Grandstream Networks, Inc.

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



GXW-410x User's Manual
Firmware Version 1.0.0.32

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GXW-410X USER'S MANUAL

1.0 Welcome

Thank you for purchasing the Grandstream GXW-410x Analog FXO IP Gateway. You've made an excellent choice and we believe you will find the GXW-410x easy to use and easy to manage.

Gateway GXW-410x Overview

The GXW-410x offers an easy to manage, easy to configure IP communications solution for any business with virtual and/or branch locations. The GXW-410x is a next generation IP voice and video gateway that features full interoperability with leading IP-PBXs, SoftSwitches and SIP platforms.

The Analog Gateway GXW-410x series converts SIP/RTP IP calls to traditional PSTN calls. There are two models - the GXW-4104 and GXW-4108, which have either 4 and 8 FXO ports respectively. The installation is the same for either model. Either model can be configured to work with any leading SIP server, for a pure media gateway to access PSTN networks.

This document is subject to changes without notice. The latest electronic version of this user manual is available for download from the following location: http://www.grandstream.com/user_manuals/GXW-410x.pdf

2.0 Packaging & Installation

2.1. Packaging

The GXW-410x package includes:

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

Safety Compliances

The GXW-410x 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-410x 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.

2.2 Installation

1. Connect power and cables
2. Connect Ethernet cable to a hub and power adapter to power wall outlet
3. Assign GXW-410x IP Address
4. Use another Ethernet cable to connect your PC to the same hub that GXW-4100 is connected.
5. Assign your PC to a static IP that is within the same subnet of default IP address of GXW-4100 192.168.0.160. i.e. 192.168.0.x. X can be any number between 2 and 254 except 160.
6. Launch web browser and type http://192.168.0.160 at Address of web browser. This will let you to connect to build-in web server of GXW-4100.

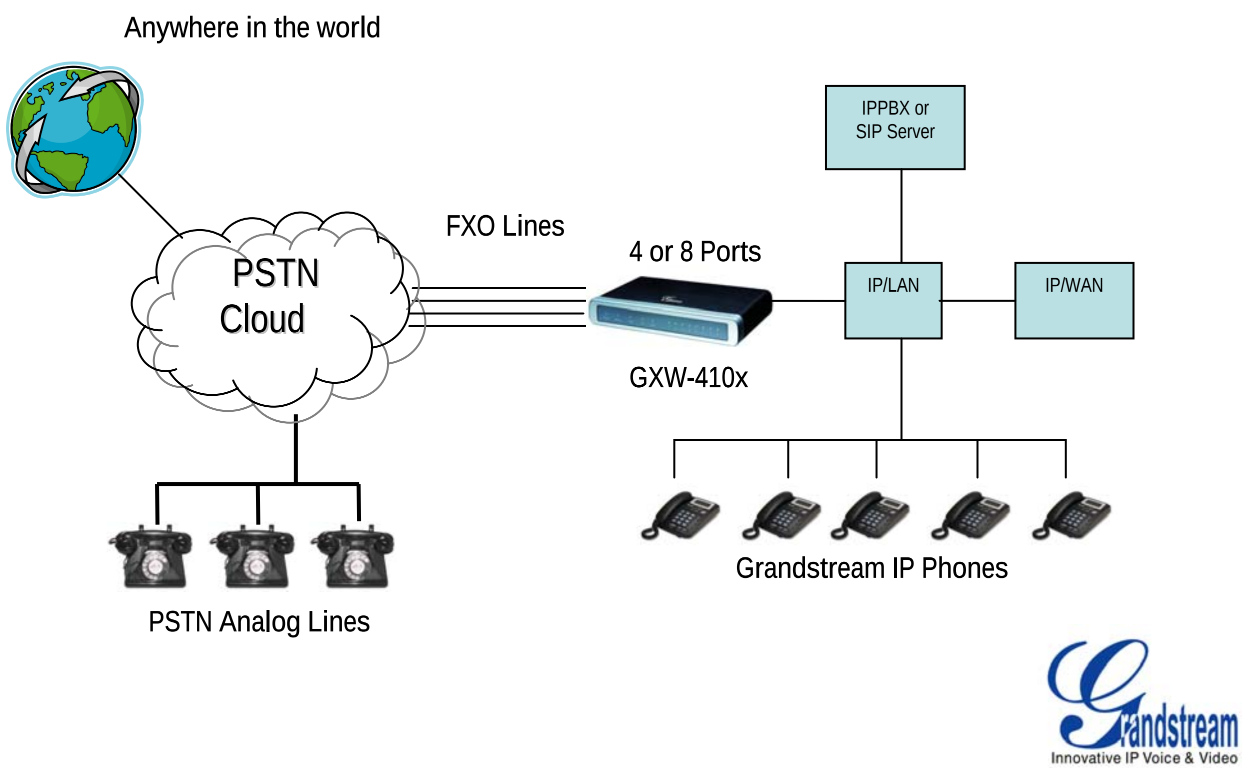
You may change to use DHCP or PPPoE connection or another static IP address according to your local network environment.

3.0 Application Description

A SIP proxy server such as Asterisk or a SIP registrar server can be deployed with the GXW-410x series. In this environment, the SIP server handles SIP registration and call control and the GXW-410x processes media conversion between IP and PSTN calls. By design, the system supports the North American call progress tones and signaling standards on PSTN sides.

Figure 1 illustrates how the GXW-410x can be deployed in a typical enterprise topology.

Figure 1: Functional Diagram of IP-PBX (Asterisk) & GXW-410x



Simple Configuration: Media Gateway to Access PSTN Networks

GXW-410x 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-410xgateway 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-410x.

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-410x.

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-410x 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-410x 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-410x 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-410x 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-410x

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.

Please see the *Quick Install Guide* at www.grandstream.com.

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 those values and users don't know what the correct values are, please use the default values. Contact product support if you still have questions about configuring your GXW-410x.*

4.0 Features

The Gateway series includes the GXW-4104 and GXW-4108, each offering superb voice and video quality, traditional telephony functionality, easy deployment, and 4 or 8 FXO ports respectively. The GXW-410x is the only small business gateway that offers a video surveillance component in the industry. A complete list of features can be found at www.grandstream.com.

4.1 Software Features:

- TFTP and HTTP firmware upgrade support
- Multiple SIP accounts (8), Eight total number of physical line ports, each account corresponding to one SIP user id and belonging to one of multiple SIP profile
- Multiple SIP profiles, max of 3 profiles per system. Each profile hosts 0 to multiple number of SIP accounts, depending on user need
- One stage and two stage dialing
- VoIP to PSTN call setup and teardown
- PSTN to VoIP call setup and teardown
- Configurable Channel Voice Settings. [Voice volume (db), Audio input (db), Silence suppression, Line echo cancellation]
- Codec Support: G711u, G711a, G729, G723, and GSM (pending)
- Line echo canceller g.168 support
- Flexible DTMF transmission method (In-audio, RFC2833, and SIP Info)
- VoIP to PSTN : channel configurable for one stage or two stage dialing
- Configurable channel dialing to improve dial-out reliability [digit length (ms), digit volume (db), dial pause between digits (ms), wait for dial-tone]
- Configurable PSTN Termination [Current Disconnect, AC Termination Impedance, Busy/Re-order tones]
- Configurable Call Progress/Termination tones via Pattern Matching

4.2 Hardware Specification

Figure 2: Hardware Specification of GXW-410x

	GXW-410x
LAN interface	2xRJ45 10/100Base-T
LED	8 LEDs in RED color
Universal Switching Power Adaptor	Input: 100-240V AC, 50/60Hz, 0.5A Max Output: 12V DC, 1.25A **UL** certified
Dimension	230mm (L) 134mm (W) 32mm (H)
Weight	0.29 lbs (3.5 oz)
Temperature	32~104°F 0~40°C
Humidity	10% - 90% (non-condensing)
Compliance	FCC, CE

5.0 Configuration Guide

5.1 Configuration with Web Browser

The GXW-410x 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 a Web browser such as Microsoft's IE.

