

Grandstream Networks, Inc.

Peering HT8XX with GXW410X



Table of Contents

OVERVIEW	3
SCENARIO 1: PEERING ONE HT8XX WITH GXW410X	4
HT8XX Configuration	4
GXW410X Configuration	6
One-To-One mapping	9
Port-To-Port mapping	10
SCENARIO 2: PEERING MULTIPLE HT8XX WITH GXW410X	12
Step 1: First HT8XX Configuration	13
Step 2: Second HT8XX Configuration	13
Step 3: GXW410X Configuration	13

Table of Figures

Figure 1 : Scenario 1: Peering one HT8XX with GXW410X	3
Figure 2 : Scenario 2: Peering multiple HT8XX with GXW410X	3
Figure 3 : Peering one HT8XX with GXW410X	4
Figure 4 : Profile Settings	5
Figure 5 : FXS Ports	5
Figure 6 : General Settings	6
Figure 7 : NAT Traversal	6
Figure 8 : SIP Registration	6
Figure 9 : Number of Rings Settings	7
Figure 10 : Dialing to PSTN	7
Figure 11: Port Scheduling Schema	8
Figure 12 : Calling to VoIP	9
Figure 13: Port Scheduling Schema	9
Figure 14: One-To-One Mapping	10
Figure 15: Port-To-Port Mapping call flow	11
Figure 16: Peering Multiple HT8XX with GXW410X	12
Figure 17: FXS Ports settings of the first HT8XX	13
Figure 18: FXS Ports settings of the second HT8XX	13
Figure 19: General Settings	13
Figure 20: Calling to VoIP	15



OVERVIEW

This document describes basic configuration to peer HT8XX series with GXW410X. This configuration applies to users seeking to add a GXW410X not only as a remote extension but also as an external PSTN trunk.

The document will demonstrate these two scenarios to set up HT8XX series with the GXW410X:

