



# Grandstream Networks, Inc.

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Analog IP Gateway GXW410x  
4 or 8 FXO Ports  
User Manual



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## GUI INTERFACES

### GXW410X USER MANUAL

[http://www.grandstream.com/products/gxw\\_series/gxw410x/documents/gxw410x\\_gui.zip](http://www.grandstream.com/products/gxw_series/gxw410x/documents/gxw410x_gui.zip)

1. SCREENSHOT OF ADVANCED SETTINGS CONFIGURATION PAGE
2. SCREENSHOT OF BASIC SETTINGS CONFIGURATION PAGE
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4. SCREENSHOT OF FXO LINES CONFIGURATION PAGE
5. SCREENSHOT OF PROFILE 1 CONFIGURATION PAGE
6. SCREENSHOT OF STATUS CONFIGURATION PAGE

## WELCOME

Thank you for purchasing the Grandstream GXW410x IP Analog FXO Gateway. The GXW410x is a cost effective, easy to use and easy to configure IP communications solution for any business. The GXW410x supports popular voice codecs and is designed for full SIP compatibility and interoperability with 3<sup>rd</sup> party SIP providers, thus enabling you to fully leverage the benefits of VoIP technology, integrate a traditional phone system into a VoIP network, and efficiently manage communication costs.

This manual will help you learn how to operate and manage your GXW FXO Analog IP Gateway and make the best use of its many upgraded features including simple and quick installation, multi-party conferencing, etc. This IP Analog Gateway is very easy to manage and scalable, specifically designed to be an easy to use and affordable VoIP solution for the small – medium business or enterprise. Enable the video surveillance port to give piece of mind while you are away from your business.

### GATEWAY GXW410X OVERVIEW

The GXW410x offers an easy to manage, feature rich feature IP communications solution for any small business or businesses with virtual and/or branch locations who want to leverage their broadband network and/or add new IP Technology to their current phone system. The Grandstream Enterprise Analog VoIP Gateway GXW410x series converts SIP/RTP IP calls to traditional PSTN calls and vice versa. There are two models - the GXW4104 and GXW4108, which have either 4 or 8 FXO ports respectively. The installation is the same for either model.

### SAFETY COMPLIANCES

The GXW410x is compliant with various safety standards including FCC/CE. Its power adaptor is compliant with UL standard. **Warning:** use only the power adapter included in the GXW410x package. Using an alternative power adapter may permanently damage the unit.

**Caution:** GXW410x is designed and recommended for indoor use only to avoid possible damage caused by over-voltage or over current situations. Not respecting this recommendation may cause a system lock which will require user to perform a power cycle of the unit.

### WARRANTY

Grandstream has a reseller agreement with our reseller customer. End users should contact the company from whom you purchased the product for replacement, repair or refund.

If you purchased the product directly from Grandstream, contact your Grandstream Sales and Service Representative for a RMA (Return Materials Authorization) number. Grandstream reserves the right to remedy warranty policy without prior notification.

**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.

- This document is contains links to Grandstream GUI Interfaces. Please download the GUI examples [http://www.grandstream.com/products/gxw\\_series/gxw410x/documents/gxw410x\\_gui.zip](http://www.grandstream.com/products/gxw_series/gxw410x/documents/gxw410x_gui.zip) for your reference.
- This document is subject to change without notice. The latest electronic version of this user manual is available on our website:  
[http://www.grandstream.com/products/gxw\\_series/gxw410x/documents/gxw410x\\_usermanual\\_english.pdf](http://www.grandstream.com/products/gxw_series/gxw410x/documents/gxw410x_usermanual_english.pdf)

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## PACKAGING

Unpack and check all accessories. Equipment included in the package:

- 1) One GXW410x Unit
- 2) One universal power adaptor
- 3) One Ethernet cable

## CONNECTING THE GXW410X

FIGURE 1: DIAGRAM OF GXW410X BACK PANEL

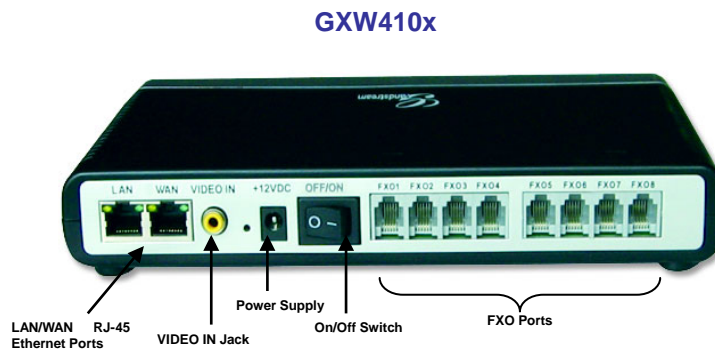


TABLE 1: DEFINITIONS OF THE GXW CONNECTORS

<b>LAN (OR PC)</b>	CONNECT YOUR PC TO THIS PORT. IT WILL THEN BE ASSIGNED AN IP ADDRESS FROM YOUR ROUTER/DHCP SERVER. THE GXW410X ACTS AS A SWITCH ONLY.
<b>WAN</b>	CONNECT TO THE INTERNAL LAN NETWORK OR PUBLIC INTERNET.
<b>VIDEO IN</b>	Connection for Analog based Video Surveillance Camera (RCA) <b><u>AVAILABLE IN HW REVISION 1 ONLY</u></b>
<b>RESET</b>	Factory Reset button. Press for 7 seconds to reset factory default settings.
<b>POWER SUPPLY</b>	Power adapter connection
<b>OFF/ON</b>	Off/On switch
<b>FX01 - FX08</b>	FXO ports to be connected to physical PSTN lines from a traditional PSTN PBX or PSTN Central Office.

**NOTE:** GXW410x acts as bridge only, if a device is connected to the LAN port, this device will get an ip in same subnet as the WAN IP (NAT is disabled).

FIGURE 2: DIAGRAM OF GXW410X DISPLAY PANEL



TABLE 2: DEFINITIONS OF THE GXW DISPLAY PANEL

<b>Power LED</b>	Indicates Power. Remains ON when Power is connected and unit is turned ON.
<b>Ready LED</b>	Remains ON after boot-up.
<b>LAN LED</b>	Indicates WAN port activity in the back side
<b>PC LED</b>	Indicates LAN port activity in the back side
<b>Video LED</b>	Remains solid green on boot-up. If Video IN terminal is connected, indicates video activity. <b>Available in HW revision 1 Only</b>
<b>LEDs 1 - 8</b>	Indicate status of the respective FXO Ports on the back panel <b>Busy - ON</b> <b>Available – OFF</b>

**NOTE:** All LEDs display green when ON. The **Ready** light will only be ON when the network interface is ready and the Web User Interface is accessible.

During a *firmware upgrade* or *configuration download* the following LED pattern will be observed:

**Power, Ready, Video** and **WAN** LEDs will be **ON**. The **FXO port** LED will keep flashing during download and then stay OFF while the new files are written. The entire process may take between 20 to 30 minutes. The firmware upgrade is complete when you can login into the web configuration pages.

## APPLICATION DESCRIPTION

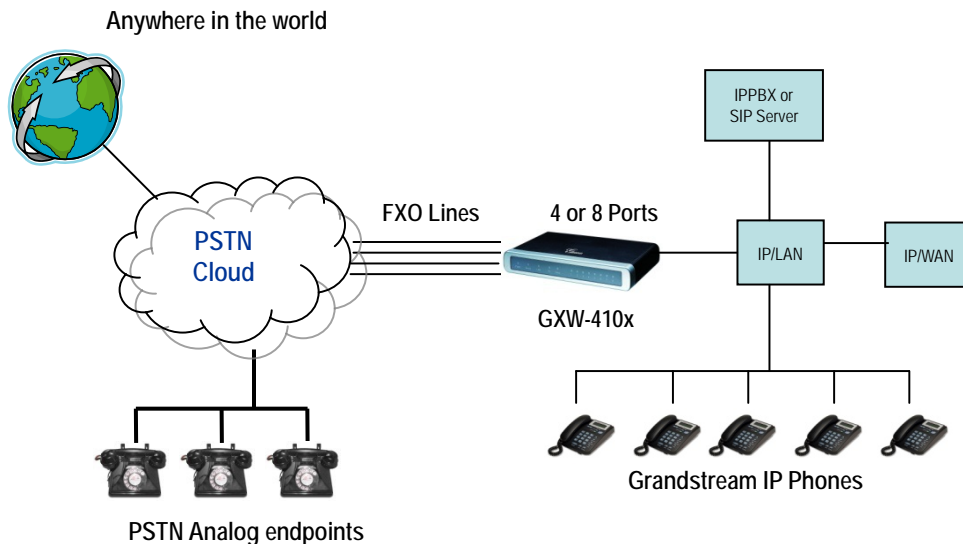
### IP PBX / SIP SERVER WITH GXW410X

A SIP proxy server such as Asterisk or a SIP registrar server can be deployed with the GXW410x series. In this environment, the SIP server handles SIP registration and call control and the GXW410x processes media conversion between IP and PSTN calls.

