

## PEERING ONE GXW42XX WITH MULTIPLE GXW410X

A common scenario which involves a GXW42xx (FXS gateway) and multiple GXW410x (FXO gateway) but doesn't involve any SIP server. This scenario allows organization with remote location to access FXO trunks through IP network.

### CONFIGURATION OF THE GXW42XX & MULTIPLE GXW410XX SCENARIO

#### GXW42XX CONFIGURATION

##### Maintenance -> Network Settings

- STUN Server – Blank

##### Profiles -> Profile 1

General Settings:

- SIP server – Set to IP address of 1<sup>st</sup> GXW 410x

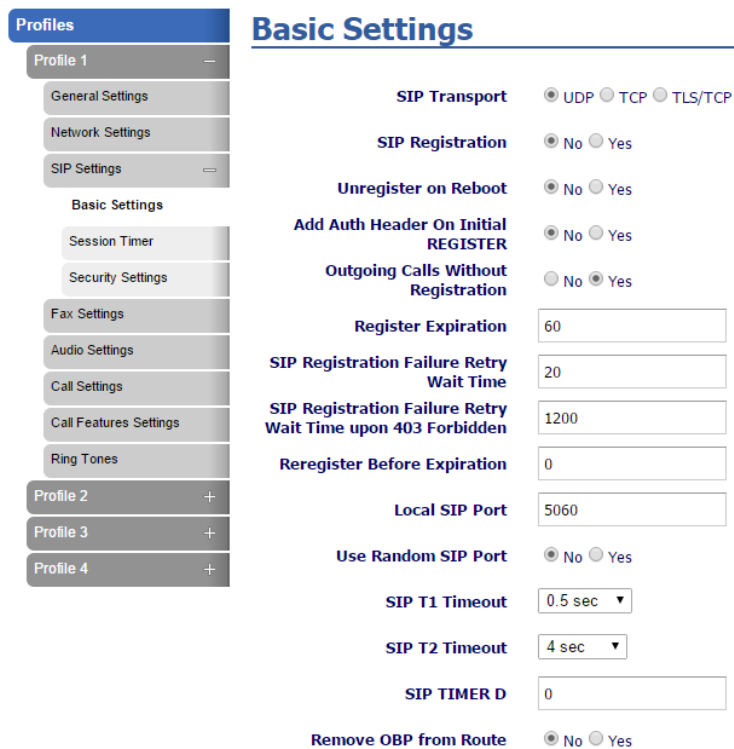
Network Setting

- NAT traversal – No

SIP Settings -> Basic Settings:

- SIP registration - No
- Outgoing Call without Registration – NO
- Local SIP Port - 5060

Maintenance -> Network Settings



The screenshot shows the configuration interface for Profile 1. On the left is a sidebar with a 'Profiles' menu containing Profile 1 (selected), Profile 2, Profile 3, and Profile 4. Under Profile 1, there are sub-menus for General Settings, Network Settings, SIP Settings, Basic Settings, Session Timer, Security Settings, Fax Settings, Audio Settings, Call Settings, Call Features Settings, and Ring Tones. The main area is titled 'Basic Settings' and contains the following configuration options:

- SIP Transport:** Radio buttons for UDP (selected), TCP, and TLS/TCP.
- SIP Registration:** Radio buttons for No (selected) and Yes.
- Unregister on Reboot:** Radio buttons for No (selected) and Yes.
- Add Auth Header On Initial REGISTER:** Radio buttons for No (selected) and Yes.
- Outgoing Calls Without Registration:** Radio buttons for No and Yes (selected).
- Register Expiration:** Text input field with value 60.
- SIP Registration Failure Retry Wait Time:** Text input field with value 20.
- SIP Registration Failure Retry Wait Time upon 403 Forbidden:** Text input field with value 1200.
- Reregister Before Expiration:** Text input field with value 0.
- Local SIP Port:** Text input field with value 5060.
- Use Random SIP Port:** Radio buttons for No (selected) and Yes.
- SIP T1 Timeout:** Dropdown menu with value 0.5 sec.
- SIP T2 Timeout:** Dropdown menu with value 4 sec.
- SIP TIMER D:** Text input field with value 0.
- Remove OBP from Route:** Radio buttons for No (selected) and Yes.

