



Grandstream Networks, Inc.

GXE5102/5104/5108/5116

All-in-one Hybrid IPPBX Appliance

User Manual

Grandstream Networks, Inc.

www.grandstream.com

GXE5102/5104/5108/5116 User Manual

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CHANGE LOG

This section documents significant changes from previous versions of GXE5102/5104/5108/5116 user manuals. Only major new features or major document updates are listed here. Minor updates for corrections or editing are not documented here.

FIRMWARE VERSION 1.0.0.19

- This is the initial version.

WELCOME

Thank you for purchasing Grandstream GXE5102/5104/5108/5116. GXE5102/5104/5108/5116 is an innovative, all-in-one hybrid IP PBX appliance designed for small to medium business. Powered by an advanced hardware platform with robust system resources, the GXE5102/5104/5108/5116 offers a highly versatile state-of-the-art Unified Communication (UC) solution for converged voice, video, data, fax and video surveillance application needs. Incorporating industry-leading features and performance, the GXE5102/5104/5108/5116 offers quick setup, deployment with ease and unrivaled reliability all at an unprecedented price point.

 **Caution:**

Changes or modifications to this product not expressly approved by Grandstream, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.

 **Warning:**

Please do not use a different power adaptor with the GXE5102/5104/5108/5116 as it may cause damage to the products and void the manufacturer warranty.

This document is subject to change without notice. The latest electronic version of this user manual is available for download here:

<http://www.grandstream.com/support>

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PRODUCT OVERVIEW

FEATURE HIGHLIGHTS

- 1GHz ARM Cortex A8 application processor, large memory (512MB DDR RAM, 4GB NAND Flash), and dedicated high performance multi-core DSP array for advanced voice processing
- Integrated 2/4/8/16 PSTN trunk FXO ports, 2 analog telephone FXS ports, and up to 50 SIP trunk options
- Gigabit network port with integrated PoE, USB, SD; integrated NAT router with advanced QoS support (GXE5102 only)
- Supports a wide range of popular voice codes (including G.711 A-law/U-law, G.722, G.723, G.726, G.729A/B, iLBC, GSM), video codec (including H.264, H.263, H.263+), and Fax (T.38)
- Hardware DSP based 128ms-tail-length carrier-grade line echo cancellation (LEC)
- Supports up to 60 concurrent calls and up to 32 conference attendees
- Flexible dial plan, call routing, site peering, call recording
- Automated detection and provisioning of IP phones, video phones, ATA and other endpoints for easy deployment
- Hardware encryption accelerator to ensure strongest security protection using SRTP, TLS, and HTTPS

TECHNICAL SPECIFICATIONS

Table 1: GXE5102/5104/5108/5116 TECHNICAL SPECIFICATIONS

Interfaces	
Analog Telephone FXS Ports	2 ports
PSTN Line FXO Ports	2 ports (GXE5102); 4 ports (GXE5104); 8 ports (GXE5108); 16 ports (GXE5116)
Network Interfaces	Single or Dual (GXE5102 only) 10M/100M/1000M RJ45 Ethernet port (s) with integrated PoE Plug (IEEE 802.3at-2009)
NAT Router	Yes (GXE5102 only)
Peripheral Ports	USB, SD
LED Indicators	Power/Ready, Network, PSTN Line, USB, SD
LCD Display	128x32 graphic LCD with DOWN and OK button
Reset Switch	Yes
Voice/Video Capabilities	

Voice-over-Packet Capabilities	LEC with NLP Packetized Voice Protocol Unit, 128ms-tail-length carrier grade Line Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711
Voice and Fax Codecs	G.711 A-law/U-law, G.722, G.723.1 5.3K/6.3K, G.726, G.729A/B, iLBC, GSM; T.38
Video Codecs	H.264, H.263, H.263+
QoS	Layer 3 QoS
Signaling and Control	
DTMF Methods	In Audio, RFC2833, and SIP INFO
Provisioning Protocol and Plug-and-Play	TFTP/HTTP/HTTPS, auto-discovery and auto-provisioning of Grandstream IP endpoints
Network Protocols	TCP/UDP/IP, RTP/RTCP, ICMP, ARP, DNS, DDNS, DHCP, NTP, TFTP, SSH, HTTP/HTTPS, PPPoE, SIP (RFC3261), STUN, SRTP, TLS/SIP
Disconnect Methods	Call Progress Tone, Polarity Reversal, Hook Flash Timing, Loop Current Disconnect, Busy Tone
Security	
Media	SRTP, TLS, HTTPS, SSH
Physical	
Universal Power Supply	Output: 12VDC, 1.5A; Input:n 100-240VAC, 50-60Hz
Environmental	Operating: 32 - 104°F / 0 - 40°C, 10-90% (non-condensing); Storage: 14 - 140°F / -10 - 60°C
Dimensions	GXE5102/5104: 226mm (L) x 155mm (W) x 34.5mm (H) GXE5108/5116: 440mm (L) x 185mm (W) x 44mm (H)
Mounting	Wall mount and Desktop
Additional Features	
Caller ID	Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF, SIN 227 - BT, NTT Japan (pending)
Polarity Reversal/ Wink	Yes, with enable/disable option upon call establishment and termination
Call Center	Multiple configurable call queues, automatic call distribution (ACD) based on agent skills/availability busy level, in-queue announcement
Customizable Attendant	Auto Up to 5 layers of IVR (Interactive Voice Response)
Concurrent Calls	Up to 30 (GXE5102), 45 (GXE5104), or 60 (GXE5108/5116)

	simultaneous calls
Conference Bridges	Up to 3 (GXE5102/5104) or 6 (GXE5108/GXE5116) password-protected conference bridges allowing up to 25 (GXE5102/5104) or 32 (GXE5108/5116) simultaneous PSTN or IP participants
Call Features	Call park, call forward, call transfer, DND, ring/hunt group, paging/intercom etc
Compliance	FCC: Part 15 (CFR 47) Class B, Part 68 CE: EN55022 Class B, EN55024, EN61000-3-2, EN61000-3-3, EN60950-1, TBR21, RoHS A-TICK: AS/NZS CISPR 22 Class B, AS/NZS CISPR 24, AS/NZS 60950, AS/ACIF S002 ITU-T K.21 (Basic Level); UL 60950 (power adapter)

INSTALLATION

EQUIPMENT PACKAGING

Table 2: GXE5102/5104 EQUIPMENT PACKAGING

Main Case	Yes (1)
Power Adaptor	Yes (1)
Ethernet Cable	Yes (1)
Quick Start Guide	Yes (1)

Table 3: GXE5108/5116 EQUIPMENT PACKAGING

Main Case	Yes (1)
Power Adaptor	Yes (1)
Ethernet Cable	Yes (1)
Quick Start Guide	Yes (1)
Wall Mount	Yes (2)
Screws	Yes (6)

CONNECTING YOUR GXE5102/5104/5108/5116

The following example shows GXE5104 connection.



Figure 1: GXE5104 Front View

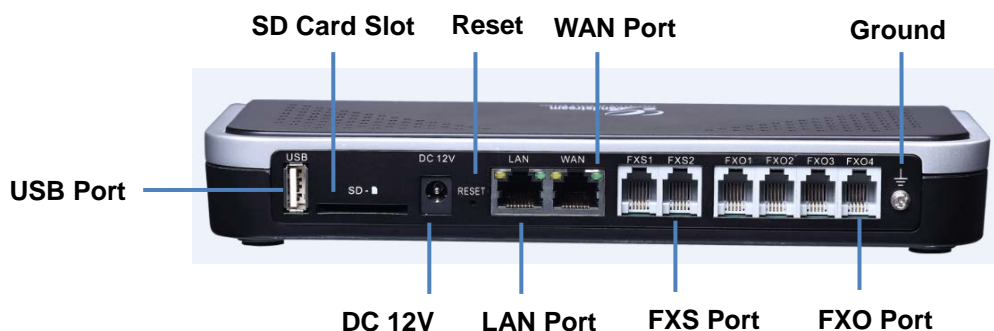


Figure 2: GXE5104 Back View

To set up the GXE5102/5104 (WAN port and LAN port), follow the steps below:

1. Connect one end of an RJ-45 Ethernet cable into the WAN port of the GXE5102/5104;
2. Connect the other end of the Ethernet cable into the uplink port of an Ethernet switch/hub;
3. Connect the 12V DC power adapter into the 12V DC power jack on the back of the GXE5102/5104. Insert the main plug of the power adapter into a surge-protected power outlet;
4. Let the GXE5102/5104 boot up for the first time. You will know it is finished with the boot process when the LCD is on. The model and hardware information will show up in the LCD.
5. Once connected to network, the Network LED indicator is solid green and the LCD shows up the IP address;
6. Connect PSTN lines from the wall jack to the FXO ports, and analog lines (phone and fax) the FXS ports. IP phones will be connected in a later step.

To set up the GXE5108/5116 (LAN port only), follow the steps below:

1. Connect one end of an RJ-45 Ethernet cable into the LAN port of the GXE5108/5116;
2. Connect the other end of the Ethernet cable into the uplink port of an Ethernet switch/hub;
3. Connect the 12V DC power adapter into the 12V DC power jack on the back of the GXE5108/5116. Insert the main plug of the power adapter into a surge-protected power outlet;
4. Let the GXE5108/5116 boot up for the first time. You will know it is finished with the boot process when the LCD is on. The model and hardware information will show up in the LCD.
5. Once connected to network, the Network LED indicator is solid green and the LCD shows up the IP address;
6. Connect PSTN lines from the wall jack to the FXO ports, and analog lines (phone and fax) the FXS ports. IP phones will be connected in a later step.

SAFETY COMPLIANCES

The GXE5102/5104/5108/5116 complies with FCC/CE and various safety standards. The GXE5102/5104/5108/5116 power adapter is compliant with the UL standard. Use the universal power adapter provided with the GXE5102/5104/5108/5116 package only. The manufacturer's warranty does not cover damages to the device caused by unsupported power adapters.

WARRANTY

If the GXE5102/5104/5108/5116 was purchased from a reseller, please contact the company where the device was purchased for replacement, repair or refund. If the device was purchased directly from Grandstream, contact the Grandstream Sales and Service Representative for a RMA (Return Materials Authorization) number before the product is returned. Grandstream reserves the right to remedy warranty policy without prior notification.

 **Warning:**

Use the power adapter provided with the GXE5102/5104/5108/5116. Do not use a different power adapter as this may damage the device. This type of damage is not covered under warranty.

GETTING TO KNOW GXE5102/5104/5108/5116

USING THE LCD MENU

NAVIGATION IN LCD MENU

- **Default LCD display.** By default, when the device is powered on, the LCD will show device model, hardware version and IP address.
- **Enter Menu.** Press "Down" or "OK" button to start browsing menu options.
- **Navigate in the menu options.** Press the "Down" arrow key to browser different menu options.
- **Select menu entries.** Press the "OK" button to select an entry.
- **Exit.** There is "Back" option in the menu. Select it to go back to previous menu.
- The LCD will come back to default display after being idle in menu for more than 20 seconds.

