GXE502X Users Manual

For Release 1.0.1.50

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1. Introduction

Thank you for purchasing the Grandstream GXE502X IP-PBX, a SIP-based, affordable, feature-rich converging communication platform designed to meet the communication requirements for small to medium sized enterprises.

The GXE502X provides the cutting-edge IP-based communications that businesses demand while leveraging existing infrastructure and providing a smooth transition into IP telephony. Based on open-standard SIP, the GXE502X can easily integrate into and interoperate with other components of your existing communications network while providing a rich set of features to reduce costs and increase productivity. Built-in FXO and FXS ports enable the GXE502X to interface with analog lines and devices while concurrently registering to SIP trunks and SIP-based trunk gateways to maximize available communications resources. The PoE (power over Ethernet) capability (LAN port) and low power consumption (15Watt Max.) solidify the GXE as an environment friendly solution (compared to similar PC based systems). This also ensures non-stop communication even during power outages (necessary UPS required)

The GXE502X also includes an Express Setup wizard and an intuitive user interface that allows users to quickly configure extensions. .. Voicemail, fax mail, faxmail-to-email, conference bridges and other enhanced features can be turned on with minimal effort via the GXE502X web configuration interface. The GXE's broad feature set, ease of operation and affordable price range make it ideal for any SMB.

- **Equipment Packaging**

  The GXE502X IP-PBX package contains:

  1) One GXE502X IP-PBX unit
  2) One 12 Volt power adapter
  3) One Ethernet cable

2. Extension Configuration

Phone Extensions

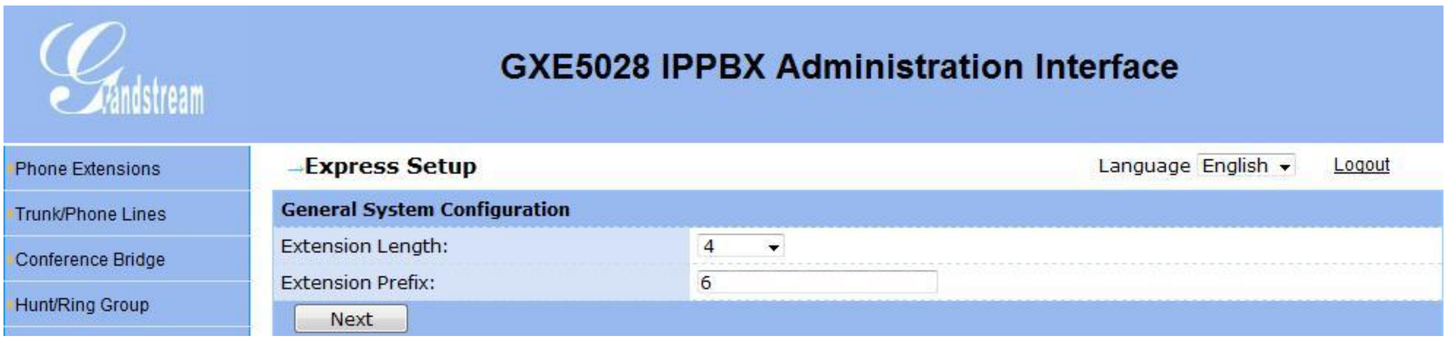
The front end menu of the GXE web interface is the Phone Extensions menu. This menu and its several different sub menus allow users to configure and add extensions with ease.

Extensions for internal users can be created from the **Express Setup** menu of a factory-default GXE502X or from the **Phone Extensions** menu at any time. The extensions can be local users (shown in Green in the Extensions Directory) in the internal network, or remote users connecting over the open Internet (shown in Blue in the Extensions Directory). This flexibility allows GXE502X users to bring their office extensions virtually anywhere that has a broadband Internet connection. This gives users easy access to personnel (home office and road warriors alike) without incurring any PSTN toll charges.

- Express Setup

The **Express Setup** process allows users to automatically configure extensions by auto-provisioning Grandstream phones[1]. This sub menu will appear ONLY on factory default GXE units or after performing a factory reset. Once the Express setup is executed, the sub menu will disappear. Grandstream highly recommends running the Express Setup even though the user might not be ready to install any phones at the moment. When we run the Express Setup the GXE will auto-configure the extensions for the conference bridges, FXS ports and more importantly it will create a digit map under the internal call routing profile. We will explain call routing profiles in a different section but they are a key element to allow transparent communication.
If you are not going to configure any phones at the moment you can start the auto-provisioning and then stop it a few seconds after so that the GXE can create all these items.

- **Using the Express Setup wizard to create extensions and auto-provision phones.**



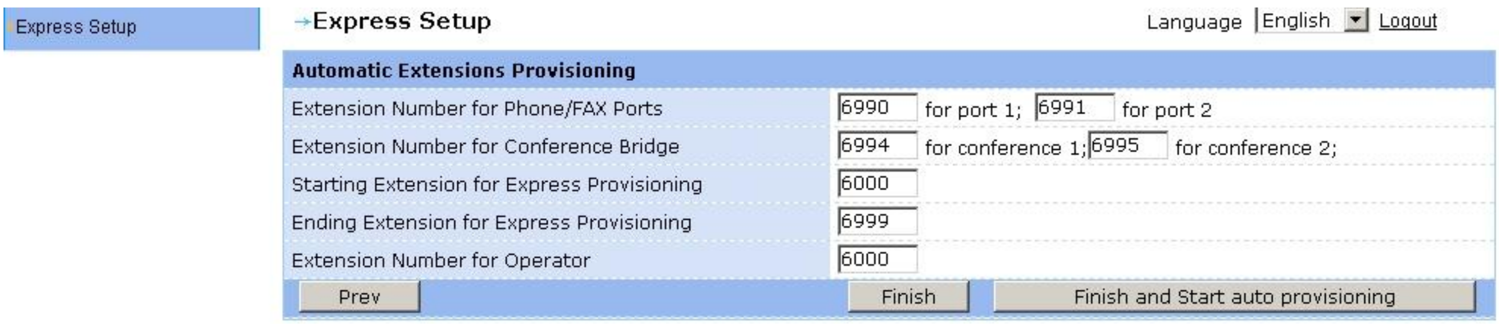
Please refer to the *GXE502X Quick Start Guide* for detailed instructions on setting up your GXW with the Express Setup wizard.
The Express Setup Wizard will be the initial point of configuration for your telephone system. The GXE needs to know these values first so that it can properly configure the system.

- *Extension Length*:
  All extensions will have the number of digits specified here. Please select a digit from 3 to 6.
  For example if you select 3, then the extensions will be like 101,102, 103…199.
  If you select 4, then the extensions would be like 1001, 1002, 1003….1999.
  For advanced users: You can create different digit maps in the internal call routing profiles with different extension lengths.
- *Extension Digit Prefix*:
  All extensions will begin with this digit.
  Other digits can be used for peer systems, trunks or feature access codes.
  For example if you select 6, then the extensions will be like 602,605,611.
  If you select 3, then the extensions will be like 307,356,322.
  For advanced users: You can also setup other local extensions with different prefixes in the in the internal call routing profile.

[1] Auto provisioning is currently only available for Grandstream phones

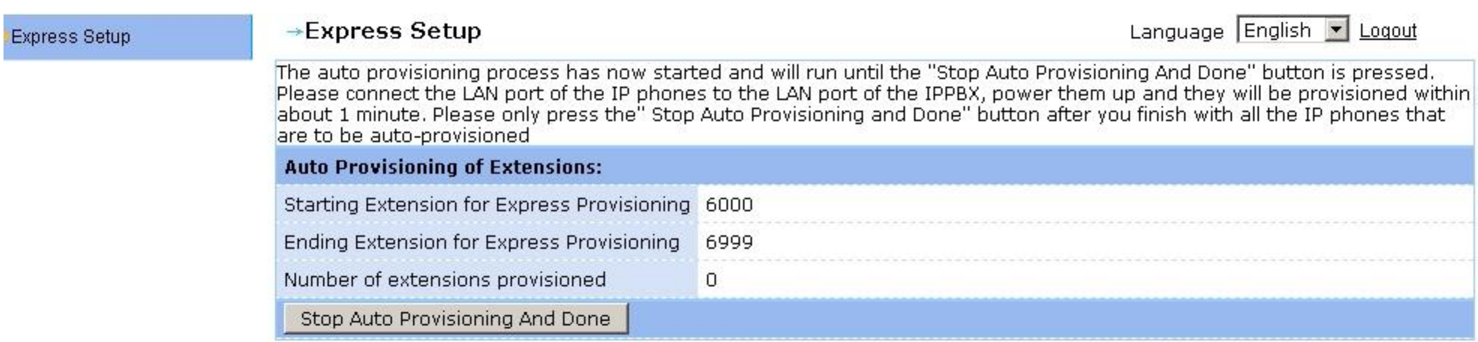
After you input these two numbers, you will find the following screen:

For this example I selected 6 to be my extension prefix and 4 to be my extension length as well.
As you can see the GXE has preconfigured some of the extension numbers that will take effect only after we start the auto-provisioning. If you wish to change any of these numbers please do it at this point. If you want to modify the extension Prefix, press the "Prev" button, otherwise press the "Finish" or the "Finish and Start auto provisioning" buttons; press “Finish" if you do not want to start auto provisioning, the GXE will reboot by itself after you press this button. Press the "Finish and Start auto provisioning" button to begin the Auto provisioning process.
The extension numbers that you see in the example are as follows:



- Extension Number for Phone/FAX Ports: These are the two FXS ports labeled TEL1 and TEL2 in the back of the GXE. You can connect analog phones or fax machines to these ports. Additionally if you have the GXE on PoE and a PSTN line connected they can be used as your lifeline in case of a power outage.
- Extension Number for Conference Bridge: These are the extensions that correspond to each conference room in the GXE. The 5024 model comes with 2 conference rooms and the 5028 model comes with 4 conference rooms. So each conference room is in reality a virtual extension. We will discuss more about the conference bridges later on.
- The starting and ending extensions for express provisioning will determine which range of extensions will be provisioned if they are connected locally, using a switch, to the LAN port of the GXE.
- The extension number for the operator is simply created automatically as the first extension since this is usually the easiest number to remember and typically used by an operator or secretary.

Once you get the auto-provisioning running you should see the screen below.



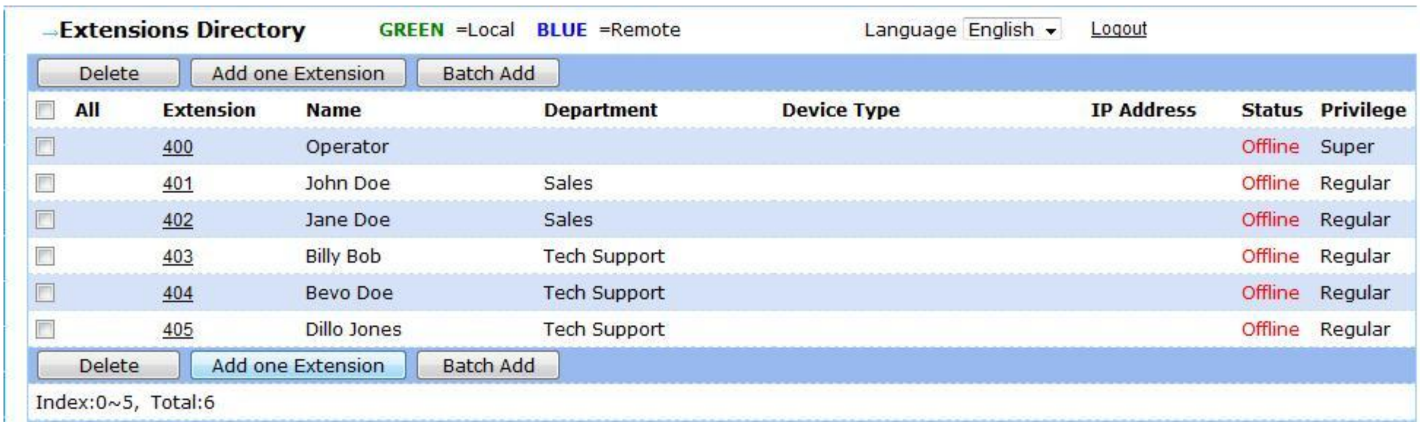
When you are done with the auto-provisioning, simply click on the "stop auto provisioning and done" button. Then click on the OK button on the pop up screen to finish. The GXE will then auto-reboot to finish the configuration process

- Extensions Directory

The **Extensions Directory** sub menu allows user to add, batch add, modify, and delete extensions, after the Express Setup process.

- **Using the Extensions Directory sub menu to create, modify, view, and delete extensions:**

The Extensions Directory section displays all SIP phone extensions configured on the GXE502X. This page also displays the IP address, firmware version and registration status of the end points. The following actions can be performed in this section:



All	Extension	Name	Department	Device Type	IP Address	Status	Privilege
	400	Operator				Offline	Super
	401	John Doe	Sales			Offline	Regular
	402	Jane Doe	Sales			Offline	Regular
	403	Billy Bob	Tech Support			Offline	Regular
	404	Bevo Doe	Tech Support			Offline	Regular
	405	Dillo Jones	Tech Support			Offline	Regular

- *Add an extension*: Clicking on the **Add One Extension** button will load the extension configuration page. This page allows users to set all of the extension's settings. Click on the **Submit** button to add the extension or **Cancel** to abort the add extension process.
  Configuration Field Explanations:
  1) User Name: Name of User as shown on the Extensions Directory screen
  2) Department Name: The Name of the Department that the extension belongs to. For example Accounting, Sales, Shipping, Support, etc.
  3) Extension: The Extension number. This is usually the SIP user ID in the IP phone or ATA.
  4) Privilege: This is used to determine the storage quota in flash memory for voicemail/video mail/fax mail, based on a percentage of the total space.
  5) SIP Password: The password required to allow phones to register to GXE502X. This password must be synchronized between the GXE502X and registered SIP end points (i.e.. IP Phone & ATAs).
  6) Voicemail Allowed: Toggles voicemail on and off for this extension.
  7) Ring Attempts before Forward to Voicemail: Configures the amount of rings, in seconds, before calls are forwarded to voicemail. (When voicemail is enabled)
  8) Fax mail Allowed: Enables the fax mail feature for this extension.
  9) Forward Voice/Fax mail to Email: This is the email address where the GXE will forward all voicemails and fax mails.
  10) Password to Retrieve Voicemail/Fax mail: The password required to retrieve Voicemail or Fax mail.

11) Call Forward[2]: Toggles Call Forwarding on and off. The factory default setting is OFF.
12) Call Forward To: The phone number the call will be forwarded to. It could be another extension or a phone number.
13) Call Forward Rule: Determines the conditions under calls are forwarded.
14) Time for No-Answer-Forwarding: This is the timer that once expired will forward the calls to the call forward number. Please make sure this timer is shorter than the voicemail timer if the extension has Voicemail enabled.

Advanced Options:

15) RTP Port Detection: Toggles the GXE's ability to detect RTP ports for remote extensions. This field is set to "Yes" on factory default units.
16) Do Not Disturb: When enabled, calls will go directly to voicemail (when VM is enabled). Calls will be rejected if voicemail is disabled.
17) Reboot Peer: When set to "Yes", this feature will reboot registered (online) Grandstream endpoints when the "Submit" button is clicked. The GXE502X can force extensions to reboot via this method. This is the check-sync method used in Notify messages.
18) Session Keep Alive: This feature lets you configure different methods to keep the session alive: None, Update, Re-Invite or Session Timer/Automatic. The factory default value is Automatic. Grandstream recommends using the default setting.
19) Session Expiration: Please refer to RFC 4028 for detailed information
20) Min-SE: Please refer to RFC 4028 for detailed information
21) Call Routing Profile: These are the profiles the extension will be using to place outbound calls. The administrator can select multiple call routing profiles from the available list to enable the extension in particular to be able to use them. For example the administrator could have created a local, cell phone, international and peers call routing profiles so he can decide what kind of call each extension can make. Simply select or unselect each profile from the list to activate them. Remember to click on the submit button.
22) Authorization Profile: The administrator can assign a predefined authority profile to this extension so that they might require a password to perform certain calls.
23) Voice mail storage time: This is the period of time in days that the GXE-502X will keep you voice/fax mail messages stored in flash memory. After the time has passed the GXE-502X will delete the messages automatically to save space in memory. Default 30 days.
24) Entry voicemail option: This will select in between 3 options for accessing voicemail from the user's extension using the feature code for voice mail.
    Extension and password: after dialing the feature, code the user will be prompted to enter both the extension number and the password to access the extension's voicemail.
    Password only: after dialing the feature code, the user will be prompted to enter his password only to listen to his voicemails
    Direct Call: After dialing the feature code, the user will start listening his voicemails bypassing all passwords.

- *Batch add extensions*: The **Batch Add** button allows you to manually add multiple extensions. The batch add configuration page lets you specify the range of extensions to add, as well as some general settings to apply to them. You may also modify the settings of these extensions through the Extensions Directory page. Click on the **Submit** button to add the extensions or **Cancel** to abort the batch add process.

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[2] Note: Call forward can also be activated by using feature code.

- *Modify an extension*: If you want to modify any of the extension's properties simply click on the extension number under the extensions directory page. To reboot the Grandstream phone/ATA (remotely), set the Reboot Peer setting to "Yes", then click on the **Submit** button to modify and/or reboot the extension or **Cancel** to go back to the extensions directory page.

- *Delete*: Check the square box to the left of the extension you wish to delete. You will be prompted for confirmation via a dialog box; click **OK** to confirm or **Cancel** to go back to the extensions directory page. You can select multiple boxes and delete several extensions at once. Note that this only affects extensions on the current page; extensions on other pages will not be deleted. You will be prompted for confirmation via a dialog box; click **OK** to confirm or **Cancel** to go back to the extensions directory page. You can also select the All checkbox to automatically select all extensions in the displayed page.

- **Busy Lamp Field and Message Waiting Indication**

Busy Lamp Field (BLF) and Message Waiting Indication (MWI) support are enabled on the GXE502X by default. New voicemail messages will automatically trigger the MWI light on phones that support it to notify the user of new voicemail messages.

To monitor the status of other extensions in the GXE502X, configure the BLF keys on a (Grandstream) IP phone to the extension numbers to be monitored. The BLF key configuration can be found on the basic settings tab of your Grandstream GXP phone's web UI. The GXE502X will trigger the LED of BLF keys to signal when the monitored extension is idle (Green), ringing (Blinking Red) or busy (Solid Red).

- General Settings

The **Auto Provision** sub menu provides users a method to Auto Provision Grandstream endpoints after the Express Setup process.

- **Auto Provisioning Extensions after Express Setup**

Click on "Auto Provision" under the Phone Extensions menu. Enter the starting and ending extensions for the phones (Make sure to enter numbers in the established digit frame structure) before connecting your Grandstream phones.

Click the "Start" button to start the auto provisioning process. Once the Auto-Provision page starts refreshing you may connect your phones to your switch/hub. Click "Stop" when done.

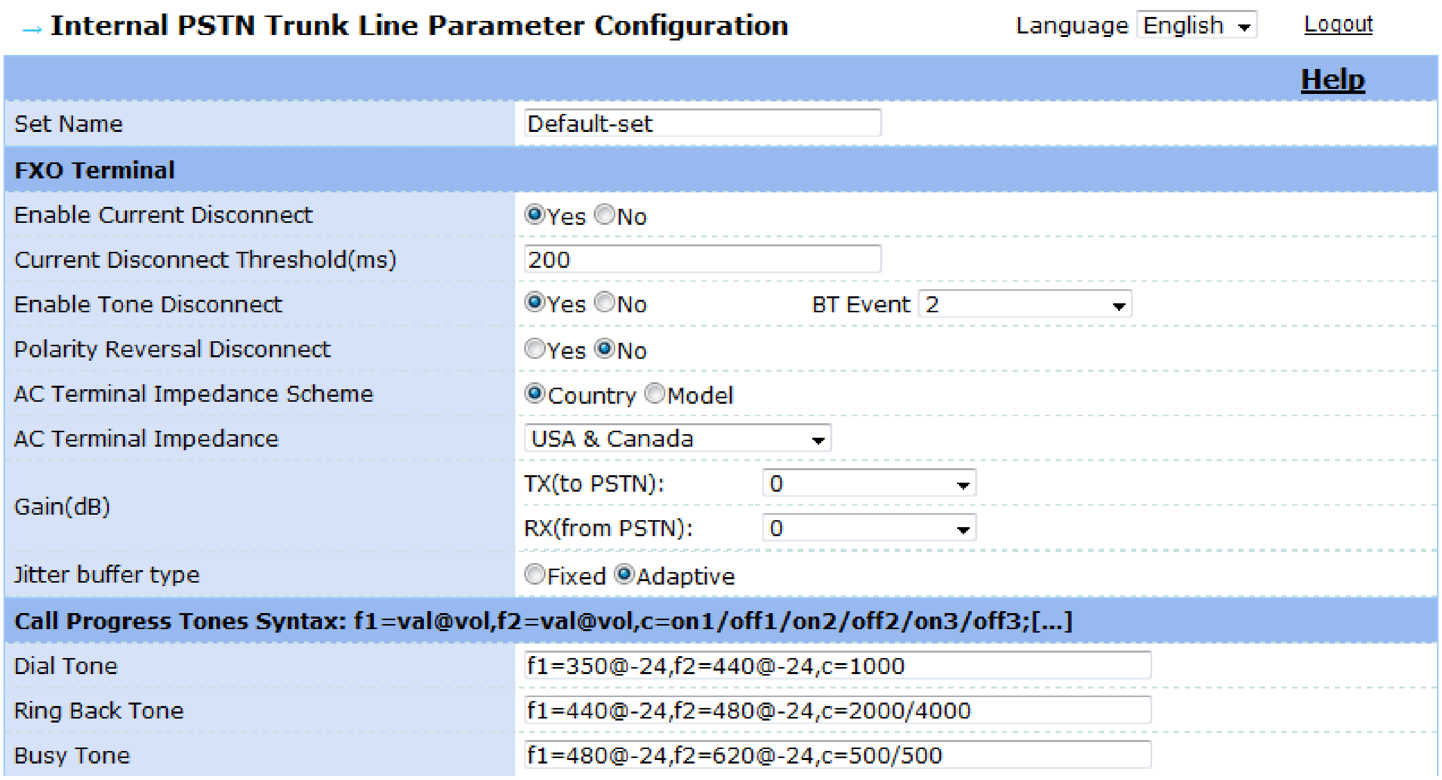
3. Trunk Configuration

PSTN trunks, SIP trunks and SIP-based PSTN trunks (extended via Grandstream FXO Gateways) can all be configured on the GXE502X in the **Trunk/Phone Lines** menu.
The FXO and FXS port/line electric specification parameters and settings can be configured in this area as well. All of these will allow users of the GXE502X to make calls anywhere through multiple trunk types.

- Internal PSTN Trunk Line Configuration (via FXO ports)

The Internal PSTN Trunk Line Configuration page lets your configure the FXO ports on the back of the GXE502X to make and receive calls to and from the PSTN.

Compatibility with different PSTN lines can be achieved by adjusting the related parameters on the Internal PSTN Trunk Line Parameter page. The default settings should function in most cases. Correct technical data must be obtained from local Telco carriers and implemented on the GXE502X before the lines will function. This data depends on the regional PSTN line characteristics.



PSTN Device List

Starting on version 1.0.1.32 the GXE support different line parameters assigned to each FXO port. Click on the Add button to start the process of adding FXO ports to your GXE.

Add FXO Device

- Trunk Name: This is the name used to identify this set of trunk(s). For example: carrier A, carrier B, Telco C, etc.
- Line: Please specify the FXO ports that you want to include in this profile either one by one or as a range. For example: 1, 2-5, 8.
- Prefix: This is the prefix that the GXE will use to create the outbound call routing profile so that it will know which trunk to grab for an outgoing call. When the user wants to make a call he will have to use this prefix to grab this particular set of lines.
- Call Routing Profile: This section will indicate the GXE where to route the calls coming into this PSTN trunk. You will see two boxes, one with the available list and another one with the selected call routing profiles. It is recommended to use either the play voice menu profile which will reproduce the auto-attendant menus according to the playing rules set, or to select the general inbound which will route the call accordingly to what the user has selected in the general inbound conditions and values.

Advanced settings:

Click on the Advanced link to show the advanced settings options, here you will find the PSTN Line Parameter Settings and the PSTN Port Parameter Assignments.

PSTN Line Parameter Sets: This section lets you specify the line disconnect signaling, impedance, as well as the caller ID standard and line dialing settings.
Set Name: This is the name that corresponds to this line or set of lines to be configured. For example a user could have 2 sets of PSTN lines from different providers with different settings.

- ***Enable Current Disconnect***: Default is Yes.
  This value should be used when the PSTN provider or local Telco Carries uses a line power drop to indicate call completion to the subscribing end point. In this case the FXO port of GXE502X will search for a power drop with the preconfigured time frame to disconnect call from a VoIP extension.
- ***Current Disconnect Threshold***:
  This is a preconfigured value of duration for a line power drop used by specific service providers or Telco Carriers. For example, with a configured value of 500ms, the device will ignore any random voltage drops on the line if the duration of such a drop is less than 500ms. The call will NOT be considered as terminated, but will clear a call if a drop is equal or more than 500ms. This is useful to prevent unnecessary call drops in some low quality PSTN lines. However inappropriate parameters may cause calls to drop unexpectedly or cause the line not to release as expected. Please consult with the local Telco carrier before changing this parameter.
- ***Enable Tone Disconnect***: Default is YES
  This feature causes a Busy Tone to be used as the FXO line disconnection signal when set to YES. It is used by some Telcos or enterprise POTS PBX.
- **BT Event or Busy Tone Event**: This is the number of cadence cycles to be detected in order to disconnect the call. In other words; in the cases where the PSTN provider signals a call termination by playing a busy tone the GXE will listen to how many times these tones are

played. Typically these tones are made up of 2 or 3 frequencies and are reproduced according to an on and off cadence. The "on" cycle means that the tone is played for certain duration of time, and the "off" cycle means that only silence is played for certain duration of time. In order to avoid false tone detections, the BT event can be configured so that the user can specify how many cadence cycles need to be detected in order for a call to be disconnected.

- ***Polarity Reversal Disconnect:***
  If set to "Yes", calls will be terminated after a polarity signal reversal. The default value is No.
- ***AC Terminal Impedance Scheme****:*
  Users can select one of two options for the AC termination scheme: **Country** (rough and easy) or the **Model** of related Impedance values (real and complex).
  Please make note that inappropriate parameters may cause echo or popping noises on the lines.
  Please consult with your local Telco Carrier before configuring these fields.
- ***AC Terminal Impedance:***
  This depends on whether Country or Model is selected, either 17 Countries are selectable or users have to select the Impedance value used by the local PSTN carriers.
- ***Gain(dB):***
  This option allows the user to increase or decrease the power level of the signal (audio) sent or received on the FXO ports. Remember that we are using a logarithmic scale in this case and that increasing the level by 3dB will actually double the power level sent to the PSTN. In a similar way, decreasing the power level by 3dB will cut in half the power level of the signal received.
  Transmit (TX) to the PSTN: This controls the power level sent out to the PSTN.
  Receive (RX) from the PSTN: This controls the power level received from the PSTN.
- ***Jitter Buffer Type:***
  This option allows the user to toggle between an adaptive or fixed jitter buffer.
  This will help in eliminating jitter in the communications by buffering some packets before transmitting them.

