G-Series User Guide

Publication Number 70-00007

Document Date: 04/05/2010

Print Date: 04/05/2010

Publication Information

G-Series User Guide
Part number: 70-00007

Publication date: March 5, 2010

Published By
Positron Telecommunication Systems Inc.
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International: 1-514-345-2220

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Chapter 1

General Information

About this Guide

This guide introduces you to the Positron Telecommunication Systems Inc. G-Series appliance its features and applications, and describes how to install it. This guide was designed to be read from beginning to end.

Related Documentation

As they are developed, other guides, white papers, FAQs and help files will be posted on our website at www.PositronTelecom.com.

Positron Telecommunication Systems Products and Services

Positron Inc. offers a line of sophisticated VoIP equipment for enterprise communication and collaboration through communication service providers. The company's products integrate VoIP and traditional telephony in stand-alone systems that combine ease of use with powerful functionality.

Positron's VoIP devices connect analog devices (telephone, fax and modem) to IP-Networks allowing customers to take advantage of converged voice and data services. The products support SIP Proxies, can integrate with Microsoft OCS through a combined Mediation server / PBX and provide visibility into PBX attributes through a detailed operator panel.

Full details and contact information are available at www.PositronTelecom.com

Compliance Information

FCC Part 15

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

Request Service Information

A plug and jack used to connect this equipment to the premises wiring and telephone network must comply with the applicable FCC Part 68 rules and requirements adopted by the ACTA. A compliant telephone cord and modular plug is provided with this product. It is designed to be connected to a compatible modular jack that is also compliant. For details, see installation instructions.

The Ringer Equivalent Number (REN) is used to determine the number of devices that may be connected to a telephone line. Excessive RENs on a telephone line may result in the devices not ringing in response to an incoming call. In most but not all areas, the sum of RENs should not exceed three (3.0). To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company. For products approved after July 23, 2001, the REN for this product is part of the product identifier that has the format US:AAAEQ##TXXXX. The digits represented by ## are the REN without a decimal point (for example, 03 is a REN of 0.3). For earlier products, the REN is separately shown on the label.

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

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

If trouble is experienced with the Positron Telecommunication Systems Inc. product, please contact the Positron Repair department at 1-800-661-4311 for repair or warranty information. If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.

Positron Telecommunication Systems Inc. located at 5101 Buchan street, Montreal in Canada hereby certifies that the Positron Telecommunications Systems Inc. bearing labeling identification numbers mentioned above complies with the Federal Communications Commission's ("FCC") Rules and Regulations 47 CFR Part 68, and the Administrative Council on Terminal Attachments (ACTA)-adopted technical criteria TIA-968-A-2, Telecommunications - Telephone Terminal Equipment -Technical Requirements for Connection of Terminal Equipment To the Telephone Network, January 2004.

Product Safety

This equipment is compliant with CSA CAN/CSA-C22.2 No. 60950-1-03

Service and Support

Positron Contact Information

General information:	Positron Telecommunication Systems Inc. 5101 Buchan Street, Suite 200 Montreal, Quebec, Canada H4P 2R9 US and Canada: 1-888-577-5254 International: 1-514-345-2220 Fax: 514-345-2271 E-mail: info@ PositronTelecom.com Website: www. PositronTelecom.com
Customer Service and Repairs:	US and Canada: 1-888-577-5254 International: 1-514-345-2220 E-mail: support@PositronTelecom.com

Technical Customer Support

Positron is committed to providing excellent ongoing technical support to its customers. A team of specialists is always available for telephone consultations or for on-site visits to assist in the maintenance and troubleshooting of Positron equipment.

For pricing information or assistance in the planning, configuration and implementation of the installation of equipment, contact Customer Service.

Warranty Repairs

All warranty repairs are performed at no cost. Positron reserves the right to repair or replace any equipment that has been found to be defective.

For information about out-of-warranty repairs, contact Positron's Repair Department. Due to the varied nature of repairs, no specific turnaround can be guaranteed, but average turnaround time is 20 working days from date of receipt. In emergency situations, special arrangements can be made. All repaired items are warranted for a period of 90 days.

Before returning any items to Positron for repair, warranty repair or replacement, call the Repair department to obtain a Return Material Authorization (RMA) number. Parts returned without RMA numbers cannot be accepted. The RMA number must always be clearly marked on all boxes, crates, and shipping documents. Bulk repairs (more than five items) will require additional processing time, so please take this into consideration when requesting an RMA number.

To accelerate the repair process, whenever possible, include a report detailing the reason for return with the unit(s). Also, please include the name and phone number of a person who can be contacted should our Repair department need further information.

When packing items being returned for repair, please ensure they are properly packed to avoid further damage. Positron Telecommunication Systems Inc. plug-in cards should never be shipped while installed in a shelf; this will cause damage that can extend the repair period.

Positron Telecommunication Systems Inc. Warranty

Subject to the provisions of this paragraph, Positron warrants that the equipment shall perform in accordance with Positron's specifications. The warranty remains valid for two (2) years from the date of shipment. The warranty fully covers workmanship, materials and labor. Positron shall, at its sole discretion, repair or replace the problem unit.

Freight costs to ship defective equipment to Positron are borne by the Customer, with return of replaced or repaired equipment to be at Positron's expense.

Limitation of Liability

Subject to anything to the contrary contained herein, Positron's sole obligation and liability and the customer's sole remedy for Positron's negligence, breach of warranty, breach of contract or for any other liability in any way connected with or arising out of, the equipment or any services performed by Positron shall be as follows:

- In all situations involving performance or non-performance of the equipment or any component thereof, the customer's sole remedy shall be, at Positron's option, the repair or replacement of the equipment or said component.
- For any other claim in any other way related to the subject matter of any order under, the customer shall be entitled to recover actual and direct damages; provided that Positron's liability for damages for any cause whatsoever, and regardless of the form of the action, whether in contract or in tort (including negligence), shall be limited to the value of the order.

Positron shall not be obligated to repair or replace any item of the equipment which has been repaired by others, abused or improperly handled, improperly stored, altered or used with third party material or equipment, which material, or equipment may be defective, of poor quality or incompatible with the equipment supplied by Positron, and Positron shall not be obligated to repair or replace any component of the equipment which has not been installed according to Positron specifications.

IN NO EVENT SHALL POSITRON BE LIABLE FOR ANY INDIRECT, INCIDENTAL, SPECIAL, CONSEQUENTIAL, PUNITIVE, EXEMPLARY OR SIMILAR OR ADDITIONAL DAMAGES INCURRED OR SUFFERED INCLUDING LOSS OF PROFITS, LOSS OF REVENUES, LOSS OF DATA, LOSS OF BUSINESS INFORMATION, LOSS OF GOODWILL, LOSS OF EXPECTED SAVINGS OR BUSINESS INTERRUPTION ARISING OUT OF OR IN CONNECTION WITH THE EQUIPMENT, A PURCHASE ORDER SUPPLIES, MAINTENANCE SERVICES OR OTHER SERVICES FURNISHED HEREUNDER, EVEN IF POSITRON HAS BEEN ADVISED OR IS AWARE OF THE POSSIBILITY OF SUCH DAMAGES.

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Chapter 2

Overview

The G-Series Family

Positron Telecom's the G-Series family of appliances provide standalone PBX and telephony ports creating a seamless gateway to the cellular, traditional telephone and VoIP worlds by combining them into a single integrated device.

The Positron Telecom G-Series products are an affordable, scalable solution for single point of contact communication needs, enabling customers to communicate through either VoIP or telephone lines, or a mix of the two, provides a centralized communication point and routes calls accordingly to desk, home or cellular phone. The G-Series also provides customized greetings per user, lower cost long distance and a true one inbox solution for email, fax and voicemail.

The G-Series appliances provide both telephony and PBX on the device itself. The G-Series are standalone systems that provide all the functionality integrated into the device itself including streaming music-on-hold port, overhead paging system port, integrated telephony ports, USB ports for external storage support, Ethernet switch, power failure port by-pass and much more.

The G-Series Overview

G-124 - Analog

- Sophisticated Integrated PBX functions in a single-board
- 4 Analog FXO Ports
- Support for up to 4 VoIP Lines
- 2 FXS Port for analog phone or fax machine
- Audio input jack for streaming music on hold
- Audio output jack from overhead paging systems
- 128 ms echo-canceller in hardware
- Four port Ethernet Switch
- On-board storage through a Compact Flash Interface or dual USB ports
- Local and Remote Web-based configuration

Figure 1 G-124

Indicators

- 7 LEDs on Front Plate
    - 4 – FXO – Orange
    - 1 – Globe
    - 1 – Tick
    - 1 Power
- Ethernet Connector on rear plate
    - LED 1 – 10/100 MBs Indictor
    - LED 2 - Activity

G-1212 - Analog

- Sophisticated Integrated PBX functions in a single-board
- 12 Analog FXO Ports
- Support for up to 4 VoIP Lines
- 2 FXS Port for analog phone or fax machine
- Audio input jack for streaming music on hold
- Audio output jack from overhead paging systems
- 128 ms echo-canceller in hardware
- Four port Ethernet Switch
- Expandable, on-board storage through a Compact Flash Interface or dual USB ports
- Local and Remote Web-based configuration

Figure 2 G-1212

Indicators

- 15 LEDs on Front Plate

  - 12 – FXO – Orange
  - 1 – Globe
  - 1 – Tick
  - 1 Power
- Ethernet Connector on rear plate
  - LED 1 – 10/100 MBs Indictor
  - LED 2 - Activity

G-224 - ISDN

- Sophisticated Integrated PBX functions in a single-board
- 4 ISDN Bri Ports (S/T)
- Support for up to 4 VoIP Lines
- 2 FXS Port for analog phone or fax machine
- Audio input jack for streaming music on hold
- Audio output jack from overhead paging systems
- 128 ms echo-canceller in hardware
- Four port Ethernet Switch
- Expandable, on-board storage through a Compact Flash Interface or dual USB ports
- Local and Remote Web-based configuration

Figure 3 G-224

Indicators

- 7 LEDs on Front Plate
  - 4 – ISDN – Orange
  - 1 – Globe
  - 1 – Tick
  - 1 Power
- Ethernet Connector on rear plate
  - LED 1 – 10/100 MBs Indictor
  - LED 2 - Activity

G-320– E1/T1 PRI

- Sophisticated Integrated PBX functions in a single-board
- Single E1/T1 PRI Ports
- Support for up to 4 VoIP Lines
- 2 FXS Port for analog phone or fax machine
- Audio input jack for streaming music on hold
- Audio output jack from overhead paging systems
- 128 ms echo-canceller in hardware
- Four port Ethernet Switch
- Expandable, on-board storage through a Compact Flash Interface or dual USB ports
- Local and Remote Web-based configuration

Figure 4 G-320 – E1T1 PRI

Indicators

- 7 LEDs on Front Plate
  - 4 – E1/T1 Status – Yellow / Red / Blue
  - 1 – Globe
  - 1 – Tick
  - 1 Power
- Ethernet Connector on rear plate
  - LED 1 – 10/100 MBs Indictor
  - LED 2 - Activity

Basic Features

In addition to the default PBX features like call switching, call completion, call connection, call termination and accounting, the following features can be enabled:

Call Routing Features

Automated Attendant

An automatic system to answer phones with the ability to build phone menu systems, add call menus, transfer to voice mail and create flexible and programmable rules to handle all of these features.

Call Menus

Flexible call management menus with user selectable options – a more advanced version of the traditional phone tree/menu systems. Support is available for multiple sets of menus and even change them based on time or on information gleaned from caller ID.

Managing Extensions

Features to help the phone system administrator, such as the ability to add new extensions, remove unneeded extensions, change extension locations and much more from a Web-based control panel.

Call Forwarding

Automatic, programmed or manual call forwarding to any number.

Call Transfer

The ability to transfer calls between extensions without going back to a central switchboard.

Call Parking

Put the caller on hold in a waiting area so that any other phone system user can pick the call up.

Messaging and Management Features

Voice Mail and Voice Mailboxes

An almost infinite number of voice mailboxes are available through the use of expandable CF card memory.

Call Hold:

System allows placing callers on hold with no drop off in queues with user selectable hold music and programmable options about handling hold time length.

Follow Me

Calls can be routed to other numbers, in the office or externally if not answered within a set time.

Meet Me Conference Calling:

System handles multiparty conference calls, internally and externally.

Web-Based Management and Administration:

Administrator can manage phone system directly from a Web browser.

Operation

Setup and configuration of the units is achieved through an integrated web-based interface.

The system allows the seamless integration of VoIP and analog telephones into the same PBX. Analog PSTN lines and VoIP lines or a mix of the two can be configured and later reconfigured as needed. Users can take advantage of the quality, availability and reliability of analog lines as well as the low long-distance rates and expandability of VoIP services. In the event of Internet connection failure, calls can be made through regular phone lines.

An integrated module within the G-Series products seamlessly detects and installs many types of SIP phones. These hardware or software SIP phones can be located locally, connected through a managed or unmanaged switch or remotely via an IP connection.

Many call-handling features can be configured locally or through service providers.

The system can accept, store and convert voicemails to WAV or MP3 file email attachments. Users can retrieve their voicemails on a computer or mobile phone, creating a true one-inbox messaging solution for emails and voicemails.

G-Series Features:

Auto SIP phone provisioning for these brands

- Linksys
- Polycom
- Snom
- Aastra
- Cisco

Voicemail to email conversion

Fax to Email conversion using TIFF format

Fax pass-though to FXS port

Support for these Voice Codecs:

- G.711 (ulaw & alaw)

Trunk Support

- SIP
- IAX
- FXO,
- ISDN,
- E1
- T1

Time of Day service

Find me / Follow me

Conference rooms

Music on Hold configurable per user

Corporate or Home Directory (Auto Attendant
Positron's auto-attendant allows callers to dial into a main number then dial a feature code or an extension. It can be used in combination with Direct Inward Dial to allow, for example, providing a directory to allow callers to look up a name and be transferred to the corresponding extension. Its features include:

- greetings
- extended greetings

- music-on-hold
- voice message forwarding
- message appending

The PBX plays music or prerecorded messages to customers on hold. Music can be sorted into various folders. Separate auto-attendant feature sets can be used for different situations. The voicemail tree supports directories by department, employee, extension, etc, offering flexibility and giving small organizations a more professional telephone appearance.

Dial by Name

Inbound callers can route their calls to the appropriate person without knowing their extension. This allows for either first or last name directory look up. Assuming voicemail is set up correctly, Dial by Name allows an outside caller to get help in finding the extension number of the person they wish to call as long as they know the person's name

Dial by Extension

Inbound callers can route their calls to the appropriate person if they know the correct extension number

Dial by Group

Inbound callers can route their calls directly to the auto attendant of a group or department

Configuration and Maintenance through Local or Remote Web Interface

Status Display of All Connections

Line Status

PBX Features:

- Call Hold
- Call Waiting
- Call Transfer - Attended and Blind
- Call Conferencing
- Call Forwarding - Unconditional, No Answer, On Busy
- Call Log (60 entries each): Made, Answered, Missed Calls
- Multiple Ring Tones with Selectable Default Ring Tone per Line
- Call Duration with Call Time Stamp Stored in Call Logs
- Syslog, Debug, Report Generation and Event

Caller ID

Represents the digits passed from the carrier (or PABX) to the end user device (or between PABXs) that identify who the caller is. Also known as CLI (Calling line identification) or ANI (Automatic Number Identification)

Corporate Call Back

Allows you to set up a callback destination that calls a user back and provides access to an application. An example of this would be a caller that dials your system, disconnects, is called back and then provided a DISA application to make a phone call. This is a basic service for reducing costs international calls and mobile phone charges

Advanced Call forwarding rules

Example: Unanswered inbound calls - the caller is prompted to speak recipient's name. The call then gets forwarded over a VoIP line to the additional forwarding numbers provided in the recipient's forwarding profile. The call is forwarded with the CallerID of the inbound caller (not the PBX). The recipient will see the inbound call and can answer or ignore. Ignored calls get sent back to the recipient's voicemail box. If the recipient answers the call, they are prompted with the recording and may accept or reject the call. A rejected call is still transparent to the original caller, and sent back to voicemail.

Configurable extension lengths (2,3,4 digits)

Desktop paging

Certain desktop phones with built in speakers can have the 'paging' function enabled which will automatically answer a paged call and play the audio without end user intervention

Outbound and Inbound configurable call rules

- Example outbound rule**:**
  Member of 'sales team' can dial long distance numbers while 'support' cannot
- Example inbound rule**:**
  Number 'xxx-xxx-xxxx' is a fax machine and routed directly to the FXS connected fax machine

External Media Support (voicemail, Music on Hold)

- Compact Flash

Chapter 3

Hardware Installation

Hardware Installation

This chapter of the G-Series User Guide covers:

- Setup of G-Series
- Verifying operations of the hardware
- Connecting to the Internet
- Rebooting the G-Series
- Testing the Phones
- Verifying the Dial Sequence
- Verifying Extensions

G-Series Setup

We recommend that the G-Series appliances be connected to a surge protector or UPS (uninterruptible power supply). This will help minimize damage in the event of power fluctuations or power surges.

Installation Equipment

To complete the installation and configuration of the G-Series the following equipment is suggested:

- A computer, referred to as “notebook” for configuration. Alternatively, any computer capable of running a browser on the local area network can be used. If such a computer is not available, a standard Ethernet cable can be used to perform the configuration on the host computer itself.
- The notebook browser must have Java capabilities to run the Web Interface.

Grounding and Handling

Before removing the product from its packaging, ensure that you are grounded. To ground yourself it is recommended to use an Anti-Static wrist band, or at least, ensure that you touch some grounded object before handling the product.

Inspection

Inspect the product for any signs of physical damage. Report any damages directly to the shipper. Keep all packaging material in the event that the unit has to be shipped for servicing.

Inventory

The complete package contains:

- G-Series appliance
- Telephone line pigtails
    - G-124 and G-224 should contain two
    - G-1212 will contain four
- Ethernet cable
- Documentation
- CD-ROM containing driver and sample files

Environment

The selected installation site should provide a stable operating environment, clean and free from temperature and humidity extremes, shock, and vibration. The operating temperature should be kept below 100 degrees F (38°C).

It is highly recommended that the product be located in or near the equipment cabinet and in proximity to the customer’s network equipment.

PSTN – G-124, G-1212

The appliance has been designed to be connected to the PSTN and should not be connected to any other type of telecommunications service or services. Doing so will void the warranty and could cause network and / or equipment damage.

ISDN – G-224

The appliance has been designed to be connected to the ISDN as an S/T and should not be connected to any other type of telecommunications service or services. Doing so will void the warranty and could cause network and / or equipment damage.

E1/T1 – G-320

The appliance has been designed to be connected to the E1/T1 network and should not be connected to any other type of telecommunications service or services. Doing so will void the warranty and could cause network and / or equipment damage.

Install Compact Flash Storage or USB before powering on the appliance

Before setting up the device, determine whether CF (Compact Flash) or USB storage is required. Used for storing voicemail messages, memory should be installed at initial setup. Installing memory at a later date will result in the loss of voicemail messages stored up to that point.

- NOTE: Adding or removing CFmemory must be done with the power off, and requires a reboot following modification.
- NOTE: We recommend use of the Sandisk CF cards for complete system compatibility. Other CF cards may be compatible. Generally, if an

incompatible CF card has been installed, the appliance will not successfully boot.

To install Compact Flash memory,

- Ensure that power to the system is off
- Insert the memory into the memory holder
- Power on the system

G-Series Installation

To setup the G-Series

- The G-Series does not have DHCP enabled by default, the static IP address will be 192.168.1.1
- Connect an Ethernet cable directly to the any of the ports on the Ethernet switch
- Ensure the computer you are using has an IP address in the same range as the G-Series (192.168.1.X)
- Open a Web browser to the IP address: 192.168.1.1
- **G-124, G-1212 Only:** If using more than two analog line ports, install the FXO line pigtails. Each pigtail handles two lines. One pigtail supports lines 1 and 3, the other, lines 2 and 4 and so on. Labels on the pigtails should be observed. Connect the pigtails to the appropriate FXO line appearances. The outermost RJ-11 jack on the backplane plate is used for Lines 1 and 3, the RJ-11 jack next to it accommodates lines 2 and 4.
- **G-224 Only:** If using more than two ISDN line ports, install the line pigtails. Each pigtail handles two lines. One pigtail supports lines 1 and 3, the other, lines 2 and 4. Labels on the pigtails should be observed. Connect the pigtails to the appropriate ISDN line appearances. The outermost RJ-45 jack on the backplane plate is used for Lines 1 and 3, the RJ-45 jack next to it accommodates lines 2 and 4.

NOTE: If only two analog lines or 2 ISDN line are used, they can be connected directly to the two ports without the use of the pigtails.

- Replace the power cord and turn on the system
- Observe the red LED on the appliance..