LCD MENU OPTIONS

The following table shows the LCD menu options.

Table 4: GXE5102/5104/5108/5116 LCD MENU OPTIONS

View Events	<ul style="list-style-type: none"> • Critical Events • Other Events
Device Info	<ul style="list-style-type: none"> • Hardware: Hardware version number • Software: Software version number • P/N: Part number • MAC: MAC address • Uptime: System up time
Network Info	<ul style="list-style-type: none"> • Mode: DHCP, Static IP, or PPPoE • IP: IP address • Subnet Mask
Network Menu	<ul style="list-style-type: none"> • LAN Mode Select LAN mode as DHCP, Static IP or PPPoE
Factory Menu	<ul style="list-style-type: none"> • LCD Test Patterns

- **Fan Mode:** Auto or On
- **Language Test**

USING THE WEB GUI

The GXE5102/5104/5108/5116 embedded Web server responds to HTTP/HTTPS GET/POST requests. Embedded HTML pages allow users to configure the GXE through a Web browser such as Microsoft's IE, Mozilla Firefox, Google Chrome and etc.



Figure 3: GXE5102/5104/5108/5116 Web GUI - Login

ACCESSING WEB GUI

To access the Web GUI:

1. Connect the computer to the same network as the GXE5102/5104/5108/5116;
2. Make sure the GXE is turned on and shows its IP address on the LCD;
3. Open a Web browser on your computer;
4. Enter the GXE's IP address in the address bar of the browser. The Web GUI URL format is:

http(s)://GXE-IP-Address:Port


where the GXE-IP-Address is the IP address displayed on the GXE5102/5104/5108/5116 LCD. By default, the protocol is HTTPS and the Port number is 8089.

For example, if the GXE LCD shows 192.168.40.167, please enter:

https://192.168.40.167:8089

5. Enter the administrator's login and password to access the Web Configuration Menu. The default administrator's username and password is "admin" and "admin".

SAVING AND APPLYING CHANGES

After configuring from web GUI options, click on  to save the change. Then click on "Apply Changes" button (if displayed) on the top right of the web page to submit the changes. Follow the prompted message to reboot the GXE if it's required.

GXE5102/5104/5108/5116 CONFIGURATIONS

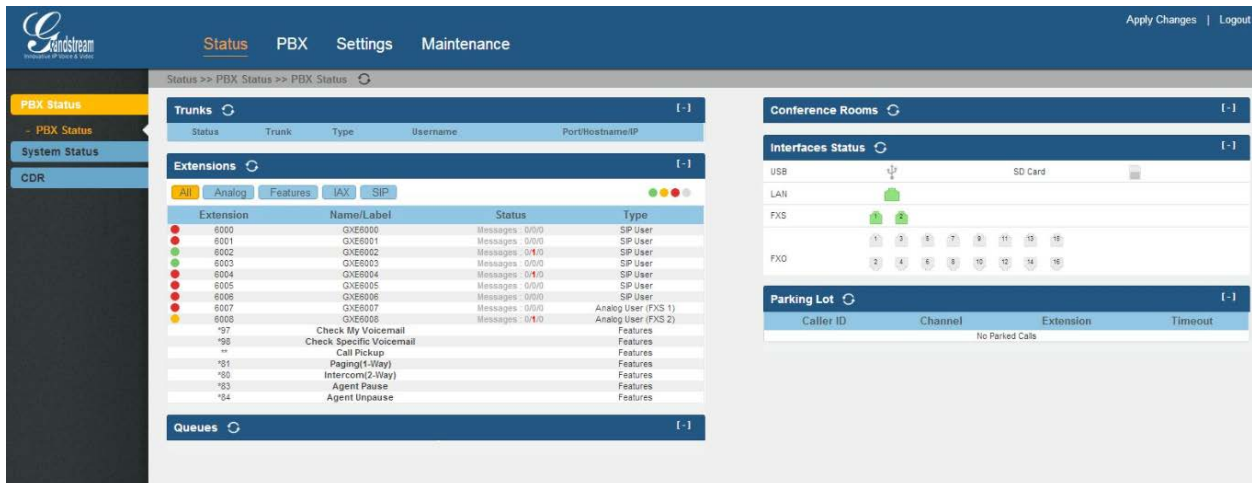
This section describes the options in the GXE5102/5104/5108/5116 Web GUI.

- **Status:** Displays PBX status, System Status and CDR.
- **PBX:** To configure extensions, trunks, call control options (inbound routes, conference, IVR, Ring Group and etc), internal options (call features, music on hold, IVR Prompt, FAX and etc), IAX Settings and SIP Settings.
- **Settings:** To configure network settings, change password, LDAP Server, HTTP Server, Email Settings and Time Settings.
- **Maintenance:** To configure syslog, upgrade, backup, reset, reboot, and perform troubleshooting.

STATUS PAGE DEFINITIONS

STATUS/PBX STATUS

In PBX Status page, there are different sections to display status for Trunks, Extensions, Queues, Conference Rooms, Interfaces and Parking lot.



The screenshot displays the 'Status -> PBX Status' page in the Grandstream Web GUI. The page has a dark blue header with the Grandstream logo and navigation tabs for 'Status', 'PBX', 'Settings', and 'Maintenance'. A left sidebar contains a menu with 'PBX Status' highlighted. The main content area is divided into several sections:

- Trunks:** A table with columns for Status, Trunk, Type, Username, and Port/Hostname/IP.
- Extensions:** A table with columns for Extension, Name/Label, Status, and Type. It includes a sub-menu for 'All' and 'Analog'. The table lists extensions 6000 through 6008, each with a status indicator (red or green dot) and a 'Messages' count (0/0). Extension 6008 is an 'Analog User (FXS 1)'. Below this table are several feature settings like 'Check My Voicemail', 'Call Pickup', 'Paging(1-Way)', etc.
- Queues:** A section with a refresh icon and a count of 1.
- Conference Rooms:** A section with a refresh icon and a count of 1.
- Interfaces Status:** A section showing the status of various interfaces: USB (SD Card), LAN, FXS (with a grid of 16 ports), and FXO (with a grid of 16 ports).
- Parking Lot:** A table with columns for Caller ID, Channel, Extension, and Timeout. It currently shows 'No Parked Calls'.

Figure 4: Status->PBX Status

STATUS/SYSTEM STATUS

Status ->System Status -> General

Model	Product model of the GXE.
Part Number	Product part number.
System Time	System time of the GXE.
Up Time	System up time since the last reboot.
Idle Time	System idle time since the last reboot.
Boot	Boot version.
Core	Core version.
Base	Base version.
Program	Program version. This is the main software release version.
Recovery	Recovery version.

Status -> System Status -> Network

MAC Address	Global unique ID of device, in HEX format. The MAC address will be used for provisioning and can be found on the label coming with original box and on the label located on the back of the device.
IP Address	IP address of the device.
Gateway	Default gateway of the device.
Subnet Mask	Subnet mask of the device.
DNS	DNS Server of the device.

Status -> System Status -> Disk Usage

Disk cfg	Displays total, available and used space for disk cfg.
Disk data	Displays total, available and used space for disk data.

Status -> System Status -> Resource Usage

CPU Usage	Displays percentage of CPU usage.
Memory Usage	Display available and used space for memory.



Figure 5: Status->System Status -> Disk Usage

STATUS/CDR

CDR status page shows call detail information as well as statistics. This can be used for business analysis.

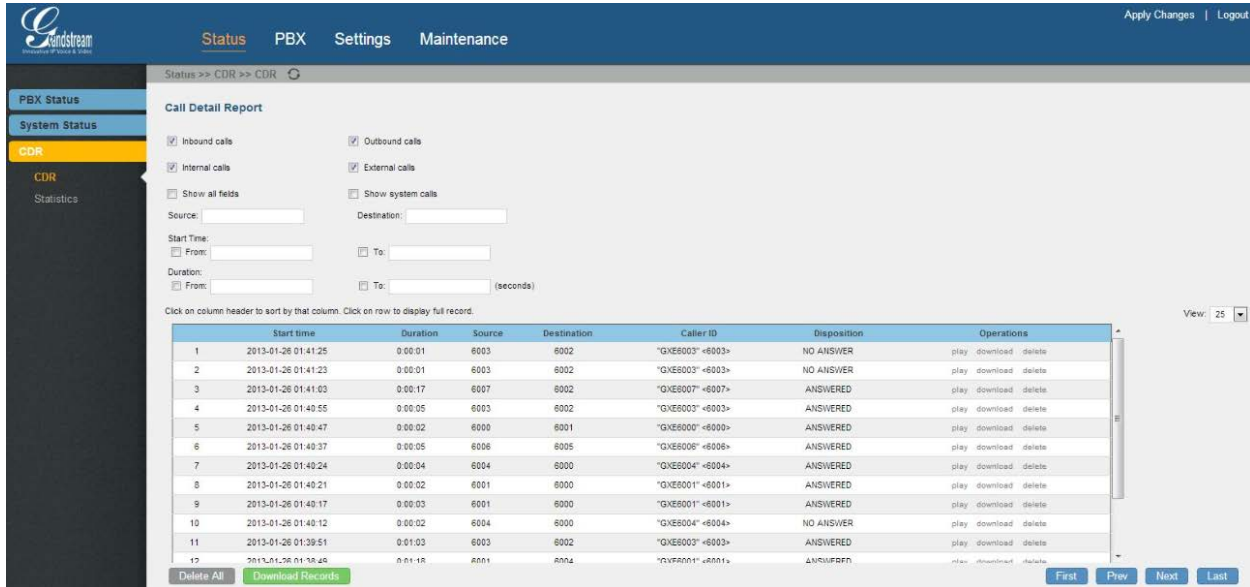


Figure 6: Status->System Status -> CDR

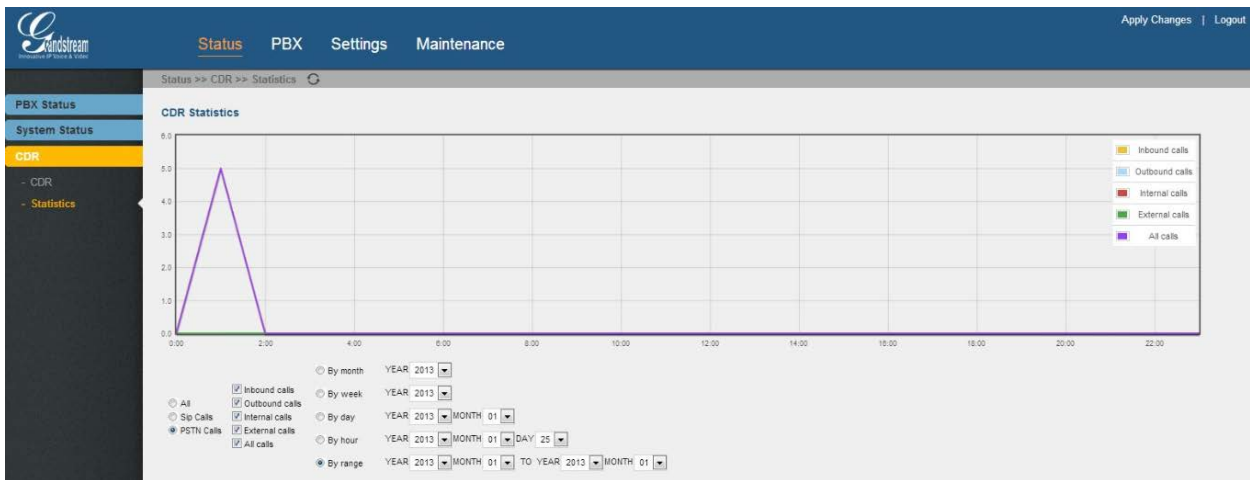


Figure 7: Status->System Status -> CDR Statistics

PBX PAGE DEFINITIONS

BASIC/EXTENSIONS

In this page, users could view, create, edit and delete extensions. The extension status will show in the list with Caller ID Name, IP: Port, connection status and etc.