- **Scenario 1: Configure one HT8XX to be peered with GXW410X**

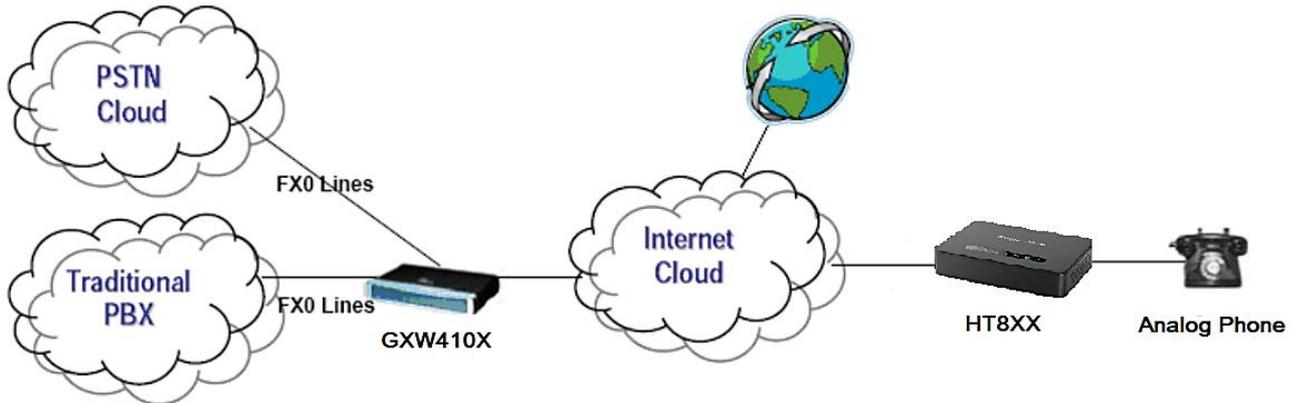


Figure 1 : Scenario 1: Peering one HT8XX with GXW410X

- **Scenario 2: Configure two HT8XX to be peered with GXW410X**

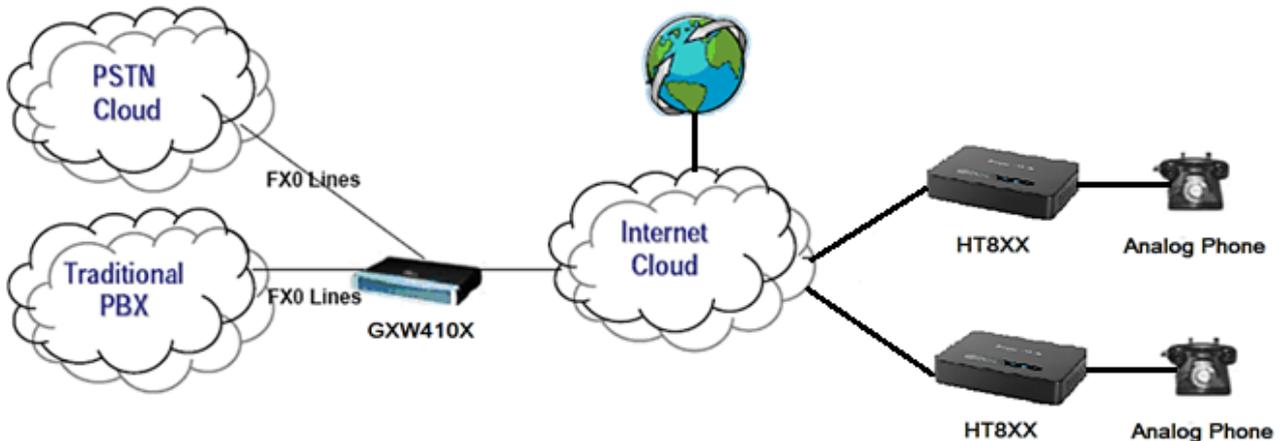


Figure 2 : Scenario 2: Peering multiple HT8XX with GXW410X

Note: GXW410X series includes GXW4104 and GXW4108

SCENARIO 1: PEERING ONE HT8XX WITH GXW410X

A common scenario which involves one HT8XX (ATA) and GXW410X (FXO gateway) but doesn't involve any SIP server. This scenario allows organization with remote location to access FXO trunks through IP network.

In this scenario, we will proceed first from the web GUI of HT8XX, then on the GXW41XX in order to configure the Peer Trunk on both sides.

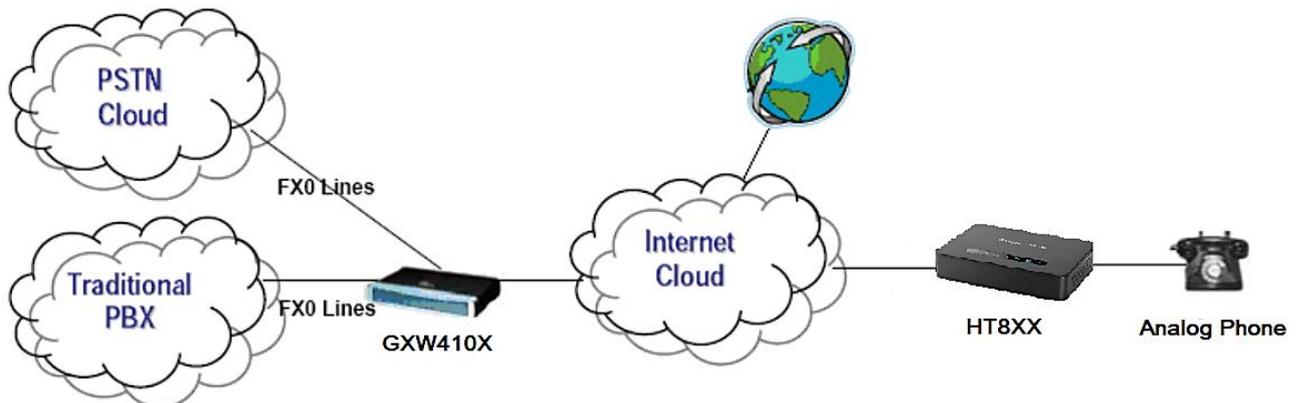


Figure 3 : Peering one HT8XX with GXW410X

HT8XX Configuration

On the HT8XX web GUI, access to "Profile1", then set the following:

1. **Profile Active** → YES
Activates this account In order to get the Profile1 operational.
2. **Primary SIP Server** → Enter here the IP address of the GXW410X
3. **SIP Transport** → UDP
In our Example, we will use the protocol UDP as transport protocol.
4. **NAT Traversal** → NO
This setting will force the HT8XX to use its private IP address while making calls through peering link with GXW410X.
5. **SIP Registration** → NO
Registration not needed since this example is a peering between GXW410X and HT8XX.
6. **Outgoing Call without Registration** → YES
This enables the ability to place outgoing calls even if the account is not registered.

Grandstream Device Configuration

STATUS
BASIC SETTINGS
ADVANCED SETTINGS
PROFILE 1
PROFILE 2
FXS PORTS

Profile Active: No Yes

Primary SIP Server: (e.g., sip.mycompany.com, or IP address)

Failover SIP Server: (Optional, used when primary server no response)

Prefer Primary SIP Server: No Yes (yes - will register to Primary Server if Failover registration expires)

Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)

Backup Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)

Prefer Primary Outbound Proxy: No Yes (yes - will reregister via Primary Outbound Proxy if registration expires)

Allow DHCP Option 120 (override SIP server): No Yes

SIP Transport: UDP TCP TLS (default is UDP)

SIP URI Scheme When Using TLS: sip sips

Use Actual Ephemeral Port in Contact with TCP/TLS: No Yes

NAT Traversal: No Keep-Alive STUN UPnP

DNS Mode: A Record SRV NAPTR/SRV

DNS SRV use Registered IP: No Yes

Tel URI:

Use Request Routing ID in SIP INVITE Header: No Yes

SIP Registration: No Yes

Unregister On Reboot: No Yes

Outgoing Call without Registration: No Yes

Figure 4 : Profile Settings

Notes:

Once the settings above are set, make sure to set the following additional settings as bellow:

7. **Local SIP Port → 5060**
Defines the local port to use by the HT814 for listening and transmitting SIP packets
8. **Use Random SIP Port → NO**
This option forces the HT8XX to use port configured under “Local SIP Port” option

FXS Ports Settings:

- Port 1: SIP User ID: 123 | Password: 123 | Name: 123 | Profile ID: Profile 1 | Enable Port: Yes

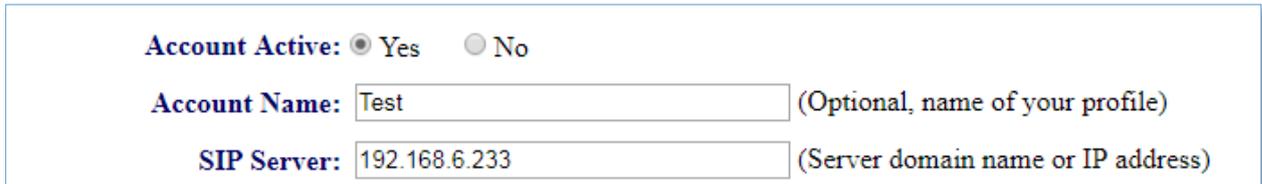
User Settings							
Port	SIP User ID	Authenticate ID	Password	Name	Profile ID	Hunting Group	Enable Port
1	<input type="text" value="123"/>	<input type="text"/>	<input type="text" value="123"/>	<input type="text" value="123"/>	Profile 1 ▼	None ▼	<input type="radio"/> No <input checked="" type="radio"/> Yes
2	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	Profile 1 ▼	None ▼	<input type="radio"/> No <input checked="" type="radio"/> Yes
3	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	Profile 1 ▼	None ▼	<input type="radio"/> No <input checked="" type="radio"/> Yes
4	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	Profile 1 ▼	None ▼	<input type="radio"/> No <input checked="" type="radio"/> Yes