To verify the basic operation of the G-Series hardware:

- Connect an analog phone to the FXS port.
- Lift the receiver and verify that you receive a dial tone.
- Dial 6001 to hear a voice prompt

G-124 Technical Specifications

Table 1: Electrical Specifications

Parameter	**Specification**
Input voltage requirement:	
Power consumption:	
REN	3 @ 100 ft

Table 2: Physical Specifications

Parameter	**Specification**
Operating temperature range:	0℃ to 40℃
Storage temperatures:	-20℃ to 85℃
Humidity:	-20℃ to 85℃
Power consumption:	
Height:	107 mm (4.2 inches)
Depth:	168 mm (6.6 inches)
Weight:	

Chapter 4

Configuring the G-Series

Information Required for Initial Configuration

- **To complete the configuration of the card, you will require the following information:**
  - The network address of the G-Series product as assigned by your network administrator
  - The address of a Time Server on your network (if present)
  - The DNS addresses used on your network
  - The quantity and telephone numbers of analog telephone lines to be used for incoming and outgoing calls and their physical location
  - The location and telephone number of a dedicated fax line (if present)
  - The configuration information from your VOIP SIP provider, typically:
    - SIP account name
    - SIP account addresses
    - Password
    - Codec type
  - To set up telephone extensions, the names, extension numbers and department groupings of employees
  - List of extensions that will form ring groups (ringing all extensions in a department handling common incoming calls)
  - Filenames and locations of prerecorded audio files for menus and music on hold

- **To connect to the integrated web-based interface:**
  - Connect a notebook computer to the Ethernet port on the G-Series using a standard Ethernet cable.
  - Set the notebook's Static IP address to any address other than 192.168.1.2 . (For example 192.168.1.10)

  NOTE: The interface requires a full Java implementation running on the browser..

  - Use the notebook's Firefox internet browser to access `http://192.168.1.2`.
  - The Home page of the G-Series Web Interface appears.

- **If a password screen is presented:**
  - In the Login field, type `admin`

- In the Password field, type `password`

Initial Configuration Steps

The following steps should be followed in the sequence listed below as information from a previous step will affect menus and options available in subsequent steps. In the event that some information has been missed or must be changed, full editing capabilities are provided by the G-Series interface.

Table 3: Configuration Steps

Step and Menu Selection	Information Needed
1. Configure analog and VOIP Trunks **PBX -> Trunks / Lines**	Number, type and telephone numbers of analog lines (optional) Fax machine info Account information for VoIP lines
2. Configure Dial plans **PBX -> Dial Plans**	Routing for outgoing calls - which will be routed to analog lines, which to VoIP, etc.
3. Configure rules related to dial plans **PBX -> Dial Plans -> Rules**	Plan for outgoing calls, which will be routed to analog lines, which to VoIP, etc.
4. Load music on hold and/or IVR files **PBX -> Sound Manager -> Music on Hold, IVR**	Uploading .wav sound files to be used for music on hold (MoH), IVR menus.
5. Configure Music on Hold **PBX -> Music on Hold**	Selection of audio files to use
6. Verify and Edit user templates **PBX -> User Templates**	Creating groups with similar outgoing call permissions.
7. Configure Extensions **Users -> Extensions**	List of users, extension numbers to assign, email addresses for voicemails. Policy for assigning voicemail passwords. Attaching user templates to extensions
8. Configure time frames **PBX -> Time Frames**	Determination of outgoing call rules varying by time of day, weekdays, holidays, etc.
9. Configure IVR	Design of custom IVR

PBX -> IVR Menus	programs.
10. Configure conferencing **Users -> Conference Bridge**	Configuration of conference “rooms”.
11. Configure ring groups **Users -> Ring Groups**	Manage ring groups. A ring group is a group of extensions that can be set to ring at the same time or in sequence.
12. Configure incoming call handling **PBX -> Incoming Calls**	Plan and policy for routing of incoming calls on VoIP lines and analog lines. Determination of how fax calls are to be routed.
13. Configure PBX general settings **PBX -> PBX Settings**	Determination of general PBX configuration settings.

Problem Solving

The product has been designed to aid you in diagnosing and solving possible problems. These problems are rarely serious, usually incorrect configuration or a disconnected or damaged cable. If this section does not solve your problem, contact your supplier for information.

➤ **Perform these fault isolation actions first:**

- Ensure that any associated network equipment is powered on
- Check the following:
- Ensure the green Power LED, is ON steadily. The green Power LED glows steadily when the card's internal software has been loaded and verified. This signifies that the card is ready to handle calls.
- If no lights are visible after 1 minute:
    - If the fault LED glows red AND a new CF card was installed in the card immediately before, then most likely the CF card is incompatible and should be removed and replaced.

➤ **To determine whether the CF card is compatible with the G-Series:**

- Remove the CF card from the appliance
- Power up the G-Series
- Confirm that the green LED glows steadily. If the green LED glows steadily, the fault most likely lies with the CF card.

Chapter 5

The Web Interface

Web Interface Home Screen

The Positron Telecom GUI (graphical user interface) provides the means to access different Configuration Panel features.

- NOTE: Only the browsers supporting the full Java implementation are supported, A list of recommended browsers is available in the Appendix. If a recommended browser is not used, an error message may result, and it may not display some features or appear to hang.

Figure 5 Web Interface Tab Bar

Along the top of the web interface screen is a tab bar offering section menus and a language pull-down menu which presents language choices which include: English, French, Spanish and Portuguese.

- The language pull-down will change only the on-screen text and will not affect language sound files used by the system.
- NOTE: Any configuration changes made in the user interface WILL NOT be active until the **System - > Create Configuration** menu item is clicked.

System Defaults:

- NOTE: Consult Appendix 2 for factory default settings for lines, extensions, voicemails, IVRs etc.

Login

The Login menu will be presented when first logging in.

Figure 6 Web Interface Tab Bar

The default username for the system is: **`admin`**

The default password is: **`password`**

- NOTE: It is recommended that the password be changed as it is necessary to secure the web interface.
- NOTE: In the event that the login information is forgotten or misplaced, then a system reset must be performed and the System Restore to the last saved Backup must be performed.

Status

The Status screen presented after login or by clicking the Status tab provides system information.

Figure 7 Status Screen

The Status screen information includes:

VoIP lines/trunks. (Configured through **PBX -> Trunks / Lines**)

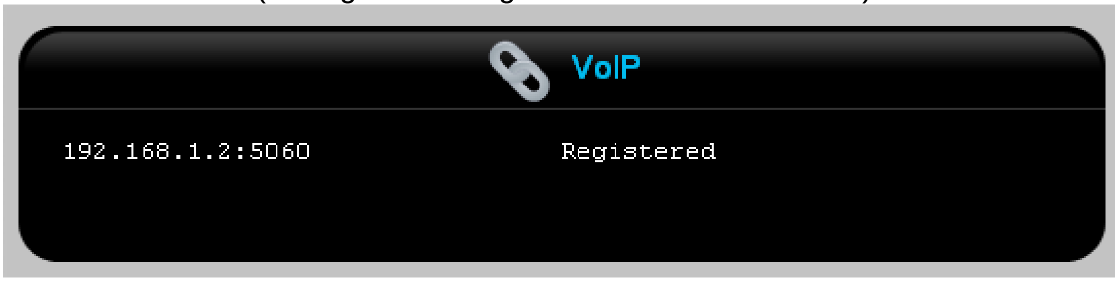


Figure 8 Status Screen - VoIP Lines/Trunks

- **Peers** (extensions and trunks). (Configured through **Users -> Extensions)**

Shows hardware and software phone information in Asterisk format along with IP address and status.

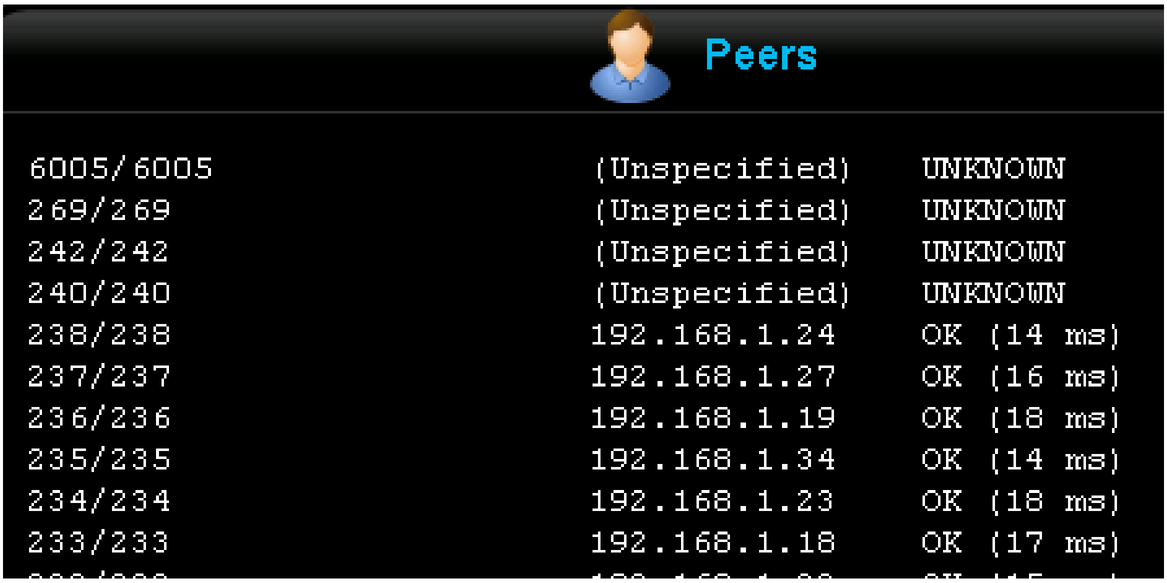


Figure 9 Status Screen - Peers

- **Network** configuration IP addresses information (Configured through **System -> Network)**

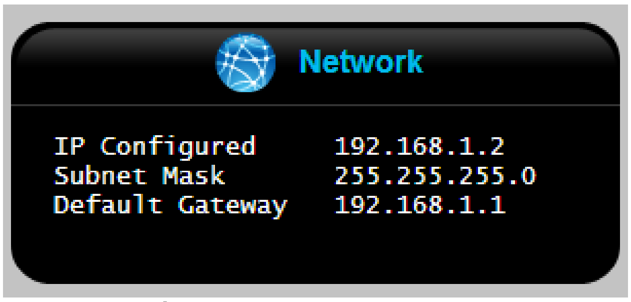


Figure 10 Status Screen - Network

- **Lines** (analog). Configured through (**PBX -> Trunks / Lines).**

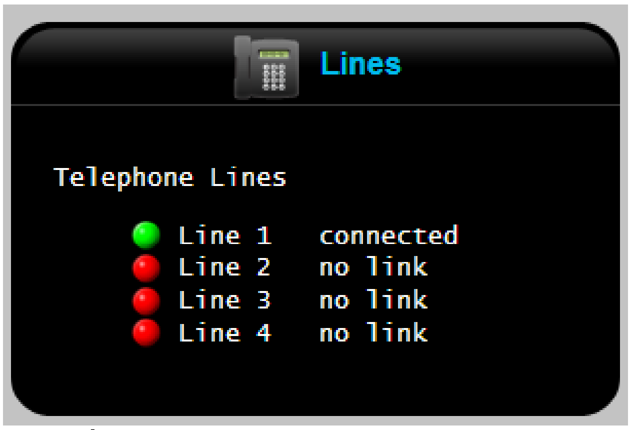


Figure 11 Status Screen - Lines

- **System** Firmware versions, uptime, fixed on-board and virtual disk (flash memory) use

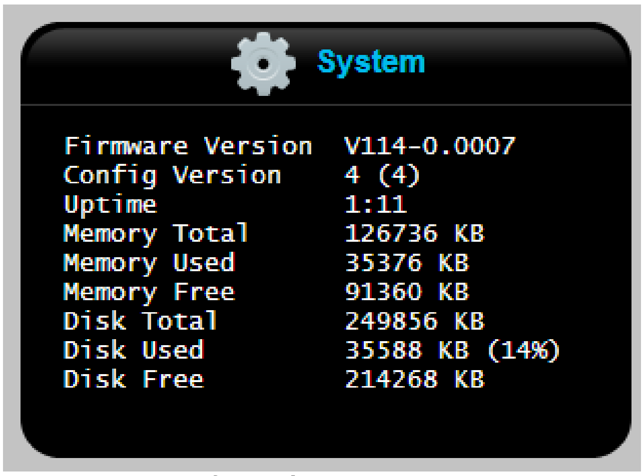


Figure 12 Status Screen – System Information

- **Storage** capacity and use of removable memory

G-series units have CF and USB memory. CF memory cannot be inserted or removed while the systems are running.

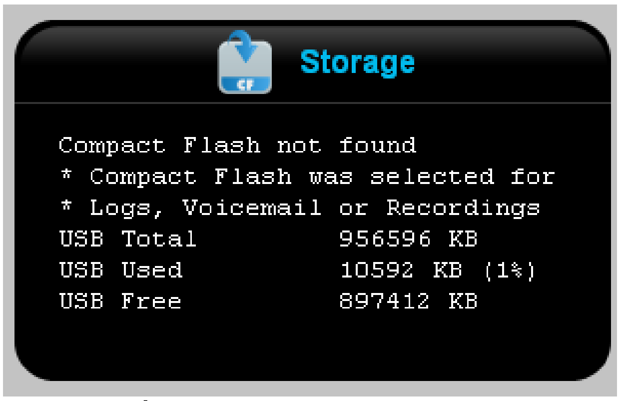


Figure 13 Status Screen – G-series Storage

Shown is the status of a system with no formatted CF memory.
The warning message indicates that the system was configured to expect CF memory for storage of logs, voicemail and/or recordings and either no CF is present, or the CF storage has not been formatted. The G-series can be configured to use USB instead of or in addition to CF storage.

Diagnostics Tab

The **Diagnostics Tab** provides two menu choices – Diagnostics and Logs

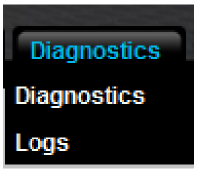


Figure 14 Diagnostics Tab

Diagnostics

The Diagnostics Menu can be used to verify network connections. It offers an input box where ping information can be entered to verify connection with the IP network and to determine whether a particular server is active.

➤ **To verify that the card is connected to a network:**

- Enter the IP address or URL of a destination server that is known to be active
- Click the Go button.

The results will show whether the server has been found, whether it is active and the length of time in milliseconds required to perform a round-trip to it.

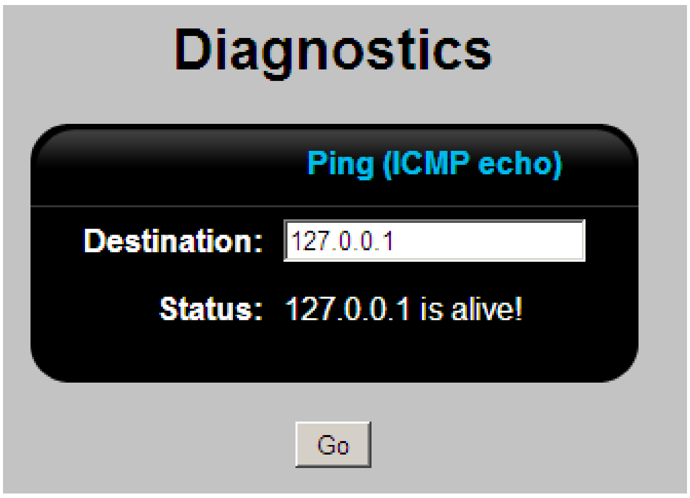


Figure 15 Diagnostics Screen

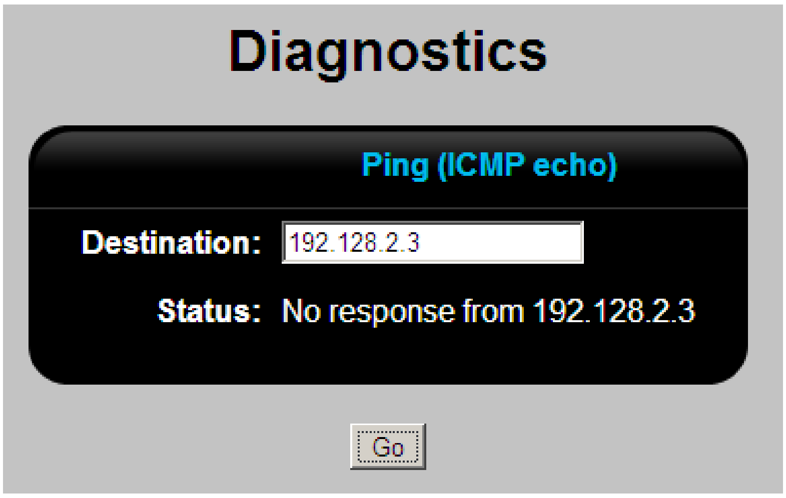


Figure 16 Diagnostics Screen

Logs

Displays:

The Logs Screen displays information generally used in system debugging. A drop-down menu provides:

- Syslog Screen
- Asterisk Messages
- GUI Log
- Web Server logs.

Buttons:

- To update the displayed logs, click **Refresh**
- To receive the files in tar.gz format, click **Download**

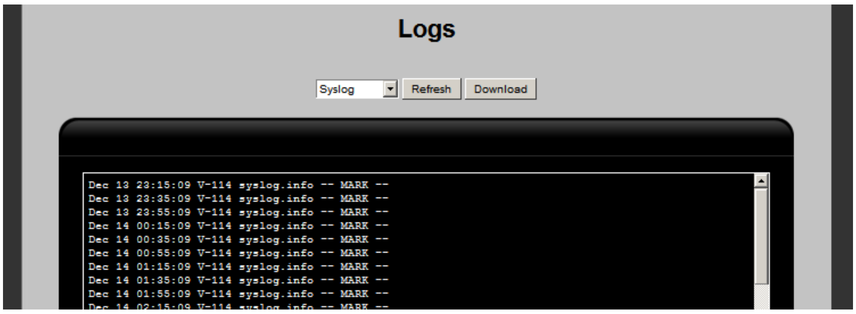


Figure 17 Call Detail Records Syslog Screen

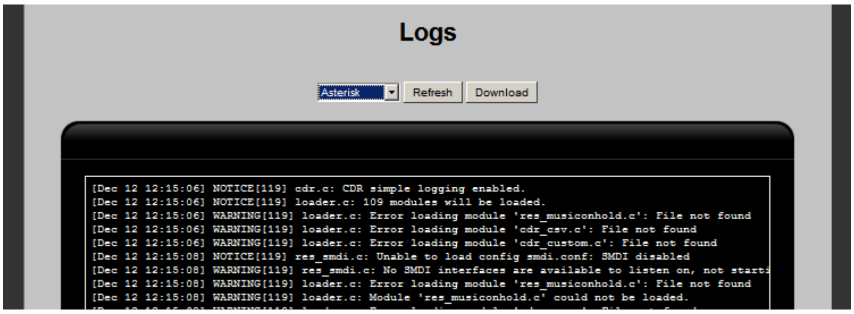


Figure 18 Asterisk Messages Log Screen

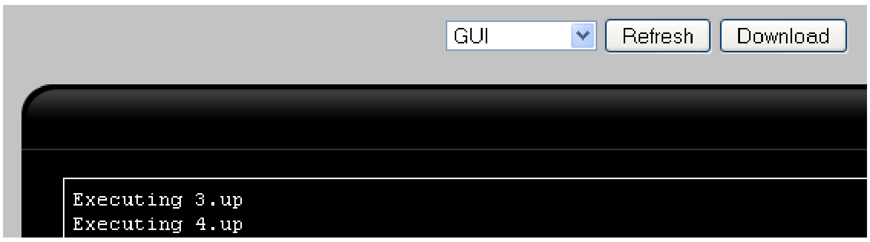


Figure 19 GUI Messages Log Screen

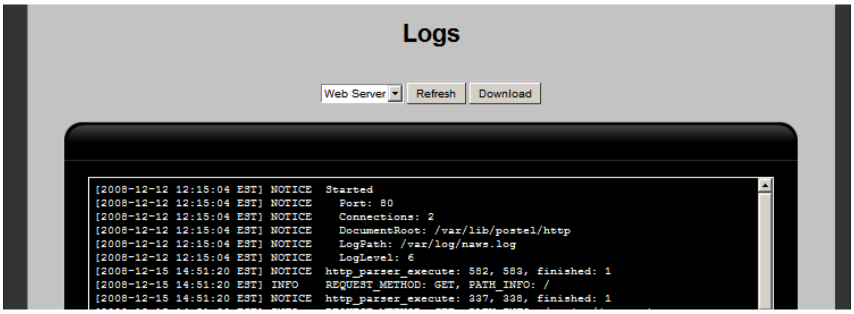


Figure 20 Web Server Log Screen

System Tab

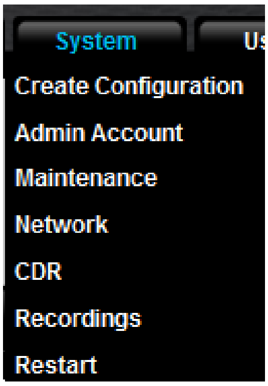


Figure 21 System Tab

The **System Tab** provides access to:

- **Create Configuration** to update system configuration files
- **Admin Account** to set system password
- **Maintenance** to set date and time information, and provides tabs for: firmware upgrading, resetting to defaults settings, backup/restore functions and formatting CF memory.
- **Network** to configure time zones, system servers and IP addresses
- **CDR** to display Call Detail Records
- **Recordings** to list and control of playing and deleting phone call recordings
- **Restart** to reset the system

Create Configuration

The **Configuration Control** updates system configuration files following changes done in the Web Interface.

