



## Grandstream Networks, Inc.

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Analog IP Gateway GXW-410xv  
4 or 8 FXO Ports w/Video Surveillance



GXW-410xv User's Manual  
Firmware Version 1.0.0.36

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## GUI INTERFACES GXW-410XV USER MANUAL

[http://www.grandstream.com/user\\_manuals/GUI/GUI\\_GXW410x.rar](http://www.grandstream.com/user_manuals/GUI/GUI_GXW410x.rar)

1. SCREENSHOT OF ADVANCED SETTINGS CONFIGURATION PAGE
2. SCREENSHOT OF BASIC SETTINGS CONFIGURATION PAGE
3. SCREENSHOT OF CHANNELS CONFIGURATION PAGE
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 GXW-410x IP Analog FXO Gateway. The GXW-410xv is a cost effective, easy to use and easy to configure IP communications solution for any business. The GXW-410x 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, and direct IP-IP Calling. 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 GXW-410x Overview

The GXW410x offers an easy to manage, feature rich 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 GXW-4104v and GXW-4108v, which have either 4 and 8 FXO ports respectively. The installation is the same for either model.

A SIP proxy server such as Asterisk or a SIP registrar server can be deployed with the GXW-410xv series. In this environment, the SIP server handles SIP registration and call control and the GXW-410xv processes media conversion between IP and PSTN calls. By design, the system supports the call progress tones and PSTN signaling standards for North America, Europe, Latin America, Asia, and various other countries/regions.

**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 remember to download these examples [http://www.grandstream.com/user\\_manuals/GUI/GUI\\_GXW410x.rar](http://www.grandstream.com/user_manuals/GUI/GUI_GXW410x.rar) for your reference.
- This document is subject to change without notice. The latest electronic version of this user manual is available for download from the following location:  
[http://www.grandstream.com/user\\_manuals/GXW410x\\_User\\_Manual.pdf](http://www.grandstream.com/user_manuals/GXW410x_User_Manual.pdf)
- *Reproduction or transmittal of the entire or any part, in any form or by any means, electronic or print, for any purpose without the express written permission of Grandstream Networks, Inc. is not permitted.*

## PACKAGING

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

- 1) One GXW-410xv Unit
- 2) One universal power adaptor
- 3) One Ethernet cable

## SAFETY COMPLIANCES

The GXW-410xv 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 GXW-410xv package. Using an alternative power adapter may permanently damage 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.

## CONNECTING THE GXW-410XV

FIGURE 1: DIAGRAM OF GXW-410XV BACK PANEL

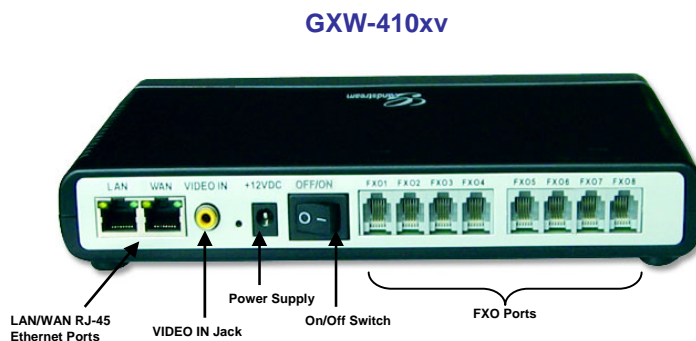


TABLE 1: DEFINITIONS OF THE GXW CONNECTORS

<b>LAN (or PC)</b>	Connect your PC to the LAN to find IP address from your Router/DHCP Server. The GXW-410xv acts as a switch.
<b>WAN (or LAN)</b>	Connect to the internal LAN network or router.
<b>VIDEO IN</b>	Connection for Analog based Video Surveillance Camera (RCA)
<b>RESET</b>	Factory Reset button. Press for 7 seconds to reset factory default settings.
<b>POWER IN</b>	Power adapter connection
<b>OFF/ON</b>	Off/On switch
<b>FXO1 - FXO8</b>	FXO ports to be connected to physical PSTN lines from a traditional PSTN PBX or PSTN Central Office.

**FIGURE 2: DIAGRAM OF GXW-410XV DISPLAY PANEL**



**TABLE 2: DEFINITIONS OF THE GXW DISPLAY PANEL**

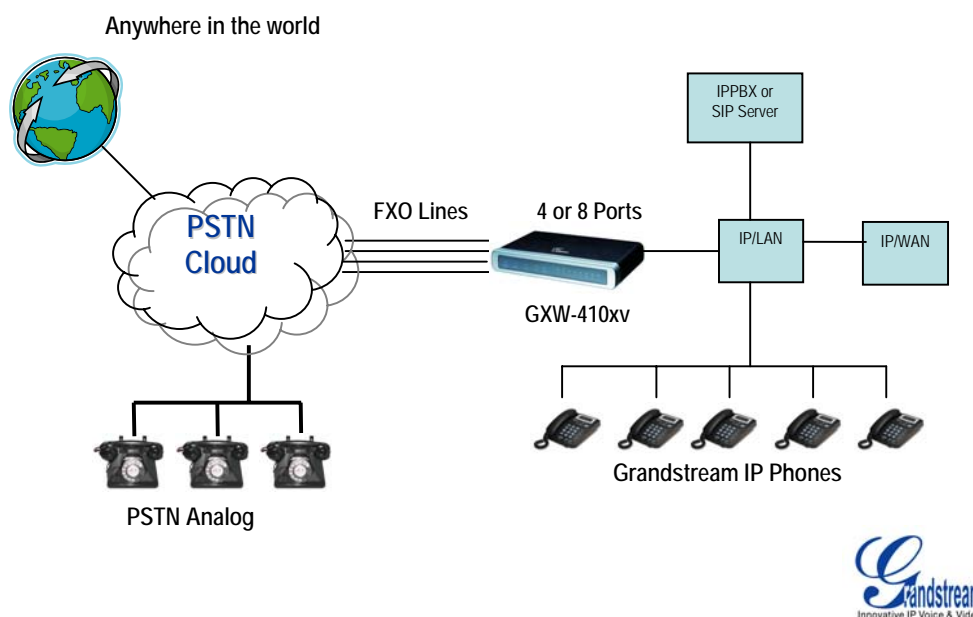
<b>Power LED</b>	Indicates Power. Remains ON when Power is connected and unit is turned ON.
<b>Ready LED</b>	Remains ON after boot-up.
<b>LAN LED</b>	Indicates LAN (or WAN) port activity
<b>PC LED</b>	Indicates PC (or LAN) port activity
<b>Video LED</b>	Remains solid green on boot-up. If Video IN terminal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

**NOTE:** All LEDs display green when ON.

## Application Description

A SIP proxy server such as Asterisk or a SIP registrar server can be deployed with the GXW-410xv series. In this environment, the SIP server handles SIP registration and call control and the GXW-410xv processes media conversion between IP and PSTN calls. By design, the system supports the call progress tones and PSTN signaling standards for North America, Europe, Latin America, Asia, and various other countries/regions.

