



GXE5024 Users Manual



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1. Introduction

Thank you for purchasing the Grandstream GXE502x IP-PBX, a SIP-based, affordable, high-quality converged communications platform designed to enhance small to medium business enterprises.

The GXE502x is capable of providing the cutting-edge IP-based communications to businesses demanding the latest of technologies, while leveraging existing infrastructure and providing a most friendly transition into IP telephony for others. Supporting open-standard SIP, the GXE502x can easily integrate into and interoperate with other components of your IP-based 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.

An auto-provisioning Express Setup wizard and an intuitive user interface allow the GXE502x to be set up and installed quickly and confidently. Voicemail, voicemail-to-email, conference bridges, and other enhanced features can be enabled and functioning with just minimal effort via user-friendly web configuration pages. Its broad feature set, ease of operation, and quality and value makes it ideal for any business communication environment.

- 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. Administering Extensions on the GXE

Extensions for the internal users can be created from the **Express Setup** menu of a factory-default GXE, or from the **Phone Extensions** menu at any time. This can be local users in the internal network, or even remote users connecting over the open Internet. This flexibility allows users to have their office extension with them virtually anywhere, keeping accessibility to personnel high without incurring any PSTN toll charges for the worker at home or the road-warrior.

The **Express Setup** provides a quick wizard to complete initial setup of all extensions on a new system, including auto-provisioning of the phones. The **Phone Extensions** menu allows you to add, batch add, modify, reboot, delete and batch delete extensions.

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

GXE5028 IPPBX Administration Interface

Phone Extensions

Trunk/Phone Lines

Conference Bridge

Hunt/Ring Group

Auto-Attendant

System Configuration

Advanced Options

Express Setup

[Logout](#)

General System Configuration:

| | |
|-------------------------------------|----------------------------------|
| Extension Length | <input type="text" value="4"/> |
| Extension Digit Prefix | <input type="text" value="6"/> |
| Identifier of this IPPBX | <input type="text" value="010"/> |
| <input type="button" value="Next"/> | |

The Express Setup wizard will allow you to set various system extensions on the GXE and create your phone extensions. It will also auto-provision your phones with the extensions you create. Please refer to the *GXE Quick Start Guide* for detailed instructions on setting up your GXW with the Express Setup wizard.

- Using the Phone Extensions menu to create, modify, view, and delete extensions.

In the General Settings section, the following settings can be set.

GXE5028 IPPBX Administration Interface

Phone Extensions

Extensions Directory

[General Settings](#)

Trunk/Phone Lines

Conference Bridge

Hunt/Ring Group

Auto-Attendant

System Configuration

Advanced Options

Status

General Settings

[Logout](#)

| | | |
|---|-----------------------------------|---|
| Extension Length | <input type="text" value="4"/> | |
| Leading Digit of Extensions | <input type="text" value="6"/> | (common prefix digits shared among all extensions on this system, up to 10 digits) |
| Local IPPBX Number | <input type="text" value="010"/> | |
| Extension Number for Paging | <input type="text" value="6992"/> | |
| Auto-Attendant | <input type="text" value="6993"/> | |
| Extension Number for Internal Phone/FAX Ports | <input type="text" value="6990"/> | for Port 1: <input type="text" value="6991"/> for Port 2: <input type="text" value=""/> |

Extension Length: all extensions must have the number of digits specified here.

Leading Digit of Extensions: all extensions must begin with this digit.

Local IPPBX Number: a unique identifier number for this local GXE system.

Extension Number for Paging: users may dial this number for group paging.

Auto-Attendant: set the extension to reach the auto-attendant menu here.

Extension Number for Internal Phone/FAX ports: Set the extension number for the TEL1 and TEL2 FXS ports on the back of the GXE here.

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

In the Extensions Directory section, all SIP phone extensions configured on the GXE, their details, and their registration status are displayed. The following actions can be performed in this section.

GXE5028 IPPBX Administration Interface

Phone Extensions

[Extensions Directory](#)

[General Settings](#)

[Trunk/Phone Lines](#)

[Conference Bridge](#)

[Hunt/Ring Group](#)

[Auto-Attendant](#)

[System Configuration](#)

[Advanced Options](#)

[Status](#)

[Reports](#)

Extensions Directory

GREEN=local extension BLUE=remote extension [Logout](#)

| All <input type="checkbox"/> | Extension | Name | Department | Device Type | IP Address | Status | Privilege | Action |
|------------------------------|-----------|-----------|------------|-------------|------------|----------|-----------|---|
| <input type="checkbox"/> | 6000 | Operator | Reception | | | Off-line | Super | Modify Delete |
| <input type="checkbox"/> | 6001 | John Doe | Sales | | | Off-line | Regular | Modify Delete |
| <input type="checkbox"/> | 6002 | Jane Doe | Sales | | | Off-line | Regular | Modify Delete |
| <input type="checkbox"/> | 6003 | Jim Bob | Support | | | Off-line | Regular | Modify Delete |
| <input type="checkbox"/> | 6004 | Billy Bob | Support | | | Off-line | Regular | Modify Delete |
| <input type="checkbox"/> | 6005 | Janet Doe | Support | | | Off-line | Regular | Modify Delete |

[Delete](#)
[Add One Extension](#)
[Batch Add](#)

Add an extension: Click on the **Add One Extension** button. The extension details page will be displayed, allowing you to set all of the extension's settings. When done, click on the **Submit** button to add the extension or **Cancel** to go back.

Batch add extensions: Click on the **Batch Add** button. The next page allows you to specify the range of extensions to add, as well as some general extension settings to apply to them. You may go back to the extensions directory and modify the extensions to set extension-specific settings. When done, click on the **Submit** button to add the extensions or **Cancel** to go back.

Modify and/or reboot a SIP extension: Click on the **Modify** button to the right of the row displaying information for the extension you wish to modify. The extension details page will be displayed, allowing you to modify all of the extension's settings. To reboot the phone, set the *Reboot Peer* setting to "Yes". When done, click on the **Submit** button to modify and/or reboot the extension or **Cancel** to go back.

Delete: Click on the **Delete** button on the far right of the row displaying the information for 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.

Batch delete: Delete multiple extensions in one step by checking each checkbox of all extensions to be deleted, and clicking on the **Delete** button below the column of checkboxes. To check all displayed extensions on the current page, check the uppermost checkbox, labeled as "All". 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.

- Busy Lamp Field and Message Waiting Indication

Busy Lamp Field (BLF) support and Message Waiting Indication (MWI) support are enabled on the GXE. New voicemail messages will automatically trigger an MWI light on phones that support it to notify the user of new voicemail messages. To monitor the statuses of other users on the GXE, configure BLF keys on an IP phone to the extension numbers of

the users to be monitored. The GXE will trigger the BLF keys to signal when the monitored user is idle, has a call ringing, or is busy.

3. Configuring Trunks on the GXE

PSTN trunks via the FXO ports, SIP trunks, and SIP-based PSTN trunk gateways can be configured on the GXE in the **Trunk/Phone Lines** menu. Also, the FXO and FXS port line electrical settings can be configured here as well. This will allow users on the GXE to call destinations anywhere through the trunks configured on the GXE.

- Configuring internal PSTN trunks via FXO ports

In the Internal PSTN Trunk Line section, the FXO ports on the back of the GXE can be configured to interface with and send/receive calls to/from the PSTN.

Compatibility with PSTN lines can be achieved by adjusting the following types of settings. The default settings should function in most cases, depending on your regional PSTN line characteristics.

GXE5028 IPPBX Administration Interface

Phone Extensions

Trunk/Phone Lines

[Internal PSTN Trunk Line](#)

[Internal Phone/Fax Port](#)

[SIP Trunk](#)

[External PSTN Trunk Line](#)

Conference Bridge

Hunt/Ring Group

Auto-Attendant

System Configuration

Advanced Options

Status

Reports

Internal PSTN Trunk Line Configuration
[Logout](#)

FXO Termination

| | |
|----------------------------------|---|
| Enable Current Disconnect | <input checked="" type="radio"/> No <input type="radio"/> Yes |
| Current Disconnect Threshold(ms) | 200 (100-10000, default 200) |
| Enable Tone Disconnect | <input checked="" type="radio"/> No <input type="radio"/> Yes |
| Polarity Reversal Disconnect | <input checked="" type="radio"/> No <input type="radio"/> Yes |
| AC Termination Impedance | USA & Canada |

Call Progress Tones (Syntax: f1=val@vol,f2=val@vol,c=on1/off1/on2/off2/on3/off3;[...])

| | |
|----------------|-----------------------------------|
| Dial Tone | f1=350@-13,f2=440@-13,c=1000 |
| Ring Back Tone | f1=440@-19,f2=480@-19,c=2000/4000 |
| Busy Tone | f1=480@-24,f2=620@-24,c=500/500 |

Line Dialing Settings

| | | |
|---------------------------|------------|-----------------------|
| DTMF Digit Length(ms) | ch1-8:80 | (Example: ch1-8:35;) |
| DTMF Digit Volume(dBm) | ch1-8:-11 | (Example: ch1-8:-11;) |
| DTMF Dial Pause(ms) | ch1-8:100 | (Example: ch1-8:35;) |
| Wait for Dial-Tone(Y/N) | ch1-8:N | (Example: ch1-8:Y;) |
| Minimum Delay Before Dial | ch1-8:1000 | (Example: ch1-8:100;) |

Caller ID Standard

| | |
|------------------|----------|
| Caller ID Scheme | Bellcore |
|------------------|----------|

FXO Termination: Specify line disconnect signaling and impedance.