There are 2 ways to configure GXW410x when using with a SIP Server:

1. **With SIP accounts configured on Channels page.** In this case, the GXW acts like an endpoint requesting registration from the SIP Server. Under the Channels webpage you will need to fill in the information like SIP User ID, Password, etc. Now, when you try to make calls from IP, the call will be routed to the SIP Server which will forward it to one of the SIP accounts on the GXW410x, which will then forward it to the PSTN line.
2. **Without SIP accounts.** In this case, you simply have to configure the SIP Server to perform forwarding of the SIP INVITE message with the FXO destination number to the gateway IP Address. The GXW410x will receive the digits and immediately forward them on the FXO lines to the destination PSTN. Most of the configuration on the Gateway for this case will remain default, except Stage Method needs to be set to 1, and SIP Server IP Address/DNS name has to be filled.

**FIGURE 3: FUNCTIONAL DIAGRAM OF IP-PBX & GXW410X**



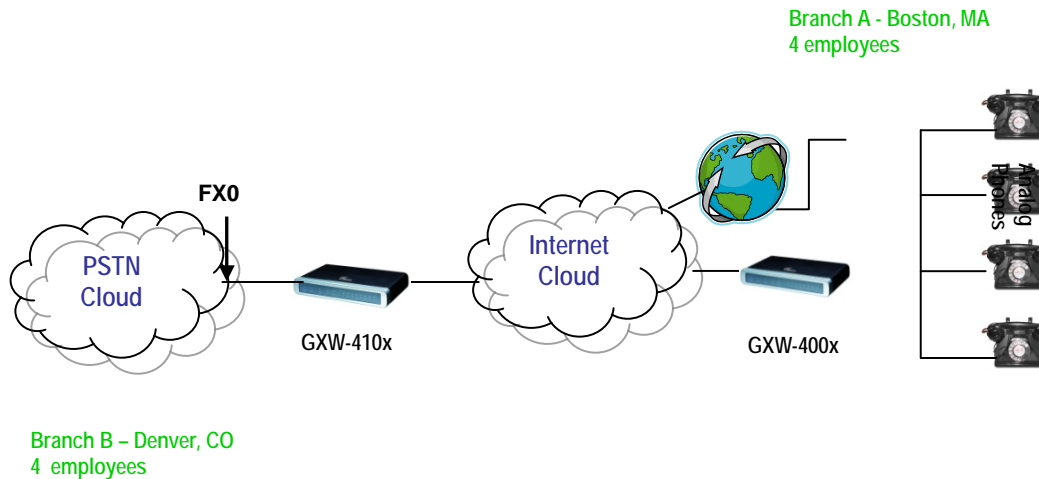
For incoming calls from the PSTN analog endpoints to the GXW410x, the device will auto forward each call to a configured IP extension. The SIP Server can then route the call based on its own configuration or IVR system.



## FXS GATEWAY WITH GXW410X [NO SIP SERVER REQUIRED]

Alternatively, the GXW410x can be used without a SIP Server. You can use it in conjunction with a FXS Gateway (Ex. GXW400x) and still be able to originate and terminate calls from IP to PSTN and vice versa. All you need to make sure is that the 2 gateways are able to locate each other (they should be on the same LAN or on Public IP addresses).

**FIGURE 4: GXW400X & GXW410X SCENARIO/TOLL- FREE CALLING BETWEEN LOCATIONS**



In this diagram, configure the SIP Server field to be the IP Address of the other gateway (i.e. configure IP address of FXS gateway to be SIP Server of GXW410x and vice versa). Please be sure you set SIP Registration to No.

**EXPECTED CALL FLOW:** Analog Phone (GXW400x) picks up and dials destination PSTN number. The call gets routed to the GXW410x which dials out the digit string onto the FXO Lines, thus reaching the destination PSTN endpoint. On the reverse, incoming calls from PSTN endpoints will be routed automatically to the FXS Gateway through the GXW410x.

**TABLE 3: FXS AND FXO GATEWAY CONFIGURATION EXAMPLE**

<b>GXW400X GATEWAY</b>	<b>GXW410X GATEWAY</b>
<b>Profile 1</b> SIP Server - Set it to IP Address of GXW410x SIP Registration - No Outgoing Call without Registration - Yes NAT traversal - No	<b>Advanced Settings</b> STUN Server - Blank Use Random Port - No
<b>Advanced Settings</b> STUN Server - Blank	<b>FXO lines</b> Wait for Dial Tone - Y or N (whichever works for your PSTN Service Provider) Stage Method - 1 Unconditional Call Forward to VOIP: ch1-8:444; @ch1-8:p1; ch1-8:5060++
	<b>Channels</b>

	1-8 5060 Profile 1 Local SIP Listen port (For VOIP to PSTN calls) - 5060++
	<b>Profile 1</b> SIP Server - Set it to IP Address of GXW400x SIP Registration - No NAT traversal - No

## FEATURES

GXW410x is a next generation IP voice and video gateway that features full interoperability with leading IP-PBXs, SoftSwitches and SIP platforms. The Gateway series offers superb voice and video quality, traditional telephony functionality, simple configuration, feature rich functionality and an additional video port that enables the gateway to act like a video surveillance gateway.

### SOFTWARE FEATURES OVERVIEW

- 4 and 8 FXO port media gateways
- Video surveillance port (**Available in HW revision 1 only**)
- External power supply
- Two RJ-45 ports (switched or routed)
- TFTP and HTTP firmware upgrade support
- Multiple SIP accounts, multiple SIP profiles (choice of 3 profiles per account)
- Supports Audio Codecs: G711U/A, G723, G729A/B and GSM
- Supports Video Codecs: H.264
- G.168 – echo cancellation
- Flexible DTMF transmission: In Audio, RFC2833, SIP Info or any combination of the 3
- Selectable, multiple LBR coders per channel
- T.38 compliant

### HARDWARE SPECIFICATION

**TABLE 4: HARDWARE SPECIFICATIONS OF GXW410X**

<b>LAN interface</b>	2xRJ45 10/100Mbps
<b>LED</b>	8 LEDs (GREEN)
<b>Universal Switching Power Adaptor</b>	<b>Input:</b> 100-240V AC, 50/60Hz, 0.5A Max <b>Output:</b> 12V DC, 1.25A <b>UL certified</b>
<b>Dimension</b>	225mm (L) x 172mm (W) x 42mm (H)
<b>Weight</b>	0.29 lbs (3.5 oz)
<b>Temperature</b>	32~104°F 0~40°C
<b>Humidity</b>	10% - 90% (non-condensing)
<b>Compliance</b>	FCC, CE

**TABLE 5: GXW410X SOFTWARE FEATURES**

	<b>GXW410x FXO Analog Gateway Series</b>
<b>IP settings</b>	GXW4104: 4 ports; 4 SIP accounts w/ choice of 3 SIP Server profiles GXW4108: 8 ports; 8 SIP accounts w/ choice of 3 SIP Server profiles Round-robin port scheduling to ensure available lines to access PSTN networks
<b>Telephone Interface</b>	FXO, RJ11


<b>Network Interface</b>	Two (2) 10/100 Mbps, RJ45
<b>LED Indicators</b>	Power, Video, and Line LEDs
<b>On/Off Switch</b>	Yes
<b>Voice over Packet Capabilities</b>	G.168 compliant Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711
<b>Voice Compression</b>	G.711U, G711A, G.723, G.729A/B, GSM
<b>Video Surveillance</b>	Real-time H.264 base CIF resolution
<b>DHCP Server/Client</b>	Switch Mode and PPPoE
<b>Fax over IP</b>	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through
<b>QoS</b>	Diffserv, TOS, 802.1 P/Q VLAN tagging
<b>IP Transport</b>	RTP/RTCP and RTSP
<b>PSTN Signaling</b>	FXO Loop start, Current Disconnect.
<b>DTMF Method</b>	Flexible DTMF transmission method, User interface of In-audio, RFC2833, and SIP Info
<b>IP Signaling</b>	SIP (RFC 3261)
<b>Provisioning</b>	TFTP and HTTP
<b>Media</b>	SRTP
<b>Control</b>	TLS and SIPS (pending)
<b>Management</b>	Syslog support, HTTPS and telnet (pending), remote management using Web browser
<b>Caller ID</b>	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID
<b>Polarity Reversal / Wink</b>	Yes (Detection only). The PSTN lines will need to be subscribed to PR service from the Service Provider.
<b>EMC</b>	GXW410x: EN55022 Class B, CFR Part 15 Class B, EN55024; GXW4104: FCC, CE (in addition)
<b>Safety</b>	GXW410x: EN60950-1 GXW4108: UL60950-1 (in addition)