**FIGURE 3: FUNCTIONAL DIAGRAM OF IP-PBX & GXW-410xv**



### SIMPLE CONFIGURATION: MEDIA GATEWAY TO ACCESS PSTN NETWORKS

GXW-410xv can be configured to work with any leading SIP server, for a pure media gateway to access PSTN networks. In such applications, the user only needs to configure GXW-410xv gateway Stage Dialing field and Sip Server field.

For a simple set-up, users only need to configure a SIP server field for default SIP Profile 1. This field should be configured to point to the SIP server to be used with the GXW-410xv.

For advanced applications, the user is required to choose at least one SIP server field from the SIP profiles and one stage dialing under system Channel configuration table. On SIP server sides, the SIP server must be configured to forward user PSTN calls to the GXW-410xv.

Please be aware that by default, the system uses North American PSTN settings and TWO STAGE dialing to access PSTN networks for VOIP to PSTN calls and PSTN to VOIP calls. Two stage dialing means the end-user will hear dial-tone twice. First dial-tone is used to let users to input destination number in the same network of the calling networks. Second dial-tone is used to let users to input final destination number.





The GXW-410xv also supports ONE STAGE dialing, which means users only need to input one final PSTN number when first dial-tone is heard for calls from VOIP to PSTN. This requires configuring both SIP server and GXW-410xv to one stage dialing (see last section of quick guide and user manual for one stage dialing). For one stage PSTN to VOIP calls, user needs to configure off-hook auto dial field (see last section on sample configuration)

## **EXTENSIVE CONFIGURATION: MEDIA GATEWAY CONFIGURATION FOR MULTIPLE USERS**

The GXW-410xv can be configured to work with a variety of SIP server features and traditional PBX on PSTN networks, with a different SIP server on each physical port. Each port may have its own voice setting, dialing settings, PSTN termination setting, and DTMF transmission settings.

## **OFF-HOOK AUTO DIAL**

The FXO interface currently does not support direct inward dialing (DID). The GXW-410xv implements an off-hook auto dial feature for each physical port. Configure off-hook auto dial to forward PSTN incoming call to a specific SIP number, call center or hunt group.

## **SAMPLE CONFIGURATIONS - IP PBX PEERS WITH GXW-410xv**

There are 2 methods to configure GXW to work with IP PBX:

- 1) Configure GXW with SIP Accounts in IP PBX, this will enable you to put GXW behind a NAT/Firewall.
- 2) Configure GXW without SIP Accounts in IP PBX, this makes GXW function as a PEER gateway.

See the *Quick Install Guide* at [http://www.grandstream.com/user\\_manuals/GXW41xx\\_QuickIG.pdf](http://www.grandstream.com/user_manuals/GXW41xx_QuickIG.pdf) for more information for quick installation.

**Note:** In regions other than North American, the user is also required to configure call progress tones and PSTN line termination fields. Check with local PSTN service carriers on values service providers use on the lines. If service provider doesn't provide these values and users don't know what the correct values are, please use the default values. Contact product support for additional help in configuring your GXW-410xv.

## Features

GXW-410x 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
- 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: G711, G723, G729 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

**TABLE 3: GXW-410XV SOFTWARE FEATURES**

<b>GXW– 410x FX0 Analog Gateway Series</b>	
<b>Telephone Interfaces</b>	GXW-4104: 4 ports; 4 SIP accounts w/ choice of 3 profiles GXW-4108: 8 ports; 8 SIP accounts w/ choice of 3 profiles 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.711, G.723, G.726, G.729A/B, GSM
<b>Video Surveillance</b>	Real-time H.264 base CIF resolution
<b>DHCP Server/Client</b>	Yes, NAT Router or Switched Mode
<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 (pending)
<b>QoS</b>	Diffserve, TOS, 802.1 P/Q VLAN tagging
<b>IP Transport</b>	RTP/RTCP & PPPoE
<b>PSTN Signaling</b>	FXO Loop start
<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 Round-robin port scheduling to ensure available lines to access PSTN networks
<b>Media</b>	SRTP
<b>Control</b>	TLS/SIPS (pending)
<b>Management</b>	Syslog support, HTTPS and telnet (pending), remote management using Web browser
<b>Short and long haul</b>	REN3: Up to 150 ft on 24 AWG line
<b>Caller ID</b>	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID
<b>Polarity Reversal / Wink</b>	Yes
<b>EMC</b>	GXW-410x: EN55022 Class B, CFR Part 15 Class B, EN55024; GXW-4104: FCC, CE (in addition)
<b>Safety</b>	GXW-410x: EN60950-1 GXW-4108: UL60950-1 (in addition)

## HARDWARE SPECIFICATION

TABLE 4: HARDWARE SPECIFICATION OF GXW-410XV

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

## CONFIGURATION GUIDE

### CONFIGURATION WITH WEB BROWSER

The GXW-410xv has an embedded Web server that will respond to HTTP GET/POST requests. It also has embedded HTML pages that allow a user to configure the IP phone through any common web browser. Examples of GUI interfaces can be downloaded @ [http://www.grandstream.com/user\\_manuals/GUI/GUI\\_GXW410x.rar](http://www.grandstream.com/user_manuals/GUI/GUI_GXW410x.rar).

### ACCESSING THE WEB CONFIGURATION MENU

1. Connect the Power to the GXW-410xv unit.
2. Connect an Ethernet cable between the LAN port on GXW-410xv to your PC.
3. You will have to assign a dummy IP with the same subnet as the GXW IP Address, which is 192.168.0.160 by default. So, set an IP address like 192.168.0.x for your PC.
4. Launch web browser and type <http://192.168.0.160> at address of web browser. This connects you to the GXW-410xv web server.

You may choose to use DHCP or PPPoE connection or another static IP address according to your local network environment.

The Gateway Web Configuration Menu can be then accessed by the following URI: <http://Gateway-IP-Address> where the *Gateway-IP-Address* is the IP address of the Gateway.