5.1.1 Accessing the Web Configuration Menu

1. Use another Ethernet cable to connect your PC to the same hub that GXW-410x is connected.
2. Assign your PC to a static IP that is within the same subnet of default IP address of GXW-410x 192.168.0.160. **i.e.** 192.168.0.x. X can be any number between 2 and 254 except 160.
3. Launch web browser and type http://192.168.0.160 at Address of web browser. This will let you to connect to build-in web server of GXW-410x.
4. You may change to use DHCP or PPPoE connection or another static IP address according to your local network environment.

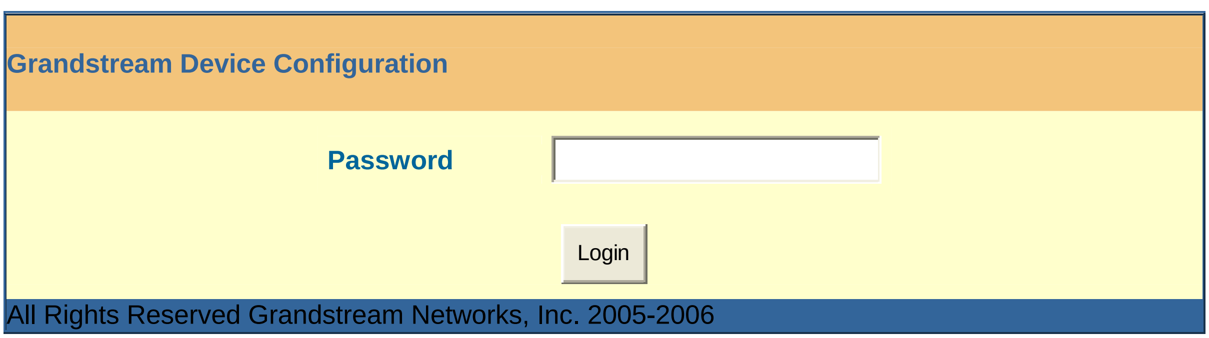
The Gateway Web Configuration Menu can be then accessed by the following URI: *http://Gateway-IP-Address* where the *Gateway-IP-Address* is the IP address of the Gateway.

NOTE: *To type IP address into browser to get into the configuration page, please strip out the leading "0" as the browser will parse in octet. e.g.: if the IP address is: 192.168.001.014, please type in: 192.168.1.14.*

5.1.2 End User Configuration

Once this HTTP request is entered and sent from a Web browser, the GXW-410x will respond with the following login screen:

Figure 3: Screen-shot of GXW-410x Log-in Screen – End User Configuration



The password is case sensitive with maximum length of 25 characters and the factory default password for End User is "123".

After a correct password is entered in the login screen, the embedded Web server will respond with the Configuration page which is explained in details below.

Figure 4: Screen-shot of GXW-410x Configuration Page

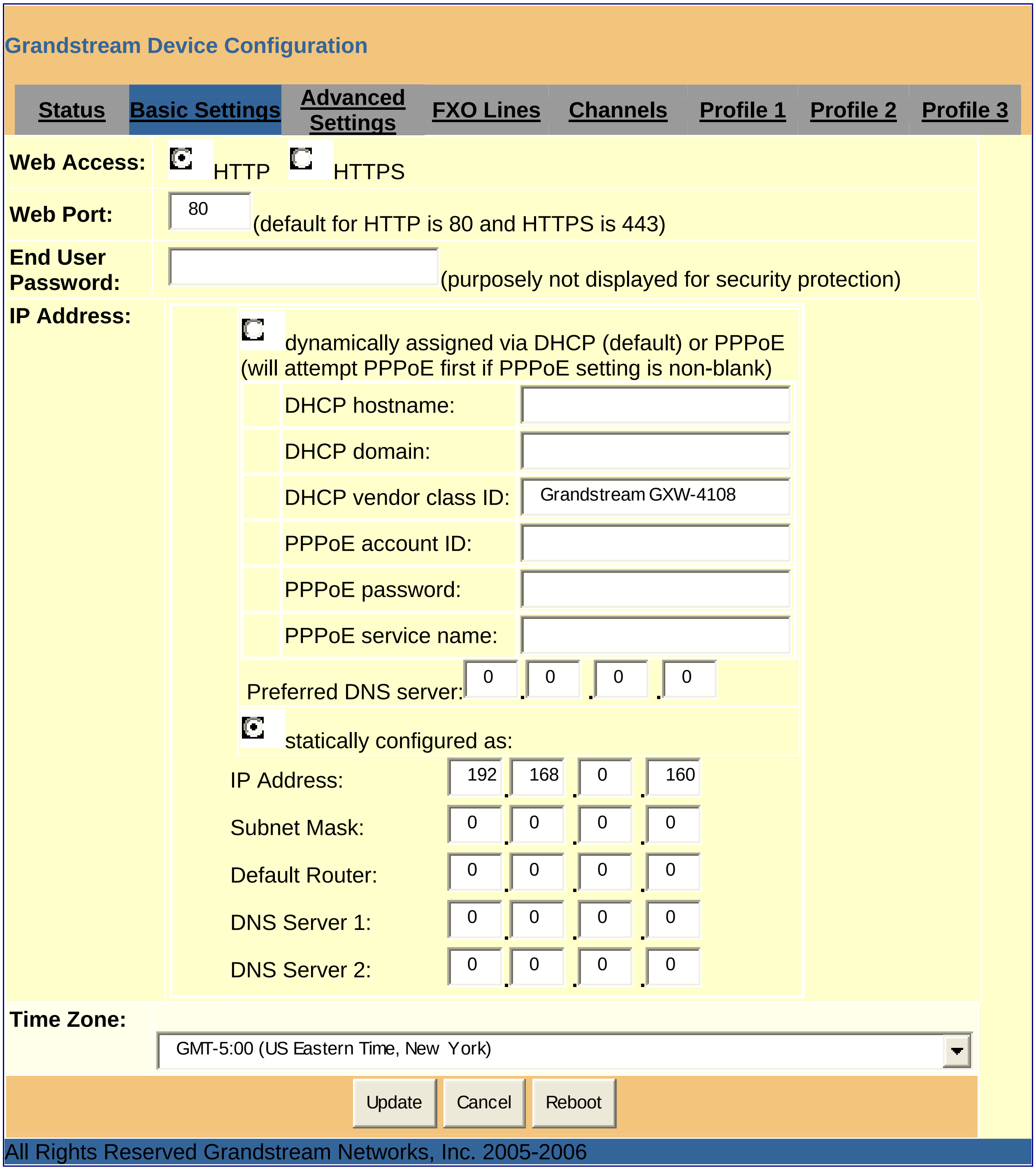


Figure 5: Web Log-in Definitions

Web Access	Select HTTP or secure HTTPS protocol for Web Access
Web Port	By default, HTTP uses port 80 and HTTPS uses port 443. This field is for customizable web port.
End User Password	This contains the password to access the Web Configuration Menu. This field is case sensitive with a maximum length of 25 characters.
IP Address	There are two modes under which the GXW-410x can operate: If DHCP mode is enabled, then all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The GXW-410x will acquire its IP address from the first DHCP server it discovers from the LAN it is connected. To use the PPPoE feature the PPPoE account settings need to be set. The GXW-410x will attempt to establish a PPPoE session if any of the PPPoE fields is set. If Static IP mode is enabled, then the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields will need to be configured. These fields are set to zero by default.
Time Zone	This parameter controls how the date/time is displayed according to the specified time zone.