Figure 5 : FXS Ports



GXW410X Configuration

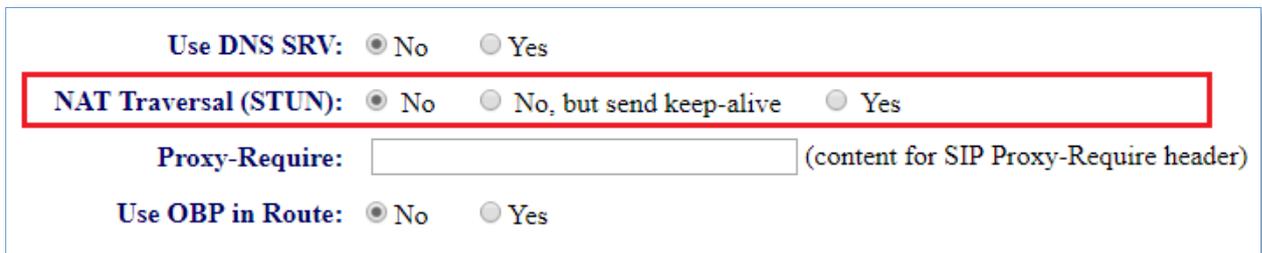
1. In order to activate the profile configured on GXW410X, please access to GXW410X web GUI under "Accounts" → "Account 1" → "General Settings" then set the Following:
 - a. **Account Active** → YES
 - b. **Account Name** → Specify a name to identify this profile.
 - c. **SIP Server** → Enter the IP address of the HT8XX that you are peering with GXW410X.



Account Active: Yes No
Account Name: (Optional, name of your profile)
SIP Server: (Server domain name or IP address)

Figure 6 : General Settings

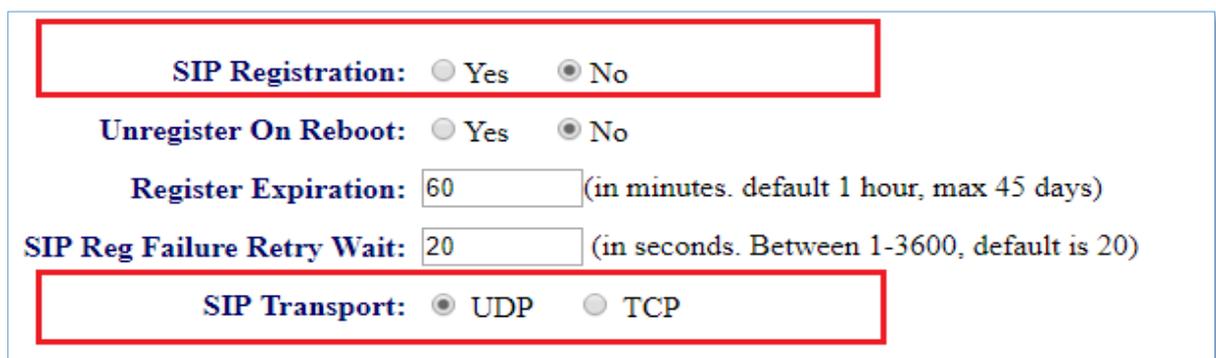
2. Now under "Accounts" → "Account 1" → "Network Settings"
 - a. **NAT traversal (STUN)** → NO
This setting will force the GXW410X to use its private IP address while making calls through peering link with HT8XX



Use DNS SRV: No Yes
NAT Traversal (STUN): No No, but send keep-alive Yes
Proxy-Require: (content for SIP Proxy-Require header)
Use OBP in Route: No Yes