Call Progress Tones: Configure tones to match regional settings.

Line Dialing Settings: Adjust line dialing properties to adhere to PSTN line requirements.

Caller ID Standard: Set caller ID handling to match PSTN settings.

The following Line Call Control settings allow you to specify how inbound and outbound calls are to be handled.

GXE5028 IPPBX Administration Interface

Phone Extensions

Trunk/Phone Lines

[Internal PSTN Trunk Line](#)

Internal Phone/Fax Port

SIP Trunk

External PSTN Trunk Line

Conference Bridge

Hunt/Ring Group

Auto-Attendant

System Configuration

Advanced Options

Status

Reports

| | | |
|---------------------------|------------|----------------------|
| Wait for Dial-Tone(Y/N) | ch1-8:N | (Example:ch1-8:Y;) |
| Minimum Delay Before Dial | ch1-8:1000 | (Example:ch1-8:100;) |

Caller ID Standard

Caller ID Scheme Bellcore

Line Call Control

| | | | | | |
|---|---------------|---|--|--|--|
| 1. Line 1-4 | (e.g., 1-4,7) | Dial Prefix 9 | Inbound Call Answer Auto-Attendant | None | Delete |
| 2. Line 5-7 | (e.g., 1-4,7) | Dial Prefix 8 | Inbound Call Answer Direct Extension | 6000 | Delete |
| 3. Line 8 | (e.g., 1-4,7) | Dial Prefix 7 | Inbound Call Answer Voice Mail | None | Delete Add |

Allow Other Peer Systems Use None

Submit Cancel

Line: Specify a port or a range of ports to apply the same line call control rule to.

Dial Prefix: Configure the prefix digit on outbound calls to specify this trunk. When the call is sent out via this trunk, the prefix digit is removed.

Inbound Call Answer: In the first drop-down box of each row, select the type of destination for inbound calls on this trunk. Select the exact destination of that type in the second drop-down box, if applicable.

Delete: Delete an existing line call control rule.

More: Allow another line call control rule to be set.

Allow Other IPPBX Use: Allow other peered IPPBXs to dial through the local GXEs internal PSTN trunks.

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

- Configuring internal phone/fax devices via FXS ports

In the Internal Phone/Fax Port section, the line electrical and DTMF signaling settings of the FXS ports on the back of the GXE can be configured for compatibility with analog phone/fax devices. The extension numbers for these ports are set in the **Phone Extensions** menu, under the General Settings section.

GXE5028 IPPBX Administration Interface

Phone Extensions
Trunk/Phone Lines
Internal PSTN Trunk Line
Internal Phone/Fax Port
SIP Trunk
External PSTN Trunk Line
Conference Bridge
Hunt/Ring Group
Auto-Attendant
System Configuration
Advanced Options

Internal Phone/Fax Port

Logout

| | | |
|---------------------------------|--|--|
| DTMF Transport Type | ch1-2:T=2833 | (Example:ch1:T=<2833/audio/signal>;ch2:T=<2833/audio/signal>;) |
| SLIC Setting | USA | |
| Caller ID Scheme | Bellcore/Telcordia | |
| Polarity Reversal | <input checked="" type="radio"/> No <input type="radio"/> Yes (reverse polarity upon call establishment and termination) | |
| Current Disconnect | <input type="radio"/> No <input checked="" type="radio"/> Yes | |
| Current Disconnect Duration(ms) | 200 | (100-10000) |
| Hook Flash Timing(ms) | Min 100 (100-2000), Max 500 (100-2000) | |

Submit

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

- Configuring SIP trunks

In the SIP Trunk section, SIP trunks can be viewed, created, or modified. All configured SIP trunks as well as their details and current status are displayed. The following actions can be performed in this section.

Phone Extensions
Trunk/Phone Lines
Internal PSTN Trunk Line
Internal Phone/Fax Port
SIP Trunk
External PSTN Trunk Line

SIP Trunk

Logout

| Name | Active | SIP Server URL | Account ID | Max Concurrent Calls | Outbound Prefix | Status | Action |
|--------|--------|------------------------|------------|----------------------|-----------------|--------------|-------------------------------------|
| Trunk1 | Enable | trunk1.grandstream.com | 8 | 8 | 5 | Disconnected | <div>Modify</div> <div>Delete</div> |
| Trunk2 | Enable | trunk2.grandstream.com | 8 | 8 | 4 | Disconnected | <div>Modify</div> <div>Delete</div> |

Add

Add: Click on the **Add** button. The SIP trunk details page will be displayed, allowing you to enter SIP account registration information.

GXE5028 IPPBX Administration Interface

Phone Extensions

Trunk/Phone Lines

Internal PSTN Trunk Line

Internal Phone/Fax Port

SIP Trunk

External PSTN Trunk Line

Conference Bridge

Hunt/Ring Group

Auto-Attendant

System Configuration

Advanced Options

Status

Reports

Add SIP Trunk

[Logout](#)

| | | | |
|------------------------------|---|-----------|--|
| Trunk Name | <input type="text"/> | | |
| Trunk Active | <input type="radio"/> Enable <input checked="" type="radio"/> Disable | | |
| SIP Server URL | <input type="text"/> | | |
| Outbound Proxy URL | <input type="text"/> | | |
| Account Name | <input type="text"/> | | |
| Account ID | <input type="text"/> | | |
| Authenticate ID | <input type="text"/> | | |
| Password | <input type="text"/> | | |
| Registration Retry Interval | <input type="text" value="600"/> | second(s) | |
| Heart Beat | <input checked="" type="radio"/> Yes <input type="radio"/> No | | |
| Max Concurrent Calls Allowed | <input type="text" value="8"/> | | |
| Dial Prefix | <input type="text"/> | | |
| Prepend Prefix | <input type="text"/> | | |
| Inbound Call Answer | <input type="text" value="Auto-Attendant"/> <input type="text" value="None"/> | | |
| Unregister On Reboot | <input type="radio"/> Yes <input checked="" type="radio"/> No | | |

Enable the trunk: Enter a descriptive name in the *Trunk Name* field, and set *Trunk Active* to *Enable*.

Set SIP registration: Enter the SIP server and, if available, outbound proxy information along with SIP account and password details. Set the registration expiration time in the *Registration Retry Interval* setting, and set *Heart Beat* to *Yes* to Enable or disable availability detection of the remote end (requires compatibility on the remote end)..

Outbound call handling: Set the *Max Concurrent Calls Allowed* to prevent too many concurrent attempts to send calls out this trunk.

Set the *Dial Prefix* digit to specify an outbound call to be dialed through this trunk. When the call is sent out via this trunk, the dial prefix digit is removed.

The *Prepend Prefix* setting allows a prefix to be automatically added by the GXE to the outbound dialed digits.

Inbound call handling: Set the destination to route incoming calls on this trunk to.

Set *Unregister on Reboot* to *Yes* if the SIP server allows it.

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

Modify: Click on the **Modify** button to the right of the row displaying information for the SIP trunk you wish to modify. The SIP trunk details page will be displayed, allowing you to modify all of the SIP trunk's settings. When done, click on the **Submit** button to save your changes or **Cancel** to go back.

Delete: Click on the **Delete** button on the far right of the row displaying the information for the SIP trunk you wish to delete. You will be prompted for confirmation via a dialog box; click **OK** to confirm or **Cancel** to go back.

- Configuring external PSTN trunk gateways via SIP

In the External PSTN Trunk Line section, external PSTN trunk gateways can be viewed, created, or modified. All configured external PSTN trunks as well as their details and current status are displayed. The following actions can be performed in this section.

The screenshot shows the 'GXE5028 IPPBX Administration Interface' with a sidebar on the left containing links like 'Phone Extensions', 'Trunk/Phone Lines', 'Internal PSTN Trunk Line', 'Internal Phone/Fax Port', 'SIP Trunk', and 'External PSTN Trunk Line'. The main content area is titled 'External PSTN Trunk Line' and includes a 'Logout' link. It displays a table with columns: Name, Active, Gateway URL, Other UDP, Concurrent calls, Outbound Prefix, and Action. Two rows are shown: one for 'GXW4104' and another for 'GXW4108', both with 'Enable' status and 'Modify'/'Delete' buttons. An 'Add' button is at the bottom left of the table.

| Name | Active | Gateway URL | Other UDP | Concurrent calls | Outbound Prefix | Action |
|---------|--------|----------------------|-----------|------------------|-----------------|---|
| GXW4104 | Enable | gxw1.grandstream.com | | 4 | 3 | Modify Delete |
| GXW4108 | Enable | gxw2.grandstream.com | | 8 | 2 | Modify Delete |

[Add](#)

Add: Click on the **Add** button. The external PSTN trunk details page will be displayed, allowing you to enter SIP peer connection information to connect with the external PSTN trunk gateway.

The screenshot shows the 'GXE5028 IPPBX Administration Interface' with a sidebar on the left. The main content area is titled 'Add External PSTN Trunk Line' and includes a 'Logout' link. It contains a form with fields for 'Trunk Name', 'Active' (radio buttons for Enable/Disable), 'SIP Gateway URL', 'Other UDP List', 'Heart Beat' (radio buttons for Yes/No), 'Max Concurrent Calls Allowed' (text input), 'Dial Prefix', 'Inbound Call Answer' (dropdown menu), and 'Allow Other IPPBX Use' (radio buttons for Yes/No). 'Submit' and 'Cancel' buttons are at the bottom.