In addition to the Basic Settings configuration page, the end user also has access to the Device Status page. The following is a screen shot of the device Status Page.

Figure 6: Screen-shot of GXW-410x Device Status Page

Grandstream Device Configuration	
Status / Basic Settings / Advanced Settings / FXO Lines / Channels / Profile 1 / Profile 2 / Profile 3	
Hardware Revision:	Main-- 0.1 Rev A Interface-- 0.0 Rev A
MAC Address:	00.0B.82.08.A5.19
IP Address:	72.72.74.217
Product Model:	GXW-410x
Software Version:	Program-- 1.0.0.31 Loader-- 1.0.0.3 Boot-- 1.0.0.13
System Up Time:	0 day(s) 3 hour(s) 28 minute(s)
Registered:	Phone Number 1: Yes Phone Number 2: Yes Phone Number 3: Yes Phone Number 4: Yes Phone Number 5: Yes Phone Number 6: Yes Phone Number 7: Yes Phone Number 8: Yes
PPPoE Link Up:	disabled
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Figure 7: Status Page Definitions

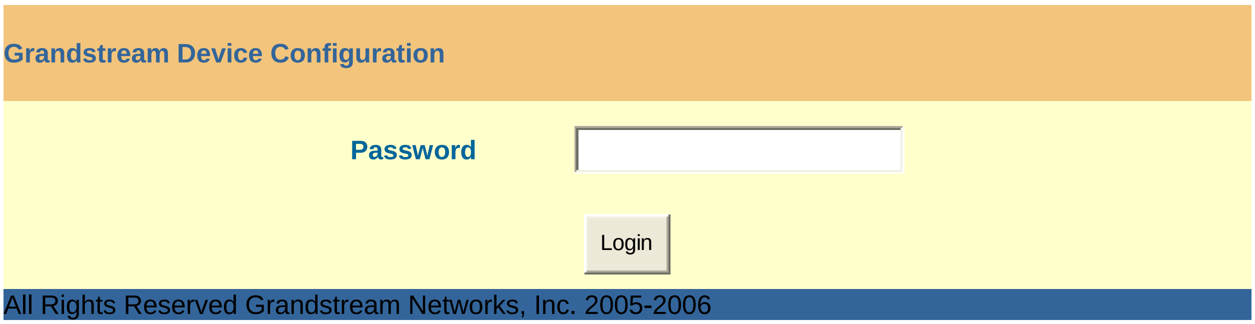
Hardware Revision	Hardware version number: Main Board, Interface Board
MAC Address	The device ID, in HEX format. This is a very important ID for ISP troubleshooting.
IP Address	This field shows LAN IP address of GXW-410x
Product Model	This field contains the product model info.
Software Version	gram: This is the main software release, its number is always used for firmware upgrade. ›tloader: This is normally not changed.
System Up Time	This field shows system up time since the last reboot.
Registered	This field indicates whether the device is registered to the SIP server(s).
PPPoE Link Up	This field shows whether the PPPoE connection is up if connected to DSL modem.
Detected NAT Type	This field shows what kind NAT the GXW-410x is connected to via its LAN port. It is based on STUN protocol.

5.2 Advanced User Settings

5.2.1 Advanced User Configuration

To login to the Advanced User Configuration page, please follow the instructions in section 5.2.1 to get to the following login page. The password is case sensitive with a maximum length of 25 characters and the factory default password for Advanced User is “admin”.

Figure 8: Screenshot of Advanced User Configuration



Advanced User configuration includes not only the end user configuration, but also advanced configuration such as SIP configuration, Codec selection, NAT Traversal Setting and other miscellaneous configuration.