**Profiles -> Profile 2**

**General Settings:**

- SIP server – Set to IP address of 2<sup>nd</sup> GXW 410x

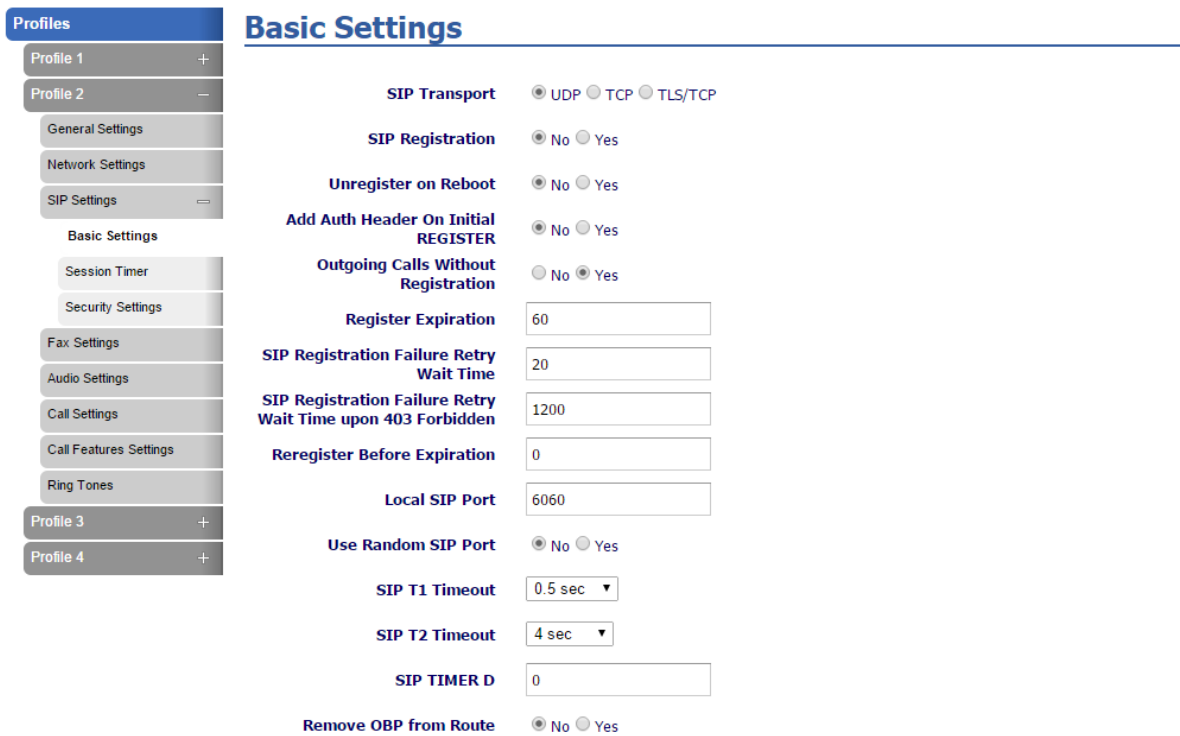
**Network Setting**

- NAT traversal – No

**SIP Settings -> Basic Settings:**

- SIP registration - No
- Outgoing Call without Registration – NO
- Local SIP Port - 6060

Maintenance -> Network Settings



The screenshot shows the 'Basic Settings' configuration page for Profile 2. On the left is a sidebar with a 'Profiles' menu containing Profile 1, Profile 2 (selected), Profile 3, and Profile 4. Below the profiles are various settings categories: General Settings, Network Settings, SIP Settings (expanded), Basic Settings (selected), Session Timer, Security Settings, Fax Settings, Audio Settings, Call Settings, Call Features Settings, and Ring Tones. The main content area is titled 'Basic Settings' and contains the following configuration options:

- SIP Transport:** Radio buttons for UDP (selected), TCP, and TLS/TCP.
- SIP Registration:** Radio buttons for No (selected) and Yes.
- Unregister on Reboot:** Radio buttons for No (selected) and Yes.
- Add Auth Header On Initial REGISTER:** Radio buttons for No (selected) and Yes.
- Outgoing Calls Without Registration:** Radio buttons for No and Yes (selected).
- Register Expiration:** Text input field with value 60.
- SIP Registration Failure Retry Wait Time:** Text input field with value 20.
- SIP Registration Failure Retry Wait Time upon 403 Forbidden:** Text input field with value 1200.
- Reregister Before Expiration:** Text input field with value 0.
- Local SIP Port:** Text input field with value 6060.
- Use Random SIP Port:** Radio buttons for No (selected) and Yes.
- SIP T1 Timeout:** Dropdown menu with value 0.5 sec.
- SIP T2 Timeout:** Dropdown menu with value 4 sec.
- SIP TIMER D:** Text input field with value 0.
- Remove OBP from Route:** Radio buttons for No (selected) and Yes.