Figure 7 : NAT Traversal

3. Under "Accounts" → "Account 1" → "SIP Settings" :
 - a. **SIP Registration** → NO
Registration not needed since this example is a peering between GXW410X and HT8XX
 - b. **SIP Transport** → UDP
On our Example, we will use the protocol UDP as transport protocol



SIP Registration: Yes No
Unregister On Reboot: Yes No
Register Expiration: (in minutes. default 1 hour, max 45 days)
SIP Reg Failure Retry Wait: (in seconds. Between 1-3600, default is 20)
SIP Transport: UDP TCP

Figure 8 : SIP Registration



4. Under “FXO lines” → “Setting” then set the following:

a. **Number of Rings Before Pickup → 4**

This is the number of rings the gateway will wait to send the call to the VOIP side in case the Caller ID has yet to be detected. If there is CID, information the call will be sent right away. If your lines do not have the CID service set this to 1.

Port Caller ID Setting	
Number of Rings Before Pickup:	ch1-4:4; (1-50, default 4)

Figure 9 : Number of Rings Settings

5. Under “FXO lines” → “Dialing” → “Dialing to PSTN” and set the following:

a. **Wait For Dial Tone → Yes or No**

Set it to “YES” or “NO” depending on the setting that suits your FXO lines.

b. **Stage Method → 1**

Setting this parameter to 1 will direct PSTN call from the VOIP endpoint.

c. **Min Delay Before Dialing Out → 500**

It's the time that GXW410X will wait before dialing out; available range 500ms to 65000ms).

Dialing to PSTN	
Wait for Dial-Tone(Y/N):	ch1-8:Y; (default No)
Stage Method(1/2):	ch1-8:1; (default 2 stage dialing)
Min Delay Before Dialing Out:	ch1-8:500; (default 500ms, 50 ~ 65000ms)

Figure 10 : Dialing to PSTN

6. Access to GXW410X web GUI under “FXO lines” → “Dialing” → “Port Scheduling Schema”, then set as follow:

Round-robin and/or Flexible → rr:1-4;

With this setting, the GXW410X will behave as follow:

- The VoIP endpoint wants to make an outgoing call through PSTN line.
- GXW410X receives the INVITE from the VOIP endpoint.
- GXW410X checks first the availability of FXO-4 since it has the highest Port-ID.
- If FXO-4 is available, the call will try to go through FXO-4.
- If FXO-4 is busy, the call will try to go through FXO-3.
- Then If FXO-3 is busy, the call will try to go through FXO-2.
- Then If FXO-2 is busy, the call will try to go through FXO-1.



Port Scheduling Schema	
Round-robin and/or Flexible:	rr:1-4; (default rr:1-8;) (Syntax: rr: port_group; [...]; Default: rr:1-8; round-robin of all ports)
Prefix to Specify Port:	99 (1 stage dial, default 99) (Syntax: prefix# + ch# + dialing# will request the ch# per call. Note that this code has to prefix dialplan number and prefix doesn't impact round-robin)

Figure 11: Port Scheduling Schema

7. Under “Settings” → “Channels Settings” then set the following:

→ **SIP Channel Settings:**

- a. **Local SIP Listen Port** → ch1-4:5060++;
The ++ indicates increments by two, so port 1 is set at 5060, port at 5062 and so on. This setting can be used with Round Robin and/or Flexible setting below to configure different ports to be placed under different Round Robin groups.

