

GXW-410x IP ANALOG GATEWAY – ASTERISK™ CONFIGURATION

BASIC CONFIGURATION OF THE GXW410x WITH ASTERISK™

Due to the various deployment possibilities of the GXW410x and Asterisk™, this configuration represents a basic GXW – Asterisk™ configuration as either a peer gateway or using SIP Accounts. Individual configurations can be customized and may slightly differ from either of these basic configurations.

METHODS OF CONFIGURATION

Method 1 Peer Gateway	Configure the GXW410x to function as a PEER gateway (no SIP accounts are needed; only used for one-stage dialing).
Method 2 SIP Accounts	Configure the GXW410x with SIP accounts in Asterisk™; this will enable you to configure the GXW410x behind a NAT/firewall (used for one-stage and two-stage dialing).

Please reboot your system prior to making test calls.

METHOD 1: PEER GATEWAY CONFIGURATION

In the sip.conf file, add the GXW410x:

```
[gxw410x]
type=peer
context=incoming
host=gxw410x_ip_address
insecure=port
```

On the GXW410x, enter a SIP server IP address or FQDN under the Profile 1 web configuration page.

Set SIP Registration setting to “**No**”.

METHOD 2: SIP ACCOUNTS CONFIGURATION

1. Configure the SIP Accounts for the GXW410x.

- a) In the sip.conf file, add SIP accounts for the GXW410x’s channels. You may add up to four or eight depending on the number of channels you have:

```
[101]
type=friend
secret=password
host=dynamic
context=internal
insecure=port
```

- b) On the GXW410x, enter a SIP server IP address or FQDN under the Profile 1 web configuration page.

- c) Under the Channels web configuration page, enter the SIP User IDs, Authentication IDs, and Authentication passwords as well as their corresponding profiles.

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

2. Configure the GXW410x to use either one-stage dialing or two-stage dialing. Please reference [GXW410x FAQs](#) for definitions of one-stage and two-stage dialing.

One-stage Dialing:

1. In the extensions.conf file, specify what numbers dialed from the VoIP side will be sent through the GXW410x. To specify ten-digit numbers and eleven-digit numbers conforming to NANPA specifications as well as 911 emergency services to be sent through the GXW410x, use the following syntax in the same context as the VoIP caller or in an included context:

- a. `exten => _NXXNXXXXXX,1,Dial(SIP/${EXTEN}@gxw410x)`
- b. `exten => _1NXXNXXXXXX,1,Dial(SIP/${EXTEN}@gxw410x)`
- c. `exten => 911,1,Dial(SIP/${EXTEN}@gxw410x)`

2. If configuring the GXW410x using SIP accounts, please replace “gxw410x” with a SIP account configured on the GXW410x channels, such as “100”.
3. Numbers such as (626) 956-0260, 1(617)566-9300, and 911 will be dialed through the GXW410x.
4. Calls coming into the GXW410x from the PSTN side will be directly auto-dialed to a configurable VoIP extension when using one-stage dialing. Set the auto dial extension in the extensions.conf file, within the same context that the GXW410x is in or in an included context. For example: forward incoming PSTN calls to extension 200, which may be a SIP device or a greeting (generally stored in /var/lib/Asterisk™/sounds):

- a. `exten => 200,1,Dial(SIP/200)`
- or*
- b. `exten => 200,1,Background(greeting_filename)`

5. Under FXO Lines web configuration page set the following:
 - a. Wait for Dial-Tone(Y/N) to Y.
 - b. Stage Method(1/2) to 1.
 - c. Offhook Auto Dial(VoIP) to the VoIP number to forward incoming PSTN calls to. Using Asterisk™ configuration example, this would be 200.

Two-stage Dialing (used only when configured with SIP accounts)

To use two-stage dialing, dial the SIP accounts configured on the channels of the GXW410x. For example, if one of the SIP accounts is SIP/101, we add this line in the extensions.conf file, within the same context as the VoIP caller or in an included context:

```
exten => 100,1,Dial(SIP/101)
```

With this rule, VoIP users may call extensions 101 and the call will be connected with the GXW410x channel, where they will receive the PSTN dial tone to make PSTN calls.

On the GXW410x, under FXO Lines web configuration page set the following:

- a. Wait for Dial-Tone(Y/N) to N.
 - b. Stage Method(1/2) to 2.
 - c. Offhook Auto Dial(VoIP) does not need to be set as it is not used. Incoming PSTN calls will simply receive VoIP dial tone, where the PSTN caller can then dial a VoIP number.
3. Most settings on the GXW410x web configuration pages are set to default values. Be sure to configure the GXW410x settings according to the local environment.

OTHER GENERAL SETTINGS

A. PSTN Settings

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

1. Call Progress Tones located under the *Channels Web Configuration* 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 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 located under *FXO Lines Web Configuration* page. There are five 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 DTMF sending methods from PSTN to VoIP.

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

For example, to set all channels to use RFC2833, set the DTMF Method field under the Channels web configuration page as follows:

DTMF Method (1-7): ch1-8:2

Please ensure the DTMF methods you set match those enabled on Asterisk™. RFC2833 is the recommended method; which can be set under the [general] area in the sip.conf file with this line:

```
dtmfmode=rfc2833
```

C. NAT Settings

1. If you place the gateway within a private network behind a firewall, we recommend using a STUN Server. In the STUN Server field under the Advanced Settings web configuration page, enter a STUN server IP or FQDN. You can find free public STUN servers on the internet. Set NAT Traversal (STUN) under the Profile web configuration pages to Yes.
2. Depending on your network settings, you may set the Use Random Port setting under the Advanced Settings web configuration page to Yes or No. If using a public IP address, set this setting to No.

D. Preferred Vocoder (Codec)

Under the *Profile Web Configuration* pages, you can choose from five different codecs:

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

Please ensure that the codecs you set match those enabled on Asterisk™. Codecs may be enabled system-wide in Asterisk or for specific users. For example: to enable PCMU and GSM system-wide, place the following lines under the [general] settings in the sip.conf file:

```
allow=ulaw  
allow=gsm
```

MAKE A TEST CALL

After configuring the GXW410x, reload the chan_sip.so module and the extensions in Asterisk, or simply restart the service. The commands vary depending on the version of Asterisk. Update the changes made to the GXW410x and reboot the device. You may now place inbound and outbound test calls to test your configuration.