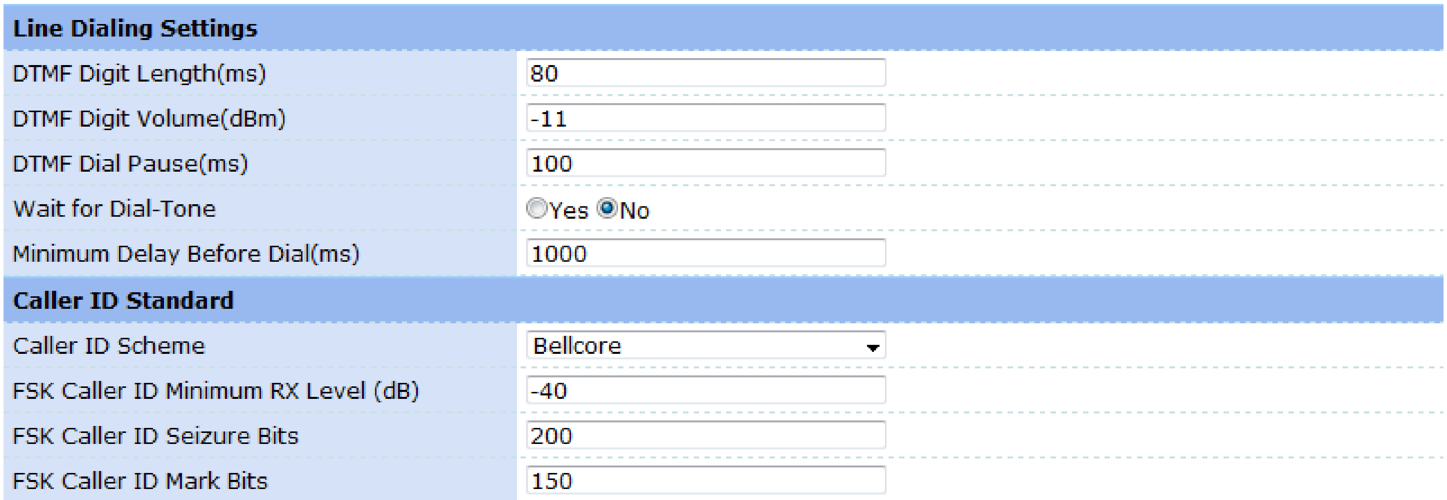
Call Progress Tones: This feature lets you configure tones to match the FXO line regional settings.

In certain countries, the carriers or central office will send special tones to indicate certain events on the PSTN side. For example, when a call is disconnected from the remote side the other side will hear a busy tone indicating that the call is terminated. Also, a dial-tone is used by the central office to signal that it is ready to receive digits and process the call. Users can pre-configure these tones in the GXE502X. However users must know the correct frequency values and cadences of these tones from the PSTN carrier. Wrong parameters may cause the GXE502X to malfunction due to incorrect tone settings.
In other words, the user MUST configure the call progress tones in order for the GXE to work properly. For more information about troubleshooting this please refer to the HELP section under this menu on the GXE's web UI.

Here is an example for the syntax for a busy tone in the U.S.A:

Syntax: **f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3; [...]**
(Note: freq: 0 - 4000Hz; vol: -30 - 0dBm; c: (cadence) in msec, several cadences allowed)

Default US Busy Tone: f1=480@-24,f2=620@-24,c=500/500)



Line Dialing Settings: This feature adjusts line dialing properties to adhere to the PSTN line requirements.

- ***DTMF Digit Length:*** The DTMF Digit Length is the play time of DTMF in milliseconds for each digit dialing out from the VOIP side to PSTN
- ***DTMF Digit Volume***: The digit output power in dBm.
- ***DTMF Dial Pause:*** Dial pause is the time between 2 digits when dialing out through the PSTN.
- ***Wait for Dial-Tone:*** If set to Yes, the device will first obtain a PSTN line and wait for a dial tone from the central office. After obtaining the dial tone, the digits dialed will be sent to the central office**.**
- ***Minimum Delay Before Dial:*** This parameter lets users configure a variable delay before dialing the first digit into the PSTN.

Caller ID Standard: This feature configures caller ID handling to match local PSTN settings.

- ***Caller ID scheme:*** This feature lets you select the model used by local PSTN provider. Please check with local PSTN carrier for configuration information. Current supporting including: Bellcore, ETSI_RING, ETSI_TAS, DTMF or NTT
- ***FSK Caller ID Minimum RX Level:*** This is an adjustable threshold value for the Caller ID signal strength received to help the device to recognize Caller ID from different networks. (-96 - 0dB. Default -40dB)
- ***FSK Caller ID Seizure Bits***: This field lets you choose a value of alternating bits (zeros and ones) used for CID channel signaling. The Default is 200.
- ***FSK Caller ID Mark Bits***: This feature lets you choose a value for the frequency modulated signal of alternating bits (zeros and ones) used for synchronization. Default 150.

Automatic Detection:

This section will be of great help to the user in order to determine the impedance, call progress tones and current disconnect settings on the FXO ports of the GXE.

Remember that the impedance on the FXO ports must be as close as possible to the impedance on your PSTN lines in order to reduce echo. If we do not match the impedance then we will have problems with echo on calls, calls getting dropped and faxes not getting received.

Call progress tones will tell the GXE when the line has dial tone, when the line is getting ringback tone and when to disconnect the line based on the busy tone heard at the end of a conversation. These tones vary from country to country and the user must configure them according to the geographical location where the GXE will be installed.

Current disconnect is a method used in a few countries, including the USA, that removed the current from the line for a split second following the end of a conversation. This method is used to determine when the line has been disconnected.

If the user ignores these settings the GXE may not work properly. No country in the world should be using current disconnect and tone disconnect simultaneously, but these settings depend completely on the central office. Impedance on the PSTN lines could vary from one to another as the physical medium is susceptible to many changes.

Before starting the auto detection please make sure that the wires in your PSTN line are in the correct order and that they work properly connected to an analog phone directly. You will need to know the PSTN numbers corresponding to each wire.

Caller FXO (detected FXO): This will be the FXO port that we will detect the values for. Please insert the line number (FXO port in the back of the GXE) and the PSTN line number for it. In other words if we have a PSTN line connected to FXO port 1 with the phone number 469-241-0100. We should type in 1 for the line number and 4692410100 for the PSTN number.
The important thing is to enter the PSTN numbers as if they were to dial each other locally (because they will). So you do not need to include country codes or area codes if your PSTN provider does not require you to input them. Do not include any dashes.

Called FXO: This is the FXO port that the caller FXO will call in order to run the tests. Simply input the FXO port number and the PSTN number as explained above.

You can only test for one parameter at a time!
This means that you can only test for impedance or current disconnect or tone disconnect at a single time.
Once you start the test all the services on the GXE will be stopped, and the user will have to reboot the GXE for the services to restart.

Select only one of the tests and then click on start detect. The auto detection process will take a few minutes; once it is done it will pop up a window with the test results. You must make a note of these results and then apply them yourself manually after reboot. In other words the results do not take place automatically!

- Internal Phone/Fax Port:

In the Internal Phone/Fax Port section, the electrical and DTMF signaling settings of the two FXS ports on the back of the GXE502X can be configured for compatibility with local analog phone/fax devices. The extension numbers for these ports are usually set in the Express Setup page.

- ***DTMF Transport Type***: This field is used to specify the method of sending DTMF digits to the VoIP side. There are three possible options: in-band (audio), via RTP (RFC2833), or via SIP INFO. It is possible to select several methods. For example: ch1:T=<2833/audio/signal>;ch2:T=<2833/audio/signal> . Default: ch1-2:T=2833.
- ***SLIC Setting***: *subscriber line interfacing circuit* configuration. This field should be configured based on local Telco standard and analogue phone/fax type.
- ***Caller ID Scheme***: The following schemes are currently supported: Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF, SIN 227 – BT, & NTT Japan.
- ***Polarity Reversal***: If set to "Yes", the polarity will be reversed upon call establishment and termination; this is usually used for billing purposes. The Default is No.
- ***Current Disconnect***: If set to "Yes", this feature will use a momentarily voltage drop to signal remote party disconnects.
- ***Current Disconnect Duration***: A configurable time threshold where the FXS port will drop off voltage on the line to indicate to the local party that the call is disconnected from the remote side. (100-10000 ms. Default 100 ms)
- ***Hook Flash Timing***: The time period when the cradle is pressed (Hook Flash) to simulate a FLASH. Adjusting this time value correctly can prevent unwanted activation of the Flash/Hold and automatic ring-back. These values are important when adjusting the Hook Flash timing for FXS gateways registered to the GXE
- ***Call Routing Profile***: The user can select here which call routing profiles will be allowed on these extensions for outgoing call purposes. By default the Internal Call and General Inbound are added automatically, the user can later restrict them to only Internal Calls for example by removing the General Outbound profile.
- **Voice mail storage time**: This is the period of time in days that the GXE-502X will keep you voice/fax mail messages stored in flash memory. After the time has passed the GXE-502X will delete the messages automatically to save space in memory. Default 30 days.
- **Entry voicemail option**: This will select in between 3 options for accessing voicemail from the user's extension using the feature code for voice mail.
  Extension and password: after dialing the feature, code the user will be prompted to enter both the extension number and the password to access the extension's voicemail.
  Password only: after dialing the feature code, the user will be prompted to enter his password only to listen to his voicemails
  Direct Call: After dialing the feature code, the user will start listening his voicemails bypassing all passwords.

DTMF Transport Type	ch1-2:T=2833
SLIC Settings	USA
Caller ID Scheme	Bellcore/Telcordia
Polarity Reversal	Yes / No (Reverse polarity upon call setup and termination)
Current Disconnect	Yes / No
Current Disconnect Duration(ms)	100
Hook Flash Timing(ms)	Min 100 (100-2000), Max 500 (100-2000)
PORT1:	
Extension	7990
Call Routing Profile	**Available List**: General Inbound, PlayVoiceMenu, International → ← **Selected List**: Internal Call, General Outbound
Authorization Profile	Default Authority
Voice mail storage time	30 (In days)
Entry voicemail option	Extension and password
PORT2:	
Extension	7991
Call Routing Profile	**Available List**: General Inbound, PlayVoiceMenu, International → ← **Selected List**: Internal Call, General Outbound
Authorization Profile	Default Authority
Voice mail storage time	30 (In days)

When finished, click on the **Submit** button to save your changes.

- SIP trunk

SIP trunks can be viewed, created, or modified on the SIP Trunk Configuration page. All configured SIP trunks as well as their details and current status are displayed. Please see the following detailed explanations of the fields on the SIP Trunk Configuration page:

SIP Trunk Language English Logout

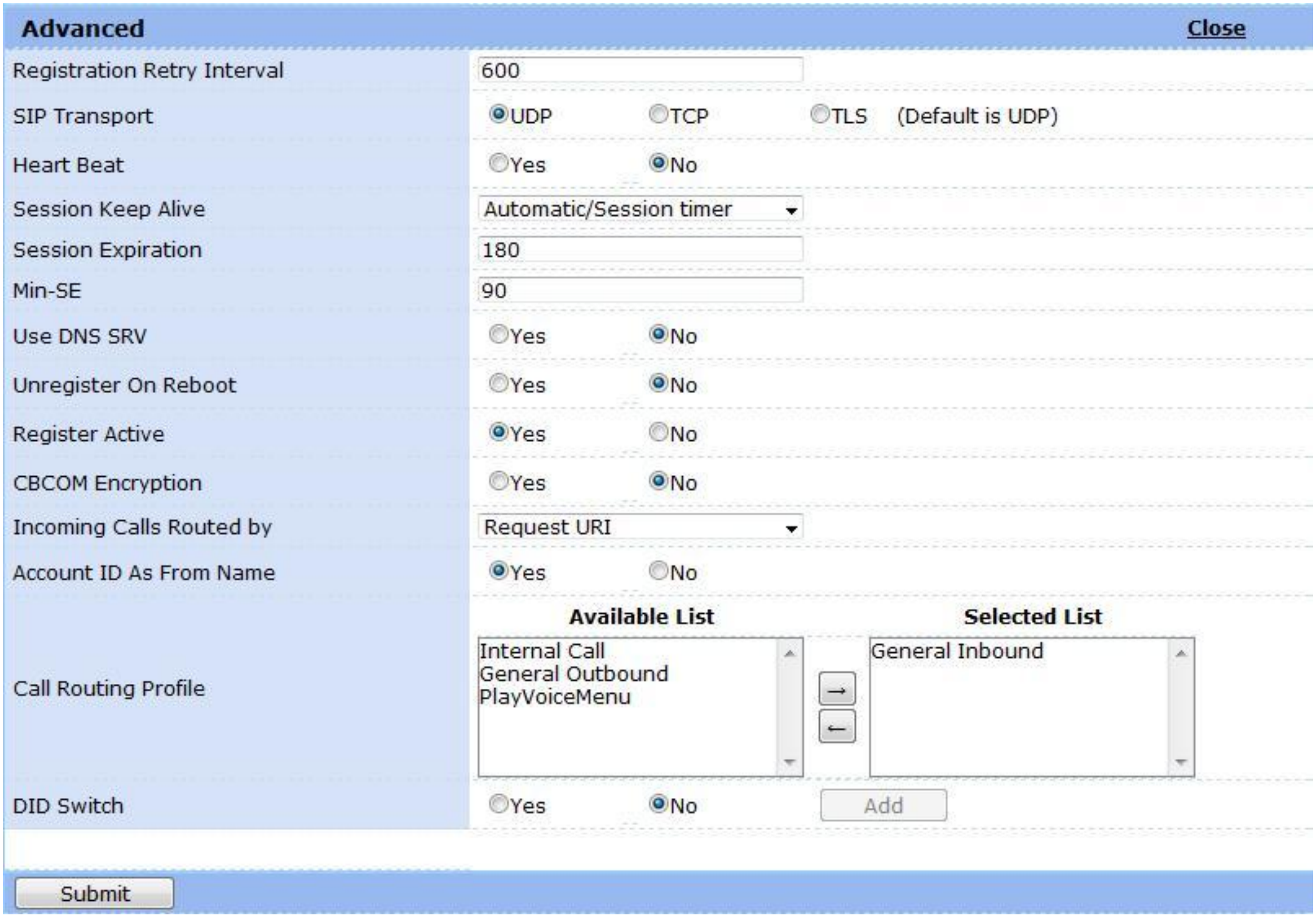
All	Name	SIP Server URL	Account ID	Max Concurrent Calls	Current Active Sessions	Status
☐	Trunk 1	trunk1.grandstream.com		8	0	Disconnected
☐	Trunk 2	trunk2.grandstream.com		8	0	Disconnected

Delete Add

One Button Provisioning Click here to get related SIP Trunking Service Provider Information and Promotional Offers!

- ***Add***: Clicking on the **Add** button will display the Add SIP Trunk page. This page allows users to enter the SIP trunk account and registration information.

Trunk Active	(•) Enable ( ) Disable
SIP Server URL	
Outbound Proxy URL	
Account Name	
Account ID	
Authenticate ID	admin
Password	••••••••
Max Concurrent Calls Allowed	8
Dial Prefix	
Advanced	**Close**
Registration Retry Interval	600
SIP Transport	(•) UDP ( ) TCP ( ) TLS (Default is UDP)
Heart Beat	( ) Yes (•) No
Session Keep Alive	Automatic/Session timer
Session Expiration	180
Min-SE	90
Use DNS SRV	( ) Yes (•) No
Unregister On Reboot	( ) Yes (•) No
Register Active	(•) Yes ( ) No
CBCOM Encryption	( ) Yes (•) No
Incoming Calls Routed by	Request URI
ReInvite Delay	0 (In seconds)
Account ID As From Name	(•) Yes ( ) No
Enable Video	(•) Yes ( ) No
Call Routing Profile	**Available List**: Internal Call, General Outbound, PlayVoiceMenu, International, Call Center LA, Call Center Boston → ← **Selected List**: General Inbound
DID Switch	( ) Yes (•) No Add



- *Trunk Name:* Enter a descriptive name in this field to identify the configured trunk.
- *Trunk Active:* Set to *enable* to activate the SIP trunk to be configured.
- *SIP Server URL:* The SIP server IP or URL provided by your SIP trunk service provider.
- *Outbound Proxy URL:* The Outbound Proxy or Session Border Controller IP or URL provided by the SIP trunk service provider.
- *Account Name:* Lets you specify a name to identify the SIP trunk service provider.
- *Account ID:* The Account ID provided by your service provider, usually the VoIP line/trunk number.
- *Authenticate ID:* The Authenticate ID used by your VoIP or SIP trunk service provider.
- *Password:* The password of the account provided by your service provider.
- *Max. Concurrent Calls Allowed:* The maximum number of concurrent calls allowed by your SIP trunk service provider. This depends on whether dynamic or static SIP trunks are used. The number can range from 1 to 20 given that there is enough available bandwidth on the physical link pipe.
- *Dial Prefix:* Sets the *Dial Prefix* digit(s) required to make an outbound call through this trunk. When the call is sent out via this trunk, the dial prefix digits will be removed. By default these digits will automatically create a new digit map on the General Outbound profile. The new digit map will include dial prefix followed by a dot (.); these dial prefix will be removed by the digit manipulation field and routed according to the trunk value. For more information please refer to the call routing section.

- *Registration Retry Interval:* The expiration time interval for the GXE502X to retry or update registration. This is usually regulated by the SIP server configured here. Reducing this time interval will keep registration more synchronized with the SIP server but with the cost of more registration packets sent to server. The default value is 600 seconds or 10 minutes.
- *SIP Transport:* This will be the transport method used by the SIP Trunk. This method is given by the SIP trunk provider. The options are UDP (default), TCP or TLS.
- *Heart Beat:* if set to Yes, the GXE502X will send a *SIP OPTIONS Request* frequently to check whether the SIP Proxy is still online. If the Proxy does not support by answering with a SIP 2*00 OK* then the GXE502X will consider this trunk as disconnected. This feature is also used to keep a UDP port open when the GXE502X is behind a NAT firewall.
- *Session Keep Alive:* This field allows users to enable or disable usage of the session timer. This is enabled by default to: Automatic/Session Timer.
- *Session Expiration:* The session timer enables SIP sessions to be periodically "refreshed" via a SIP request. If enabled, before the session interval expires, the GXE502X will send a SIP re-INVITE message to the SIP server, if this message is not replied the session will be terminated. This will avoid stalled sessions occupying the channel/bandwidth. Session Expiration is the timer (in seconds) at which the session is considered timed out if no successful session refresh transaction occurs beforehand. The default value is 180 seconds. Incorrect configurations could cause calls to get dropped.
- *Min-SE:* The minimum session expiration (in seconds). The default value is 90 seconds.
- *Use DNS SRV:* Default is No. Select "Yes" if the domain name of the SIP server or Outbound Proxy is compliant with RFC2782. The SIP trunk will not function properly if this field is not configured correctly.
- *Unregister on Reboot:* Default is No. If set to "Yes", The GXE502X will first send a registration request to remove all previous or multiple bindings by adding "*" in the SIP Contact Header. Please use this feature ONLY if the proxy supports removal binding requests. Otherwise the SIP trunk will not function properly.
- *Register Active:* This parameter controls whether the GXE502X needs to send REGISTER messages to the Proxy Server before making or receiving calls. The default setting is "Yes" (for most dynamic SIP trunks).

Note:
For security reasons: If this is set to "No" (for most static SIP trunks) any incoming request MUST match the SIP server IP or the FQDN resolution must match the sending IP address.

- *CBCOM Encryption:* If set to Yes, RTP will be encrypted as per the algorithms of CBCOM. The Default setting is No.
- *Incoming Calls Routed by:* If set to Requested-URI, then all calls are routed base on the Request URI field in SIP Invite. If set to To-URI, then calls will be routed base on To-URI, this is normally the case when To-URI matches the DID assigned to this SIP Trunk.
- *ReInvite Delay:* This option allows the user to delay the SIP ReInvite sent out by the GXE in order to finish establishing the call. The delay can be set from 0 to 4 seconds. This is very helpful in establishing interoperability with certain service providers or platforms who cannot handle ReInvites too quick.
- *Account ID as From Name:* If set Yes, then all outgoing calls from this SIP Trunk will use the Account ID (In this case the account name for SIP Registration) in From Header of the SIP Invite.

- *Enable Video:* This option allows the user to enable or disable video offering on the SDP body of the GXE's SIP replies. If your SIP trunk does not support video, please disable this option to avoid any interoperability problems.
- *Add PAI Header:* This option enables the insertion of the P-Asserted-Identity header in the SIP invite that is used on outgoing calls.
- *Call Routing Profile:* This section will indicate the GXE where to route the calls coming into this SIP trunk. You will see two boxes, one with the available list and another one with the selected call routing profiles. It is recommended to use either the play voice menu profile which will reproduce the auto-attendant menus according to the playing rules set; or to select the general inbound which will route the call accordingly to what the user has selected in the general inbound conditions and values.
- *DID Switch:* This field is used to route incoming calls when a DID (Direct Inward Dialing #) is used instead of the account ID. It is very helpful when there are several DIDs related to the same (static) trunk provided by your SIP Trunk Service Provider.

Note:
All SIP requests are verified by the GXE502X so the User Part of the SIP URI has to match the account ID or the DIDs. For outgoing calls when DID switch is enabled, the DID number in the list will be used in the From, Contact and PAI (P-Asserted-Identity) value.
If a prepend prefix is configured, it will also be used as prefix for the DIDs.
For example, if the DID is +16175669300 and for outgoing calls the service provider requires "+", then configure the prepend prefix to "+" and add the DID 16175669300. Then +16175669300 will be used to match both incoming calls and also outgoing calls.

Click the **Submit** button to add the trunk.

- ***Modify:*** Click the SIP trunk name to modify the trunk's parameters. The SIP trunk details page will be displayed, allowing users to modify all of the SIP trunk's settings. When done, click the **Submit** button to save all the changes.
- ***Delete:*** Click the **Delete** button on the bottom of the row to delete the SIP trunk. Users will be prompted for confirmation via a dialog box; click **OK** to confirm or **Cancel** to go back.

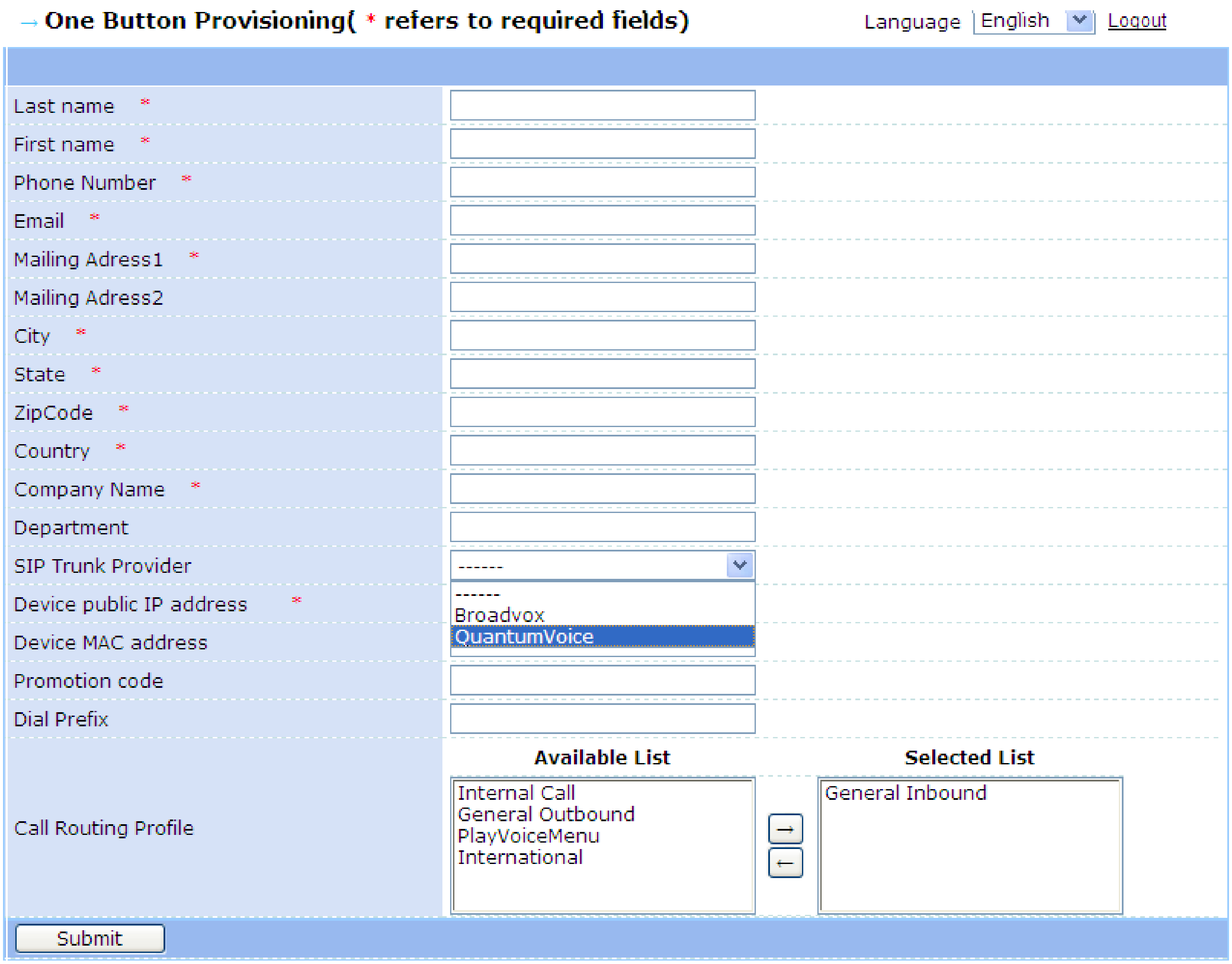
One Button Provisioning

"One-Button-Provisioning" for SIP Trunk provisioning allows the user to provision Grandstream Certified SIP Trunk Providers with just a few steps:

- Fill in the contact information
- Select the certified SIP Trunk service provider
- Click on "submit" button to activate the service

Note:

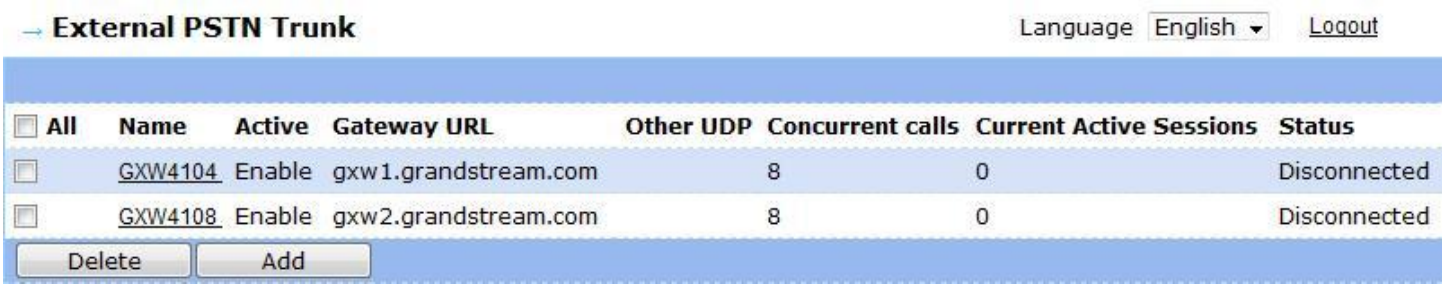
- The Device Public IP Address: This is for some SIP Trunk Service Provider to provision service base on Static IP, GXE5000 will automatically detect it and fill it once WAN port IP is detected.
- Dial Prefix: If you want to use a prefix for dialing out, in this case, no Call Routing Profile needs to be configured.
- Call Routing Profile: This is the same as all other Call Routing Profile configuration.