List of User Extensions

Extension	CallerID Name	Technology	IP and Port	Status	Options
6000	GXE6000	SIP	192.168.40.102:12111	Online	Reboot Edit Delete
6001	GXE6001	SIP	192.168.40.225:24150	Online	Reboot Edit Delete
6002	GXE6002	SIP	192.168.40.191:5060	Online	Reboot Edit Delete
6003	GXE6003	SIP	192.168.40.177:5060	Online	Reboot Edit Delete
6004	GXE6004	SIP	192.168.40.181:5060	Online	Reboot Edit Delete
6005	GXE6005	SIP	192.168.40.179:5060	Online	Reboot Edit Delete
6006	GXE6006	SIP	192.168.40.184:5060	Online	Reboot Edit Delete
6007	GXE6007	FXS1	--	Offline	Reboot Edit Delete
6008	GXE6008	FXS2	--	Offline	Reboot Edit Delete

Total 9 Show: 1/1 Jump to:

Figure 8: PBX->Basic->Extensions

Click on "Create New User" button, the following window will show.

X
Create New User

General

Extension:
 CallerID Name:

CallerID Number:
 Permission:

SIP/IAX Password:
 Enable Voicemail:

VoiceMail Password:
 Email Address:

Technology

SIP:
 IAX:

Analog Station:

SIP Settings

NAT:
 Can Reinvite:

DTMF Mode:
 Insecure:

Other Options

SRTP:
 FaxDetect:

Strategy:
 Disable Pin:

Codec Preference:

Selected Codecs

- PCMU
- PCMA
- GSM
- G.726
- H.264

Available Codec

- ILBC
- G.722
- ADPCM
- LPC10
- G.729

Figure 9: Create New User

BASIC/ANALOG TRUNKS

In this page, users could view, create and manage analog trunks. To create new trunk, click on "Create New Analog Trunk" and fill out the information in the prompted window as below.

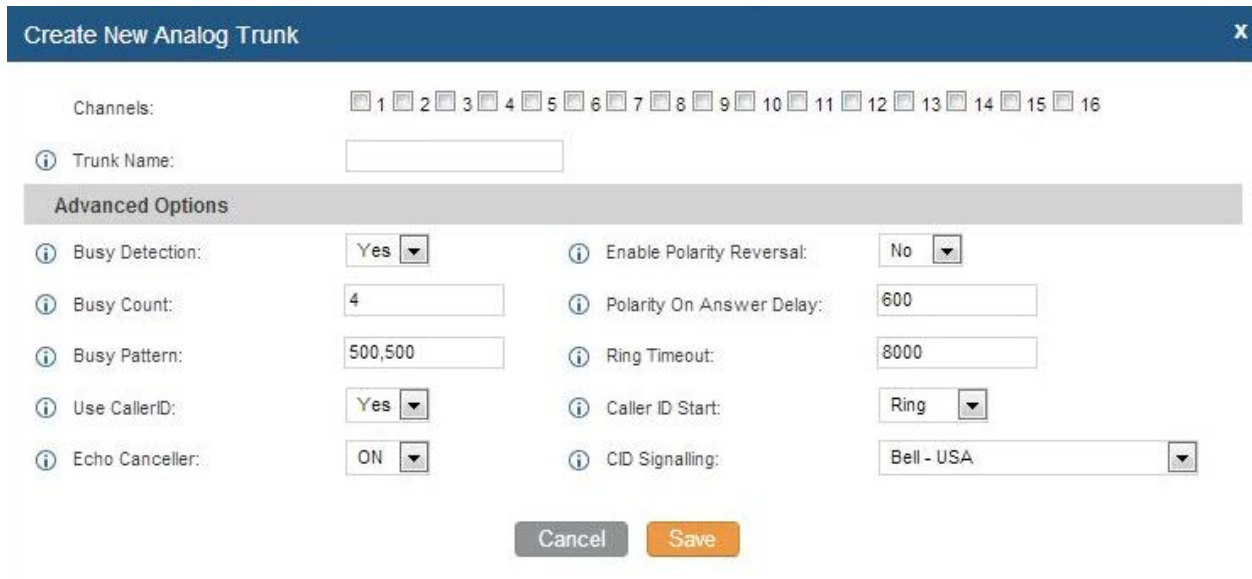


Figure 10: Create New Analog Trunk

BASIC/VOIP TRUNKS

In this page, users could view, create and manage SIP/IAX trunks. To create new trunk, click on "Create New SIP/IAX Trunk" and fill out the information in the prompted window as below.

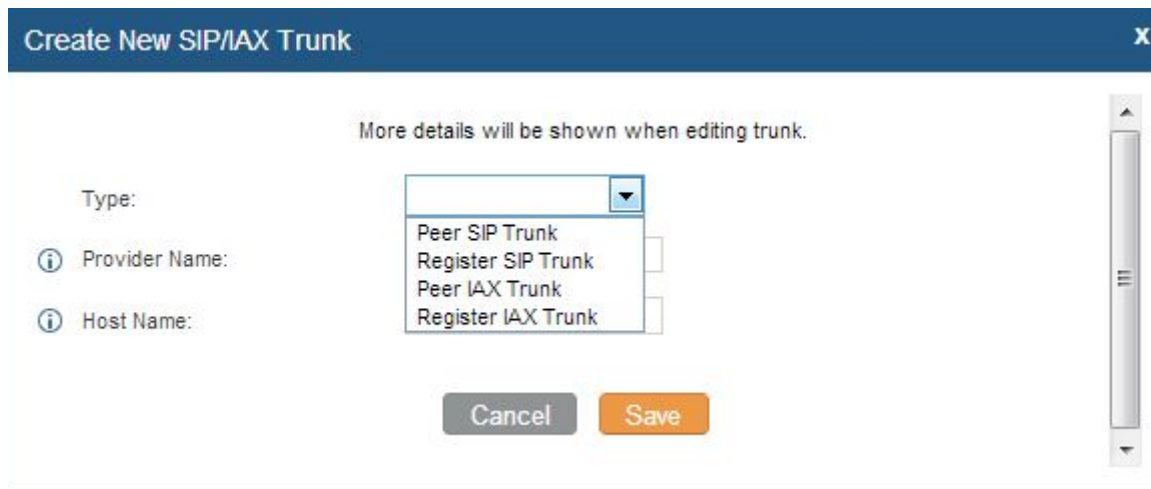
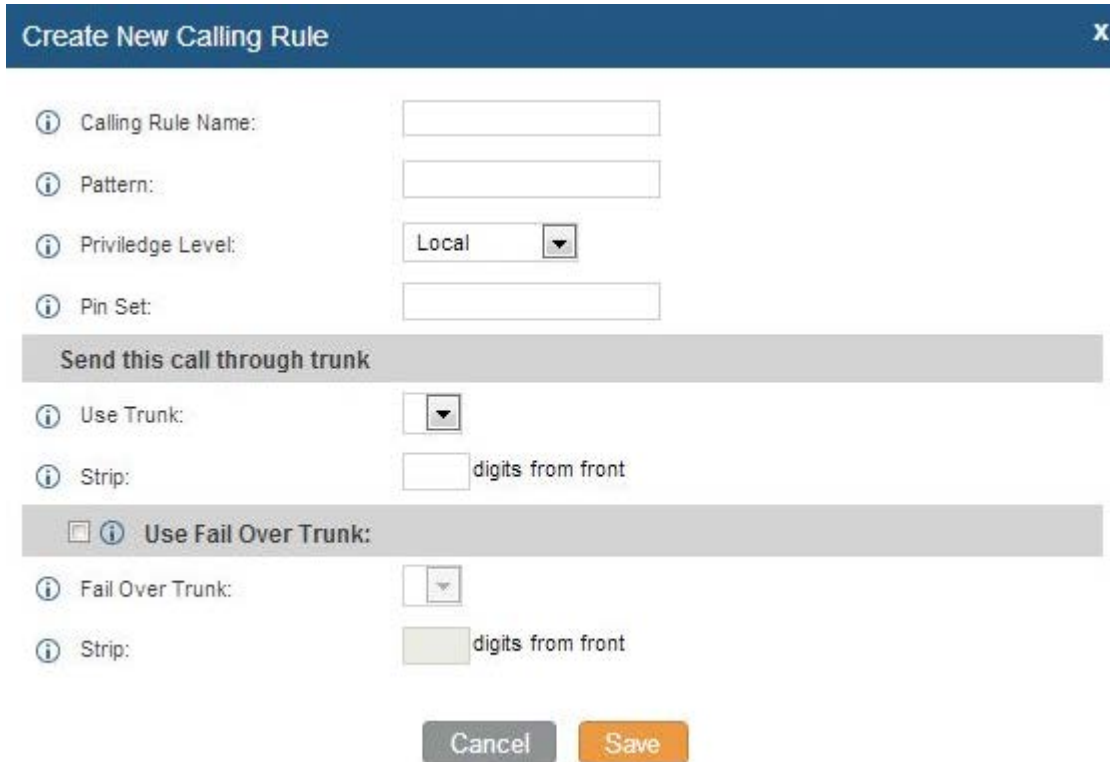


Figure 11: Create New SIP/IAX Trunk

BASIC/OUTBOUND ROUTES

In this page, users could view, create and manage calling rules. To create new route, click on "Create New Calling Rule" and fill out the information in the prompted window as below.



Create New Calling Rule X

Calling Rule Name:
 Pattern:
 Privilege Level: Local
 Pin Set:

Send this call through trunk

Use Trunk:
 Strip: digits from front

Use Fail Over Trunk:

Fail Over Trunk:
 Strip: digits from front

Figure 12: Create New Calling Rule

BASIC/ZERO CONFIG

By default, zero config feature is turned on in GXE5102/5104/5108/5116 so the devices connected in the same LAN will be auto provisioned by the GXE. There are three methods of auto provision: SIP subscribe, DHCP option 66 and mDNS. Basically, when the device boots up, it will send the SUBSCRIBE to LAN. The GXE will find it, create and assign an extension to the device, and then return the url of config file for the device to download.