Enable the trunk: Enter a descriptive name in the *Trunk Name* field, and set *Trunk Active* to *Enable*.

Set SIP peer information: Enter the SIP gateway IP address or domain name in the *SIP Gateway URL* field.

Set *Heart Beat* to *Yes* to Enable or disable availability detection of the remote side (requires compatibility on the remote end)..

Outbound call handling: Set the *Max Concurrent Calls Allowed* to prevent too many concurrent attempts to send calls out this trunk.

Set the *Dial Prefix* digit to specify an outbound call to be dialed through this trunk. When the call is sent out via this trunk, the dial prefix digit is removed

Inbound call handling: Set the destination to route incoming calls on this trunk to.

Share trunk with peer systems: If this trunk is to be available for use by peer systems, set the *Allow Other IPPBX Use* setting to *Yes*.

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

Modify: Click on the **Modify** button to the right of the row displaying information for the external PSTN trunk you wish to modify. The external PSTN trunk details page will be displayed, allowing you to modify all of the external PSTN trunk's settings. When done, click on the **Submit** button to save your changes or **Cancel** to go back.

Delete: Click on the **Delete** button on the far right of the row displaying the information for the external PSTN trunk you wish to delete. You will be prompted for confirmation via a dialog box; click **OK** to confirm or **Cancel** to go back.

4. Configuring Conference Bridges

- View conference bridge statuses

The GXE supports up to four conference bridges, each one supporting up to ten participants. In the **Conference Bridge** menu, the status, number of attendees, and the duration of an ongoing conference is displayed.

GXE5028 IPPBX Administration Interface

[Phone Extensions](#)
[Trunk/Phone Lines](#)
[Conference Bridge](#)
[Hunt/Ring Group](#)
[Auto-Attendant](#)
[System Configuration](#)
[Advanced Options](#)

Conference Bridge
[Logout](#)

| | Extension | Password | Status | Attendees | Duration(mins) |
|--------|-----------|----------|--------|-----------|----------------|
| Room 1 | 6994 | 123456 | Vacant | 0 | 0 |
| Room 2 | 6995 | 123456 | Vacant | 0 | 0 |
| Room 3 | 6996 | | Vacant | 0 | 0 |
| Room 4 | 6997 | | Vacant | 0 | 0 |

- Assign extension numbers and passwords to conference bridges

To assign an extension number to a conference bridge, enter the extension number into the *Extension* field of a conference bridge room. If you wish to require users to enter a password before entering, enter a numeric password into the *Password* field. When done, click on the **Submit** button to save your changes.

5. Configuring Ring Groups

Ring groups can be configured on the GXE to allow multiple users to provide a higher level of availability to incoming callers. Multiple ring methods are supported, and ring groups can have their own voicemail boxes as well.

- Viewing ring groups

The **Hunt/Ring Group** menu displays all configured ring groups and their details and allows you to create, modify, or delete ring groups.

The screenshot shows the 'GXE5028 IPPBX Administration Interface'. On the left is a navigation menu with options: Phone Extensions, Trunk/Phone Lines, Conference Bridge, **Hunt/Ring Group** (highlighted in yellow), and Auto-Attendant. The main content area is titled 'Hunt/Ring Group' and includes a 'Logout' link in the top right. Below the title is a table with columns: Extension, Group Name, Members, and Action. The table lists two groups: 'Sales' (Extension 6200, Members 6000;6001;6002;6003) and 'Support' (Extension 6210, Members 6004;6005;6006;6007;6008;6009). Each row has 'Modify' and 'Delete' buttons. An 'Add' button is located at the bottom left of the table area.

| Extension | Group Name | Members | Action |
|-----------|------------|-------------------------------|---|
| 6200 | Sales | 6000;6001;6002;6003 | Modify Delete |
| 6210 | Support | 6004;6005;6006;6007;6008;6009 | Modify Delete |

[Add](#)

- Adding and modifying ring groups

Add: Click on the **Add** button. The ring group details page will be displayed, allowing you to configure the ring group settings:

GXE5028 IPPBX Administration Interface

Phone Extensions

Trunk/Phone Lines

Conference Bridge

Hunt/Ring Group

Auto-Attendant

System Configuration

Advanced Options

Status

Reports

Add Hunt/Ring Group
[Logout](#)

| | |
|---|--|
| Extension | 0000 |
| Group Name | |
| Ring Mode | <input checked="" type="radio"/> Parallel <input type="radio"/> Serial |
| Serial Ring Attempts Per Member | 1 |
| Serial Ring Interval | 2 (in seconds) |
| Waiting Tone | System Music |
| <div style="border: 1px solid #ccc; padding: 5px;">Group Members</div> <div style="font-size: small; color: gray;">(e.g., 51000;5101;5200, up to maximum 15 members)</div> | |
| <div style="border: 1px solid #ccc; padding: 5px;">Round Robin of Serial Ring Starting Attempt</div> <div style="font-size: small; color: gray;">among the first 1 of the Group Member List</div> | |
| <div style="border: 1px solid #ccc; padding: 5px;">Email For Message Delivery</div> <div style="font-size: small; color: gray;">(if empty, will send to each member's email box)</div> | |

Submit
Cancel

- **Extension:** Enter an extension number to assign to this ring group. The leading digit for this extension must correspond with the *Leading Digit of Extensions* setting in the General Settings section of the **Phone Extensions** menu.
- **Group Name:** Enter a descriptive name to assign to this ring group.
- **Ring Mode:** Select a ring method for this ring group. Parallel will ring all ring group members at the same time, while Serial will ring members one at a time, starting from the first ring group member. A round-robin variation of Serial is available, and is enabled below. This allows the starting position of each ring group call to be the next ring group member of the previous ring group call, instead of from the first ring group member in the list.
- **Serial Ring Attempts Per Member:** If using Serial ring mode, select the number of attempts to ring all ring group members before sending the call to voicemail.
- **Serial Ring Interval:** If using Serial ring mode, select the number of rings for each ring group member before moving onto the next ring group member.
- **Waiting Tone:** Choose either ringback tone or hold-music to be played back to the ring group caller.
- **Group Members:** List all extensions who are members of this ring group, separating each with a semicolon.
- **Round Robin of Serial Ring Starting Attempt:** Enter the digit to specify the number of ring group members starting from the first in the list to round-robin the starting position of incoming ring group calls amongst. To disable the round-robin mode, enter 1. To round-robin amongst all ring group members, enter the total number of ring group members listed.
- **Email For Message Delivery:** Enter the email address to deliver voicemail-to-email messages to for voicemails left for this ring group.

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

Modify: Click on the **Modify** button to the right of the row displaying information for the ring group you wish to modify. The ring group details page will be displayed, allowing

you to modify all of the ring group's settings. When done, click on the **Submit** button to save your changes or **Cancel** to go back.

Delete: Click on the **Delete** button on the far right of the row displaying the information for the ring group you wish to delete. You will be prompted for confirmation via a dialog box; click **OK** to confirm or **Cancel** to go back.

6. Configuring Auto-Attendants

Incoming calls can be directed to auto-attendants to provide immediate and professional service to callers and automatically route calls to intended parties. You may schedule different auto-attendants to play based on the date, time, and day of the week as well. The **Auto-Attendant** menu displays all configured auto-attendants menus.

The screenshot displays the 'GXE5028 IPPBX Administration Interface'. On the left is a blue sidebar menu with options: Phone Extensions, Trunk/Phone Lines, Conference Bridge, Hunt/Ring Group, Auto-Attendant (highlighted in yellow), and Voice Menu (linked in blue). The main content area is titled 'Voice Menu' and includes a 'Logout' link in the top right. Below the title is a table with two columns: 'Voice Menu Name' and 'Action'. The table lists 'After Hours' and 'Business Hours', each with 'Modify' and 'Delete' buttons. An 'Add' button is located at the bottom left of the table area.

| Voice Menu Name | Action |
|-----------------|---------------|
| After Hours | Modify Delete |
| Business Hours | Modify Delete |

Add

- Configuring auto-attendant menus

The following actions can be performed in the Voice Menu section.

Add: Click on the **Add** button. The voice menu details page will be displayed, allowing you to configure an auto-attendant:

GXE5028 IPPBX Administration Interface

Phone Extensions

Trunk/Phone Lines

Conference Bridge

Hunt/Ring Group

Auto-Attendant

Voice Menu

Playing Rules

System Configuration

Advanced Options

Status

Reports

Add Voice Menu

[Logout](#)

| | | |
|---|---|------|
| Voice Menu Name | <input style="width: 100%;" type="text"/> | |
| <input type="checkbox"/> Press 0 to trigger | Hunt/Ring Group | None |
| <input type="checkbox"/> Press 1 to trigger | Hunt/Ring Group | None |
| <input type="checkbox"/> Press 2 to trigger | Hunt/Ring Group | None |
| <input type="checkbox"/> Press 3 to trigger | Hunt/Ring Group | None |
| <input type="checkbox"/> Press 4 to trigger | Hunt/Ring Group | None |
| <input type="checkbox"/> Press 5 to trigger | Hunt/Ring Group | None |
| <input type="checkbox"/> Press 6 to trigger | Hunt/Ring Group | None |
| <input type="checkbox"/> Press 7 to trigger | Hunt/Ring Group | None |
| <input type="checkbox"/> Press 8 to trigger | Hunt/Ring Group | None |
| <input type="checkbox"/> Press 9 to trigger | Hunt/Ring Group | None |

No entry time out (second), play warning and repeat voice menu for up to time(s)

- **Voice Menu Name:** Enter a name for this auto-attendant.
- **Press x to trigger:** Check the box next to the row for each keypad digit you would like to provide as an option for callers to choose. Callers automatically have the option to enter internal extension numbers from the auto-attendant, so no special configuration needs to be enabled for that. In the first drop-down box of each row, select the type of destination you would like this keypad digit to route the caller to. Select the exact destination of that type in the second drop-down box, if applicable.
- **No entry time out:** Enter the time to wait for the caller 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 to repeat when the caller does not enter any menu options, before exiting.