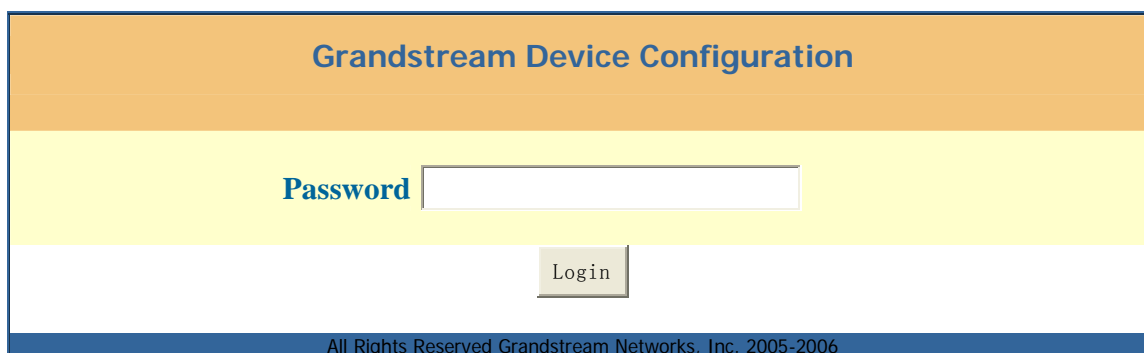
**NOTE:** To access the configuration page, type the GXW IP address into the browser, stripping out the leading "0" because the browser will parse in octet. **e.g.** if the IP address is: 192.168.001.014, please type in: 192.168.1.14.

## END USER CONFIGURATION

Once this HTTP request is entered and sent from a Web browser, the GXW-410xv 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.

FIGURE 4: SCREEN-SHOT OF GXW-410XV LOG-IN SCREEN



After login, the next configuration page is the Basic Configuration page, explained in detail in **Table 6: Web Log-in Definition**.

TABLE 6: WEB LOG-IN DEFINITIONS

<b>Web Access</b>	Select HTTP or secure HTTPS protocol for Web Access
<b>Web Port</b>	By default, HTTP uses port 80 and HTTPS uses port 443. This field is for customizable web port.
<b>End User Password</b>	This contains the password to access the Web Configuration Menu. This field is case sensitive with a maximum length of 25 characters.
<b>IP Address</b>	There are two modes to operate the GXW-410xv: <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 GXW-410xv acquires its IP address from the first DHCP server it discovers from the LAN it is connected. <i>Using the PPPoE feature:</i> set the PPPoE account settings. The GXW-410xv will establish a PPPoE session if any of the PPPoE fields is set. <b>Static IP mode:</b> configure the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields. These fields are set to zero by default.
<b>Time Zone</b>	Controls how the date/time is displayed according to the specified time zone.

You may also access the Device Status page which provides details of the GXW product. The Device Status page terms are defined in **Table 7: Status Page Definitions**.

TABLE 7: STATUS PAGE DEFINITIONS

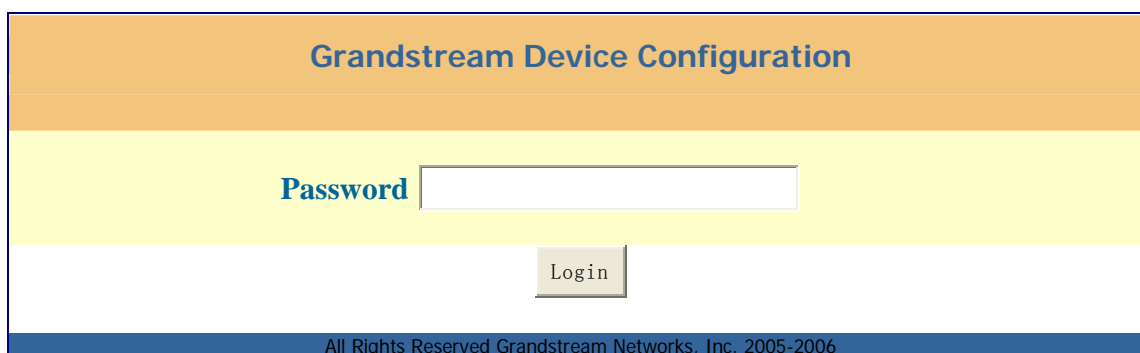
<b>Hardware Revision</b>	Hardware version number: Main Board, Interface Board
<b>MAC Address</b>	The device ID in HEX format. This is a very important ID for ISP troubleshooting.
<b>IP Address</b>	This field shows LAN IP address of GXW-410xv
<b>Product Model</b>	This field contains the product model info.
<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 Channels are 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
<b>PPPoE Link Up</b>	This field shows whether the PPPoE connection is running if connected to DSL modem.
<b>Detected NAT Type</b>	This field shows what kind NAT the GXW-410xv is connected to via its LAN port. It is based on STUN protocol.

## ADVANCED USER SETTINGS

### ADVANCED USER CONFIGURATION

The end-user needs to login to the advanced user configuration page the same way as for the basic configuration page.

**FIGURE 5: SCREENSHOT OF ADVANCED USER CONFIGURATION**



The screenshot shows a web interface titled "Grandstream Device Configuration". It features a yellow background for the main content area. In the center, there is a text label "Password" followed by a white input field. Below the input field is a "Login" button. At the bottom of the page, there is a footer that reads "All Rights Reserved Grandstream Networks, Inc. 2005-2006".

Advanced User configuration includes the end user configuration and advanced configurations including: SIP configuration, Codec selection, NAT Traversal Setting and other miscellaneous configuration.

**TABLE 8: ADVANCED CONFIGURATION PAGE DEFINITIONS**

<b>Admin Password</b>	Administrator password. Only the administrator can configure the “Advanced Settings” page. Password field is purposely left blank for security reasons. The maximum password length is 25 characters.
<b>g723 Rate</b>	G723 encoding rate (6.3kbps or 5.3kbps)
<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.
<b>Layer 2 QoS</b>	This contains the value used for layer 2 VLAN tag. Default setting is blank.
<b>Inter Digit Timeout</b>	Default is 4 seconds.
<b>Local RTP port</b>	This parameter defines the local RTP-RTCP port pair the GXW-410xv 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. The default value is 5004.
<b>Use Random Port</b>	This parameter, when set to Yes, will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple GXW-410xvs are behind the same NAT.
<b>Keep-alive interval</b>	This parameter specifies how often the GXW-410xv sends a blank UDP packet to the SIP server in order to keep the “hole” on the NAT open. Default is 20 seconds.
<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 server.
<b>Firmware Upgrade &amp; Provisioning</b>	This radio button will enable GXW-410xv to download firmware or configuration file through either TFTP or HTTP.
<b>Via TFTP Server</b>	<p>This is the IP address of the configured TFTP server. If selected and it is non-zero or not blank, 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><u>Note:</u> 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 15 or 20 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/1.0.0.36 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><u>Note:</u> If Auto Upgrade is set to No, GXW-410xv will only do HTTP download once at boot up.</p>

<b>Automatic Upgrade</b>	Choose <b>Yes</b> to enable automatic upgrade and provisioning. In “Check for new firmware every” field, enter the number of days to enable GXW-410xv to check the server for firmware upgrade or configuration in the defined period of days. When set to No, GXW-410xv will only do upgrade once at boot up. “Always check for New Firmware.” Check New Firmware only when F/W pre/suffix changes”
<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>	Select the ATA 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: <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:</p> <p>GS_LOG: [device MAC address][error code] error message</p> <p>Here is an example: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up</p>
<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>Enable Video Surveillance</b>	Work in progress.
<b>RTSP Port</b>	By default it is 554.



