Grandstream
Innovative IP Voice & Video

Analog IP Gateway GXW410x

Quick Installation Guide

SW version 1.0.0.36

Updated – 10/24/06

Grandstream Analog IP Gateway GXW410x
Quick Installation Guide
SW version 1.0.0.36

WARNING: Please DO NOT power cycle the GXW410x when LED lights are flashing during system boot up or firmware upgrade. You may corrupt firmware images and cause the unit to malfunction.

WARNING: use only the power adapter included in the GXW410x package. Using an alternative non-qualified power adapter may possibly damage the unit.

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 GXW4104 and GXW4108, 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-410x series. In this environment, the SIP server handles SIP registration and call control and the GXW410x 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.

Package

- Unpack and check all accessories included:
- one 12V universal power adapter,
- one Ethernet cable,
- one device unit.

APPLICATION Diagram

GXW410x series can be deployed in the following enterprise topology:-

![Grandstream logo]
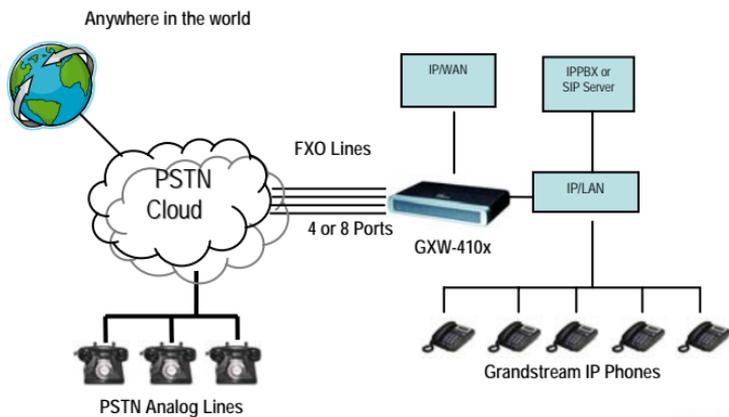
ACCESSING THE WEB CONFIGURATION PAGES

1. Connect the Power to the GXW410x unit.
2. Connect an Ethernet cable between the LAN port on GXW410x to your PC.
3. Assign a dummy IP with the same subnet as the GXW410x IP Address, which is 192.168.0.160 by default. For example, 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 built-in web server of GXW410x.

You may then change it to use DHCP (or PPPoE) or another static IP address according to your local network environment.

CONFIGURATION

There are 2 methods to configure GXW410x to work with an IP PBX:

Method	Description
Method 1	Configure GXW410x with SIP Accounts in IP-PBX, this will enable you to put GXW410x behind a NAT/Firewall. (Used for one-stage and two-stage dialing)
Method 2	Configure GXW410x to function as a PEER gateway. (No SIP accounts are needed; only used for one-stage dialing)

METHOD 1 Configuration:

1. Enter a SIP Server IP Address (or FQDN ex. sip.mysipserver.com) in the SIP Server field under Profile 1 webpage.

2. Next, go to the Channels webpage and enter up to 8 accounts/extensions and their Authentication ID and Authentication Passwords, and select the corresponding Profile.

Channels	SIP User ID	Authentication ID	Authentication Password	Profile ID
1	101	101	***	Profile1
2	102	102	***	Profile1
3	103	103	***	Profile1
4	104	104	***	Profile1
5	105	105	***	Profile1
6	106	106	***	Profile1
7	107	107	***	Profile1
8	108	108	***	Profile1

3. One of the most important settings on the GXW410x is the "Stage Method" setting under FXO Lines webpage. You can set this different (1 or 2) for each Channel. For simplicity, keep Stage Method to 2. **Ex.** Stage Method (1/2) : Ch1-8:2

4. Now, click on Update and reboot the unit. When it boots up, click on Status webpage to check if the Account/extensions show up as "Registered:Yes". If not, ensure the passwords or network settings are correct.

5. Next, place a VOIP to PSTN call (accounts must be registered to your SIP server and your physical PSTN Lines are connected to the FXO Ports at the back panel of the unit).

   Example of a VOIP to PSTN call flow:
   - Accounts 101 to 108 registered on the Channels webpage to the SIP Server A.
   - IP Phone with Account 201 is registered to same SIP Server A
   - PSTN Line X is connected to FXO1 port on the gateway unit.
       a. 201 dials 101 (or 102/103/104/105/106/107/108)
       b. You will hear ringback and then PSTN dialtone played from PSTN Line X
       c. Now, 201 can dial out onto PSTN Line X

   Example of a PSTN to VOIP call flow:
   - PSTN number Y dials PSTN number X (connected to FXO1 on the gateway)
   - You will hear ring-back and then VOIP dial-tone played from 101 (only)

   Note One: VOIP to PSTN calls function in round robin method, so the next available port is selected to route the call. PSTN to VOIP calls depend on the PSTN line you are calling and will be routed to the corresponding VOIP Account on that channel.

   Note Two: For *two-stage dialing*, the "Wait for Dial Tone" field under FXO Lines webpage should be set to N (No) **Ex**. Wait for Dial Tone: ch1-8:N. For *one stage dialing*, configure the SIP Server to forward certain calls (ex. external calls dialed with a prefix, etc) to the gateway, <u>before</u> you configure the gateway.