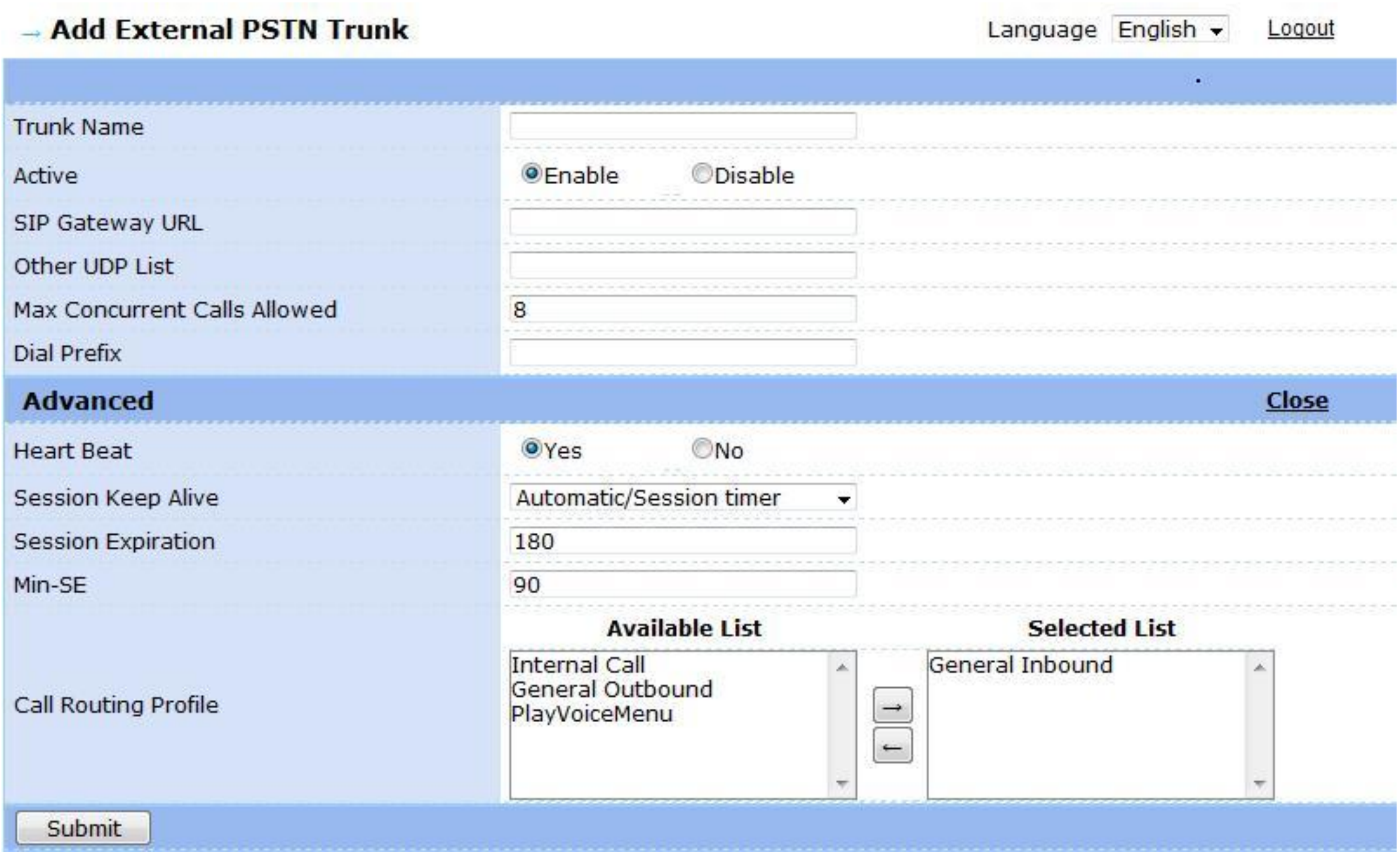
- External PSTN Trunk Line

External (Grandstream) PSTN Gateways can be added to expand the GXE502X's PSTN trunks on the External PSTN Trunk Line Configuration page. This is very useful when we want to have access to PSTN lines in remote places.

- **External PSTN Trunk Line:** When clicked, the external PSTN trunk configuration page is shown. This page allows users to administrate SIP peer connections via external PSTN trunk gateways.



- ***Add***: Clicking on the **Add** button will load the Add External PSTN Trunk Line page:



  - *Trunk Name:* A name to identify the configured trunk. Ex., A name for a Grandstream FXO Gateway could be GXW4104 or GXW4108.
  - *Active:* This allows users to turn the trunk ON or OFF.
  - *SIP Gateway URL:* The IP address or FQDN domain name of the external PSTN Gateway.
  - Other UDP list: Please configure this field if you expect the GXE to receive SIP messages on additional ports (other than 5060) from your SIP gateway URL or IP.:
    For example, If a GXW4104 is used, each FXO port will use a UDP port: port1=5060, port2=5062, port3=5064, port=5066; therefore, user needs to add *Other UDP List* using the following values: *5062; 5064; 5066*. In the case of an 8 port FXO gateway (GXW 4108) the list would be 5060;5062;5064;5066;5068;5070;5072;5074.

- *Max Concurrent Calls Allowed:* The maximum amount of concurrent calls that are allowed on the gateway. This is limited by the amount of FXO ports on the gateway or the bandwidth of the link in between the GXE and the FXO gateway.
- *Dial Prefix:* This sets the *Dial Prefix* digit(s) required for outbound calls to be routed through the trunk. When the call is sent out via this trunk, the dial prefix digits will be removed.

Advanced

- *Heart Beat:* If set to Yes, the GXE502X will send a *SIP OPTIONS Request* periodically to check whether the Gateway is still online. If the Gateway does not respond by answering with a SIP 2*00 OK* then the GXE502X will consider this trunk as disconnected. This feature is also used to keep UDP ports open when the GXE502X is behind a NAT firewall.
- *Session Keep Alive:* This field allows users to enable or disable the usage of the session timer. This is enabled by default to: Automatic/Session Timer. The session timer enables SIP sessions to be periodically "refreshed" via a SIP request. When enabled, before the session interval expires, the GXE502X will send a SIP re-INVITE message to the SIP server. If there is no reply to this message the session will be terminated. This will avoid stalled sessions occupying the channel/bandwidth.
- *Session Expiration:* The Session Expiration is the timer (in seconds) at which the session is considered timed out if no successful session refresh transaction occurs beforehand. The default value is 180 seconds. Incorrect configurations could cause calls to be dropped or suspended.
- *Min-SE:* The minimum session expiration (in seconds). The default value is 90 seconds.
- *Call Routing Profile:* This section will indicate the GXE where to route the calls coming from this external PSTN trunk. You will see two boxes, one with the available list and another one with the selected call routing profiles. It is recommended to use either the play voice menu profile which will reproduce the auto-attendant menus according to the playing rules set; or to select the general inbound which will route the call accordingly to what the user has selected in the general inbound conditions and values.

Click the **Submit** button to add the trunk or **Cancel** to go back.

- ***Modify:*** Clicking on the external PSTN trunk name will load the trunk details page. This page is where users can modify all of the SIP trunk's settings. When done, click the **Submit** button to save all the changes.
- ***Delete:*** Click the **Delete** button on the bottom of the trunk status/details to delete the trunk. Users will be prompted for confirmation via a dialog box; click **OK** to confirm..

4. Conference Bridge

The GXE502X supports 2 (GXE5024) or 4 (GXE5028) password protected conference bridges that allow up to 12 (GXE5024) or 20 (GXE5028) simultaneous participants from PSTN trunks, SIP trunks or Internal Extensions.

The Conference Bridge/Room numbers are configured automatically by GXE502X during the Express Setup process; this is one of the reasons why it is important to always run the ExpressSetup process at the beginning. At the factory default settings the conference bridges do not have password protection. Users can change the extension numbers of the bridges and set a password to by selecting the **Conference Bridge** menu.

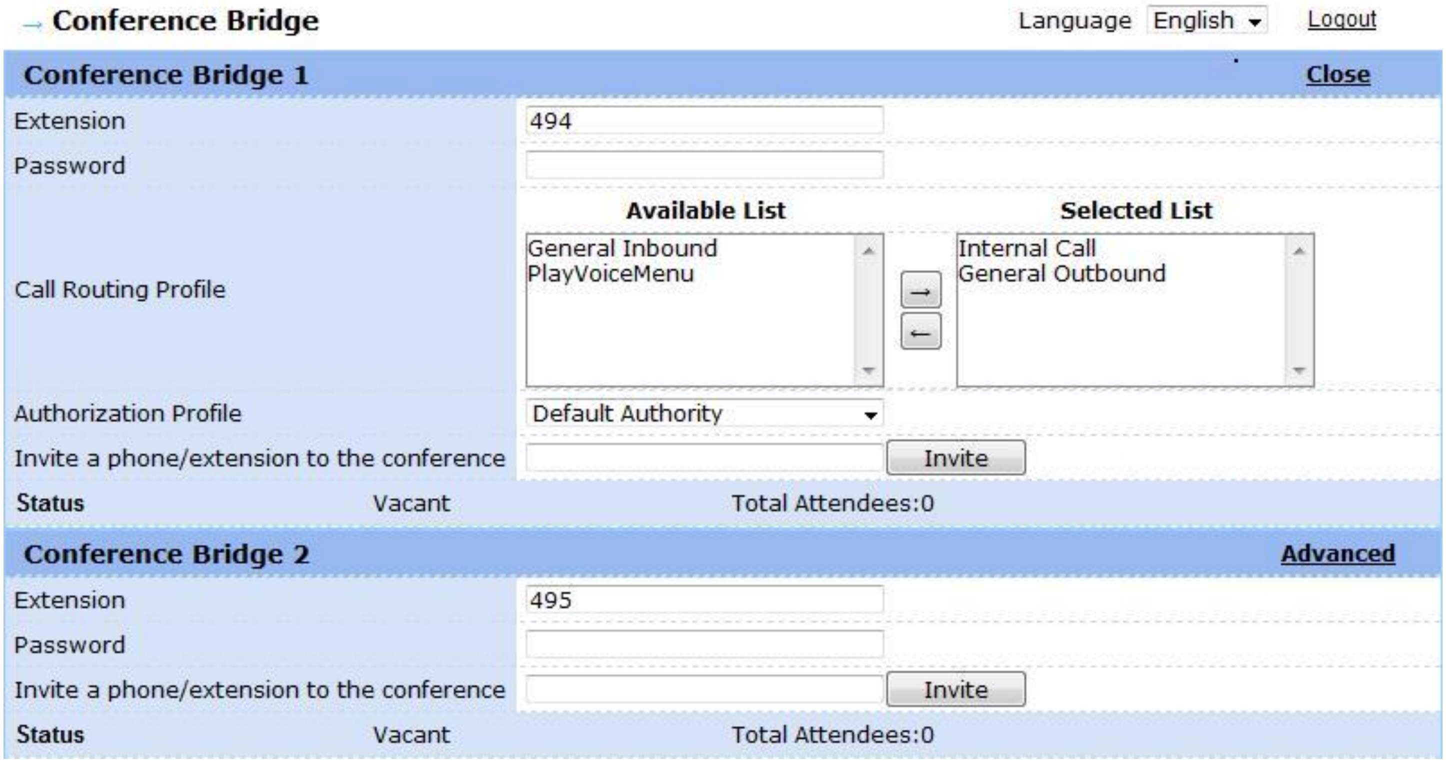
- Renaming the conference bridge extension numbers and passwords:
  To rename an extension number on a conference bridge, enter the extension number (please make sure the number is not currently in use by a local extension) into the *Extension* field of a conference room. If you want to require users to enter a password before they can enter the room, input a numeric password in the *Password* field. When done, click the **Submit** button to save the changes. The changes will be written into the Flash memory.

- Invite a phone/extension to the conference:
  The administrator can invite and extension number or any number (PSTN or DID) to join the conference by simply typing in the extension number or the PSTN number as long as it follows the dial plan rules. The GXE will then display the total number of attendees, the people who are in the conference, the start time and where they are joining from. Additionally the administrator can mute some of the attendees (moderator mode) or he can kick them out by pressing the delete button.

- Call Routing Profile:
  Since the conference bridges are a set of virtual extensions, the GXE needs to assign them a call routing profile. By default the Internal Call and General Outbound are automatically added so that the administrator can invite other extensions as well as outside phone numbers into the conference. Additionally the administrator can add extra call routing profiles to expand the ability of the conference bridge to invite people; so the conference bridge acts just as another extension in the GXE that has the ability to create a multi party conference, but it is restricted by its call routing profile just like all other extension.

- Authorization profile:
  Just like the other internal extensions in the GXE, the administrator can limit the authorization profile of each conference bridge according to the rules set in the call routing menu.

- Conference Room Status
  When the conference bridges are occupied, users can click the **Conference Bridge** menu to check the status of each conference room:

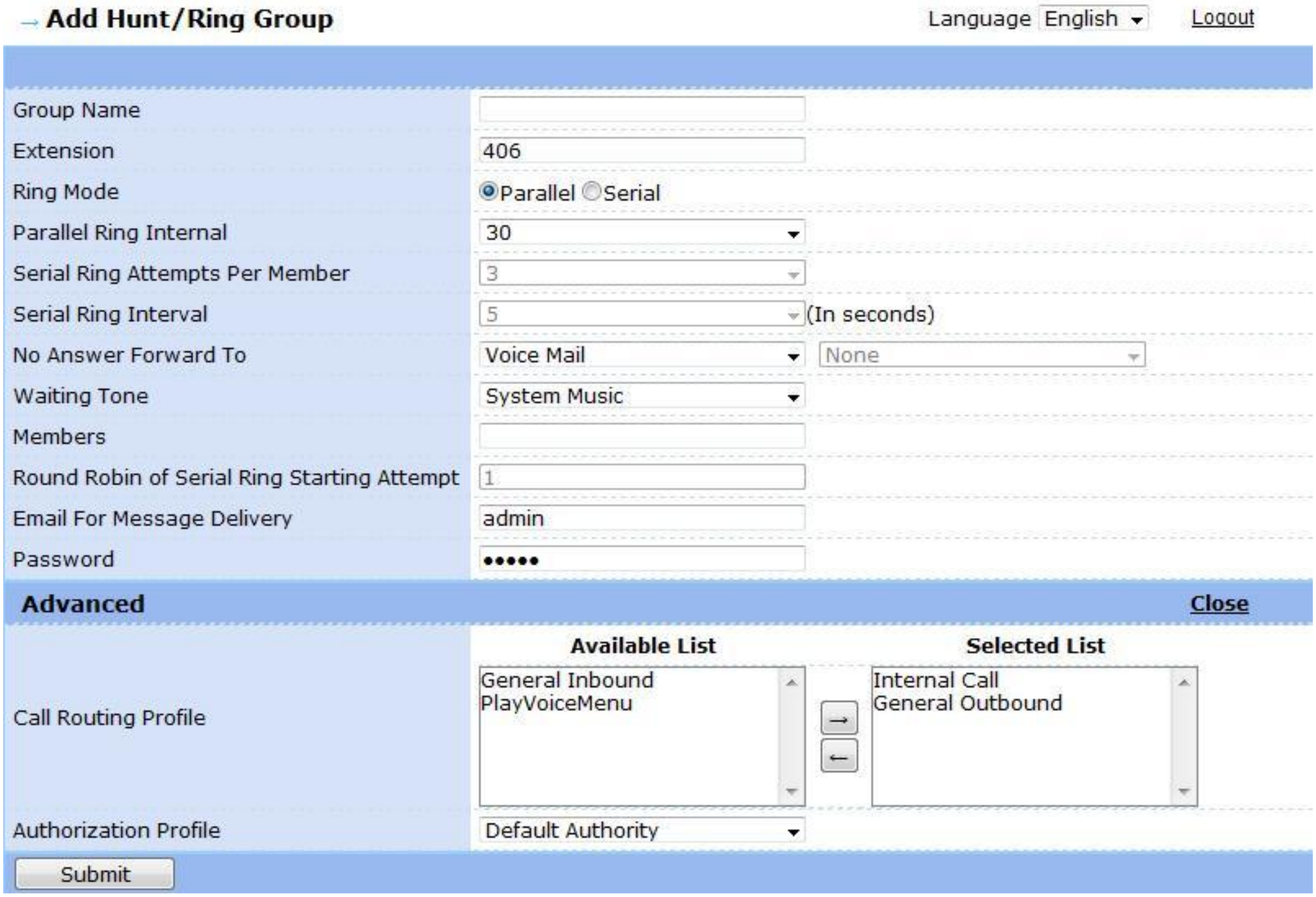


5. Hunt/Ring Group Configuration

Hunt/Ring groups can be configured on the GXE502X to balance the call traffic for multiple users and give callers a higher level of availability for incoming calls. Multiple ring methods and voicemail are supported.

- ***Hunt/Ring Group:*** Click the **Hunt/Ring Group** menu will display all of the configured hunt/ring groups. This page lets users add, modify, and delete ring groups.

- ***Add:*** Clicking on the **Add** button displays the Add Hunt/Ring Group page where users can configure the hunt/ring group settings:



- ***Group Name:*** Enter a name for the group you want to create. For example: customer service, sales, shipping, security, etc.
- ***Extension*****:** Enter an extension number for this ring group. The leading digit for this extension must correspond with the *Leading Digit of Extensions*. If you are an advanced user the leading digit can correspond to any of the leading digits of the profiles associated with Internal Calls under call routing.
- ***Ring Mode*****:** Select a ring method for this hunt/ring group. Parallel will ring all group members simultaneously, while Serial will ring members one at a time, starting from the first group member. A round-robin variation of Serial is also available, and is explained below. Setting this option to Serial will increase the time in which incoming calls are forwarded to voicemail compared to Parallel if there are more than 5 members in the group.
- ***Parallel Ring Interval:*** This is the time in seconds (from 20 to 180) that all the extensions in the hunt/ring group will ring simultaneously.
- ***Serial Ring Attempts Per Member*****:** This field lets you select the number of ring attempts to each group member before forwarding to the next member. Please make note that this field is available only if the Serial ring mode is enabled.
- ***Serial Ring Interval*****:** This field lets users select the ring time interval between ring group members (Serial only).
- ***No Answer Forward To:*** This field will allow the administrator to select where to route the call in case no one in the hunt/ring group was able to answer. It can be sent to another

Hunt/Ring group, an extension, a voice menu (auto-attendant) or to the hunt/ring group's voice mail box.

- ***Waiting Tone*:** Lets users choose either a normal ring-back tone or the system music to be played back to the ring group caller.
- ***Members*:** List the phone extensions of the members of the ring group, separated by semicolons.
- ***Round Robin of Serial Ring Starting Attempt*:** This allows the starting position of the ring group to increment for the next call to the group, instead of starting at the first member for all calls.

  Enter a number to specify the number of ring group members to be included in the round robin ring mode.
  To disable the round-robin mode, enter "1".
  To round-robin among all ring group members, enter the total number of ring group members listed.
  Example: - Ring group members: 6001;6002;6003;6004
  - Round Robin of Serial Ring Starting Attempt: 3

  The ring group incoming calls will round robin between 6001, 6002 and 6003. But 6004 will always be the last one to receive calls.
- ***Email for Message Delivery*[4]:** Enter the email address to deliver voicemail-to-email messages for this ring group.
- ***Password to Retrieve Voicemail:*** Set a password to retrieve this voicemail/fax mail box.

Advanced:

- ***Call Routing Profile:*** By default the Hunt/Ring group will get assigned the internal call and general outbound profiles. This is done so that the group can call all the extensions that are members or forward a call to any of the extensions when no one picks up.
- ***Authorization Profile:*** The administrator can assign a predefined authority profile to this extension so that they might require a password to perform certain calls
- **Voice mail storage time**: This is the period of time in days that the GXE-502X will keep the voice/fax mail messages stored in flash memory. After the time has passed the GXE-502X will delete the messages automatically to save space in memory. Default 30 days.

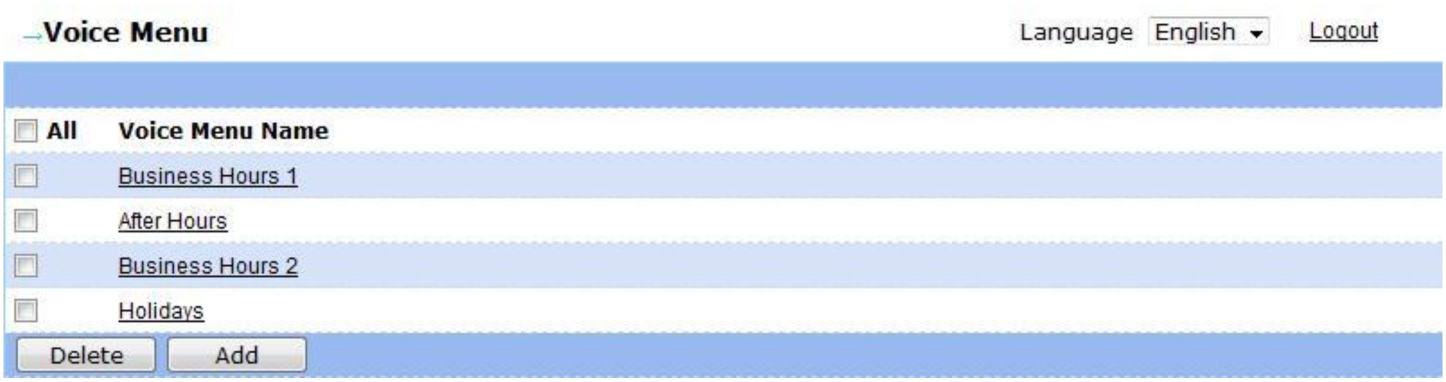
When finished, click the **Submit** button to save the changes.

- ***Modify:*** Click the extension number to select the ring group to be modified. This will load the hunt/ring group details page which allows users to modify all of the group's settings. When done, click the **Submit** button to save the changes.
- ***Delete:*** Click the **Delete** button on the bottom to delete the ring group. Users will be prompted for confirmation via a dialog box; c**lick OK** to confirm or **Cancel** to go back.

6. Auto-Attendant Configuration

[4] Note: Multiple email addresses can be entered, separated by ; .

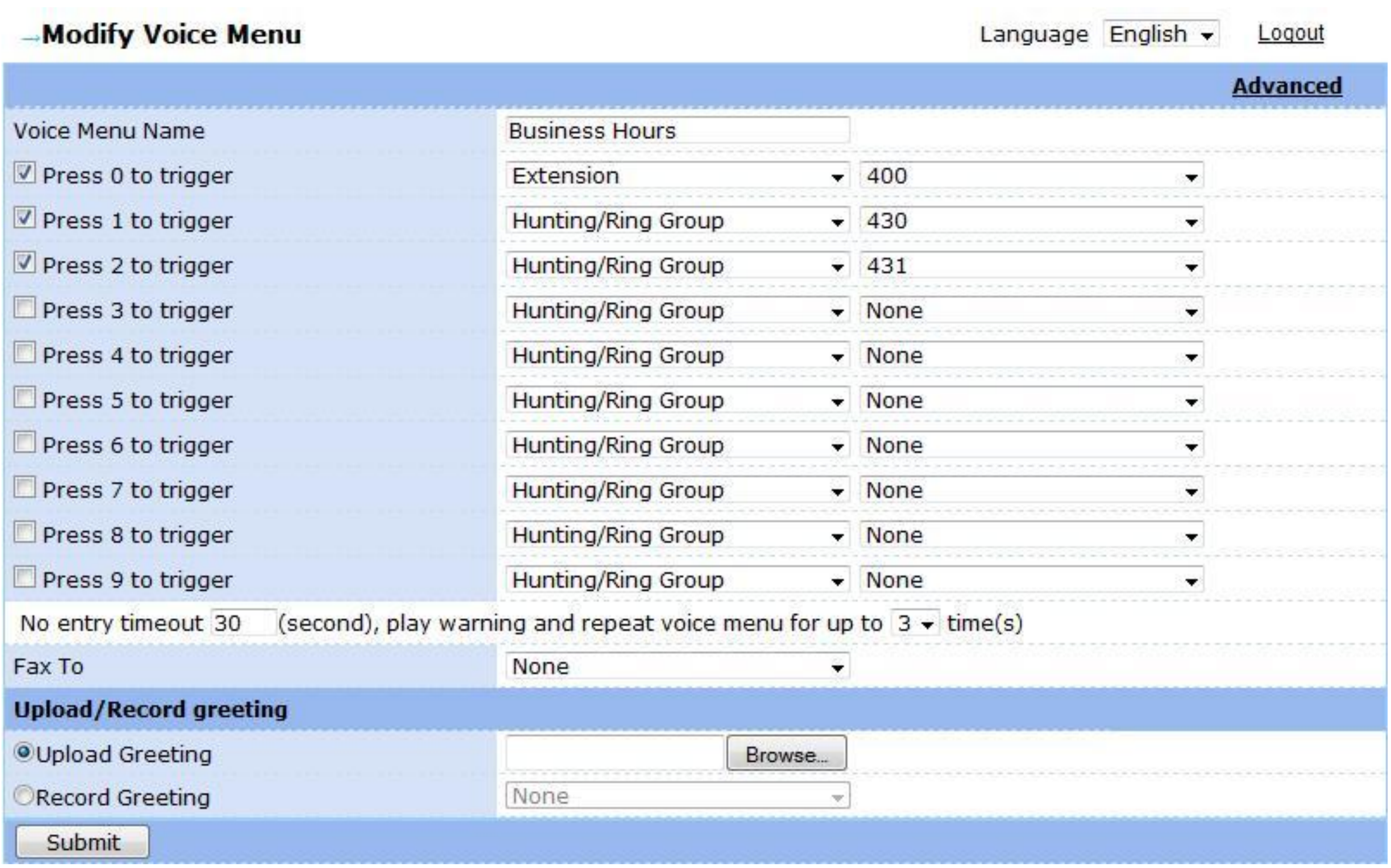
Incoming calls can be directed to the Auto-Attendant to provide immediate and professional service to callers and automatically route calls to intended parties. It is also possible to schedule different auto-attendants rules to be played based on the time, date and day of the week. The **Auto-Attendant** menu displays all configured auto-attendant rules.



- Voice Menu Configuration:

The following actions can be performed in the Voice Menu sub menu section. Voice Menu is used by the Auto-Attendant or IVRs (Interactive Voice Response System).

- ***Add:*** Click the **Add** button. The voice menu details page will be displayed, allowing users to configure the voice menu for their new auto-attendant:



- ***Voice Menu Name*****:** This field lets users enter a name for their auto-attendant.
- ***Press x to trigger*****:** Check the box next to each keypad digit to specify the flow of your auto attendant menu. Callers automatically have the option to enter internal extension numbers from the auto-attendant. To select the type of destination that calls will be forwarded to, select an option from the first drop-down box. Users can then select from available destinations of that type in the second drop-down box (if applicable).
- ***No entry time out*****:** This field lets the administrator to enter the minimum time callers have to enter a menu option before repeating the voice menu or exiting.
- ***Play warning and repeat voice menu for up to*****:** Select from the drop-down box the number of times for this auto-attendant (IVR) to be repeated when the caller does not enter any menu option, before exiting.
- ***Fax To*****:** This field allows the auto attendant to automatically route all incoming faxes to a specific extension.

The Auto Attendant Greeting voice prompt can be uploaded from a computer or recorded directly from a registered GXE extension:
- **Upload file:** Click the *Browse* button to search for voice prompt files on a local computer, select the file and click on the **Submit** button to add the file to auto-attendant. Please make sure that all uploaded files are converted via the correct tools
- **Record Prompt:** This feature lets the users specify a local extension to record their auto attendant greeting message. Clicking on the submit button will cause the selected extension to receive a call from GXE502X that will prompt the user to record the voice prompt.

Note: - Uploaded audio files must be in the following format: 8 KHz/16bit/ MONO .wav.
- It is highly recommended to use the GXE502X IVR/System Prompt Conversion Tool to convert the wav file before uploading it to GXE502X. Using this tool will generate a "ZIP" file that can be uploaded via the same field.
- Once the voice prompt is uploaded correctly, the message "Voice Menu File does NOT EXIST" will disappear and the user can preview the Voice menu by clicking the **Preview** button shown on the screen.

