

Basic configuration of the GXW410x with Trixbox

Please note that due to the customizable nature of both the GXW410x and Trixbox and the vast deployment possibilities, these instructions should be taken as a basic tutorial sample of getting the GXW410x to work with Trixbox. Your implementation may be customized and differ from this basic configuration as according to your needs.

There are two methods to configure the GXW410x to work with Trixbox:

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

1. Depending on your deployment scenario, please choose one method and follow the instructions for that method below:

Configuring the GXW410x as a peer gateway

In FreePBX, click on the Trunks section and choose Add SIP Trunk. Assign a name to the trunk. For example:

Trunk Name: gwx_trunk

In the PEER Details section, enter the following:

```
context=from-trunk
host=gxw410x_ip_address
insecure=port
type=peer
dtmfmode=preferred_dtmf_mode
```

If you are having problems, you may try setting *insecure=very* instead.

Replace *preferred_dtmf_mode* with your DTMF mode of choice. Based on our experience, rfc2833 has proven to be the most reliable DTMF method. To set rfc2833, use the following syntax:

```
dtmfmode=rfc2833
```

Click Submit Changes and Apply Configuration Changes.

On the GXW410x, enter the Trixbox server IP address or FQDN under the Profile 1 web configuration page. Set the SIP Registration setting to No.

Configuring the GXW410x using SIP accounts

If using one-stage dialing:

In FreePBX, click on the Trunks section and choose Add SIP Trunk. Assign only a numeric ID for the Trunk Name as the GXW410x will only register numeric SIP account user IDs. For example:

Trunk Name: 101

In the PEER Details section, enter the following:

```
context=from-trunk
host=dynamic
secret=101
type=friend
dtmfmode=preferred_dtmf_mode
```

Replace *preferred_dtmf_mode* with your DTMF mode of choice. Based on our experience, rfc2833 has proven to be the most reliable DTMF method. To set rfc2833, use the following syntax:

```
dtmfmode=rfc2833
```

Click Submit Changes. Repeat until all desired SIP accounts have been configured as Trunks, then click Apply Configuration Changes.

If using two-stage dialing:

In FreePBX, click on the Extensions section and a Generic SIP Device. Assign only a numeric ID for the User Extension as the GXW410x will only register numeric SIP account user IDs. Enter any Display Name. For example:

User Extension: 101
Display Name: GXW Channel 1

Scroll down and in the Device Options section, set the dtmfmode field. Based on our experience, rfc2833 has proven to be the most reliable DTMF method:

```
dtmfmode:              rfc2833
```

Click Submit Changes. Repeat until all desired SIP accounts have been configured as Trunks, then click Apply Configuration Changes.

Please note that when using two-stage dialing, the PSTN caller has as much access in the VoIP side of the system as the context the configured SIP account is in allows. This may present a security concern to some if unauthorized callers can commit toll fraud by calling in and using another trunk to make unauthorized outbound calls, or gain other unauthorized system access.

On the GXW410x, enter the Trixbox server IP address or FQDN under the Profile 1 web configuration page. Set the SIP Registration setting to Yes.
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. Next, the GXW410x will need to be configured to use one-stage dialing or two-stage dialing. Please see the Frequently Asked Questions section for the definitions of one-stage and two-stage dialing. Once you have chosen your stage method, please configure Trixbox and the GXW410x using the corresponding procedure below:

Using one-stage dialing

In FreePBX, click on the Outbound Routes section and enter any name in the Route Name field. Enter the Dial Patterns you would like to match and dial out the GXW410x. Please use the Dial patterns wizards if you need help with dial patterns. For example, we may name the route GXW_outbound:

Route Name: GXW_outbound

We may also specify all ten-digit numbers and eleven-digit long-distance numbers conforming to NANPA specifications as well as 911 emergency services to be sent through the GXW410x by entering the following:

Dial Patterns: NXXNXXXXXX
 1NXXNXXXXXX
 911

Lastly, we select the SIP trunk(s) configured earlier to send the call to in the Trunk Sequence section. Then click Submit Changes and Apply Configuration Changes.

With this Outbound Route, numbers such as (626)956-0260, 1(617)566-9300, and 911 will be dialed through the GXW410x.

Calls coming into the GXW410x from the PSTN side are directly auto dialed to a VoIP extension when using one-stage dialing. In FreePBX, click on the Inbound Routes section and enter a number in the DID Number field to catch the auto dialed number from the GXW410x. Although this number can be arbitrary, we may set it to your main DID for understandability. For example:

DID Number: 6175669300

Scroll down to the Set Destination field and choose the destination for incoming PSTN calls to reach. For example, you may choose a Ring Group, an IVR menu, a Time Condition, etc. Then click Submit Changes and Apply Configuration Changes.

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

1. Wait for Dial-Tone(Y/N) to N.
2. Stage Method(1/2) to 1.
3. Unconditional Call Forward to VoIP to the VoIP. The user should construct a valid URI where to forward the call. The User ID can be the extension number like ch1-4:6175669300; The SIP server will be using the IP address or FQDN indicated in the profile page, we use p1 for profile1, p2 or profile2 and p3 for profile3. The SIP destination port should be the one where the Trixbox is listening for incoming SIP communication.

Using two-stage dialing (can only be used when configuring GXW410x with SIP accounts)

In Trixbox, nothing will need to be configured.

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

1. Wait for Dial-Tone(Y/N) to N.
2. Stage Method(1/2) to 2.
3. Unconditional Call Forward to 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. However, some settings should be configured according to the local environment if necessary.

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; 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 analog PBX that you are using with the gateway. The format and syntax are shown on the web configuration page. By default, they are set according to North American PSTN settings.

Please remember that the time in ms for the cadences is multiplied by a factor of 10ms.

Remember that no carrier uses both current disconnect and tone disconnect at the same time to signal the disconnection of a call.

2. Caller ID Scheme; under the FXO Lines web configuration page

There are eleven different options to choose from:

- 1 - Bellcore/Telcordia, (USA)
- 2 - ETSI-FSK during ringing
- 3 - ETSI-FSK prior to ringing with DTAS (dual tone alert signal)
- 4 - ETSI-FSK prior to ringing with LR (line reversal)
- 5 - ETSI-FSK prior to ringing with RP (ring pulse)
- 6 - ETSI-DTMF during ringing
- 7 - ETSI-DTMF prior to ringing with DTAS
- 8 - ETSI-DTMF prior to ringing with LR
- 9 - ETSI-DTMF prior to ringing with RP
- 10 - SIN 227 - BT (UK)
- 11 - NTT, Japan type

B. DTMF Methods (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, find the DTMF Method field under the Channels web configuration page and set it to the following:

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

C. NAT Settings

1. If you are planning to place the gateway within a private network separated from the Trixbox server from 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 G.711 u-law)
2. PCMA (or G.711 a-law)
3. G.729A/B
4. G.723.1
5. GSM

Please make sure that the codecs you set match those enabled on Trixbox.

4. Update the changes made to the GXW410x and reboot the device. You may now place inbound and outbound test calls to test your configuration.

Please send any comments or suggestions regarding this document to support@grandstream.com.