXO - Foreign Exchange Office

FXS - Foreign Exchange Station

GUI – Graphical User Interface

IAX – Inter Asterisk Exchange

IP – Internet Protocol

ITSP - Internet Telephony Service Provider

IVR - Interactive Voice Response

LAN – Local Area Network

MAC – Media Access Control

MIME - Multipurpose Internet Mail Extensions

MTU - Maximum Transmission Unit

MWI – Message Waiting Indicator

NAT - Network Address Translation

NTP – Network Time Protocol

OS - Operating System

PBX – Private Branch Exchange

PIN - Personal Identification Number

RFC - Request for Comments

RTP - Real-time Transport Protocol

RX – Receive

SIP - Session Initiation Protocol

TOS – Type of Service

TTL – Time to Live

TX - Transmit

UDP - User Datagram Protocol

URI - Uniform Resource Identifier

VoIP – Voice Over Internet Protocol

Zap – Zaptel

Appendix 2

Default Settings

The following are default settings currently shipped with the G-Series Gateways:

NOTE: As the G-Series Gateway systems evolve, the following information will change.

Login
System -> Admin Account Menu

IP Address: `192.168.1.2`
Username: `admin`
Password: `password`

Analog Sets
PBX -> Users -> Users menu
FXS (Analog) telephones: `6002, 6003`

Analog Lines
PBX -> Trunks / Lines -> Trunks / Lines menu
FXO (Analog) telephone lines 1 - 4: `Enabled`

Power Failure Configuration:
In the event of a power failure to the Positron device:

- The analog telephone connected to FXS port 2 is connected to the analog PSTN line connected to FXO port 1.
- The default extension number of the analog phone is 6003.

PBX General Settings
PBX -> PBX Settings -> General tab

System Name:	Positron G-xxx
Country:	United States/Canada
Language:	English
Maximum Users:	20
Maximum Conference Rooms:	2

SIP Settings
PBX -> PBX Settings -> SIP tab

SIP Realm:	asterisk
Registration Timer:	120
SIP Video:	not enabled

Voicemail

PBX -> PBX Settings -> Voicemail tab

Maximum Greeting Time:	60
Minimum Message Time:	1
Maximum Message Time:	60
Maximum Messages:	15
Say Message Caller ID:	enabled
Say Message Duration:	enabled
Dial '0' for Operator:	enabled
Allow Users to Review:	enabled
Send Messages by Email Only:	not enabled
Attach Recording:	enabled
From:	Voicemail
Subject:	[PTS-PBX]: New message ${VM_MSGNUM} in mailbox ${VM_MAILBOX} from ${VM_CALLERID}
Body:	Dear ${VM_NAME}: just wanted to let you know you were just left a ${VM_DUR} long message (number ${VM_MSGNUM})in mailbox ${VM_MAILBOX} from ${VM_CALLERID}, on ${VM_DATE}, so you might want to check it when you get a chance. Thanks! --Positron Telecom

Recordings:

PBX -> PBX Settings -> Recordings tab

Record Calls:	Never
Start OnDemand Recording:	#8
Stop OnDemand Recording:	#9

Outlook

PBX -> PBX Settings -> Outlook tab

Outlook Username:	
Outlook Password:	
Outlook Port:	1234

Trunks/Lines

PBX -> PBX Settings -> Trunks/Lines tab

Fax Detect (Analog):	not enabled

International Prefix:	00
National Prefix:	0
Music on Hold:	not specified

Telephone Lines

Type:	ISDN
Controller:	1
Mode:	TE
Link:	PTMP
Relax DTMF:	not enabled
Immediate:	enabled

VoIP

Type:	VoIP
Type:	IAX2
Provider:	not specified
Port:	4569
Qualify:	yes
Jitter Buffer:	enabled
Trunk:	enabled
Transfer:	no
Requie Call Token:	yes
Codec 1:	uLaw
Codec 2:	gsm

Network Settings

System -> Network -> Network tab

Host Name:	G-xxx
TFTP Server:	192.168.1.1
NTP Server:	pool.ntp.org
Time Zone:	Canada Eastern (GMT-5)
NTPD Server:	Not enabled
DHCP Server:	Not enabled
DHCP IP Address Range:	192.168.1.100
To:	192.168.1.199
DHCP Lease Time (sec):	120
DHCP Boot Server:	
DHCP Subnet Mask:	255.255.255.0
DHCP Gateway:	
DHCP DNS Server:	
Port 4:	LAN
IP Address:	192.168.1.2
Default Gateway:	192.168.1.1
DNS Server:	192.168.1.2
Mailhub:	mail

Extensions and General Settings

DTMF: United States / Canada frequencies and timing setting

Call Handling Settings
PBX -> PBX Settings -> Call Handling tab

Blind Transfer of calls:	#1
Warm Transfer:	#2
Pick-up:	#7
Operator Extension:	6005
Voicemail Extension:	6050
Remote Voicemail Extension:	6051
‘0’ Extension (Voicemail):	6005
IVR Recording Extension:	*95
Audio-In Extension:	*97
Paging Extension:	*96
Desktop (Phone) Paging Extension:	*94
Parking Extension:	700
Parking Rooms:	701-720
Parking Time:	45
Cell Phone Prefix:	7
BLF Pickup Prefix:	*7
Call Accounting:	not selected
PSIP External IP:	blank

Interactive Voice Response (IVR)

IVR Menu:	6000
Allow calling other extensions:	Enabled

Time Frames:(PBX -> Time Frames)

Open:	9am to 5pm
Christmas:	Dec. 25, all day
New Year's Day:	Jan. 1, all day

Language Setting
PBX -> Sound Manager -> Sound Manager menu
Language Setting: English