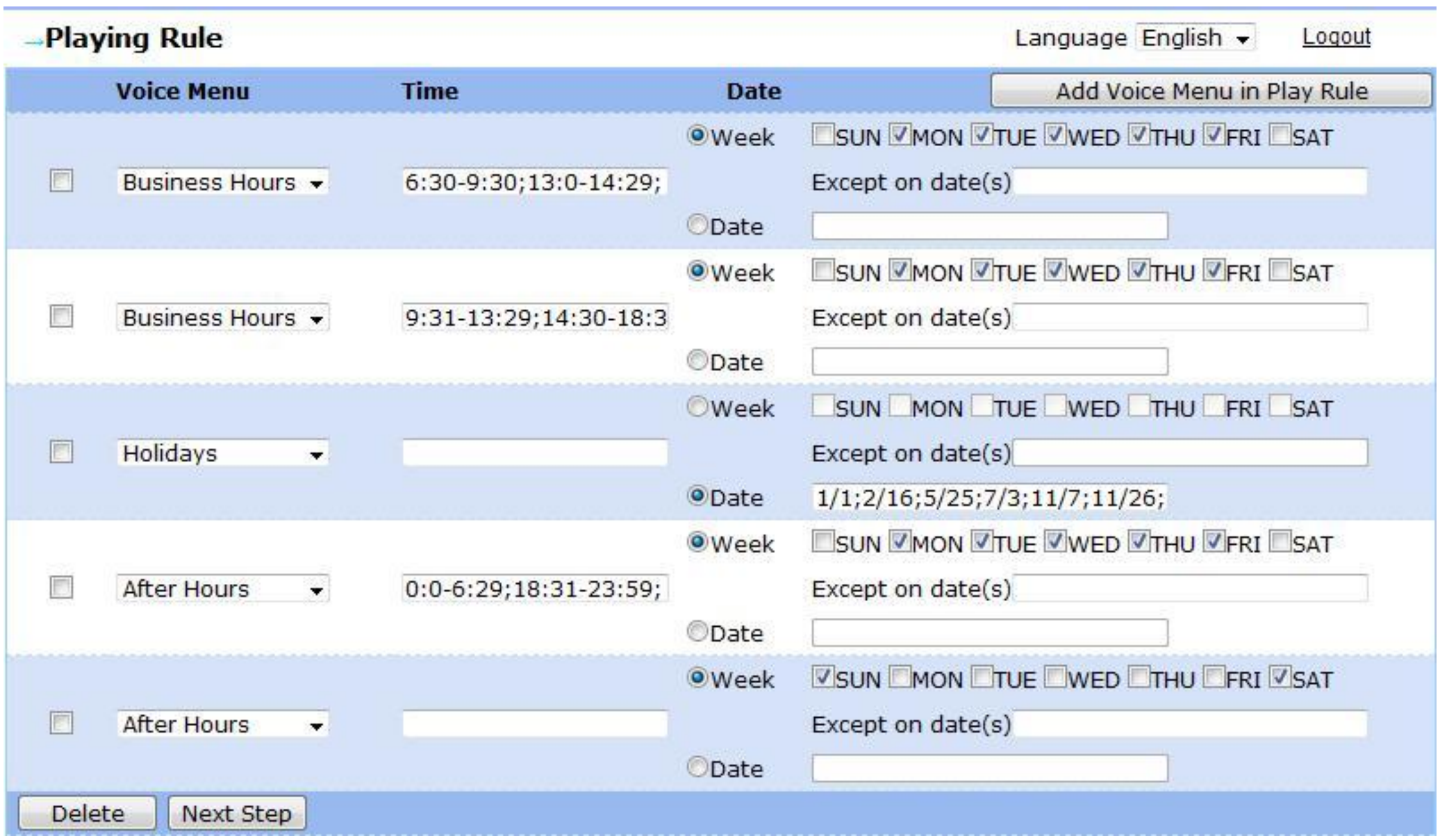
Advanced:

- **Call Routing Profile:** This section will determine which call routing profiles are accessible to the user once he reaches the IVR. This is particularly useful when we are trying to configure a DISA (Direct Inward System Access). Once the user reaches the IVR he can start dialing DTMF and reach extensions, or grab PSTN trunks and SIP trunks if allowed in this call routing profile. By default the auto-attendant will have the internal call profile selected. This is so that the caller can reach internal extensions, hunt groups, conference rooms, etc. Additionally the user can add other profiles to make it work as a DISA to provide access to SIP trunks and PSTN trunks. For example an employee could dial into the auto-attendant from his cell phone and use the DISA to make a long distance call through a SIP trunk without incurring in long distance charges on his cell phone. Remember to dial accordingly to the call routing profile rules set.
- **Authorization Profile:** This is very important to configure especially if we are allowing access to tolled trunks in the auto-attendant. Turn the authority profile to a password protected setting if you want users to enter the trunk password before the GXE allows them to dial through a tolled trunk.

- ***Modify:*** Click the **Modify** button to the right of the voice menu to select the auto-attendant/IVR to be modified. This will load the Modify Voice Menu page which allows users to re-configure the auto-attendant or IVR (Interactive Voice Response System)
- ***Delete:*** To remove a voice menu: Click the **Delete** button on the far right of the voice menu you wish to delete. Users will be prompted for confirmation via a dialog box; click **OK** to confirm or **Cancel** to go back.

- Playing Rules Configuration:

Playing Rules can be configured to dictate which auto-attendant is played based on user defined time and date conditions. The Playing Rules section displays all configured *Voice Menu* playing rules, and allows the following actions to be performed:



- ***Add Voice Menu in Play Rule*:** Clicking the **Add Voice Menu in Play Rule** button will display a new row which will allow users to configure the time conditions required to play a specific auto-attendant Voice Menu:

   - ***Voice Menu*:** This field lets users select the name of the auto-attendant voice menu that they wish to play from a drop-down box. These menus have to be created first under the Voice menu section of the Auto-attendant. For example the user could have created previously voice menus for business hours, after hours, holidays, etc. Now he will be able to tell the GXE when to play these menus.

- ***Time*: This field let you enter the time range to play the auto-attendant during the days it is set to play. For example if we have a business hours menu the range could be 08:00-17:00. Additionally we can create separate time ranges using semicolons: 08:00-12:00;13:00-18:00

> **Important:** Make sure that the current date and time are configured correctly under **System Settings.**

- ***Week***: This field lets users configure the auto-attendant voice menu to be played on specific days of the week. Click on the *Day of Week* radio button and check the boxes beside each day of the week that you would like the auto-attendant to be played on.

  Except on date(s): If you have configured the auto-attendant to be played based on selected days of the week, you may enter exceptions such as a holiday where you would like another auto-attendant to be played. To do this you must enter the exception dates in the blank text field.
- ***Date***: This field lets users configure the auto-attendant voice menu selected to the left to be played on a preset date or a range of dates. To specify by date, select the radio button beside *Date* and enter the date or a range of dates in the text field.

Next Step: Click on the Next Step button at the bottom of the page to indicate the GXE which incoming trunks will be routed according to the play rule that you just setup. This means that all incoming calls into the selected trunks will be directed to the IVR selected according to the date and time that you just specified in the previous section. Press the Finish button to apply these changes. All the incoming calls into the SIP trunk, PSTN trunk or External PSTN trunks that you have selected will be routed now into the IVR according to the time and date.

- ***Modify:*** Click***ing the Mo*****dify** button on right of the row will load the Modify Playing Ryles page. This page allows user to modify all of the voice menu's playing rules. When done, click the **Submit** button to save the changes or **Cancel** to go back.

- ***Delete:*** Click the **Delete** button on the far right of the row to delete unwanted rules. You will be prompted for confirmation via a dialog box; click **OK** to confirm or **Cancel** to go back.

7. Call Queues Configuration

Call Queues give users (i.e. call centers) an efficient means to have their calls answered in the order they were received to deliver top tier customer service.

- In-Queue Announcements Configuration

The In-Queue Announcements sub menu lets users configure the In-Queue Announcements for their group.

- **In-Queue Announcement Name:** This field allows users to enter a name (20 character limit) for the uploaded In-Queue Announcement file.

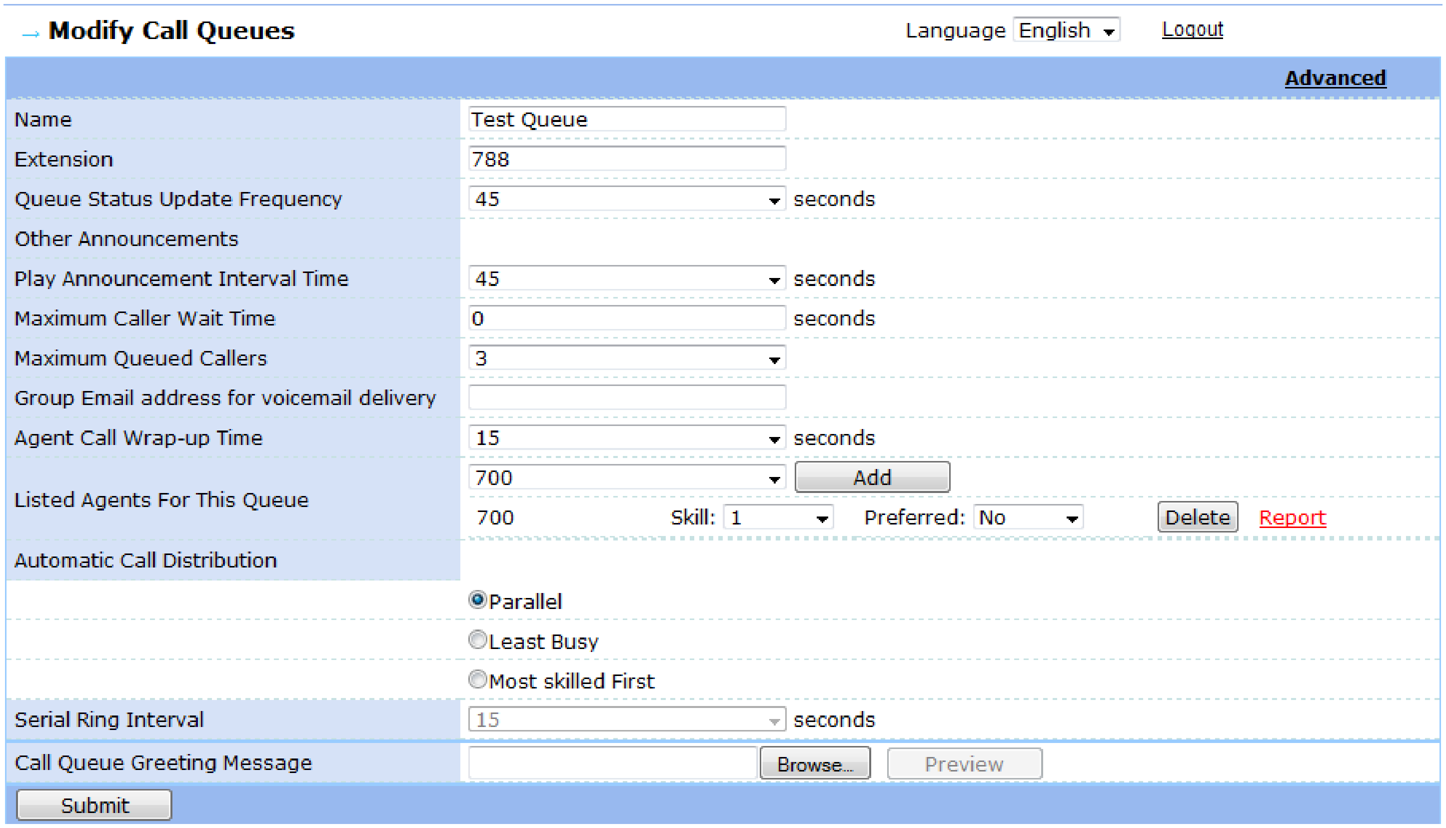
- **In-Queue Announcement File:** This field lets users specify the location of the audio file which will be uploaded. Users can type the path manually or click the *Browse…* button to navigate and select the file on their computer.

<u>Note:</u> - The In-Queue Announcement File must be in the following format: 8 KHz/16bit/MONO .wav or .ZIP (if you use the GXE502X IVR/System Prompt Conversion Tool).
- It is highly recommended to use the GXE502X IVR/System Prompt Conversion Tool to convert files to the .wav format before uploading them to the GXE502X.

<u>Note:</u> Agents must have the **"PUBLISH For Presence" feature** enabled in their phone's web UI. Having the Agent's extension registered on the phone side does not mean the call queue agent is online and ready to receive queued calls. Users can check the agent's status on the GXE502X Web UI (see image below).

- Call Queues Configuration

- ***Add:*** Click on the Add button to load the Add Call Queue configuration page. This is the main configuration page for call queues configured on the GXE502X.



- ***Name:*** Enter the name of the call queue here.
- ***Extension:*** Enter the extension of the call queue here.
- ***Queue Status Update Frequency:*** This determines how often callers will be updated on the status of the queue via an uploaded update message. This message will tell the caller what his position in the queue is.
- ***Other Announcements:*** This determines the frequency in which any other announcements that the user has added will be played to callers in the queue.
- ***Maximum Caller Wait Time:*** This field lets you set the maximum amount of time that callers will wait within the queue before being forwarded to voicemail.
- ***Maximum Queued Callers:*** This field allows users to set how many callers can be within the queue simultaneously.
- ***Group Email Address for Voicemail Delivery:*** Enter the email address where all voicemail for the queue/group will be delivered.
- ***Agent Call Wrap-Up Time:*** This setting allows users to specify the amount of wrap-up time an agent will have before receiving another call (time between two calls).
  For example, an agent may need 1 minute of wrap-up time to document a call.
- ***Listed Agents For This Queue:*** All configured agents for the queue will be displayed here. The supervisor can select the available agents from the drop-down box and click the **Add** button to add the agent to the queue.
- ***Skill:*** In this scale 9 is the most skilled agent and 1 is the least skilled agent

- ***Preferred:*** Since the agent can be included in more than queue, this option will put the preferred queue on top of the list displayed in the LCD screen of the GXP phone when the agent presses the sign in softkey.
- ***Report:*** The supervisor can click on the report link to the right of each agent's properties. This will bring up a new page which informs the supervisor about the agent's login status, answered calls, time calls were answered, customers helped, etc
- ***Automatic Call Distribution:*** This setting lets users configure, enable and disable skill-based call routing. If skill-based routing is enabled, users can configure the call to be routed to the least skilled or most skilled agent first. The skill level for each agent can be configured on the agent 's properties.
- ***Parallel:*** This feature is similar to that of hunt/ring groups as supervisors can configure the mode in which agents of the same skill level will ring.

  Parallel - All agents ring simultaneously.
  Least Busy - The least busy agent will ring first.
  Most skilled first - The most skilled agent will have his phone ring first
- ***Serial Ring Interval*:** If the serial ring mode is selected, users may set the ring time interval (in seconds) between call queue members.
- ***Call Queue Greeting Message:*** Upload an audio file for the queue greeting message. Users can type the path manually or click the *Browse* button to upload the file from your computer.

Note: - The Call Queue Greeting Message must be in the following format: 8 KHz/16bit/MONO .WAV or a ZIP file if you use the GXE502X IVR/System Prompt Conversion Tool.
- It's highly recommended to use the GXE502X IVR/System Prompt Conversion Tool to convert the wav file before uploading it to GXE502X.

- ***Delete:*** Click the **Delete** button on the far right of the unwanted Call Queue. Users will then be prompted for confirmation via a dialog box; click **OK** to confirm or **Cancel** to go back.
- ***Status:*** Click on the status button to the right of each call queue to get information about the agents that belong to this call queue. The page will display information like calls answered per agent, average time per call answered, missed calls, average waiting time, etc.

Note: If you are using Grandstream GXP phones (2020, 2010, 1200, 280 only) and the extension on these phones belong to any of the queue configured in the GXE, then the softkeys on the phones should allow you to log in or log out of any of the call queues that the extension is a member of.

8. System Configuration

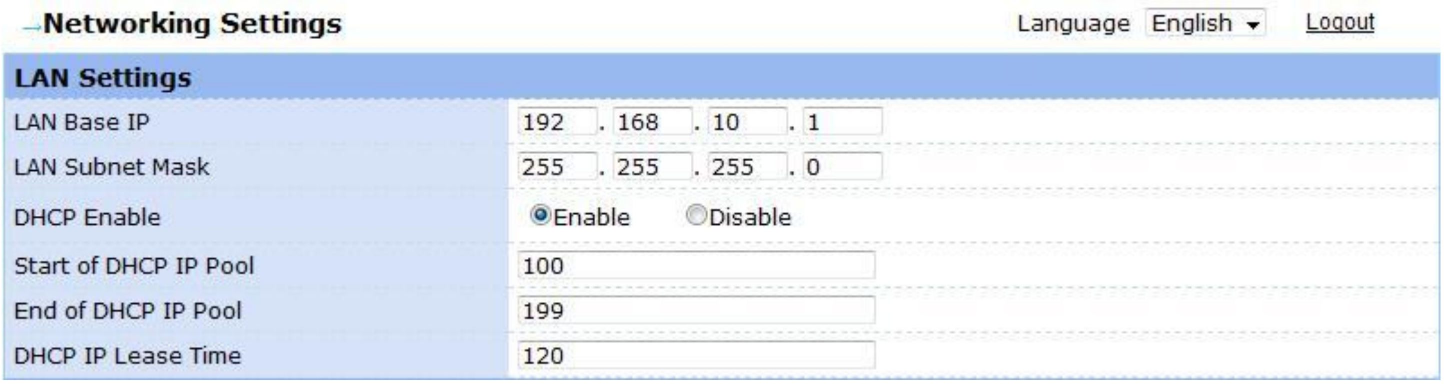
System configuration and administration can be configured via the **System Configuration** menu. The following sub menus can be used to manage different system configurations..

- Networking Setting

The *Networking Setting* sub menu allows users to configure the LAN-side IP address and DHCP server settings, as well as the WAN-side settings like IP address of the WAN port, Dynamic DNS and port-forwarding settings.

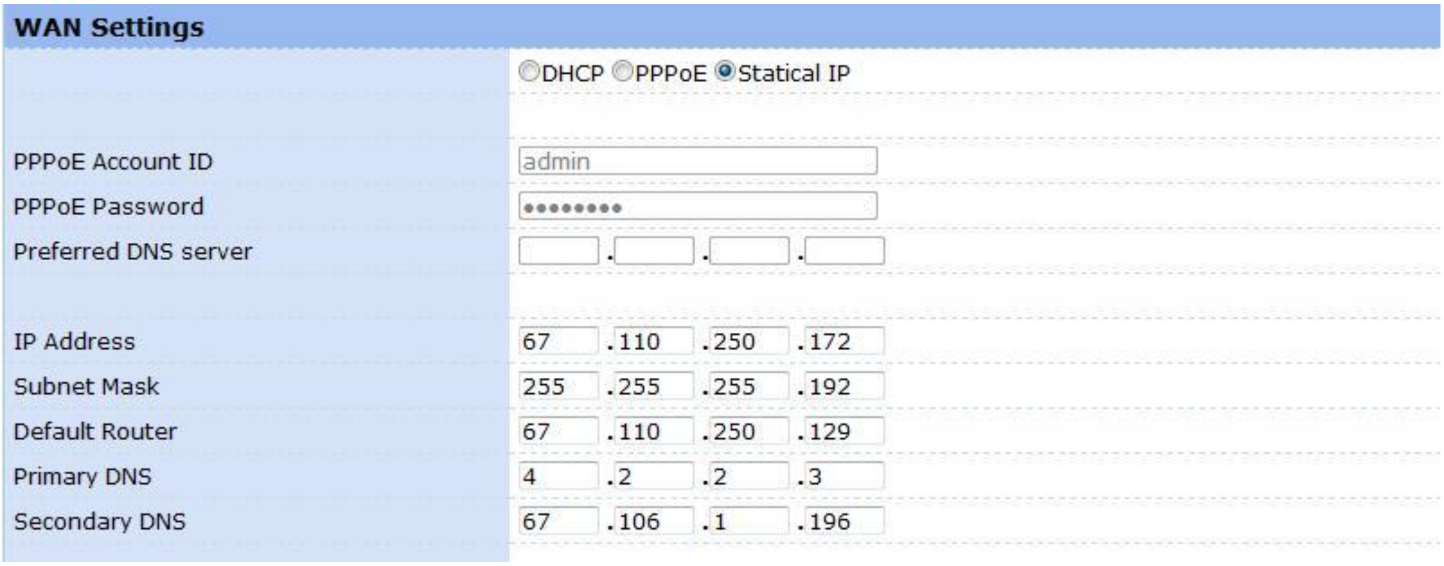
- **LAN Setting:**
  Set the LAN-port IP address by entering the IP address used as the *LAN Base IP*. Enter the subnet mask in the *LAN Subnet Mask* setting.

Note: Please be advised that this setting should be configured prior to auto-provisioning phones through the **Express Setup** menu. Otherwise the GXE502X will use an incorrect IP address for the SIP server after the LAN-port IP address changes.



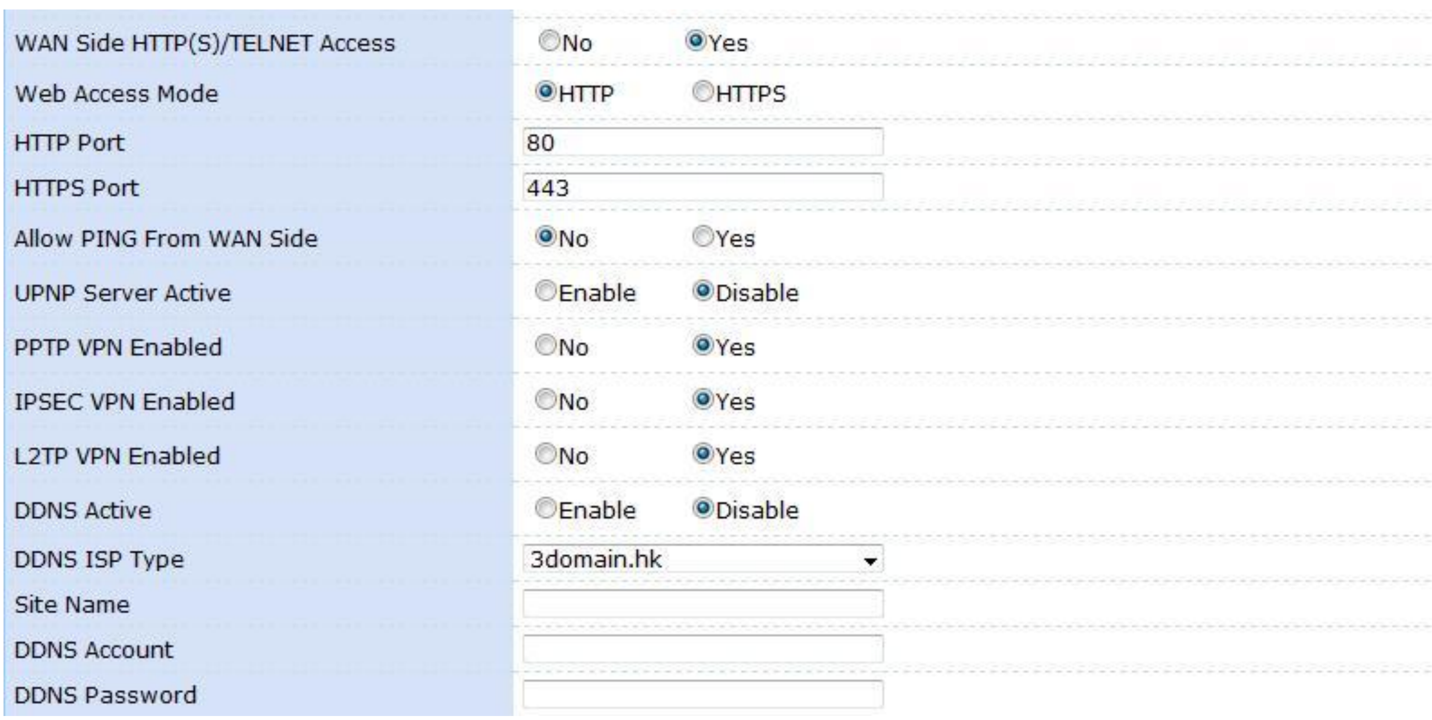
  - ***DHCP Configuration:***
  To use the GXE502X as a DHCP server on the LAN-side, set the *DHCP Enable* setting to *Enable* (It is enabled in the default settings). Users may specify the range of the IP addresses offered by the DHCP server by setting the *Start of DHCP IP Pool* and *End of DHCP IP Pool* settings. The *DHCP IP Lease Time* allows users to specify the number of hours an IP address is leased to a device before renewal.

- **WAN Setting:**
  - The WAN-port can be configured to be dynamically assigned via DHCP, to use PPPoE, or to be static. Select the radio button beside the option that you wish to use. To use PPPoE, enter the PPPoE account, password and DNS server information provided by the ISP in the provided fields. If statically configuring the WAN-port IP address, please enter the IP address details from the ISP in the provided fields.



WAN Settings	
	DHCP / PPPoE / (•) Statical IP
PPPoE Account ID	admin
PPPoE Password	••••••••
Preferred DNS server	. . .
IP Address	67 .110 .250 .172
Subnet Mask	255 .255 .255 .192
Default Router	67 .110 .250 .129
Primary DNS	4 .2 .2 .3
Secondary DNS	67 .106 .1 .196

- *WAN-side Access and Security*:
  Accessibility and Security for management are considerations for whether or not to enable or disable HTTP(S)/Telnet access from the WAN-side of the GXE502X. The default setting for *WAN Side Http/Telnet Access* is YES. You can also select to use either HTTP or secure HTTP (HTTPS) here. The default port for HTTP access is 80 and for HTTPS is port 443. At the default settings UPNP is turned OFF, but PPTP, IPSEC and L2TP VPN are enabled.



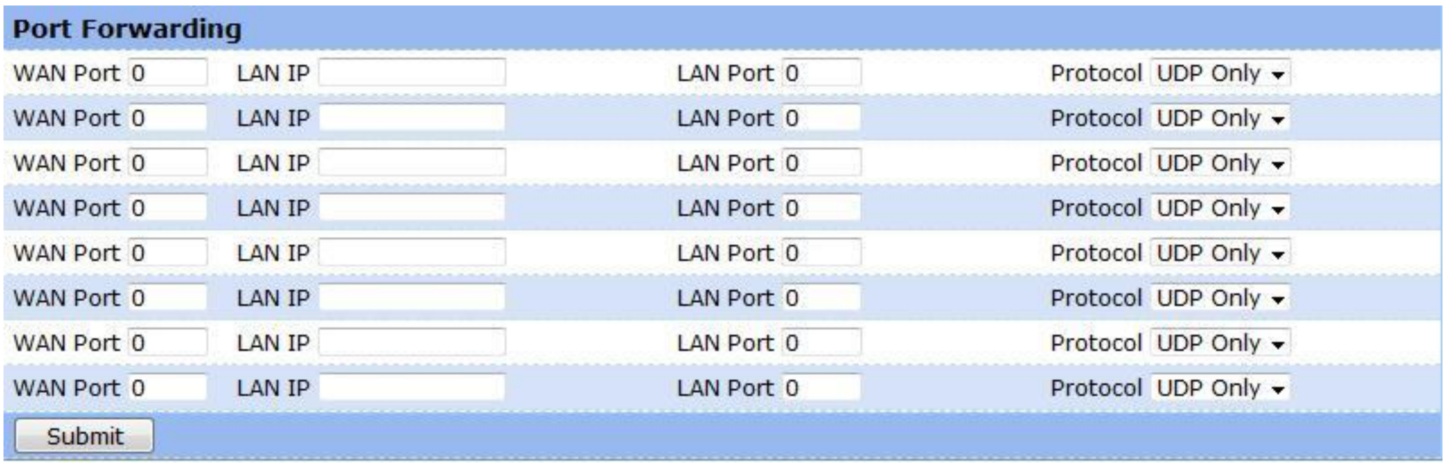
WAN Side HTTP(S)/TELNET Access	No	(•) Yes
Web Access Mode	(•) HTTP	HTTPS
HTTP Port	80	
HTTPS Port	443	
Allow PING From WAN Side	(•) No	Yes
UPNP Server Active	Enable	(•) Disable
PPTP VPN Enabled	No	(•) Yes
IPSEC VPN Enabled	No	(•) Yes
L2TP VPN Enabled	No	(•) Yes
DDNS Active	Enable	(•) Disable
DDNS ISP Type	3domain.hk	
Site Name		
DDNS Account		
DDNS Password		

- *Using Dynamic DNS*:
  If the WAN-port IP of the GXE502X is dynamically assigned by your ISP, the GXE502X can use Dynamic DNS to obtain a fixed domain name which is always synchronized with the IP address of the GXE502X. Enable Dynamic DNS by setting the *DDNS Active* setting to

Enable and setting the DDNS ISP type, site name, and account and password in the provided fields. If you are not using Dynamic DNS, set the *DDNS Active* setting to *Disable*.