## CONFIGURING THE FXO CHANNELS

Configuring the FXO channels on the GXW – 410xv is an easy process. Follow the GUI interfaces. The Device Status page terms are defined in **Table 9: 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 9.

**TABLE 9: FXO LINES CONFIGURATION DEFINITIONS**

<b>Enable Current Disconnect</b>	When set to Y, Current Disconnect is enabled. Certain PSTN Cos require this to be enabled, in order to realize correct disconnect for PSTN side. Default it Y.
<b>AC Termination Impedance</b>	Selects the impedance of the analog Line connected to the FXO port on the GXW-410xv.
<b>Silence Timeout</b>	Terminate call after long silence detected. Default is 60 seconds, max 65536.
<b>DTMF Digit Length</b>	Default value is 100ms.
<b>DTMF Digit Volume</b>	Default value is -11dB.
<b>DTMF Dial Pause</b>	Default value is 100ms.
<b>Wait Dial-tone</b>	It is recommended to set this to Y in case Dialing stage is set to 1.
<b>Dialing Stage</b>	Dialing stage can be set to 1 or 2. <b>Note:</b> When set to 1, the Server needs to be configured to allow forwarding and receipt of SIP messages from GXW IP address directly.
<b>Off-hook Auto Dial</b>	This parameter allows users to configure a User ID or extension number to be automatically dialed upon off-hook. Please note only the user part of the SIP address needs to be entered here. The GXW-410xv will automatically append the '@' and host portion of the corresponding SIP address.

**FIGURE 6: SCREEN-SHOT OF GXW-410XV CHANNEL DIALING**

Channel Dialing	
1. DTMF Digit Length(X10ms):	ch1-8:10; (1-200, default 10)
2. DTMF Digit Volume(dB):	ch1-8:-11; (-31-0, default -11)
3. DTMF Dial Pause(X10ms):	ch1-8:10; (1-200, default 10)
4. Wait Dial-Tone(Y/N):	ch1-8:Y; (default Yes)
5. Dialing Stage(1/2):	ch1-8:2; (default 2)
6. Off-hook Auto Dial(VoIP):	ch1-8:605;
Update	

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**TABLE 10: CHANNELS PAGE DEFINITIONS**

<b>SIP User ID</b>	User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
<b>Authentication ID</b>	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
<b>Authentication Password</b>	SIP service subscriber's account password for GXW-410xv to register to (SIP) servers of ITSP.
<b>Profile ID</b>	Select the corresponding Profile ID (1/2/3)
<b>Call Progress Tones</b>	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. <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>
<b>Channel Voice Settings</b>	Channel voice settings mentioned below.
<b>Tx to PSTN Audio Gain (dB)</b>	Allows user to set a value in dB for transmission to PSTN Audio Gain.
<b>Rx from PSTN Audio Gain (dB)</b>	Allows user to set a value in dB for receive from PSTN Audio Gain.
<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>Channel specific Setting</b>	Channel specific settings mentioned below.
<b>DTMF Method</b>	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. <ul style="list-style-type: none"> <li>1 – in-audio</li> <li>2 – RFC2833</li> <li>3 – in-audio and RFC2833</li> <li>4 – SIP Info</li> <li>5 – in-audio and RFC2833</li> <li>6 – SIP Info and RFC2833</li> <li>7 – in-audio, RFC2833, and SIP Info</li> </ul>
<b>No Key Entry Timeout</b>	Default is 4 seconds.

**TABLE 11: GRANDSTREAM CONFIGURATION PAGE DEFINITIONS**

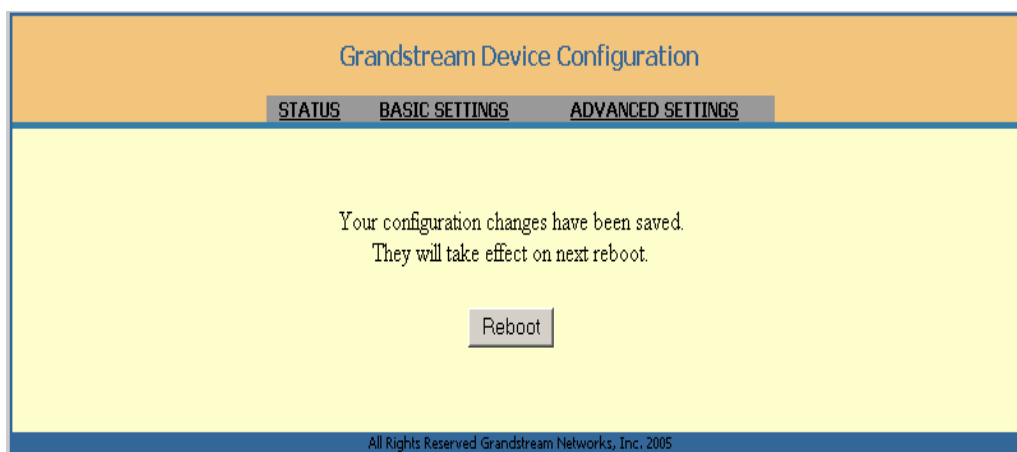
<b>Activate Profile</b>	When set to Yes the SIP Profile is activated.
<b>Profile Name</b>	A name to identify a Profile.
<b>SIP Server</b>	SIP Server's IP address or Domain name provided by VoIP service provider.
<b>Outbound Proxy</b>	IP address or Domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by GXW-410xv 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>Use DNS SRV:</b>	Default is No. If set to Yes the client will use DNS SRV to look up server.
<b>User ID is Phone Number</b>	If the GXW-410xv 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>SIP Registration</b>	This parameter controls whether the GXW-410xv needs to send REGISTER messages to the proxy 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 GXW-410xv 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>Local SIP port</b>	This parameter defines the local SIP port the GXW-410xv will listen and transmit. The default value for Account 1 is 5060. It is 5062, 5064, 5066 for Account 2, Account 3 and Account 4 respectively.
<b>NAT Traversal</b>	This parameter defines whether the GXW-410xv NAT traversal mechanism will be activated or not. If activated (by choosing "Yes") and a STUN server is also specified, then the GXW-410xv will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the GXW-410xv 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 GXW-410xv 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 GXW-410xv 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>Early Dial</b>	Default is No. Use only if proxy supports 484 response.
<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>Send Anonymous</b>	If this parameter is set to “Yes”, the “From” header in outgoing INVITE message will be set to anonymous, essentially blocking the Caller ID from displaying.
<b>Preferred Vocoder</b>	The GXW-410xv 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>Special Feature</b>	Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Nortel, Broadsoft, etc.

### SAVING THE CONFIGURATION CHANGES

Once a change is made, press the “Update” button in the Configuration Menu. The GXW-410xv will display the following screen to confirm that the changes have been saved. To activate changes, reboot or power cycle the GXW-410x after all changes are made.

