

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

Profile 1

General Settings:

- SIP server – Set to IP address of 1st GXW 410x

SIP Settings → Basic Settings:

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

Advanced settings :

- STUN Server – Blank

Profiles	Basic Settings
Profile 1	
General Settings	
Network Settings	
SIP Settings	
Basic Settings	
Session Timer	
Security Settings	
Fax Settings	
Audio Settings	
Call Settings	
Ring Tones	
Profile 2	
Profile 3	
Profile 4	

SIP Transport	☒ UDP ☐ TCP ☐ TLS/TCP
SIP Registration	☒ No ☐ Yes
Unregister on Reboot	☒ No ☐ Yes
Outgoing Calls Without Registration	☐ No ☒ Yes
Register Expiration	60
SIP Registration Failure Retry Wait Time	20
Reregister Before Expiration	0
Local SIP Port	5060
Use Random SIP Port	☒ No ☐ Yes
SIP T1 Timeout	0.5 sec
SIP T2 Timeout	4 sec
Remove OBP from Route	☒ No ☐ Yes

Profile 2

General Settings:

- SIP server – Set to IP address of 2nd GXW 410x

SIP Settings → Basic Settings:

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

Advanced settings:

- STUN Server – Blank

Profiles		Basic Settings	
Profile 1	+		
Profile 2	–		
General Settings			
Network Settings			
SIP Settings			
Basic Settings			
Session Timer			
Security Settings			
Fax Settings			
Audio Settings			
Call Settings			
Ring Tones			
Profile 3	+		
Profile 4	+		

SIP Transport	☒ UDP ☐ TCP ☐ TLS/TCP
SIP Registration	☒ No ☐ Yes
Unregister on Reboot	☒ No ☐ Yes
Outgoing Calls Without Registration	☐ No ☒ Yes
Register Expiration	60
SIP Registration Failure Retry Wait Time	20
Reregister Before Expiration	0
Local SIP Port	6060
Use Random SIP Port	☒ No ☐ Yes
SIP T1 Timeout	0.5 sec
SIP T2 Timeout	4 sec
Remove OBP from Route	☒ No ☐ 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

Advanced Port Settings

FXO Mapping

Port Settings

Port	SIP User ID	Authenticate ID	Password	Name	Profile	Enable FXS (TR-069)
FXS 1	123				Profile 1	☐ No ☒ Yes
FXS 2	123				Profile 1	☐ No ☒ Yes
FXS 3	123				Profile 1	☐ No ☒ Yes
FXS 4	123				Profile 1	☐ No ☒ Yes
FXS 5	123				Profile 1	☐ No ☒ Yes
FXS 6	123				Profile 1	☐ No ☒ Yes
FXS 7	123				Profile 1	☐ No ☒ Yes
FXS 8	123				Profile 1	☐ No ☒ Yes
FXS 9	123				Profile 2	☐ No ☒ Yes
FXS 10	123				Profile 2	☐ No ☒ Yes
FXS 11	123				Profile 2	☐ No ☒ Yes
FXS 12	123				Profile 2	☐ No ☒ Yes
FXS 13	123				Profile 2	☐ No ☒ Yes
FXS 14	123				Profile 2	☐ No ☒ Yes
FXS 15	123				Profile 2	☐ No ☒ Yes
FXS 16	123				Profile 2	☐ No ☒ Yes

Save

Save and Apply

Reset

1ST GXW 410X CONFIGURATION:

Advanced Settings:

- STUN server – Blank
- Use Random Port – No

FXO lines:

Channel dialing to PSTN:

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

Channel dialing to VOIP -> Unconditional call forward:

- User ID: ch1-8:123;
- Sip Server: ch1 – 8; p1;
- Sip Destination Port: 5060++
- Number of Rings Before Pickup : ch1 – 8:4;

Profile 1:

- SIP Server: Set it to IP address of GXW42xx
- SIP registration – No
- NAT traversal – No

Channel Dialing to PSTN

- | | | |
|----------------------------------|---|-------------------------------------|
| 1. Wait for Dial-Tone(Y/N): | <input type="text" value="ch1-8:N;"/> | (default No) |
| 2. Stage Method(1/2): | <input type="text" value="ch1-8:1;"/> | (default 2 - 2 stage dialing) |
| 3. Min Delay Before Dialing Out: | <input type="text" value="ch1-8:500;"/> | (default 500ms, range 50 ~ 65000ms) |

Channel Dialing to VoIP

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|--------------------------------|--|-------------------------|
| 1. Unconditional Call Forward: | | |
| User ID: | <input type="text" value="ch1-8:123;"/> | (i.e ch1-2:223;ch3:224) |
| Sip Server: | @ <input type="text" value="ch1-8:p1;"/> | (ch1-2:p1;ch3:p2) |
| Sip Destination Port: | : <input type="text" value="ch1-8:5060++;"/> | (ch1-2:5060;ch2:7080) |

PSTN to VOIP Caller ID Setting

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|-----------------------------------|---------------------------------------|-------------------|
| 1. Number of Rings Before Pickup: | <input type="text" value="ch1-8:4;"/> | (1-50, default 4) |
| 2. Caller ID Scheme: | <input type="text" value="ch1-8:1;"/> | (1-11, default 1) |

2ND GXW 410x CONFIGURATION:

Advanced Settings:

- STUN server – Blank
- Use Random Port – No

FXO lines:

Channel dialing to PSTN:

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

Channel dialing to VOIP -> Unconditional call forward:

- User ID: ch1-8:123;
- Sip Server: ch1 – 8; p1;
- Sip Destination Port: 6060++
- Number of Rings Before Pickup : ch1 – 8:4;

Profile 1:

- SIP Server: Set it to IP address of GXW42xx
- SIP registration – No
- NAT traversal – No

Channel Dialing to PSTN

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|----------------------------------|---|-------------------------------------|
| 1. Wait for Dial-Tone(Y/N): | <input type="text" value="ch1-8:N;"/> | (default No) |
| 2. Stage Method(1/2): | <input type="text" value="ch1-8:1;"/> | (default 2 - 2 stage dialing) |
| 3. Min Delay Before Dialing Out: | <input type="text" value="ch1-8:500;"/> | (default 500ms, range 50 ~ 65000ms) |

Channel Dialing to VoIP

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|--------------------------------|--|-------------------------|
| 1. Unconditional Call Forward: | | |
| User ID: | <input type="text" value="ch1-8:123;"/> | (i.e ch1-2:223;ch3:224) |
| Sip Server: | @ <input type="text" value="ch1-8:p1;"/> | (ch1-2:p1;ch3:p2) |
| Sip Destination Port: | : <input type="text" value="ch1-8:6060++;"/> | (ch1-2:5060;ch2:7080) |

PSTN to VOIP Caller ID Setting

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|-----------------------------------|---------------------------------------|-------------------|
| 1. Number of Rings Before Pickup: | <input type="text" value="ch1-8:4;"/> | (1-50, default 4) |
| 2. Caller ID Scheme: | <input type="text" value="ch1-8:1;"/> | (1-11, default 1) |