→ **Calling to VOIP:**

- a. **User ID** → ch1-4:123;
This parameter allows users to configure a User ID or extension number to be automatically dialed upon FXO line off-hook. (In our example, we will send it to User ID 123)
- b. **SIP Server** → ch1-4: p1;
Specify the Profile of the user id configured above (p1 stands for Profile 1, p2 stands for Profile 2 and so on).
- c. **SIP Destination Port** → ch1-4: 5060;
Setting this port to 5060 will allow this GXW410X to always redirect the incoming PSTN calls to the HT8XX which is reachable via port 5060



SIP Channel Setting	
DTMF Methods(1-7):	ch1-4:1; (default 1) (1:in-audio, 2:RFC2833, 3:1+2, 4:SIP Info, 5:1+4, 6:2+4, 7:1+2+4)
No Key Entry Timeout(X1s):	ch1-4:4; (1-9, default 4)
Local SIP Listen Port:	ch1-4:5060++; (default ch1-8:5060++;)
SRTP Mode(1-3):	ch1-4:1; (default 1) (1:disabled, 2:enabled but not forced, 3:enabled and forced)
Calling to VoIP	
Unconditional Call Forward to Following:	
User ID:	ch1-4:1000; (i.e ch1-2:223;ch3:224)
SIP Server:	ch1-4:p1; (ch1-2:p1;ch3:p2)
SIP Destination Port:	ch1-4:5060; (ch1-2:5060;ch2:7080)

Figure 12 : Calling to VoIP

Note:

If your setup requires that FXS port 'x' user be able to access only FXO port 'x' for outbound calls, you can use either the One-To-One mapping or the Port-To-Port Mapping feature.

- One-To-One mapping between HT8XX and GXW410X. [One-To-One mapping]
- Port-To-Port Mapping between HT8XX and GXW410X. [Port-To-Port mapping]

One-To-One mapping

The One-to-One mapping feature will allow to the HT8XX FXS port to specify the port number 'x' that user will use to access GXW410X FXO port 'x' for outbound calls.

Port Scheduling Schema	
Round-robin and/or Flexible:	rr:1-4; (default rr:1-8;) (Syntax: rr: port_group; [...]; Default: rr:1-8; round-robin of all ports)
Prefix to Specify Port:	99 (1 stage dial, default 99) (Syntax: prefix# + ch# + dialing# will request the ch# per call. Note that this code has to prefix dialplan number and prefix doesn't impact round-robin)

Figure 13: Port Scheduling Schema

To enable this One-To-One Mapping feature between HT8XX FXS ports and GXW410X FXO ports, you will need to access to GXW410X web GUI under "FXO lines" → "Dialing" → "Port Scheduling Schema", then set as follow:



Prefix to specify Port → 99

This setting allows you the power to send outbound calls from the VOIP endpoint to a specific FXO line. With this setting, the GXW410X will behave as follow:

1. User tries to make an outbound call to an external number 5669300 via GXW410X FXO port 1
2. User should add the prefix **991** to the external called numbers.
3. In this case the user will dial **99-1-5669300** where :
 - **99**: represents the prefix;
 - **1** : represents port number 1
 - **5669300** : represents external number
4. Once received, the GXW4108 gateway will strip off **991** from the dialed number **9915669300**
5. GXW410X will route the call to port 1 and make call to **5669300** over the PSTN network connected to FXO1.



Figure 14: One-To-One Mapping

Port-To-Port mapping

Port to port mapping feature will force all calls placed on a specific FXS port "x" to be completed through the PSTN network that is connected to a specific FXO port "x". This is similar to a one to one mapping configuration except it is hardcoded to be always routed through a specific FXO port. In this case, the End-user is not required to dial any extra numbers (see example above in page 10).

Port to port mapping is configured using the Dial Plan feature on the HT8XX. The HT8XX supports \$P, \$p as port numbers. Port to port mapping from the HT8XX to GXW410X is configured as follows:

- **GXW410X** : prefix to specific port: 99 ---- default settings on FXO GXW
- **HT8XX** : dial plan: { <=99\$P>x+ }

So, if the HT8XX configured with this dial plan, any number dialing from a specific port will be prefixed with 99 and a port number:

1. User dials **5669300** from his analog phone which is connected to port 1
2. HT8XX will append this number with the appropriate dial plan configured (HT8XX will be **9915669300**)

3. Once received , The GXW4108 gateway will strip off **991** from the dialed number **9915669300**
4. GXW410X will route the call to port 1 and make call to 5669300 over the PSTN network connected to FXO1.