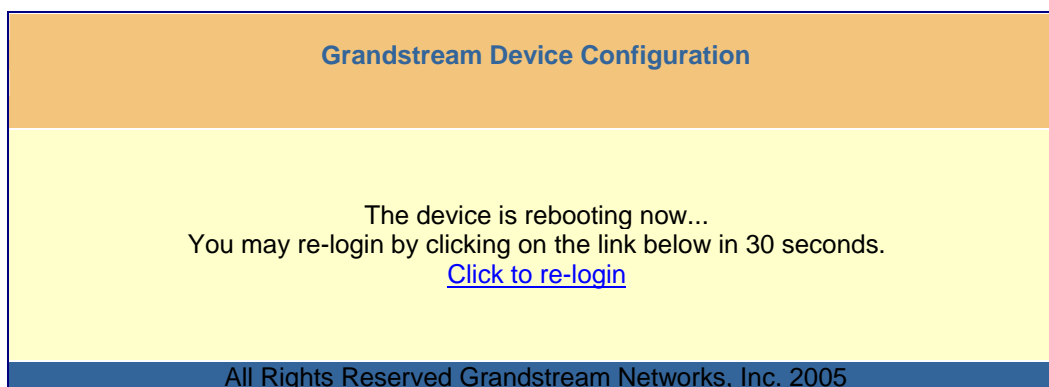
**FIGURE 7: SCREEN-SHOT OF SAVE CONFIGURATION**



### 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.

**FIGURE 8: SCREEN-SHOT OF REBOOTING**



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

## VIDEO SURVEILLANCE

The GXW-410xv can be used with an Analog Surveillance Camera to perform video surveillance function. This application should be used in a LAN environment or when both sides have public IP address.

### VIDEO SURVEILLANCE PROCEDURES

➤ **Gateway side:**

1. In the ADVANCED SETTING page, find the following field and change from default setting NO to YES, reboot the device.

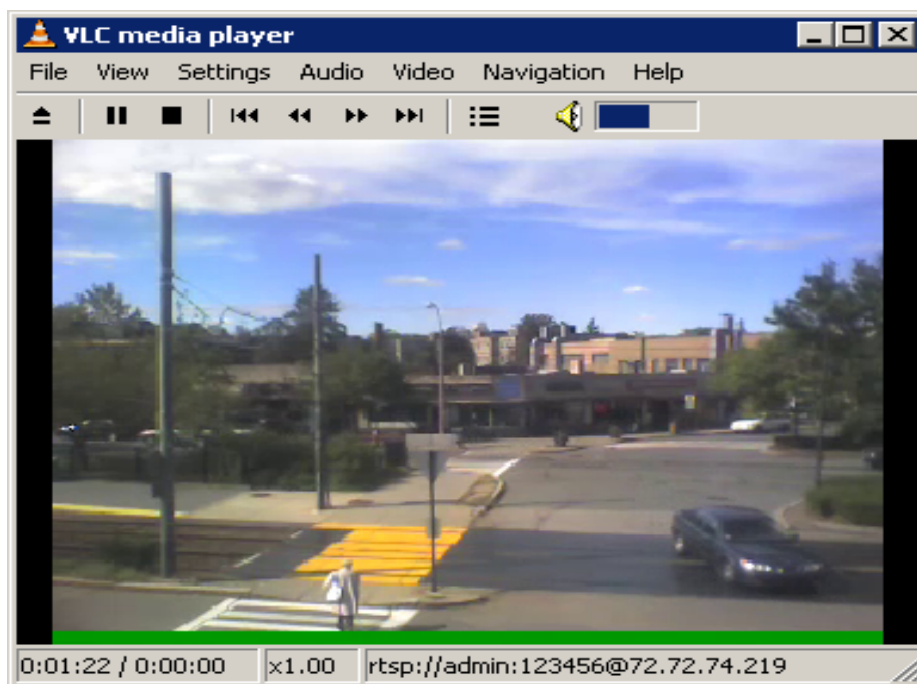
*Enable Video Surveillance:*    No    Yes

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 works and support 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 9: SCREEN-SHOT OF VIDEO SURVEILLANCE\*



\* PC client side running VLC as monitoring station

## FIRMWARE UPGRADE

Our latest official release can be downloaded from: <http://www.grandstream.com/firmware.html>.

Firmware (or software) upgrades can be done either via TFTP or HTTP. The corresponding configuration settings are on the configuration page. End users should NOT touch the configuration settings that are useful for ITSPs. To upgrade your unit firmware, follow these steps:

1. Under Advanced Settings webpage, enter your TFTP or HTTP Server IP address (or FQDN) next to the “Firmware Upgrade: Upgrade Server” field.
2. Select via TFTP or HTTP accordingly.
3. If you plan to use Automatic Upgrade, set it to “Yes”, otherwise No (this will make it check for upgrade every time you reboot).

### UPGRADE THROUGH HTTP

To upgrade firmware via HTTP, the field “*Firmware Upgrade and Provisioning: Upgrade Via*” needs to be set to HTTP. The “*Firmware Server Path*” should be set to where the firmware files are located.

For example, the user can use the following URL in the Firmware Server Path: [firmware.mycompany.com:6688/Grandstream/1.0.0.29](http://firmware.mycompany.com:6688/Grandstream/1.0.0.29) where [firmware.mycompany.com](http://firmware.mycompany.com) is the FQDN of the HTTP server. It can also be in IP address format. “:6688” is the TCP port the HTTP server listening to, default http server listens to port 80. “/Grandstream/1.0.0.29” is the RELATIVE directory to the root dir on HTTP web server.

### UPGRADE THROUGH TFTP

To upgrade firmware via TFTP, set the field “*Firmware Upgrade and Provisioning: Upgrade Via*” to TFTP. The TFTP server can be configured in either IP address format or FQDN.

To configure the TFTP server via the Web configuration interface, follow these five steps:

1. Open your browser to input the IP address of the GXW-410xv.
2. Enter the admin password to enter the configuration screen.
3. Enter the TFTP server address or URL in the “Firmware Server Path” field near the bottom of the configuration screen.
4. Once the “Firmware Server Path” is set, update the change by clicking the “Update” button.
5. Reboot or power cycle the unit.

If the configured updating server is found and a new code image is available, the GXW-410xv will retrieve the new image files by downloading them into the GXW-410xv’s SRAM. During this stage, the GXW-410xv’s LED will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will be saved into the Flash. If TFTP fails for any reason (e.g., TFTP server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the GXW-410xv will stop the TFTP process and simply boot using the existing code image in the flash.

Firmware upgrading may take as long as 20 minutes over the Internet, or just 20+ seconds if it is performed on a LAN. Grandstream recommends conducting firmware upgrades in a controlled LAN environment if possible.