When done, click on the **Next** button or **Cancel** to go back.

The IVR upload page will allow uploading of the voice recording for the new auto-attendant. Click on the **Browse...** button to search for files on your computer. Click on the **Upload** button to upload this file to the GXE. After the file is successfully uploaded, click on the **Finish** button to add this auto-attendant.

Modify: Click on the **Modify** button to the right of the row displaying information for the auto-attendant you wish to modify. The voice menu details page will be displayed, allowing you to re-configure the auto-attendant. When done, click on the **Next** button; you may then upload a new voice recording for the auto-attendant if you wish. When done, click on the **Finish** button to modify this auto-attendant.

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

- Configuring time-based playing rules for auto-attendants

Playing rules may be created to dictate which auto-attendant is played to the incoming caller, based on the times you set. The Playing Rules section displays all configured voice menu playing rules, and allows the following actions to be performed.

The screenshot shows the 'GXE5028 IPPBX Administration Interface' with a sidebar menu on the left containing: Phone Extensions, Trunk/Phone Lines, Conference Bridge, Hunt/Ring Group, Auto-Attendant (highlighted), Voice Menu, and Playing Rules. The main content area is titled 'Playing Rules' and includes a 'Logout' link. It displays a table with the following data:

| Voice Menu | Playing Dates | Playing Day Of Week | Playing Time | Action |
|----------------|---------------|---------------------|-----------------------|---|
| Business Hours | 1/1-12/31 | --- | 8:00-16:59 | Modify Delete |
| After Hours | 1/1-12/31 | --- | 0:00-7:59/17:00-23:59 | Modify Delete |

Below the table is an 'Add' button.

Add: Click on the **Add** button. The playing rules details page will be displayed, allowing you to configure the time conditions for playing an auto-attendant:

The screenshot shows the 'GXE5028 IPPBX Administration Interface' with a sidebar menu on the left containing: Phone Extensions, Trunk/Phone Lines, Conference Bridge, Hunt/Ring Group, Auto-Attendant (highlighted), Voice Menu, Playing Rules, System Configuration, and Advanced Options. The main content area is titled 'Add Playing Rules' and includes a 'Logout' link. It contains the following form fields:

- Play Voice Menu:** A drop-down box currently set to 'None'.
- According to:**
 - ☒ **Date:** A text field for entering a date or range (e.g., 5/1-5/7).
 - ☐ **Day of Week:** Checkboxes for SUN, MON, TUN, WEN, THU, FRI, and SAT.
 - ☐ **Except on date(s):** A text field for entering exception dates (e.g., 5/1-5/7) where calls are answered by voice menu.
- Play Time:** A text field for entering a time range (e.g., 0:00-8:00/18:30-24:00).

At the bottom are 'Submit' and 'Cancel' buttons.

- **Play Voice Menu:** Using the drop-down box, select the auto-attendant you wish to configure.
- **Date:** You may set the auto-attendant selected above to play either on a preset date or range of dates, or to play on all selected days of the week. To specify by date, select the radio button beside *Date* and enter the date or range of dates in the text field to the right.
- **Day of Week:** To specify the auto-attendant to play on all selected days of the week, select the radio button beside *Day of Week* and check each checkbox beside each day of the week you would like the auto-attendant to play on.
- **Except on date(s):** If setting the auto-attendant to play based on selected days of the week, you may enter exceptions such as holiday dates where you would like another auto-attendant to be played. Enter exception dates in the blank text field,

and use the drop-down box on the far right to select the auto-attendant to use instead on the listed exception dates.

- *Play Time*: Enter in the blank text field the time of day to play the auto-attendant during days the auto-attendant is set to play.

Modify: Click on the **Modify** button to the right of the row displaying information for the playing rules you wish to modify. The playing rules details page will be displayed, allowing you to modify all of the playing rules. When done, click on the **Submit** button to save your changes or **Cancel** to go back.

Delete: Click on the **Delete** button on the far right of the row displaying the information for the playing rule you wish to delete. You will be prompted for confirmation via a dialog box; click **OK** to confirm or **Cancel** to go back.

7. Call Queues

This screen allows you to upload an audio file in .wav (file/sample rate 8K/sample data 16bit/single channel) that will play while you are in a call queue.

- Setting up In-Queue Announcements

In-Queue Announcement Name

This field lets you enter a name (20 character limit) for the queue that you are uploading.

In-Queue Announcement File

This field lets you specify the location of the .wav file that you are uploading. You can type the path manually or click the browse button to navigate the file through Windows Explorer.

- Setting up Agents

Add an Agent

This page lets you add an agent and configure its rules within the queue. You must specify the name, extension, SIP password, skill level, and email of the agent. Your typical queue will start with the agent of the highest skill level.

| | |
|------------------------|--|
| Phone Extensions | <h3>Add Agent</h3> <hr/> <div> <div>Name</div> <input type="text"/> </div> <div> <div>Extension</div> <input type="text"/> </div> <div> <div>Skill</div> <div>1 ▾</div> </div> <div> <div>SIP Password</div> <input type="password"/> </div> <div> <div>Email</div> <input type="text"/> </div> <div> <input type="button" value="Submit"/> <input type="button" value="Cancel"/> </div> |
| Trunk/Phone Lines | |
| Conference Bridge | |
| Hunt/Ring Group | |
| Auto-Attendant | |
| Call Queues | |
| In-Queue Announcements | |
| Agents | |
| Call Queues | |

- Call Queues

Add a Call Queue

This is the main configuration page for any call queues that you add to your system.

Name

Enter the name of your call queue here.

Extension

Enter the extension of the call queue here. See the drop down box for a list of current extensions.

Priority

Set the priority of the call queue.

Queue status update frequency

This determines how often the in-queue announcement message will play.

Other Announcements

This determines the frequency in which any other announcements you add will be played to the caller 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.

Minimum Caller Wait Time

This field lets you set the minimum amount of time that callers will wait within the queue before being forwarded to voicemail.

Maximum Queued Callers

This lets you set how many callers can be within the queue simultaneously.

Group Email Address for Voicemail Delivery

Enter the email address in which all voicemail for the queue/group will be delivered.

Agent Call Wrap-Up Time

This setting lets you specify the amount of wrap-up time an agent will have before receiving another call. For example an agent may need 5 minutes of wrap-up time to document a call.

Listed Agents For This Queue

All configured agents for the queue will be displayed here.

Automatic Call Distribution

This setting lets you configure enable and disable skill-based routing. If skill-based routing is enabled, you can configure it to rout by least-skilled first or more skilled first.

Ring Mode to Agents of Same Skill Level

These settings are similar to hunt/ring groups in that you can configure the order/rules in which agents within the group/queue ring.

- Serial - Agents ring one at a time based on availability.
- Parallel - All agents ring simultaneously.
- Circular -A different agent will ring first each time a caller enters the queue.
- Least Busy -The least busy agent will ring first.

Call Queue Greeting Message

Upload a .wav file (file/sample rate 8K/sample data 16bit/single channel) for the queue greeting message. You can type the path manually or click the browse button to navigate the file through Windows Explorer.

Phone Extensions

Trunk/Phone Lines

Conference Bridge

Hunt/Ring Group

Auto-Attendant

Call Queues

In-Queue Announcements

Agents

Call Queues

System Configuration

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| | |
|--|---|
| Name | <input type="text"/> |
| Extension | <input type="text"/> (for reference, current list of extensions <input type="text" value="7000"/>) |
| Priority | <input type="text" value="1"/> ▾ |
| Queue Status Update Frequency | <input type="text" value="45"/> seconds |
| Other Announcements | Play the following in-queue announcements at an interval of <input type="text" value="45"/> seconds |
| Maximum Caller Wait Time | <input type="text" value="0"/> seconds(default is 0, unlimited) |
| Minimum Caller Wait Time | <input type="text" value="0"/> seconds(default is 0, unlimited) |
| Maximum Queued Callers | <input type="text" value="3"/> ▾ |
| Group Email address for voicemail delivery | <input type="text"/> |
| Agent Call Wrap-up Time | <input type="text" value="15"/> seconds |
| Listed Agents For This Queue | |

8. System Configuration

System configuration and administration may be performed from the **System Configuration** menu. The following sections detail the tasks which may be performed in this menu.

- Configuring network settings

The Networking Setting section allows you to configure the LAN-side IP addressing and DHCP server settings. For the WAN-side settings, IP addressing of the WAN port can be set as well as Dynamic DNS and port-forwarding settings. The below actions and their associated settings appear in order from top to bottom on the GXE web UI as well.