Layer 3 QoS: This enables Layer 3 QoS differentiated services or precedence values for the voice packets coming out of the GXE.

- *Configuring port forwarding*:
  The GXE502X can be configured to perform port forwarding. In the *Port Forwarding* settings, enter the port to forward from the WAN-side, the LAN-side device IP address to forward to, the LAN-side device port to forward to, and the protocol (TCP, UDP, or both) to forward in the respective fields.



When done, click on the **Submit** button to save your changes.

- Route Configuration

This sub menu is used to add static routes into the routing table. This feature is useful when there are destinations that cannot be reached using the default gateway or when there are several subnets in your local network. The interface can accept a network or host address to route. For example:

Network=192.168.22.0/24 can be reached through router 192.168.1.50, then:
Destination IP: 192.168.22.0
Mask: 255.255.255.0
Gateway: 192.168.1.50

Static route to host 192.168.22.5 through router 192.168.1.50 then:
Destination IP: 192.168.22.5
Mask: 255.255.255.255
Gateway: 192.168.1.50

Click the *Add* button on the right to add the route in the routing table. Click the *Delete* button to delete the related route from the routing table.

-Blacklist

The administrator of the system can type in an IP address in the field in order to block this IP. To add an IP address, simply click on the Add button. This is particularly useful when the administrator detects that he is getting attacked from a particular IP and he wants to stop all responses to that IP in particular.

- System Setting

System Setting sub menu Configuration:

The System Settings sub menu contains various important internal system settings and allows configuration of several key features and functions of the GXE502X. The actions below and the associated settings appear in order from top to bottom on the GXE502X web UI:

- *Setting the Administrator login password*:
  This is where you can change the administrator password of the GXE502X. The default password is “admin”.
  For security reasons, it is recommended to change the administrator password immediately after Express Setup. Please document the password in a secure place. For further security, you can disable WAN-side HTTP access or PING.
- *Administrator Contact information*:
  These fields are optional and provided to store the administrator’s name, phone number, and email address so that they can be contacted for any issue requiring attention.
- *System Name:* This System Name will be used in email for voicemail and fax mail to identify the sending GXE5000.

Fax Options:

- *Manual Selection of Fax:* The default setting for this is yes. This is used when a caller reaches an extension's voicemail. The prompt played will be "Press 1 to leave a message, press 2 to send fax". If set to No, voice prompt 'press 2 to send fax' is removed and FAX reception will automatically start upon detecting FAX tone.

System Music Options:

- *System Music:* This tells the GXE which music on hold audio to play. You can select the music files stored in the GXE or you can select to play music from the audio-in port in the back of the GXE. Users can select a Music on Hold (MoH) source by setting the System Music drop-down box to choose either System Music Files of internally stored audio files or Audio-In to play audio from an external audio source (like an iPod or Radio). The source must be connected to the AUDIO IN jack on the back of the GXE502X.
- *Upload system music:* This field allows you to select an audio file from your computer and upload it to the GXE-502X by clicking on the browse button. This audio file will be used as your music on hold and it can be up to 3 minutes long.
  The file must be in the following format: 8 KHz/16bit/MONO .WAV

Email Settings:

To configure the GXE502X to send out voicemail via email, the related SMTP setting must be configured. The receiving email address can be configured in the *Extensions Directory* sub menu of *Phone Extensions* configuration.

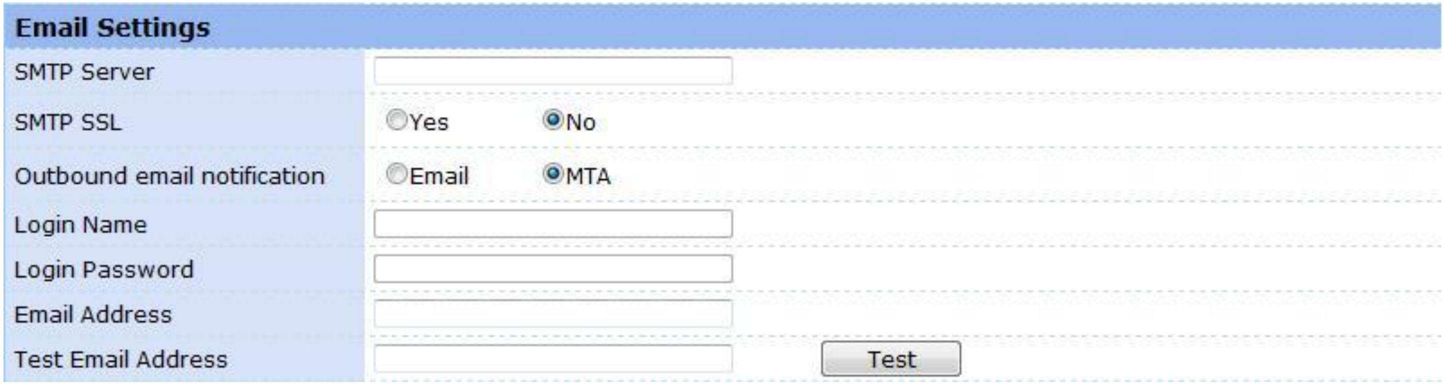
There are two ways to configure the GXE502X to send emails:

1) Configure as email client. This requires an SMTP server to forward all outgoing emails via that particular authorized email client account.

Configure as MTA. This usually must be handled by the system administrator. This will analyze the destination email address and resolve it using the domain name of the destination email server.

Note: For spam prevention, many email service providers reject or delete incoming emails originating from a dynamic IP address or emails that are not from a FQDN. If this is the scenario, the authorized email client account is strongly recommended to prevent email loss.

To send using the "as email client method," the user must enter the SMTP server, login name and password, as well as the email address to send from. This will allow the GXE502X to send voice messages through email to users and ring groups with the email address configured when receiving voicemail messages. The user can also select to use SMTP SLL to send the email messages over port 465. There is also a test email address field that the GXE502X will sue to send a test email to verify if the SMTP settings are working.



It is recommended to use a special email account for voicemail to email, for example: bostonpbx@company.com

System Time Configuration:

- *Time*: If NTP set up on either WAN or LAN is disabled, the starting time has to be set manually in the *Time* field for system time.
- *Self-Defined Time Zone*:
  Select the correct time zone for the location of the GXE502X using the *Time Zone* drop-down box. Users may also enter a *Self-Defined Time Zone*, using the following syntax, to configure Day Light Savings Time.

MTZ+6MDT+5, M4.1.0, M11.1.0

The Syntax: std offset dst [offset], start [/time], end [/time]

Default is set to: **MTZ+6MDT+5, M3.2.0, M11.1.0**

MTZ+6MDT+5, (Time Zone)
This indicates a time zone with a 6 hour offset with 1 hour ahead which is U.S central time. If it is positive (+) then the local time zone is west of the Prime Meridian, and negative (-) is east of Prime Meridian (A.K.A: International or Greenwich Meridian)

M3.2.0, M11.1.0 ***(Daylight Saving Time Rule)***
The 1st number indicates Month: 1,2,3.., 12 (for Jan, Feb, .., Dec)

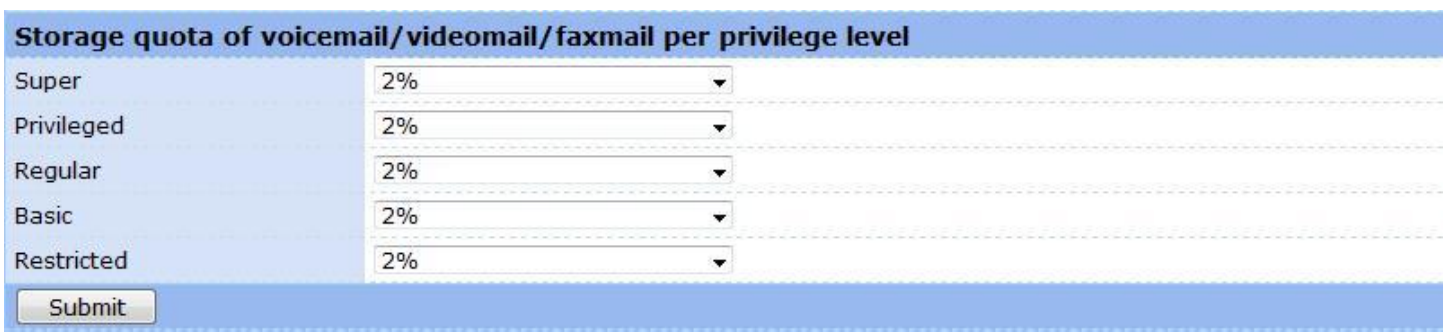
The 2nd number indicates the nth iteration of the weekday: (1st Sunday, 3rd Tuesday…)
The 3rd number indicates weekday: 0,1,2,..,6(for Sun, Mon, Tues,..,Sat)
Therefore, this example is the DST which starts from the second Sunday of March to the 1st Sunday of November.

- *Synchronize with NTP Server*: Whether to Synchronize system time with NTP Server.
- *NTP Server:* NTP Server IP or FQDN.
- *WAN side NTP server:* Default NO. If this option is enabled, the GXE502X will try to synchronize the current time from the NTP server path configured and adjust to the specific time zone by selecting a time-zone or using a self-defined time zone.
- *LAN side NTP server:* Default YES. If enabled, the GXE502X will function as an NTP server on the LAN side and will synchronize the time with the phones connected to the LAN port. This is ideal for a GXE502X installation without WAN side NTP access.

Voicemail/Fax Storage Configuration per privilege level:

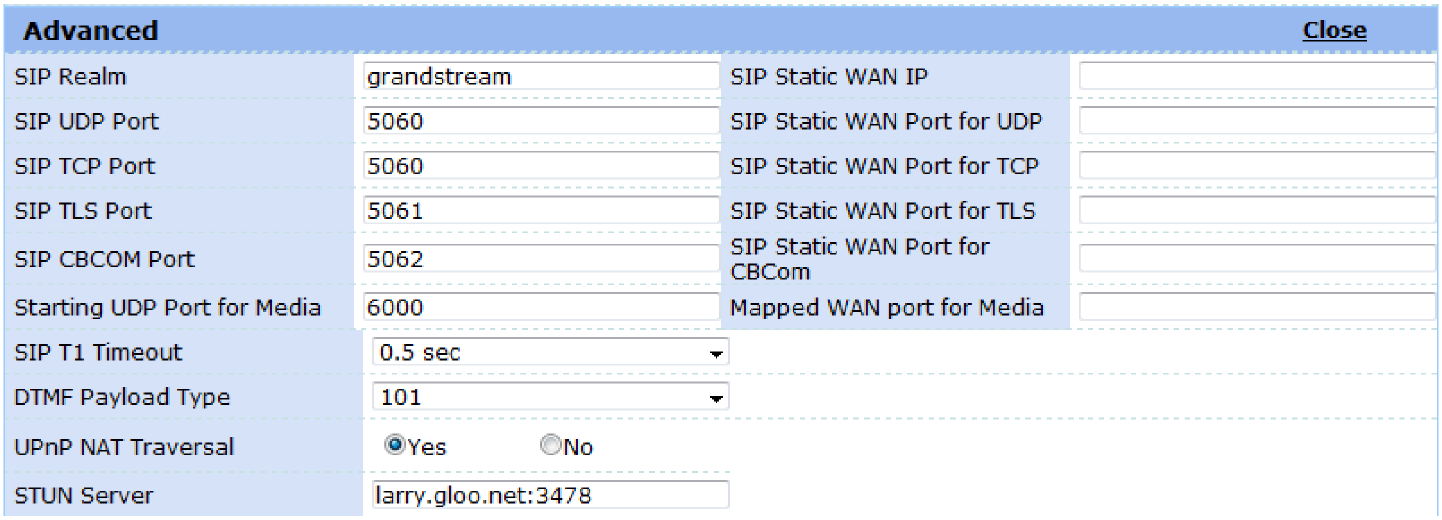
- *Voicemail/Fax mail box storage quota*:

The voicemail and fax mail storage quota can be limited by different extension privilege levels. The mailbox storage limits (in percentages of total system memory) are also specified in the drop-down boxes beside each user privilege level.



Advanced Settings:

SIP Related System Setting:



- *SIP Realm*: If configured, the system will use the entered name in its SIP Realm field instead of default "grandstream". This is used for authentication purposes.
- *SIP UDP Port*: The local listening UDP port for SIP messages.
- *SIP TCP Port*: The local listening TCP port for SIP messages.
- *SIP TLS Port*: The local listening TLS port for SIP messages.
- *SIP CBCOM Port*: The local listening UDP port for SIP messages with CBCOM encryption.
- *Start of UDP Port for Media Transfer*

When this value is set, the GXE502X will use the value within "value+2000" for the local UDP port for its media; this is useful when the GXE502X is behind a NAT and the NAT is configured to do port forwarding with "value+2000" for UDP ports. For example if we set this value at port 6000, then the GXE502X will start listening for media transfer on ports 6000 to 8000. Make sure you configure this dynamic range of ports on your NAT Router.

- *SIP Static Mapped WAN IP:* The public IP of the uplink to which GXE5000 is under. This is typically the public IP address of the router that you have the GXE connected to. You should log into your router's configuration page and look for its WAN IP address. Enter its WAN IP address here.
- *SIP Static Mapped WAN port for UDP*: The SIP port for the GXE502X can be set to either the default of 5060 in the *SIP Port* field, or another port number may be used.
- *SIP Static Mapped WAN port for TCP*: The SIP port for the GXE502X can be set to either the default of 5060 in the *SIP Port* field, or another port number may be used
- *SIP Static Mapped WAN port for TLS*: The SIP port for the GXE502X can be set to either the default of 5061 in the *SIP Port* field, or another port number may be used
- *SIP Static Mapped WAN Port for CBCOM:* This Value is reserved for China's CBCOM SIP trunk provider.
- *Mapped WAN Port for Media:* This is in correlation to the "Start of UDP Port for Media Transfer". When GXE5000 is behind the NAT and need to open WAN port for media transfer, it should map the local UDP port sequentially.
- *SIP T1 Timeout:* This value sets the time to wait for a retransmission. This is the RTT time. For more information please refer to RFC 3261 section 17.1.1.1.
- *DTMF Payload type:* This parameter is used to configure the payload type of the DTMF sent and received according to RFC 2833. This will affect all SIP trunks, external PSTN trunks, peer systems and extensions connected to the GXE. The default value is 101.

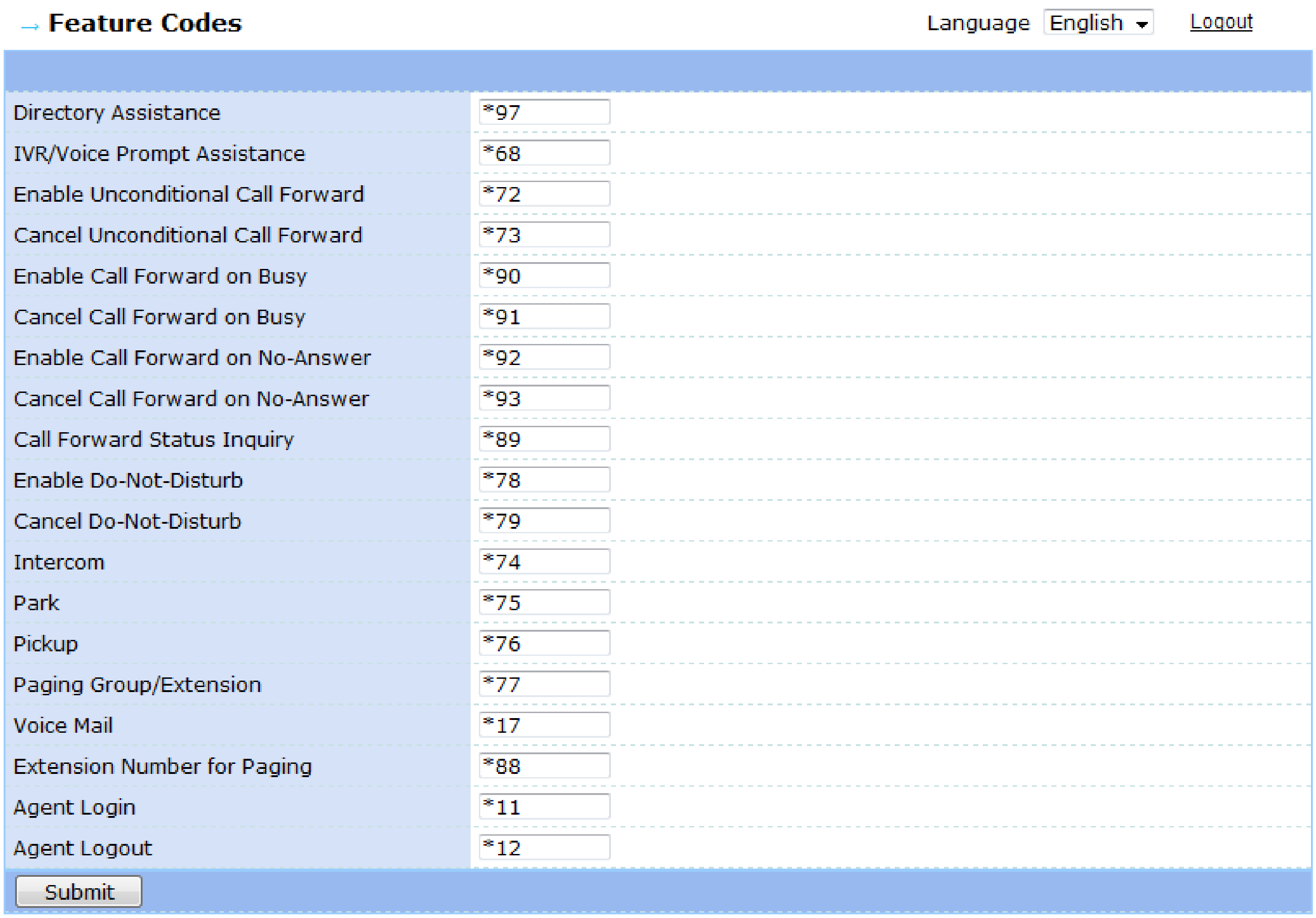
- *UPnP NAT Traversal:* If the router that the GXE502X is connected to supports UPnP, then enable this option to avoid NAT Traversal issues. The GXE502X will create a “tunnel” through which all the media packets will travel, preventing one way audio or no audio issues. This created port range will take care of the RTP traffic.
- *STUN Server (for NAT traversal)*:

If the GXE502X is behind a non-symmetric NAT router, it may be necessary to use STUN to allow GXE502X to reliably communicate via IP through the router. Enter a STUN server IP address or domain name in the *STUN Server* field. For a list of public STUN servers please refer to: http://www.voip-info.org/wiki/view/STUN

When done, click on the **Submit** button to save your changes.

- Feature Codes

Feature codes allow GXE502X users to set features such as Call Forwarding and Do Not Disturb on related extensions, as well as access destinations such as the voicemail system. This page also allows users to customize the feature codes. The feature codes may be viewed or modified in the text field beside each feature name.



Directory Assistance	*97
IVR/Voice Prompt Assistance	*68
Enable Unconditional Call Forward	*72
Cancel Unconditional Call Forward	*73
Enable Call Forward on Busy	*90
Cancel Call Forward on Busy	*91
Enable Call Forward on No-Answer	*92
Cancel Call Forward on No-Answer	*93
Call Forward Status Inquiry	*89
Enable Do-Not-Disturb	*78
Cancel Do-Not-Disturb	*79
Intercom	*74
Park	*75
Pickup	*76
Paging Group/Extension	*77
Voice Mail	*17
Extension Number for Paging	*88
Agent Login	*11
Agent Logout	*12

- ***Directory Assistance***: Dial from a phone to have the GXE502X playback the phone's extension number.
- ***IVR/Voice Prompt Assistance***: Used for listening to and recording voice prompts. Dial the feature code plus * 0 * [prompt number] * to listen to the voice prompt. Dial the feature code plus * 1 * [prompt number] * to record the voice prompt.

<u>Example:</u> The voice prompt file number 350 "Please enter the voicemail extension and #" and the code feature is *68.

- If you want to listen to this voice prompt, enter: *68*0*350*
- If you want to record the voice prompt number 350, enter: *68*1*350*

You can also change the music on hold file in a similar manner. The voice prompt file number for the music on hold is 27. So if you want to listen to the music on hold file simply dial *68*0*27* SEND from any extension. If you want to record the music on hold from any extension then dial *68*1*27* SEND and you will be able to replace the music on hold file.

You can also create a file named 27.wav using the IVR prompt conversion tool and upload it under the system prompt image field.

- ***Enable Unconditional Call Forward*****:** Dial the feature code plus [forward-to number] to enable unconditional call forwarding for the extension you are dialing from. For example: dial *72*617-555-5555 and press SEND. You will hear a prompt confirming the activation of the call forward feature.
- ***Cancel Unconditional Call Forward*****:** Cancel unconditional call forward, calls will ring in.
- ***Enable Call Forward on Busy*****:** Dial the feature code plus [forward-to number] to turn on busy call forwarding for the extension you are dialing from.
- ***Cancel Call Forward on Busy*****:** Cancels call forward on busy; calls will ring when the user is on the phone.
- ***Enable Call Forward No-Answer*****:** Dial the feature code plus [forward-to number] to turn on no-answer call forwarding for the extension you are dialing from.
- ***Cancel Call Forward No-Answer*****:** Cancel call forward on no answer, calls will go to voicemail.
- ***Call Forward Status Inquiry:***
- ***Enable Do-Not-Disturb*****:** Calls will go straight to voicemail, or return busy if voicemail is disabled.
- ***Cancel Do-Not-Disturb*****:** Cancel DND, calls will ring in normally.
- ***Intercom:*** *Dial the Feature code + * + phone extension to page a specific extension.*

> **Example:** The phone extension to page is 601 and the intercom feature code is *74.
> - To page that extension, you dial: *74*601.
>
> **Important:** To use the intercom feature correctly, the phone paged should have the option "***Allow Auto Answer by Call-Info***" enabled.

- ***Park*****:** When in a call, an extension can transfer the other party to this feature code to park the call.

How do Call Park and Call Pick up work?

Call Park is done using an Attended-Transfer. For example, if you have a GXP2000 that takes an incoming call on Line 1, you can press Line 2 (or any line in this case) to automatically put Line 1 on hold. You can then enter your Call-Park feature code (ie. *75), this will prompt a parking announcement: "Parked At 000." You can then press Transfer followed by the Line1 button to transfer the call on Line1 into parking extension 000. To pick up the call, dial ***76***000 to pick up the call, assuming *76 is your Call-Pickup feature code. The parking extension number starts at 000.

> **Important:** This must be performed as an attended transfer. During the transfer, the GXE502X will tell you the parking extension used for this call.

- ***Pickup*****:** Dial the feature code plus * [parking extension] to pick up a call parked by the parking extension.

> **Example:** the pickup feature code is: *76 and the parking extension that the GXE502X gave to this parked call is 100.
> **-** To pick the parked call, you dial *76*100.

- ***Paging Group/Extension:*** Dial the feature code plus * [group/extension] to page an extension or a group.

Example: the paging group/extension feature code is: *77, 601 is an extension number and a 610 is a ring group number (ring group members are: 603, 604, and 605).
- To page the extension 601, dial *77*601.
- To page the group 610, dial *77*610 (will page all ring group members 603/604/605).

Important: If the phone paged does not have the option "***Allow Auto Answer by Call-Info***" enabled, the phone will only ring and not answer. Such parameter needs to be enabled in the phone's web UI.

- ***Voice Mail***: Dial the appropriate feature code to access the voicemail system. In some Grandstream IP phone models you will find a field "Voice Mail User ID" under the account information, if you type in the voice mail feature codes here then you can press the mail button on the phone and go directly to voice mail.
- ***Extension Number for Paging:*** This feature code is used to connect to the audio out jack in the back of the GXE502X. Some users might want to connect a PA system to this jack and use it for loudspeakers announcements. The user can simply dial this feature code to make an announcement through the loudspeakers. Think of it as an extension with auto-answer.
- ***Agent Login:*** This feature code is used by any non GXP phones (with softkeys) or ATAs that do not support the Grandstream protocol for call queue agent recognition. In case we have a non Grandstream phone or ATA, the user can dial for example *11*call queue number +Send to log into the desired queue. If successful, the user should hear the "operation succeeded prompt"
- ***Agent Logout:*** This feature is similar to the agent login feature, but it is used to logout the user from the queue. To log out simply dial for example *12*call queue number + Send from the agent's extension, and a prompt should inform the user that it has been successfully logged out of the queue.

When done, click on the **Submit** button to save your changes.

- **Firmware Upgrade**

The *Firmware Upgrade* sub menu allows users to upload new firmware versions to the GXE502X and upload custom voice prompts.

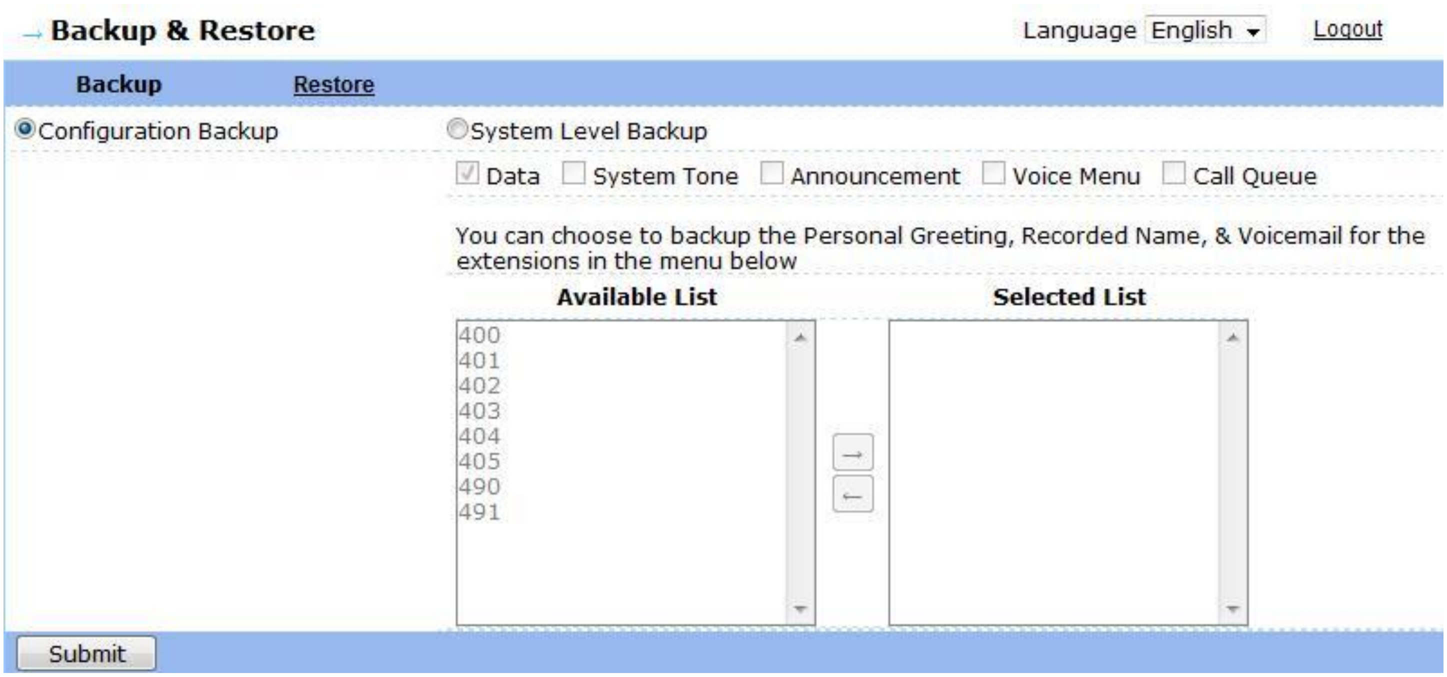
Please read the release notes carefully prior to upgrading your firmware to document the changes. Grandstream also recommends saving a **backup** file of all GXE502X configurations (please see the next section). The firmware upgrade will not overwrite your existing GXE502X configuration, but it is good practice to keep a backup copy in case.