- Note: Until the control is clicked, changes made to configurations will not be applied.

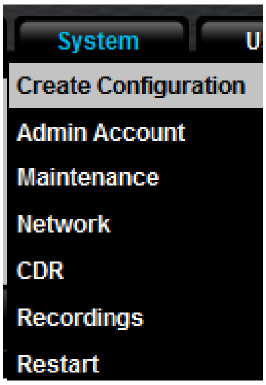


Figure 22 Create Configuration Control

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

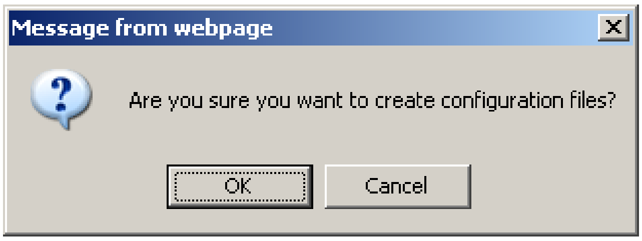


Figure 23 Create Configuration confirmation dialog

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

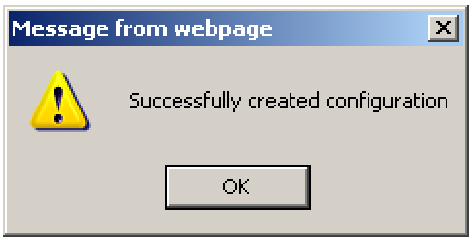


Figure 24 Create Configuration confirmation dialog

Admin Account

This menu allows for the setting and changes to the username and password which can be used to secure the web interface.

The default username and password for the system is listed on Page 44.

- NOTE: In the event that the login information is forgotten or misplaced, then a system reset must be performed and the System Restore to the last saved Backup must be performed.

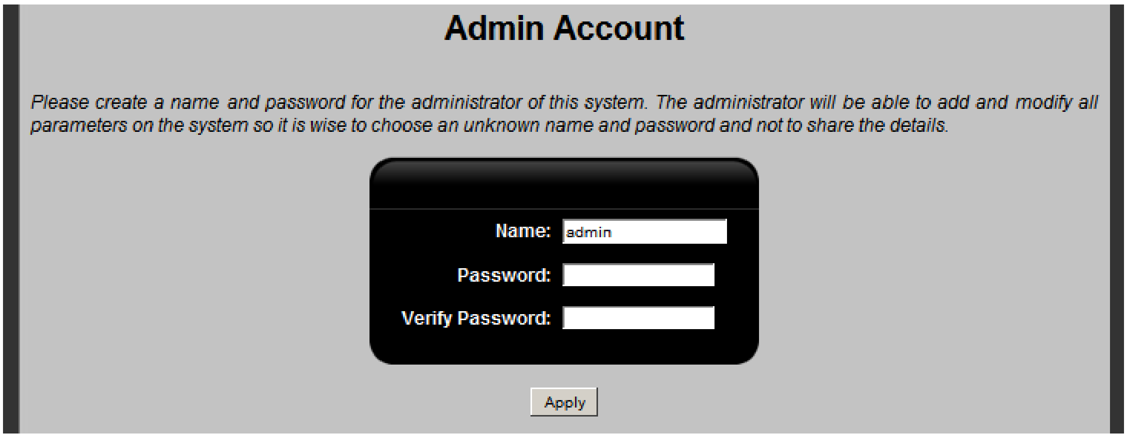


Figure 25 Admin Account Menu Screen

Maintenance

The Maintenance Menu comprises:

- **Date and Time**
- **Firmware Upgrade**
- **Reset to Defaults**
- **Backups**
- **Storage**

Date and Time

- **This menu is to be used only where it is not possible to connect to an NTP time server.**
- The timezone and NTP are set in **System -> Network**.

The G-series has battery backed-up clocks, and as a result, when the system has been reset, the system time will be reset automatically. Time information is important as it provides the timestamps for recordings, system operation and controls the operation of Time-of-Day dialing rules.

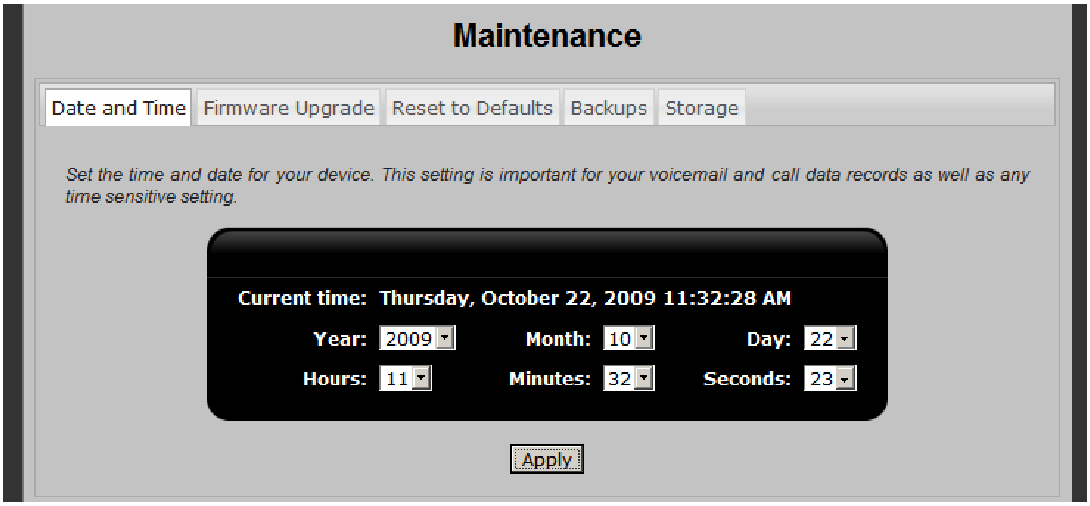


Figure 26 Maintenance Menu Date and Time Tab Screen

➘ To set system time equal to host time:

- Note: In normal operation, the time server specified in the **System->Network** screen will provide the system time.
  - Click **Apply** to set the time on the equal to the time of the host computer.
- The timezone and NTP (time server) are selected in **System -> Network**.

Firmware Upgrade

- When system firmware upgrades are available, they can be loaded into system memory using the Firmware Upgrade Tab.

Obtaining Firmware Upgrades

To obtain a firmware upgrade file using an Internet-connected browser:

Firmware upgrades are available through the Positron Telecom website at `www.PositronTelecom.com/xxx`

Where (depending on product) `xxx` is:

- `G124`
- `G1212`
- `G320`
- `G224`

Screens shown in the following examples are for a V-series system, similar information applies for other products.

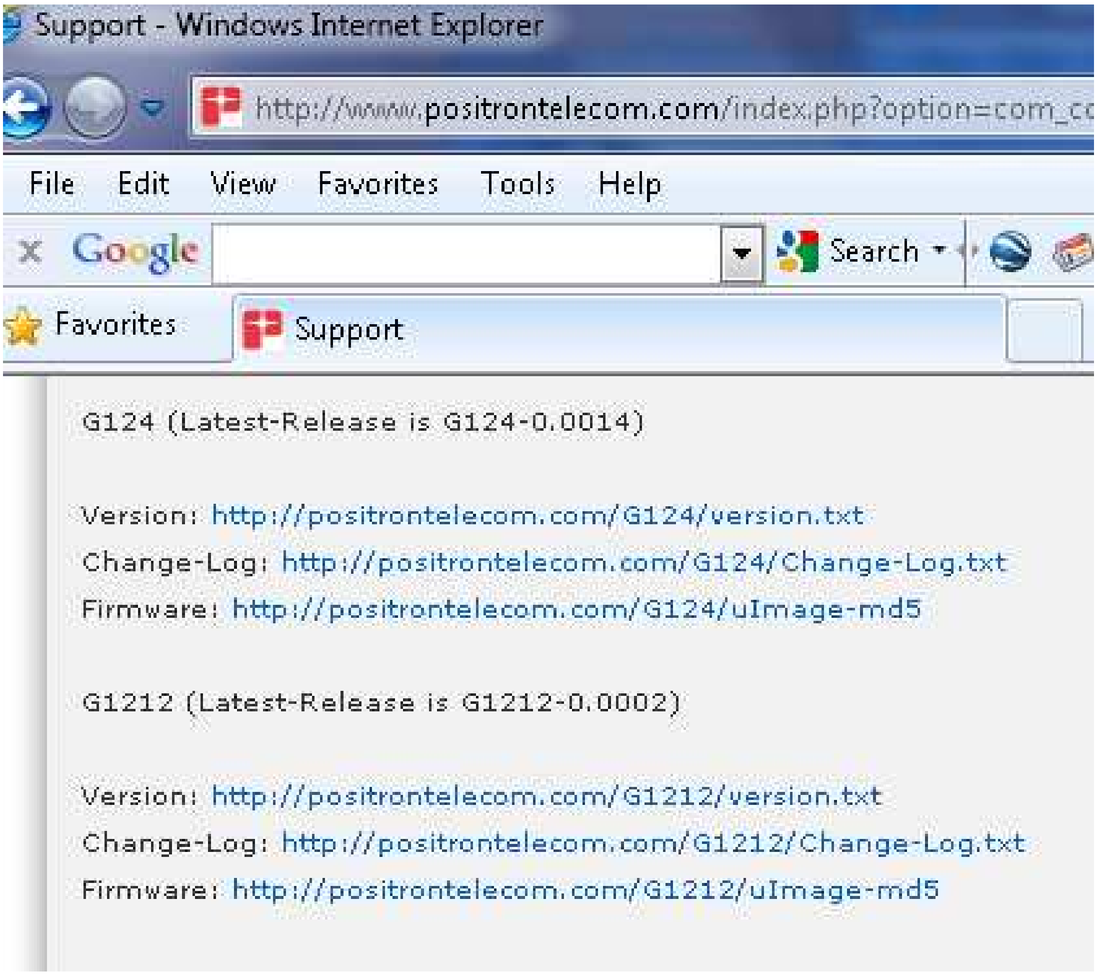


Figure 27 Positron Telecom Upgrade Page

- Locate and read the Version.txt file to confirm

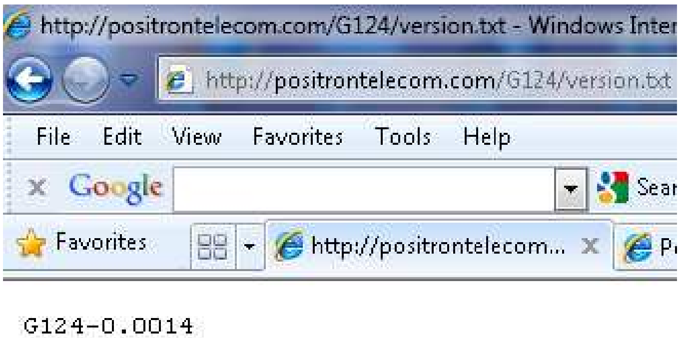


Figure 28 Positron Telecom version.txt Page

- Locate and read the change-log.txt page for the upgrade to verify its applicability to your situation.

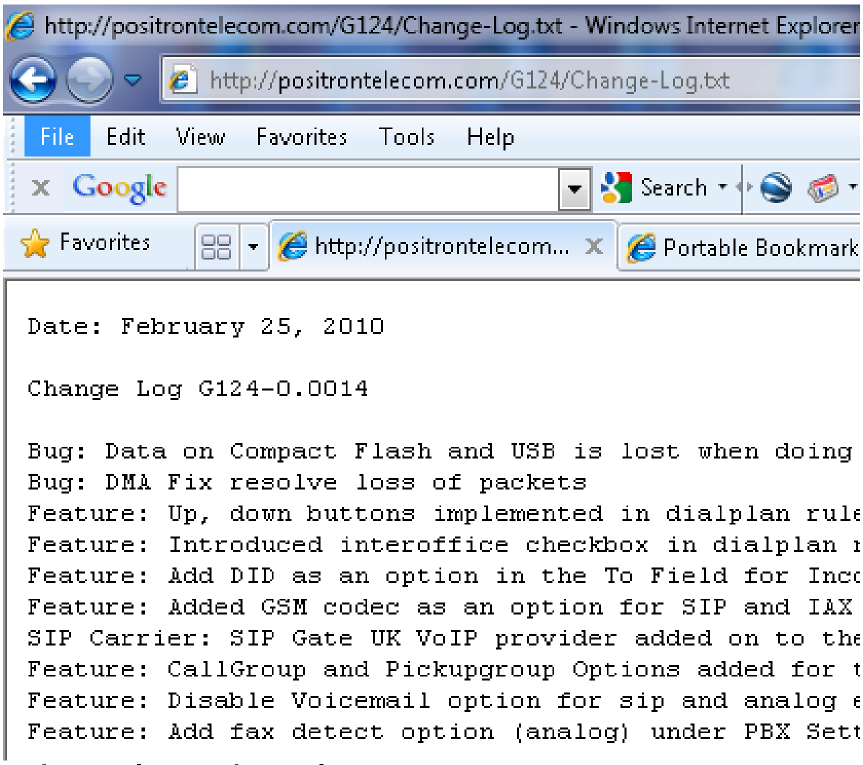


Figure 29 Positron Telecom change-log.txt Page at URL : http://positrontelecom.com/G124/Change-Log.txt

- Right-click the appropriate link and do a Save Target As…

Figure 30 Positron Telecom menu Page

- Locate the directory for your download on your local file system.
- Give it the filename exactly as shown on the web page. In this example use:

`"uImage-md5"`
(Include the quotation marks)

➘ To upgrade the firmware using the Positron Telecom Web Interface

- Ensure that the network is connected to the Internet
- Click HTTP and enter the URL for the firmware. Example: `http://positrontelecom.com/G124/uImage-md5`

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

- Click **Upgrade**

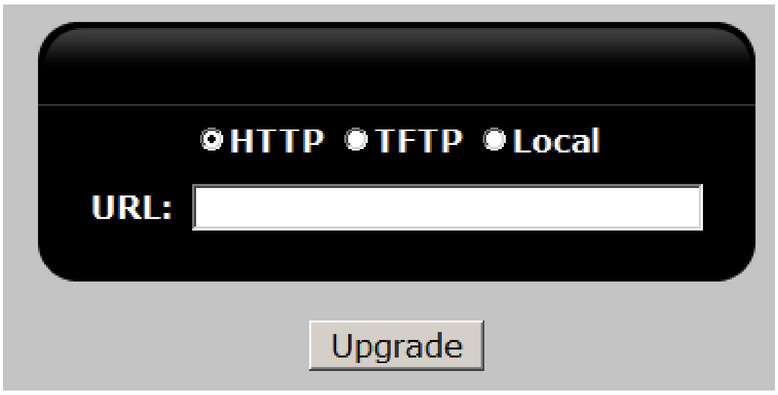


Figure 31 Maintenance Menu Firmware Upgrade Tab HTTP Screen

- A confirmation dialog box will be shown allowing cancellation of the upgrade.

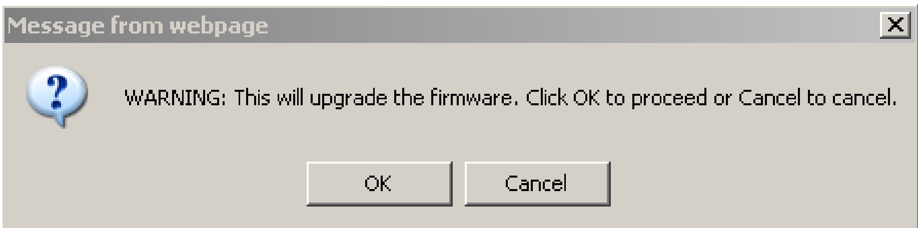
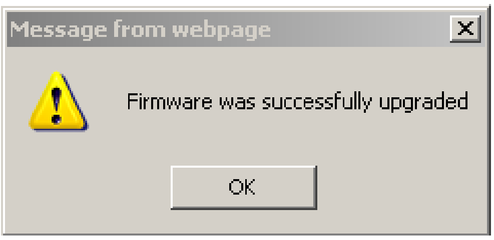


Figure 32 Maintenance Menu Firmware Upgrade Confirmation

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



or

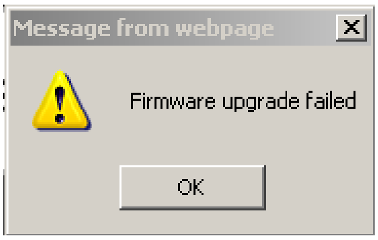


Figure 33 Maintenance Menu Firmware Upgrade Completion

To upgrade the firmware using TFTP:

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

- Click **TFTP**, enter **the IP address** and **filename** of the upgrade file
- Click **Upgrade**
- The dialog boxes above will be shown allowing cancellation and showing whether the operation completed successfully.

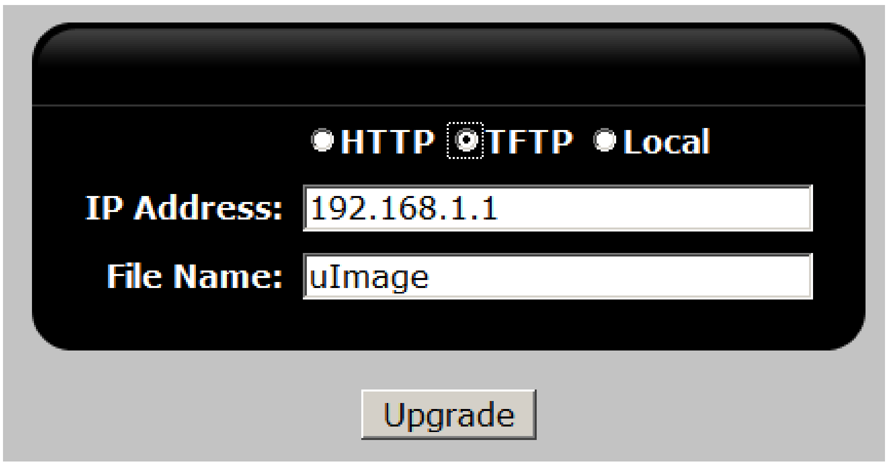


Figure 34 Maintenance Menu Firmware Upgrade Tab TFTP Screen

➘ To upgrade the firmware using a file on the local file system:

- Click **Local**
- Click **Browse…** and locate the file on your local file system

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

- Click **Upload**
- Click **Upgrade**
- The dialog boxes above will be shown allowing cancellation and showing whether the operation completed successfully.

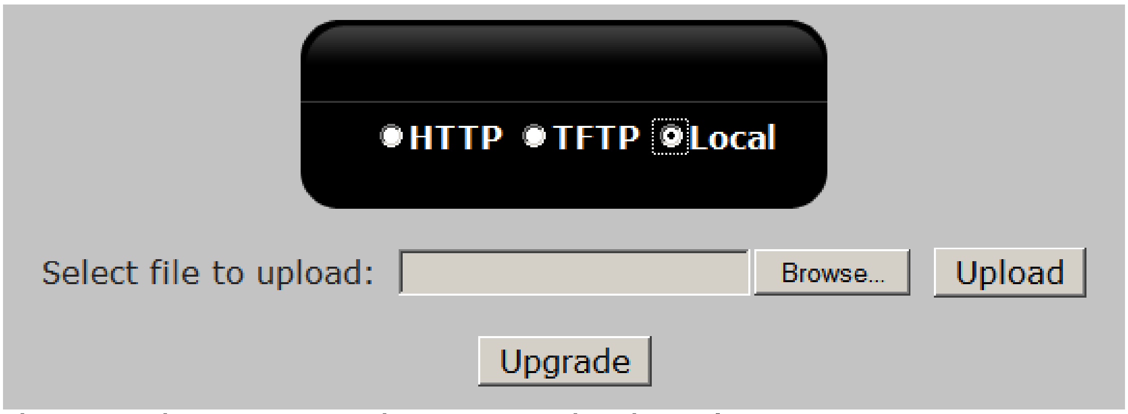


Figure 35 Maintenance Menu Firmware Upgrade Tab Local Screen

Reset to Defaults

Clicking this tab resets the system to its default mode using the last applied firmware.

- This operation will disconnect all current calls and should not be done during operating hours.
- It is highly recommended that a backup is taken of current configuration files as they must be restored following the restart.
- The Web Interface will not be available until the system has been reset. Its IP address will be reset to 192.168.1.2 and the password will return to its default. A configuration file saved before the reset will have to be restored in Maintenance -> Backups

To reset the system

- Following a configuration backup, choose **System -> Restart.**
- A confirmation dialog box will allow the reset to be cancelled.

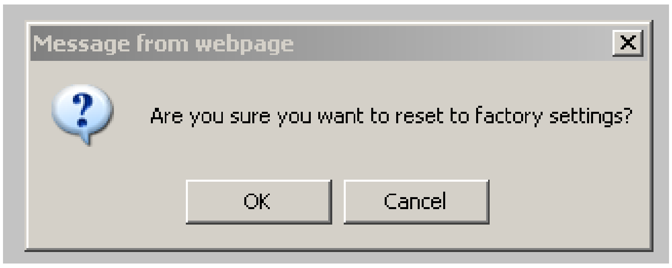


Figure 36 Reset to Factory Settings confirmation dialog

- Click **OK** to restart the system.
- A message will be displayed indicating a 100-second wait. The system will be reset when the unit's green LED lights continuously.
- Enter the web interface default address of `192.168.1.2` in the browser as well as the default system password as indicated in the Login section on page 43

Backups

Used to perform configuration file backup and restore functions. Backups are stored on the card's onboard memory. They can be downloaded as .tar.gz files onto the host's filesystem.