Incoming Calls
PBX -> Incoming Calls menu

Incoming Calls: **FXO calls routed to IVR**

Users

Sample IVR: **6000**
Operator: **6005**

Analog User: **6002**
Analog User: **6003**

User Template
PBX -> Users -> User Templates menu

PTS-Internal:

Name:	PTS-Internal
Remote Extension:	no
DTMF Mode:	rfc2833
Outgoing Call Rules:	PTS-default
Music on Hold:	DefaultMoH
Qualify:	yes
Enable BLF:	not enabled
Audio Codecs	
Codec 1:	uLaw
Codec 2:	
Codec 3:	
Permitted Networks	none
Permit All:	not enabled
Network:	/
IVR Menus	
Extension:	6000
Name:	Sample IVR
Allow dialing other extensions:	enabled

Group: Main

Step	Parameter
Ringing	
Wait	2
Answer	
Background	if-u-know-ext-dial.gsm
WaitExten	3
Background	if-u-know-ext-dial.gsm
WaitExten	3
Hangup	

Group: 0

Step	Parameter
GoToExtension	6005 “Operator”

Time Frames

PBX -> Incoming Call Features -> Time Frames menu

Open

Name:	Open
Mode:	Permanent
Start Time:	9:00
End Time:	17:00
Start Week Day:	mon
End Week Day:	fri
Start Month Day:	*
End Month Day:	
Start Month:	*
End Month:	

Christmas

Name:	Christmas
Mode:	Permanent
Start Time:	
End Time:	
Start Week Day:	*
End Week Day:	
Start Month Day:	25
End Month Day	25
Start Month:	dec
End Month:	dec

New Year's

Name:	NewYear
Mode:	Permanent
Start Time:	
End Time:	
Start Week Day:	*
End Week Day:	
Start Month Day:	1
End Month Day:	1
Start Month:	jan
End Month:	jan

Conference Bridges

PBX -> Incoming Call Features -> Conference Bridge menu

6080

Extension:	6080
Name:	Conference 1
PIN Code:	0806
Admin PIN Code:	12340806
Announce Callers:	Enabled
Music on Hold:	DefaultMoH

6081

Extension:	6081
Name:	Conference 2
PIN Code:	1806
Admin PIN Code:	12341806
Callers announced:	Enabled
Music on Hold:	DefaultMoH

Outward Call Rules

_9NXXXXXXXXX : **9 followed by 10 digits will pass**
10 digit number to available analog line

911 : **will pass 911 to available analog line**

Outgoing Call Rules
PBX -> Call Handling -> Outgoing Call Rules Menu

Rule: FXS Lines

Rule:	`Analog 0`		
Pattern:	`911`		
Description:	`blank`		
Inter-Office:	`not enabled`		
CallerID (NAME <NUMBER>):			
Group	Strip Digits	Prepend	Ring Time
Analog	0		40
If Failed:			

Rule: Analog 1

Pattern:	`_9NXXXXXXXXX`		
Description:	`blank`		
Inter-Office:	`not enabled`		
CallerID (NAME <NUMBER>):			
Group	**Strip Digits**	**Prepend**	**Ring Time**
Analog	1		40
If Failed:	*speak-to-the-operator.gsm*		

Rule: PTS-default

Rule:	`Analog 0`		
Pattern:	`911`		
Description:	`blank`		
Inter-Office:	`not enabled`		
CallerID (NAME <NUMBER>):			
Group	**Strip Digits**	**Prepend**	**Ring Time**
Analog	0		40
If Failed:			

Rule: Analog 1

Pattern:	`_9NXXXXXXXXX`		
Description:	`blank`		
Inter-Office:	`not enabled`		
CallerID (NAME <NUMBER>):			
Group	**Strip Digits**	**Prepend**	**Ring Time**
Analog	1		40
If Failed	*speak-to-the-operator.gsm*		

Incoming Call Rules

PBX -> Call Handling -> Incoming Call Rules menu

Rule: All -> 6000

DID:	`blank`
To:	`6000 "Sample IVR"`
CallerID Prefix:	`blank`
Answer:	`not enabled`

Music on Hold

music.gsm

IVR

speak-to-the-operator.gsm
T-to-rtrn-to-main-menu.gsm
if-u-know-ext-dial.gsm
T-to-leave-msg.gsm
pls-enter-vm-password.gsm
to-reach-operator.gsm
enter-password.gsm
please-try-again-later.gsm
thanks-for-calling-today.gsm
pls-hold-while-try.gsm
this-call-may-be-monitored-or-recorded.gsm
T-to-hear-cur-ancmnt.gsm
pls-enter-conf-password.gsm
T-to-rec-ancmnt.gsm
goodbye.gsm
thank-you-for-calling.gsm
this-is-the-voice-mail-system.gsm
loud-ring.gsm

Language: English

Music on Hold

DefaultMoH Source:	File
File Name:	music.gsm
DefaultAudioIn Source:	Audio-In

3G

System -> 3G menu

Enable:	not enabled
Voice and Data:	not enabled
Telephone Line Failover:	not enabled
Ethernet Failover:	not enabled
IP Address 1:	198.133.219.25
Port 1:	80
IP Address 2:	216.255.83.40
Port 2:	80
Provider:	Standard
DCE-DTE Options:	"ATQ0 V1 E1 S0=0 &C1 &D2 +FCLASS=0"
Packet Data Protocol:	AT+CGDCONT=1,"IP","internet.com","",0,0
Dial Number:	ATDT*99***1#
Probe Response Timeout:	5
Probe Interval:	10
Retry Count Till Down:	3
Retry Count Till Up:	3
Switching Delay:	60

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