**FXS Ports:**

Port Settings:

- Port 1 - 8 -> User ID: 123
- Port 1 – 8 -> Profile 1
- Port 1 – 8 -> Enable FXS – Yes
- Port 9 - 16 -> User ID: 123
- Port 9 – 16 -> Profile 2
- Port 9 – 16 -> Enable FXS – Yes

FXS Ports		Port Settings						
<ul style="list-style-type: none"> <li>Port Settings</li> <li>Advanced Port Settings</li> <li>FXO Mapping</li> </ul>		Port	SIP User ID	Authenticate ID	Password	Name	Profile	Enable FXS (TR-069)
FXS 1	123					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes	
FXS 2	123					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes	
FXS 3	123					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes	
FXS 4	123					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes	
FXS 5	123					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes	
FXS 6	123					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes	
FXS 7	123					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes	
FXS 8	123					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes	
FXS 9	123					Profile 2	<input type="radio"/> No <input checked="" type="radio"/> Yes	
FXS 10	123					Profile 2	<input type="radio"/> No <input checked="" type="radio"/> Yes	
FXS 11	123					Profile 2	<input type="radio"/> No <input checked="" type="radio"/> Yes	
FXS 12	123					Profile 2	<input type="radio"/> No <input checked="" type="radio"/> Yes	
FXS 13	123					Profile 2	<input type="radio"/> No <input checked="" type="radio"/> Yes	
FXS 14	123					Profile 2	<input type="radio"/> No <input checked="" type="radio"/> Yes	
FXS 15	123					Profile 2	<input type="radio"/> No <input checked="" type="radio"/> Yes	
FXS 16	123					Profile 2	<input type="radio"/> No <input checked="" type="radio"/> Yes	

**1<sup>ST</sup> GXW 410x CONFIGURATION:**

**Settings -> General Settings:**

- STUN server – Blank

**FXO lines -> Dialing:**

Dialing to PSTN:

- Wait for dial tone – Y or N (whichever suits your FXO lines)
- Stage Method – 1

**Settings -> Channels Settings**

Calling to VoIP:

- User ID: ch1-8:123;
- Sip Server: ch1-8; p1;
- Sip Destination Port: 5060++

**FXO lines -> Settings:**

Port Caller ID Setting:

- Number of Rings Before Pickup: ch1-8:4;

**Accounts -> Account 1:**

General Settings:

- SIP Server: Set it to IP address of GXW42xx

SIP Settings:

- SIP registration – No

Network Settings:

- NAT traversal – No

**Dialing to PSTN**

**Wait for Dial-Tone(Y/N):**  (default No)

**Stage Method(1/2):**  (default 2 stage dialing)

**Min Delay Before Dialing Out:**  (default 500ms, 50 ~ 65000ms)

**Calling to VoIP**

**Unconditional Call Forward to Following:**

**User ID:**  (i.e ch1-2:223;ch3:224)

**SIP Server:**  (ch1-2:p1;ch3:p2)

**SIP Destination Port:**  (ch1-2:5060;ch2:7080)

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### Port Caller ID Setting

**Number of Rings Before Pickup:**  (1-50, default 4)

**Caller ID Scheme:**  (1-11, default 1)

- 1 - Bellcore/Telcordia
- 2 - ETSI-FSK during ringing
- 3 - ETSI-FSK prior to ringing with DTAS
- 4 - ETSI-FSK prior to ringing with LR
- 5 - ETSI-FSK prior to ringing with RP
- 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
- 11 - NTT - Japan

**2<sup>ND</sup> GXW 410X CONFIGURATION:**

**Settings -> General Settings:**

- STUN server – Blank

**FXO lines -> Dialing:**

Dialing to PSTN:

- Wait for dial tone – Y or N (whichever suits your FXO lines)
- Stage Method – 1

**Settings -> Channels Settings**

Calling to VoIP:

- User ID: ch1-8:123;
- Sip Server: ch1-8; p1;
- Sip Destination Port: 6076++

**FXO lines -> Settings:**

Port Caller ID Setting:

- Number of Rings Before Pickup: ch1-8:4;

**Accounts -> Account 1:**

General Settings:

- SIP Server: Set it to IP address of GXW42xx

SIP Settings:

- SIP registration – No

Network Settings:

- NAT traversal – No

**Dialing to PSTN**

**Wait for Dial-Tone(Y/N):**  (default No)

**Stage Method(1/2):**  (default 2 stage dialing)

**Min Delay Before Dialing Out:**  (default 500ms, 50 ~ 65000ms)

**Calling to VoIP**

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**SIP Server:**  (ch1-2:p1;ch3:p2)

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