Before performing a Firmware Upgrade or Reset to Defaults it is highly recommended that the configuration file is backed up for restoration after the system is restarted.

Backups can be performed during normal system operation however Firmware Upgrades and Resets will terminate all calls.

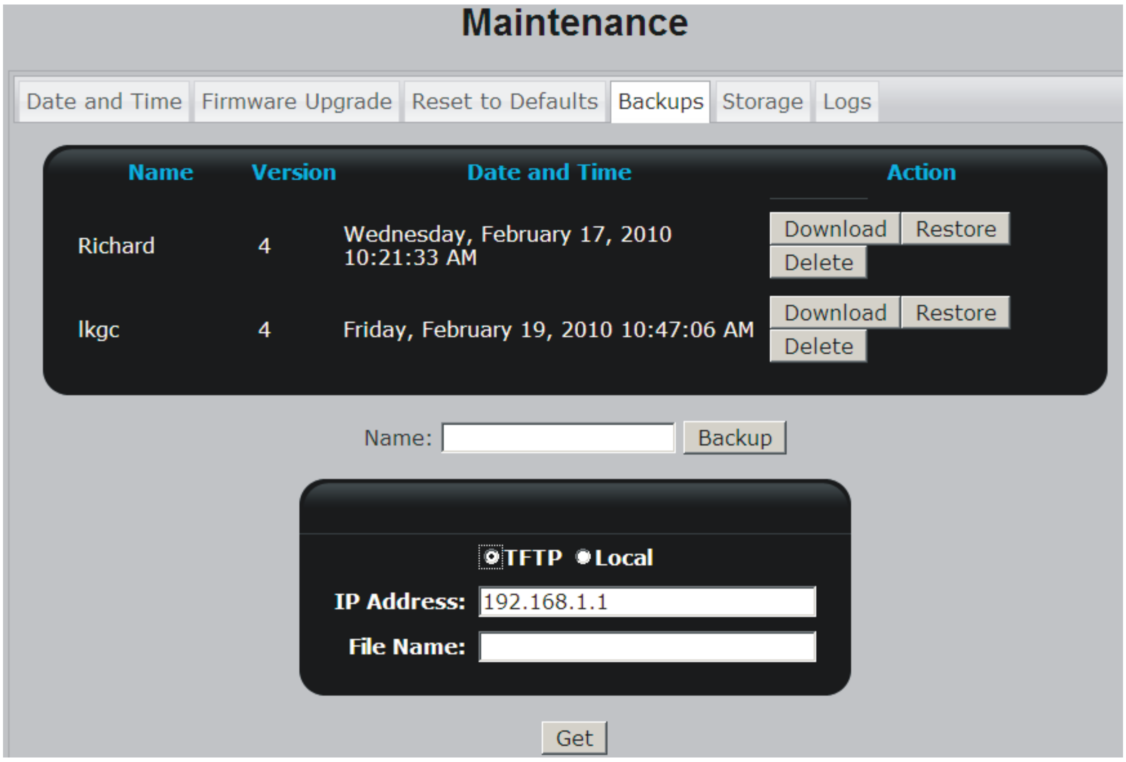


Figure 37 Maintenance Menu TFTP Section

To set the TFTP server:

- Click **System / Network**
- Enter the correct value in the **TFTP Server** box.
- Click **Get**

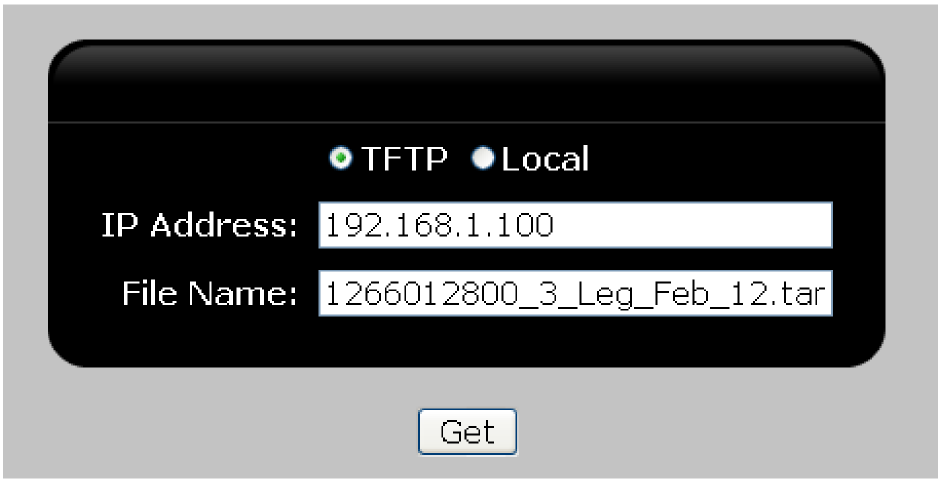


Figure 38 Maintenance Menu Local Section

To backup configuration files:

- 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 unit's onboard memory.
- A confirmation dialog box will be presented if the backup is successful.

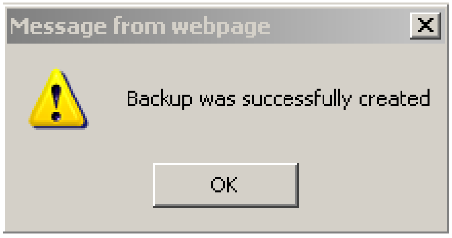


Figure 39 Maintenance Menu Confirmation Dialog box

- 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 Card:

- 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.

The restore operation can be performed while the system is running.

To restore configuration files located on the TFTP server:

- Click the TFTP radio button
- Enter the IP address of the TFTP server and the name of the restore file in the File Name box
- Click **Get**.

To restore configuration files located on the local file system:

- Click the Local radio button
- Locate the file on the local file system using the **Browse** button
- Click **Upload**.

Storage

Used to format the onboard Compact Flash storage used for voicemail and general storage. This control is useful when the Status screen shows that the Compact Flash memory is installed but unrecognized.

- NOTE: Estimated time for a format is 1.5 minutes/GB
- NOTE: A Reboot (**System -> Restart**) must be done in order to recognize the newly-formatted memory.

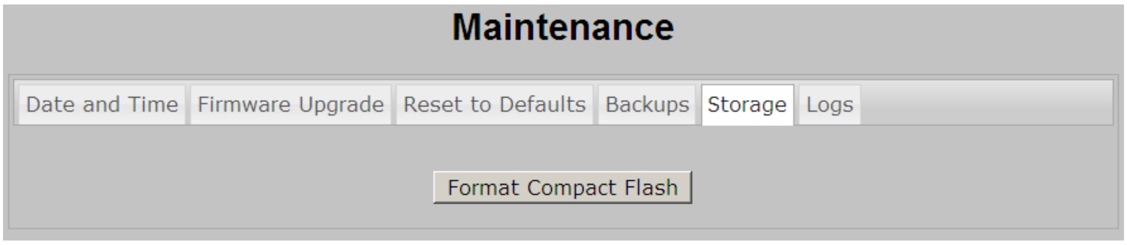


Figure 40 Maintenance Menu Storage Tab Screen

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

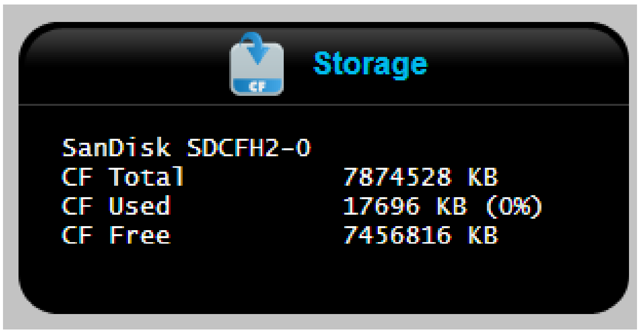


Figure 41 Status Screen – CF Storage

Logs

This menu allows the specification of which logs will be created and their location on storage devices, if available.

- It is highly recommended to establish these settings during system setup to ensure that the associated information is captured when the PBX goes online.
- Data will not be captured until the checkboxes have been selected and **System -> Create Configuration** has been performed.

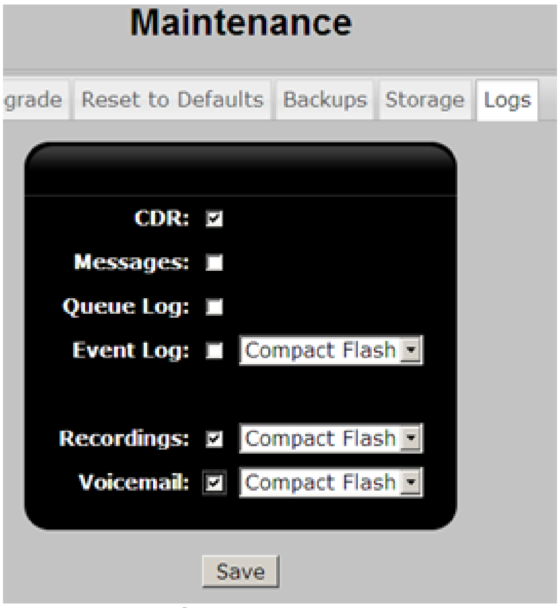


Figure 42 Maintenance Menu Logs Tab Screen

The choices available are:

- CDR (Call Detail Records): - Ensures that CDRs are stored.
- Messages: - Asterisk messages will be stored and made available under **Diagnostics -> Logs**
- Queue Log: - Asterisk queue logs will be stored and made available under **Diagnostics -> Logs**
- Event Log: - Asterisk event logs will be stored and made available under **Diagnostics -> Logs**
- Recordings: - If recordings are enabled on the system in **PBX Settings -> Record Calls** they will be stored in the associated location.

- Voicemail : - Messages will be stored in the associated location.

A drop-down menu is presented listing available storage locations..

- CF Storage is not hot-swappable, system power must be removed to add or remove CF Storage.
- USB storage can be added or removed while the system is active.
- To safely remove or replace USB devices for devices which support them (G-series), without interrupting production systems, consult the latest videos on the www.PositronTelecom.com website.

Network

Allows for the review and editing of networking parameters.

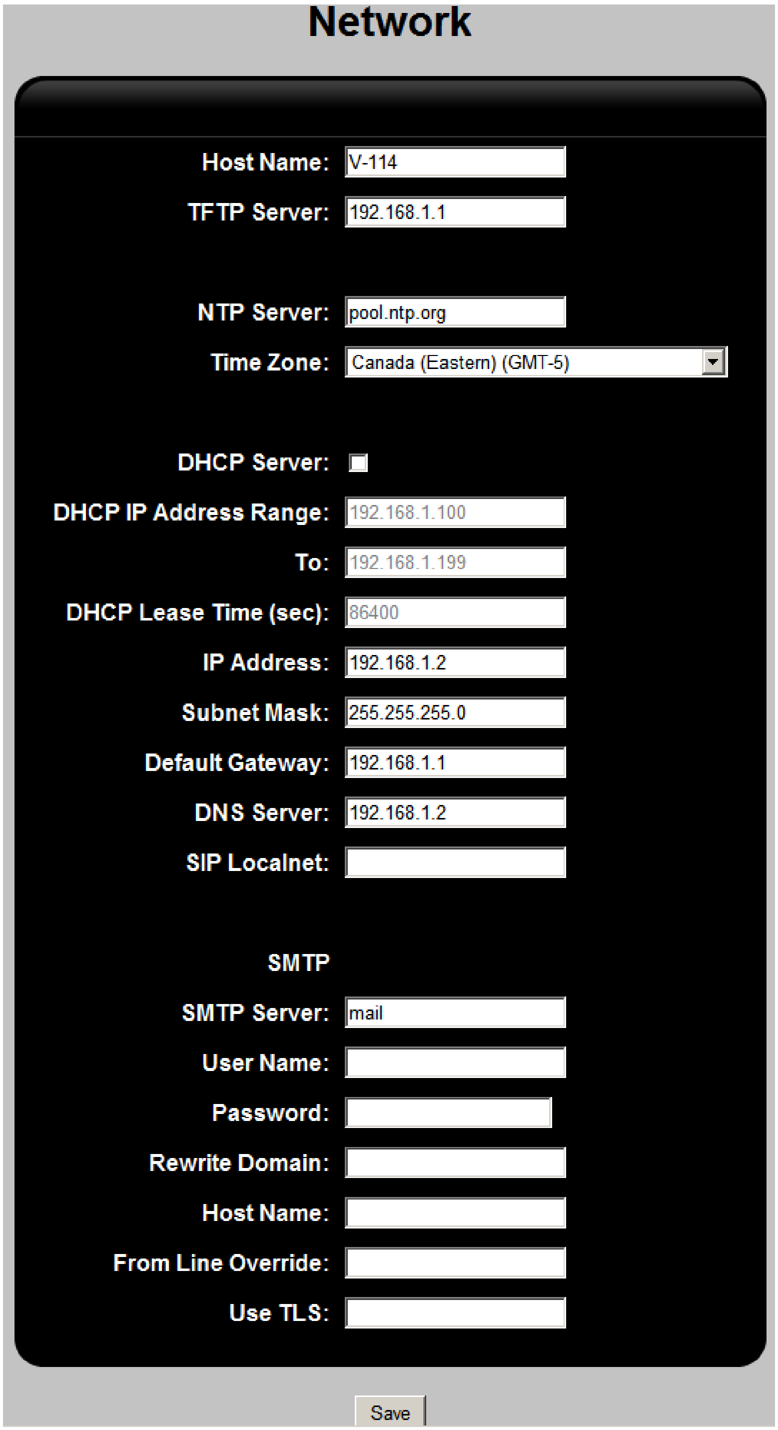


Figure 43 Network Menu Screen

Host Name:

Sets and displays the name of the host of the system and Web interface.

TFTP Server:

Sets and displays the IP address of the TFTP (Trivial File Transfer Protocol) server which typically holds firmware upgrades, sound files and system backups.

NTP Server:

Sets and displays the name of the time server. If there is a local NTP server, the address will be supplied by the local area network administrator.

Time Zone:

Sets and displays the current time-zone.

Figure 44 Network Menu NTP and Time Zone Section

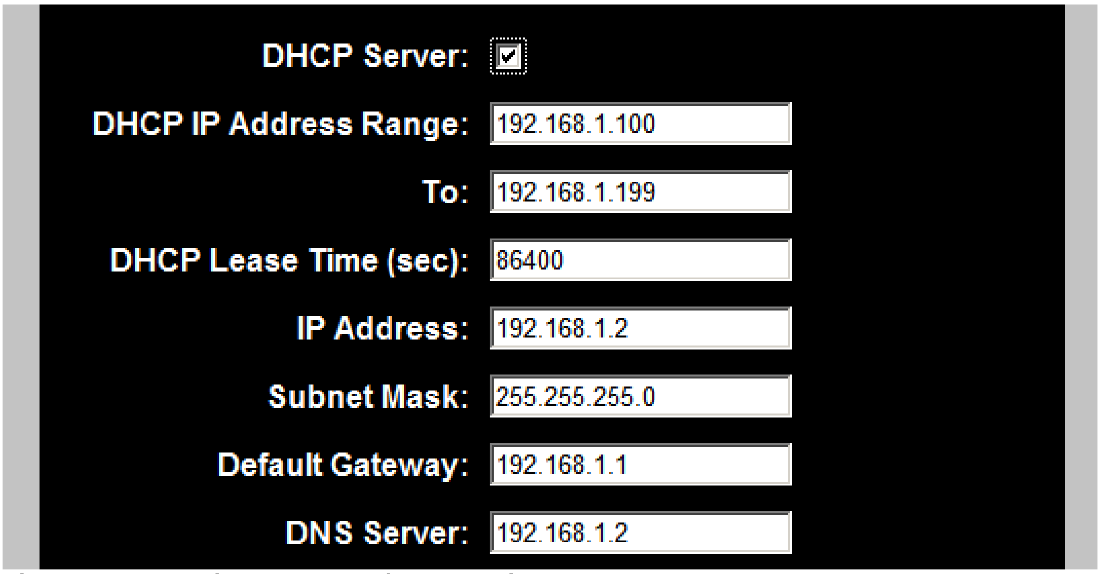


Figure 45 Network Menu DHCP/DNS Section

DHCP Server:

Controls whether the card’s address is set by the DHCP server. Normally this is NOT recommended.

DHCP IP Address Range (From and To):

If DHCP is enabled on a dedicated LAN used only for the telephone sets, this determines the address range of the devices which will be allocated by the Positron unit.

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. The default time set is 24 hrs. 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.

IP Address:

Sets and displays the IP address of the card and its web interface.

Subnet Mask:

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

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.

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.

SIP Localnet:

This setting is used in offices which use VPN (Virtual Private Networks) which will establish multiple networks where phones are registered. More details and assistance in setting this are available from Positron Telecom.

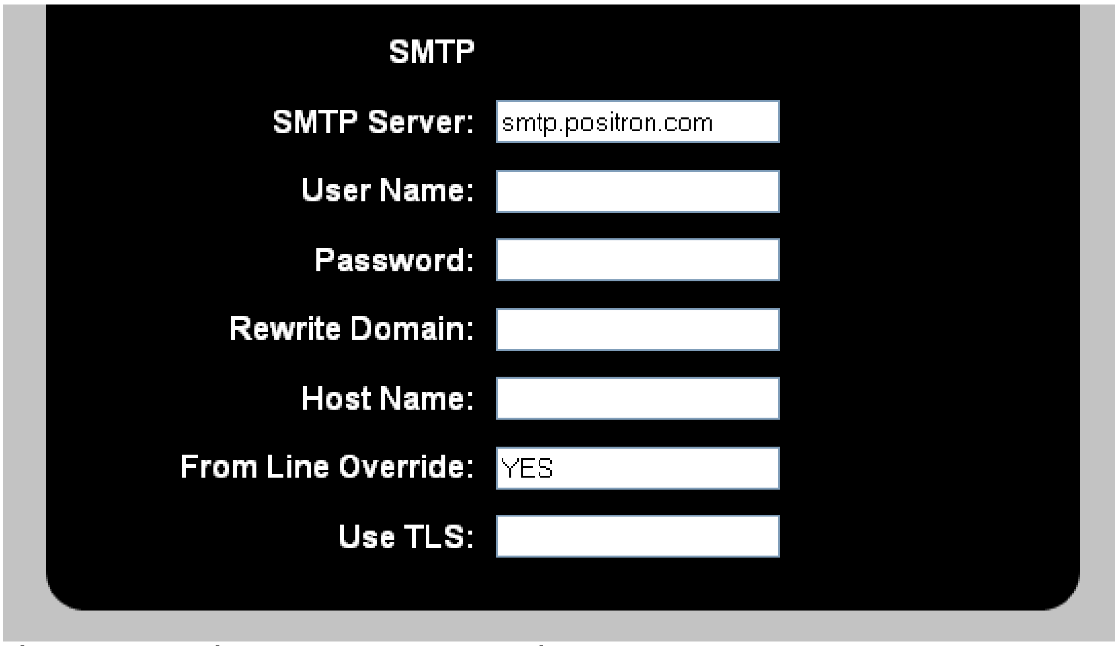


Figure 46 Network Menu Screen SMTP Section

SMTP Server:

Sets and displays the address of the network's mail system.
This information is used to allow the sending of voicemail attachments to emails.

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.

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 setting can be obtained from the network administrator.

CDR

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

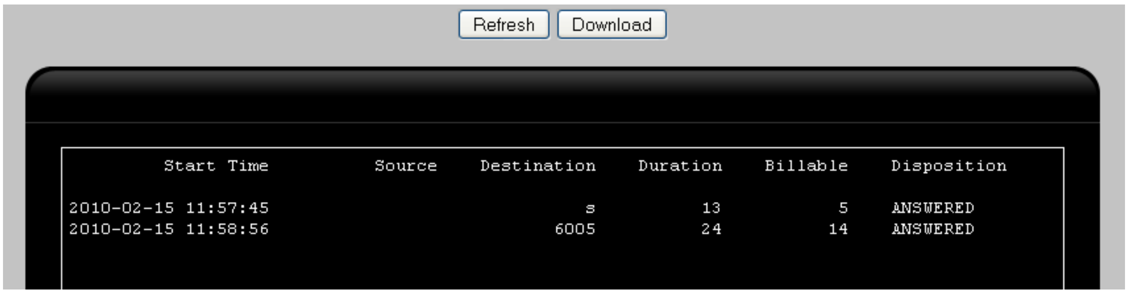


Figure 47 Call Detail Records

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

Action to perform on the recording.

Recordings

Provides a list of recordings made by the system and controls to allow individual and all recordings to be deleted. Typically recordings are individually downloaded then deleted.