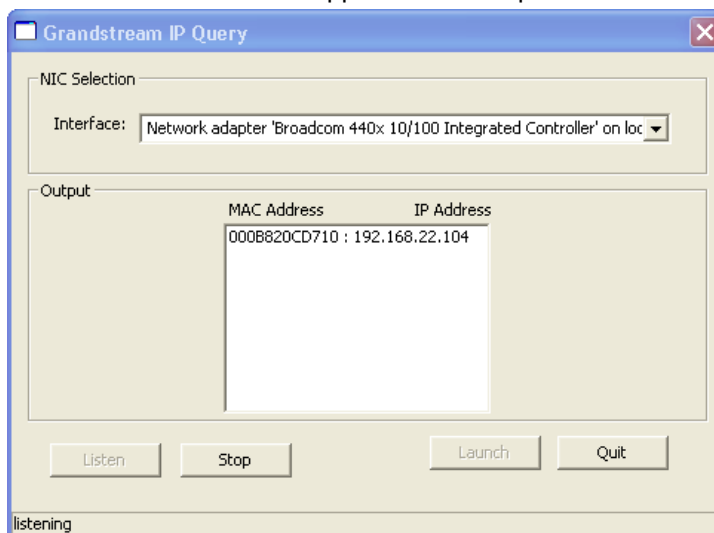
## CONFIGURATION GUIDE

### CONFIGURATION WITH WEB BROWSER

The GXW410x has an embedded Web server that will allow a user to configure the IP phone through any common web browser.

#### ACCESSING THE WEB CONFIGURATION MENU

1. Navigate your browser to:  
[http://www.grandstream.com/sites/default/files/Resources/IPQuery\\_0.zip](http://www.grandstream.com/sites/default/files/Resources/IPQuery_0.zip)
2. Run the Grandstream IPQuery tool that you just downloaded.
3. Click on  button in order to begin device detection
4. The detected devices will appear in the Output field



#### END USER CONFIGURATION

Once this HTTP request is entered and sent from a Web browser, the GXW410x will respond with a login screen. There are two default passwords for the login page:

User Level:	Password:	Webpages allowed:
End User Level	123	Only Status and Basic Settings
Administrator Level	admin	All pages can be browsed.

After login, the next configuration page is the **Basic Configuration** page, explained in detail in **Table 6: Maintenance**.

**TABLE 6: MAINTENANCE DEFINITIONS**

<b>MAINTENANCE</b>	
<b>Web/Telnet access</b>	
<b>Web Access</b>	Select HTTP or secure HTTPS protocol for Web Access
<b>Web Port</b>	This option defines Web Port desired to use. This field is optional. Default is 80 for HTTP and 443 for HTTPS.
<b>End-User Password</b>	This contains the password to access the End-user Web Configuration Menu (Status and Basic Settings). This field is case sensitive with a maximum length of 25 characters.
<b>Admin Password</b>	Contains the password to access administrative settings other than Basic Settings and Status Page.
<b>Upgrade/Provisioning</b>	
<b>Firmware Upgrade &amp; Provisioning</b>	This radio button will enable GXW410x to download firmware or configuration file through either TFTP or HTTP.
<b>Via TFTP Server</b>	<p>If selected, the GXW410x will attempt to retrieve new configuration file or new code image from the specified TFTP server at boot time. It will make up to 5 attempts before timeout and then it will start the boot process using the existing code image in the Flash memory. If a TFTP server is configured and a new code image is retrieved, the new downloaded image will be verified and then saved into the Flash memory.</p> <p><b>Note:</b> Please do NOT interrupt the TFTP upgrade process (especially the power supply) as this will damage the device. Depending on the network environment this process can take up to 25 or 30 minutes.</p>
<b>Via HTTP Server</b>	<p>The URL for the HTTP server used for firmware upgrade and configuration via HTTP. For example, ttp://provisioning.mycompany.com:6688/Grandstream/1.0.0.54 Here “:6688” is the specific TCP port that the HTTP server is listening at, it can be omitted if using default port 80.</p> <p><b>Note:</b> If Auto Upgrade is set to No, GXW410x will only do HTTP download once at boot up.</p>
<b>Firmware Server Path</b>	IP address or domain name of firmware server.
<b>Config Server Path</b>	IP address or domain name of configuration server.
<b>Firmware File Prefix</b>	Default is blank. If configured, GXW410x will request firmware file with the prefix. This setting is useful for ITSPs. End user should keep it blank.
<b>Firmware File Postfix</b>	Default is blank. End user should keep it blank.
<b>Config File Prefix</b>	Default is blank. End user should keep it blank.
<b>Config File Postfix</b>	Default is blank. End user should keep it blank.
<b>Allow DHCP Option 66 to override server</b>	Default value is No. If set to Yes, configuration file will originate from the DHCP server.
<b>Automatic Upgrade</b>	<p>Choose <b>Yes</b> to enable automatic upgrade and provisioning. In “Check for new firmware every” field, enter the number of minutes to enable GXW410x to check the server for firmware upgrade or configuration. When set to <b>No</b>, GXW410x will only do upgrade once at boot up. Other options are:</p> <p>“ Always check for New Firmware.”</p> <p>“ Check New Firmware only when F/W pre/suffix changes”</p> <p>“ Always skip the Firmware check”</p>
<b>Syslog Setup</b>	
<b>Syslog Server</b>	The IP address or URL of System log server. This feature is especially useful for ITSP (Internet Telephone Service Provider)

<b>Syslog Level</b>	<p>Select the GXW to report the log level. Default is NONE. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events:</p> <ol style="list-style-type: none"> <li>1. product model/version on boot up (INFO level)</li> <li>2. NAT related info (INFO level)</li> <li>3. sent or received SIP message (DEBUG level)</li> <li>4. SIP message summary (INFO level)</li> <li>5. inbound and outbound calls (INFO level)</li> <li>6. registration status change (INFO level)</li> <li>7. negotiated codec (INFO level)</li> <li>8. Ethernet link up (INFO level)</li> <li>9. SLIC chip exception (WARNING and ERROR levels)</li> <li>10. memory exception (ERROR level)</li> </ol> <p>The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: GS_LOG: [device MAC address][error code] error message</p> <p><u>Example:</u> May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up</p>
<b>Security</b>	
<b>Download Configure</b>	Downloads the current configuration of the GXW410X

**TABLE 7: NETWORKS DEFINITIONS**

<b>NETWORKS</b>	
<b>Basic Settings</b>	
<b>IP Address</b>	<p>There are two modes to operate the GXW410x:</p> <p><b>DHCP mode:</b> all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The GXW410x acquires its IP address from the first DHCP server it discovers from the LAN it is connected.</p> <p><u>Using the PPPoE feature:</u> set the PPPoE account settings. The GXW410x will establish a PPPoE session if any of the PPPoE fields is set.</p> <p><b>Static IP mode:</b> configure the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields.</p>
<b>DHCP hostname</b>	This option specifies the name of the client. This field is optional but may be required by some Internet Service Providers. Default is blank.
<b>DHCP domain</b>	This option specifies the domain name that client should use when resolving hostnames via the Domain Name System. Default is blank.
<b>DHCP vendor class ID</b>	Used by clients and servers to exchange vendor-specific information. Default is Grandstream GXW410x.
<b>PPPoE account ID</b>	PPPoE username. Necessary if ISP requires you to use a PPPoE (Point to Point Protocol over Ethernet) connection.
<b>PPPoE password</b>	PPPoE account password.
<b>PPPoE Service Name</b>	This field is optional. If your ISP uses a service name for the PPPoE connection, enter the service name here. Default is blank.
<b>Preferred DNS server</b>	This field will let the user enter a preferred DNS server to be used instead of the one acquired by the service provider.
<b>Time Zone</b>	Controls how the date/time is displayed according to the specified time zone.
<b>Allow DHCP Option 2 to override Time Zone Settings</b>	Default is No. If set to Yes, time zone settings will originate from the DHCP server.
<b>Advanced Settings</b>	
<b>Layer 3 QoS</b>	This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv or MPLS.