## **No LOCAL TFTP SERVER**

For users who do not have a local TFTP server, Grandstream provides a NAT-friendly TFTP server on the public Internet for users to download the latest firmware upgrade automatically. Please check the Services section of Grandstream's Web site to obtain this TFTP server IP address. Alternatively, user can download and install a free TFTP or HTTP server in his LAN for a firmware upgrade.

A free Windows version TFTP server can be downloaded from:

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

### **Directions For Downloading TFTP Server:**

1. Unzip the file and put all of the files under the root directory of the TFTP server.
2. Put the PC running the TFTP server and the GXW-410x in the same LAN segment.
3. 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.

You can also download the free HTTP server from <http://httpd.apache.org> or just use Microsoft IIS web.

## 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.

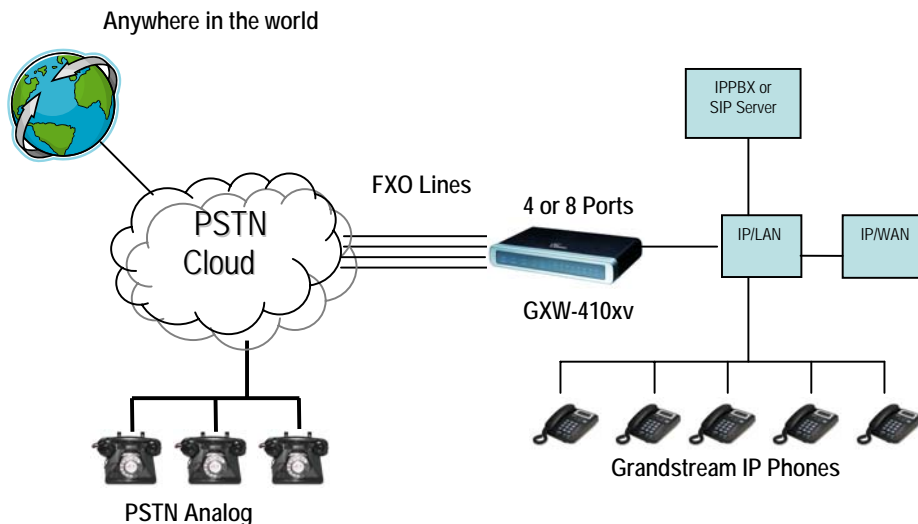
The ONLY way to restore default factory settings is as follows:

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. Enter a needle like object in this hole and keep it pressed for about 7 seconds.
4. You will see the LAN port LEDs (green and orange) go off and on simultaneously; this indicates the reset went through.
5. All settings have been erased and the gateway is back to factory settings.

