



GXW410x Analog FXO Gateway Quick Installation Guide

Part Number: 418-02008-10
Document Version Number: 1.0



Grandstream Analog IP Gateway GXW410x Quick Installation Guide

WARNING: ⚠ Please DO NOT power cycle the GXW410x 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, easy to configure 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. There are two models - the GXW4104 and GXW4108, which have either 4 and 8 FXO ports respectively. The installation is the same for both models.

A SIP proxy server such as Asterisk or a SIP registrar server can be deployed with the GXW410x series. In this environment, the SIP server handles SIP registration and call control and the GXW410x processes media conversion between IP and PSTN calls. By design, the system supports the North American call progress tones and signaling standards on PSTN sides.

Equipment Packaging

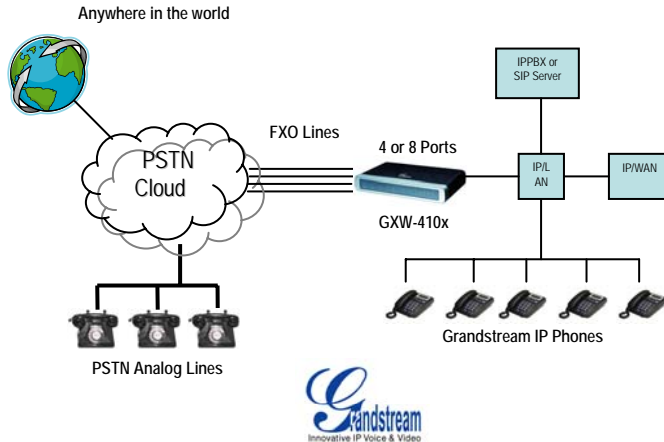
Unpack and check all accessories

- one device unit
- one Ethernet cable
- one 12V universal power adapter



Application Diagram

GXW410x series can be deployed in the following enterprise topology:



Connect the GXW410x

Please use the equipment included in the packaging. Using any other power source may damage the product.

1. Connect one or more PSTN (PBX) telephone lines to FXO ports.
2. Insert the Ethernet cable into the WAN port and connect the other end of the Ethernet cable to an uplink port (a router or a modem, etc.).
3. Connect a PC to the LAN port (optional).
4. Insert the power adapter into the GXW-410x and connect it to an electrical outlet.
5. Using the GXW-400x embedded web server, you can further configure the device.

Configuring the GXW410x using a Web Browser

1. By default, GXW410x has DHCP option turned on.
2. To find out GXW410x's IP address, please download "Grandstream Device IP Discovery" tool from <http://www.grandstream.com/resources.html>. Please follow the instructions below on how to use the tool:
 - a. Run "IP Query.exe"
 - b. Select the network interface that has access to the GXW410x.
 - c. Click "Listen" to start sniffing on the network interface.
 - d. Reboot the GXW410x.
 - e. Select the desired entry from output box
3. Type the GXW410x's IP address in your PC web browser.
4. Log in using password "admin" to configure the unit.

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

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

METHOD	Description
METHOD 1	Configure GXW410x with SIP Accounts in IP PBX, this will enable you to put GXW-410x 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 needed; only used for one-stage dialing)
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METHOD 1 (Configuration):

Please follow the steps below:

1. Enter a SIP Server IP Address (or FQDN ex. sip.mysipserver.com) in the SIP Server field under Profile 1 page.
2. Go to the Channels page and enter up to 8 accounts/extensions along with their Authentication ID and Authentication Passwords, and select the corresponding Profile.

Channels	SIP User ID	Authen ID	Authen 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 page. You can set this different (1 or 2) for each Channel. For simplicity purposes please keep Stage Method to 2.

Ex. Stage Method (1/2): Ch1-8:2;

4. Click on Update and Reboot the unit. When the GXW410x boots up, click on "Status Page" to check if the Account/extensions show up as "Registered:Yes". If not, please double check the passwords or network settings.