Setting the LAN-port IP address:

Set the LAN-port IP address by entering the IP address you wish to use in the *LAN Base IP* setting. Enter the subnet mask in the *LAN Subnet Mask* setting. Please note that this should be configured prior to auto-provisioning your phones through the **Express Setup** menu, as they will be using an incorrect IP address for the SIP server after the LAN-port IP address changes.

Configuring DHCP:

To use the GXE as the DHCP server on the LAN-side, set the *DHCP Enable* setting to *Enable* (this is enabled by default). You may specify the starting and ending IP

addresses of the range of 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 you to specify the number of hours an IP address is leased to a device before renewal.

| | | |
|---|-----------------------|---|
| Auto-Attendant System Configuration Networking Setting System Setting Feature Codes | DHCP Enable | <input checked="" type="radio"/> Enable <input type="radio"/> Disable |
| | Start of DHCP IP Pool | 100 |
| | End of DHCP IP Pool | 199 |
| | DHCP IP Lease Time | 120 (in units of hours, default is 120 hours c |

Setting the WAN-port IP address:

The WAN-port can be configured to be dynamically assigned via DHCP, to use PPPoE, or to be statically configured. Select the radio button beside the option you wish to use. To use PPPoE, you will need to enter the PPPoE account and password information and DNS server information in the provided fields. If statically configuring the WAN-port IP address, please enter the IP addressing details in the provided fields.

| | | | | | |
|--|--|-----|------|------|------|
| Firmware Upgrade Backup & Restore Configure Reset & Reboot Syslog Configuration Advanced Options Status Reports | WAN Setting | | | | |
| | <input checked="" type="radio"/> Dynamically Assigned Via DHCP | | | | |
| | <input type="radio"/> Use PPPoE | | | | |
| | PPPoE Account ID | | | | |
| | PPPoE Password | | | | |
| | Preferred DNS server | 0 | .0 | .0 | .0 |
| | <input type="radio"/> Statically Configured | | | | |
| | IP Address | 192 | .168 | .0 | .160 |
| | Subnet Mask | 255 | .255 | .255 | .0 |
| | Default Router | 192 | .168 | .0 | .1 |
| Primary DNS | | . | . | . | |
| Secondary DNS | | . | . | . | |

Enabling/disabling WAN-side HTTP access:

Security and accessibility for management are considerations for whether or not you wish to enable or disable HTTP/Telnet access from the WAN-side of the GXE. If you decide to use the default setting of enabled, set *WAN Side Http/Telnet Access* to Yes; otherwise, disable it by setting it to No.

| | | |
|--|-----------------------------|---|
| | WAN Side Http/Telnet Access | <input type="radio"/> No <input checked="" type="radio"/> Yes |
|--|-----------------------------|---|

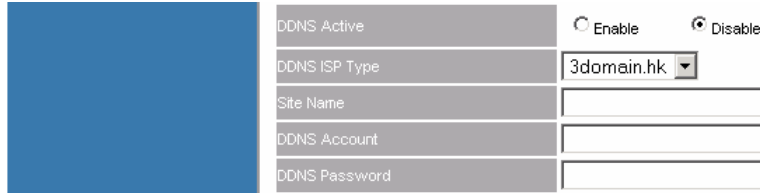
Placing a LAN-side device in the DMZ:

To specify a device on the LAN-side of the GXE to act as a default DMZ server, enter the IP address of the device in the *DMZ IP* field. This allows all packets destined for services undefined on the GXE to be routed to the default DMZ server.

| | | |
|--|--------|---|
| | DMZ IP | <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> |
|--|--------|---|

Using Dynamic DNS:

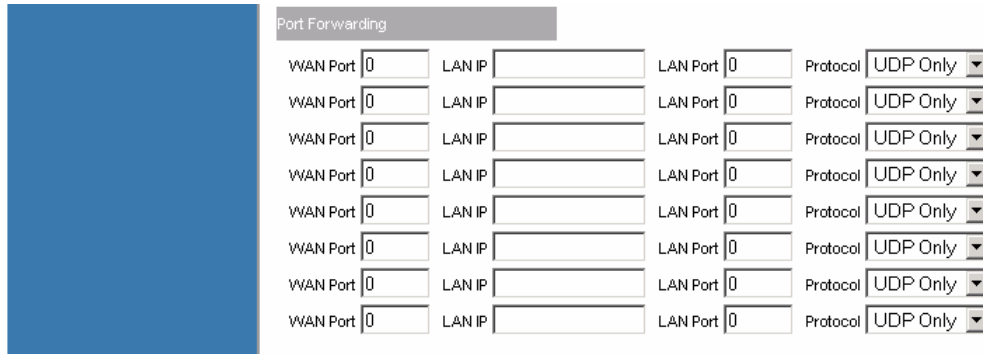
If the WAN-port IP of the GXE is dynamically assigned, you may use Dynamic DNS to assign it a fixed domain name which is always synchronized with the IP address of your GXE. Enable Dynamic DNS by setting the *DDNS Active* setting to *Enable* and setting the DDNS ISP type, site name, and account and password information in the provided fields. If not using Dynamic DNS, set the *DDNS Active* setting to *Disable*.



| | |
|---------------|---|
| DDNS Active | <input type="radio"/> Enable <input checked="" type="radio"/> Disable |
| DDNS ISP Type | 3domain.hk |
| Site Name | |
| DDNS Account | |
| DDNS Password | |

Configuring port forwarding:

The GXE 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.



| Port Forwarding | | | | | | | |
|-----------------|---|--------|--|----------|---|----------|----------|
| WAN Port | 0 | LAN IP | | LAN Port | 0 | Protocol | UDP Only |
| WAN Port | 0 | LAN IP | | LAN Port | 0 | Protocol | UDP Only |
| WAN Port | 0 | LAN IP | | LAN Port | 0 | Protocol | UDP Only |
| WAN Port | 0 | LAN IP | | LAN Port | 0 | Protocol | UDP Only |
| WAN Port | 0 | LAN IP | | LAN Port | 0 | Protocol | UDP Only |
| WAN Port | 0 | LAN IP | | LAN Port | 0 | Protocol | UDP Only |
| WAN Port | 0 | LAN IP | | LAN Port | 0 | Protocol | UDP Only |
| WAN Port | 0 | LAN IP | | LAN Port | 0 | Protocol | UDP Only |

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

- Configuring system settings

The System Setting section contains various important internal system settings and allows configuration of several features and functions of the GXE. The below actions and their associated settings appear in order from top to bottom on the GXE web UI as well.

Setting the web UI login password:

To change the password for accessing the GXE web configuration pages, enter a new password in the *Login Password* field. Do not lose this password. For further security, you may disable WAN-side web UI access in the Networking Setting section.



| | |
|-------------------|-----------------------|
| Trunk/Phone Lines | Administrator Setting |
| Conference Bridge | Login Password |

Storing administrator contact and information:

Optional fields have been provided to store the administrator's name, phone numbers, and email address so that the administrator may be contacted for any issues requiring attention. The system name may also be labeled in the *System Name* setting to identify this GXE.

| | | |
|------------------------|----------------|----------------------|
| Hunt/Ring Group | Name | <input type="text"/> |
| | Contact Phone | <input type="text"/> |
| | Contact Mobile | <input type="text"/> |
| | Contact Email | <input type="text"/> |
| | System Name | <input type="text"/> |

Setting SIP IP and port:

The SIP port for the GXE can be set to either the default of 5060 in the *SIP Port* field, or another port number may be used. The *Static Mapped WAN IP for SIP* and the *Static Mapped WAN Port for SIP* may also be changed in their respective fields.

| | | |
|-----------------------------|--------------------------------|-----------------------------------|
| Upload & Restore | SIP Port | <input type="text" value="5060"/> |
| | Static Mapped WAN IP for SIP | <input type="text"/> |
| | Static Mapped WAN Port for SIP | <input type="text"/> |

Using STUN for NAT traversal:

If the GXE is behind a NAT router, it may be necessary to use STUN to allow the GXE to reliably communicate via IP through the router. Enter a STUN server IP address or domain name in the *STUN Server* field.

| | | |
|---------------------------|-------------|--|
| Reset & Reboot | STUN Server | <input type="text" value="stun.grandstream.com"/> (e.g., my_stunserver_ip_or_url:port) |
|---------------------------|-------------|--|

Enabling voicemail-to-email:

To have the GXE email users with their received voicemail files, enter the SMTP server and login name and password information, as well as the email address to send from. This will allow the GXE to send voicemail emails to users and ring groups with email addresses configured when they receive voicemail messages.

| | | |
|-------------------------|----------------|----------------------|
| Advanced Options | SMTP Server | <input type="text"/> |
| | Login Name | <input type="text"/> |
| | Login Password | <input type="text"/> |
| | Email Address | <input type="text"/> |

Configuring a Call Detail Record server:

To specify a Call Detail Record (CDR) server for the GXE to upload call detail information to, enter the IP address or domain name of the server in the *CDR TFTP Server* field.

| | | |
|--|-----------------|--|
| | CDR TFTP Server | <input type="text" value="192.168.0.6"/> |
|--|-----------------|--|

Setting the system time:

Select the correct time zone for the location of the GXE in the *Time Zone* drop-down box. Conversely, you may enter a *Self-Defined Time Zone*, using the following syntax: MTZ+6MDT+5,M4.1.0,M11.1.0

The syntax starts with “MTZ” followed by your time offset from Greenwich Mean Time (GMT). To set Pacific Standard Time, you would use “MTZ-8”. This is followed by “MDT” followed by your time offset from GMT during daylight savings time. A comma then follows to begin definition of the daylight savings start date.