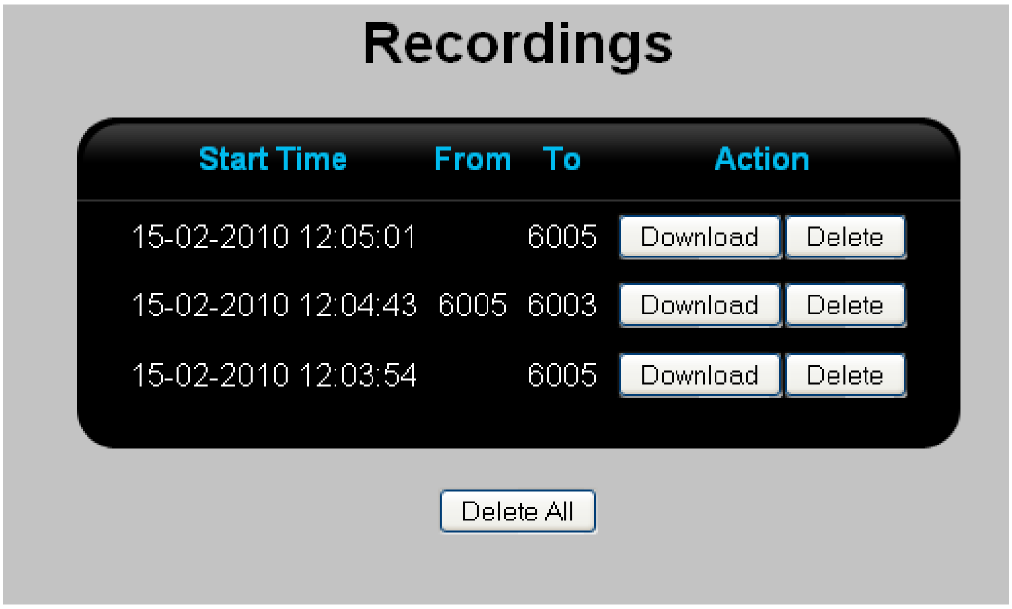


Figure 48 Recordings menu

Restart

Restarts the system using the existing configuration files.

NOTE: All calls in progress will be dropped during Restart.

To perform a system reset:

- Click Cancel to return without resetting or Click OK to begin reset.

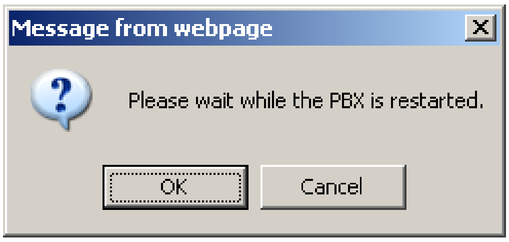


Figure 49 Reset Dialog Box

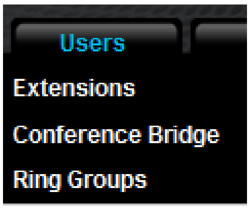
- A 60-second countdown timer will be displayed after which the web interface should be available again. The card's green Ready LED will glow steadily when reset is complete.
- The system's Login screen will be presented, the IP address and Password will be the ones stored in the current system configuration.

NOTE: Resetting the system will clear all logs stored in on-board memory including Call Detail Records (CDR).

NOTE: Resetting the system may return the system clock to its default start time in units without battery backup clocks. If the network cannot call the NTP time server, the system time should be set using **System - > Maintenance.**

It is important to the system's recordkeeping and general operation that the time be set correctly.

Users



The **Users Tab** permits access to the:

- **Extensions** menu, for the creation and initial configuration of all extensions, including hardware and software sets and virtual extensions for voicemail boxes, IVR menus, conference bridges and ring groups
- **Conference Bridge** configuration,
- **Ring Groups** configuration.

Extensions

Displays a list of extensions, allows for the editing (configuration) and deletion of extensions.

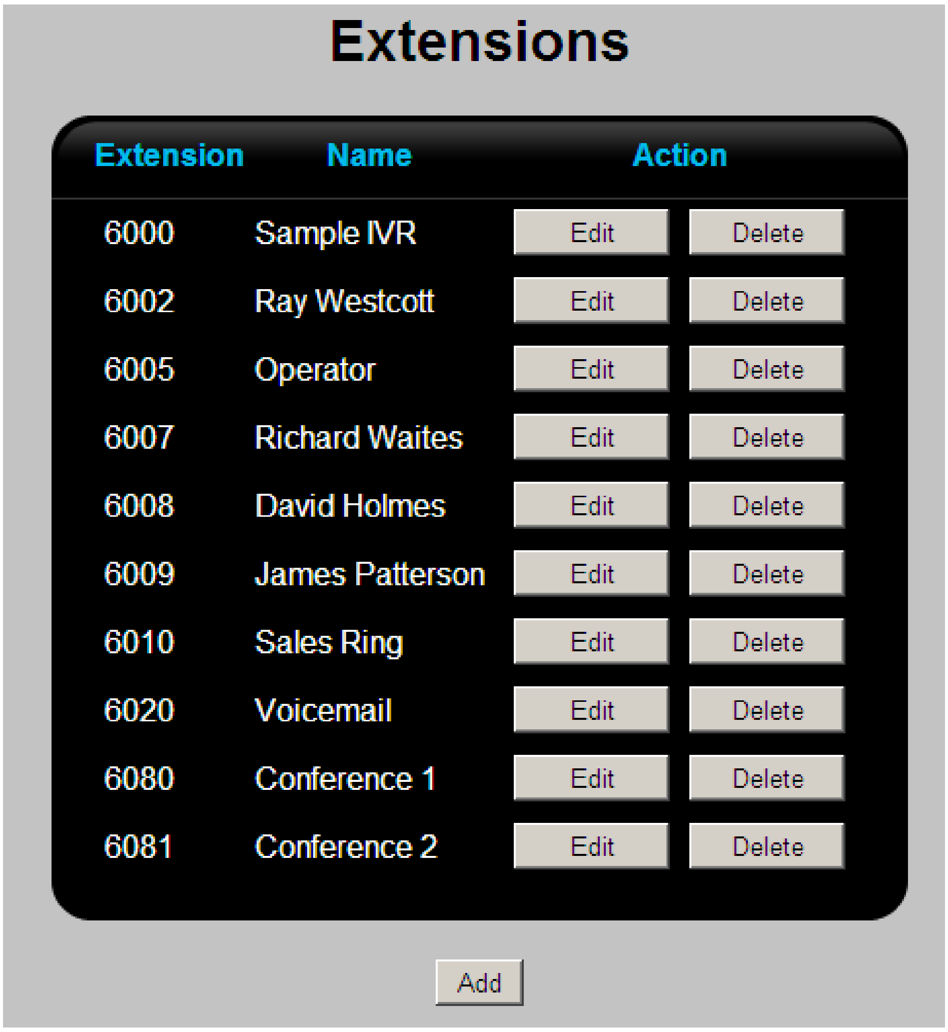


Figure 50 Extension Menu Screen

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

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

Extensions can be:

- Analog telephone sets

- SIP telephone sets
- SIP software phone
- Virtual Extensions:
    - IVR systems
    - Voicemail boxes
    - Conference bridges
    - Ring groups.

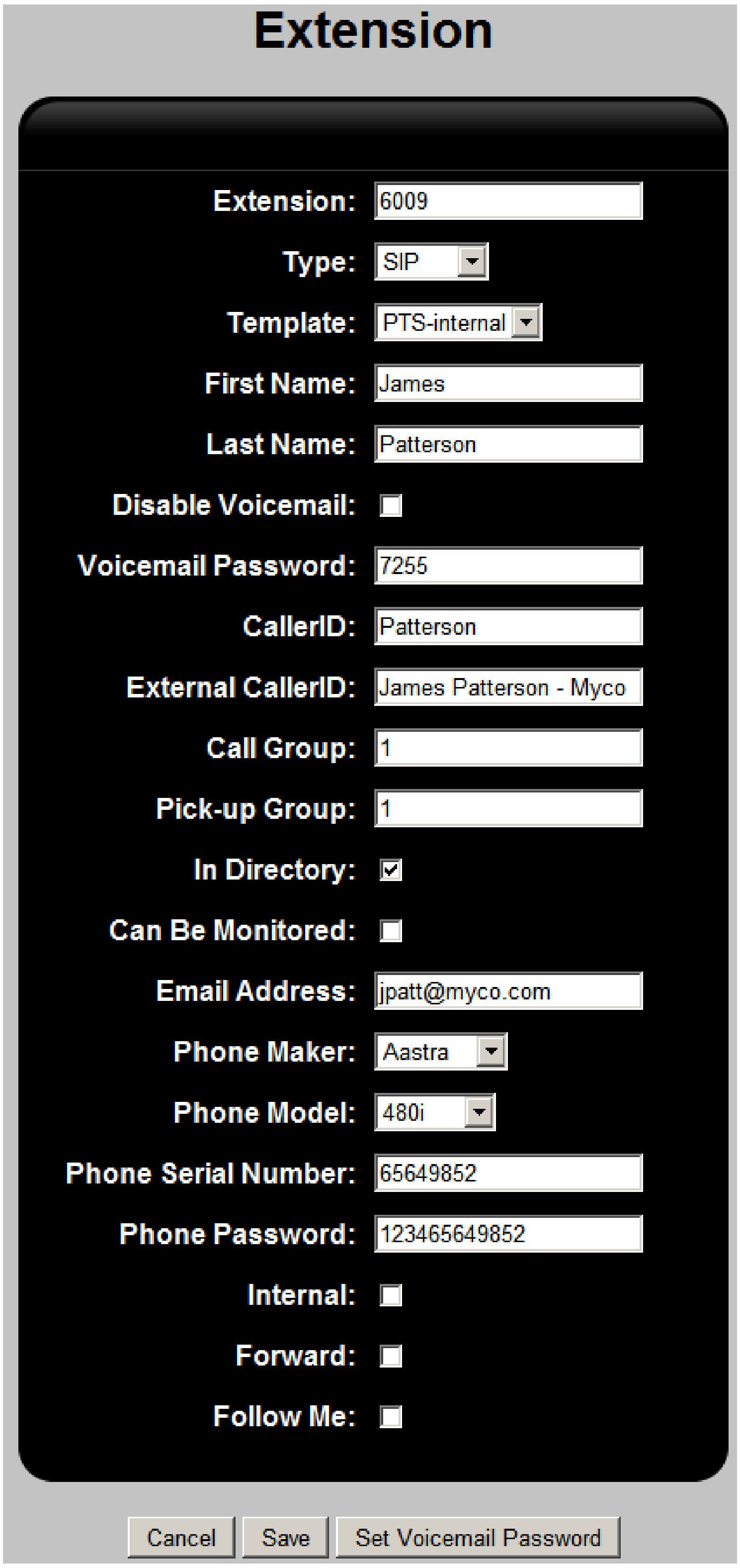


Figure 51 Extension Screen – SIP parameters

Extension:

The extension number (typically four digits). An error message will result if the entered extension number is already in use.

Type:

Select the type of extension:

SIP:

Extension is a SIP hardware phone or SIP software phone

Analog

Extension is an analog phone. Often used for an analog fax machine or analog telephone set.

Virtual

A virtual extension is used for shared or departmental Voicemail boxes, IVRs, Conference Bridges and Ring Groups.

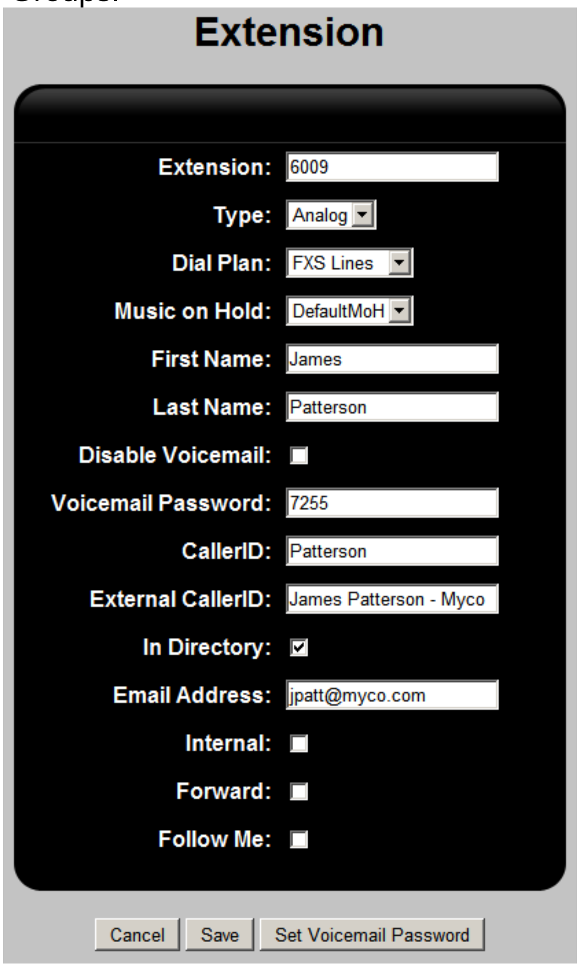


Figure 52 Extension Screen – Analog parameters

Subtypes:

Voicemail:

Establishes a voicemail box which is not associated with a physical extension. For 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.

See field descriptions below.

IVR:

Establishes a new IVR (Interactive Voice Response) system. Requires only the extension number and name of the IVR.

NOTE: Configuration of the IVR is done in the **PBX -> IVR Menus** menu.

Figure 53 Extension Screen – Virtual parameters

Conference Room:

Establishes a new Conference Room.

NOTE: Configuration of the Conference Room is done through **Users-> Conference Rooms**.

Ring Group

Establishes a new Ring Group.

- NOTE: Configuration of the Ring Group is done using **Users-> Ring Groups**.

Template:

Selects the extension template to use for default configuration of the extension.

NOTE: See the **PBX -> User Templates** section for details.

First Name:, Last Name:

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

NOTE: Choice of searching the first or last name for directory is done in the main IVR menu.

Voicemail Access Code:

Accommodates up to 8 digits.

Password:

The user's numeric password for voicemail access.

CallerID:

Certain analog and VOIP carriers will permit the setting of the caller ID displayed on the called phone.

NOTE: Enter the CallerID as numeric or alphanumeric text as applicable.

External CallerID:

In Directory:

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

Can be monitored:

Email Address:

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

Phone Model:

Provides configurations for certain IP telephone set manufacturers.

Phone Serial Number:

The MAC address of the telephone set.

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

Phone Password:

Password for the phone to be used on phone registration

Internal:

Extension can be dialed from within the office but not from outside. This can be useful for extensions like ring group voicemails.

Forward:

Forwards calls to the designated extension.

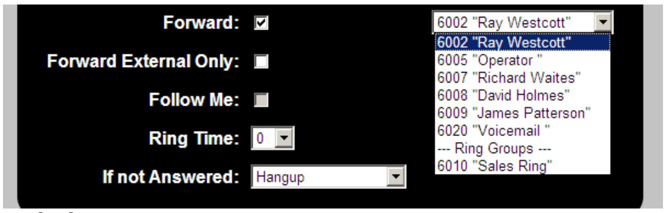


Figure 54 Forward Extended Menu

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:

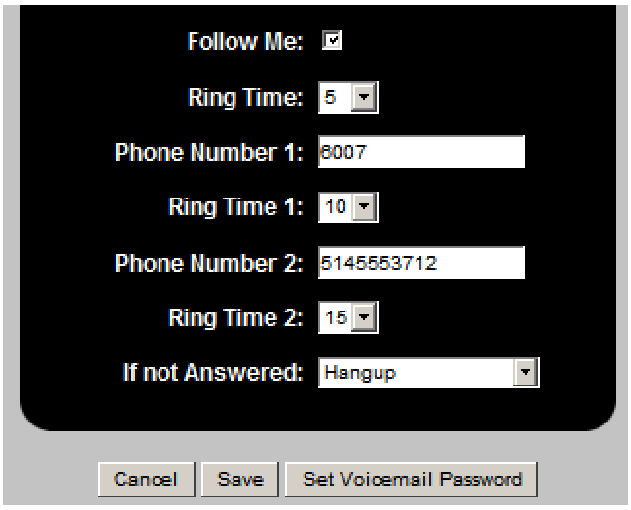


Figure 55 Follow Me Extended Menu

Ring Time:

After the specified number of seconds, (in this example, 5) the call will be forwarded to the number listed in Phone Number 1.

Ring Time 1,2:

After the specified number of seconds the call will be forwarded to the numbers listed below each.

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. Voicemail boxes are established using the Extensions menu and sub-type of Voicemail.

Goto IVR

Displays available IVR systems. IVR systems are established using the Extensions menu and sub-type of IVR.

Hangup

Hangs up the call if it has not been answered in the specified times by the specified Follow Me numbers.

Conference Bridge

Conference bridging allows quick, ad-hoc conferences, also known as Meet Me conferences, with or without PIN security.

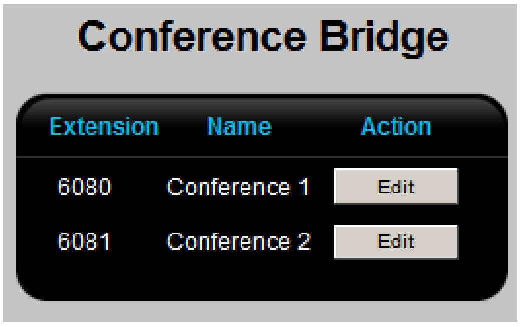


Figure 56 Conference Bridge Menu

The Conference Bridge Menu allows editing of existing Conference Bridges.

- NOTE: The conference extension must be set up in the Extensions menu with the Conference sub-type selected.
- NOTE: The number of conference bridges available on the system is set to a default of 2. Changes to this number can be made using **PBX -> PBX Settings**.
- NOTE: After adding extensions, be sure to click **Create Configuration** to make the new extensions active.

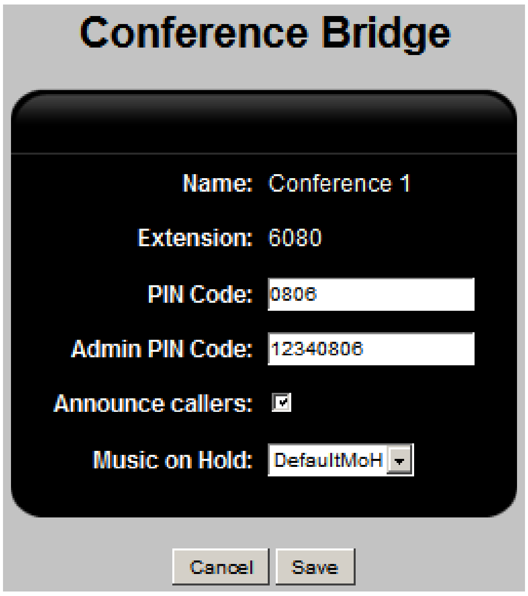


Figure 57 Conference Bridge Edit Screen

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 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 message will be played to existing conference members when a new member joins the conference.

Music on Hold:

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

Ring Groups

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

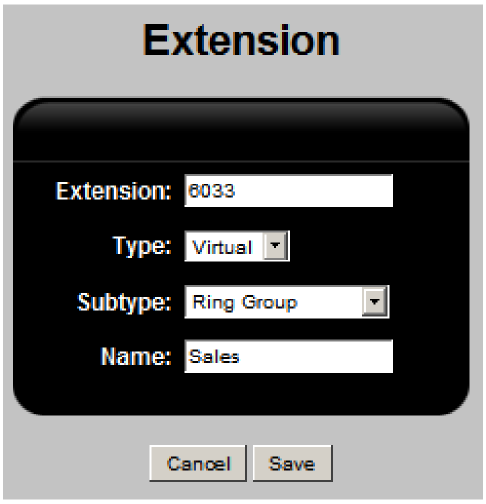


Figure 58 Ring Groups Menu Screen

- NOTE: If the Ring Group to be configured is not shown in the list, use **User -> Extensions** to create a virtual extension with a name and extension number.
- NOTE: Changes to the name and extension number of the Ring Group can also be made through **User -> Extensions.**
- NOTE: After making changes to Ring Groups, be sure to click Create Configuration to make the changes active.

To Create a Ring Group

- Click the **Users -> Ring Groups** menu
- Select the Ring Group extension to be configured and click Edit
- The name and extension fields will be filled in.