## Examples of GXW-410xv Configurations

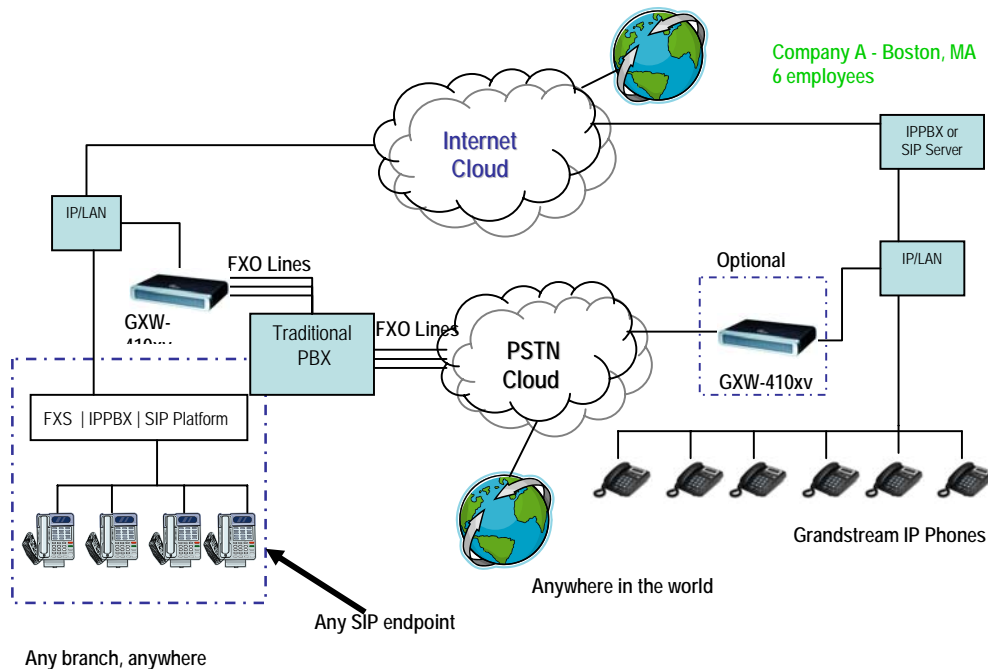
### APPLICATION ONE: GXW CONNECTED WITH AN IP-PBX OR SIP SERVER

**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).



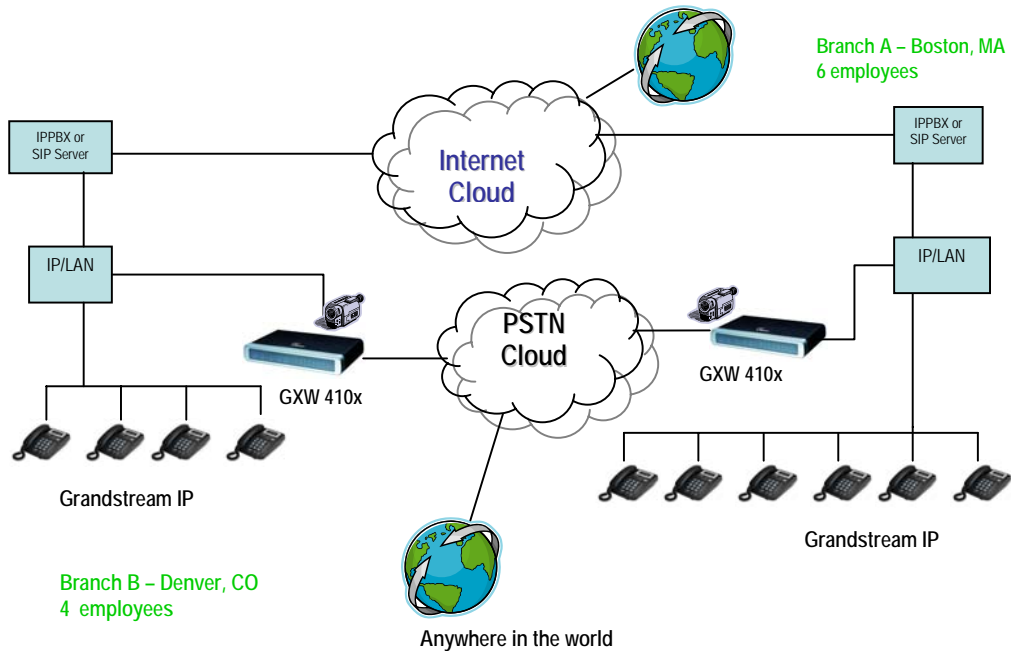
### APPLICATION TWO: GXW TO EXTEND A TRADITIONAL PBX SCENARIO

**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)



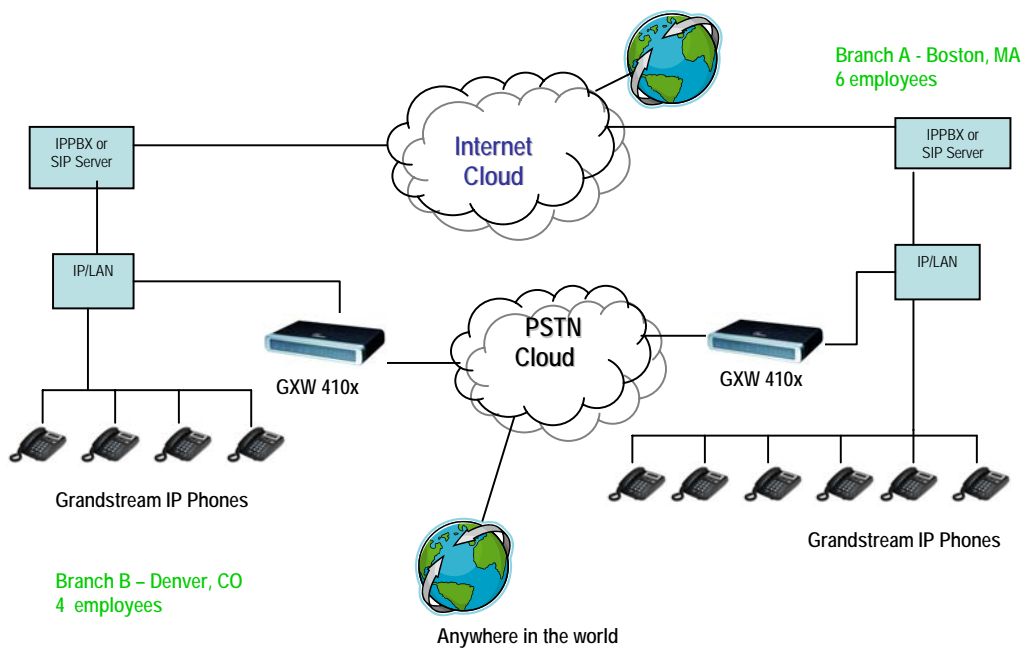
### APPLICATION THREE: GXW CONNECTED WITH AN IP-PBX OR SIP SERVER AND VIDEO SURVEILLANCE

**Scenario:** The GXW-410xv offers an additional video surveillance port which can be configured for surveillance. It is the only small business analog gateway that offers this security feature.



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

**Scenario Four:** The GXW-410xv offers an IP to IP pure IP Communications System configuration, where all locations use IP phones.



## GLOSSARY OF TERMS

**ADSL** Asymmetric Digital Subscriber Line: Modems attached to twisted pair copper wiring that transmit from 1.5 Mbps to 9 Mbps downstream (to the subscriber) and from 16 kbps to 800 kbps upstream, depending on line distance.

**AGC** Automatic Gain Control is an [electronic system](#) found in many types of devices. Its purpose is to control the [gain](#) of a system in order to maintain some measure of performance over a changing range of real world conditions.

**ARP** Address Resolution Protocol is a protocol used by the [Internet Protocol \(IP\) \[RFC826\]](#), specifically IPv4, to map [IP network addresses](#) to the hardware addresses used by a data link protocol. The protocol operates below the network layer as a part of the interface between the OSI network and OSI link layer. It is used when [IPv4 is used over Ethernet](#)

**ATA** Analogue Telephone Adapter. Covert analogue telephone to be used in data network for VoIP, like Grandstream HT series products.

**CODEC** Abbreviation for Coder-Decoder. It's an analog-to-digital (A/D) and digital-to-analog (D/A) converter for translating the signals from the outside world to digital, and back again.

**CNG** Comfort Noise Generator, generate artificial background [noise](#) used in [radio](#) and [wireless](#) communications to fill the [silent](#) time in a transmission resulting from [voice activity detection](#).

**DATAGRAM** A data packet carrying its own address information so it can be independently routed from its source to the destination computer

**DECIMATE** To discard portions of a signal in order to reduce the amount of information to be encoded or compressed. Lossy compression algorithms ordinarily decimate while sub-sampling.

**DECT** Digital Enhanced Cordless Telecommunications: A standard developed by the European Telecommunication Standard Institute from 1988, governing pan-European digital mobile telephony. DECT covers wireless PBXs, telepoint, residential cordless telephones, wireless access to the public switched telephone network, Closed User Groups (CUGs), Local Area Networks, and wireless local loop. The DECT Common Interface radio standard is a multi-carrier time division multiple access, time division duplex (MC-TDMA-TDD) radio transmission technique using ten radio frequency channels from 1880 to 1930 MHz, each divided into 24 time slots of 10ms, and twelve full-duplex accesses per carrier, for a total of 120 possible combinations. A DECT base station (an RFP, Radio Fixed Part) can transmit all 12 possible accesses (time slots) simultaneously by using different frequencies or using only one frequency. All signaling information is transmitted from the RFP within a multi-frame (16 frames). Voice signals are digitally encoded into a 32 Kbit/s signal using Adaptive Differential Pulse Code Modulation.

**DNS** Short for *Domain Name System* (or *Service* or *Server*), an [Internet](#) service that translates [domain names](#) into IP addresses

**DID** Direct Inward Dialing. The ability for an outside caller to dial to a PBX extension without going through an attendant or auto-attendant.

**DSP** Digital Signal Processor. A specialized CPU used for digital signal processing. Grandstream products all have DSP chips built inside.

**DTMF** Dual Tone Multi Frequency. The standard tone-pairs used on telephone terminals for dialing using in-band signaling. The standards define 16 tone-pairs (0-9, #, \* and A-F) although most terminals support only 12 of them (0-9, \* and #).

**FQDN** Fully Qualified Domain Name. A FQDN consists of a host and domain name, including top-level domain. For example, [www.grandstream.com](http://www.grandstream.com) is a fully qualified domain name. www is the host, Grandstream is the second-level domain, and .com is the top level domain.

**FXO** Foreign eXchange Office. An FXO device can be an analog phone, answering machine, fax, or anything that handles a call from the telephone company like AT&T. They should also operate the same way when connected to an FXS interface.

- An FXO interface will accept calls from FXS or PSTN interfaces. All countries and regions have their own standards.
- FXO is complimentary to FXS (and the PSTN).