Figure 8: Screenshot of Advanced Configuration Page

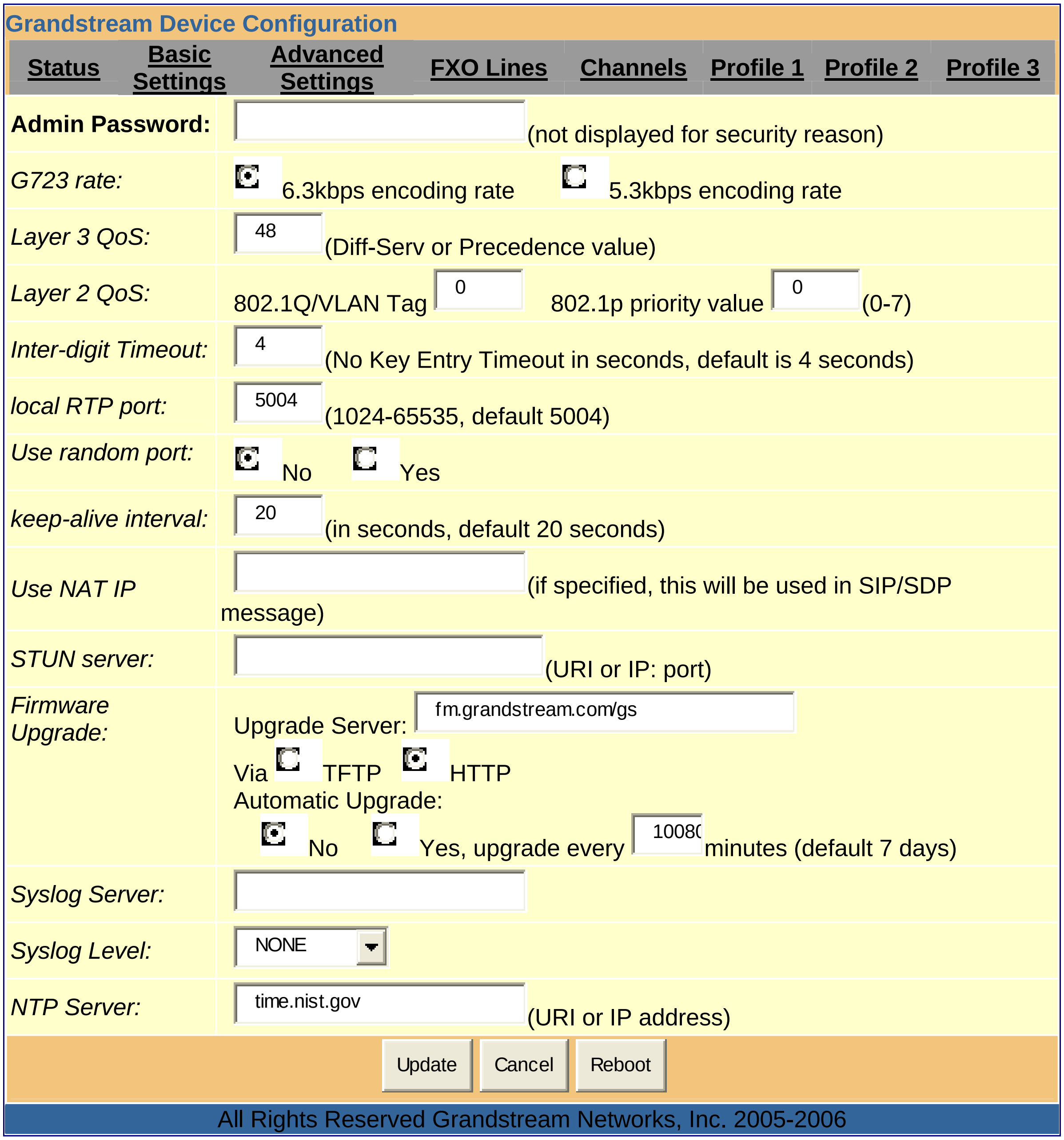


Figure 9: Advanced Configuration Page Definitions

Admin Password	Administrator password. Only the administrator can configure the "Advanced Settings" page. Password field is purposely left blank for security reason after clicking update and saved. The maximum password length is 25 characters.
g723 Rate	g723 encoding rate (6.3kbps or 5.3kbps)
Layer 3 QoS	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.
Layer 2 QoS	This contains the value used for layer 2 VLAN tag. Default setting is blank.
Inter Digit Timeout	Default is 4 seconds.
Local RTP port	This parameter defines the local RTP-RTCP port pair the GXW-410x 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.
Use Random Port	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-410xs are behind the same NAT.
Keep-alive interval	This parameter specifies how often the GXW-410x sends a blank UDP packet to the SIP server in order to keep the "hole" on the NAT open. Default is 20 seconds.
Use NAT IP	NAT IP address used in SIP/SDP message. Default is blank.
STUN Server	IP address or Domain name of the STUN server.
Firmware Upgrade & Provisioning	This radio button will enable GXW-410x to download firmware or configuration file through either TFTP or HTTP.
Via TFTP Server	This is the IP address of the configured TFTP server. If selected and it is non-zero or not blank, the BudgeTone 200 will attempt to retrieve new configuration file or new code image from the specified TFTP server at boot time. It will make up to 3 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. Note: 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.
Via HTTP Server	The URL for the HTTP server used for firmware upgrade and configuration via HTTP. For example, ttp://provisioning.mycompany.com:6688/Grandstream/1.0.0.26 Here ":6688" is the specific TCP port that the HTTP server is listening at, it can be omitted if using default port 80. Note: If Auto Upgrade is set to No, GXW-410x will only do HTTP download once at boot up.

Automatic Upgrade	Choose Yes to enable automatic upgrade and provisioning. In “Check for new firmware every” field, enter the number of days to enable GXW-410x to check the server for firmware upgrade or configuration in the defined period of days. When set to No, GXW-410x will only do upgrade once at boot up. “Always check for New Firmware.” Check New Firmware only when F/W pre/suffix changes”
Syslog Server	The IP address or URL of System log server. This feature is especially useful for ITSP (Internet Telephone Service Provider)
Syslog Level	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: 1. product model/version on boot up (INFO level) 2. NAT related info (INFO level) 3. sent or received SIP message (DEBUG level) 4. SIP message summary (INFO level) 5. inbound and outbound calls (INFO level) 6. registration status change (INFO level) 7. negotiated codec (INFO level) 8. Ethernet link up (INFO level) 9. SLIC chip exception (WARNING and ERROR levels) 10. memory exception (ERROR level) 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 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
NTP server	URI or IP address of the NTP (Network Time Protocol) server, which will be used by the phone to synchronize the date and time.