For the GXE to find the accurate time, an NTP server IP address or domain name will need to be entered in the *NTP Server* field.

| | |
|------------------------|--|
| Time Zone | GMT-12:00(International Date Line West) |
| Self-Defined Time Zone | MTZ 6MDT 5,M4.1.0,M11.1.0 (For example: "MTZ+6MDT+5,M4.1.0,M11.1.0") |
| NTP Server | us.pool.ntp.org |

Selecting Music on Hold source:

Specify a Music on Hold (MOH) source by setting the *System Music* drop-down box to either *System Music Files* for internally stored audio files or *Audio-In* to play audio from an external audio source connected to the AUDIO IN jack on the back of the GXE.

| | |
|--------------|----------|
| System Music | Audio-In |
|--------------|----------|

Setting mailbox storage quota:

Voicemail, videomail, and faxmail storage can be limited for the different types of system users. The mailbox storage limits (in percentages of total system memory) are specified in the drop-down boxes beside each user privilege level.

| Storage quota of voicemail/videomail/faxmail per privilege level | |
|--|----|
| Super | 2% |
| Privileged | 2% |
| Regular | 2% |
| Basic | 2% |
| Restricted | 2% |

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

- Setting feature codes

Feature star codes allow GXE users to set features such as forwarding and Do Not Disturb on their individual extensions, as well as reach destinations such as the voicemail system. The Feature Codes section allows customization of the feature codes. The feature codes may be viewed or modified in the text box beside each feature name.

GXE5028 IPPBX Administration Interface

| | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
|---|--|----------------------|------|-------------------------|-------|-----------------------------------|-------|-----------------------------------|-------|--------------------------|-------|--------------------------|-------|-------------------------------|-------|-------------------------------|-------|---------------|-------|-------------------------|-------|-------------------------|-------|-----------------|-------|-----------------|-------|----------------|-------|------|-------|--------|-------|------------|--------|
| <div style="text-align: center; font-weight: bold;">Phone Extensions</div> <div style="text-align: center; font-weight: bold;">Trunk/Phone Lines</div> <div style="text-align: center; font-weight: bold;">Conference Bridge</div> <div style="text-align: center; font-weight: bold;">Hunt/Ring Group</div> <div style="text-align: center; font-weight: bold;">Auto-Attendant</div> <div style="text-align: center; font-weight: bold; color: yellow;">System Configuration</div> <div style="text-align: center;">Networking Setting</div> <div style="text-align: center;">System Setting</div> <div style="text-align: center; color: yellow;">Feature Codes</div> <div style="text-align: center;">Upload & Restore</div> <div style="text-align: center;">Firmware Upgrade</div> <div style="text-align: center;">Backup & Restore Configure</div> <div style="text-align: center;">Reset & Reboot</div> <div style="text-align: center;">Syslog Configuration</div> <div style="text-align: center; font-weight: bold;">Advanced Options</div> <div style="text-align: center; font-weight: bold;">Status</div> <div style="text-align: center; font-weight: bold;">Reports</div> | <div style="text-align: right; font-weight: bold; font-size: small;">Logout</div> <div style="text-align: center; font-weight: bold; margin-bottom: 10px;">Feature Codes</div> <table border="1" style="width: 100%; border-collapse: collapse;"> <tr><td style="padding: 2px;">Directory Assistance</td><td style="padding: 2px;">*128</td></tr> <tr><td style="padding: 2px;">Prompt Voice Assistance</td><td style="padding: 2px;">*7701</td></tr> <tr><td style="padding: 2px;">Enable Call Forward Unconditional</td><td style="padding: 2px;">*7702</td></tr> <tr><td style="padding: 2px;">Cancel Call Forward Unconditional</td><td style="padding: 2px;">*7703</td></tr> <tr><td style="padding: 2px;">Enable Call Forward Busy</td><td style="padding: 2px;">*7704</td></tr> <tr><td style="padding: 2px;">Cancel Call Forward Busy</td><td style="padding: 2px;">*7705</td></tr> <tr><td style="padding: 2px;">Enable Call Forward No-Answer</td><td style="padding: 2px;">*7706</td></tr> <tr><td style="padding: 2px;">Cancel Call Forward No-Answer</td><td style="padding: 2px;">*7707</td></tr> <tr><td style="padding: 2px;">Query Forward</td><td style="padding: 2px;">*7708</td></tr> <tr><td style="padding: 2px;">Enable Do-Not-Disturbed</td><td style="padding: 2px;">*7712</td></tr> <tr><td style="padding: 2px;">Cancel Do-Not-Disturbed</td><td style="padding: 2px;">*7713</td></tr> <tr><td style="padding: 2px;">Enable Intercom</td><td style="padding: 2px;">*7709</td></tr> <tr><td style="padding: 2px;">Cancel Intercom</td><td style="padding: 2px;">*7710</td></tr> <tr><td style="padding: 2px;">Query Intercom</td><td style="padding: 2px;">*7711</td></tr> <tr><td style="padding: 2px;">Park</td><td style="padding: 2px;">*7730</td></tr> <tr><td style="padding: 2px;">Pickup</td><td style="padding: 2px;">*7731</td></tr> <tr><td style="padding: 2px;">Voice Mail</td><td style="padding: 2px;">*12345</td></tr> </table> <div style="text-align: center; margin-top: 10px;"> <input type="button" value="Submit"/> </div> | Directory Assistance | *128 | Prompt Voice Assistance | *7701 | Enable Call Forward Unconditional | *7702 | Cancel Call Forward Unconditional | *7703 | Enable Call Forward Busy | *7704 | Cancel Call Forward Busy | *7705 | Enable Call Forward No-Answer | *7706 | Cancel Call Forward No-Answer | *7707 | Query Forward | *7708 | Enable Do-Not-Disturbed | *7712 | Cancel Do-Not-Disturbed | *7713 | Enable Intercom | *7709 | Cancel Intercom | *7710 | Query Intercom | *7711 | Park | *7730 | Pickup | *7731 | Voice Mail | *12345 |
| Directory Assistance | *128 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Prompt Voice Assistance | *7701 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Enable Call Forward Unconditional | *7702 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Cancel Call Forward Unconditional | *7703 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Enable Call Forward Busy | *7704 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Cancel Call Forward Busy | *7705 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Enable Call Forward No-Answer | *7706 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Cancel Call Forward No-Answer | *7707 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Query Forward | *7708 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Enable Do-Not-Disturbed | *7712 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Cancel Do-Not-Disturbed | *7713 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Enable Intercom | *7709 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Cancel Intercom | *7710 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Query Intercom | *7711 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Park | *7730 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Pickup | *7731 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Voice Mail | *12345 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |

Directory Assistance: Dial from a phone to have the GXE playback the phone's extension number.

Prompt Voice Assistance: 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.

Enable Call Forward Unconditional: Dial the feature code plus [forward-to number] to enable unconditional call forwarding for the extension you are dialing with.

Cancel Call Forward Unconditional: Cancel unconditional call forward, calls will ring in.

Enable Call Forward Busy: Dial the feature code plus [forward-to number] to turn on busy call forwarding for the extension you are dialing with.

Cancel Call Forward Busy: Cancel busy call forward; calls will ring when 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 with.

Cancel Call Forward No-Answer: Cancel call forward on no answer, calls will go to voicemail.

Query Forward: Dial the feature code to have the GXE playback the call forwarding status and forward-to number.

Enable Do-Not-Disturbed: Calls will go straight to voicemail, or return busy if no voicemail.

Cancel Do-Not-Disturbed: Cancel DND, calls will ring in normally.

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

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

Voice Mail: Dial the appropriate feature code to access the voicemail system.

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

- Uploading voice prompts and configuration templates

Custom voice prompts may be uploaded in the Firmware Upgrade section, as well as configuration templates for use when auto-provisioning phones. To upload these files, browse to your firmware/prompt files in the Firmware Upgrade and System Prompt Image fields 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.

Note: If you would like the naming scheme for custom voice prompt files, contact us and we will happily send it to you.

The screenshot shows the 'GXE5024 IPPBX Administration Interface' header. Below it, the 'Firmware Upgrade' section is active, with a 'Language' dropdown set to 'English' and a 'Logout' link. The 'Firmware Upgrade' section contains a text input field, a 'Browse...' button, and a 'Submit' button. Below this, the 'System Prompt Image' section also contains a text input field, a 'Browse...' button, and a 'Submit' button.

- Upgrading firmware

The Firmware Upgrade section allows you to upload firmware from your computer to the GXE and upload custom voice prompts.

Prior to the firmware upgrade, please read the release notes to review the changes. In addition, backup a copy of your GXE configuration, detailed in the next section. The GXE will not overwrite your existing configuration, but it is good practice to keep a backup copy in all cases.

Once you have downloaded new firmware from Grandstream, click on the **Browse...** button to find the file on your computer. After the file has been selected, click on the **Submit** button and wait for the file to finish uploading. When finished, reboot the GXE to load the new firmware.

- Backing up and restoring configuration files

Configuration files may be downloaded to or uploaded from your computer in the Backup & Restore Configure section. It is good practice to keep a backup copy of your configuration file at all times.