Here is the Auto Provision Setting:

Auto Provision Setting X

Auto provision is automatically provides a extension to device . There are three methods of auto provision: SIP subscribe, DHCP option 66, mDNS.

For extension, one devices boot, it will send the subscribe broadcast, the server will find it and create an account, then return it a url of config file.

Enable Zero Config:

<input type="checkbox"/>	Module Name	Version
<input checked="" type="checkbox"/>	mdns	--
<input checked="" type="checkbox"/>	dhcp	--
<input checked="" type="checkbox"/>	sip	--

Automatically Assign User:

User Starting Number:

Generate Random Password:

Default Password:

Figure 13: Zero Config: Auto Provision Setting

The following picture shows the provisioned devices from the GXE zero config feature.

List of Devices Extensions

Filter: All

<input type="checkbox"/>	No.	Mac	IP	Extension	Version	Vendor	Model	Connect State	Config File	<input type="button" value="Edit"/>	<input type="button" value="Delete"/>
<input type="checkbox"/>	1.	000B823E1D8B	192.168.40.225	6001	1.0.1.33	Grandstream	GXP2200	Connected	Yes	<input type="button" value="Edit"/>	<input type="button" value="Delete"/>
<input type="checkbox"/>	2.	000B823E175D	192.168.40.102	6000	1.0.1.33	Grandstream	GXP2200	Connected	Yes	<input type="button" value="Edit"/>	<input type="button" value="Delete"/>
<input type="checkbox"/>	3.	000B822B1641	192.168.40.179	6005	1.0.5.15	Grandstream	GXP2100	Connected	Yes	<input type="button" value="Edit"/>	<input type="button" value="Delete"/>
<input type="checkbox"/>	4.	000B8240E529	192.168.40.184	6006	1.0.5.15	Grandstream	GXP1160	Connected	Yes	<input type="button" value="Edit"/>	<input type="button" value="Delete"/>
<input type="checkbox"/>	5.	000B822B2A24	192.168.40.181	6004	1.0.5.15	Grandstream	GXP2110	Connected	Yes	<input type="button" value="Edit"/>	<input type="button" value="Delete"/>
<input type="checkbox"/>	6.	000B823AB2E1	192.168.40.157	6002	1.0.5.15	Grandstream	GXP2124	Connected	No	<input type="button" value="Edit"/>	<input type="button" value="Delete"/>
<input type="checkbox"/>	7.	000B823A045	192.168.40.177	6003	1.0.5.15	Grandstream	GXP1450	Connected	Yes	<input type="button" value="Edit"/>	<input type="button" value="Delete"/>

Figure 14: Zero Config: Provisioned Devices

CALL CONTROL/CONFERENCE

In this page, users could view, create, edit and delete conference rooms. The conference room status and activity will show in the list. Click on "Create New Conference Bridge" to create and configure conference room.

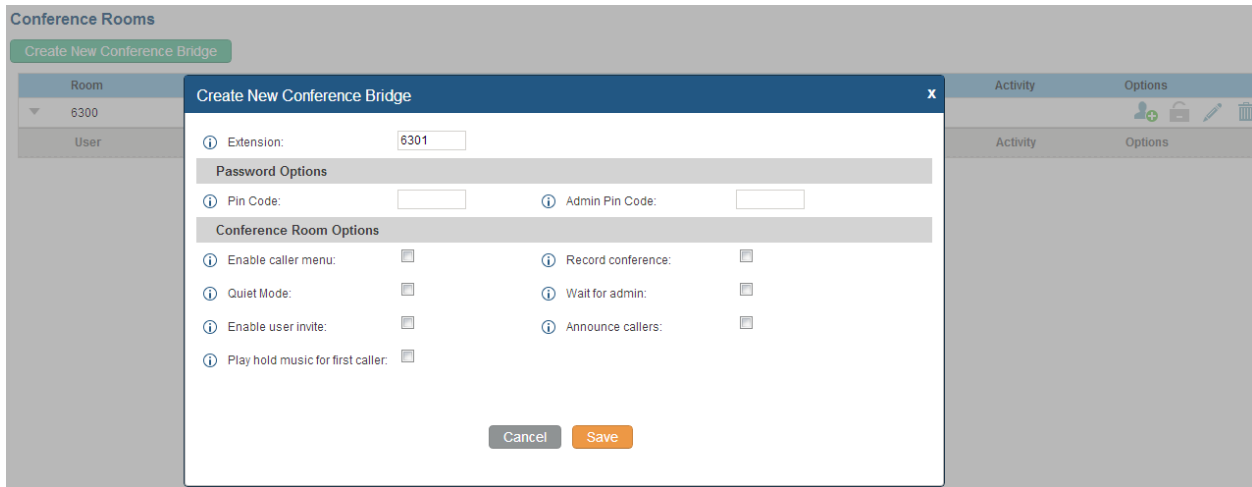


Figure 15: Create New Conference Bridge

CALL CONTROL/IVR

The IVR in the PBX system can be configured under **Call Control->IVR** page. Click on "Create New IVR" and fill in the information in the window as below.

Create New IVR X

ⓘ Name:

ⓘ Extension:

ⓘ Dial Other Extensions:

ⓘ Welcome Prompt: None [Prompt](#)

ⓘ Timeout:

ⓘ Timeout Prompt: None

ⓘ Invalid Prompt: None

ⓘ Repeat Loops: 1

KeyPress Events

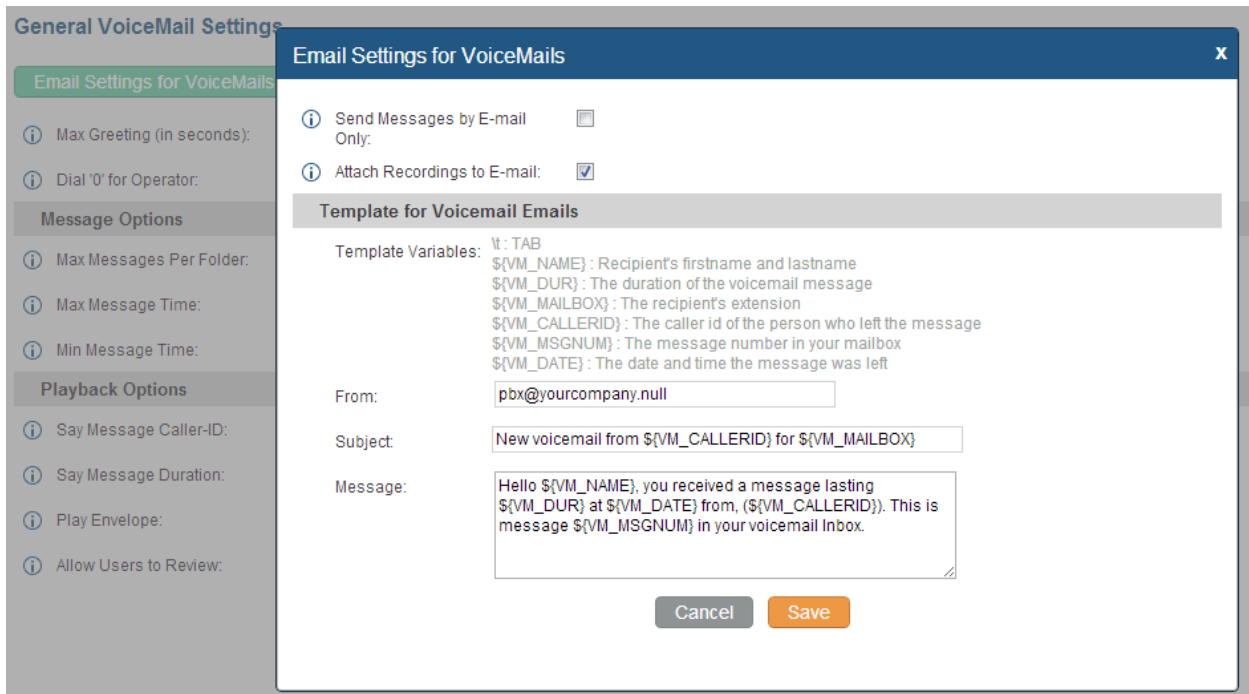
Press 0:	Select an Option
Press 1:	Select an Option
Press 2:	Select an Option
Press 3:	Select an Option
Press 4:	Select an Option
Press 5:	Select an Option
Press 6:	Select an Option
Press 7:	Select an Option
Press 8:	Select an Option
Press 9:	Select an Option
Press #:	Select an Option
Press *:	Select an Option

Cancel
Save

Figure 16: Create New IVR

CALL CONTROL/VOICEMAIL

General Voicemail settings can be configured under **Call Control->Voicemail** page. Users could also set up email for the voicemails. Click on "Create New IVR" and fill in the information in the window as below.



General VoiceMail Settings

Email Settings for VoiceMails

Send Messages by E-mail Only:
 Attach Recordings to E-mail:

Template for Voicemail Emails

Template Variables:
 \t : TAB
 \${VM_NAME} : Recipient's firstname and lastname
 \${VM_DUR} : The duration of the voicemail message
 \${VM_MAILBOX} : The recipient's extension
 \${VM_CALLERID} : The caller id of the person who left the message
 \${VM_MSGNUM} : The message number in your mailbox
 \${VM_DATE} : The date and time the message was left

From:
 Subject:
 Message:

Figure 17: Voicemail Settings

CALL CONTROL/VOICEMAIL GROUPS

In this page, users could create extension for voicemail group which contains members that will receive the voicemail if the group extension has voice messages. Click on "New VoiceMail Group" to add groups.

List of VoiceMail Groups

[New VoiceMail Group](#)

Extension for VoiceMail Group	Label	Member MailBoxes	
6600	GXE_VM1	6000, 6001, 6002	<input type="button" value="Edit"/> <input type="button" value="Delete"/>

Figure 18: Voicemail Groups Settings

CALL CONTROL/RING GROUP

In this page, users could create extension for ring group which contains members that will receive the call with specific ring strategy if the group extension has incoming calls.

Create New Ring Group
X

Ring Group Name:

Extension:

Ring Group Members

- ⊕
6000 "GXE6000"
▲
- ⊕
6001 "GXE6001"
▲
- ⊕
6002 "GXE6002"
▲
- ⊕
6003 "GXE6003"
▲

Available Users

- ⊖
6004 "GXE6004"
▲
- ⊖
6005 "GXE6005"
▲
- ⊖
6006 "GXE6006"
▲
- ⊖
6007 "GXE6007"
▲
- ⊖
6008 "GXE6008"
▲
- ⊖
6009
▲

Ring Group Options

Ring Strategy:

i Seconds to Ring Each Member:

i Enable Voicemail:

Secret:

Email Address:

Cancel
Save

Figure 19: Ring Group Settings

CALL CONTROL/PAGE AND INTERCOM GROUPS

Paging and intercom can be configured in group level or per extension. In this page, users could add paging and intercom groups.