Figure 10: Screen-shot of GXW-410x FXO Lines Configuration Page

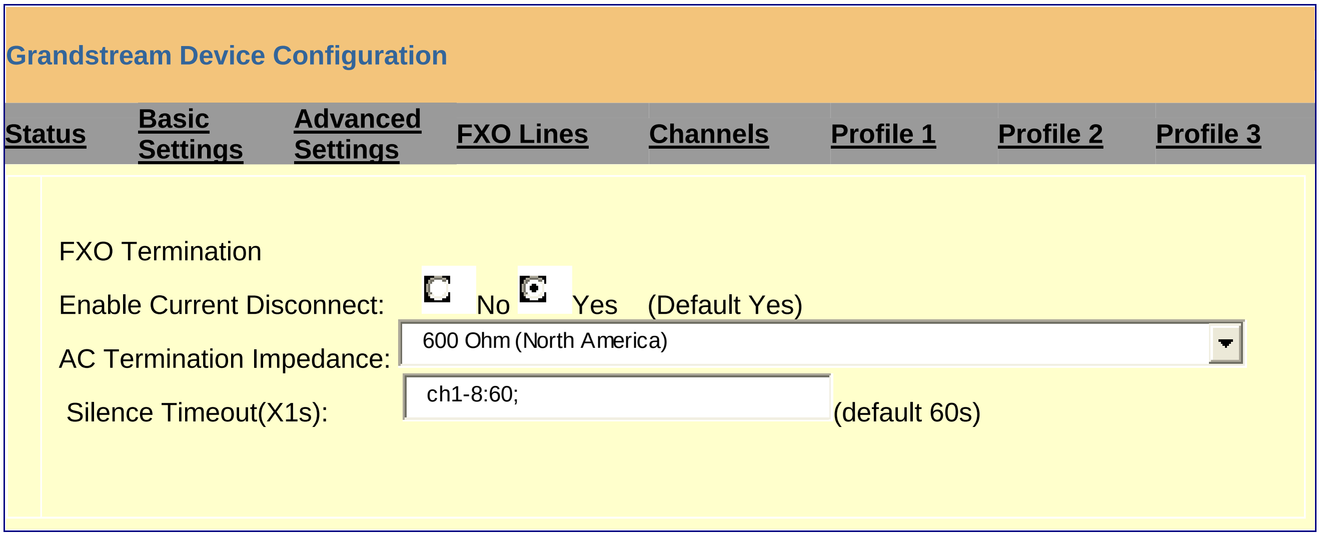


Figure 10a: Screen-shot of GXW-410x Channel Dialing

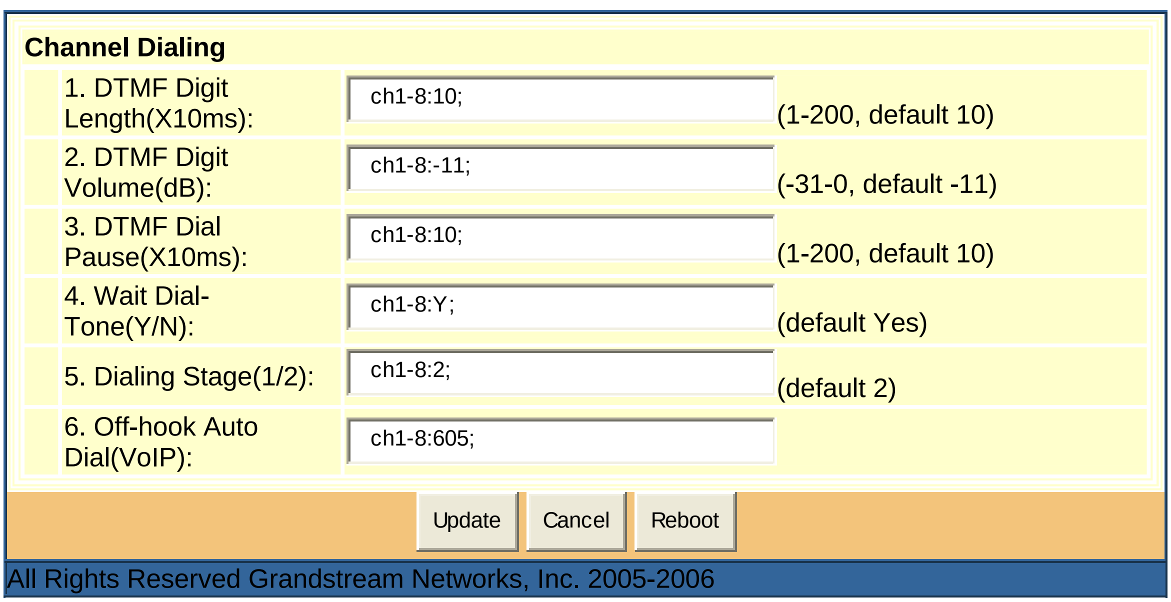


Figure 11: FXO Lines Configuration Definitions

Enable Current Disconnect	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.
AC Termination Impedance	Selects the impedance of the analog Line connected to the FXO port on the GXW-410x.
Silence Timeout	Terminate call after long silence detected. Default is 60 seconds, max 65536.
DTMF Digit Length	Default value is 100ms.
DTMF Digit Volume	Default value is -11dB.
DTMF Dial Pause	Default value is 100ms.
Wait Dial-tone	It is recommended to set this to Y in case Dialing stage is set to 1.
Dialing Stage	Dialing stage can be set to 1 or 2. **Note**:- When set to 1, the Server needs to be configured to allow forwarding and receipt of SIP messages from GXW IP address directly.
Off-hook Auto Dial	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-410x will automatically append the '@' and host portion of the corresponding SIP address.

Figure 12: Screen- shot of GXW-410x Channels Page

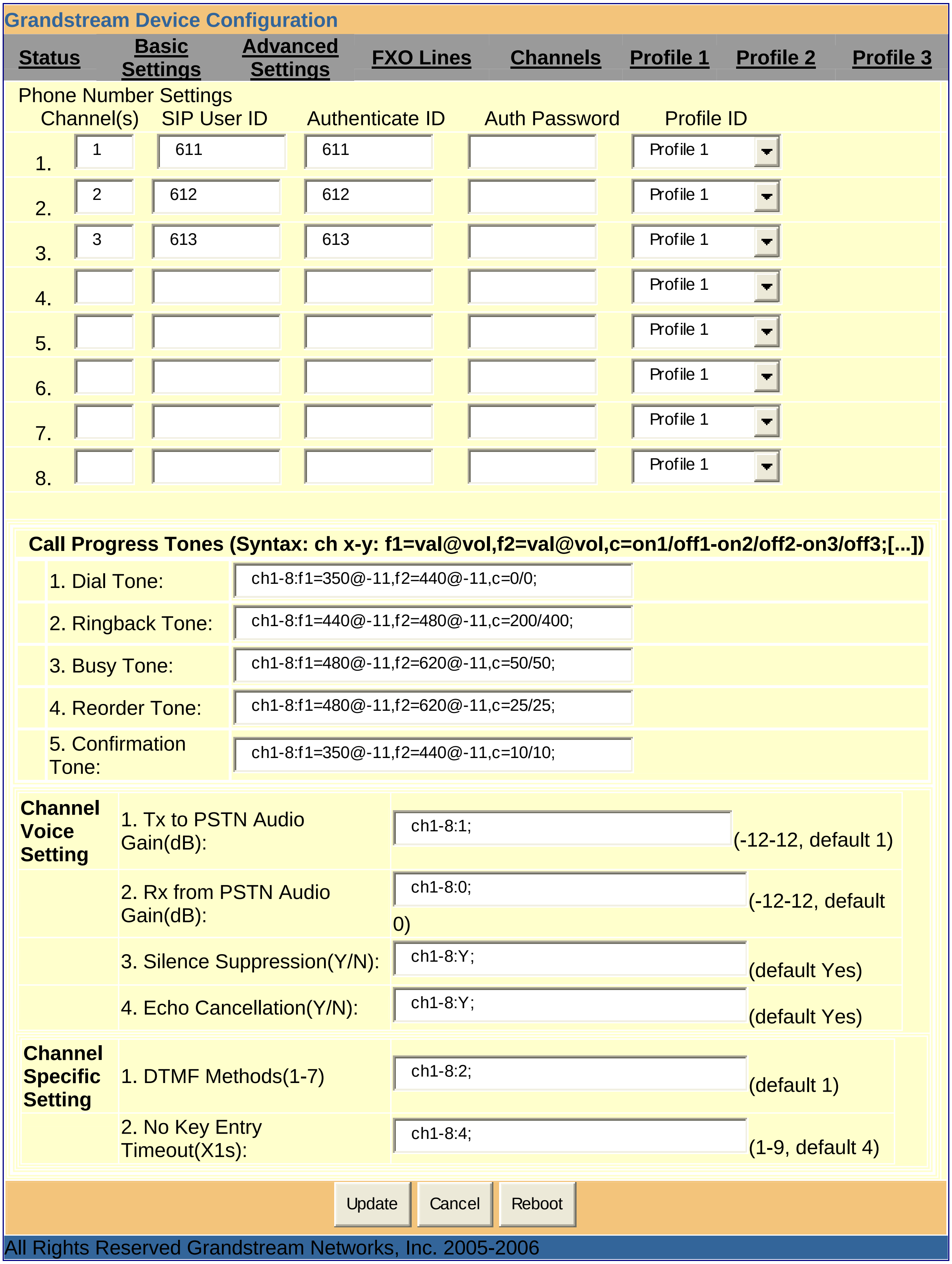
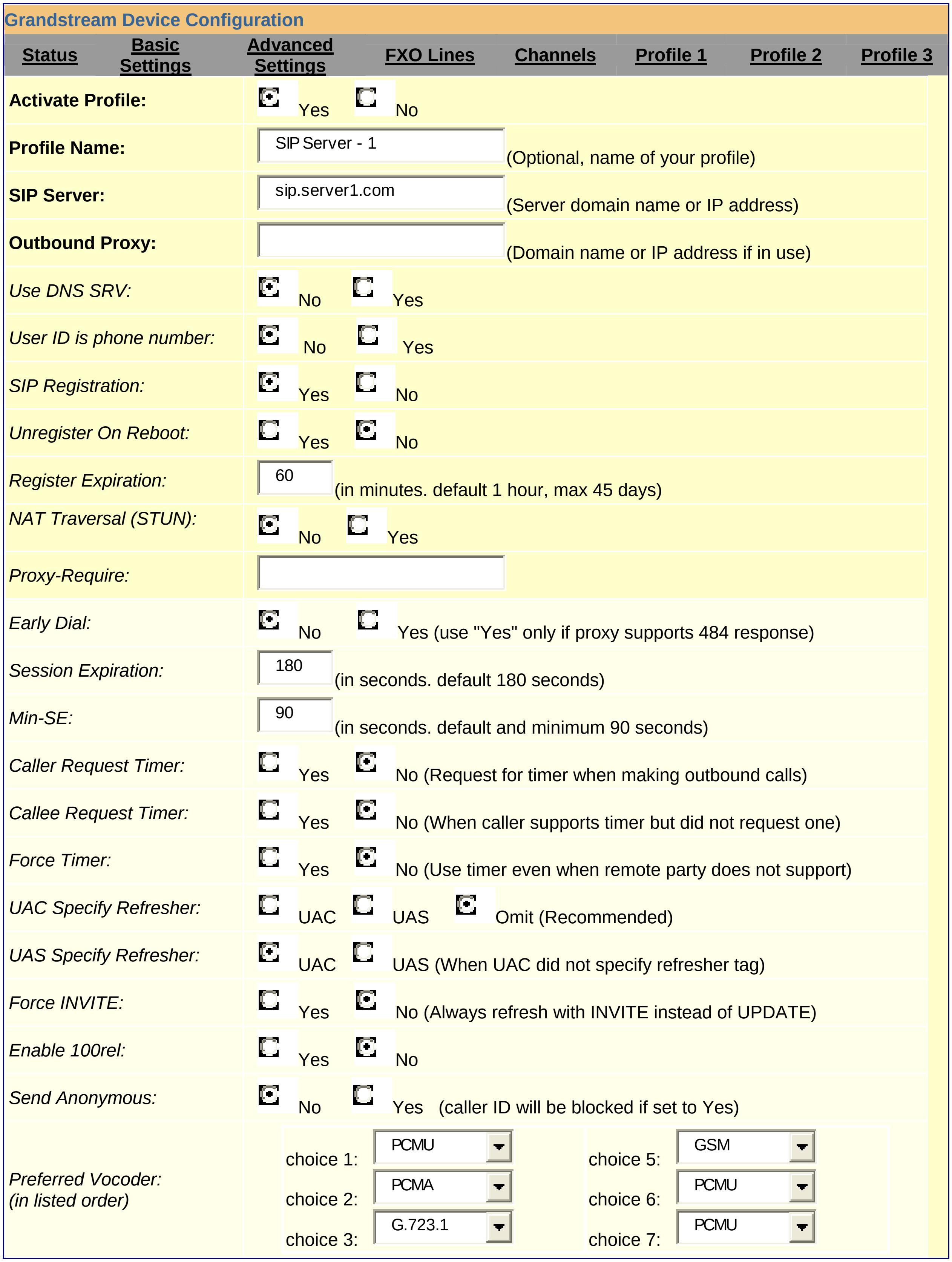