	Default value is 48. Its range lies from 0 to 63.
<b>Layer 2 QoS</b>	<p>This contains the value used for layer 2 VLAN tag.        802.1q / VLAN tag: Default value is 0. Range lies from 0 to 4095.        802.1p Priority value: Default value is 0. Range lies from 0 to 7.</p> <p>*** The above 2 settings need to be supported on the network and then configured accordingly on the GXW410x. Incorrect configuration will cause blocked access, which will result in Factory Reset as the only option to renew access.</p>
<b>Video Surveillance (HW version 1 only)</b>	Default is No. Set to Yes, in order to enable the video in port. And configure the RTSP port number here(default port number 554).
<b>Date &amp; Time</b>	
<b>NTP server</b>	URI or IP address of the NTP (Network Time Protocol) server, which will be used by the phone to synchronize the date and time.
<b>Allow DHCP Option 42 to override an NTP server</b>	Default value is No. If set to Yes, the NTP server will originate from the DHCP server.
<b>Self-Defined Time Zone (Yes/No)</b>	<p>This parameter controls whether the displayed time will be daylight savings time or not. If set to "Yes" and the Optional Rule is empty, then the displayed time will be 1 hour ahead of normal time. The "Automatic Daylight Saving Time Rule" shall have the following syntax: start-time;end-time;saving</p> <p>Both start-time and end-time have the same syntax:        Month ,day ,weekday ,hour ,minute        month: 1,2,3,...,12 (for Jan, Feb, ..., Dec)        day: [+ -]1,2,3,...,31        weekday: 1, 2, 3, ..., 7 (for Mon, Tue, ..., Sun), or 0 which means the daylight saving rule is not based on week days but based on the day of the month.        hour: hour (0-23),        minute: minute (0-59)</p> <p>If "weekday" is 0, it means the date to start or end daylight saving is at exactly the given date. In that case, the "day" value must not be negative.</p> <p>If "weekday" is not zero and "day" is positive, then the daylight saving starts on the first "day"th iteration of the weekday (1st Sunday, 3rd Tuesday etc). If "weekday" is not zero and "day" is negative, then the daylight saving starts on the last "day"th iteration of the weekday (last Sunday, 3rd last Tuesday etc). The saving is in the unit of minutes. The saving time may also be preceded by a negative (-) sign if subtraction is desired instead of addition. The default value for "Automatic Daylight Saving Time Rule" shall be set to        "03,11,0,02,00;11,04,0,02,00;60" which is the rule for US.</p> <p><u>Examples:</u>        US where daylight saving time is applicable: 03,11,0,02,00;11,4,0,02,00;60        This means the daylight saving time starts from 11<sup>th</sup> March at 2AM and ends November 4th at 2AM. The saving is 60 minutes (1hour).</p>
<b>Optional Rule:</b>	

**TABLE 8: SETTINGS DEFINITIONS**

<b>SETTINGS</b>	
<b>General settings</b>	
<b>Use NAT IP</b>	NAT IP address used in SIP/SDP message. Default is blank.
<b>STUN Server</b>	IP address or Domain name of the STUN (Simple Traversal of UDP through NATs) server.
<b>Call Settings</b>	
<b>G723 Rate</b>	G723 encoding rate (6.3kbps or 5.3kbps)
<b>Voice Frames per Tx</b>	This field contains the number of voice frames to be transmitted in a single packet. When setting this value, the user should be aware of the requested packet time (used in SDP message) as a result of configuring this parameter. This parameter is



		<p>associated with the first vocoder in the above vocoder Preference List or the actual used payload type negotiated between the 2 conversation parties at run time.</p> <p>e.g., if the first vocoder is configured as G723 and the “Voice Frames per TX” is set to be 2, then the “ptime” value in the SDP message of an INVITE request will be 60ms because each G723 voice frame contains 30ms of audio. Similarly, if this field is set to be 2 and if the first vocoder chosen is G729 or G711 or G726, then the “ptime” value in the SDP message of an INVITE request will be 20ms.</p> <p>If the configured voice frames per TX exceeds the maximum allowed value, the BudgeTone 200 will use and save the maximum allowed value for the corresponding first vocoder choice. The maximum value for PCM is 10(x10ms) frames; for G726, it is 20 (x10ms) frames; for G723, it is 32 (x30ms) frames; for G729/G728, 64 (x10ms) and 64 (x2.5ms) frames respectively.</p>
<b>Local RTP port</b>		This parameter defines the local RTP-RTCP port pair the GXW410x will listen and transmit. It is the base RTP port for channel 0. When configured, channel 0 will use this port _value for RTP and the port_value+1 for its RTCP; channel 1 will use port_value+2 for RTP and port_value+3 for its RTCP and so on. The default value is 5004.
<b>RTP Loopback</b>		Default value is No. If set to Yes, means no RTP if RTP streams between 2 internal ports.
<b>Channel Settings</b>		
<b>DTMF Method</b>		<p>This parameter specifies the mechanism to transmit DTMF digits. There are 7 modes supported: in audio which means DTMF is combined in audio signal (not very reliable with low bit-rate codec), via RTP (RFC2833), or via SIP INFO. Multiple DTMF transmission schemas can be selected.</p> <p>1 – in-audio  2 – RFC2833  3 – in-audio and RFC2833  4 – SIP Info  5 – in-audio and RFC2833  6 – SIP Info and RFC2833  7 – in-audio, RFC2833, and SIP Info</p>
<b>No Key Entry Timeout</b>		Default is 4 seconds.
<b>Local SIP Listen Port</b>		Default is ch1-8:5060++;. The ++ indicates increments by 2, so port 1 is set at 5060, port at 5062 and so on. This setting can be used with Round Robin and/or Flexible setting below to configure different ports to be placed under different Round Robin groups.
<b>SRTP Mode</b>		Default is disabled for all ports. The user can select to either enable it but not force it or force it on an individual port basis. When used the communication will be sent using Secure RTP.
<b>Unconditional Forward to VOIP:</b>	<b>Call</b>	<p>This is an extremely important setting to make sure incoming PSTN calls are picked up and forwarded to the correct VOIP destination.</p> <p><b>User ID</b> - This parameter allows users to configure a User ID or extension number to be automatically dialed upon FXO line off-hook.</p> <p><b>SIP Server</b> - You also need to specify the Profile of the user id configured above (p1 stands for Profile 1, p2 stands for Profile 2 and so on).</p> <p><b>SIP Destination Port</b> - Along with the user-id and Profile, you also have the option to choose the destination port where you would like to send the call. By default it should be set to ch1-x:5060; (x can be 4 or 8 depending on number of ports).</p> <p>We can also specify a different destination for each port. For example under User ID we can type in: ch1:104;ch2:227;ch3-5:501;ch6,7:856.  Under Sip Server we can type in: ch1:p1;ch2-4:p2;ch5:p3  Under Sip Destination Port we can type in: ch1-2:5060;ch2:7080;ch3-8:5066++</p>
<b>T.38 Setting</b>		<p>This setting allows you to make several options related to facsimile.</p> <p>You can select the method: T.38 or Pass through (G711)</p> <p>You can select the fax transmission rates (2400/4800/7200/9600/12000/14400bps)</p> <p>You can enable or disable ECM (Error Checking Mode)</p> <p><b>Note:</b> The user can only test the parameters for only one of the PSTN lines at the</p>

same time. In all cases please enter the telephone numbers as if the lines were to dial each other locally.

For the AC Impedance Test we only need to select the line to be tested by clicking on the AC impedance box corresponding to that line, telephone numbers are optional. Remember that the AC impedance test is usually used to reduce the echo that might be present in the line.

For the CPT test (call progress tones) we will test current disconnect as well. You will need 2 telephone numbers to perform the test. You can only perform the test on one line (row) at the same time and it will be the one that has the box checked for testing. This tested line will use another line connected to the gateway to perform the test by calling into it, this is why you will have to enter the telephone number for a second line to help with the test.

For CID detection you will need 2 telephone numbers to perform the test. You can only perform the test on one line (row) at the same time and it will be the one that has the box checked for testing. This tested line will use another line connected to the gateway to perform the test by calling into it, this is why you will have to enter the telephone number for a second line to help with the test.

To perform the test, please select the line you want to test and the desired test to be performed. Enter the information for this line as well as a second line if necessary. Then click on the update button and then reboot. Log back in and now you should see the information for the line selected as well as the check box already marked already there. Go ahead and start the test now, please wait a few minutes until the test is done.

Notes:

It is not required to enter a telephone number when testing for impedance, as the system does not place any actual calls for the test.

If you log into the Web Interface while the test is running will not interrupt the process.

