

Grandstream Networks, Inc.

Peering GXW42XX FXS Gateway with HT8x1 FXO Gateways
Configuration Guide

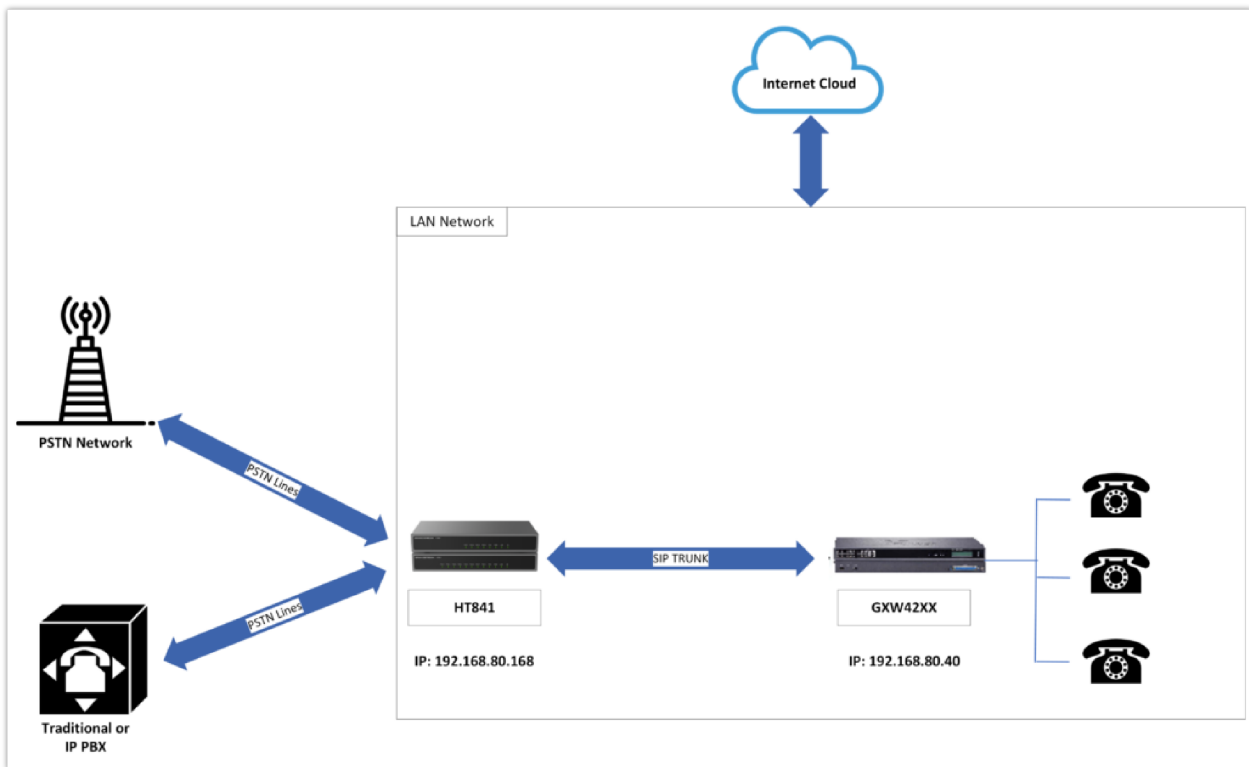


Peering GXW42XX FXS Gateway with HT8x1 FXO Gateways

Introduction

A common scenario which involves a GXW42xx (FXS gateway) connected to an HT841/HT881 (FXO gateway) but doesn't involve any SIP server. This scenario is useful when we want to deploy analog phones in our LAN with different PSTN lines connected to our FXO gateway, without the need to deploy a SIP Server.

The illustration below demonstrates the set up we want to achieve:



GXW42xx & HT8x1 connection

Note

Please note in order for this setup to work, it is important that both the FXO gateway HT841/HT881 and the FXS Gateway GXW42xx are located on the same LAN OR have Public Static IPs. In short, both devices should be able to locate each other.