X
Create New Page/Intercom Group

Extension:

Type: 2-Way Intercom ▼

Page/Intercom Group Member

SIP/6000 "GXE6000"

SIP/6001 "GXE6001"

SIP/6002 "GXE6002"

SIP/6003 "GXE6003"

SIP/6004 "GXE6004"

⏪ ⏩

Available Users

SIP/6005 "GXE6005"

SIP/6006 "GXE6006"

SIP/6009

SIP/6010

SIP/6011

SIP/6012

Cancel
Save

X
Paging/Intercom Group Settings

Settings for Paging & Intercom

i Alert-Info Header:

Settings For Paging Individual Extensions

Please go to [Call Features](#) for setting paging individual extensions.

Cancel
Save

Figure 20: Paging/Intercom Group Settings

CALL CONTROL/FOLLOW ME

Follow Me feature allows users to add internal extensions or external numbers to follow existed extensions when there is incoming call. If the main extension has incoming call rejected or unanswered, the call will be routed to the FollowMe numbers in specified order or simultaneously. The caller will hear music on hold when calling the main extension number.

Edit Extension: 6000
X

i Status: Enable Disable

i 'Music On Hold' Class: default

Follow Me Numbers

6000 (30 seconds)

6001 & 6002 (30 seconds) ⏏ ⏴ ⏵

i New FollowMe Number: Dial Local Extension Dial Outside Number

6003 "GXE6003" for 30 Seconds

i Dial Order: Ring after Trying previous extension/number
 Ring along with previous extension/number

Cancel
↑ Add

Figure 21: Follow Me Settings

CALL CONTROL/CALL QUEUE

Call queues can be configured under **Call Control->Call Queue**. The login and logout code can be configured under "Agent Login Settings".

Create New Queue X

ⓘ Extension: ⓘ Name:
 ⓘ Strategy: ⓘ Music On Hold:
 ⓘ Leave When Empty: ⓘ Join Empty:
 ⓘ Dynamic Login PIN:

Queue Options

ⓘ TimeOut: ⓘ Wrapup Time:
 ⓘ Max Len: ⓘ Report Hold Time:
 ⓘ Wait Time:

Agents ⓘ

Static Agents	Available Users
<input type="checkbox"/> SIP/6000 "GXE6000" <input type="checkbox"/> SIP/6001 "GXE6001" <input type="checkbox"/> SIP/6002 "GXE6002"	<input type="checkbox"/> SIP/6003 "GXE6003" <input type="checkbox"/> SIP/6004 "GXE6004" <input type="checkbox"/> SIP/6005 "GXE6005" <input type="checkbox"/> SIP/6006 "GXE6006" <input type="checkbox"/> SIP/6009 <input type="checkbox"/> SIP/6010

Agent Login Settings X

Agent Login Settings

ⓘ Agent Login Extension Postfix:
 ⓘ Agent Logout Extension Postfix:

Example: If Queue Extension is 6500,
 Agent Login Extension Postfix is *,
 Agent Logout Extension Postfix is **,
 Dial [6500*] to Login, [6500**] to Logout.

Figure 22: Call Queue Settings

INTERNAL OPTIONS/GENERAL

Global OutBound CID	This is the default global CallerID that is used for all outgoing calls when no other CallerID is defined. If the "User" tab or "VoIP Trunks" tab does not have defined CallerID neither, this Global OutBound CID will be used for CallerID.
Global OutBound CID Name	This is the global CallerID Name that is used for all outgoing calls. If this value is defined, all outgoing calls will have a "CallerId Name" set to this value. Usually this value could be your company name. Leave this value blank if you would like to have the users' "CallerID Name" display on outbound calls.
Operator Extension	The operator extension is the number dialed when users press "0" to exit Voicemail. It's also available in IVR option.
Ring Timeout	Number of seconds to ring an extension before sending to the user's voicemail box.

Users could also configure the extension preference for different functions. It is recommended to keep this feature on so the extensions could be properly arranged and used.

Extension preferences

Disable Extension Ranges:

User Extensions: -

Conference Extensions: -

IVR Extensions: -

Ring Group Extensions: -

Queue Extensions: -

VoiceMail Group Extensions: -

Figure 23: Extension Preferences

INTERNAL OPTIONS/CALL FEATURES

In this page, users could configure feature code for the following call features in the PBX. The default setting is listed in the following table.

Blind Transfer	#1
Attended Transfer	*2
Disconnect	*0
Call Parking	#72
Audio Record	*1
Audio Mix Record	*3
Do Not Disturb (DND) Active	*77
Do Not Disturb (DND) Deactive	*78
Call Forward Busy Active	*71
Call Forward Busy Deactive	*72
Call Forward No Answer Active	*73
Call Forward No Answer Deactive	*74
Call Forward Uncondition Active	*75
Call Forward Uncondiion Deactive	*76
Feature Digit Timeout	1000
Extension to Dial to Park a Call	700
Extensions for Parked Calls	701-720
Parked Call Timeout (in secs)	120
Dial Voice Mail	*98
Voice Mail Main	*97
Agent Pause	*83
Agent Unpause	*84
Paging Prefix	*81
Intercom Prefix	*80
Call Pickup	**

Call Features

Feature Maps

Reset All
Default All

ⓘ Blind Transfer: <input style="width: 40px;" type="text" value="#1"/> Neither	ⓘ Attended Transfer: <input style="width: 40px;" type="text" value="2"/> Neither	ⓘ Disconnect: <input style="width: 40px;" type="text" value="0"/> Neither
ⓘ Call Parking: <input style="width: 40px;" type="text" value="#72"/> Neither	ⓘ Audio Record: <input style="width: 40px;" type="text" value="1"/> Neither	ⓘ Audio Mix Record: <input style="width: 40px;" type="text" value="3"/> Neither

Feature DND/Forward

Reset All
Default All

ⓘ Do Not Disturb (DND) Active: <input style="width: 40px;" type="text" value="*77"/>	ⓘ Do Not Disturb (DND) Deactive: <input style="width: 40px;" type="text" value="*78"/>	ⓘ Call Forward Busy Active: <input style="width: 40px;" type="text" value="*71"/>
ⓘ Call Forward Busy Deactive: <input style="width: 40px;" type="text" value="*72"/>	ⓘ Call Forward NoAnswer Active: <input style="width: 40px;" type="text" value="*73"/>	ⓘ Call Forward NoAnswer Deactive: <input style="width: 40px;" type="text" value="*74"/>
ⓘ Call Forward Uncondition Active: <input style="width: 40px;" type="text" value="*75"/>	ⓘ Call Forward Uncondition Deactive: <input style="width: 40px;" type="text" value="*76"/>	

Feature Misc

Reset All
Default All

Feature Digit Timeout: <input style="width: 40px;" type="text" value="1000"/>	Extension to Dial to Park a Call: <input style="width: 40px;" type="text" value="700"/>	ⓘ Extensions for Parked Calls: <input style="width: 40px;" type="text" value="701-720"/>
Parked Call Timeout (in secs): <input style="width: 40px;" type="text" value="120"/>		

Feature Code

Reset All
Default All

ⓘ Dial Voice Mail: <input style="width: 40px;" type="text" value="*98"/>	ⓘ Voice Mail Main: <input style="width: 40px;" type="text" value="*97"/>	ⓘ Agent Pause: <input style="width: 40px;" type="text" value="*83"/>
ⓘ Agent Unpause: <input style="width: 40px;" type="text" value="*84"/>	ⓘ Paging Prefix: <input style="width: 40px;" type="text" value="*81"/>	ⓘ Intercom Prefix: <input style="width: 40px;" type="text" value="*80"/>
ⓘ Call Pickup: <input style="width: 40px;" type="text" value="**"/>		

Cancel
Save

Figure 24: Call Features

INTERNAL OPTIONS/MUSIC ON HOLD

In this page, users could configure music on hold class and the music files. The music file uploaded has to be 8 KHz Mono format with size less than 5M.

Manage 'Music On Hold' Classes

[Create New MOH class](#)

Manage Music-on-Hold Classes:

Upload an 8 KHz Mono Music file (size less then 5M):

Choose file to Upload:

List of Sound Files

Sound File	Options
macroform-cold_day.wav	<input type="button" value="Delete"/>
macroform-robot_dity.wav	<input type="button" value="Delete"/>
macroform-the_simplicity.wav	<input type="button" value="Delete"/>
manolo_camp-morning_coffee.wav	<input type="button" value="Delete"/>
reno_project-system.wav	<input type="button" value="Delete"/>

Figure 25: Music On Hold Settings

INTERNAL OPTIONS/IVR PROMPT

In this page, users could record new IVR prompt or upload a new file for IVR prompt. The uploaded file has to be small than 5M in 8KHz mono, 16 bits GSM or WAV format. Or it could be raw ulaw/alaw file with the .ulaw/.alaw suffix.

List of Custom IVR Prompts

[Record New IVR prompt](#) [Upload IVR prompt](#)

No custom IVR prompts found !!

You can record a new IVR Prompt by clicking on the 'Record a new IVR prompt' or click on the 'Upload a IVR prompt' button to upload a custom IVR.

Upload IVR prompt x

Choose voice prompt to upload :

Each file uploaded must be less than 5 megabytes, in 8KHz mono, 16bits and in GSM or WAV format, or raw ulaw/alaw file with the .ulaw/.alaw suffix.

Choose file to Upload:

Figure 26: IVR Prompt Settings

INTERNAL OPTIONS/FAX T.38

In this page, users could create and configure FAX extensions on the GXE.

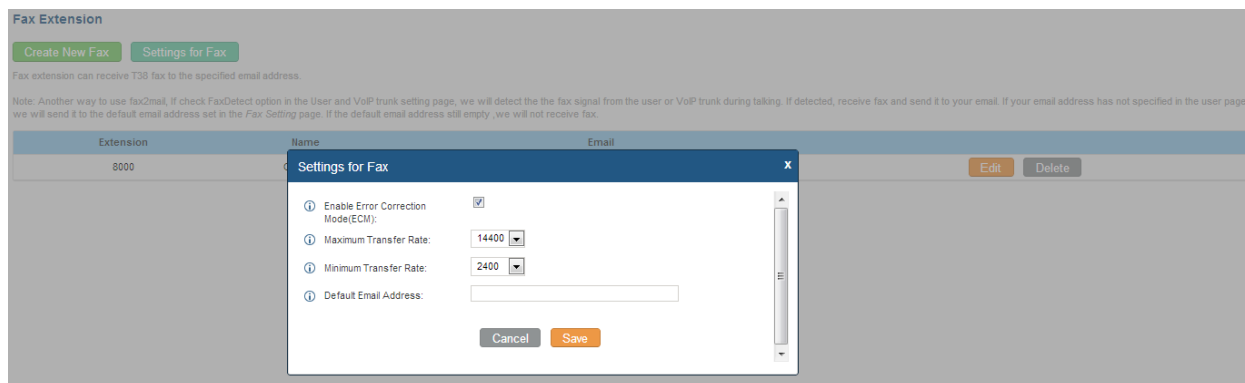


Figure 27: FAX Settings

INTERNAL OPTIONS/RTP SETTINGS

RTP Start	RTP port starting address. The default setting is 10000.
RTP End	RTP port ending address. The default setting is 20000.
Strict RTP	Enables/disables strict RTP protection. When enabled, RTP packets that do not come from the source of the RTP stream will be dropped. The default setting is "Disable".
RTP Checksums	Enables/Disables RTP Checksums. The default setting is "Disable".