**TABLE 9: ACCOUNTS DEFINITIONS**

<b>ACCOUNTS</b>	
<b>General Settings</b>	
<b>Account Active</b>	When set to Yes the SIP Profile is activated.
<b>Account Name</b>	A name to identify a Profile.
<b>SIP Server</b>	SIP Server's IP address or Domain name provided by VoIP service provider.
<b>Outbound Proxy</b>	IP address or Domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by GXW410x for firewall or NAT penetration in different network environments. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy can correct the problem.
<b>Networks Settings</b>	
<b>Use DNS SRV:</b>	Default is No. If set to Yes the client will use DNS SRV to look up server.
<b>NAT Traversal</b>	This parameter defines whether the GXW410x NAT traversal mechanism will be activated or not. If activated (by choosing "Yes") and a STUN server is also specified, then the GXW410x will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the GXW410x will attempt to detect if and what type of firewall/NAT it is sitting behind through communication with the specified STUN server. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the GXW410x will attempt to use its mapped public IP address and port in all of its SIP and SDP messages. If the NAT Traversal field is set to "Yes" with no specified STUN server, the GXW410x will periodically (every 20 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open.
<b>Proxy-Require</b>	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.

<b>Use OBP in Route</b>	Utilizes outbound proxy in route.
<b>SIP Settings</b>	
<b>SIP Registration</b>	This parameter controls whether the GXW410x needs to send REGISTER messages to the SIP Server. The default setting is “Yes”.
<b>Unregister on Reboot</b>	Default is No. If set to yes, the SIP user’s registration information will be cleared on reboot.
<b>Register Expiration</b>	This parameter allows the user to specify the time frequency (in minutes) for the GXW410x to refresh its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
<b>SIP Registration Failure Retry Wait Time</b>	This parameter is mostly used by Service Providers. It prevents message REGISTER overload of SIP Server in case of downtime due to maintenance or power failure. By increasing interval length, common message load is decreased. Interval range is 1 – 3600 seconds.
<b>SIP Transport</b>	User can select UDP or TCP. Please make sure you’re SIP Server or network environment supports SIP over the selected transport method. Default is UDP.
<b>Session Expiration</b>	Grandstream implemented SIP Session Timer. The session timer extension enables SIP sessions to be periodically “refreshed” via a SIP request (UPDATE, or re-INVITE. Once the session interval expires, if there is no refresh via a UPDATE or re-INVITE message, the session will be terminated. Session Expiration is the time (in seconds) at which the session is considered timed out, if no successful session refresh transaction occurs beforehand. The default value is 180 seconds.
<b>Min-SE</b>	The minimum session expiration (in seconds). The default value is 90 seconds.
<b>Caller Request Timer</b>	If selecting “Yes” the phone will use session timer when it makes outbound calls if remote party supports session timer.
<b>Callee Request Timer</b>	If selecting “Yes” the phone will use session timer when it “receives inbound calls with session timer request.
<b>Force Timer</b>	If selecting “Yes” the phone will use session timer even if the remote party does not support this feature. Selecting “No” will allow the phone to enable session timer only when the remote party support this feature. To turn off Session Timer, select “No” for Caller Request Timer, Callee Request Timer, and Force Timer.
<b>UAC Specify Refresher</b>	As a Caller, select UAC to use the phone as the refresher, or UAS to use the Callee or proxy server as the refresher.
<b>UAS Specify Refresher</b>	As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the phone as the refresher.
<b>Force INVITE</b>	Session Timer can be refreshed using INVITE method or UPDATE method. Select “Yes” to use INVITE method to refresh the session timer.
<b>Enable 100rel</b>	The use of the PRACK (Provisional Acknowledgment) method enables reliability to be offered to SIP provisional responses (1xx series). This is very important if PSTN inter-networking is to be supported. A user’s request to use reliable provisional responses is invoked by the 100rel tag which is appended to the value of the required header of initial signalling messages.
<b>Refer-to uses Target Contact</b>	Default is NO. If set to YES, then for Attended Transfer, the “Refer-To” header uses the transferred target’s Contact header information.
<b>INVITE Ring-no-answer Timeout</b>	In case incoming call has arrived from PSTN to VoIP and INVITE message was generated by GXW device, the call will be disconnected after preconfigured timeout if not answered by VoIP extension.
<b>Accept INVITE from Proxy Only</b>	Default is YES. The device will authenticate and accept only incoming INVITE messages from the peered SIP server.
<b>Audio Settings</b>	
<b>Preferred Vocoder</b>	The GXW410x supports up to 5 different Vocoder types including G.711 A-/U-law, GSM, G.723.1, G.729A/B. The user can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder in this list can be entered by choosing the appropriate option in “Choice 1”. Similarly, the last Vocoder in this list can be entered by choosing the appropriate option in “Choice 8”.
<b>Call Settings</b>	
<b>User ID is Phone Number</b>	If the GXW410x has an assigned PSTN telephone number, this field should be set to “Yes”. Otherwise, set it to “No”. If “Yes” is set, a “user=phone” parameter will be attached to the “From” header in SIP request.

<b>Early Dial</b>	Default is No. Use only if proxy supports 484 response.
<b>User Accounts</b>	<b>Note</b> – The channels here are basically SIP endpoints that will act as clients registering to the SIP Server configured under the appropriate Accounts page.
<b>Channels</b>	It should be set same as the channel number (i.e 1, 2..4 or 8 depending on number of FXO ports). <i>It is NOT the same as SIP Account ID.</i>
<b>SIP User ID</b>	This is the SIP account information. Enter the SIP User ID part of the account.
<b>Authentication ID</b>	SIP service subscriber's Authenticate ID used for authentication. It can be identical to, or different from SIP User ID.
<b>Authen Password</b>	SIP account password needs to be entered here. Note: After entering the password, it will show up as blank but the password still remains active.

## CONFIGURING THE FXO CHANNELS

Configuring the FXO channels on the GXW – 410x is an easy process. Follow the GUI interfaces. The Device Status page terms are defined in **Table 8: FXO Lines Configuration Definitions**. An example of the Channel Dialing Configuration is shown in Figure 6. Please note the default is always configured. The user has the option to change the default settings as described in the Table 8.

**TABLE 10: FXO LINES (SETTINGS / DIALING)**

FXO LINES	
<b>Settings</b>	
<b>Call Progress Tones</b>	<p>Using these settings, user can configure tone frequencies according to user preference. By default, the tones are set to North American frequencies. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. <b>ON</b> is the period of ringing (ON time in ms) while <b>OFF</b> is the period of silence. In order to set a continuous ring, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern.</p> <ul style="list-style-type: none"> <li>• “Dial tone”</li> <li>• “Ringback tone”</li> <li>• “Busy/Re-order tone”</li> <li>• “Confirmation tone”</li> </ul> <p>Please refer the document below to determine your local call progress tones (<a href="http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf">http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf</a>) or run the FXO Line Test (Table 9).</p>
<b>Tx to PSTN Audio Gain (dB)</b>	Allows user to set a value in dB for transmission to PSTN Audio Gain. Default is 1. Range is from -12 to 12dB.
<b>Rx from PSTN Audio Gain (dB)</b>	Allows user to set a value in dB for receive from PSTN Audio Gain. Default is 0. Range is from -12 to 12dB.
<b>Silence Suppression</b>	This controls the silence suppression/VAD feature of G723 and G729. If set to “Yes”, when a silence is detected, small quantity of VAD packets (instead of audio packets) will be sent during the period of no talking. If set to “No”, this feature is disabled.
<b>Echo Cancellation</b>	When set to Y, Echo cancellation is enabled.
<b>Enable Current Disconnect</b>	<p>When set to Y, Current Disconnect is enabled. Certain PSTN COs require this to be enabled, in order to realize disconnect signal from PSTN side. Default is Y.  <b>If enabled use threshold:</b> Default is 100ms. Range is 40ms to 800ms.</p> <p>Certain PSTN service providers have a threshold time within which the line stabilizes after off-hook. It is entirely dependent on provider, however if you experience PSTN line detection issues, please modify this setting appropriately in 100ms increments.</p> <p>If you are not sure if this option should be enabled please refer to Table 9 (FXO Lines Test Tab Definition). This tool will run an automated test to determine the proper PSTN configuration that the gateway should have to work with your Service Provider or analog PBX.</p>
<b>Enable Tone Disconnect</b>	<p>Default is No. If PSTN provider uses call progress tones then it should be set to Yes in order to realize disconnect tone. Please configure accurate Call Progress Tones on Channels webpage based on PSTN provider (or traditional PBX) settings.</p> <p>If you are not sure if this option should be enabled or what Call Progress Tones are required please refer to Table 9 (FXO Lines Test Tab Definition). This tool will run an automated test to determine the proper PSTN configuration that the gateway should have to work with your Service Provider or analog PBX.</p>
<b>Enable Polarity Reversal</b>	<p>Default is No. This should be set to Yes only if the FXO lines are subscribed to PR service from PSTN Service provider. It is merely a PR detect feature.</p> <p>***Note: If there is no PR service from provider on the FXO line, and this setting is configured to Yes, calls will not be successful.</p>
<b>Enable Call Answer Supervision</b>	Default is No. If PSTN providers use the CAS then this option should be enabled. The Call Answer Supervision (CAS) is for billing—the telephone exchange and the