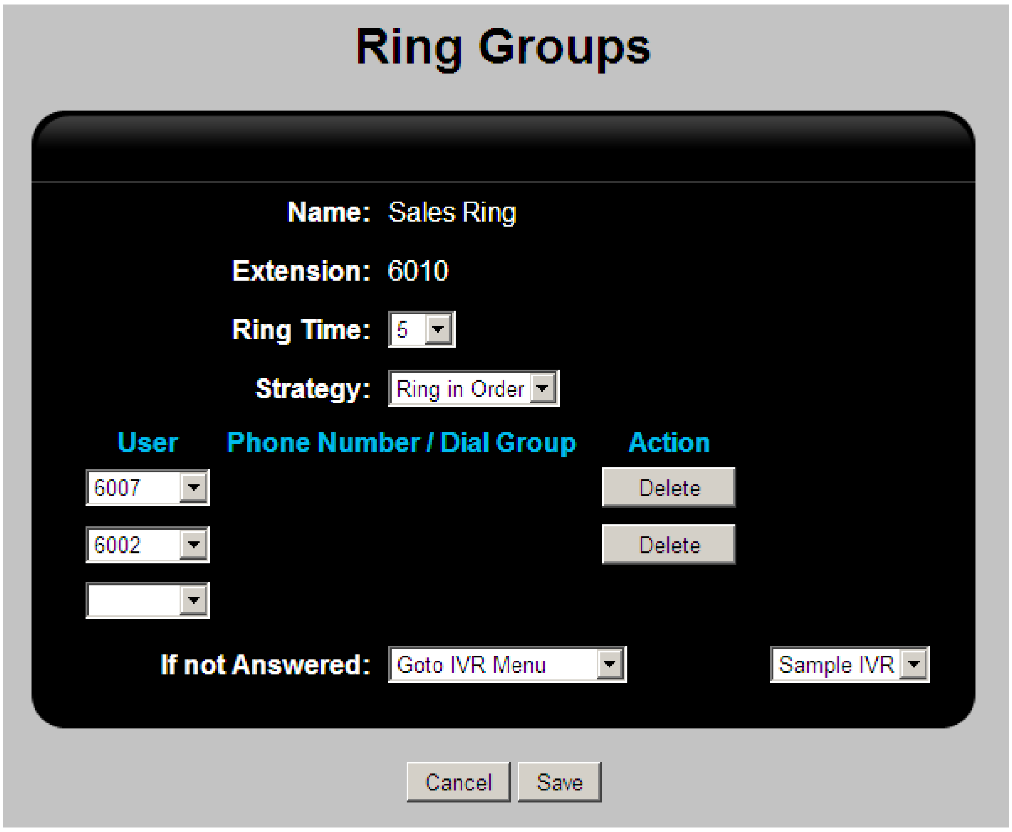


Figure 59 Ring Groups Edit Screen

Ring Time:

The number of seconds to allow ringing for the strategy in the following box:

Strategy:

Ring in Order specifies that one extension at a time will be rung
Ring All will cause extensions to be rung simultaneously.
NOTE: the call will not go to voicemail of any of these extensions unless specified in the “If not Answered” box.

User 1 -3:

Extensions: existing extensions that will be called as part of the group.
NOTE: Extensions should be created before using this drop-down menu.

If not Answered:

Determines action if none of the extensions answer the call. Choices include:

- Goto Voicemail Box,
- Goto IVR Menu

- **Hangup** from the drop-down menu.

NOTE: Selecting Goto Voicemail or Goto IVR Menu will provide a drop-down box of available choices. If the appropriate choice is not visible, it must be created using **User -> Extensions**.

PBX

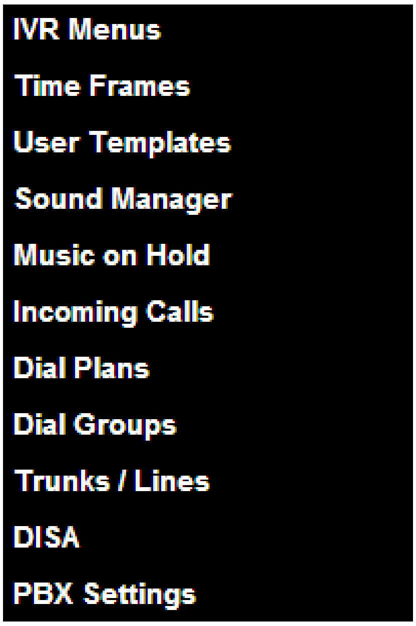


Figure 60 PBX Tab

The **PBX Tab** offers access to:

- **IVR Menus** to configure the Interactive Voice Response (IVR) system,
- **Time Frames** which are used to create time-based dialing rules,
- **User Templates** which define rules for groups of extensions,
- **Sound Manager** to manage sound files for Music on Hold and IVR applications,
- **Music on Hold** menu,
- **Incoming Calls** Menu which defines rules for the routing of incoming calls,
- **Dial Plans** which determine routing and permissions for outgoing calls,
- **Trunks / Lines** to configure analog and SIP lines
- **DISA (Direct Inward System Access)** Allows access to the PBX network from outside using a callback function. More details on the operation of this feature are available on the website at www.PositronTelecom.com

- **PBX Settings** which define overall PBX settings.

IVR Menus

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

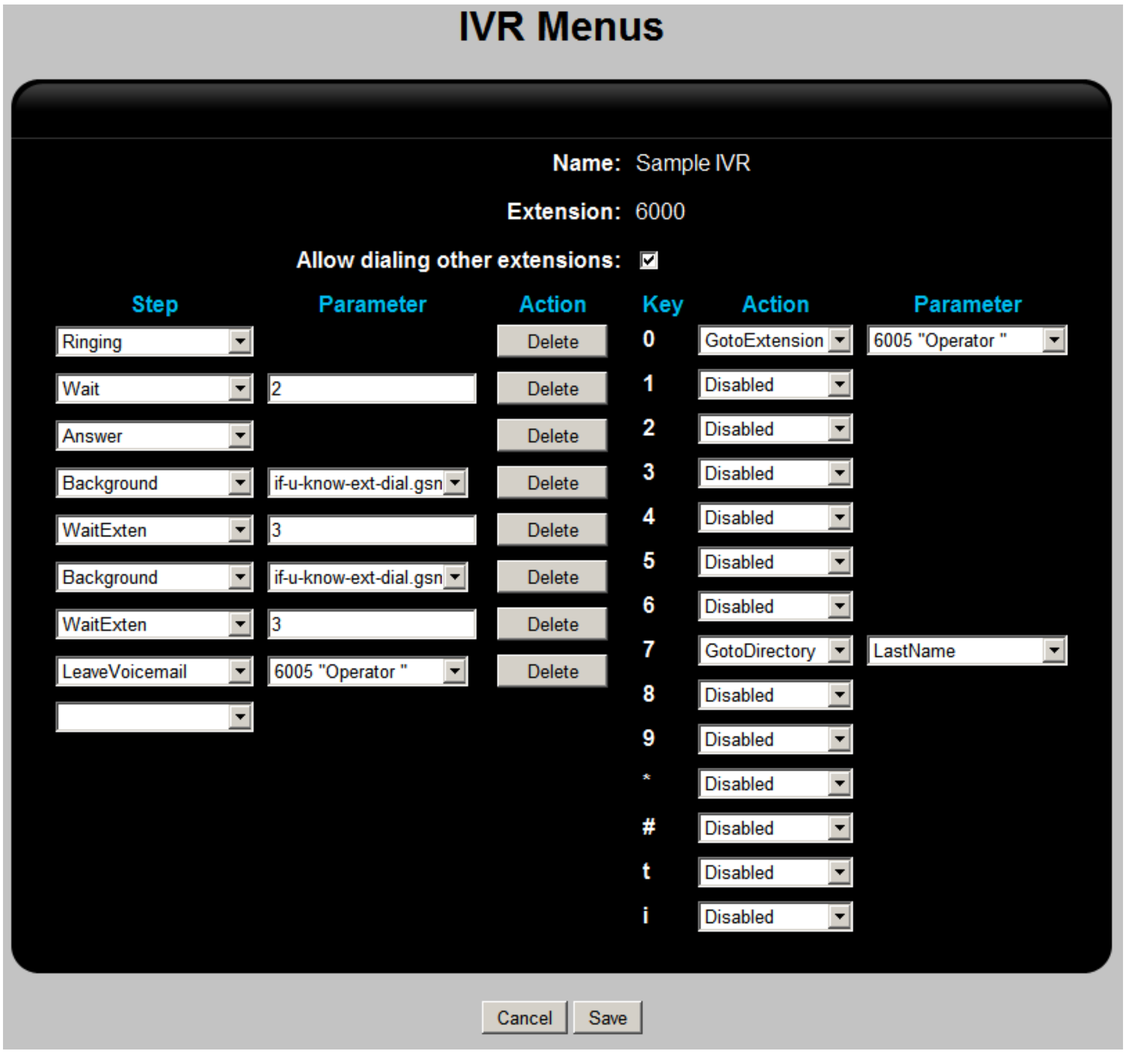


Figure 61 IVR Menu Screen

- NOTE: The IVR extension must first be created in **Users -> Extension** with the type “IVR”.
- NOTE: 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 Create Configuration to make the changes active.

To create an IVR menu:

- Open the menu **PBX -> IVR Menus** and open the new IVR by clicking Edit. The IVR screen will appear with the selected IVR application's name and extension shown at the top.
- Select the **Allow dialing other extensions** checkbox to allow users to dial those extensions at any time in the IVR process.
- **NOTE:** A malicious person may use this process to locate an outside line and use it for fraudulent purposes. Callers from outside should be confined to using individual key presses to locate extensions, and not be allowed to dial other extensions
- 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 entry from the drop-down box.

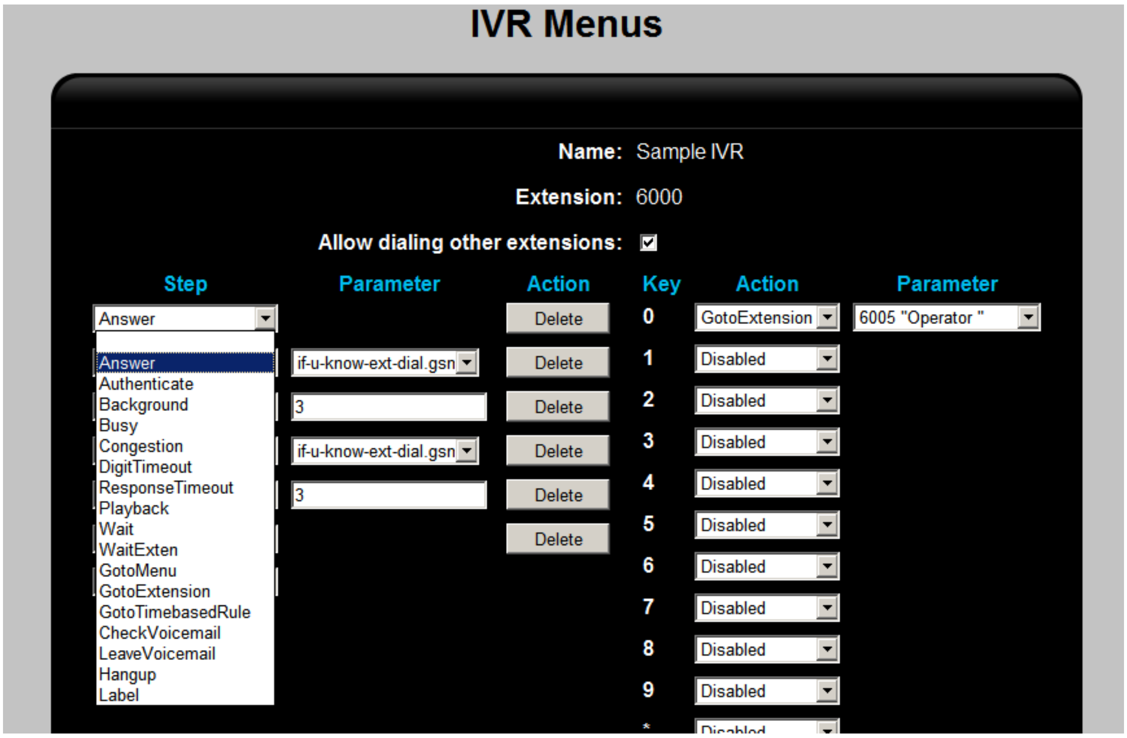


Figure 62 IVR Menus Menu Screen with Step drop-down menu

IVR Menu Commands

Answer:

First step in any new menu.

Authenticate

Used for entering PIN codes for authorization in order to proceed to the next step.

Background

Plays a sound file in background and waits for user input. Playback ends when user begins key presses. Parameter column will display a drop-down menu of all available sound files from which to choose.

Busy

Plays a busy tone.

Congestion

Plays a congestion (fast busy) tone.

Check Voicemail

Allows user to enter a voicemail box extension to retrieve messages.

DigitTimeout

Sets the maximum amount of time between keypresses. Enter the number of seconds in the Parameter column to the right.

ResponseTimeout

Terminates the call after a specified number of seconds have elapsed without a user response. Enter the number of seconds to wait in the Parameter column to the right.

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.

Leave Voicemail

Sends user to voice mailbox specified in Parameter column.

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.

Wait

Pauses execution for the specified number of seconds. Enter the number of seconds to wait in the Parameter column to the right.

Goto Menu

Sends caller to the top of the specified IVR menu file.

Goto Extension

Sends the caller to the specified extension. Select the extension from the Parameter drop-down list.

GotoTimebasedRule

Transfers control to another IVR program if a selected Time Frame is currently in effect.

Hangup

Terminates the call.

Keypress Actions

The Action column provides for the selection of events from drop-down lists depending on the user pressing the digit keys, the . #, t and I keys. Options available are:

Disabled

Key disabled

Goto Menu

Pressing this key will transfer control to the specified IVR.

Goto Extension

Pressing this key will transfer user to specified extension.

Goto Directory

Pressing this key will transfer control to specified IVR

Hangup

Terminate call.

PlayInvalid

Plays sound file to advise caller that they have made an invalid entry.

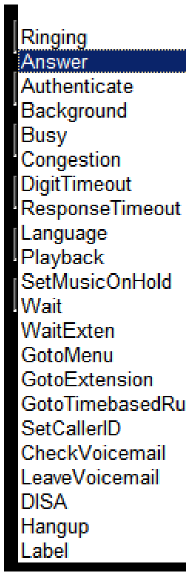


Figure 63 IVR Step drop-down menu

Parameter

Lists filenames of message fragments to be played during this step.

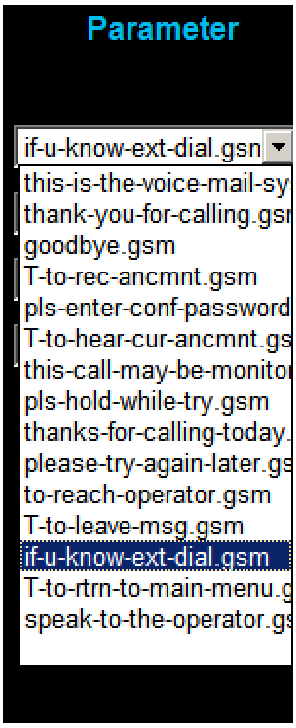


Figure 64 Parameter drop-down menu

Action

Describes step to be taken as a result of step completion.

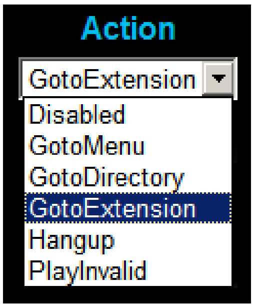


Figure 65 Action drop-down menu

Parameter

The Parameter drop down menus appear when needed and are governed by the selected Action. In the example below, the parameter which needs to be set is the extension is affected by action.

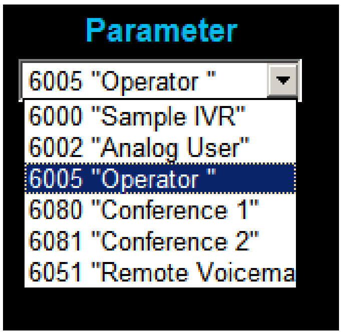


Figure 66 Sample Parameter drop-down menu

Time Frames

Determine the schedule for implementing certain calling rules. Each rule is given a name by which it is referred to in Dial Plans.

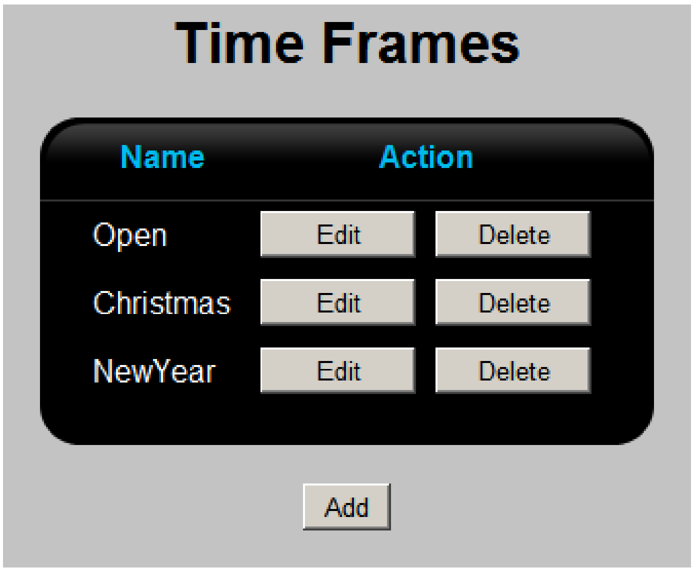


Figure 67 Time Frames Menu

Rules can be set for time of day, day of week, day of month etc.

Name:

The name of the Time Frame.

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 the Time Frame, or "" to choose any day.

Start Month Day

The day of the month to start the Time Frame, or "" to choose any day.

End Month Day

The day of the month to end the Time Frame, 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.

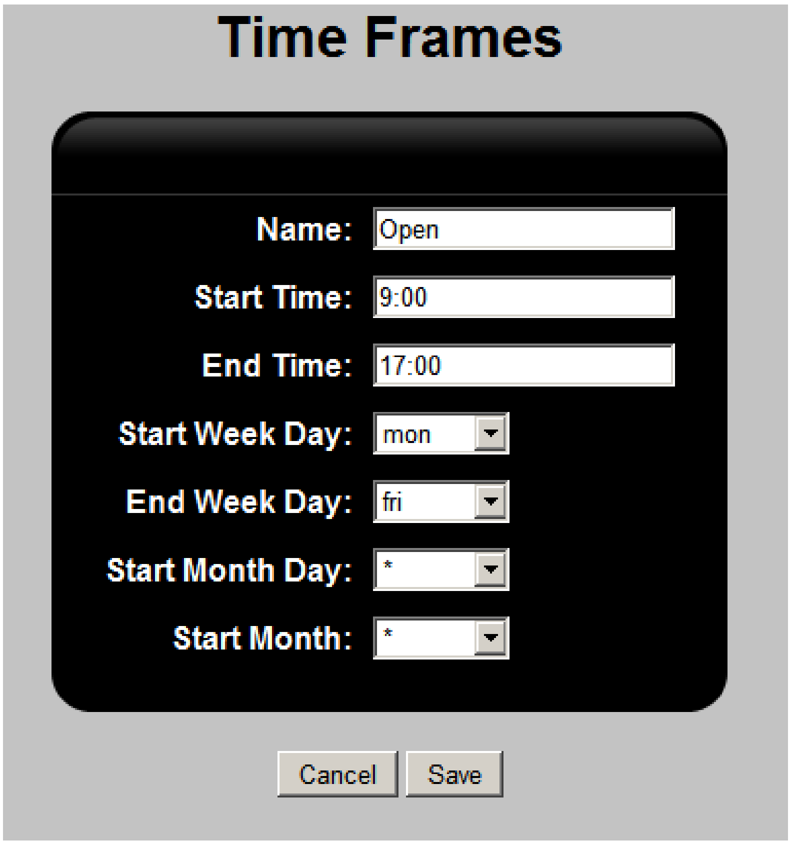


Figure 68 Time Frame Menu Screen

Example:

The Time Frame called “Open” 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 trunks during all periods while the Open time frame is in effect.

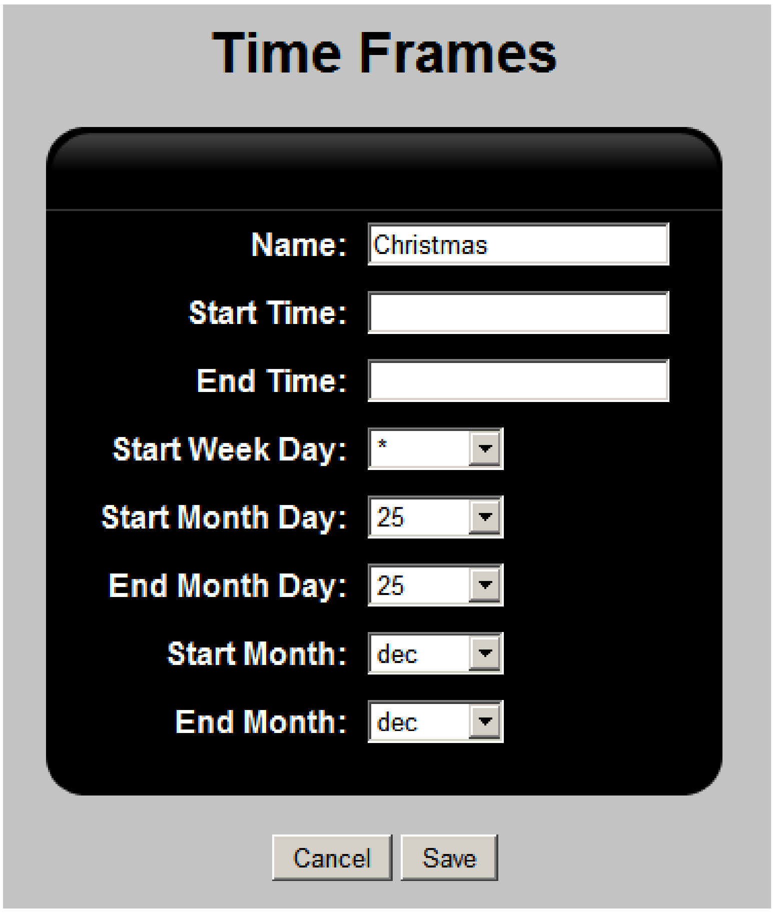


Figure 69 Time Frame Menu Screen

Example:

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

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

The organization may choose to restrict the use of outgoing analog and SIP trunks.