Figure 13: Channels Page Definitions

SIP User ID	User account information, provided by VoIP service provider (ITSP), usually has the form of digit similar to phone number or actually a phone number.
Authentication ID	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
Authentication Password	SIP service subscriber's account password for GXW-410x to register to (SIP) servers of ITSP.
Profile ID	Select the corresponding Profile ID (1/2/3)
Call Progress Tones	Using these settings, user can configure tone frequencies according to their preference. By default, the tones are set to North American frequencies. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. ON is the period of ringing (ON time in ms) while OFF 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. "Dial tone" "Ringback tone" "Busy/Re order tone" "Confirmation tone"
Channel Voice Settings	Channel voice settings mentioned below.
Tx to PSTN Audio Gain (dB)	Allows user to set a value in dB for transmission to PSTN Audio Gain.
Rx from PSTN Audio Gain (dB)	Allows user to set a value in dB for receive from PSTN Audio Gain.
Silence Suppression	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.
Echo Cancellation	When set to Y, Echo cancellation is enabled.
Channel specific Setting	Channel specific settings mentioned below.

DTMF Method	This parameter specifies the mechanism to transmit DTMF digits. There are7 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. 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
No Key Entry Timeout	Default is 4 seconds.

Figure 14: Screen-shot of Grandstream Configuration Page



Grandstream Device Configuration

Status | Basic Settings | Advanced Settings | FXO Lines | Channels | Profile 1 | Profile 2 | Profile 3

Activate Profile:	(•) Yes ( ) No
Profile Name:	SIP Server - 1 (Optional, name of your profile)
SIP Server:	sip.server1.com (Server domain name or IP address)
Outbound Proxy:	(Domain name or IP address if in use)
Use DNS SRV:	(•) No ( ) Yes
User ID is phone number:	(•) No ( ) Yes
SIP Registration:	(•) Yes ( ) No
Unregister On Reboot:	( ) Yes (•) No
Register Expiration:	60 (in minutes. default 1 hour, max 45 days)
NAT Traversal (STUN):	(•) No ( ) Yes
Proxy-Require:	
Early Dial:	(•) No ( ) Yes (use "Yes" only if proxy supports 484 response)
Session Expiration:	180 (in seconds. default 180 seconds)
Min-SE:	90 (in seconds. default and minimum 90 seconds)
Caller Request Timer:	( ) Yes (•) No (Request for timer when making outbound calls)
Callee Request Timer:	( ) Yes (•) No (When caller supports timer but did not request one)
Force Timer:	( ) Yes (•) No (Use timer even when remote party does not support)
UAC Specify Refresher:	( ) UAC ( ) UAS (•) Omit (Recommended)
UAS Specify Refresher:	(•) UAC ( ) UAS (When UAC did not specify refresher tag)
Force INVITE:	( ) Yes (•) No (Always refresh with INVITE instead of UPDATE)
Enable 100rel:	( ) Yes (•) No
Send Anonymous:	(•) No ( ) Yes (caller ID will be blocked if set to Yes)
Preferred Vocoder: (in listed order)	choice 1: PCMU; choice 2: PCMA; choice 3: G.723.1; choice 5: GSM; choice 6: PCMU; choice 7: PCMU

	choice 4:	G.729A/B	choice 8:	PCMU
Special Feature:	Standard			
	Update	Cancel	Reboot	

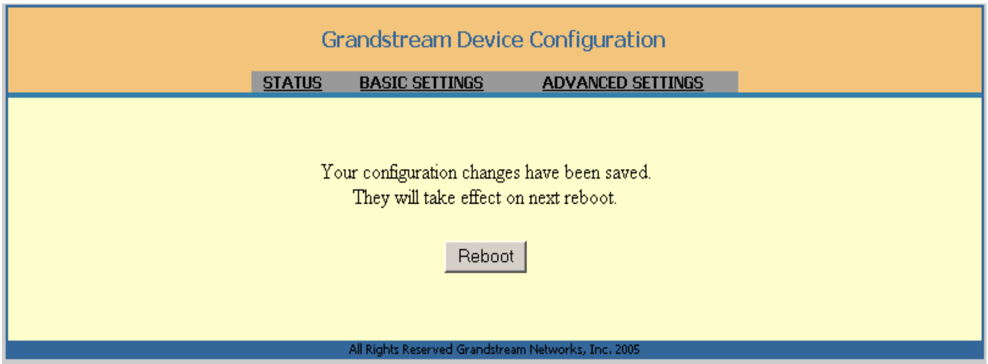
Figure 15: Grandstream Configuration Page Definitions

Activate Profile	When set to Yes the SIP Profile is activated.
Profile Name	A name to identify a Profile.
SIP Server	SIP Server's IP address or Domain name provided by VoIP service provider.
Outbound Proxy	IP address or Domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by GXW-410x 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.
Use DNS SRV:	Default is No. If set to Yes the client will use DNS SRV to look up server.
User ID is Phone Number	If the GXW-410x 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.
SIP Registration	This parameter controls whether the GXW-410x needs to send REGISTER messages to the proxy server. The default setting is "Yes".
Unregister on Reboot	Default is No. If set to yes, the SIP user's registration information will be cleared on reboot.
Register Expiration	This parameter allows the user to specify the time frequency (in minutes) for the GXW-410x 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).
Local SIP port	This parameter defines the local SIP port the GXW-410x 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.
NAT Traversal	This parameter defines whether the GXW-410x NAT traversal mechanism will be activated or not. If activated (by choosing "Yes") and a STUN server is also specified, then the GXW-410x will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the GXW-410x 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-410x 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-410x 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.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Early Dial	Default is No. Use only if proxy supports 484 response.
Session Expiration	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.