Once you have downloaded and uncompressed your new firmware, click the ***Browse…*** button to find and select the "gxe50xxfw" file on your computer. This file is located in the "image" directory of the firmware package. After the file has been selected, click the ***Submit*** button and wait for the file to finish uploading. When finished, reboot the GXE502X to load the new firmware.

Custom voice prompts may also be uploaded in the Firmware Upgrade sub menu. To upload your custom prompts browse to the prompt file in the *System Prompt Image* field and click the **Submit** button. You can upload zip files containing .wav files or a PV files that have been generated by the Grandstream Wavtools.exe prompt generator. The voice prompt file is named "gxe50xxpv.bin" .You can download this voice prompt file from www.grandstream.com/firmware and upload it under system prompt image.

- Backup & Restore Configuration

GXE502X Configuration files may be downloaded to or uploaded from the computer in the *Backup & Restore Configuration* sub menu. It is good practice to keep a backup copy of your GXE's configuration file at all times.



The System Level Backup & Restore page allows users to Backup/Restore the entire system or specific portions such as: voicemail, system tones, announcements & voice menus.

- *Configuration Backup*: Select the *Configuration Backup* radio button, and click the **Submit** button. You will be prompted to download the configuration file to your computer. The Configuration Backup will only save the information related to the data in GXE if you want to save the system tones, announcements, voice menus and call queues then you should perform a system level backup.

- *System Level Backup:* Select the System Level Backup radio button, then check the boxes next to the desired configuration components and click *"Submit"* to backup system level configuration data, System tone, Announcement, Voice Menu and/or Call queues. Users can also choose to backup voicemail and greetings for a specific extension by selecting an extension from the available list. You can select multiple extensions from the available list and put them in the selected list.

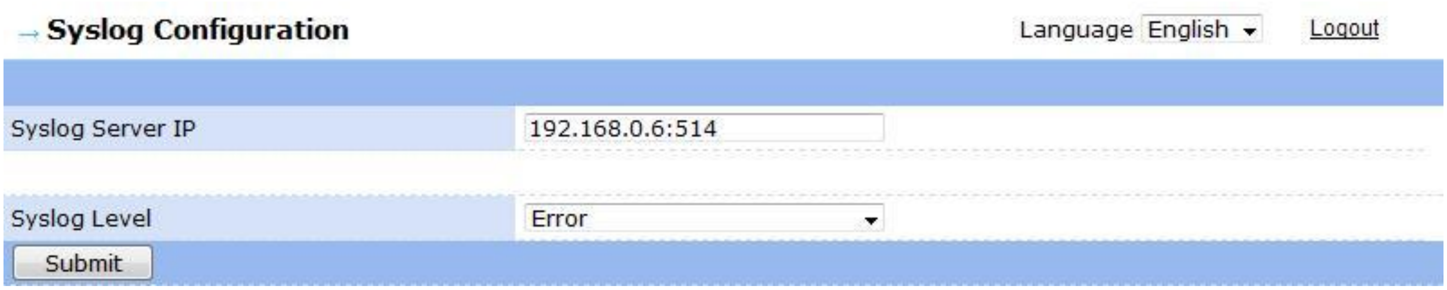
Important: - The "Data" checkbox is always checked. A backup cannot be performed without having it selected.
- The system restoration can be done only by using a backup file from the same GXE502X series (e.g. GXE5024 to GXE5024).
- Do NOT reboot the GXE502X when it is in the process of downloading the Backup file.

Configuration Restore: Click on the restore link and then select the *Configuration Restore* radio button, and click the **Browse…** button to find the configuration file on your computer. After the file has been selected, click the **Submit** button and wait for the file to finish uploading. The GXE502X will automatically reboot when finished, and load the configuration file.

System Level Restore: This lets users browse to the backup file and click the **Submit** button to restore the contents of the file to the GXE502X system configuration.

- Syslog Configuration

The GXE502X can be configured to send out several levels of Syslog messages: Info, Warning, Error, and Debug. With a Syslog server setup to collect these messages, problems are easier to find and diagnose. The Syslog Configuration sub menu allows a Syslog server IP address and the Syslog level to be set.



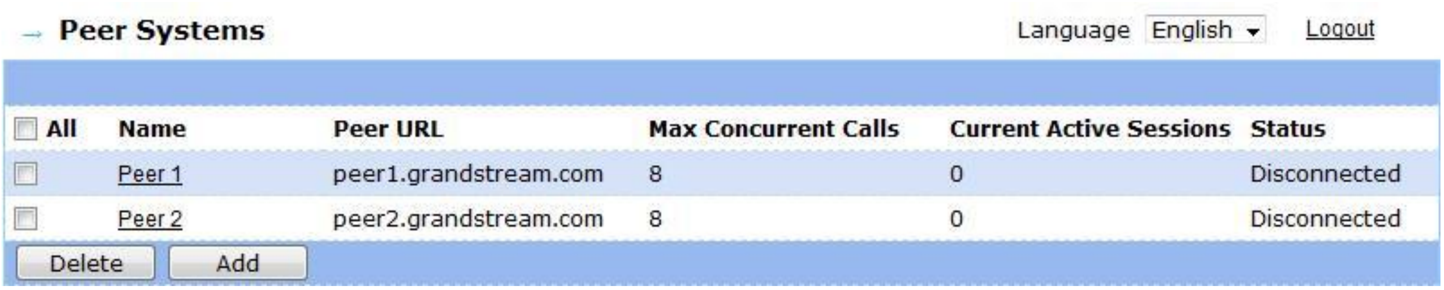
Enter the IP address of the Syslog server in the *Syslog Server IP* field, and use the *Syslog Level* drop-down box to specify the level. The Error level gives out the least amount of information, the Debug level gives out the most amount of information. When done, click the **Submit** button to save the changes.

9. Advanced Options Configuration

Remote GXE502X systems or other PBX systems can be peered with a local GXE502X. This allows local users to dial remote system extensions and the GXE502X will route the calls directly via IP to the remote system. In a networked multiple office environment, this provides the ability to reach remote colleagues through simple and familiar extension dialing but without incurring the toll costs of routing the call over the PSTN. The **Advanced Options** menu allows the user to view, add, modify, and delete networked peer systems.

- Peer Systems

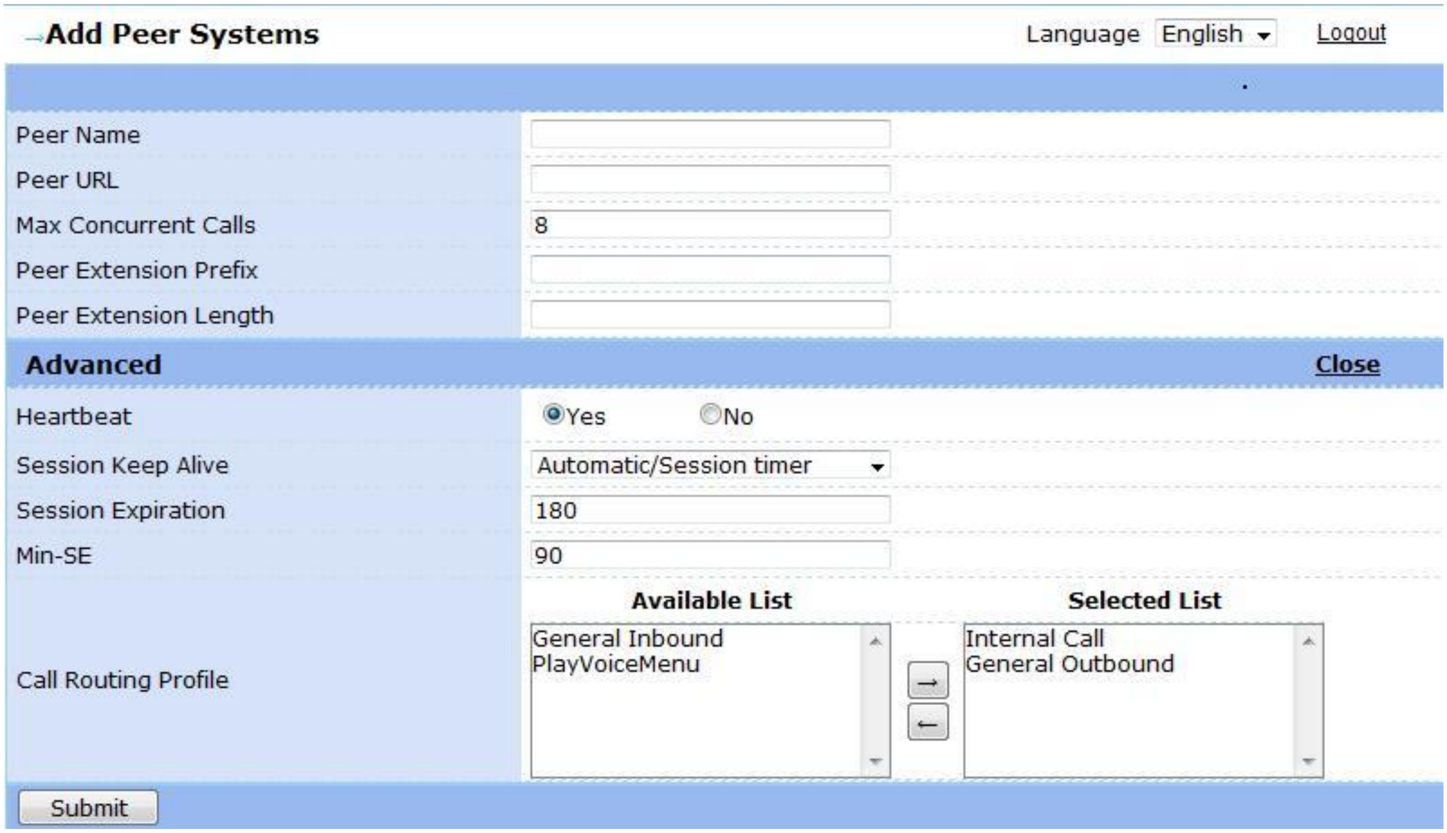
The Peer Systems sub menu displays all configured peer systems in details.



All	Name	Peer URL	Max Concurrent Calls	Current Active Sessions	Status
	Peer 1	peer1.grandstream.com	8	0	Disconnected
	Peer 2	peer2.grandstream.com	8	0	Disconnected

The following actions can be performed to manage peer systems:

- ***Add*****:** Clicking the **Add** button will load the add peer system page. where users can enter connectivity settings for the new peer system:



- *Peer Name:* This field lets you enter a name to identify this peer system
- *Peer URL*: Enter the IP address or FQDN domain name of the peer system.
- *Max Concurrent Calls*: This field specifies the maximum number of concurrent calls allowed between the peer systems. This is usually limited by the bandwidth in between the GXE502X and the peer systems.
- *Peer Extension Prefix*: The prefix digits for outbound calls to be recognized and routed to this peer system. This prefix together with the peer extension length will determine the digitmap to be added under the internal call routing profile. For example if we select 5 as the prefix and we give it an extension length of 4, then the digitmap 5XXX will be created under the internal call routing profile and it will be assigned a peer extension as its option and the value will correspond to the name given to this peer system. Users may specify multiple prefixes for a peer system by separating them with a semicolon (i.e. 6;7).
- *Peer Extension Length*: This is the number of digits that the extensions on the peer system have. For example if the extensions are like 101, 304, 506 then they are 3 digit extension length. If they look like 4001, 3445, 9800 then they are 4 digit extension length.

Advanced

- *Heart Beat:* if set to Yes, the GXE502X will send a *SIP OPTIONS Request* periodically to check whether the peer system is still online. If the peer does not respond with a SIP *200 OK* then the GXE502X will consider this trunk or peer as disconnected. This feature is also used to keep UDP ports open when the GXE502X is behind a NAT firewall.
- *Session Keep Alive:* This field allows users to enable or disable the session timer. This is set to Automatic/Session Timer in the factory default settings.

- *Session Expiration:* The session timer enables SIP sessions to be periodically “refreshed” via a SIP request. If enabled, before the session interval expires, the GXE502X will send a SIP re-INVITE message to the SIP server. If this message is not replied to the session will be terminated. This will avoid dropped sessions occupying the channel/bandwidth. Session Expiration is the timer (in seconds) at which the session is considered timed out if no successful session refresh transaction occurs beforehand. The default value is 180 seconds. Incorrect configuration could cause calls to be dropped or suspended.
- *Min-SE:* The minimum session expiration time (in seconds). The default value is 90 seconds.
- *Call Routing Profile:* Here you will select which routing profiles will be available to the callers coming in from this peer system

When done, click the **Submit** button to add the extension or **Cancel** to go back.

- ***Modify*****:** Click the **Modify** button to the right of the row to select the peer system to be modified. The modify peer system page will be load. This page allows users to modify all of the peer system’s settings. When done, click the **Submit** button to save changes or **Cancel** to go back.

- ***Delete*****:** Click the **Delete** button on the far right to remove a peer system. Users will be prompted for confirmation via a dialog box; click **OK** to confirm or **Cancel** to go back.

- Template Upload

The *Template Upload* sub menu page allows the user to upload a configuration template which determines the parameters that will be configured on Grandstream endpoints during the GXE502X Express Setup process and the general *Auto Provision* process. *Click the Browse button to locate the template file on your computer.* Next click the *Upload* button to upload the template to the GXE502X.

Note: - Custom templates can be created by using the Grandstream Configuration Template, see Appendix B for detail.

-BLF Resource list

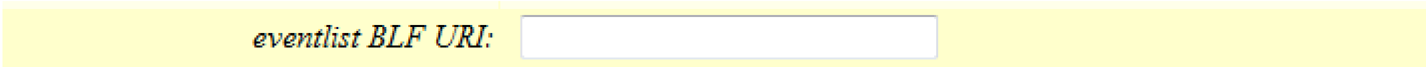
This is also known as eventlist BLF functionality. The Busy Lamp Field functionality can be used together with the Grandstream GXP phones. This function will allow the users to use the multipurpose keys (GXP-2020/2010 and 2000 only) of their phones to monitor the status of other extensions in the same GXE. If an extension is idle, the monitoring key will light up solid green, if the extension is ringing then the monitoring key will blink solid red, and if the extension is busy then it will light up solid red.

If a monitoring key is blinking red, other extensions can pick up that call by simply pushing the monitoring key button.

Click on the add button to add a new BLF Resource list. Give it a name, for example the name of your company, division that you want to monitor, branch name, etc.

Add up to 130 members, you can add them as a range separated by dashes or as individual accounts separated by semicolons.

On the GXP phones go to the Line account information of the extension. Scroll all the way down to where it says eventlist BLF URI and type in the same name you just gave to the BLF resource list. So for example if named the BLF Resource list sales, then type in sales in the eventlist BLF URI.



-Ethernet Capture

This new function in the GXE will allow the user to capture all the packet traffic coming in and out of either Ethernet interface of the GXE. This could be very helpful to debug certain configuration issues in the GXE, particularly the ones regarding SIP trunk compatibility.

Select the interface you want to capture packets on: LAN or WAN.

Select the running time for the Ethernet capture, up to 10 minutes.

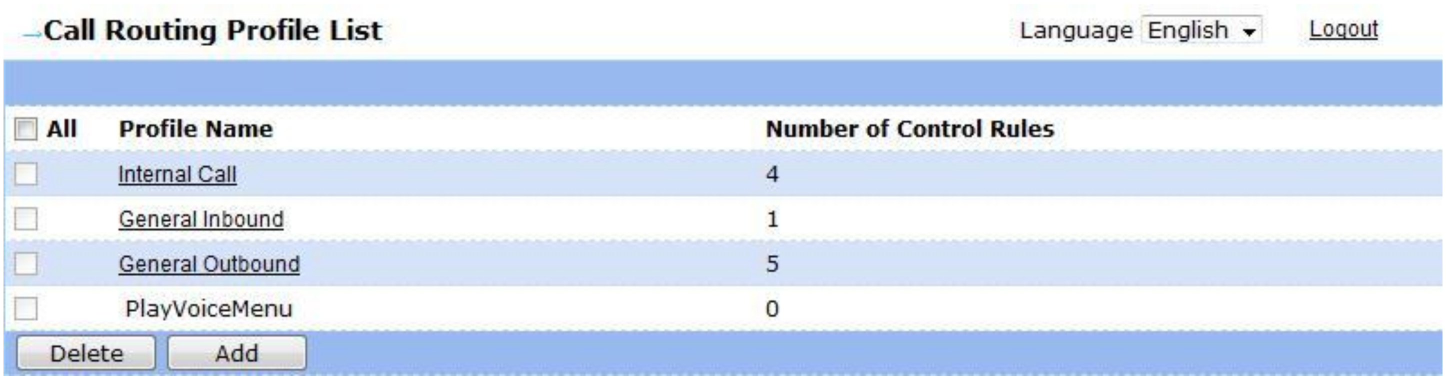
If the administrator wants he can add a filter string like: host 192.168.83.159 && udp or host 192.168.83.159 and udp port 5060).
Click on the start button to begin the capture, click on stop to stop it once you have performed the tests you wanted and then click on download to download the trace into your computer for further analysis.

-Remote Access

This feature allows the user to get technical support easily. The technician can inform the system administrator which server address and port to point the GXE to, as well as the access type to be used. The technician can access the system either via HTTP or Telnet.
Once the user has pointed the GXE to the specified server the technician will have control over the GXE over a TCP connection.

10. Call Routing

This new feature is the most noticeable addition to the new GXE-502X firmware. With this feature the users can program their own routing profiles so that we can achieve LCR (least cost routing), multiple IVRs for different lines, calls restrictions based on extensions, etc.
Initially there will be 3 call routing profiles: internal call, general inbound and general outbound. Additionally you will also see the PlayVoiceMenu call routing profile but you are not able to edit it since you can only perform this operation under the Auto-attendant menu.
The user can create additional call routing profiles for other purposes (up to 10). For example we can have one for international calls, another one for long distance, another for access to GSM gateways, etc



Call Routing Profile List — Language English — Logout

All	Profile Name	Number of Control Rules
	Internal Call	4
	General Inbound	1
	General Outbound	5
	PlayVoiceMenu	0

Delete | Add

Digit Maps

Each call routing profile contains several digit maps (up to 10). Digit maps are digit patterns created by the GXE according to some instructions or they can be created by the user. The digit map syntax is as follows:

- X: This will match any digits from 0 to 9
- N: This will match any digits from 1 to 9
- Z: This will match any digits from 2 to 9
- . (dot): This will match any digits, * (asterisk) and blank
- [ ]: This will match the digit that you put in between the brackets

These can be used to specify the length of the string dialed. For example is the digit map is XXXXXXXXXX (10 X's) then this will basically match any 10 digit number dialed which is the way people dial within the USA; now this rule may not be too practical because it is too general, instead the user may want to create something like: ZXXZXXXXXX where the first and fourth digit of the string dialed match any number from 2 to 9. This adjusts better to the reality of the North American numbering system.
We can also write the matching digits as a set of digits or a range. For example to match any digit from 2 to 9 we can also write [2-9] or we can specify all the digits as a set [2,3,4,5,6,7,8,9].
So in the example above we can write the general rule for the USA as: [2-9]XX[2-9]XXXXXX, which might be easier to understand for some users.
We can also specify certain strings. For example we can say that 911 is a digit map. This way the GXE will know the best way to route as soon as it sees an emergency call (911).
We also have the "," (comma) and "." (dot) operators. The comma introduces a 5 second delay while dialing and the dot is the wildcard, so any digits dialed after the dot will just be sent out.
The important thing is to notice that we can decide to set a string size. As explained above we can input a fixed number of Xs, Ys and Zs to indicate the GXE how long the string dialed should be. We can also replace any of the XYZs with a range of numbers like [2-9]. Following are some examples to illustrate what we just learned.

- NXXNXXXXXX: Typical US call when using 10 digit dialing. This will match numbers like 617-555-5555, 626-555-5555 or 972-555-555
- [1-9]XX[1-9]XXXXXX: Same as above but this time we are specifying the range used. Again in this case the GXE will be looking for 10 digits dialed that match this pattern in order to route the call.
- 911 or similar numbers: In most countries there will be some 3 digit numbers established for emergencies or information services. The GXE will be looking for that string in particular so the user has to actually dial those numbers in order to match the digit map.
- 1[2-9]XX[2-9]XXXXXX: Another typical USA dialing pattern, but in this case we have added a 1 at the beginning since some users might be used to dial this way.
- 1800.: In this case there is a . (dot) right after the 1-800 area code. This means that the GXE will not care about which or how many numbers are dialed after the 1800. The (.) is used as a wildcard since it will match any digits without a pre set length.
- 1866,XXXXXXX: In this case there will be a 5 second pause introduced after the 1866.
- 6XNZ[1-5],123456: This one is a little complicated, but it illustrates how we can basically match any string of numbers as we please. The first digit is 6, second digit is any digit from 0 to 9, third digit is any digit from 1 to 9, fourth digit is any digit from 2 to 9, fifth digit is any digit from 1 to 5, "," will cause 5 second delay, followed by number "123456". This will also be an 11 digit string.

Additionally we can also use the * (star) or # (pound) signs if necessary just like any other digits.

Conditions

A user can add up to 5 conditions to each digit map. These conditions are Boolean expressions which mean they are either true or false. Typically conditions will be set mostly by advanced users as the digit maps don't necessarily require them to function.
The expressions that go in the condition field consist of variables, conditional operators and values connected with logical operators. The logical operators are as follows:

- &&: Logical AND
- ||: Logical OR
- !: Logical NOT

People with a minimum of programming experience will find them familiar.

The conditional operators are as follows:

- !=: Not equal to
- ==: Equal to
- >, <, >=, <= : Comparison operators, Greater Than, Less Than, Greater Than or Equal to, Less Than or Equal to.

The variables are as follows:

- CALLER: Used to indicate a particular caller ID to be routed. For example we could have the case were we want certain people within the company to have access to the SIP trunks of the GXE in order to place international calls at a lower rate. In this case the employees could call into the GXE from their cell phones and the caller ID would get recognized, the GXE could then route the call accordingly; for example we can send this call to an auto-attendant with no restrictions or passwords.
  To do this we can type into the condition string something like: CALLER==6175555555. Or we can have multiple caller IDs to be recognized:
  CALLER==6265555555||CALLER==9725555555||CALLER==4695555555. All these caller IDs would get routed according to the set value.
- DATE, WEEKDAY, TIME: These values are input in the condition field in the following format:
  - DATE: MMDD(Month + Day of Month)
  - WEEKDAY: SUN, MON, TUE, WED, THU, FRI, SAT
  - TIME: HHMM

  For example let's say we want to define the working schedule for: Monday through Friday, and exclude the holiday that is between Dec 25 to Dec 27, and the working hours are from 8:00AM to 12:00PM and 14:00PM to 18:00PM, we would have a string like this:

  - (WEEKDAY>=MON) && (WEEKDAY<=FRI) && (!((DATE>=1225) && (DATE<=1227))) && ((TIME>0800 && TIME<1200) || (TIME>1400 && TIME<1800)).

  Using the previous condition the administrator could combine it with a SIPDID condition to route calls to certain IVR based on the time and date of the week.

  Other common business hours settings could be:

  - (WEEKDAY>=MON&&WEEKDAY<=FRI)&&((TIME>=0800&&TIME<1230)||(TIME>=1300&&TIME<1730))

  A common after hours setting could be:

  - ((WEEKDAY>=MON&&WEEKDAY<=FRI)&&(TIME<0800||TIME>=2100))||(WEEKDAY==SAT||WEEKDAY==SUN)

  A common holiday setting could be:

  - DATE==1127||DATE==1128||DATE==1225||DATE==0101||DATE==0216

- SIPDID: This variable will be used in the conditional expression to specify a particular DID associated with a SIP trunk connected to the GXE. We can use it in combination with the time and date conditions to route incoming calls to a particular IVR, extension, conference, etc that we choose.

Call Paths

Each digit map consists of up to 5 different call paths. Call paths are the routes that we choose to direct a call; they can be either outgoing or incoming. Each call path is also composed of 3 different fields: Digit Manipulation, Option and Value.
The idea behind the call paths is to have fallback routes in case one of them is blocked or not available.
For example one call path could be routing the call through a SIP trunk, but maybe the server of the service provider is down, in that case if there is a second call path configured, let's say a PSTN trunk, the GXE will try to make the call through it. Each call path can be configured completely different from the other ones and it is left at the absolute will of the administrator how to configure them.

Digit Manipulation

Digit Manipulation will allow the administrator to manipulate the digits being received by the call routing profile as he might please. This can be done in 3 ways: addition, deletion and replacement.
The rules are the following:

1. (I:NN) is to add number NN.
2. (D:XXX) is to delete any 3 digits mapped in the position, XXX.
3. (M:XXX:NN) is to replace any 3 digits in the position where XXX is with digits NN.

In other words (I:) is used to INSERT any digits in the position that the administrator wants.
This is typically done at the beginning of the number dialed but we can insert numbers in any position of the string dialed. For example:

- (I:99)XXXXXXXX will insert the digits 99 in front of an 8 digit dialed string, so the numbers dialed out will be like: 99XXXXXXXX.
- We could also put them in the middle or at the end: XXXX(I:99)XXXX or XXXXXXXX(I:99) and we would get XXXX99XXXX and XXXXXXXX99 respectively.

(D:) is used to DELETE any digits in the position that the administrator wants. Again this is typically done at the beginning of the number dialed but we can insert numbers in any position of the string dialed. For example:

- (D:X)XXXXXXXX will delete the first number dialed from the string
- (D:XXX)XXXXXX will delete the first 3 numbers of the dialed string.

Just as in the previous example we can select to delete digits on any position in the string. This is very useful when we want to force users to dial a certain prefix in order to gain access to a particular trunk.

(M:XXX:NN) is used to replace any 3 digits in the position where XXX is with the digits NN. This is called MANIPULATE. Some users might find it useful in order to create certain routes. For example:

- For (D:XX)(I:17951)XXXXXXXX(M:XXXX:4321), if dialed number is 99075526031234, then the

Number sent out would be: 17951075526034321.
In this case the Manipulate tool would grab the last 4 digits of the string and replace them with 4321.

Option

This is a drop down menu used for the administrator to select the initial step of the route that the call must follow. The options are the following:

- Trunk: Refers to internal and external PSTN trunks connected either to the back of the GXE or to an external PSTN gateway like the GXW-410X. It can also refer to a SIP trunk connected directly to the GXE from a service provider. This option is usually used to get the calls OUT through the selected trunk
- Peer Trunk: Refers to any trunk (PSTN or SIP) connected to a peer that has allowed the GXE access to their trunks. For example we could have multiple GXEs in different office locations all of them peered together, but only one has SIP trunks with access to international calls. We may want to offer access to all the other GXEs to perform international calls in this case we can route the calls towards the GXE with the SIP trunks by selecting this option.
- Peer Extension: Very similar to the one above but in this case the call can only terminate in a peer extension.
- Internal Call: This option will limit the calls to only be able to reach extensions within the same GXE
- Extension: This option will limit the call to only reach an extension in particular within the same GXE. Very useful when we are dealing with incoming calls or when we want to route calls coming in from a particular DID
- Voice Mail: This option will direct the call directly into a voice mail box of a particular extension within the same GXE. Again very useful when we want to route incoming calls.
- Fax Mail: This option will direct the call directly into the fax mail box of a particular extension within the same GXE.
- Group: This option will direct the call directly into a hunt/ring group within the same GXE.
- Group Voicemail: This option will direct the call directly into the voice mail box of a particular hunt/ring group within the same GXE.
- Conference: This option will direct the call directly into a conference room within the same GXE.
- Call Queue: This option will direct the call directly into a call queue within the same GXE.
- VoiceMenu: This option will direct the call directly into a voice menu (IVR) within the same GXE. The user can select which voice menu to reproduce from the list. In this way we can achieve multiple IVR in the same GXE based on certain conditions.