The screenshot shows the 'GXE5028 IPPBX Administration Interface' with a sidebar menu on the left containing 'Phone Extensions', 'Trunk/Phone Lines', 'Conference Bridge', 'Hunt/Ring Group', 'Auto-Attendant', and 'System Configuration' (highlighted in yellow). The main content area is titled 'Backup & Restore Configure' and includes a 'Logout' link. It features two radio buttons: 'Upload Configure' (selected) and 'Download Configure'. Below the 'Upload Configure' button is a text input field with a 'Browse...' button next to it. A 'Submit' button is located at the bottom of the section.

Configuration backup: Select the *Download Configure* radio button, and click on the **Submit** button. You will be prompted to download the configuration file onto your computer.

Configuration restore: Select the *Upload Configure* radio button, and click on the **Browse...** button to find the configuration file on your computer. After the file has been selected, click on the **Submit** button and wait for the file to finish uploading. The GXE will automatically reboot when finished, and load the configuration file.

- Rebooting and resetting to default

The Reset & Reboot section allows rebooting of the GXE or resetting of all settings to factory default. Please backup the configuration file before resetting to factory default, should you decide you need to restore the system configuration.

The screenshot shows the 'GXE5028 IPPBX Administration Interface' with the same sidebar menu. The main content area is titled 'Reset & Reboot' and includes a 'Logout' link. It features two radio buttons: 'Reboot' (selected) and 'Reset to Default'. A 'Submit' button is located at the bottom of the section.

Rebooting: Select the *Reboot* radio button, and click on the **Submit** button. The GXE will reboot itself.

Resetting to default: Select the *Reset to Default* radio button, and click on the **Submit** button. The GXE will reboot itself, and will boot up with all factory default settings. The web configuration page access password will revert back to the default password of "admin".

- Configuring Syslog logging

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

GXE5028 IPPBX Administration Interface

[Phone Extensions](#)
[Trunk/Phone Lines](#)
[Conference Bridge](#)
[Hunt/Ring Group](#)
[Auto-Attendant](#)
[System Configuration](#)
[Networking Setting](#)

Syslog Configuration
[Logout](#)

Syslog Server IP

Syslog Level Error

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

9. Configuring Peer PBX Systems

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

- Viewing peer systems

The Peer Systems section displays all configured peer systems and their details.

GXE5028 IPPBX Administration Interface

[Phone Extensions](#)
[Trunk/Phone Lines](#)
[Conference Bridge](#)
[Hunt/Ring Group](#)
[Auto-Attendant](#)
[System Configuration](#)

Peer Systems
[Logout](#)

| Prefix | Peer URL | Trunk Number | Action | |
|--------|-----------------------|--------------|---------------------------------------|---------------------------------------|
| 8 | peer1.grandstream.com | 11 | <input type="button" value="Modify"/> | <input type="button" value="Delete"/> |
| 7 | peer2.grandstream.com | 12 | <input type="button" value="Modify"/> | <input type="button" value="Delete"/> |

- Adding and modifying peer systems

The following actions can be performed in this section.

Add: Click on the **Add** button. The peer system details page will be displayed, allowing you to enter connectivity details for this peer system:

The screenshot shows the 'GXE5028 IPPBX Administration Interface'. On the left is a blue sidebar menu with the following items: Phone Extensions, Trunk/Phone Lines, Conference Bridge, Hunt/Ring Group, Auto-Attendant, System Configuration, and Advanced Options (highlighted in yellow). Under 'Advanced Options' is a link for 'Peer Systems'. The main content area is titled 'Add Peer System' and has a 'Logout' link in the top right. The form contains the following fields: 'Peer URL' (text input), 'Trunk Number' (text input with '0' entered), 'Prefix' (text input with a note '(up to 10 digits)'), 'IPPBX Number' (text input), and 'Heart Beat' (radio buttons for 'Yes' and 'No', with 'Yes' selected). At the bottom are 'Submit' and 'Cancel' buttons.

- *PeerURL:* Enter the IP address or domain name of the peer system.
- *Trunk Number:* The number of concurrent calls that can be used on the peer.
- *Prefix:* Configure the prefix digit on outbound calls to recognize this peer system as the destination. When the call is sent to this peer system, the prefix digit is not removed as it is when dialing out other trunks. You may specify multiple prefixes for a peer system by separating them with a semicolon (i.e. 6;7).
- *IPPBX Number:* Enter a unique number to identify the remote peer system.
- *Heart Beat:* Enable or disable availability detection of the remote peer system (requires compatibility on the remote end).

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

Modify: Click on the **Modify** button to the right of the row displaying information for the peer system you wish to modify. The peer system details page will be displayed, allowing you to modify all of the peer system's settings. When done, click on the **Submit** button to save your changes or **Cancel** to go back.

Delete: Click on the **Delete** button on the far right of the row displaying the information for the peer system you wish to delete. You will be prompted for confirmation via a dialog box; click **OK** to confirm or **Cancel** to go back.

10. Viewing GXE Status and Reporting Information

- GXE current status information

In the **Status** menu, the following categories of GXE status information are displayed.

GXE5028 IPPBX Administration Interface

Phone Extensions

Trunk/Phone Lines

Conference Bridge

Hunt/Ring Group

Auto-Attendant

System Configuration

Advanced Options

Status

Reports

[Logout](#)

System Status

| | |
|----------------------|-------------------|
| Product Model | GXE5028 |
| Bootloader Version | 1.0.0.4 |
| Base Version | 1.0.0.14 |
| WAN MAC Address | 00:0B:82:0F:2C:EA |
| System Up Time Since | 2007-11-08 08:02 |

Network Status

| | |
|-----------------------|----------------|
| WAN Port Link Status | |
| LAN Port Link Status | |
| WAN-side NAT Detected | dissymmetrical |
| PPPoE Link Status | Disabled |
| Active DHCP Clients | 0 |

Peripheral Status

| | |
|------------------|-----------|
| Phone/Fax Port 1 | idle |
| PSTN Line 1 | unplugged |
| PSTN Line 3 | unplugged |
| PSTN Line 5 | unplugged |
| PSTN Line 7 | unplugged |
| USB Port | |

User Activity Status

| | |
|----------------------|---|
| Current Active Calls | 0 |
| Local Online Users | 0 |

| | |
|------------------|-------------------|
| Hardware Version | 1.0.0.2 |
| Core Version | 1.0.0.14 |
| Firmware Version | 1.0.0.29 |
| LAN MAC Address | 00:0B:82:0F:2C:EB |

| | |
|---------------------|---------------|
| WAN IP Address | 172.18.31.251 |
| LAN IP Address | 192.168.2.1 |
| Mapped IP:port | |
| DDNS Status | Disabled |
| Active Peer Systems | |

| | |
|--------------------|-----------|
| Phone/Fax Port 2 | idle |
| PSTN Line 2 | unplugged |
| PSTN Line 4 | unplugged |
| PSTN Line 6 | unplugged |
| PSTN Line 8 | unplugged |
| Music-On-Hold Port | |

| | |
|------------------------|---|
| Total Configured Users | 1 |
| Remote Online Users | 0 |

System Status: System identification and firmware information and system uptime.

Network Status: Network connection information.

Peripheral Status: Physical ports connection status information.

User Activity Status: Current user and registration status and active calls information.

- System and call reports

In the **Reports** menu, useful system statistics and call statistics collected by the GXE are reported.

Viewing system statistics

Under the System Statistics section, information regarding memory usage, faxes, and voice and video messages is displayed.

GXE5028 IPPBX Administration Interface

Phone Extensions
Trunk/Phone Lines
Conference Bridge
Hunt/Ring Group
Auto-Attendant
System Configuration
Advanced Options
Status
Reports
System Statistics
Call Statistics

System Statistics

Logout

| | |
|------------------------------------|-----------|
| System NAND Flash Utilization | 0% used |
| Saved Inbound Voice Messages | 0 |
| Unread Voice Messages | 0 |
| Average Duration Per Voice Message | 0 seconds |
| Saved Inbound Fax Messages | 0 |
| Average Pages Per Fax Received | 0 pages |
| Total Outbound Fax Messages Sent | 0 |
| Average Pages Per Fax Sent | 0 pages |
| Saved Inbound Video Messages | 0 |
| Unread Video Messages | 0 |
| Average Duration Per Video Message | 0 seconds |

Viewing call statistics

Under the Call Statistics section, information regarding inbound calls and outbound calls is displayed. These statistics are reset upon reboot of the GXE.