**FXS** Foreign eXchange Station. An FXS device has hardware to generate the ring signal to the FXO extension (usually an analog phone).

- An FXS device will allow any FXO device to operate as if it were connected to the phone company. This makes your PBX the POTS+PSTN for the phone.
- The FXS Interface connects to FXO devices (by an FXO interface, of course).

**DHCP** The *Dynamic Host Configuration Protocol* (DHCP) is an Internet protocol for automating the configuration of computers that use TCP/IP. DHCP can be used to automatically assign IP addresses, to deliver TCP/IP stack configuration parameters such as the subnet mask and default router, and to provide other configuration information such as the addresses for printer, time and news servers.

**ECHO CANCELLATION** Echo Cancellation is used in [telephony](#) to describe the process of removing [echo](#) from a voice communication in order to improve voice quality on a [telephone call](#). In addition to improving quality, this process improves [bandwidth](#) savings achieved through [silence suppression](#) by preventing echo from traveling across a [network](#). There are **two types** of echo of relevance in telephony: acoustic echo and hybrid echo. [Speech compression](#) techniques and [digital processing](#) delay often contribute to echo generation in [telephone networks](#).

**H.323** A suite of standards for multimedia conferences on traditional packet-switched networks.

**HTTP** Hyper Text Transfer Protocol; the World Wide Web protocol that performs the request and retrieve functions of a server

**IP** Internet Protocol. A packet-based protocol for delivering data across networks.

**IP-PBX** IP-based Private Branch Exchange

**IP Telephony** (Internet Protocol telephony, also known as Voice over IP Telephony) A general term for the technologies that use the Internet Protocol's packet-switched connections to exchange voice, fax, and other forms of information that have traditionally been carried over the dedicated circuit-switched connections of the public switched telephone network (PSTN). The basic steps involved in originating an IP Telephony call are conversion of the analog voice signal to digital format and compression/translation of the signal into Internet protocol (IP) packets for transmission over the Internet or other packet-switched networks; the process is reversed at the receiving end. The terms IP Telephony and Internet Telephony are often used to mean the same; however, they are not 100 per cent interchangeable, since Internet is only a subcase of packet-switched networks. For users who have free or fixed-price Internet access, IP Telephony software essentially provides free telephone calls anywhere in the world. However, the challenge of IP Telephony is maintaining the quality of service expected by subscribers. Session border controllers resolve this issue by providing quality assurance comparable to legacy telephone systems.

**IVR** IVR is a software application that accepts a combination of voice telephone input and touch-tone keypad selection and provides appropriate responses in the form of voice, fax, callback, e-mail and perhaps other media.

**MTU** A Maximum Transmission Unit (MTU) is the largest size [packet](#) or [frame](#), specified in [octets](#) (eight-bit bytes), that can be sent in a packet- or frame-based network such as the Internet. The maximum for Ethernet is 1500 byte.

**NAT** Network Address Translation

**NTP** Network Time Protocol, a protocol to exchange and synchronize time over networks. The port used is UDP 123. Grandstream products use NTP to get time from Internet.

**OBP/SBC** Outbound Proxy or another name Session Border Controller. A device used in [VoIP](#) networks. OBP/SBCs are put into the signaling and media path between calling and called party. The OBP/SBC acts as if it was the called VoIP phone and places a second call to the called party. The effect of this behavior is that not only the signaling traffic, but also the media traffic (voice, video etc) crosses the OBP/SBC. Without an OBP/SBC, the media traffic travels directly between the VoIP phones. Private OBP/SBCs are used along with [firewalls](#) to enable VoIP calls to and from a protected enterprise network. Public VoIP service providers use OBP/SBCs to allow the use of VoIP protocols from private networks with [internet](#) connections using [NAT](#).

**PPPoE** Point-to-Point Protocol over Ethernet is a network protocol for encapsulating PPP frames in Ethernet frames. It is used mainly with cable modem and DSL services.

**PSTN** Public Switched Telephone Network. The phone service we use for every ordinary phone call, or called POT (Plain Old Telephone), or circuit switched network.

**RTCP** Real-time Transport Control Protocol, defined in [RFC 3550](#), a sister protocol of the [Real-time Transport Protocol](#) (RTP), It partners RTP in the delivery and packaging of multimedia data, but does not transport any data itself. It is used periodically to transmit control packets to participants in a streaming multimedia session. The primary function of RTCP is to provide feedback on the quality of service being provided by RTP.

**RTP** Real-time Transport Protocol defines a standardized packet format for delivering audio and video over the Internet. It was developed by the Audio-Video Transport Working Group of the [IETF](#) and first published in 1996 as [RFC 1889](#).

**SDP** Session Description Protocol is a format for describing [streaming media](#) initialization parameters. It has been published by the [IETF](#) as [RFC 2327](#).

**SIP** Session Initiation Protocol, An IP telephony signaling protocol developed by the IETF (RFC3261). SIP is a text-based protocol suitable for integrated voice-data applications. SIP is designed for voice transmission and uses fewer resources and is considerably less complex than H.323. All Grandstream products are SIP based.

**STUN** Simple Traversal of UDP over NATs is a [network protocol](#) allowing clients behind [NAT](#) (or multiple NATs) to find out its public address, the type of NAT it is behind and the internet side port associated by the NAT with a particular local port. This information is used to set up UDP communication between two hosts that are both behind NAT routers. The protocol is defined in [RFC 3489](#). STUN will usually work well with non-symmetric NAT routers.

**TCP** Transmission Control Protocol is one of the core protocols of the [Internet protocol suite](#). Using TCP, applications on networked hosts can create *connections* to one another, over which they can exchange data or [packets](#). The protocol guarantees reliable and in-order delivery of sender to receiver data.

**TFTP** Trivial File Transfer Protocol, is a very simple [file transfer protocol](#), with the functionality of a very basic form of [FTP](#); It uses [UDP](#) (port 69) as its [transport protocol](#).

**UDP** User Datagram Protocol (UDP) is one of the core protocols of the [Internet protocol suite](#). Using UDP, programs on networked computers can send short messages known as [datagrams](#) to one another. UDP does not provide the reliability and ordering guarantees that [TCP](#) does; datagrams may arrive out of order or go missing without notice. However, as a result, UDP is faster and more efficient for many lightweight or time-sensitive purposes.

**VAD** Voice Activity Detection or Voice Activity Detector is an algorithm used in [speech processing](#) wherein, the presence or absence of human speech is detected from the audio samples.

**VLAN** A virtual [LAN](#), known as a VLAN, is a logically-independent [network](#). Several VLANs can co-exist on a single physical [switch](#). It is usually refer to the [IEEE 802.1Q](#) tagging protocol.

**VoIP** Voice over the Internet. VoIP encompasses many protocols. All the protocols do some form of signaling of call capabilities and transport of voice data from one point to another. e.g.: SIP, H.323, etc.