5. Once the accounts are registered to the SIP Server, you should be able to make VoIP to PSTN calls (assuming you have physical PSTN Lines connected to the FXO Ports at the back panel of the unit). Below is an example showing VoIP-to-PSTN call flow:

- Accounts 101 to 108 registered on the Channels page to SIP Server A.
 - IP Phone with Account 201 registered to the same SIP Server A.
 - PSTN Line X is connected to FXO X port on GXW-410x.
- a. 201 dials 101 (or 102/103/104/105/106/107/108).
 - b. It will hear ring back tone and then PSTN dial tone played from PSTN Line X.
 - c. 201 can dial out via PSTN Line X.

6. PSTN-to-VoIP call flow:

- a. PSTN number Y dials PSTN number X (connected to FXO1 on the gateway)
- b. Y gets ring back tone and then VoIP dial tone played from 101 (only)

Note: VoIP-to-PSTN calls function in round robin fashion, 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: Generally for two-stage dialing the “Wait for Dial Tone” field under FXO Lines page should be set to N (No) Ex. Wait for Dial Tone: ch1-8:N;

7. For One Stage Dialing, before you configure the GXW-410x you need to make sure the SIP Server is configured to forward certain calls (ex. external calls dialed with a prefix, etc) to the GXW-410x.

8. On the GXW410x, you need to change the Stage Method field to 1. ex. Ch1-8:1;

9. 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 GXW-410x, which will route it to the PSTN directly.

10. Next you need to enter a VoIP account number for “Unconditional Call Forward to VOIP” under FXO Lines page. This account generally will be a dummy account on your SIP Server used simply to be a forwarding number for all incoming PSTN calls on the GXW410x. Ex. Unconditional Call Forward to VOIP: ch1-8:500;

So now all incoming PSTN calls will directly be forwarded to 500 on SIP Server A.

Note: In regions other than North American, 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. Please contact product support if you still have questions about configuring your GXW-410x.

METHOD 2 (Configuration):

1. In this scenario, you do not need any SIP accounts. So simply keep the Channels table shown above blank.
2. You will still need to enter a SIP Server IP Address (or FQDN) in the SIP Server field under Profile 1 page.
3. Since this scenario works only with one-stage dialing, you will have to set Stage Method to 1. Ex. Stage Method: ch1-8:1;
4. It is advised to have “Wait for Dial Tone” field under FXO Lines page set to Y (Yes). Ex. Wait for Dial Tone: ch1-8: Y;

Other Important Settings:

Most of the settings on the web configuration pages should be kept to default value. But there are some settings that need to be configured according to the local network environment.

A. PSTN Settings

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

1. Call Progress Tones (under Channels page)
(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 GXW-410x. 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 page)
There are 5 different options to choose from:

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www.grandstream.com

Last updated on April 1, 2008

Bellcore, ETSI_RING, ETSI_TAS, DTMF and NTT

B. DTMF Methods

Note: The DTMF Method field under Channels page applies ONLY to PSTN to VOIP calls, not 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 DTMF Method as in-audio: DTMF Method (1-7): ch1-8:1

C. NAT Settings

If the GXW410x is connected behind a firewall, an Outbound Proxy or a STUN server must be configured to solve the NAT issue. Here are four settings that can be useful in such a scenario:

1. STUN Server (under Advanced Settings page)
Enter a STUN Server IP or FQDN in this field.
2. Outbound Proxy (Under Profile pages)
Enter an Outbound Proxy server IP or FQDN.
3. Use Random Ports (under Advanced Settings page)
By default, this field is set to No. It may needs to be set to "Yes" depending on your network environment.
4. NAT Traversal (under Profile pages)
Set to Yes.

D. Preferred VOCODER (Codec)

Under Profile pages, you can choose from the following different codecs: PCMU (or G711u), PCMA (or G711a), G729A/B, G723 and GSM.

E. Firmware Upgrade

Under Advanced Settings page, enter "Firmware Server Path" (IP address or FQDN) under "Firmware Upgrade and Provisioning" field. Select upgrade protocol TFTP or HTTP accordingly. By default, "Always Check for New Firmware" is selected.