	customer need an accurate indication of calls through a network.
<b>Silence Timeout</b>	Terminate call after long silence detected. Default is 60 seconds, max 65536.
<b>Incoming Call Timeout</b>	Default value is 6 seconds. Incoming call will stop ringing when not picked up given a specific period of time.
<b>AC Termination Impedance</b>	<p>Selects the impedance of the analog line connected to the FXO port on the GXW410x. Here is some basic information which may be helpful for initial configuration:</p> <p>600 Ohm – North America;          270 Ohm + (750 Ohm    150 nF) -- Most of Europe          220 Ohm + (820 Ohm    120 nF) – Australia, New Zealand          220 Ohm + (820 Ohm    115 nF) – Austria, Bulgaria, Germany, Slovakia, South Africa          370 Ohm + (620 Ohm    310 nF) – UK., India</p> <p>If this parameter is not configured properly you may experience echo or static in the line. Please refer to Table 9 (FXO Lines Test Tab Definition). This tool will run an automated test to determine the correct impedance value to match your lines</p>
<b>Number of Rings Before Pickup</b>	Default is 4. This is the number of rings the gateway will wait to send the call to the VOIP side in case the Caller ID has yet to be detected. If there's CID information the call will be sent right away. If your lines don't have the CID service set this to 1.
<b>Caller ID Scheme</b>	<p>The GXW410x supports 5 different types of schemes:</p> <ol style="list-style-type: none"> <li>1. Bellcore (US standard)</li> <li>2. ETSI-FSK during ringing</li> <li>3. ETSI-FSK prior to ringing with DTAS</li> <li>4. ETSI-FSK prior to ringing with LR</li> <li>5. ETSI-FSK prior to ringing with PR</li> <li>6. ETSI-DTMF during ringing</li> <li>7. ETSI-DTMF prior to ringing with DTAS</li> <li>8. ETSI-DTMF prior to ringing with PR</li> <li>9. ETSI-DTMF prior to ringing with PR</li> <li>10. SIN 227 - BT</li> <li>11. NTT (Japanese standard)</li> </ol> <p>Please check with your PSTN service provider (or traditional PBX specs) for which caller ID scheme they/it support. If you are not sure about which to use please refer to Table 9 (FXO Lines Test Tab Definition). This tool will run an automated test to determine the proper Caller ID Scheme so the gateway can properly detect the Caller ID.</p> <p>Similarly to the cases explained above we can specify a caller ID scheme for each channel independently.</p>
<b>Caller ID Transport type</b>	<p>Default is “relay via From header”. You may also select :</p> <p>“relay via P_Asserted_Identity header”          “Disable” : Caller ID feature will be disabled.          “Send anonymous” : All calls forwarded to VOIP end will be sent as anonymous.</p>
<b>DIALING</b>	
<b>Wait for Dial-tone</b>	Default is Yes. When set to Yes, the gateway will recognize dial-tone from the Central Office (CO) before it completes call. If you can't make an outbound call, set this is <b>No</b> .
<b>Stage Method</b>	<p>Syntax - ch1-8:1; {all channels 1 to 8 are set to value 1 or 2}</p> <p>Stage method can be set to either 1 or 2.</p> <p>Set this parameter to 1 if you need to make a direct PSTN call from a VOIP endpoint. When you set it to 2, you will first dial one of the VOIP channel accounts from the VOIP endpoint, this will result in getting a PSTN line dial-tone to then dial out the</p>

	<p>destination PSTN number.</p> <p>Most implementations require this setting to be configured to 1.</p>
<b>Min. Delay before Dial PSTN</b>	<p>Default is 500ms. This needs to be equal to or greater than the Current Disconnect threshold setting. Once the threshold is reached the gateway can dial out. This parameter should only be used if there are PSTN line detection issues.</p>
<b>Round Robin and/or Flexible</b>	<p>Default is rr:1-8;</p> <p>The syntax is pretty straight-forward here. The rr stands for Round Robin and the numbers stand for the ports that belong to that round robin group.</p> <p><b>For example:</b></p> <p>rr:1-8; -&gt; Round robin within the first 8 ports i.e. outgoing calls will be forwarded to the next available port within the group of ports 1 to 8.</p> <p>rr:1,3-6,8;rr:2,7; -&gt; Round robin within port 1,3,4,5,6 and 8; Second round robin group within ports 2 and 7 i.e. outgoing calls to ports 1,3,4,5,6 and 8 will be forwarded to the next available port within this group ONLY. Outgoing calls to port 2 and 7 will be forwarded to the next available port between ports 2 and 7 ONLY.</p> <p><i>** In order to terminate a call on FXO port 2 or 7 you will need to change its Local SIP Listen port accordingly.</i></p>
<b>Prefix to specify Port (1 stage dialing method)</b>	<p>Default is 99.</p> <p>Syntax to USE this feature: prefix# (that is 99) + ch# (could be anything from 1 to 8) + dialing# will result in this call forwarded to FXO port (ch#) immediately.</p>
<b>Dial Plan</b>	<p>The Dial Plan feature implemented is applicable for VOIP to PSTN calls only. You may configure a dial plan based on the following grammar:</p> <ol style="list-style-type: none"> <li>Accept Digits: 1,2,3,4,5,6,7,8,9,0,*,#,A,a,B,b,C,c,D,d</li> <li>Grammar: <ul style="list-style-type: none"> <li>x - any digit from 0-9;</li> <li>xx+ - at least 2 digit number;</li> <li>xx. - at least 2 digit number;</li> <li>^ - exclude;</li> <li>[3-5] - any digit of 3, 4, or 5;</li> <li>[147] - any digit 1, 4, or 7;</li> <li>&lt;2=011&gt; - replace digit 2 with 011 when dialing</li> <li>WARNING - illegal input will fall back to default</li> </ul> </li> </ol> <p><b>Example 1:</b> {[369]11   1617xxxxxxx} - Allow 311, 611, 911, and any 10 digit numbers of leading digits 1617.</p> <p><b>Example 2:</b> {^1900x+   &lt;=1617&gt;xxxxxxx} - Block any number of leading digits 1900 and add prefix 1617 for any dialed 7 digit numbers.</p> <p><b>Example 3:</b> {1xx[2-9]xxxxxx   &lt;2=011&gt;x+} - Allow any length of number with leading digit 2 and 10 digit-numbers of leading digit 1 and leading exchange number between 2 and 9;</p> <p>If leading digit is 2, replace leading digit 2 with 011 before dialing</p> <p><b>Example 4:</b> { [x#]+   [x*]+ } - Allow any length of number with leading * or # in number to dial.</p> <p>Default: PSTN Outgoing - {x+}</p> <p><b>Note:</b> If you do not plan to use this feature set to default {x+}</p>
<b>Hookflash Duration (X10ms)</b>	<p>Default 600ms. This value can accept any value in the 100-2000ms range.</p>
<b>Use DTMF Parameter from RFC2833 or SIP Info</b>	<p>Default Yes, No means to use DTMF parameter settings according to DTMF Digit Length, DTMF Digit Volume and DTMF Dial Pause.</p>
<b>DTMF Digit Length</b>	<p>Default value is 100ms. Please note that the value will be multiplied by 10ms</p>
<b>DTMF Digit Volume</b>	<p>Default value is -11dB.</p>
<b>DTMF Dial Pause</b>	<p>Default value is 100ms. Please note that the value will be multiplied by 10ms.</p>