INTERNAL OPTIONS/HARDWARE CONFIG

In this page, users could configure the signaling preference for each FXS and FXO ports, as well as region preference for Tone and Opermode. Other advanced settings such as PCMA Override, FXS Honor Mode, Boost Ringer, Fast Ringer, Low Power, Ring Detect, MWI Mode and etc can also be configured here as the figure shown below.

Tone Region: United States/North America

Advanced Settings

Opermode: USA

PCMA Override: PCMU

FXS Honor Mode: Apply Opermode to FXS Modules Only

Boost Ringer: Normal

Fast Ringer: Normal

Low Power: Normal

Ring Detect: Standard

MWI Mode: None

Figure 28: Hardware Configuration: Advanced Settings

INTERNAL OPTIONS/STUN MONITOR

STUN Server	<p>Configures the STUN server to query.</p> <p>Valid format: [(hostname IP-address) [:' port]</p> <p>The default port number is 3478 if not specified.</p> <p>Leave this field blank to disable STUN.</p>
STUN Refresh	<p>Number of seconds between STUN Refreshes. The default setting is 30 seconds.</p>

INTERNAL OPTIONS/IAX GENERAL

Bind Port	<p>Allows iax2 to listen to another port. The default setting is 4569.</p>
Bind Address	<p>Forces iax2 to bind to a specific address instead of all addresses. The default setting is 0.0.0.0.</p>
IAX1 Compatibility	<p>Enables/disables iax1 style compatibility.</p>
No Checksums	<p>Enables/disables checksums.</p>
Delay Reject	<p>Enables/disables iax2 to delay reject of calls to avoid DOS.</p>
ADSI	<p>Enables/disables ADSI phone compatibility.</p>
Music On Hold Interpret	<p>Specifies Music On Hold class.</p>

Music On Hold Suggest	Suggests Music On Hold for the channel.
Language	Configures default language for the channel. This can be used by prompts.
Bandwidth	Configures allowed codecs for different bandwidth requirement. The default setting is "Low".

INTERNAL OPTIONS/CODECS

The following codes are supported in GXE5102/5104/5108/5116.

- PCMU
- PCMA
- GSM
- ILBC
- G.722
- G.726
- ADPCM
- LPC10
- G.729
- G.723
- H.263
- H.263p
- H.264

INTERNAL OPTIONS/IAX JITTER BUFFER

Enable Jitter Buffer	Enables the use of jitter buffer on the receiving side of a SIP channel.
Force Jitter Buffer	Forces the use of jitter buffer on the receiving side of a SIP channel.
Drop Count	Configures drop count.
MAX Jitter Buffer	Configures the maximum time (in milliseconds) 0 for the buffer.
MAX Interpolation Frames	Configures the maximum number of interpolated frames the jitter buffer should return consecutively.
Recync Threshold	Jumps in the frame timestamps over where the jitter buffer is resynchronized. This feature is useful to improve the quality of voice with big jumps in/broken timestamps sent from exotic devices and programs. The default setting is 1000.
Max Excess Buffer	Configures the maximum number (in milliseconds) to pad the jitter buffer.
Min Excess Buffer	Configures the minimum number (in milliseconds) to pad the jitter buffer.

Jitter Shrink Rate	Configures the jitter shrink rate.
--------------------	------------------------------------

INTERNAL OPTIONS/IAX REGISTRATION

Min Reg Expire	Minimum duration (in seconds) of registrations/subscriptions. The default setting is 60.
Max Reg Expire	Maximum duration (in seconds) of incoming registration/subscriptions. The default setting is 3600.
IAX Thread Count	Configures number of IAX threads.
IAX Max Thread Count	Configures maximum number of IAX threads.
Auto Kill	When set to "yes", the connection will be terminated if ACK for the NEW message is not received in 2000ms. Users could also specify number (in milliseconds) in addition to "yes" and "no".
Authentication Debugging	Enables/disables IAX related debug output in log messages.
Codec Priority	Configures codec negotiation priority to Caller, Host, Disabled or Reonly.
Type of Service	Configures ToS bit for preferred IP routing.
Trunk Frequency	Configures frequency of trunk frames measured in milliseconds.
Trunk Time Stamps	Enables/disables attaching time stamps to trunk frames.

INTERNAL OPTIONS/IAX SECURITY

Call Token Optional	A single IP address or a range of IP addresses for which call token validation is not required in the form 11.11.11.11 or 11.11.11.11/22.22.22.22.
Max Call Numbers	Limits the amount of call numbers allowed for a single IP address.
Max Nonvalidated Call Numbers	Limits the amount of nonvalidated call numbers for all IP addresses combined.
Call Number Limits	Limits the call numbers for a given IP range.

SIP SETTINGS/GENERAL

Realm For Digest Authentication	Realms MUST be globally unique according to RFC 3261. Configure this value as your host or domain name. The default setting is \"asterisk\". If a system name is configured in asterisk.conf, this value will be set to the configured system name.
UDP Port to Bind to	The default setting is 5060.

IP Address to Bind to	The default setting is 0.0.0.0, which means binding to all addresses.
Domain	Use comma to separate a list of domains that the GXE will be responsible for.
Allow Guest Calls	Enables/disables guest calls.
Overlap Dialing Support	Enables/disables dialing support.
Allow Transfer	Enables/disables all transfers (unless enabled in peers or users) initiated by the endpoint. The Dial() options 't' and 'T' are not related to whether SIP transfers are allowed or not.
Enable DNS SRV Lookups (on outbound calls)	Enables/disables DNS SRV lookups on calls.
MWI From	When sending MWI NOTIFY requests, this value will be used in the "From:" header as the "name\" part. If no "fromuser" is configured, the "user\" part of the URI in the "From:" header will be filled with this value as well.
From Domain	Configures the domain in the "From:" field of the SIP header. It may be required by some providers for authentication.
Auto Domain	When turned on, the GXE will add local host name and local IP to domain list.
Allow External Domains	Allow requests for domains that are not served by the GXE.

SIP SETTINGS/SIP JITTER BUFFER

Enable Jitter Buffer	Enables/disables the use of jitter buffer on the receiving side of a SIP channel.
Force Jitter Buffer	Forces the use of jitter buffer on the receiving side of a SIP channel.
Log Frames	Enable/disables jitter buffer frame logging.
Max Jitter Buffer	Configures max length of the jitter buffer in milliseconds.
Resync Threshold	Jumps in the frame timestamps over where the jitter buffer is resynchronized. This feature is useful to improve the quality of voice with big jumps in/broken timestamps sent from exotic devices and programs. The default setting is 1000.
Implementation	The Jitter buffer implementation used on the receiving side of a SIP channel. Users could select "Fixed" (with size always equals to jbmmaxsize) or "Adaptive" (with variable size which is the new jb of IAX2).

SIP SETTINGS/SIP MISCELLANEOUS

Register	Register as a SIP user agent to a SIP proxy (provider).
Register Timeout	The interval (in seconds) for the GXE to retry registration. The default setting is 20.
Register Attempts	Number of registration attempts before the GXE gives up. The default setting is 0 (keep trying until the server side accepts the registration request).
Video Max Bitrate (kb/s)	Maximum bitrate (kb/s) for video calls. The default setting is 384.
Support for SIP Video	Enables/disables SIP video support.
Generate Manager Events	Generates manager events when SIP UA performs events (e.g. hold).
Reject NonMatching Invites	When rejecting an incoming INVITE or REGISTER request, always reject with "401 Unauthorized" instead of notifying the requester that if there is a matching user or peer for the request.
NonStandard G.726 Support	If the peer negotiates G726-32 audio, use AAL2 packing order instead of RFC3551 packing order (this is required for Sipura and Grandstream ATAs).

SIP SETTINGS/SIP SESSION TIMER

Session Timers	<ul style="list-style-type: none"> • Originate: always request and run session-timers. • Accept: Run session-timers only when requested by other UA. • Refuse: Do not run session timers. <p>The default setting is "Accept".</p>
Session Expires	The maximum session refresh interval (in seconds). The default setting is 1800.
Min SE	The minimum session refresh interval (in seconds). The default setting is 90.
Session Refresher	Selects the session refresher to be UAC or UAS. The default setting is UAC.

SIP SETTINGS/SIP TLS AND TCP SETTINGS

TCP Enable	Enables/disables server for incoming TCP connections. The default setting is "No".
TCP Bindaddr	IP address for TCP server to bind to (0.0.0.0: binds to all interfaces). The default port number is 5060 if not specified.
TLS Enable	Enables/disables server for incoming TLS (secure) connections. The

	default setting is "No".
TLS Bindaddr	<p>IP address for TLS server to bind to (0.0.0.0: binds to all interfaces). The default port number is 5061 if not specified.</p> <p>Note: The IP address must match the common name (hostname) in the certificate. Please do not bind a TLS socket to multiple IP addresses. For details on how to construct a certificate for SIP, please refer to the following document: http://tools.ietf.org/html/draft-ietf-sip-domain-certs</p>
TLS Self Signed CA	<p>This is the CA certificate is the TLS server being connected to requires self signed certificate, including server's public key. This file will be renamed as "asterisk.ca" automatically.</p> <p>Note: The size of your ca file can't be larger than 2MB.</p>
TLS Cert	<p>This is the Certificate file (*.pem format only) used for TLS connections. This file will be renamed as "asterisk.pem" automatically.</p> <p>Note: The size of your certificate can't be larger than 2MB.</p>
TLS CA Cert	<p>This file must be named with the CA subject name hash value. It contains CA's (Certificate Authority) public key, which is used to verify the accessed servers.</p> <p>Note: The size of your certificate can't be larger than 2MB.</p>
TLS CA List	The list of files under the CA Cert directory.