Phone Extensions
Trunk/Phone Lines
Conference Bridge
Hunt/Ring Group
Auto-Attendant
System Configuration
Advanced Options
Status
Reports
System Statistics
Call Statistics

Call Statistics

Logout

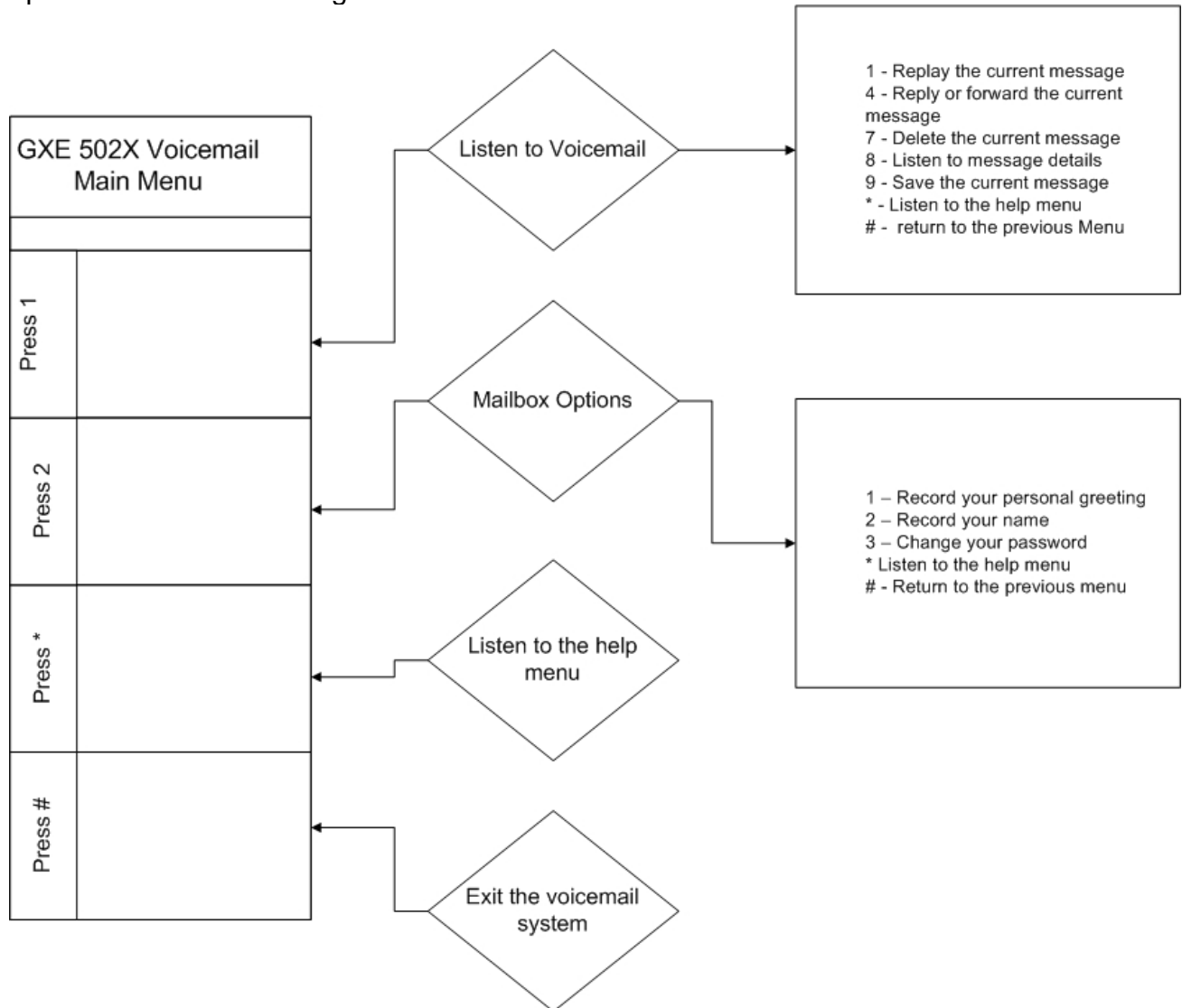
| | |
|---|-----------|
| Total Outbound Calls: | |
| Total Outbound Calls to Remote Extensions | 0 |
| Total Outbound Calls to PSTN Trunks | 0 |
| Total Outbound Calls to SIP Trunks | 0 |
| Average Duration per Outbound Call | 0 seconds |
| Total Inbound Calls: | |
| Total Inbound Calls to Remote Extensions | 0 |
| Total Inbound Calls to PSTN Trunks | 0 |
| Total Inbound Calls to SIP Trunks | 0 |
| Average Duration per Inbound Call | 0 seconds |

11. Configuring Voicemail

The GXE 502X allows users to manage voicemail via IVRs in their phones or through a personal web portal. This section summarizes how to manage voicemail and other settings using both of these features.

Configuring Voicemail through the IVR

The default feature code for voicemail access is *12345. After dialing this code, you will enter a basic IVR menu with the option to listen to/forward messages and configure your voicemail options. See the following flow chart for reference.



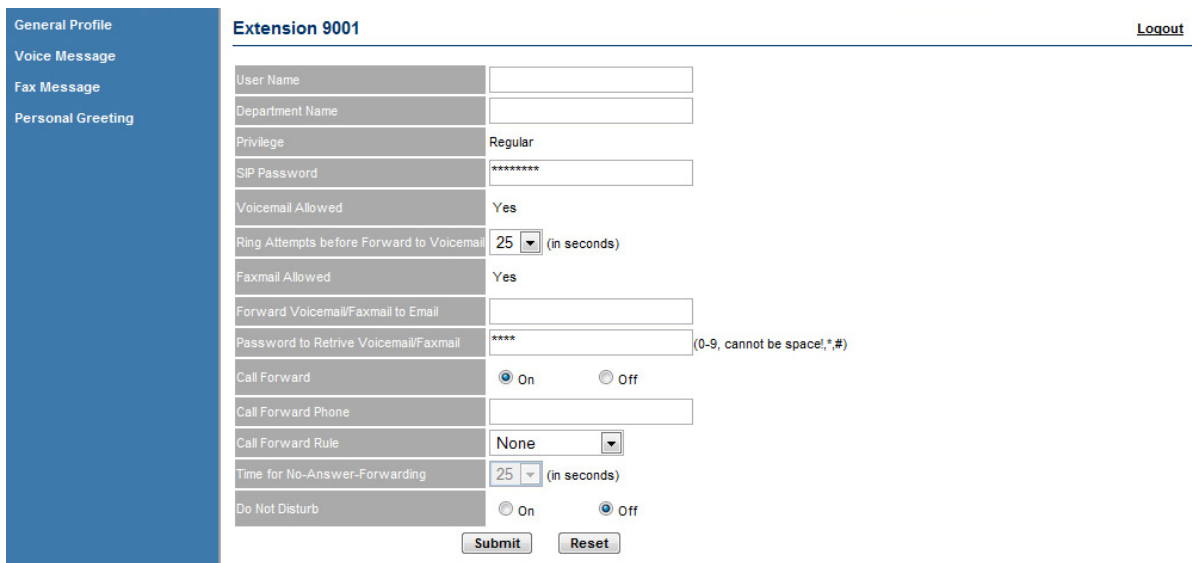
The Personal Web Portal

1. To log into the personal web portal for an extension. Log into the GXEs web GUI using the extension # as the user name and password. If you have changed your voicemail password from the default value, use your new password.



The login screen for the Grandstream Device Configuration web portal. It features a blue header with the title "Grandstream Device Configuration". Below the header is a yellow login area with two input fields: "Login ID" (containing "9001") and "Password" (containing six dots). To the right of the password field is a language dropdown menu set to "English". A "Login" button is centered below the input fields. At the bottom of the page, a blue footer contains the text "All Rights Reserved Grandstream Networks, Inc. 2007".

Once you have logged in the following page will display




The "Extension 9001" configuration page. On the left is a blue sidebar with navigation links: "General Profile", "Voice Message", "Fax Message", and "Personal Greeting". The main content area is titled "Extension 9001" and includes a "Logout" link in the top right. The configuration fields are as follows:

| | |
|---|---|
| User Name | |
| Department Name | |
| Privilege | Regular |
| SIP Password | ***** |
| Voicemail Allowed | Yes |
| Ring Attempts before Forward to Voicemail | 25 (in seconds) |
| Faxmail Allowed | Yes |
| Forward Voicemail/Faxmail to Email | |
| Password to Retrive Voicemail/Faxmail | **** (0-9, cannot be space, *, #) |
| Call Forward | <input checked="" type="radio"/> On <input type="radio"/> Off |
| Call Forward Phone | |
| Call Forward Rule | None |
| Time for No-Answer-Forwarding | 25 (in seconds) |
| Do Not Disturb | <input type="radio"/> On <input checked="" type="radio"/> Off |

At the bottom of the configuration fields are "Submit" and "Reset" buttons.

2. To view current voice mails click on the "Voice Message" button on the left hand side of the interface. Doing this will load the voice mail management page. Here you can save or delete your messages.



The "9001 Voice Message" management page. The left sidebar is the same as the previous page, but "Voice Message" is highlighted in yellow. The main content area is titled "9001 Voice Message" and includes a "Logout" link. It displays a table of messages:

| All | Message | Status | |
|--------------------------|-------------------|---|---|
| <input type="checkbox"/> | 20080108015718.vm | this is new message, receive from 9002 in 2008/01/08 01:57:18 | <input type="button" value="Save"/> <input type="button" value="Delete"/> |

Below the table is a "Delete" button for the selected message. At the bottom right, it says "Index:0~0, Total:1".

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

[General Profile](#)
[Voice Message](#)
[Fax Message](#)
[Personal Greeting](#)

9001 Greeting
[Logout](#)

Personal Greeting

Browse...
Upload
Cancel
Preview

Personal Name

Browse...
Upload
Cancel
Preview

4. Click on the Fax Message button on the left hand menu to load the fax message management page.

[General Profile](#)
[Voice Message](#)
[Fax Message](#)
[Personal Greeting](#)

7001 Fax Message

Language

English ▼

[Logout](#)

There is no FAX Message