**TABLE 11: FXO LINE ANALYSIS**

<b>LINE ANALYSIS</b>	
<b>Overview</b>	<p><b>Note:</b> The user can only test the parameters for only one of the PSTN lines at the same time. In all cases please enter the telephone numbers as if the lines were to dial each other locally.</p> <p>For the AC Impedance Test we only need to select the line to be tested by clicking on the AC impedance box corresponding to that line, telephone numbers are optional. Remember that the AC impedance test is usually used to reduce the echo that might be present in the line.</p> <p>For the CPT test (call progress tones) we will test current disconnect as well. You will need 2 telephone numbers to perform the test. You can only perform the test on one line (row) at the same time and it will be the one that has the box checked for testing. This tested line will use another line connected to the gateway to perform the test by calling into it, this is why you will have to enter the telephone number for a second line to help with the test.</p> <p>For CID detection you will need 2 telephone numbers to perform the test. You can only perform the test on one line (row) at the same time and it will be the one that has the box checked for testing. This tested line will use another line connected to the gateway to perform the test by calling into it, this is why you will have to enter the telephone number for a second line to help with the test.</p> <p>To perform the test, please select the line you want to test and the desired test to be performed. Enter the information for this line as well as a second line if necessary. Then click on the update button and then reboot. Log back in and now you should see the information for the line selected as well as the check box already marked already there. Go ahead and start the test now, please wait a few minutes until the test is done.</p> <p><u>Notes:</u>            It is not required to enter a telephone number when testing for impedance, as the system does not place any actual calls for the test.            If you log into the Web Interface while the test is running will not interrupt the process.</p>
<b>Auto Detect</b>	
<b>Line #</b>	Enter the telephone (PSTN) number that corresponds to this line. Enter it as if you were going to dial it locally.
<b>AC Impedance</b>	Select this box if you want to test for impedance on the line that is on the same row as the checked box. Remember that you can only check one item at the same time.
<b>CPT Detection</b>	Select this box if you want to test for call progress tones and current disconnect threshold on the line that is on the same row as the checked box. Remember that you can only check one item at the same time.
<b>External Number</b>	Enter an external telephone (PSTN) number to be used as an auxiliary number for the test. This is used only if we do not have at least 2 PSTN lines connected to the gateway. This is only used for CPT and current disconnect threshold testing. In order to use this function you will have to monitor the Syslog output. This is only reserved for very advanced users.
<b>External Call Timeout</b>	This is the time the GXW will wait for the external telephone number to pick up during the test.
<b>Apply test results automatically</b>	Default is No. If selected on Yes, then all the results from the test will be applied automatically. If you select No you will have to monitor the Syslog output. This is only reserved for very advanced users.
<b>Apply test results to all ports</b>	Default is No. If selected on Yes, then all the test results will be applied to all ports on the gateway. If all the lines belong to the same service provider or PBX it will make sense to apply the results to all ports.
<b>Error Timeout</b>	This is the time the gateway will wait to exit the test mode, when something unexpected or an error has occurred.



## CHECK DEVICE STATUS

You may access the Device Status page which provides details of the GXW product. The Device Status page terms are defined in **Table 11: Status Page Definitions**.

**TABLE 12: STATUS PAGE DEFINITIONS**

<b>STATUS</b>	
<b>Hardware Revision</b>	Hardware version number: Main Board, Interface Board
<b>MAC Address</b>	The device ID in HEX format. This is a very important ID for ISP troubleshooting.
<b>IP Address</b>	This field shows WAN IP address of GXW410x
<b>Product Model</b>	This field contains the product model info (GXW4104 or GXW4108)
<b>Software Version</b>	Program: This is the main software release. Boot and Loader are not changed often.
<b>System Up Time</b>	This field shows system up time since the last reboot.
<b>Registered</b>	This field indicates whether the different SIP Accounts configured under Channels page are successfully registered to the SIP server(s).
<b>FXO Line Connected</b>	This field will give the status of each physical FXO Line connected to the Gateway. It will update the status regularly. <b>Yes</b> - Connected and Idle <b>Busy</b> - Connected and Busy <b>No</b> - Not connected Additionally it will also provide real time Caller ID information of Incoming as well as Outgoing calls.
<b>PPPoE Link Up</b>	This field shows whether the PPPoE connection is running if connected to DSL modem.

## SAVING THE CONFIGURATION CHANGES

Once a change is made, press the “Update” button in the Configuration Menu. The GXW410x will display the following screen to confirm that the changes have been saved. To activate changes, reboot or power cycle the GXW410x after all changes are made.

## REBOOTING FROM REMOTE

The administrator can remotely reboot the unit by pressing the “Reboot” button at the bottom of the configuration menu. The following screen will indicate that rebooting is underway.

The user can re-login to the unit after waiting for about 30 seconds.

## VIDEO SURVEILLANCE

The GXW410x (**HW version 1 only**) can be used with an Analog Surveillance CCD Camera to perform video surveillance function. This application should be used in a LAN environment or when both sides have public IP address.

**NOTE:** The following information is essential, only if your GXW410x's hardware (Revision 1) has a Video-IN Port, otherwise It will not show this functions.

### VIDEO SURVEILLANCE PROCEDURES

➤ **Gateway side:**

1. In the NETWORKS page->Advanced Settings, find the following field and change from default setting NO to YES, reboot the device.

**Video Surveillance:**    No    Yes   RTSP Port:  (default is 554)

2. Connect an analog based surveillance camera to the VIDEOIN connection at the back panel of the unit.

➤ **PC side (Monitor Device):**

1. Download VLC from <http://www.videolan.org/vlc/>. This is the only player so far that supports RFC 3984.
2. Launch VLC.
3. Go to Preferences->Input/Codecs->Demuxers->H264, check "Advanced options" in the bottom. The option "Frames per Second" will show. Change that value to 5 and then save.
4. Go to Preferences->Input/Codecs->Access modules->Real RTSP, check "Advanced options" in the bottom. The option "Caching value (ms)" will show. Change that value to 1000 and then save. You may change it to a smaller value to reduce the delay.
5. If the viewer is under NAT, go to Preferences->Demuxers->Access modules->RTP/RTSP, check "Advanced options" in the bottom. The option "Use RTP over RTSP (TCP)" will show. Check that option box. (*Grandstream does NOT recommend this network environment*)
6. Close the Preferences window and go to File->Open Network Stream:
  - a) Select RTSP as the protocol
  - b) Enter the URL in the format of rtsp://admin:  
**ADMIN\_PASSWORD@DEVICE\_IP\_ADDRESS:DEVICE\_RTSP\_PORT.**  
 Change the blue text according to your configuration:
    - **ADMIN\_PASSWORD** is the device's web configuration password for admin.
    - **DEVICE\_IP\_ADDRESS** is the device IP.
    - **DEVICE\_RTSP\_PORT** is the RTSP port setting of the device.
 If the port uses default value 554, the port portion can be omitted from the URL
  - c) Click OK to start the video.

**FIGURE 5: SCREEN-SHOT OF VIDEO SURVEILLANCE\***



\* PC client side running VLC as monitoring station

## SOFTWARE UPGRADE

Software upgrade can be done via either TFTP or HTTP. The corresponding configuration settings are in the ADVANCED SETTINGS configuration page.

### FIRMWARE UPGRADE THROUGH TFTP/HTTP/HTTPS

To upgrade via TFTP or HTTP/HTTPS, the “Firmware Upgrade and Provisioning upgrade via” field needs to be set to TFTP HTTP or HTTPS, respectively. “Firmware Server Path” needs to be set to a valid URL of a TFTP or HTTP server, server name can be in either FQDN or IP address format. Here are examples of some valid URL.

- e.g. firmware.mycompany.com:6688/Grandstream/1.4.1.5
- e.g. firmware.grandstream.com

#### NOTES:

- Firmware upgrade server in IP address format can be configured via IVR. Please refer to the CONFIGURATION GUIDE section for instructions. If the server is in FQDN format, it must be set via the web configuration interface.
- Grandstream recommends end-user use the Grandstream HTTP server. Its address can be found at <http://www.grandstream.com/support/firmware>. Currently the HTTP firmware server address is firmware.grandstream.com. For large companies, we recommend to maintain their own TFTP/HTTP/HTTPS server for upgrade and provisioning procedures.
- Once a “Firmware Server Path” is set, user needs to update the settings and reboot the device. If the configured firmware server is found and a new code image is available, the GXW410x will attempt to retrieve the new image files by downloading them into the GXW410x 's SRAM. During this stage, the GXW410x's LEDs will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will then be saved into the Flash. If TFTP/HTTP/HTTPS fails for any reason (e.g. TFTP/HTTP/HTTPS server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the GXW410x will stop the TFTP/HTTP/HTTPS process and simply boot using the existing code image in the flash.
- Firmware upgrade may take as long as 15 to 30 minutes over Internet, or just 5 minutes if it is performed on a LAN. It is recommended to conduct firmware upgrade in a controlled LAN environment if possible.
- Grandstream's latest firmware is available <http://www.grandstream.com/support/firmware>.  
***Overseas users are strongly recommended to download the binary files and upgrade firmware locally in a controlled LAN environment.***
- Alternatively, user can download a free TFTP or HTTP server and conduct local firmware upgrade. A free windows version TFTP server is available for download from <http://www.solarwinds.com/register/?Program=52&c=7015000000CcH2>. Our latest official release can be downloaded from <http://www.grandstream.com/firmware.htm>.