SIP SETTINGS/SIP NAT

External Address	A static address (and port) that will be in outbound SIP messages if the GXE is behind NAT. If it's a hostname, it will only be looked up only.
External Host	Specifies an external host, which is similar to External Address except the hostname will be looked up every "External Refresh" interval and Asterisk will perform DNS queries periodically.
External Refresh	Configures the refresh interval for the external host.
External TCP Port	Configures the externally mapped TCP port when the GXE is behind a

	static NAT or PAT.
External TLS Port	Configures the externally mapped TLS port when GXE is behind a static NAT or PAT. The default value is 5061.
Local Network Address	A list of network addresses that are considered inside of the NAT network. Multiple entries are allowed, e.g., a reasonable set could be as follows: 192.168.0.0/255.255.0.0
NAT Mode	This is a global NAT setting that will affect all peers and users. <ul style="list-style-type: none"> • No: Use rport if the remote side requires it. • Force rport: Force rport to always be on. This is the default setting. • Yes: Force rport to always be on and perform comedia RTP handling. • Comedia: Use rport if the remote side requires it and perform comedia RTP handling. <p>Note: "comedia RTP handling" refers to the technique of sending RTP to the port where the other endpoint's RTP comes from. This can also be rephrased as "connection-oriented media".</p>
Allow RTP Reinvite	When turned on, the GXE will try to redirect the RTP media stream (audio) to go directly from the caller to the callee. <ul style="list-style-type: none"> • Yes: Enables RTP Reinvite. • NoNAT: Allows media path redirection (reinvite) but only when the peer is not behind NAT. The RTP core can determine if the peer is behind NAT or not based on the IP address where the media comes from. • Update: use UPDATE for media path redirection, instead of INVITE. <p>Note: Some devices do not support this (especially if one of them is behind NAT).</p>

SIP SETTINGS/SIP ToS

The following options are provided to configure SIP ToS on the GXE5102/5104/5108/5116.

SIP ToS

ToS for Signalling packets:	None ▾	ToS for RTP audio packets:	None ▾
ToS for RTP video packets:	None ▾		
Music On Hold Interpret:	default	Music On Hold Suggest:	
Language:		Enable Relaxed DTMF:	<input type="checkbox"/>
RTP TimeOut:		RTP HoldTimeOut:	
Trust Remote Party ID:	<input type="checkbox"/>	Send Remote Party ID:	<input type="checkbox"/>
Generate In-Band Ringing:	Never ▾	Server UserAgent:	
Allow Nonlocal Redirect:	<input type="checkbox"/>	Add 'user=phone' to URI:	<input type="checkbox"/>
DTMF Mode:	RFC2833 ▾	Send Compact SIP Headers:	<input type="checkbox"/>
Max Registration/Subscription Time:	3600	Min Registration/Subscription Time:	60
Default Incoming/Outgoing Registration Time:	120		
Min RoundtripTime (T1 Time):	100	Time between MWI Checks:	10

Figure 29: Hardware Configuration: Advanced Settings

SIP SETTINGS/DEBUG NOTIFY

Enable SIP Debugging	Enables/disables SIP debugging.
Record SIP History	Records SIP history.
Dump SIP History	Dumps SIP history at the end of SIP dialog.
Subscribe Context	Configures a specific context for SUBSCRIBE requests. This setting is useful to limit subscriptions to local extensions.
Allow Subscribe	Enables/disables support for subscriptions.
Notify on Ringing	Sends out NOTIFY on ringing status.

SETTINGS PAGE DEFINITIONS

SETTINGS/NETWORK SETTINGS

LAN	Set up IP method as Static, DHCP or PPPoE. <ul style="list-style-type: none"> Static IP: Enter IP Address, Gateway IP, Subnet Mask, Primary
-----	--

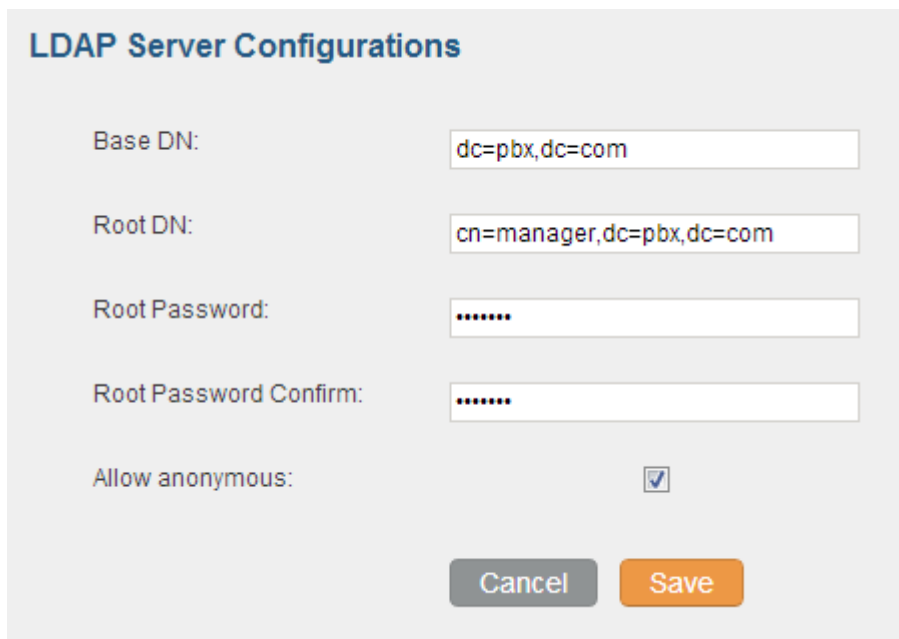
	<p>DNS, Secondary DNS.</p> <ul style="list-style-type: none"> • DHCP: Enter alternative DNS Server. • PPPoE: Enter User Name and Password.
802.1X	<p>To enable 802.1X, select 802.1X mode as "EAP-MD5", "EAP-TLS" or "EAP-PEAPv0/MSCHAPv2". Then enter the following information for the selected mode:</p> <ul style="list-style-type: none"> • Identity • MD5 Password • 802.1X CA Certificate • 802.1X Client Certificate

SETTINGS/CHANGE PASSWORD

To change the web access password, enter the old password and new password in this page. Once the web page comes back to the login interface again, enter the new password to login.

SETTINGS/LDAP SERVER

The GXE5102/5104/5108/5116 allows LDAP clients to connect to the LDAP Server in the GXE. The following options need to be configured first in the GXE.



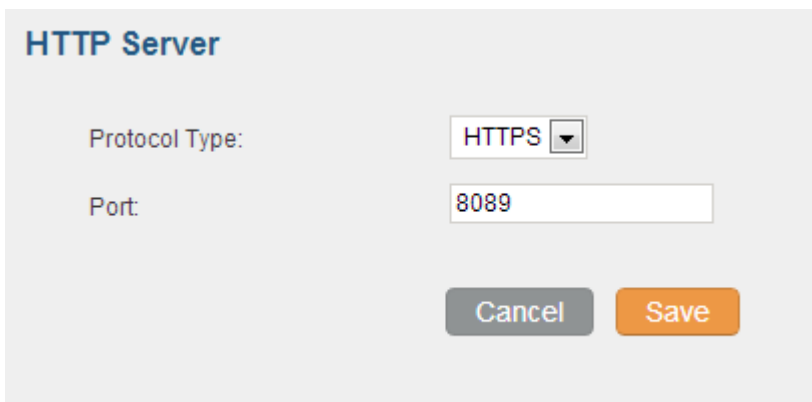
The screenshot shows a web form titled "LDAP Server Configurations". It contains the following fields and controls:

- Base DN:** Text input field containing "dc=pbx,dc=com".
- Root DN:** Text input field containing "cn=manager,dc=pbx,dc=com".
- Root Password:** Password input field with masked characters (dots).
- Root Password Confirm:** Password input field with masked characters (dots).
- Allow anonymous:** A checkbox that is checked.
- Buttons:** "Cancel" (grey) and "Save" (orange) buttons at the bottom.

Figure 30: LDAP Server Configurations

SETTINGS/HTTP SERVER

The GXE5102/5104/5108/5116 embedded Web server responds to HTTP/HTTPS GET/POST requests. In this page, users could configure the HTTP server protocol type (HTTP or HTTPS) as well as the port number.

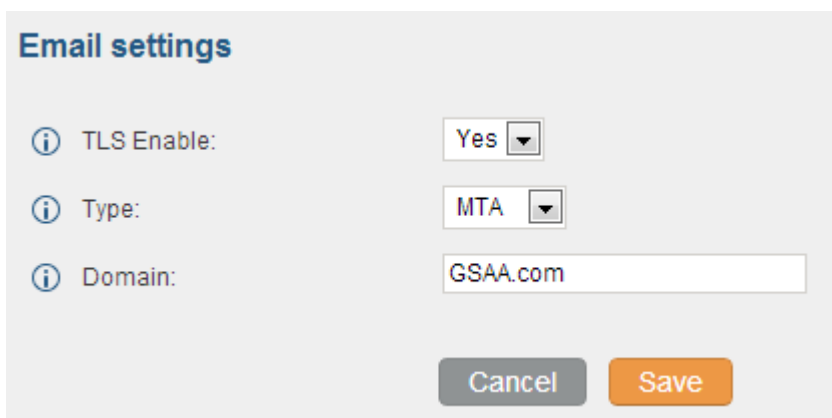


The screenshot shows a configuration window titled "HTTP Server". It contains two fields: "Protocol Type" with a dropdown menu set to "HTTPS", and "Port" with a text input field containing "8089". At the bottom, there are two buttons: "Cancel" and "Save".

Figure 31: HTTP Server Configurations

SETTINGS/EMAIL SETTINGS

The Email settings in this page configures transport protocol, type (MTA or Client) and Domain for the emails sent for FAX, Voicemail in the GXE5102/5104/5108/5116. embedded Web server responds to HTTP/HTTPS GET/POST requests. In this page, users could configure the HTTP server protocol type (HTTP or HTTPS) as well as the port number.



The screenshot shows a configuration window titled "Email settings". It contains three fields: "TLS Enable" with a dropdown menu set to "Yes", "Type" with a dropdown menu set to "MTA", and "Domain" with a text input field containing "GSAA.com". At the bottom, there are two buttons: "Cancel" and "Save".

Figure 32: Email Settings

SETTINGS/TIME SETTINGS

NTP Server	Defines the URL or IP address of the NTP server. The GXE may obtain the date and time from the server.
Enable DHCP Option 2	Allows device to get provisioned for Time Zone from DHCP Option 2 in

	the local server automatically. The default setting is "Yes".
Enable DHCP Option 42	Defines whether DHCP Option 42 should override NTP server or not. When enabled, DHCP Option 42 will override the NTP server if it's set up on the LAN. The default setting is "Yes".
Time Zone	Controls the date/time display according to the specified time zone.
Self-Defined Time Zone	<p>This parameter allows the users to define their own time zone. The syntax is: std offset dst [offset], start [/time], end [/time] Default is set to: MTZ+6MDT+5,M4.1.0,M11.1.0</p> <p>MTZ+6MDT+5 This indicates a time zone with 6 hours offset with 1 hour ahead which is U.S central time. If it is positive (+) if the local time zone is west of the Prime Meridian (A.K.A: International or Greenwich Meridian) and negative (-) if it is east.</p> <p>M4.1.0,M11.1.0 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 First Sunday of April to the 1st Sunday of November.</p>

MAINTENANCE PAGE DEFINATIONS

SYSLOG

In this page, users could configure syslog server with different levels. Select the modules you would like to send syslog to the server for different syslog levels.