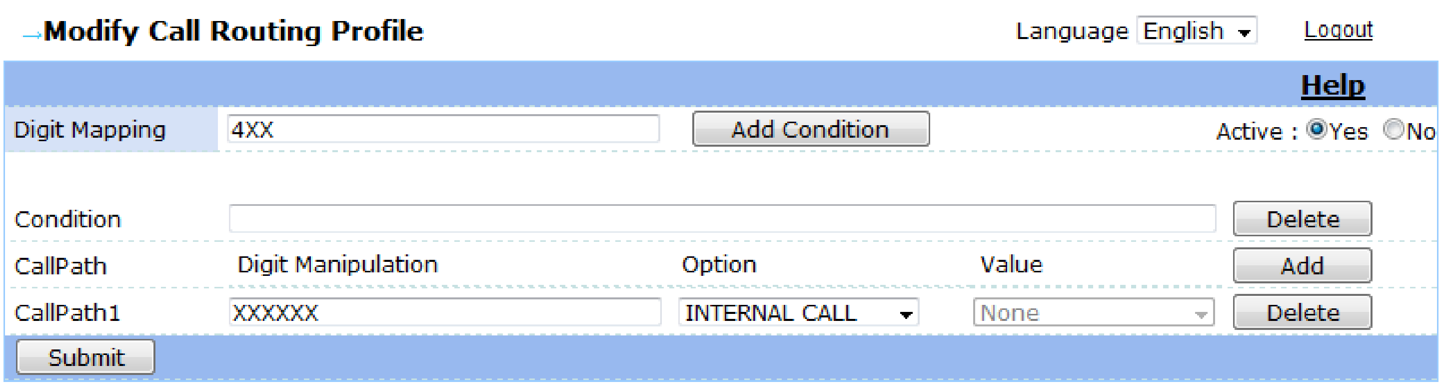
Value

The value of the option selected will vary depending on the option selected in the previous column. Usually most of these values would have been configured first when the administrator sets up the PSTN trunks, SIP trunks, peers, express provisioning, voice menu, etc. The values of the items shown in this drop down menu will correspond to the names given by the administrator so it will be easy to identify them.

We will now explain the first three call routing profiles created by default.

- Internal call: This call routing profile is the one that will be used for all internal calls between extensions. You will start with the following digit maps:
  - *. : Star dot is created by the GXE automatically so that the local extensions can have access to the feature codes. No conditions are set for it and the option is always set as an internal call since it is within the same GXE
  - NXX: If you run the express provisioning, which you should have done in the first place, then you will find that the GXE created a digit map that matches your extensions prefix and length. Again there are no conditions set and the value is always internal call. If you added any peers to the GXE you will find that the GXE also created digit maps for them in a similar way like it did with your own internal extensions, the difference is that now the option is set to Peer Extension and the value corresponds to the peer where those extensions belong.

The administrator can keep adding digit maps (up to 10) to the internal call profile. The idea is that these calls should always be routed within the limits of the company or corporation so that no call would incur in toll charges; although it is left to the administrator's decision how to route all the calls.

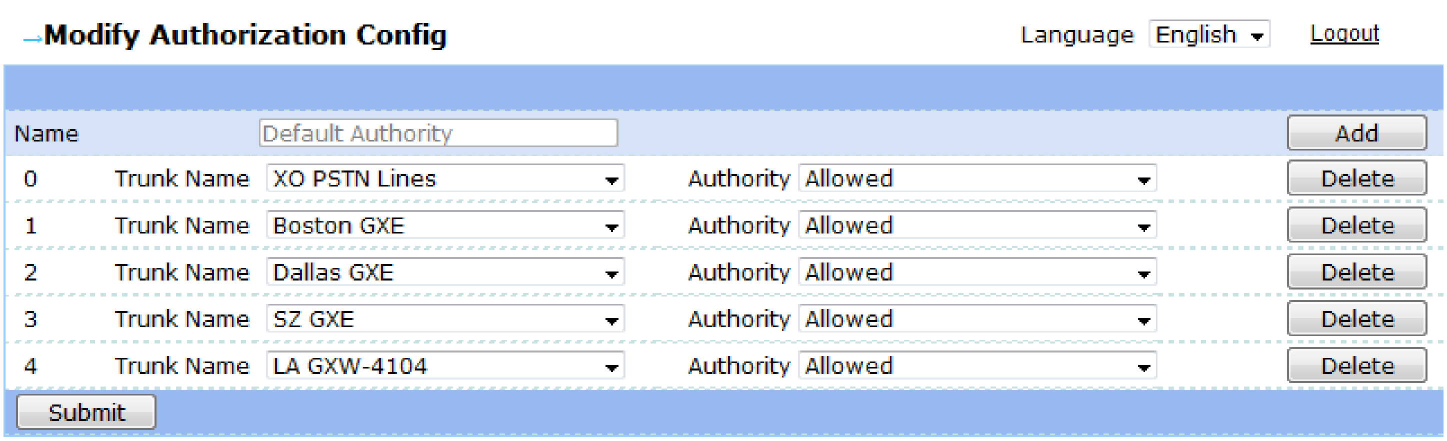


- General Inbound: If the user does not want to use the PlayVoiceMenu option like in the previous firmware versions, then he can select to use this profile. This profile offers more flexibility than the PlayVoiceMenu but it requires a certain level of knowledge in order to configure it.
  Initially the user will only find one digit map configured. The digit map will be the dot (.). Since the dot is the wildcard, this means that the profile will take anything regardless of what digits where used or how many where entered. This makes sense since it is an incoming call and the caller doesn't know the dialing rules. By default there are no conditions set in this profile and the option is set to direct the calls into the default voice menu. This means that by default the GXE is configured to take all incoming calls and route them to the default voice menu that comes pre set in the GXE. Later on the user can start creating different voice menus and combine them with time condition statements in order to reproduce

certain menus at determined times of the day. Basically incoming calls will terminate at the location established with the option and value combination.

- General Outbound: The general outbound profile will not have any configurations initially. The digit maps will be added as you add more trunks (SIP or PSTN) to the GXE502X. For example let's say we add a PSTN trunk and we select 9 as the prefix. The GXE will automatically create a digit map of 9. (nine dot) and will also add the digit manipulation field (D:X). In other words whenever the user dials any string with a prefix of 9, the GXE will strip the first digit (the 9) and will pass on the rest of the string. The option in these cases will be trunk and the values will correspond to the name of the trunk we just created.

Authorization Profile:



By default the GXE will create a single authority profile which will be the default authority profile. Additionally the administrator may want to create authority profiles were no passwords are required or where passwords are required for access to every trunk. There will be 2 columns in each profile authorization menu. The first column will display the trunk name. This could be any trunk connected to the GXE (SIP trunks, PSTN trunks and external PSTN trunks) so that the user can select which trunks will have restricted access.
The second column will display the authority level. There are three levels: Allowed, disallowed and Allowed with Password. The definition is as follows:

- Allowed: This means that the calls will go through without a password. The extensions assigned with this authorization will be allowed to use this resource.
- Disallowed: This means that the call will not go through and get an error prompt.

- Allowed with Password: This means that the calls will only go through if the caller enters his password. The user will have to enter his extension number and then his extension password. The extension does not have to be the same extension number as the calling phone extension.
  The password is the same one that the extension uses to access its own voicemail, fax mail and personal web configuration page.

Central Management:

Modify Central Management Language English Logout

Extension Trunk Other

Number	Call Routing Profile1	Call Routing Profile2	Call Routing Profile3	Call Routing Profile4	Call Routing Profile5	Authorization Profile
400	Internal Call	General Outbou	None	None	None	Default Authorit
401	Internal Call	General Outbou	None	None	None	Default Authorit
402	Internal Call	General Outbou	None	None	None	Default Authorit
403	Internal Call	General Outbou	None	None	None	Default Authorit
404	Internal Call	General Outbou	None	None	None	Default Authorit
405	Internal Call	General Outbou	None	None	None	Default Authorit

Submit

Index:0~5, Total:6

This section provides the administrator with an easy view of the call routing profiles assigned to each extension, trunk, or other features within the GXE like auto-attendants, conference rooms, hunt/ring groups and call queues. It will also display information regarding the authorization profiles assigned to each one of them.
There will be 3 links on the top: extensions, trunk and other.

Extensions Menu: This menu consists of 5 columns with drop down menus plus one column with the authorization profile options.
The options for the drop down menus will be the different call routing profiles available within the GXE.

This is an easy and quick way to determine which extensions will have access to which trunks. For example we could have created a call routing profile for international calls, another one for cell phone calls, another one for long distance, etc. In this way it would be easy for the administrator to determine which extensions have access to which trunks. In the last column we can select which authorization profile corresponds to each extension.

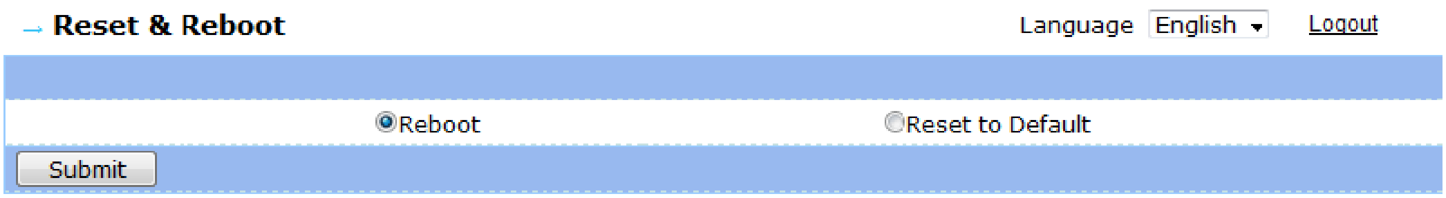
Trunk Menu: This menu will display all the trunks connected to the GXE (SIP trunks, internal PSTN and external PSTN) as well as the peer trunks, which are the other systems peered to the GXE.
In general the peer trunks (peer systems) should have the internal call routing profile selected by default. Additionally we can add some outgoing profiles to the peer trunks here so that they can have access to local resources, but this is completely left to the administrators will.
The SIP trunks, internal PSTN trunks and external PSTN trunks should usually have the general inbound profile assigned since we usually expect to get inbound calls from them.

Other Menu: This menu will show the call routing profiles for all hunt/ring groups, call queues, auto-attendants and conference rooms. Typically all of them should have the internal call routing profile selected so that they can reach other extensions within the same GXE or within the company's limits. The other profiles should only be assigned if the administrator wants to grant outbound calling access to any of them.

11. Reset & Reboot

The Reset & Reboot menu allows you to reboot the GXE502X and perform a factory reset.

Please backup the configuration file of your GXE502X before performing a factory reset.



Rebooting: Select the *Reboot* radio button, and click the **Submit** button. The GXE502X will automatically perform a soft reboot. Users can log back into the GXE502X web portal when the "ready" LED lights up.

Reset to default: Select the *Reset to Default* radio button, and enter the admin password to confirm, and then click the **Submit** button. The GXE502X will reboot itself, and will boot up with the factory default settings. The web portal administrator password will revert back to the default password:"admin."

12. Status

All the necessary information about the GXE502X is displayed in the status menu.

System Statistics			
Product Model	GXE5028	Hardware Version:	V0.1 A
Bootloader Version:	1.0.1.15	Core Version:	1.0.1.35
Base Version:	1.0.1.35	Firmware Version:	1.0.1.35
WAN MAC Address:	00:0B:82:11:DD:08	LAN MAC Address:	00:0B:82:11:DD:09
System Uptime Since:	2009-02-26 21:03	System Current Time:	2009-02-28 00:05:21
Network Status			
WAN Port Link Status:	Plugged	WAN IP Address:	192.168.1.20
LAN Port Link Status:	Unplugged	LAN IP Address:	192.168.10.1
UPNP NAT Passthrough work:	No	Mapped IP:port:	
WAN-side NAT Detected:	Unknown	DDNS Status:	Disable
PPPoE Link Status:	Disable	Active DHCP Clients:	0
Peripheral Status			
Phone/Fax Port 1:	Idle	Phone/Fax Port 2:	Idle
PSTN Line 1	Unplugged	PSTN Line 2	Unplugged
PSTN Line 3	Unplugged	PSTN Line 4	Unplugged
PSTN Line 5	Unplugged	PSTN Line 6	Unplugged
PSTN Line 7	Unplugged	PSTN Line 8	Unplugged
USB Port:	Disconnected	Music-On-Hold Port:	Unplugged
C55 Status:	Normal		

Status Menu Summary:

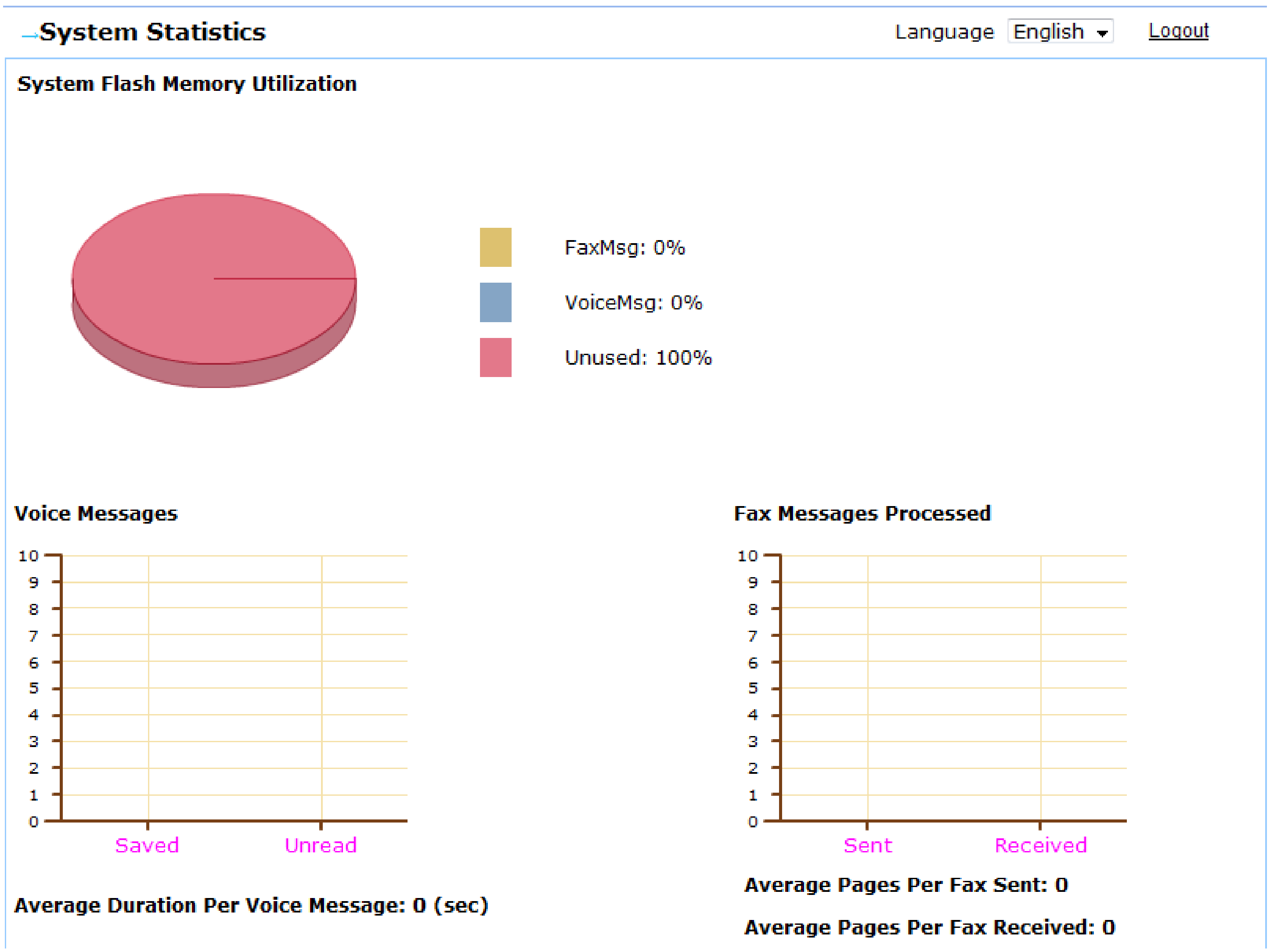
- *System Statistics*: System ID, MACs, hardware/firmware version, system uptime, etc are displayed.
- *Network Status*: Network connection information, IP of WAN and LAN, PPPoE status, DDNS.
- *Peripheral Status*: Displays the connection status of the FXO and FXS ports.
- *User Activity Status*: Displays the current user and registration status as well as the status of active calls.
- *DHCP Clients Table:* Lists the DHCP clients connected via the LAN port of GXE502X.
- *Debug Info:* Displays information used for debugging and troubleshooting.

13. Reports

In the **Reports** menu, users can access very useful reports generated by the GXE502X that document System Statistics, Call Statistics and Call Records. The Call Records can also be downloaded from the GXE502X for 3rd party billing analysis.

- System Statistics

The System Statistics sub menu section displays information regarding memory usage, faxes, and voice messages via pie charts and bar graphs.



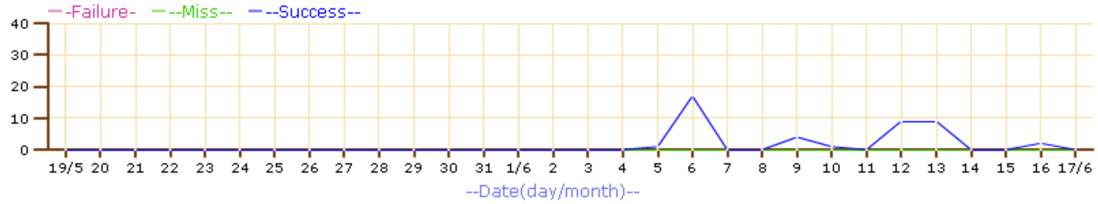
- Call Statistics

The Call Statistics sub menu page displays information regarding the number and duration of inbound and outbound calls through various trunks (PSTN, SIP, Peer Systems).

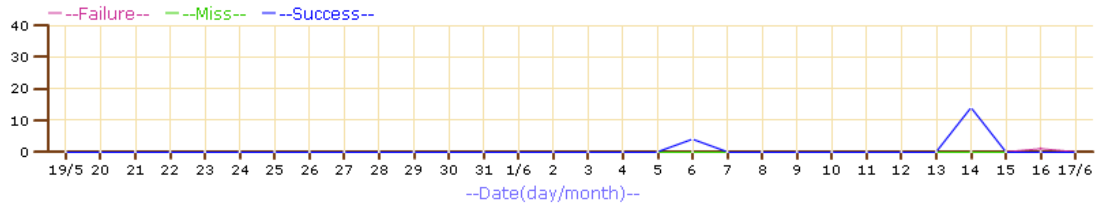
All of these statistics will be reset upon rebooting the GXE502X.

Call Statistics Language English **Logout**

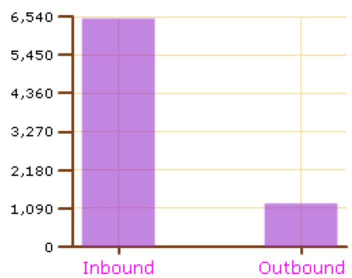
Internal PSTN Inbound Calls



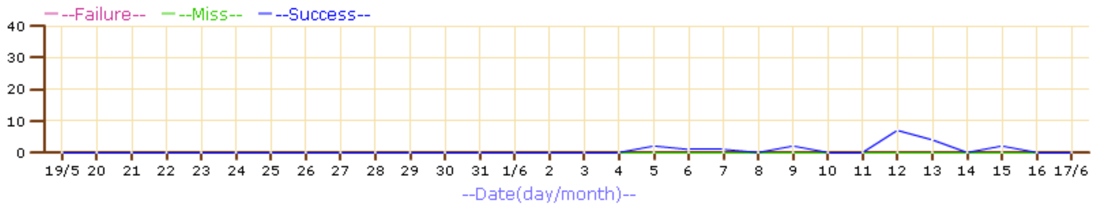
Internal PSTN Outbound Calls



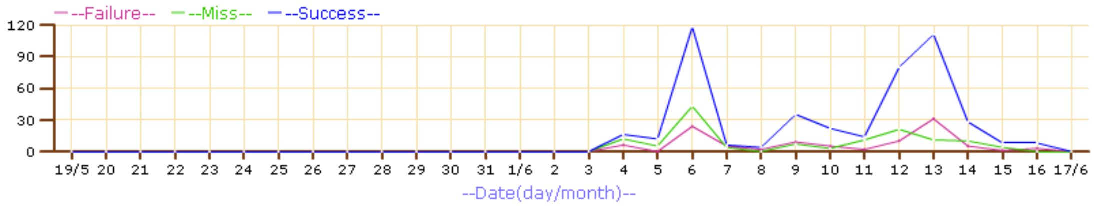
Internal PSTN Calls Total Duration(sec)



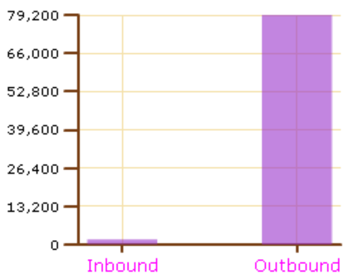
Sip Trunk Inbound Calls



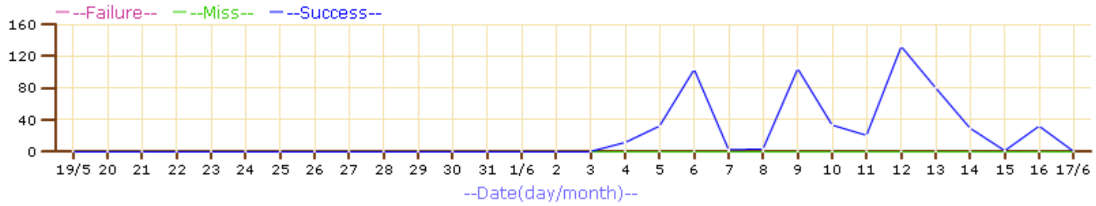
Sip Trunk Outbound Calls



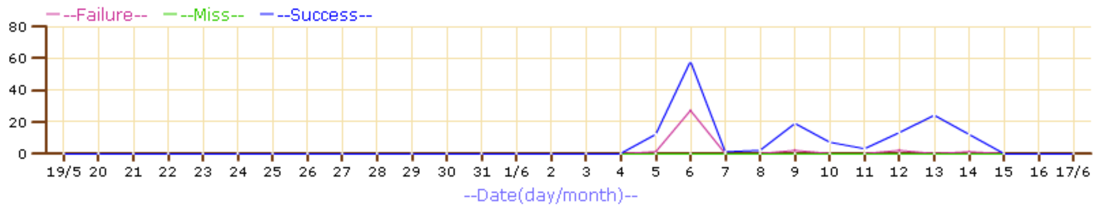
Sip Trunk Calls Total Duration(sec)



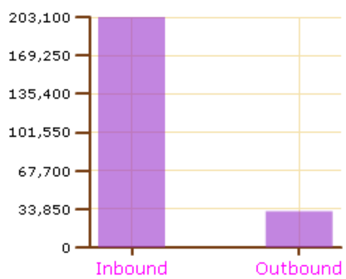
External PSTN Inbound Calls



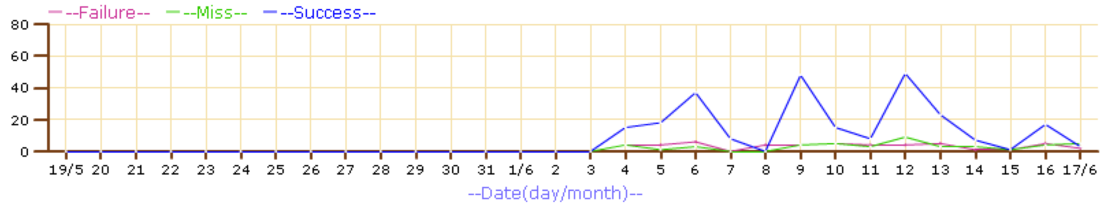
External PSTN Outbound Calls



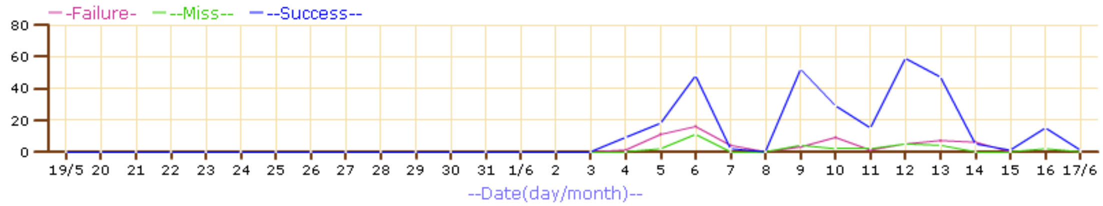
External PSTN Calls Total Duration(sec)



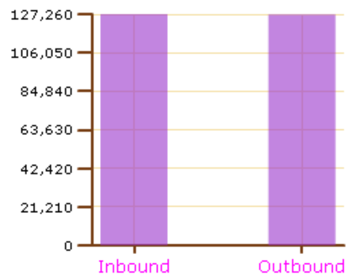
Peer System Outbound Calls



Peer System Inbound Calls

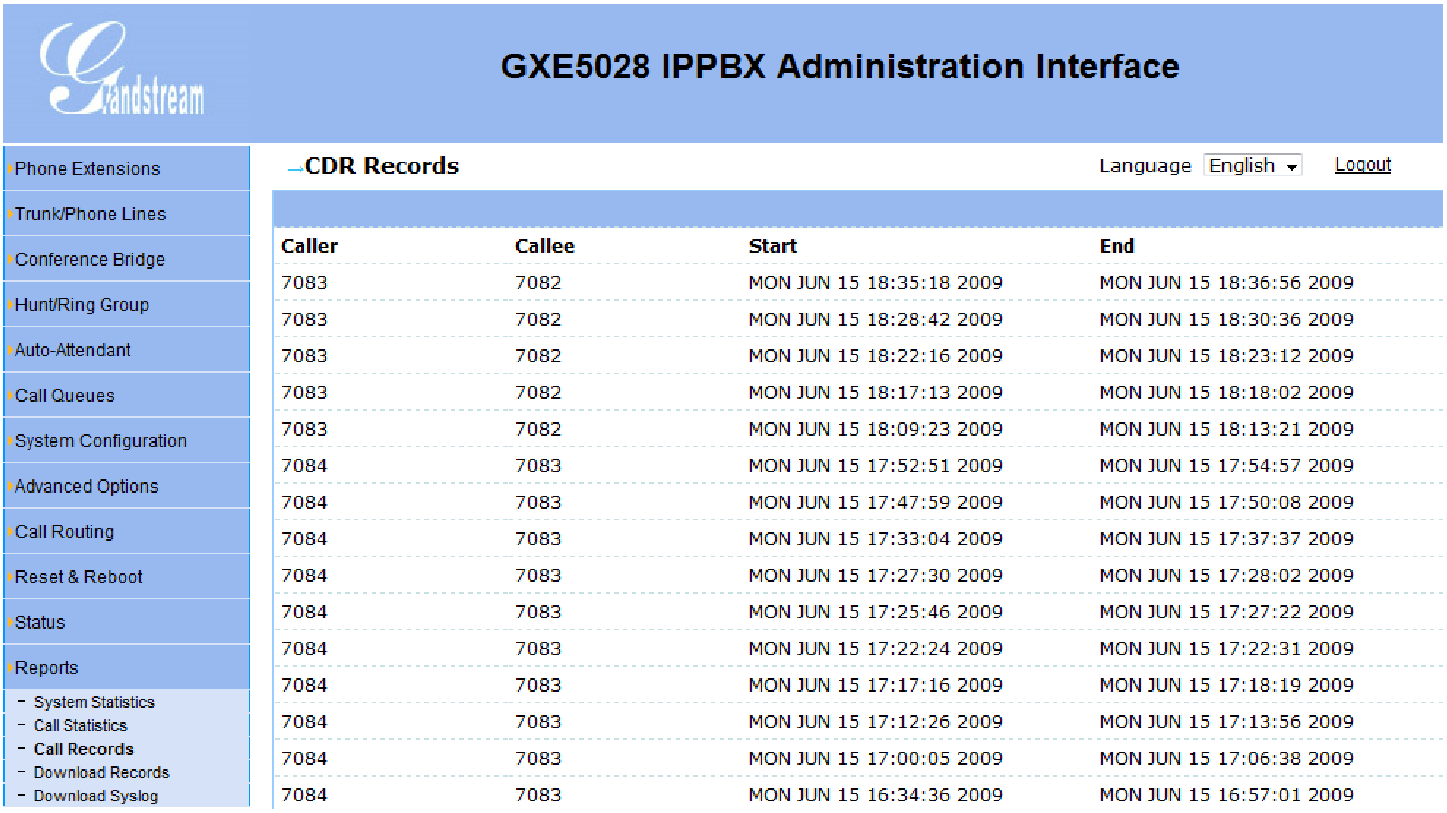


Peer System Calls Total Duration(sec)



- Call Records

The Call Records sub menu displays detailed information such as the Caller/Callee extension number as well as the start and end time for each inbound and outbound call.