F. Factory Reset

The ONLY way to set the GXW410x to factory default is by following the steps below:

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 pin in this hole and press 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.

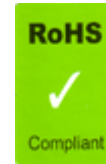
5. Take out the pin. All configuration settings are restored to factory settings.

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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.



Declaration of Conformity

**Konformitätserklärung gemäß dem Gesetz über Funkanlagen und
Telekommunikationsendeinrichtungen (FTEG) und der Richtlinie 1999/5/EG (R&TTE)**

Declaration of Conformity in accordance with the Radio and Telecommunications Terminal
Equipment Act (FTEG) and Directive 1999/5/EC (R&TTE Directive)

Hersteller /Verantwortliche Person / The manufacturer / responsible person

Manufacturer: **Grandstream Networks, Inc.**
Suite 201-205, Bldg #1 (LangFeng Bldg), Industrial Zone
#2 KeFa Road NanShan High Tech Industrial Park, ShenZhen, China

Responsible Laboratory: **Bay Area Compliance Lab Corp. ShenZhen**
Suite C, 41-D Electronics Science & Technology Building,
No. 2070 Shennanzhong Rd ShenZhen,
Guandong 518031, P.R. China
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erklärt, dass das Produkt/ declares that the product:

Series: VoIP Gateway
Model: GXW 4104/8
Attestation Number: RSZ06072501-1, RSZ06072501-2
Date of Issue: August 2, 2006

Type (ggf. Anlagenkonfiguration mit Angabe der Module):
Type (if applicable, configuration including the modules)

Telekommunikations(Tk-)endeinrichtung
Telecommunications terminal equipment

Funkanlage
Radio equipment

Telecommunication via Voice over IP (VoIP)

Verwendungszweck
Intended purpose

.....
Geräteklasse
Equipment class

bei bestimmungsgemäßer Verwendung den grundlegenden Anforderungen des § 3 und den
übrigen einschlägigen Bestimmungen des FTEG (Artikel 3 der R&TTE) entspricht.
complies with the essential requirements of §3 and the other relevant provisions of the FTEG
(Article 3 of the R&TTE Directive), when used for its intended purpose.

Gesundheit und Sicherheit gemäß § 3 (1) 1. (Artikel 3 (1) a))
Health and safety requirements pursuant to § 3 (1) 1. (Article 3(1) a))

angewendete harmonisierte Normen ... Einhaltung der grundlegenden Anforderungen auf
Harmonised standards applied... andere Art und Weise (hierzu verwendete

Standards/Spezifikationen) ...
Other means of proving conformity with the
essential requirements (standards/specifications
used)...

Schutzanforderungen in Bezug auf die elektromagn. Verträglichkeit § 3 (1) 2, Artikel 3 (1b))
Protection requirements concerning electromagnetic compatibility § 3(1)(2), (Article3(1)(b))

angewendete harmonisierte Normen
Harmonised standards applied...

EN 55022: 1998+A1:2001+A2:2003 Class B
EN 55024: 1998+A1:2001+A2:2003

Einhaltung der grundlegenden
Anforderungen auf andere Art und Weise
(hierzu verwendete
Standards/Spezifikationen)...
Other means of proving conformity with the
essential requirements
(standards/specifications used)...

Maßnahmen zur effizienten Nutzung des Funkfrequenzspektrums
Measures for the efficient use of the radio frequency spectrum

Non-applicable

Luftschnittstelle bei § 3 Funkanlagen gemäß (2) (Artikel 3(2))
Air interface of the radio systems pursuant to § 3(2) (Article 3(2))

angewendete harmonisierte
Harmonised standards applied

Normen Einhaltung der grundlegenden Anforderungen auf
andere Art und Weise (hierzu verwendete Standards/
Schnittstellenbeschreibungen)...

Non-applicable

Other means of proving conformity with the essential
requirements (standards/interface specifications used)...

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Ort, Datum
Place & date of issue

Name und Unterschrift
Name and signature



Brookline, MA, USA August 20, 2007

Bruce G. MacAloney, Vice President