Min-SE	The minimum session expiration (in seconds). The default value is 90 seconds.
Caller Request Timer	If selecting “Yes” the phone will use session timer when it makes outbound calls if remote party supports session timer.
Callee Request Timer	If selecting “Yes” the phone will use session timer when it receives inbound calls with session timer request.
Force Timer	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.
UAC Specify Refresher	As a Caller, select UAC to use the phone as the refresher, or UAS to use the Callee or proxy server as the refresher.
UAS Specify Refresher	As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the phone as the refresher.
Force INVITE	Session Timer can be refreshed using INVITE method or UPDATE method. Select “Yes” to use INVITE method to refresh the session timer.
Enable 100rel	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.
Send Anonymous	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.
Preferred Vocoder	The GXW-410x 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”.
Special Feature	Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Nortel, Broadsoft, etc.

5.2.2. Saving the Configuration Changes

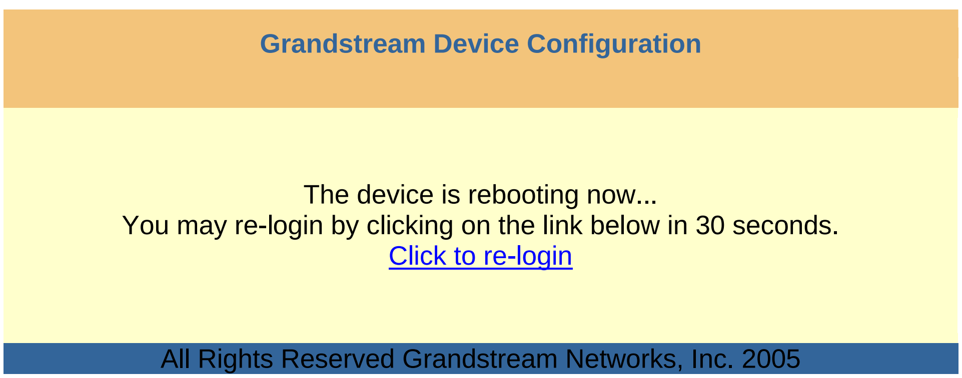
Once a change is made, press the "Update" button in the Configuration Menu. The GXW-410x will then display the following screen to confirm that the changes have been saved.



User is recommended to reboot or power cycle the GXW-410x after all the changes are made so that those changes can take effect.

5.2.3. Rebooting from Remote

The administrator of the phone can remotely reboot the unit by pressing the "Reboot" button at the bottom of the configuration menu. Once done, the following screen will be displayed to indicate that rebooting is underway.



At this point, the user can re-login to the unit after waiting for about 30 seconds.

6. Software Upgrade

Software (or firmware) upgrade 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.

6.1 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 where 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.

6.2 Upgrade through TFTP

To upgrade firmware via TFTP, the field *“Firmware Upgrade and Provisioning: Upgrade Via”* needs to be set to TFTP.

TFTP server can be configured in either IP address format or FQDN.

To configure the TFTP server via the Web configuration interface, open your browser to input the IP address of the GXW-410x. Enter the admin password to enter the configuration screen. From there, enter the TFTP server address or URL in the “Firmware Server Path” field near the bottom of the configuration screen. Once the “Firmware Server Path” is set, user needs to update the change by clicking the “Update” button. Then “Reboot” or power cycle the unit, the firmware files will be fetched upon booting up.

If the configured updating server is found and a new code image is available, the GXW-410x will attempt to retrieve the new image files by downloading them into the GXW-410x’s SRAM. During this stage, the GXW-410x’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-410x phone will stop the TFTP process and simply boot using the existing code image in the flash.

Firmware upgrading may take as long as 1 to 20 minutes over the Internet, or just 20+ seconds if it is performed on a LAN. It is generally recommended to conduct firmware upgrade in a controlled LAN environment if possible.

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 free TFTP or HTTP server in his LAN to do firmware upgrading.

A free Windows version TFTP server can be downloaded from:
http://support.solarwinds.net/updates/New-customerFree.cfm.

Our latest official release can be downloaded from:
http://www.grandstream.com/y-firmware.htm

Unzip the file and put all of the files under the root directory of the TFTP server. Put the PC running the TFTP server and the GXW-410x in the same LAN segment. Please go to File -> Configure -> Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade. Start the TFTP server, in the phone’s web configuration page, configure the Firmware Server Path with the IP address of the PC, 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 server.

7. Restore Factory Default Setting

WARNING!

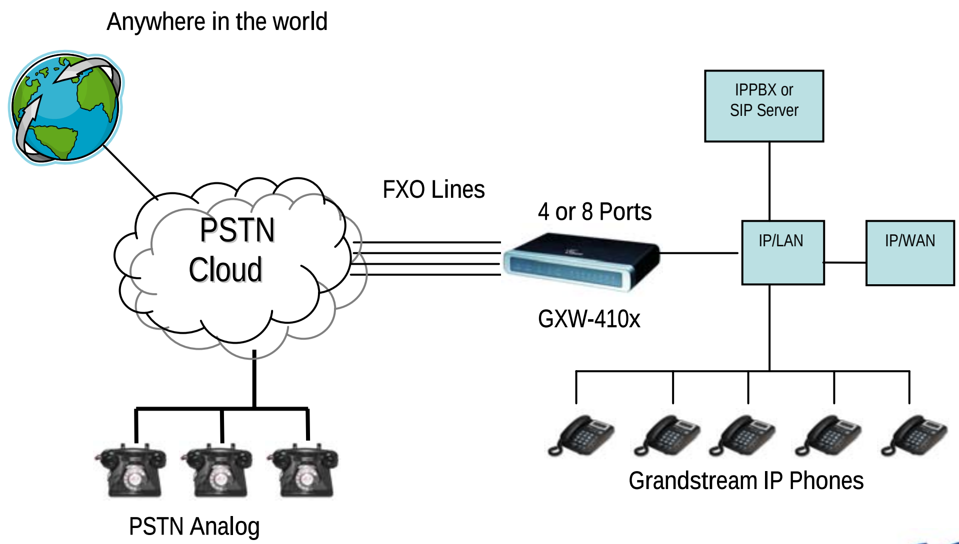
Restore the Factory Default Setting will DELETE all configuration information of the phone. Please BACKUP *or* ***PRINT out all the settings before you approach to following steps. Grandstream will not take any responsibility if you lose all the parameters of setting and cannot connect to your VoIP service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

Please disconnect the network cable and power cycle the unit before trying to reset the unit to factory default. In order to perform a Factory Reset, simply press the **reset button** for 8 seconds. The unit will reboot automatically with configuration and settings back to original Factory defaults.