User Templates

User templates permit the establishment of default settings which can be applied to a group of extensions. Multiple templates will allow like settings to be applied to groups of similar extensions.

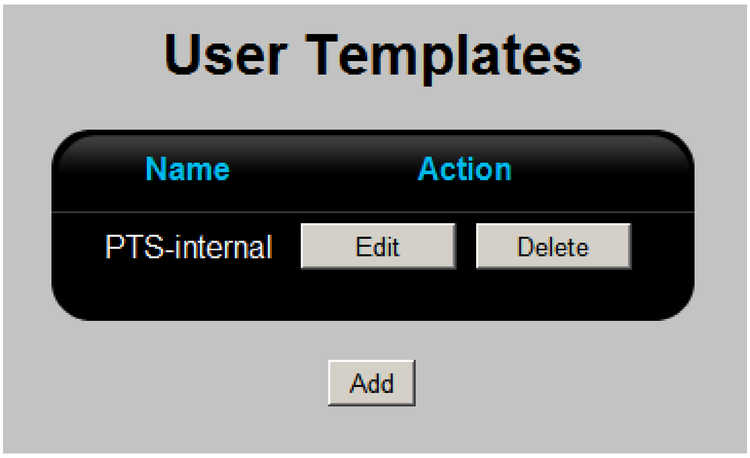


Figure 70 User Templates Menu

- The PTS-internal template is shipped with the system. It provides settings common to extensions used internally within an office.

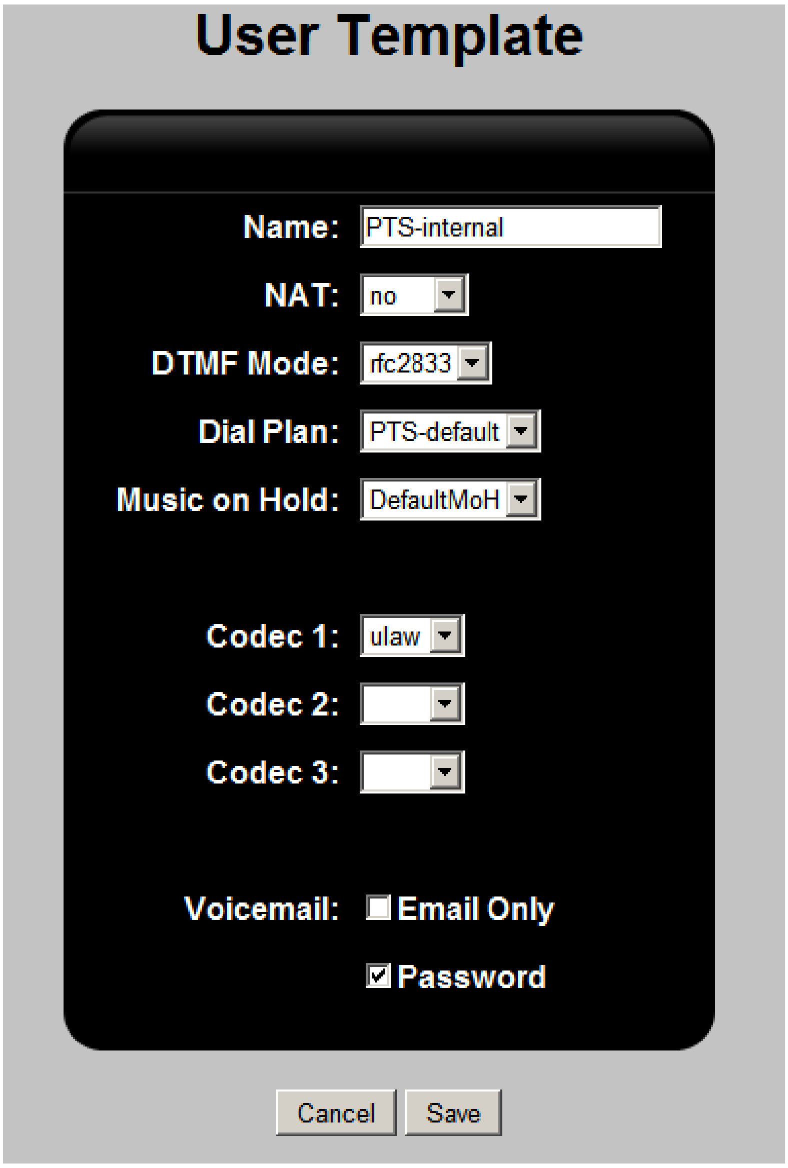


Figure 71 User Templates Edit

Name:

Name of the template

NAT: (Network Address Translation)

Determines whether to establish firewall checking. Typically, local network users will have NAT set to “No,” and remote users to “Yes.”

Dial Plan:

Default dial plan to be established for extensions.

Music on Hold:

Music file to be played for callers to these extensions when on hold.

Codec 1 – 3:

Codecs (in order of priority) to be used by these extensions.

Voicemail checkbox:

Whether these extensions have voicemail capability enabled.

Email Only:

Whether voicemail for these extensions will be delivered by email, or whether users can call for messages.

Password:

Whether voicemail boxes will require passwords for access.

Sound Manager

Used to load or delete sound files used in the Music on Hold, IVR systems and main language systems.

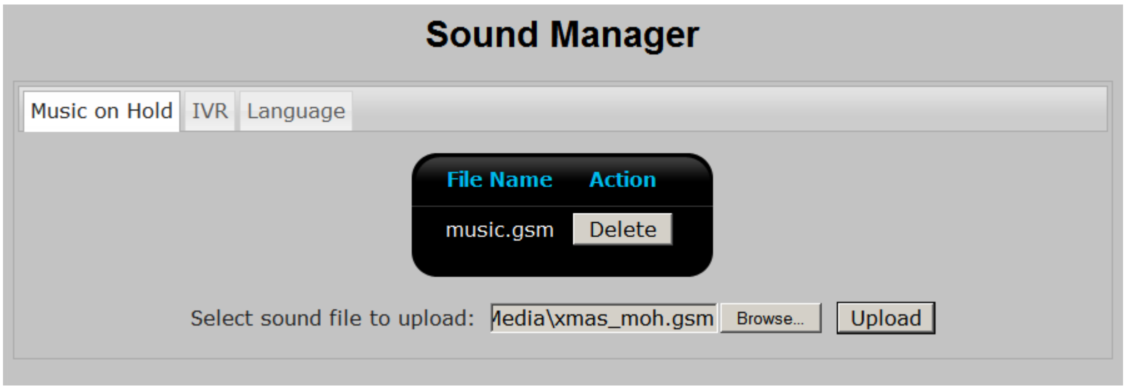


Figure 72 Sound Manager, Music on Hold

This procedure uploads the files from the host file system to the unit's memory, and identifies them as Music on Hold, IVR or Language files, making them available to choose from in subsequent steps.

Music on Hold

To upload a new Music on Hold sound file into the system:

- Click the Browse button 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 in the IVR.

To remove a Music on Hold sound file from the system:

- Click Delete beside the filename.

IVR

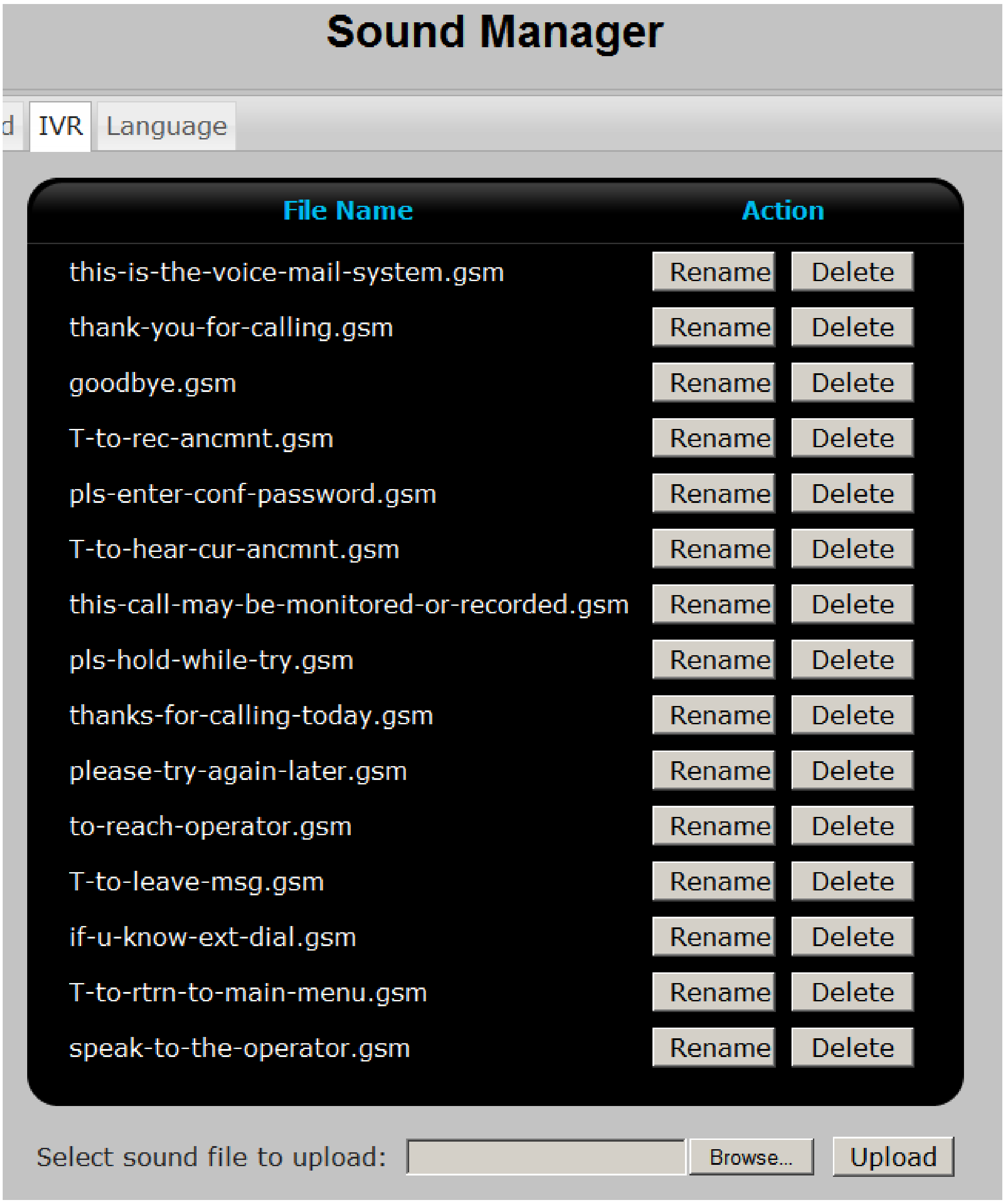


Figure 73 Upload IVR

To record a new IVR sound file:

- Using an extension dial the IVR Recording code. The default is *95.
- After the beep signal, record the new IVR message.
- Press “#” when finished.
- Options for disposition of the recording will be presented. 1 – to play, 2 – to re-record.
- Hangup the extension to end the recording.
- Go to the **PBX -> Sound Manager -> IVR** tab menu.
- Locate the new recording, identified by an extension number and timestamp.
- Click Rename, select the text and give the file a meaningful name.

NOTE: Do not leave spaces in the filename.

NOTE: The Filetype should remain .gsm

Language

Figure 74 Upload Language

- NOTE: Only one language file can be uploaded at a time. Uploading a new language file will replace the previous file.
- NOTE: Language sound files are selected using the drop-down menu in the **PBX -> PBX Settings** menu.
- NOTE: Language sound files are unaffected by the Languages: drop-down menu on the web interface.

Music on Hold

Music on Hold is played to callers who are on hold, or to users of the extension when they are awaiting the start of a conference. The Music on Hold Menu allows for the selection of the active Music on Hold sound file.

Figure 75 Music on Hold Menu Screen

Name:

Lists files that are available for the application.

Source:

"File" is the default type from the drop-down menu.

File Name:

Available files will be listed in the drop-down menu.

NOTE: Sound files must first be loaded using **PBX -> Sound Manager**.

Incoming Calls

Controls the routing of incoming calls. Typically, this will route all incoming calls to the main IVR system. Rules can cause calls received on a dedicated analog fax line, to be routed directly to the fax machine. Calls received on a specific analog or SIP line can be routed to a particular extension.

Example:

Calls received on a particular line can be sent directly to a specified department's IVR system or dedicated voicemail box. All other calls will be sent to the main IVR system.

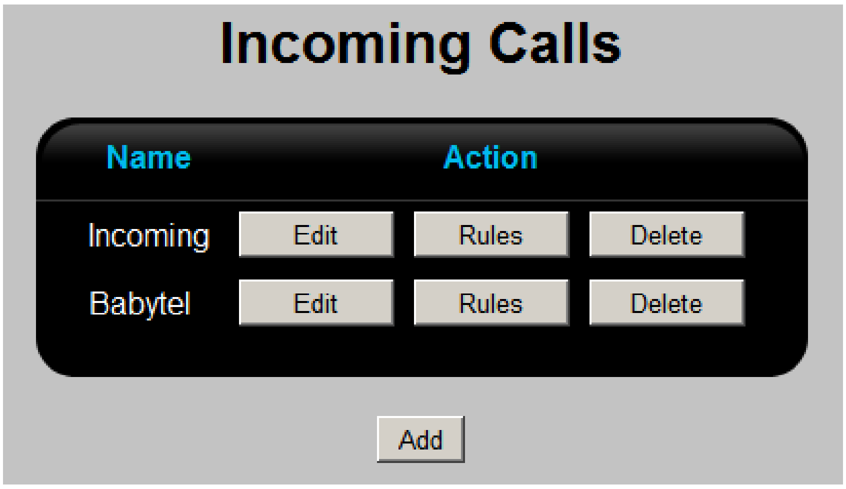


Figure 76 Incoming Calls Menu Screen

From Provider:

The drop-down menu provides a choice of provider for the incoming call. NOTE: Providers are set up in the **PBX -> Trunks/Lines** menu.

To Extension:

Select the extension to receive the call from the drop-down list. Extensions are set up in the **Users -> Extensions** menu.

DID:

Available only to calls coming in through the VoIP provider, calls received on this phone number will be routed to specific extension.

Figure 77 Incoming Calls Edit Screen

Example:

Incoming calls on the VoIP service named Babytel using the DID number of 212-555-1212 will be routed directly to extension 6002.

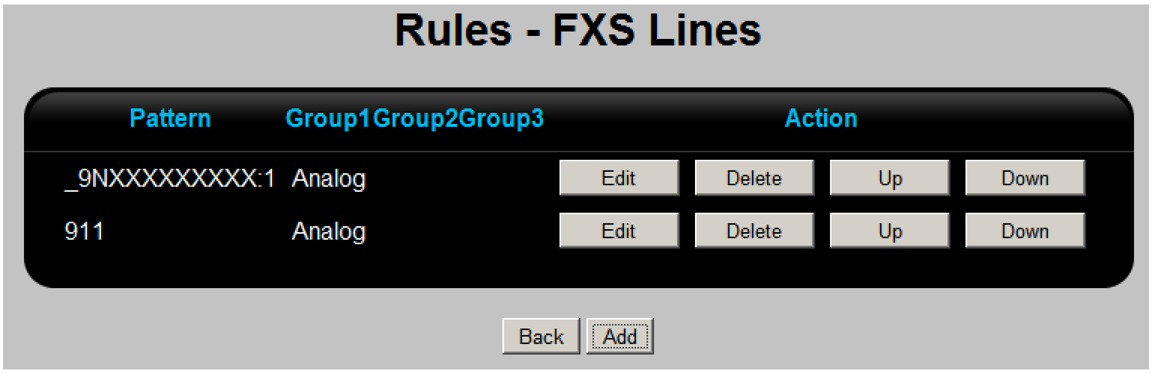
Dial Plans

The Dial Plan defines dialing permissions and least-cost routing rules. Dial plans are used in conjunction with calling rules to determine how outgoing calls are to be routed. For example an organization may choose to route local call through the analog telephone network, and long-distance calls through the VoIP system.

Calling rules within the Dial Plan define the specific outgoing call rules using a pattern-matching system. Dial Plans control how calling rules are applied to specific classes of extensions.

- NOTE: Extensions and calling rules are defined first, so that Dial Plans can be built upon them.
- NOTE: If a user dials a specific pattern which has not been defined, the call is considered to be internal and will be routed internally.
- Dial plans are executed in sequence, from top to bottom as they are listed in the Web Interface. For this reason, the rules will generally be listed with long patterns at the top, and shorter ones toward the bottom.
- The **UP** and **Down** buttons allow for repositioning of rules as they are added.

Figure 78 Dial Plan Menu Screen



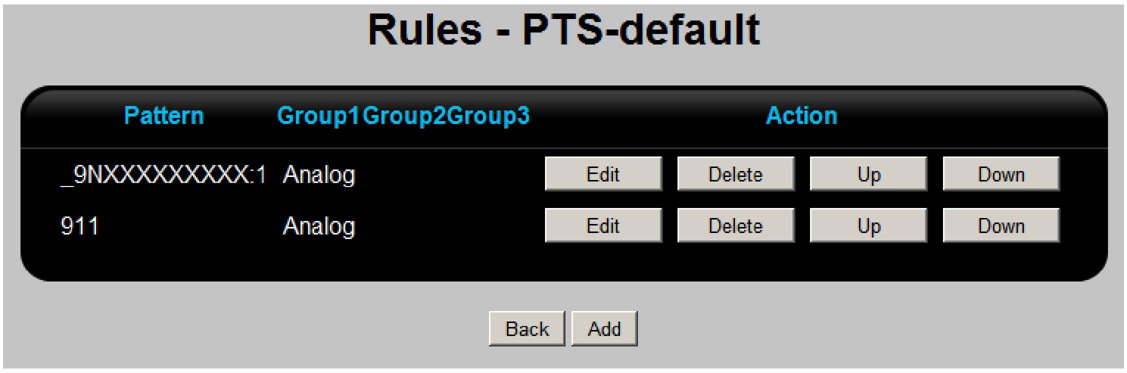


Figure 79 Calling Rules Example Menu Screens

Dial Plans

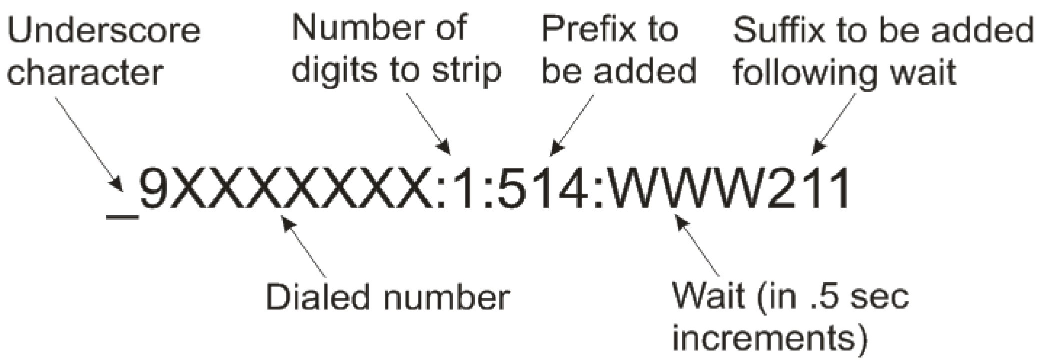
Dial-plans are calling rules which govern whether a call is internal or external.

Any dial attempt that doesn't match a Calling Rule will be considered an internal call, so the system will try to route the call to an internal resource such as another extension or a feature.

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

The general form of dial plans:

_characters:number:number



_ (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.

_91NXXXXXXXXX: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

Trunks / Lines

Trunks and lines are used to allow the system to make calls to the PSTN. The outbound lines can be VoIP lines or traditional telephony lines. When connecting to service providers, the connection made is typically referred to as a "trunk." A trunk can use regular analog lines or SIP to connect to a VoIP provider.

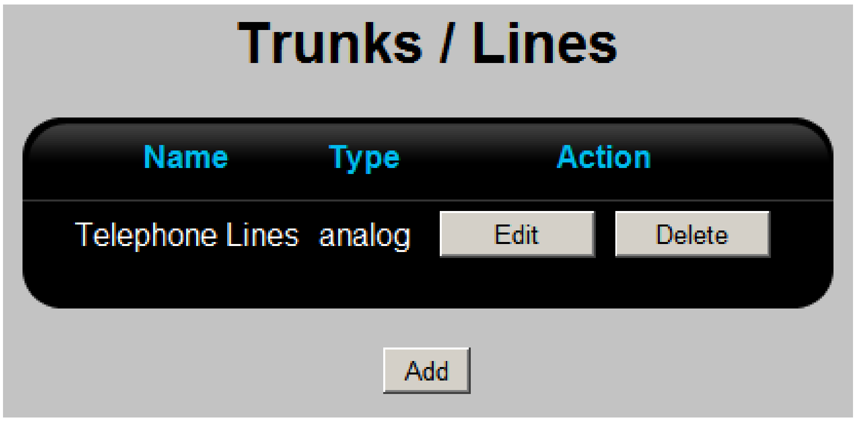


Figure 80 Trunks/Lines Main Menu Screen

Analog lines are used to connect to the traditional PSTN (Public Switched Telephone Network). For example, up to four analog lines can be accommodated in the V-114 PCI's four FXO ports. (A single analog telephone set can be connected to the FXS port.)