#### Instructions for local firmware upgrade:

1. Unzip the file and put all of them under the root directory of the TFTP server.
2. Put the PC running the TFTP server and the GXW410x device in the same LAN segment.
3. Please go to File -> 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, in the phone's web configuration page

5. Configure the Firmware Server Path with the IP address of the PC
6. Update the change and reboot the unit

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

## CONFIGURATION FILE DOWNLOAD

Grandstream SIP Device can be configured via Web Interface as well as via Configuration File through TFTP or HTTP/HTTPS. “Config Server Path” is the TFTP or HTTP/HTTPS server path for configuration file. It needs to be set to a valid URL, either in FQDN or IP address format. The “Config Server Path” can be same or different from the “Firmware Server Path”.

A configuration parameter is associated with each particular field in the web configuration page. A parameter consists of a Capital letter P and 2 to 3 (Could be extended to 4 in the future) digit numeric numbers. i.e., P2 is associated with “Admin Password” in the ADVANCED SETTINGS page. For a detailed parameter list, please refer to the corresponding firmware release configuration template.

When Grandstream Device boots up or reboots, it will issue request for configuration file named “cfgxxxxxxxxxxx”, where “xxxxxxxxxxx” is the LAN MAC address of the device, i.e., “cfg000b820102ab”. The configuration file name should be in lower cases.

## FIRMWARE AND CONFIGURATION FILE PREFIX AND POSTFIX

Firmware Prefix and Postfix allows device to download the firmware name with the matching Prefix and Postfix. This makes it the possible to store ALL of the firmware with different version in one single directory. Similarly, Config File Prefix and Postfix allows device to download the configuration file with the matching Prefix and Postfix. Thus multiple configuration files for the same device can be stored in one directory.

In addition, when the field “Check New Firmware only when F/W pre/suffix changes” is set to “Yes”, the device will only issue firmware upgrade request if there are changes in the firmware Prefix or Postfix.

## MANAGING FIRMWARE AND CONFIGURATION FILE DOWNLOAD

When “Automatic Upgrade” is set “**Yes, every**” the auto check will be done in the minute specified in this field. If set to “**daily at hour (0-23)**”, Service Provider can use P193 (Auto Check Interval) to have the devices do a daily check at the hour set in this field with either Firmware Server or Config Server. If set to “**weekly on day (0-6)**” the auto check will be done in the day specified in this field. This allows the device periodically check if there are any new changes need to be taken on a scheduled time. By defining different intervals in P193 for different devices, Server Provider can spread the Firmware or Configuration File download in minutes to reduce the Firmware or Provisioning Server load at any given time.

Automatic Upgrade:

- No  Yes, every  minutes(60-5256000).
- Yes, daily at hour  (0-23).  Yes, weekly on day  (0-6).

## RESTORE FACTORY DEFAULT SETTING

**WARNING!** Restoring the Factory Default Setting will DELETE all configuration information of the phone. Please BACKUP or PRINT out all the settings before you approach to following steps. Grandstream will not take any responsibility if you lose all the parameters of setting and cannot connect to your VoIP service provider.

### FACTORY RESET

#### Reset Button

Reset default factory settings following these four (4) steps:

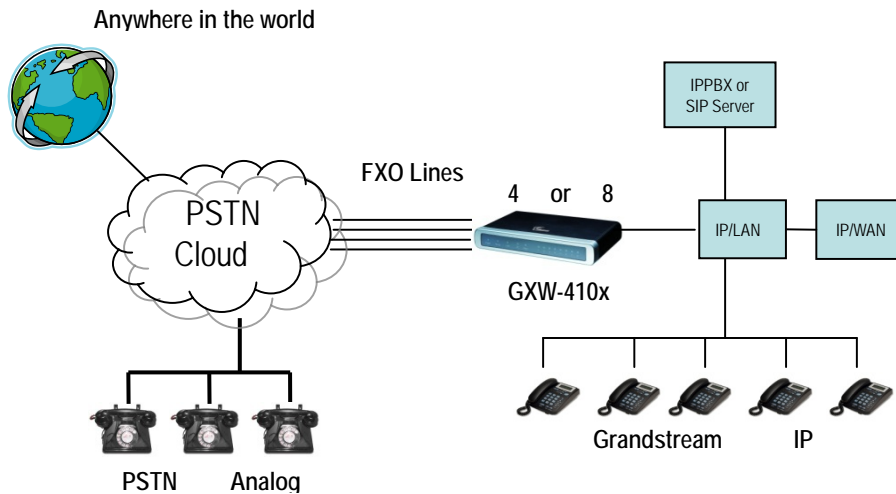
1. Unplug the Ethernet cable.
2. Locate a needle-sized hole on the back panel of the gateway unit next to the power connection.
3. Insert a pin in this hole, and press for about 7 seconds.
4. Take out the pin. All unit settings are restored to factory settings.

## EXAMPLES OF GXW410X CONFIGURATIONS

### APPLICATION 1: GXW CONNECTED WITH AN IP-PBX OR SIP SERVER

**Scenario:** A business with a traditional phone system (with or without broadband access) and an IP PBX or SIP Servers connecting to an Internet Telephone Service Provider (ITSP).

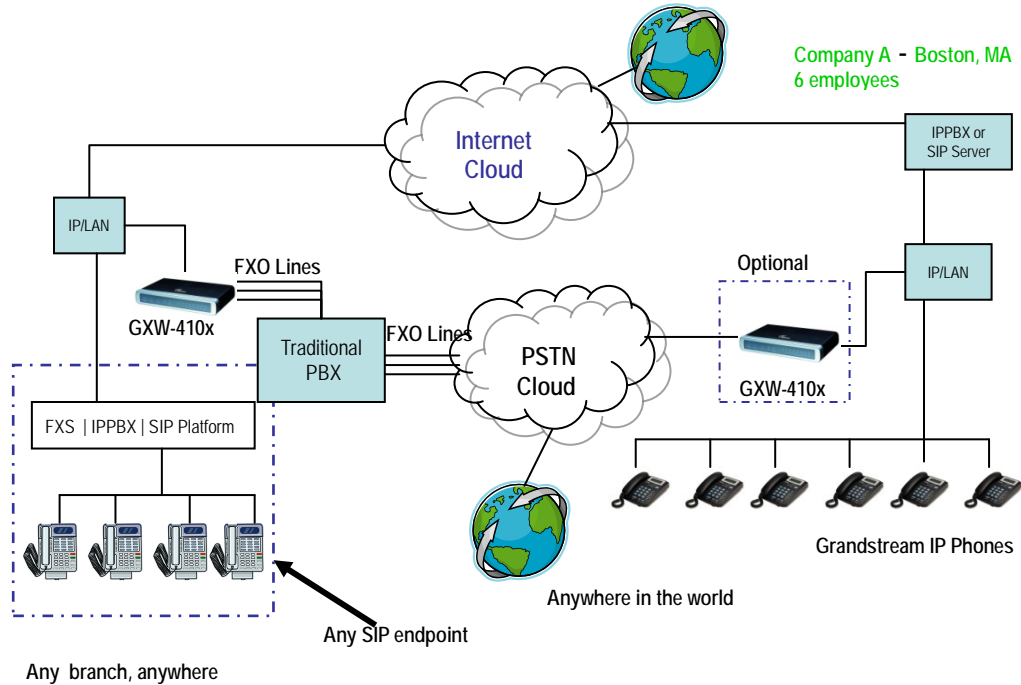
FIGURE 6: GXW CONNECTED WITH AN IP-PBX OR SIP SERVER



### APPLICATION 2: USE GXW TO EXTEND A TRADITIONAL PBX SCENARIO

**Scenario:** a small business with traditional analog PBX lines and broadband access who want to extend their traditional PBX to virtually anywhere in the world, using the internet. (Any SIP End point, such as Grandstream BugeTone, HandyTone, GXP-2000 or GXV-3000 are needed in this scenario)

FIGURE 7: GXW TO EXTEND A TRADITIONAL PBX SCENARIO



### APPLICATION 3: USING A GXW FOR PURE IP- IP COMMUNICATION CONFIGURATION

**Scenario Four:** The GXW410x offers an IP to IP pure IP Communications System configuration, where all locations use IP phones.

FIGURE 7: USING A GXW FOR PURE IP- IP COMMUNICATION CONFIGURATION