6. On the gateway, change the Stage Method field to 1. ex. Ch1-8:1

7. Now, based on the prefix you have configured on the SIP Server (ex. 91), if a user dials any number 91-xxx-xxxx, it will be routed to the gateway, which will route it to the PSTN directly.

8. Next you need to enter a VOIP account number for Off-hook Auto Dial (VOIP) under FXO Lines webpage. This account generally will be a dummy account on your SIP Server which is used as a forwarding number for all incoming PSTN calls on the gateway. **Ex**. Off-hook Auto Dial (VOIP) : ch1-8:500. Now, all incoming PSTN calls will directly be forwarded to 500 on SIP Server A.

Note: In regions other than North America, 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 does not provide those values and users don't know the correct values, please use the default values. Please contact product support if you still have questions about configuring your GXW410x.

METHOD 2 Configuration:

In this scenario, you do not need any SIP accounts. So simply keep the Channels table shown above blank.

1. Enter a SIP Server IP Address (or FQDN) in the SIP Server field under Profile 1 webpage.

2. Since this scenario works only with one-stage dialing, you will have to set Stage Method to 1. **Ex**. Stage Method: ch1-8:1

3. Set the "Wait for Dial Tone" field under FXO Lines webpage to Y(Yes). **Ex.** Wait for Dial Tone: ch1-8: Y

Other Important Settings:

Most settings on the web configuration pages are set to default values. However, some settings should be configured according to the local environment.

A. PSTN Settings

The following fields need to be set according to the PSTN Service provider (or PSTN PBX):

1. **Call Progress Tones** (under Channels webpage)
   (Dial Tone, Ring-back Tone, Busy Tone, Reorder Tone, Confirmation Tone)
   These should be set according to the PSTN Service Provider or PSTN PBX that you are using with the gateway. The format and syntax are shown on the web GUI. By default, they are set according to North American PSTN settings.

2. **Caller ID Scheme** (under FXO Lines webpage)
   There are 5 different options to choose from: Bellcore, ETSI_RING, ETSI_TAS, DTMF and NTT.

B. DTMF Methods (to be used ONLY for PSTN to VOIP calls, not for VOIP to PSTN).

The gateway allows several combinations of sending DTMF across from PSTN to the IP end.

Setting	DTMF Method
1	In-audio
2	RTP (RFC2833)
3	In-audio + RTP (RFC2833)
4	SIP INFO
5	In-audio + SIP INFO
6	RTP (RFC2833) + SIP INFO
7	In-audio + RTP (RFC2833) + SIP INFO

Ex. To set as *in-audio*: DTMF Method (1-7): ch1-8:1

C. NAT Settings

If you are planning to keep the gateway within a private network behind a firewall, we recommend using STUN Server. There are 3 settings that are useful in such a scenario:

1. STUN Server (under Advanced Settings webpage).
2. Enter a STUN Server IP (or FQDN) that you may have, or look up a free public STUN Server on the internet and enter it on this field. If using Public IP, keep this field blank.
3. Use Random Ports (under Advanced Settings webpage)
   a. This setting depends on your network settings. Set this to "*Yes*" or "*No*", based on your network settings. If using a Public IP, set this to **No.**
   b. NAT Traversal (under the Profile web pages).
   c. Set NAT Traversal to <u>Yes</u> when gateway is behind firewall on a private network.

D. Preferred VOCODER (Codec)

Under Profile webpage, you can choose from 5 different codecs:

1. PCMU (or G711u)
2. PCMA (or G711a)
3. G729A/B
4. G723.1
5. GSM

E. Firmware Upgrade

Under Advanced Settings webpage, enter your TFTP or HTTP Server IP address (or FQDN) next to the "Firmware Upgrade: "Upgrade Server" field. Select via TFTP or HTTP accordingly. If you plan to perform Automatic Upgrade, set 'firmware upgrade' to "*Yes*", otherwise select "*No*" (this process ensures the system checks for upgrades every time the GXW410x is rebooted).

F. Factory Reset

The ONLY way to reset default factory settings is through the following steps:

1. Unplug the Ethernet cable.
2. Locate a needle-sized hole on the back panel of the gateway unit next to the power connection.
3. Enter a needle like object in this hole and keep it pressed for about 7 seconds.
4. The LAN port LEDs (green and orange) go off and on simultaneously; this indicates the reset was successful.
5. All settings are erased and reset to factory settings.

For additional support, email support@grandstream.com OR call tech support @ 617-566-9300.

Grandstream Networks, Inc.
www.grandstream.com
info@grandstream.com

Corporate Headquarters
1297 Beacon Street. 2d Floor
Brookline, MA 02446
Tel: +1 (617) 566 – 9300
Fax: +1 (617) 247 – 1987

About Grandstream Networks. Grandstream Networks, Inc. is an award-winning designer and manufacturer of next generation IP voice & video products for broadband networks. Grandstream's products deliver superb sound and picture quality, rich telephony features, full compliance with industry standards, and broad interoperability with most service providers and 3rd party SIP based VoIP products. Grandstream is consistently recognized in the VoIP industry for their innovation, affordability and superior value in their products. Grandstream Networks is a private company headquartered in Brookline, MA with offices in Los Angeles, Dallas and China. For more information, please visit www.grandstream.com