Caller	Callee	Start	End
7083	7082	MON JUN 15 18:35:18 2009	MON JUN 15 18:36:56 2009
7083	7082	MON JUN 15 18:28:42 2009	MON JUN 15 18:30:36 2009
7083	7082	MON JUN 15 18:22:16 2009	MON JUN 15 18:23:12 2009
7083	7082	MON JUN 15 18:17:13 2009	MON JUN 15 18:18:02 2009
7083	7082	MON JUN 15 18:09:23 2009	MON JUN 15 18:13:21 2009
7084	7083	MON JUN 15 17:52:51 2009	MON JUN 15 17:54:57 2009
7084	7083	MON JUN 15 17:47:59 2009	MON JUN 15 17:50:08 2009
7084	7083	MON JUN 15 17:33:04 2009	MON JUN 15 17:37:37 2009
7084	7083	MON JUN 15 17:27:30 2009	MON JUN 15 17:28:02 2009
7084	7083	MON JUN 15 17:25:46 2009	MON JUN 15 17:27:22 2009
7084	7083	MON JUN 15 17:22:24 2009	MON JUN 15 17:22:31 2009
7084	7083	MON JUN 15 17:17:16 2009	MON JUN 15 17:18:19 2009
7084	7083	MON JUN 15 17:12:26 2009	MON JUN 15 17:13:56 2009
7084	7083	MON JUN 15 17:00:05 2009	MON JUN 15 17:06:38 2009
7084	7083	MON JUN 15 16:34:36 2009	MON JUN 15 16:57:01 2009

- Downloading (Call) Records

The CDR (Call Detail Record) can be downloaded from the GXE502X to your computer and exported to 3rd party billing software companies for analysis.

The file is in the .csv format and stored on a weekly (7 Day) basis with a maximum record of 10000 calls. It can be exported to an Excel Spreadsheet, other another compatible database or program for processing and printing.

Click the “Download” button to the right of file name to download the record to your computer.


GXE5028 IPPBX
Phone Extensions
Trunk/Phone Lines
Conference Bridge
Hunt/Ring Group
Auto-Attendant
Call Queues
System Configuration
Advanced Options
Call Routing
Reset & Reboot
Status
Reports
- System Statistics
- Call Statistics
- Call Records
- Download Records
- Download Syslog
Download Records
Record Name
20090614-20090620.csv Download
20090607-20090613.csv Download
20090531-20090606.csv Download
20090517-20090523.csv Download
20090510-20090516.csv Download
20090503-20090509.csv Download
20090524-20090530.csv Download
20090426-20090502.csv Download

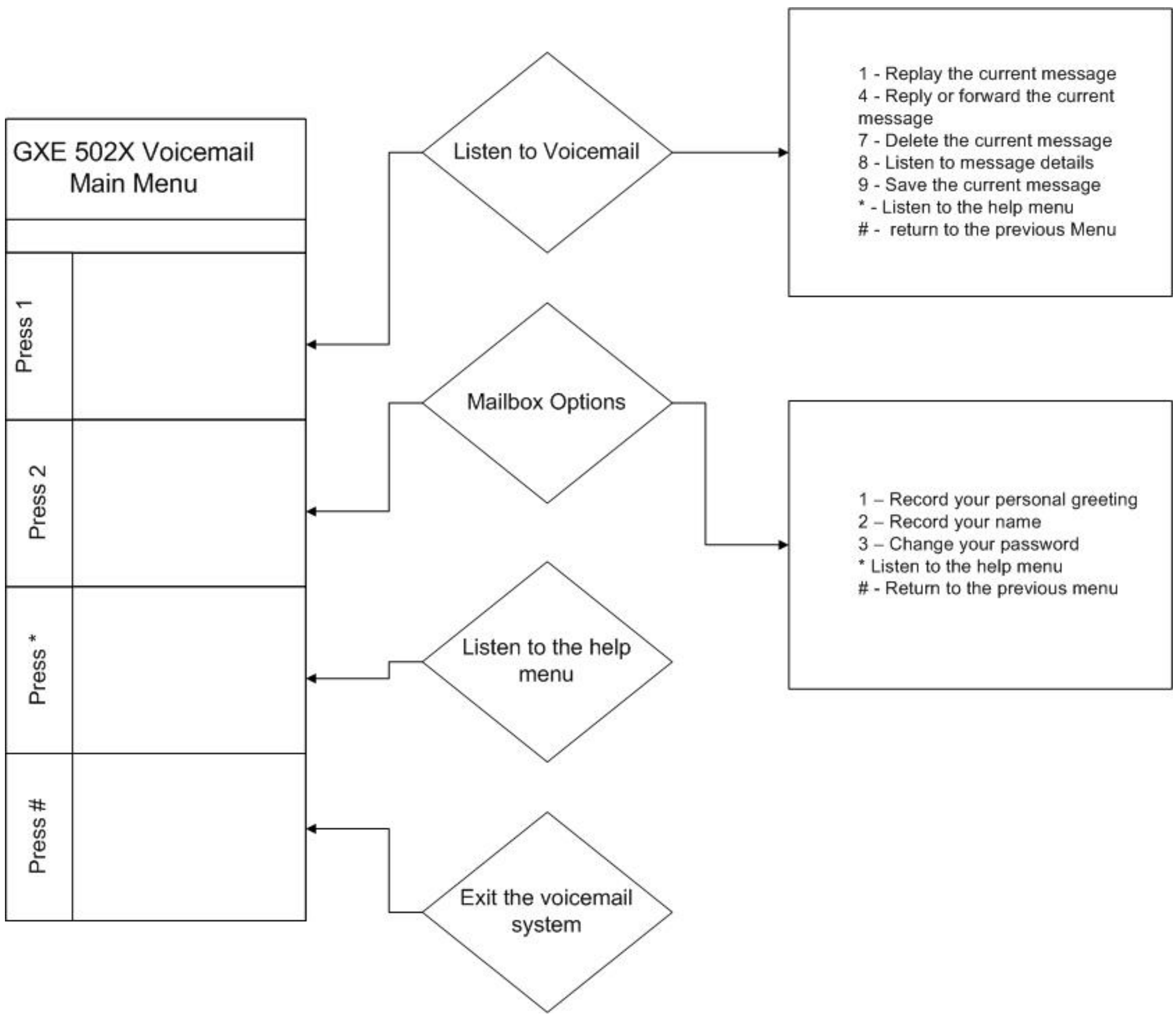
14. Voicemail Configuration

The GXE502X allows users to manage voicemail via the IVR in their phones or through a personal web portal. This section summarizes how to manage voicemail and other settings using both of these methods.

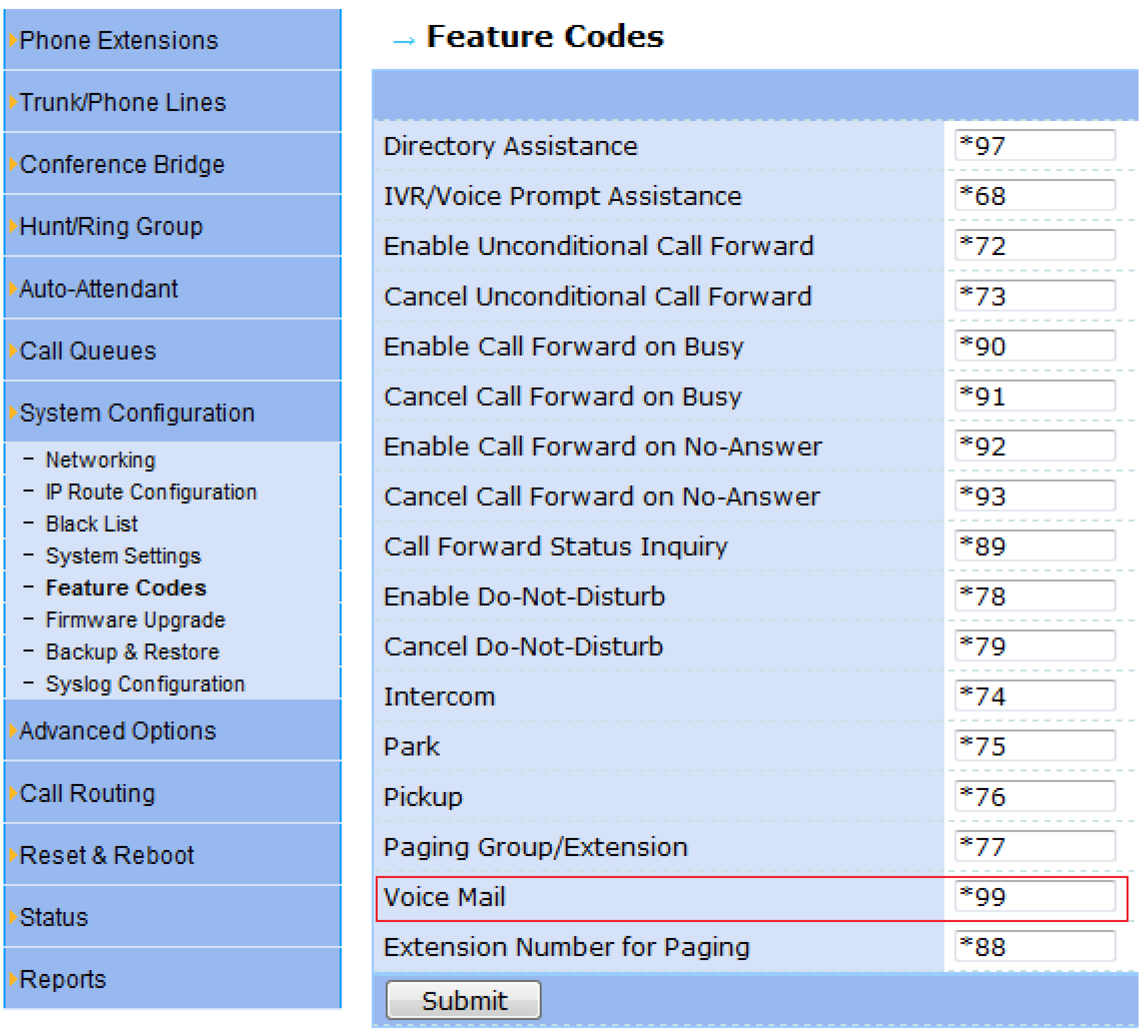
- **Configuring Voicemail through the IVR**

The default feature code for the voicemail IVR system is *99. After dialing this code users must enter their extension number and password. Once logged in, users are prompted by a basic IVR menu with multiple choices.

Following is a Quick Usage Flow Chart, for more detail Voicemail flow, please refer to appendix A

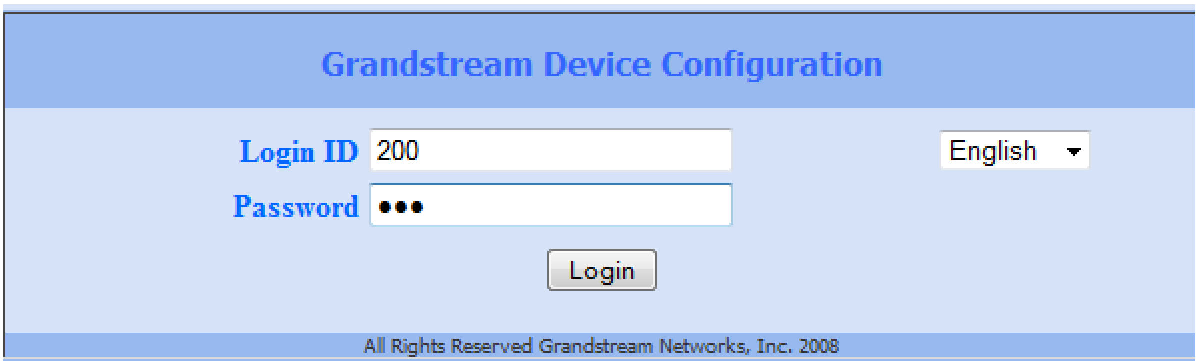


Important: - The feature code used for voice mail can be found and modified under System configuration ➔ Feature codes (see the picture below).
- Users can access the voicemail system from any phone connected to the GXE502X by dialing the voicemail feature code.
- In order to access to voicemail, the GXE502X will ask for the extension number and password. The default voicemail password is the extension number. Please make note that both the GXE502X and all registered SIP phones must be configured with the same Send DTMF setting. The default Send DTMF mode on the GXE502X is "via RFC2833".

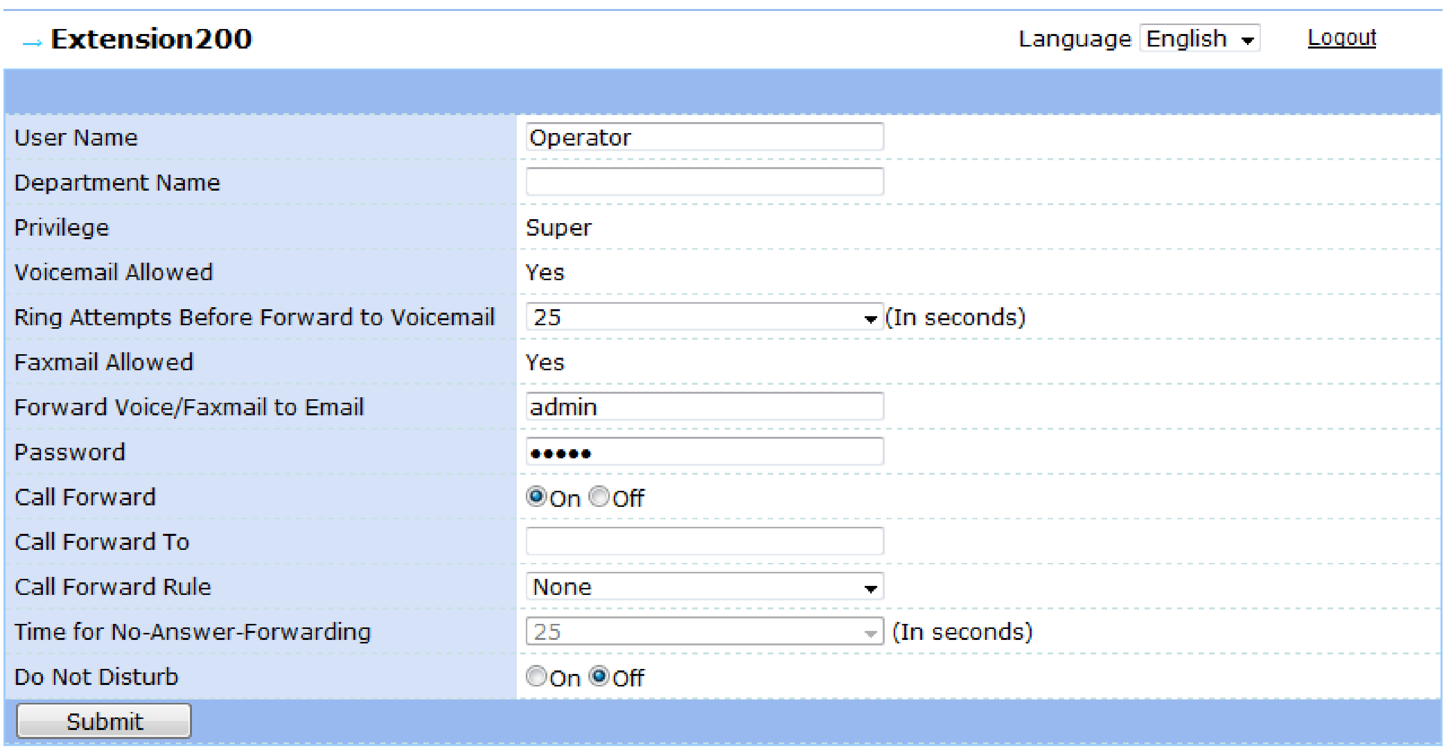


15. Personal Web Portal

1. To log into the personal web portal for an extension. Log into the GXE502X Web GUI using the extension number as Login ID and voicemail password. If you have changed the default voicemail password via the voicemail IVR phone, you must input the updated password.

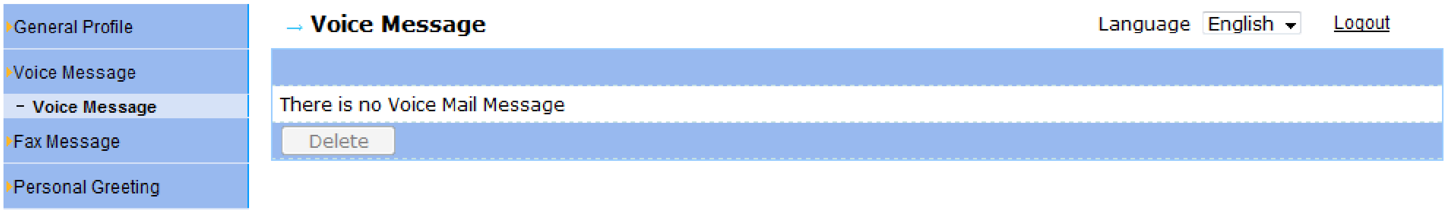


After logging in, the following page will display

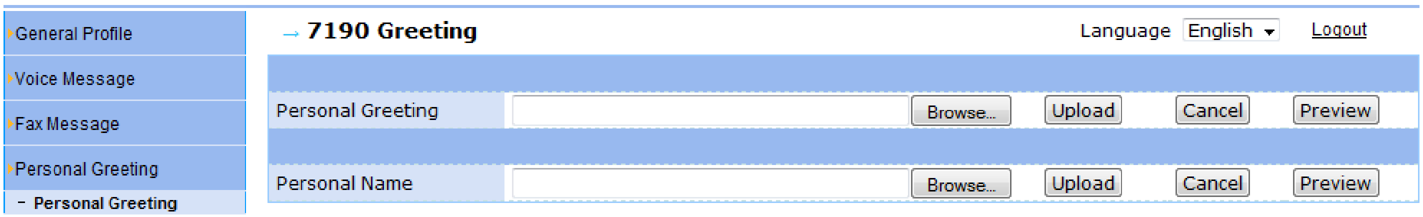


2. To view current voice mails, click the *"Voice Message"* sub menu on the left hand side of the interface. Doing this will load the voice mail management page. Users can manage (save & delete) the messages here as an alternative to using the voicemail IVR.

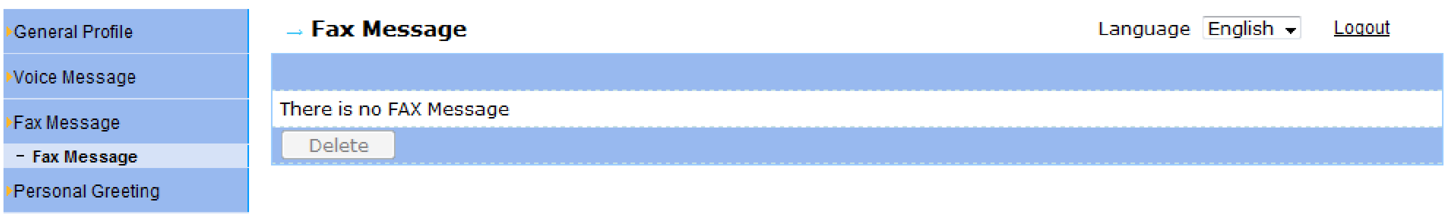
   This is a great feature for users that are constantly on the road.



3. To view the personal greeting, click the *"Personal Greeting"* sub menu button on the left hand side of the GUI. This will load the personal greeting management page for the extension. Users can upload .wav files from a PC and preview them here.

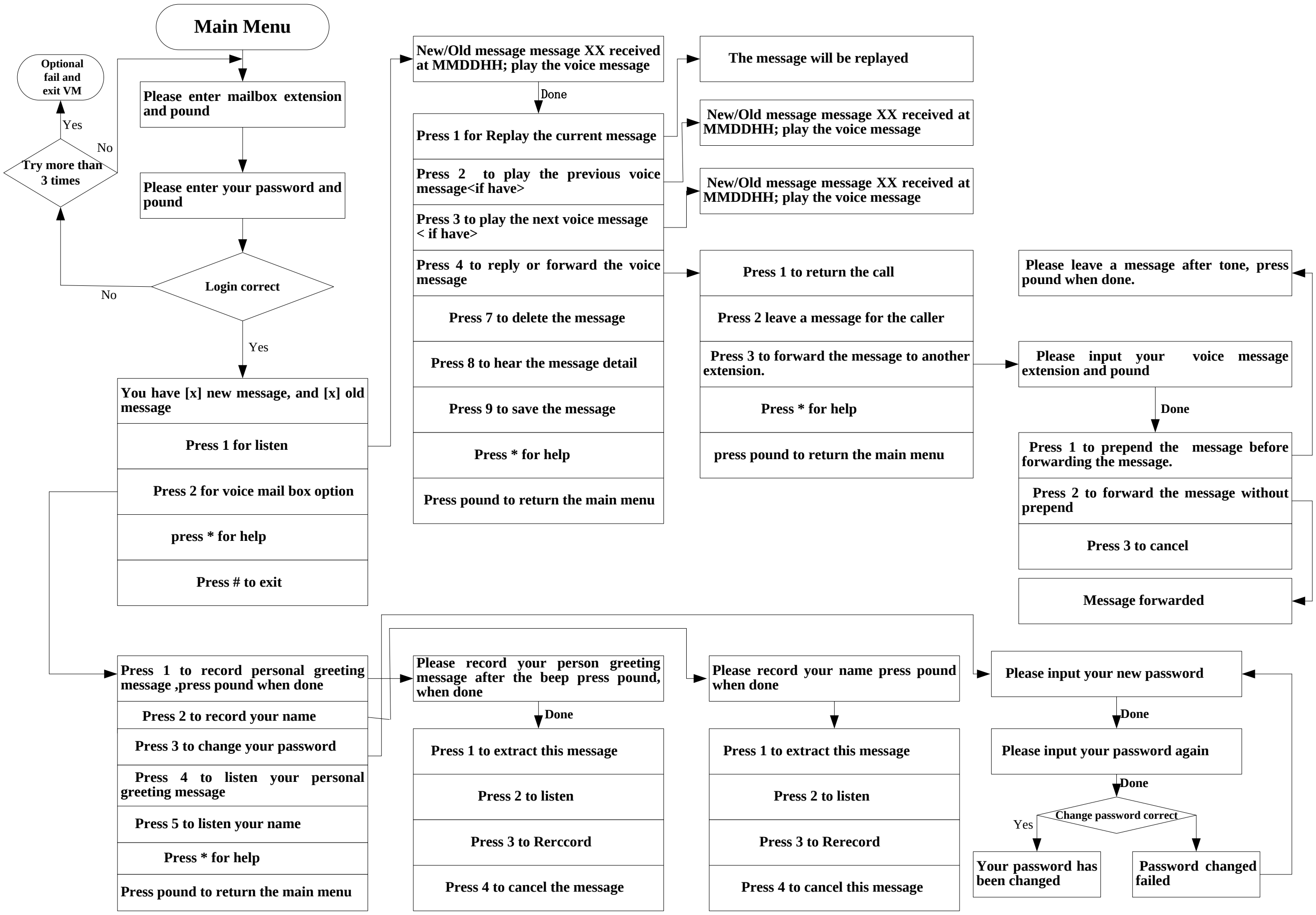


4. Click the Fax Message sub menu button on the left hand menu to load the fax message management page. Faxmail can be managed (save & delete) in a similar fashion to voicemail



Important: - If you do not know the URL or IP address to access to the Personal Web Portal, please contact your the local system administrator.
- Users can also receive voice and fax messages via email. To receive them, **users** must specify a valid email address in the "Forward voice/fax mail to email" field. **The administrator** should enable this feature by configuring the SMTP server in the system settings.
- The default voicemail/personal web portal password is the extension number. The same applies to the SIP registration password on the GXE502X. Users can change this via the IVR (press 2 ➔ press 3) or via the personal web portal (under General Profile).
- Changing the voicemail/personal web portal password will NOT change the SIP password. The SIP password can only be changed by the Administrator.

Appendix A: Complete Voicemail Flow Chart



Appendix B: Template File

1. What is a template file?

   A template file is a file that allows GXE5000 system administrator to Auto-Provision a specific terminal device(Grandstream ONLY) based on the configuration parameters set forth in the template file, with certain parameters that are configurable by GXE5000 itself.

   Template file is based on the Configuration Template that specified in http://www.grandstream.com/configurationtool.html, with 4 different type of variables that is automatically assigned by GXE5000 to certain P parameters.

   The 4 variables are:

   - Extension Number: &GS extension-XX&, here XX are numbers.
     For example, though by default only account one is configured, you can create a template that configure all 4 extensions in GXP-2000 using following 4 variables:

     &GS extension-0&
     &GS extension-1&
     &GS extension-2&
     &GS extension-3&

   - SIP Password for the extension: &GS password-XX&, here XX are numbers. For example:
     &GS password -0&
     &GS password -1&
     &GS password -2&
     &GS password -3&

   - SIP Server: &GS sipserver&
   - SIP Server Port: &GS sipserver-port&

Following template file with file name “GXP-2000-base-provision” is to configure account 1 and account 2 with account 1 to use G.729 for its codec ONLY.

```
#*************************************************************
            P35=&GS extension-0&
            P36=&GS extension-0&
            P34=&GS password-0&
            P47=&GS sipserver&:&GS sipserver-port&
            P57 = 18
            P58 = 18
            P59 = 18
            P60 = 18
            P61 = 18
            P62 = 18
            P46 = 18
            P98 = 18
            P237=&GS sipserver&
            P402=&GS sipserver&:&GS sipserver-port&
```

P404=&GS extension-1&
P405=&GS extension-1&
P406=&GS password-1&
#***

2. How does template file become configuration file and loaded into Grandstream product?

   When Grandstream Terminal Device boot up in the same LAN with GXE5000(LAN Port), it will go through following process:
   - Issue DHCP Request, acquire both IP address and the TFTP server IP through DHCP Option 66, here GXE5000 function as DHCP Server.
   - Issue TFTP request for configuration file with name “cfg000b82xxxxxx”. Here the TFTP request contains Grandstream proprietary extensions that contain Grandstream Product model information, see following example:

     Trivial File Transfer Protocol
     Opcode: Read Request (1)
     Source File: /GXV-3000/cfg000b8209ba7d
     Type: octet
     Option: blksize = 1024
     Option: tsize = 0
     Option: timeout = 4
     Option: grandstream_MODEL = GXV-3000
     Option: grandstream_NAT = 1
     Option: grandstream_ID = 000b8209ba7d
     Option: grandstream_REV_BOOT = 001.001.003.002
     Option: grandstream_REV_PHONE = 001.001.003.014

   - When GXE5000 received the TFTP request, it will base on the product model to look up the template file and generate the device configuration file dynamically.

3. Template file name format.

   Product-Model-base-provision, here **Product-Model** has to be the product model shown in the tftp option, e.g., GXP-2000 or GXV-3000, their template file will be GXP-2000-base-provision and GXV-3000-base-provision, respectively.

4. Template file management.

   Currently GXE5000 contains following product templates:

   Unknown/BT-100/BT-110/BT-200/GXP-2000/GXP-2020/GXV-3000/GXW-4004/GXW-4008/HT-286/HT-386/HT-487/HT-488/HT-496

   Here Unknown is a special template that it will only assign variables to P34/P35/P36/P47, in case a TFTP request does not contain a known product model or no model option in the TFTP request.