Syslog Configuration

Syslog Server:

PBX Modules

all level	module	error	warn	notic	verb	debug
	all modules	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	ami	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	app_adsiprogram	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	app_alarmreceiver	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	app_amd	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	app_authenticate	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	app_controlplayback	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	app_dictate	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	app_directed_pickup	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	app_directory	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	app_disa	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	app_echo	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	app_exec	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

Figure 33: Syslog Settings

UPGRADE

The GXE5102/5104/5108/5116 can be upgraded via network or local uploading.

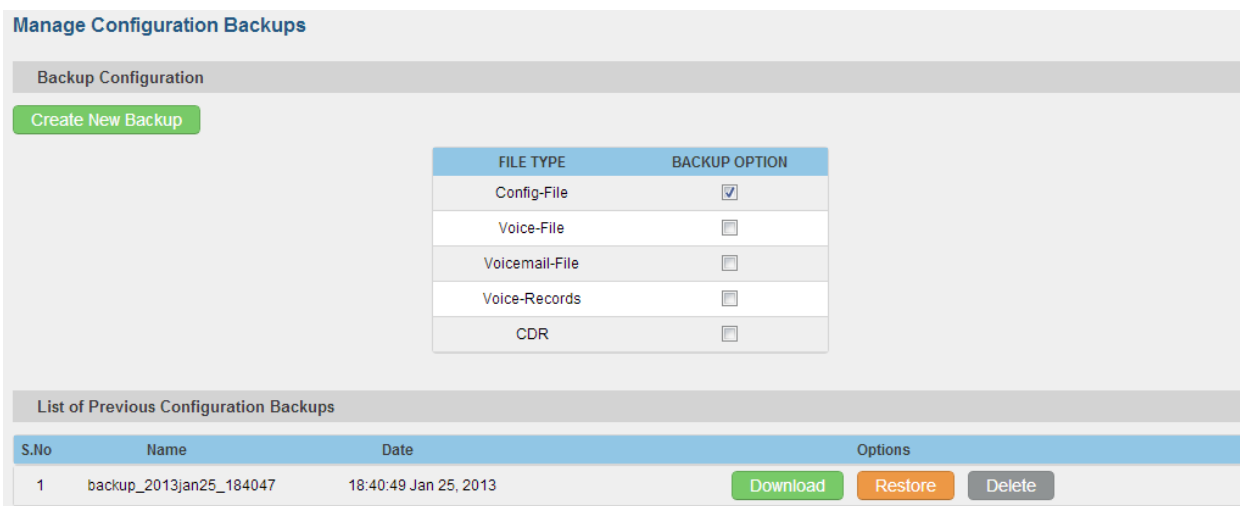
Upgrade Via	Allows users to choose the firmware upgrade method: TFTP, HTTP or HTTPS.
Firmware Server Path	Defines the server path for the firmware server.
Firmware File Prefix	If configured, only the firmware with the matching encrypted prefix will be downloaded and flashed into the GXE.
Firmware File Suffix	If configured, only the firmware with the matching encrypted postfix will be downloaded and flashed into the GXE.
HTTP/HTTPS User Name	The user name for the HTTP/HTTPS server.
HTTP/HTTPS Password	The password for the HTTP/HTTPS server.

MAINTENANCE/BACKUP

Users could backup the configurations on the GXE for restore purpose. Before creating new backup file, select the backup option first.

- If the Config-File is selected only, the backup file will be saved in the flash of the device.
- If Voice-File, Voicemail-File, Voice-Records or CDR is selected, external storage devices (USB Flash drive or SD Card) will be required because the backup file might be too large.

Once backup is done, the list of the backups will be displayed with date and time. Users then can download, restore or delete it from the GXE or the external device.



Manage Configuration Backups

Backup Configuration

Create New Backup

FILE TYPE	BACKUP OPTION
Config-File	<input checked="" type="checkbox"/>
Voice-File	<input type="checkbox"/>
Voicemail-File	<input type="checkbox"/>
Voice-Records	<input type="checkbox"/>
CDR	<input type="checkbox"/>

List of Previous Configuration Backups

S.No	Name	Date	Options
1	backup_2013jan25_184047	18:40:49 Jan 25, 2013	Download Restore Delete

Figure 34: Backup

MAINTENANCE/RESET AND REBOOT

To factory reset the device, select the mode type first. There are three different types for reset.

- User Configuration: All the Extensions, Trunks and Routing configurations, as well as the local settings (network settings, upgrading setting and etc) will be cleared.
- User Data: All the data including voicemail, recordings, IVR Prompt, Music on Hold, CDR and backup files will be cleared.
- All: All the configurations and data will be reset to factory default.

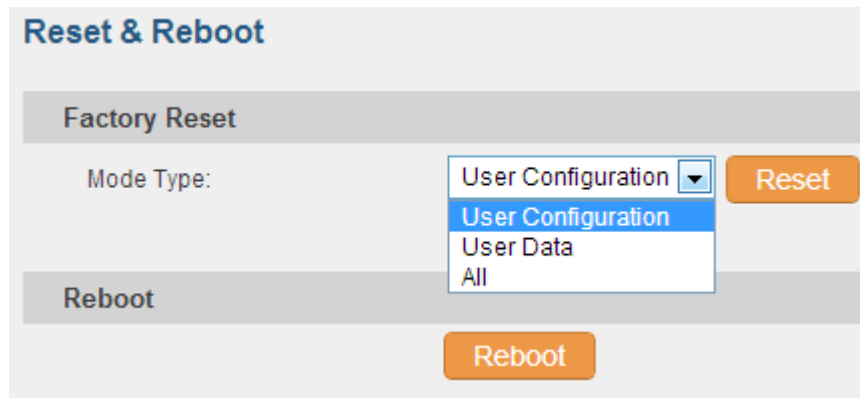


Figure 35: Reset and Reboot

MAINTENANCE/TROUBLESHOOTING

On the GXE, users could capture traces, ping remote host and traceroute remote host for troubleshooting purpose. The captured trace can be downloaded for analysis. Also the instructions or result will be displayed in the web GUI output result.

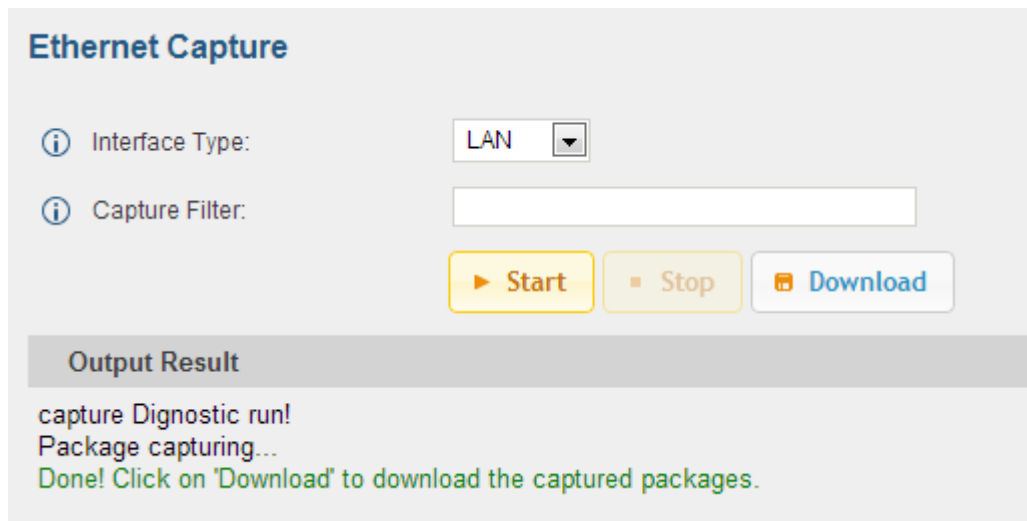


Figure 36: Ethernet Capture

UPGRADING GXE5102/5104/5108/5116

UPGRADE FROM NETWORK

The GXE5102/5104/5108/5116 can be upgraded via TFTP/HTTP/HTTPS by configuring the URL/IP Address for the TFTP/HTTP/HTTPS server and selecting a download method. Configure a valid URL for TFTP, HTTP or HTTPS; the server name can be FQDN or IP address.

Examples of valid URLs:

firmware.grandstream.com

UPLOAD FIRMWARE LOCALLING

If there is no HTTP/TFTP server, users could also upload the firmware to the GXE5102/5104/5108/5116 directly via Web GUI. Please follow the steps below to upload firmware locally.



- Download the latest GXE5102/5104/5108/5116 firmware file from the following link and save it in your PC;
<http://www.grandstream.com/support/firmware>
- Log in the Web GUI as administrator in the PC;
- Go to Web GUI->**Maintenance**->**Upgrade**, Upload the firmware file by clicking on  and select the firmware file from your PC;
- Click on  to start upgrading;
- Wait until the upgrading process is done and the GXE boots up again.



Figure 37: Local Upgrade

 **Note:**

Please do not interrupt or power cycle the GXE5102/5104/5108/5116 when the upgrading process is on.

NO LOCAL FIRMWARE SERVERS

For users that would like to use remote upgrading without a local TFTP server, Grandstream offers a NAT-friendly HTTP server. This enables users to download the latest software upgrades for their devices via this server. Please refer to the webpage:

<http://www.grandstream.com/support/firmware>.

Alternatively, users can download a free TFTP or HTTP server and conduct a local firmware upgrade. A free windows version TFTP server is available for download from :

<http://support.solarwinds.net/updates/New-customerFree.cfm>

<http://tftpd32.jounin.net/>.

Instructions for local firmware upgrade via TFTP:

1. Unzip the firmware files and put all of them in the root directory of the TFTP server;
2. Connect the PC running the TFTP server and the GXE5102/5104/5108/5116 device to the same LAN segment;
3. Launch the TFTP server and go to the File menu->Configure->Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade;
4. Start the TFTP server and configure the TFTP server in the GXE's web configuration interface;
5. Configure the Firmware Server Path to the IP address of the PC;
6. Update the changes and reboot the GXE5102/5104/5108/5116.

End users can also choose to download a free HTTP server from <http://httpd.apache.org/> or use Microsoft IIS web server.

EXPERIENCING THE GXE5102/5104/5108/5116

Please visit our website: <http://www.grandstream.com> to receive the most up- to-date updates on firmware releases, additional features, FAQs, documentation and news on new products.

We encourage you to browse our [product related documentation](#), [FAQs](#) and [User and Developer Forum](#) for answers to your general questions. If you have purchased our products through a Grandstream Certified Partner or Reseller, please contact them directly for immediate support.

Our technical support staff is trained and ready to answer all of your questions. Contact a technical support member or [submit a trouble ticket online](#) to receive in-depth support.

Thank you again for purchasing Grandstream GXE5102/5104/5108/5116, it will be sure to bring convenience and color to both your business and personal life.

FCC Warning

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions:

(1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

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

FCC 15.105 Class B

(b) For a Class B digital device or peripheral, the instructions furnished the user shall include the following or similar statement, placed in a prominent location in the text of the manual:

Note: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a

particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

—Reorient or relocate the receiving antenna.

—Increase the separation between the equipment and receiver.

—Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.

—Consult the dealer or an experienced radio/TV technician for help.