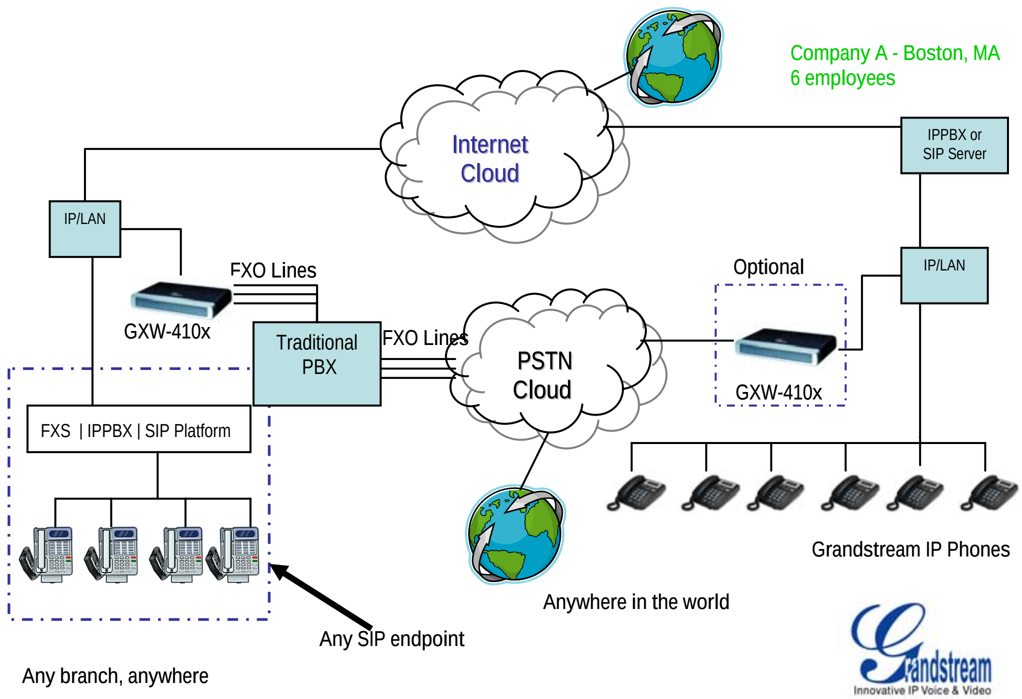
8. Examples of GXW-410x Configurations

The GXW-410x can be used in several scenarios.

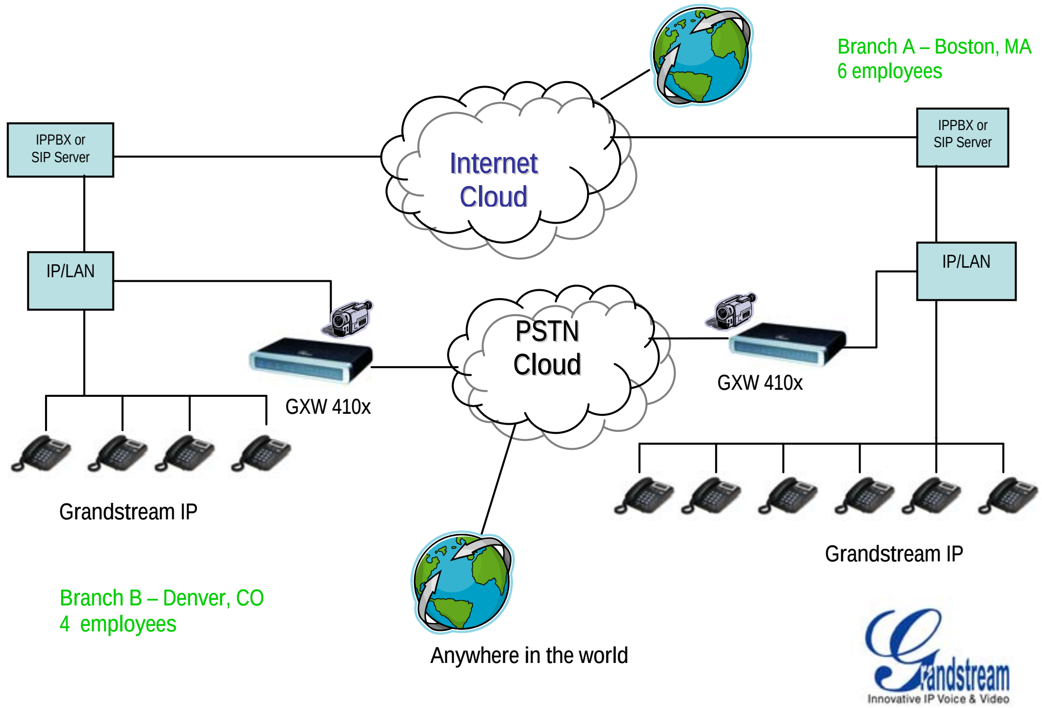
Scenario One: a business with a traditional phone system (with or without broadband access) and an IP PBX or SIP Servers connecting to an Internet Telephone Service Provider (ITSP).



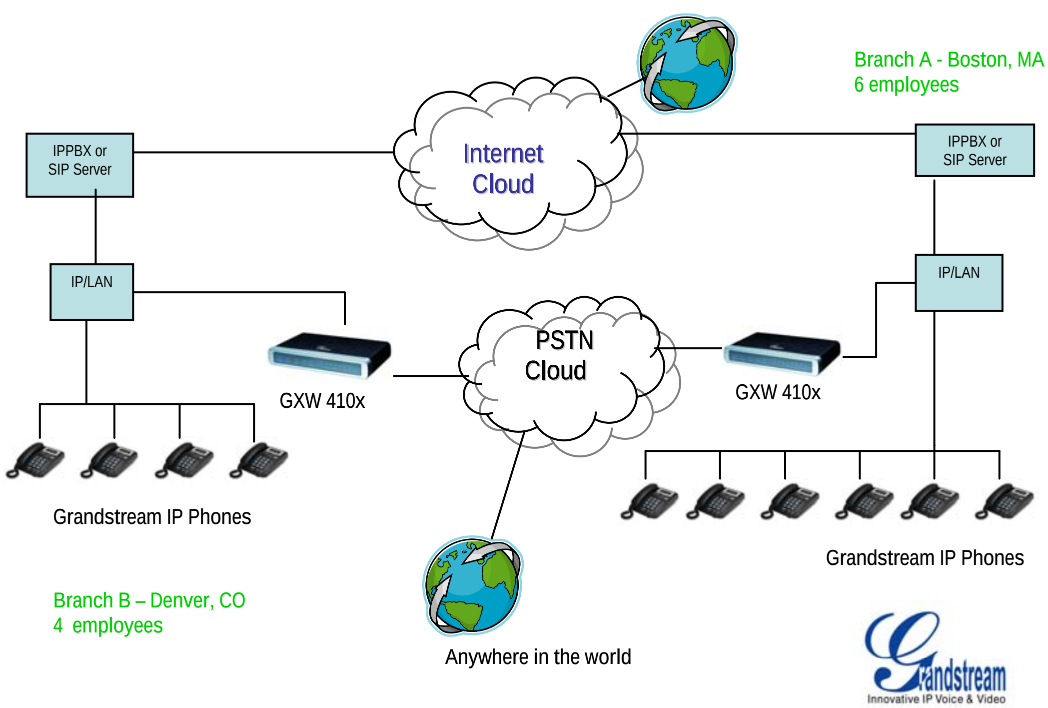
Scenario Two: a small business with traditional analog PBX lines and broadband access who want to extend their traditional PBX to virtually anywhere in the world, using the internet. (Any SIP End point, such as Grandstream BugeTone, HandyTone, GXP-2000 or GXV-3000 are needed in this scenario)



Scenario Three: The GXW-410x 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.



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



9. 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 ***D****omain* ***N****ame* ***S****ystem* (or ***S****ervice* or ***S****erver*), 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 is a fully qualified domain name. www is the host, Grandstream is the second-level domain, and 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 using 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.