The units offer Kewl Start or Loop Start for the FXO ports.

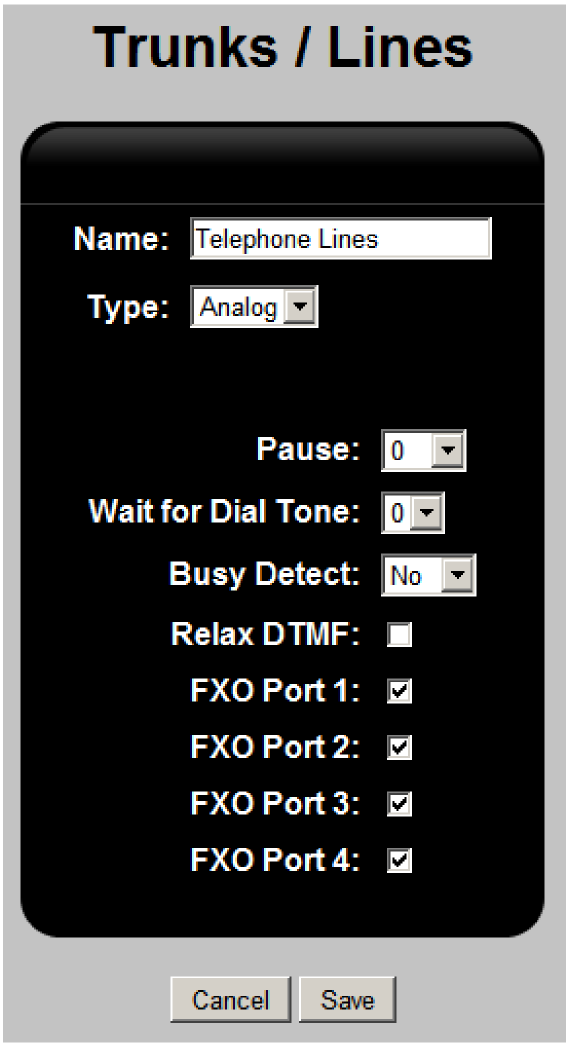


Figure 81 Trunks/Lines Analog Screen

Name:

Name to identify the line or lines being configured.

Type:

Drop-down menu allowing selection whether line is Analog or SIP

Group Number:

A group is one or more lines with common characteristics – for example, if 4 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 dialtone detection, enter 0. It is recommended NOT to enable dialtone detection.

Busy Detect:

Detects far-end hangup 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.

FXO Port 1 - 4:

Selects which port should be a member of the group

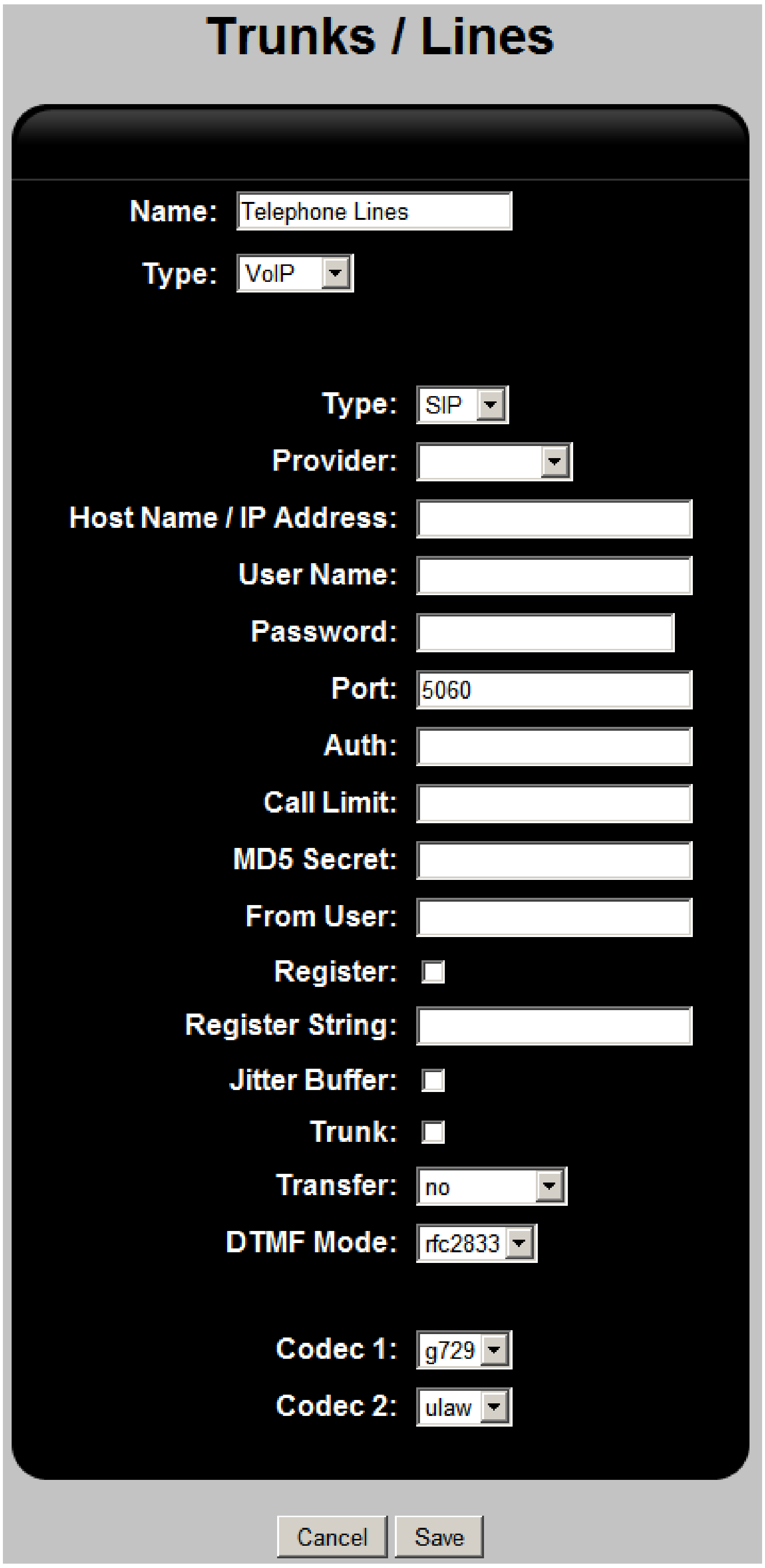


Figure 82 Trunks/Lines SIP Menu Screen

Name:

A name to identify the line or lines being configured.

Type:

Select SIP from the drop-down menu.
NOTE: Up to 4 VoIP lines can be configured.

Provider:

The name of the SIP carrier.

Host Name / IP Address:

The URL or IP address as provided by the SIP service Provider.

User Name:

User name as provided by the SIP service Provider.

Password:

Password as provided by the SIP service Provider.

Port:

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

Auth:

Md5 – information supplied by SIP carrier

Call Limit: (SIP Only)

Information supplied by SIP carrier

MD5 Secret: (SIP Only)

Information supplied by SIP carrier

From User: (SIP Only)

Information supplied by SIP 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 SIP carrier

Jitter Buffer: (IAX2 Only)

Information supplied by SIP carrier

Trunk: (SIP Only)

Information supplied by SIP carrier

Transfer: (SIP Only)

yes
no
media only

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 service provider should specify the preferred codec used by that network.

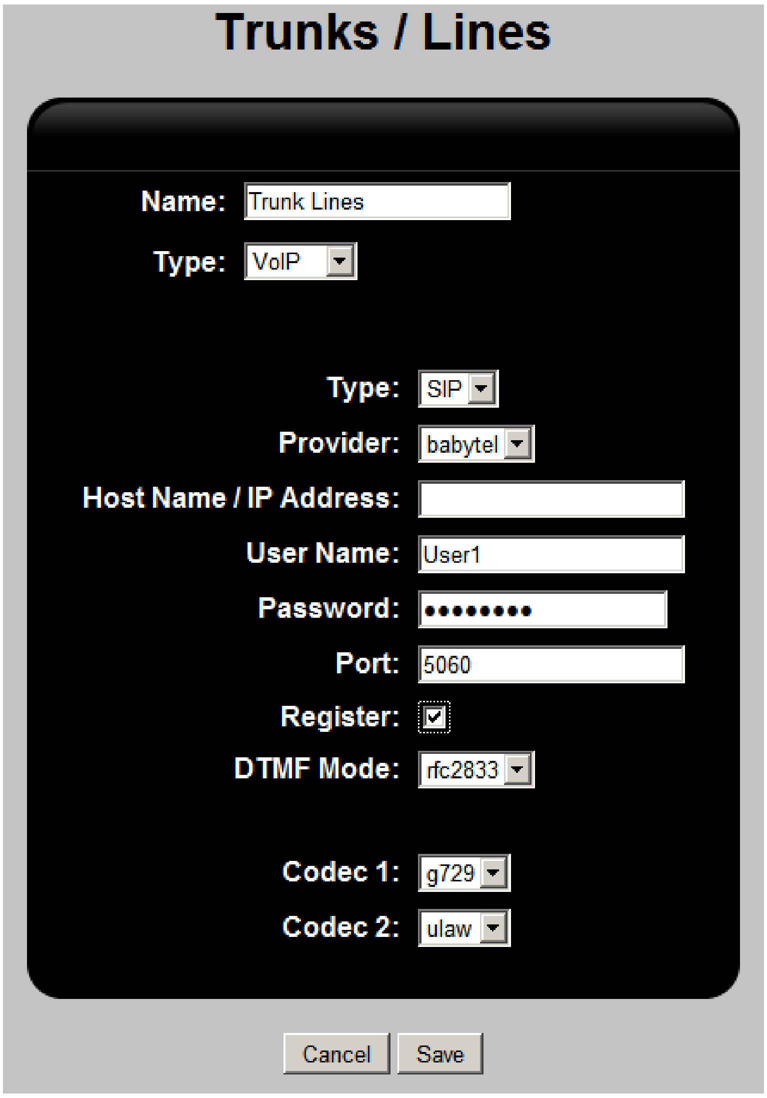


Figure 83 SIP Trunks/Lines VoIP Menu Screen

DISA

Direct Inward System Access (DISA) allows users of the PBX to access their communications functions from outside.
For example, a system-accredited salesperson on the road can make a local call to the PBX, and be given access to analog or SIP trunks to make a toll-call to a customer which will be charged to, and bear the Caller ID of the salesperson's company.

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

- User calls the system either at a specified extension or dedicated number.
- User is asked for, and enters the password.
- System hangs up.
- In one minute, the system calls the telephone number identified in the Phone Number field of the DISA menu (shown below).
- The CallerID of the system is the one shown in the CallerID field below.
- The user is asked for the password once more.
- Upon successfully entering the password, the user is given a dial tone.
- The user can place a call using the network's normal calling rules.
- The called party will see the CallerID as entered in the CallerID field below.

Figure 84 DISA Menu Screen

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 competed.

PBX Settings

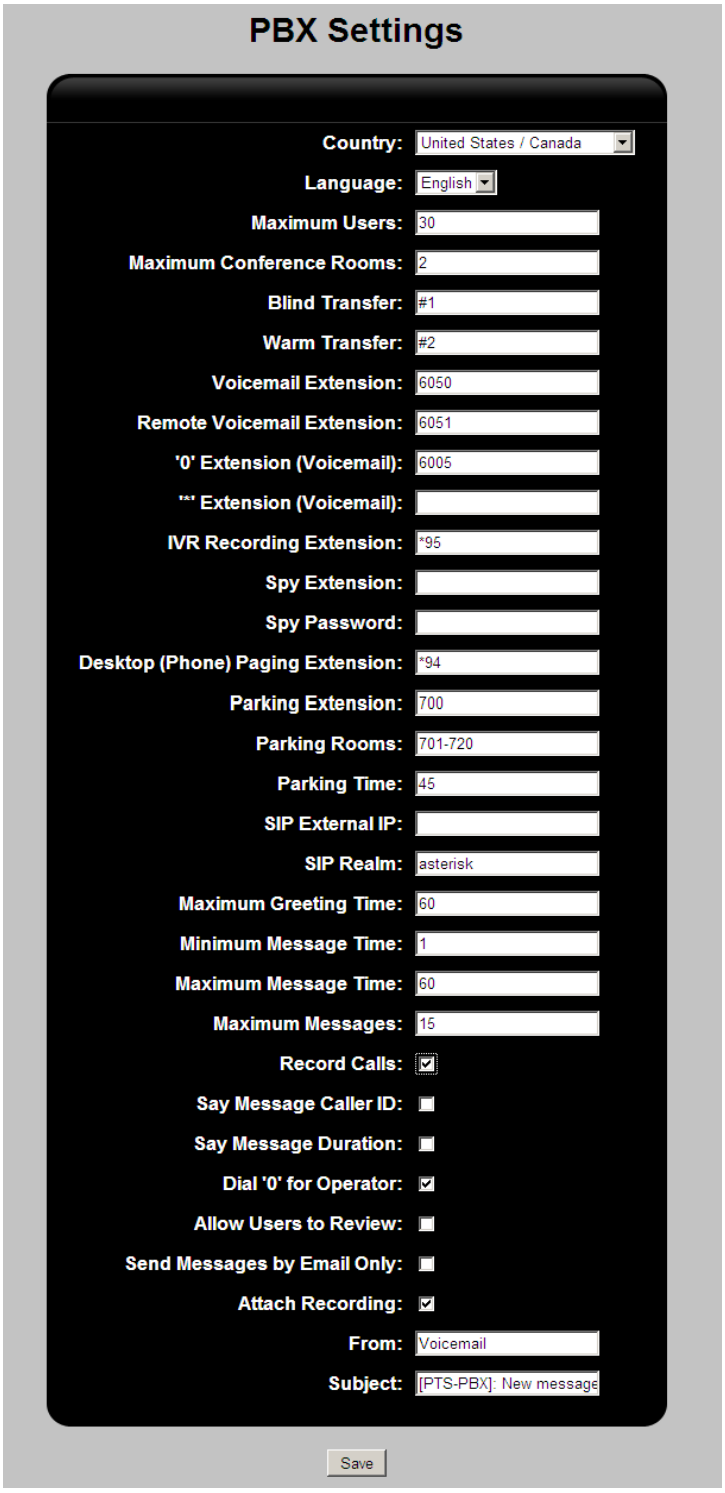


Figure 85 PBX Settings Menu Screen

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.

Blind Transfer:

When the transfer is done through the PBX, user enters “#1” and dials extension, then hangs up. This is sometimes called Cold or Unattended transfer.

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.

Pick-up:

Identification of the pickup groups available on the system – comma separated.

Voicemail Extension:

Extension to be used when system users call in for messages,

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. Defaults to 6000

'*' Extension (Voicemail):

Extension to be dialed when a user of an IVR system dials *. Defaults to 6051 (voicemail menu_.

IVR Recording Extension:

Local digits to dial to access the built-in voice recorder for IVR menus.

Spy Extension:

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

Spy Password:

Password to enter to validate a call monitoring user.

Audio-In Extension:

Allows a PBX user to monitor the Audio Input of a G-series unit.
Defaults to *91.

Paging Extension (Snoms 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.

Desktop (Phone) Paging Extension:

Extension to dial to activate the page system though sets which have local speakers.

Parking Extension:

The parking 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).

Parking Rooms:

Range of extensions that can be assigned to parking rooms

Parking Time:

Number of seconds of parking time allowed. After this time the calls are routed to the Main IVR.

➘ **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 as defined in the PBX Settings Menu. The range 701 – 710 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.
- 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.

SIP External IP:

This is the external IP of the PBX generally used to identify the address of the PBX to VoIP service providers. The information is usually kept by the network administrator.

SIP Realm:

Used in the calculation of Md5 digits. Used in SIP trunks and carriers.

Maximum Greeting Time:

Time in seconds allowed for users to create their Voicemail greetings.

Minimum Message Time:

Time in seconds allowed for incoming voicemail box messages.

Maximum Message Time:

Time in seconds allowed for incoming voicemail box messages.

Maximum Messages:

Maximum number of messages allowed in a voicemail box

Fax Detect (Analog):

If enabled, will add “Fax” to IVR menus. Will forward incoming fax calls from the IVR to extension 6002 in V-series, or 6003 in G-series units.

Record Calls:

Checkbox to enable all calls to be recorded.

Say Message Caller ID:

Checkbox to enable the system to provide spoken CallerID for the voicemail message.

Say Message Duration:

Checkbox to enable the system to provide spoken message duration time.

Dial '0' for Operator:

Checkbox to enable callers to access the Operator.

Allow Users to Review:

Checkbox to enable the callers to review their voicemail messages.

Send Messages by Email Only:

Checkbox to enable the system to send voicemail messages by email. Messages will NOT be saved on the PBX.

Attach recording:

Checkbox to enable the system to attach voicemail messages to emails.

From:

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

Subject:

Specifies format for the Subject field in email messages providing notification of missed calls.

Appendix 1

Acronyms

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

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

System Defaults

Default Settings

The following are default settings currently shipped with the V-114.

NOTE: As the systems evolve, the following information will most likely change.

Login

Username: admin
Password: password
(changes in **System -> Admin Account**)

Analog Set

FXS (Analog) telephone : extension 6002
(changes in **Users-> Extensions**)

Analog Lines

FXO (Analog) telephone lines 1 - 4: Enabled
(changes in **PBX -> Trunks / Lines**)

Network Settings

TFTP Server: 192.168.1.1
NTP Server: pool-ntp.org
Time Zone: Eastern (GMT-5)
IP Address: 192.168.1.2
Default Gateway: 192.168.1.1
DNS Server: 192.168.1.1
Mailhub: mail
(changes in System -> Network)

Outward Dialing Rules

- 9 followed by 10 digits will pass 10 digit number to available analog line
- 911 will pass 911 to available analog line

(changes in **PBX -> Dial Plans -> Rules**)

Extensions and General Settings

DTMF: United States / Canada frequencies and timing setting
Number of Conference Rooms (Bridges): 2
Blind Transfer of calls: #1
Warm Transfer: #2
Operator Extension: 6005
Voicemail Extension: 6050 (allows entry of user outgoing messages)

Remote Voicemail Ext: 6051
Parking Extension: 700
IVR Recording Extension: *95
Various settings related to voicemail
SIP External IP: blank
(changes in **PBX -> PBX Settings**)

Interactive Voice Response (IVR)

IVR Menu: 6000
Allow calling other extensions: Enabled
(changes in **PBX -> IVR Menus**)

Time and Date Call Blocking:

9am to 5pm (9:00 - 17:00 24-hour clock)
Christmas (Dec. 25, all day)
New Year's Day (Jan. 1, all day)
(changes in **PBX -> Time Frames**)

Default setting for SIP calls:

NAT: no
DTMF Mode: rfc2833
Music on Hold: file set (change **PBX -> Sound Manager**)
Codec: ulaw
Voicemail: Password protection available, messages NOT emailed
(changes in **PBX -> Time Frames**)

Language Setting:

Language Setting: English
(changes in **PBX -> PBX Sound Manager**)

Incoming Calls

Incoming Calls: FXO calls routed to IVR
(changes in **PBX -> Incoming Calls -> Edit**)

Conference Bridges

Conference Bridge: 6080
PIN Code: 0806
Admin PIN Code: 12340806
Callers announced: Enabled
Music on Hold: Enabled for first caller
Conference Bridge: 6081
PIN Code: 1806
Admin PIN Code: 12341806
Callers announced: Enabled
Music on Hold: Enabled for first caller

(changes in **PBX -> Conference Bridge**)

Appendix 3

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uClibc - a Small C Library for Linux

Erik Andersen <andersen@codepoet.org>

uClibc (aka µClibc/pronounced yew-see-lib-see) is a C library for developing embedded Linux systems. It is much smaller than the GNU C Library, but nearly all applications supported by glibc also work perfectly with uClibc. Porting applications from glibc to uClibc typically involves just recompiling the source code.

uClibc even supports shared libraries and threading. It currently runs on standard Linux and MMU-less (also known as µClinux) systems with support for alpha, ARM, cris, e1, h8300, i386, i960, m68k, microblaze, mips/mipsel, PowerPC, SH, SPARC, and v850 processors.

If you are building an embedded Linux system and you find that glibc is eating up too much space, you should consider using uClibc. If you are building a huge fileserver with 12 Terabytes of storage, then using glibc may make more sense. Unless, for example, that 12 Terabytes will be Network Attached Storage and you plan to burn Linux into the system's firmware..

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For installation instructions, see the file INSTALL.

uClibc strives to be standards compliant, which means that most documentation written for SuSv3, or for glibc also applies to uClibc functions. However, many GNU extensions are not supported because they have not been ported, or more importantly, would increase the size of uClibc disproportional to the added functionality. There is some discussion of these differences in the "docs" directory.

Additional information (recent releases, FAQ, mailing list, bugs, etc.) can be found at http://www.uclibc.org/.

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/ zlib.h -- interface of the 'zlib' general purpose compression library
version 1.2.2, October 3rd, 2004

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