Figure 15: Port-To-Port Mapping call flow



SCENARIO 2: PEERING MULTIPLE HT8XX WITH GXW410X

A common scenario which involves multiple HT8XX (ATA) and GXW410X (FXO gateway) but does not involve any SIP server. This scenario allows organization with remote location to access FXO trunks through IP network.

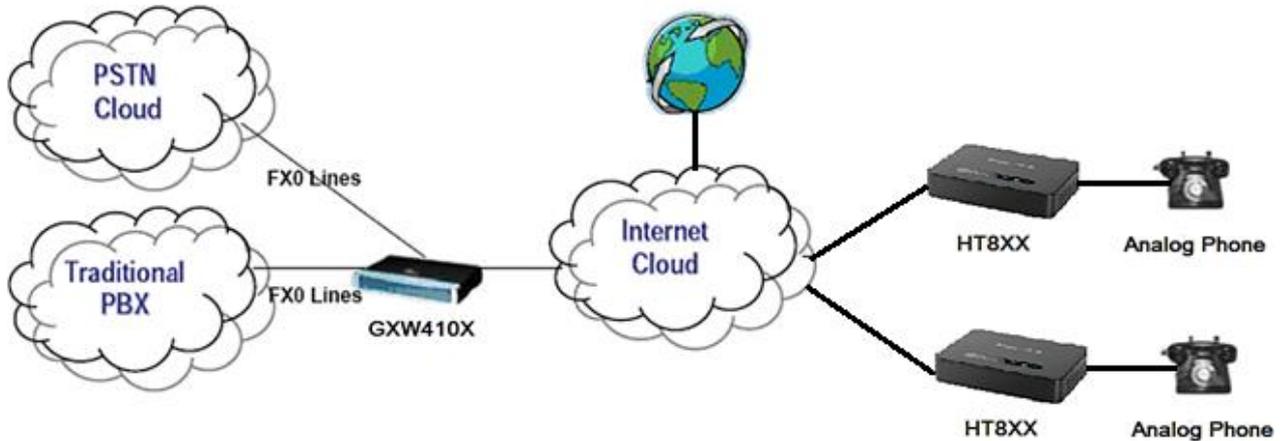


Figure 16: Peering Multiple HT8XX with GXW410X

In this scenario, we will use two HT8XX, users can apply same configuration if more than two HT8XX are used.

To configure this scenario, you will need to follow the steps below:

- **Step 1:** Configure the first HT8XX to be peered with the GXW410X
- **Step 2:** Configure the second HT8XX to be peered with the GXW410X.
- **Step 3:** Configure GXW410X to be peered with both HT8XX.

Step 1: First HT8XX Configuration

Please refer to the page 5 where you will find the steps (from step 1 until step 8) to configure the first HT8XX, then you will need to set the FXS port settings as follow:

FXS Ports Settings:

- Port 1: SIP User ID: 111 | Password: 111 | Name: 111 | Profile ID: Profile 1 | Enable Port: Yes

User Settings							
Port	SIP User ID	Authenticate ID	Password	Name	Profile ID	Hunting Group	Enable Port
1	111		...	111	Profile 1	None	<input type="radio"/> No <input checked="" type="radio"/> Yes
2					Profile 1	None	<input type="radio"/> No <input checked="" type="radio"/> Yes
3					Profile 1	None	<input type="radio"/> No <input checked="" type="radio"/> Yes
4					Profile 1	None	<input type="radio"/> No <input checked="" type="radio"/> Yes

Figure 17: FXS Ports settings of the first HT8XX

Step 2: Second HT8XX Configuration

Please refer to the page 5 where you will find steps (from step 1 until step 8) to configure the first HT8XX, then you will need to set the FXS port settings as follow:

FXS Ports Settings:

- Port 1: SIP User ID: 222 | Password: 222 | Name: 222 | Profile ID: Profile 1 | Enable Port: Yes

User Settings							
Port	SIP User ID	Authenticate ID	Password	Name	Profile ID	Hunting Group	Enable Port
1	222		...	222	Profile 1	None	<input type="radio"/> No <input checked="" type="radio"/> Yes
2					Profile 1	None	<input type="radio"/> No <input checked="" type="radio"/> Yes
3					Profile 1	None	<input type="radio"/> No <input checked="" type="radio"/> Yes
4					Profile 1	None	<input type="radio"/> No <input checked="" type="radio"/> Yes

Figure 18: FXS Ports settings of the second HT8XX

Step 3: GXW410X Configuration

- In order to activate the first Account configured on GXW410X , please access to GXW410X web GUI under “Accounts” → “**Account 1**” → “General Settings” then set the Following:
 - Account Active** → YES
 - Account Name** → Specify a name to identify this profile
 - SIP Server** → Enter the IP address of the First HT8XX that you are peering the GXW410X with

Account Active:	<input checked="" type="radio"/> Yes	<input type="radio"/> No
Account Name:	<input type="text" value="Test"/>	(Optional, name of your profile)
SIP Server:	<input type="text" value="192.168.6.233"/>	(Server domain name or IP address)

Figure 19: General Settings



2. In order to activate the second account profile configured on GXW410X , please access to GXW410X web GUI under “Accounts” → “**Account 2**” → “General Settings” then set the Following:
 - a. **Account Active** → YES
 - b. **Account Name** → Specify a name to identify this profile
 - c. **SIP Server** → Enter the IP address of the Second HT8XX that you are peering the GXW410X with
3. Please refer to the page 7 and page 8 where you will find the steps (from step 2 until step 6) which show how to configure the remaining GXW410X settings.
4. At this stage, you will need to configure the last step which is step 7 called “Channels Settings”, so under “Settings” → “Channels Settings” then set the following:

→ **SIP Channel Settings:**

- a. **Local SIP Listen Port** → ch1-4:5060++;
The ++ indicates increments by two, so port 1 is set at 5060, port at 5062 and so on. This setting can be used with Round Robin and/or Flexible setting below to configure different ports to be placed under different Round Robin groups.

→ **Calling to VoIP:**

- a. **User ID** → ch1:111; ch2:222;ch3-4;;
This parameter allows users to configure a User ID or extension number to be automatically dialed upon FXO line off-hook. (In our example, we will send it to User ID 123)
- b. **SIP Server** → ch1:p1; ch2:p2;ch3-4;;
Specify the Profile of the user id configured above (p1 stands for Profile 1, p2 stands for Profile 2 and so on).
- c. **SIP Destination Port** → ch1-4:5060;
Setting this port to 5060 will allow this GXW410X to always redirect the incoming PSTN calls to the HT8XX reachable via port 5060



SIP Channel Setting	
DTMF Methods(1-7):	ch1-4:1; (default 1) (1:in-audio, 2:RFC2833, 3:1+2, 4:SIP Info, 5:1+4, 6:2+4, 7:1+2+4)
No Key Entry Timeout(XIs):	ch1-4:4; (1-9, default 4)
Local SIP Listen Port:	ch1-4:5060++; (default ch1-8:5060++;)
SRTP Mode(1-3):	ch1-4:1; (default 1) (1:disabled, 2:enabled but not forced, 3:enabled and forced)
Calling to VoIP	
Unconditional Call Forward to Following:	
User ID:	ch1:111; ch2:222;ch3-4:: (i.e ch1-2:223;ch3:224)
SIP Server:	ch1:p1; ch2:p2;ch3-4;; (ch1-2:p1;ch3:p2)
SIP Destination Port:	ch1-4:5060: (ch1-2:5060;ch2:7080)

Figure 20: Calling to VoIP