CONFIGURATION OF THE GXW42XX & MULTIPLE HT841/HT881 SCENARIO

GXW42XX CONFIGURATION

Maintenance – Network Settings

- STUN Server – Blank

STUN Settings	
Use STUN	<input checked="" type="radio"/> No <input type="radio"/> Yes
STUN server	<input type="text"/>
Number of STUN Response Misses Allowed	<input type="text" value="3"/>
Keep-Alive Interval	<input type="text" value="20"/>

STUN Settings

Profiles – Profile 1

General Settings:

- SIP server – Set to IP address of HT8x1, followed by the default listening port for FXO port 1 defined on the HT8x1 which is 6060, we will enter the value 192.168.80.168:6060

Profiles	General Settings
Profile 1 –	<p>Profile Active <input type="radio"/> No <input checked="" type="radio"/> Yes</p> <p>SIP Server <input type="text" value="192.168.80.168:6060"/></p> <p>Failover SIP Server <input type="text"/></p> <p>Prefer Primary SIP Server <input checked="" type="radio"/> No <input type="radio"/> Yes</p> <p>Primary Outbound Proxy <input type="text"/></p> <p>Backup Outbound Proxy <input type="text"/></p> <p>Prefer Primary Outbound Proxy <input checked="" type="radio"/> No <input type="radio"/> Yes</p> <p><input type="button" value="Save"/> <input type="button" value="Save and Apply"/> <input type="button" value="Reset"/></p>
Profile 2 +	
Profile 3 +	
Profile 4 +	

SIP Server set up for HT841

Network Settings:

- NAT traversal – No

NAT Settings	
NAT Traversal	<input type="text" value="No"/>
Use NAT IP	<input type="text"/>
Proxy-Require	<input type="text"/>
<input type="button" value="Save"/> <input type="button" value="Save and Apply"/> <input type="button" value="Reset"/>	

NAT Traversal settings

SIP Settings → Basic Settings:

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

Profiles

Profile 1 -

General Settings

Network Settings

SIP Settings

Basic Settings

Session Timer

Security Settings

Fax Settings

Audio Settings

Call Settings

Call Features Settings

Ring Tones

Call Waiting Tones

Profile 2 +

Profile 3 +

Profile 4 +

Basic Settings

SIP Transport UDP TCP TLS/TCP

SIP Registration No Yes

Unregister on Reboot No Yes

Add Auth Header On Initial REGISTER No Yes

Outgoing Calls Without Registration No Yes

Register Expiration

SIP Registration Failure Retry Wait Time

SIP Registration Failure Retry Wait Time upon 403 Forbidden

Reregister Before Expiration

Enable SIP OPTIONS/NOTIFY Keep Alive No OPTIONS NOTIFY

SIP OPTIONS/NOTIFY Keep Alive Interval

SIP OPTIONS/NOTIFY Keep Alive Max Lost

Local SIP Port

SIP Registration Settings

Note

- If there's a need to set up multiple HT8x1 FXO gateways due to a shortage of FXO ports or any similar configuration requirement, the same setup procedure should be applied to the second HT8x1 device. This involves configuring the SIP trunk with the second HT8x1 FXO gateway, specifically on profile 2 of the GXW42xx.
- GXW42xx can be peered with up to four HT8x1 FXO Gateways, since it supports four profiles to be configured

FXS Ports

Port Settings:

- Port 1 → User ID: 5555 | Authenticate ID: 5555 | Name: 5555 | Profile : Profile 1
- Port 1 → Enable FXS – Yes

This enables the GXW42xx to direct calls between the analog phone linked to port 1 of the GXW42xx via the FXO gateway connected through Profile 1.

- Port 2 → User ID: 7777 | Authenticate ID: 7777 | Name: 7777 | Profile : Profile 2
- Port 2 → Enable FXS – Yes

This enables the GXW42xx to direct calls between the analog phone linked to port 2 of the GXW42xx via the FXO gateway connected through Profile 2

FXS Ports

Port Settings

FXS 1-16

Advanced Port Settings

FXS 1-16

FXO Mapping

FXS 1-16

Port Settings

Port	SIP User ID	Authenticate ID	Password	Name	Profile	Enable FXS (TR-069)
FXS 1	5555	5555			Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes
FXS 2	7777	7777			Profile 2	<input type="radio"/> No <input checked="" type="radio"/> Yes
FXS 3					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes
FXS 4					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes
FXS 5					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes
FXS 6					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes
FXS 7					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes
FXS 8					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes
FXS 9					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes
FXS 10					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes
FXS 11					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes
FXS 12					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes
FXS 13					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes
FXS 14					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes
FXS 15					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes
FXS 16					Profile 1	<input type="radio"/> No <input checked="" type="radio"/> Yes

Save **Save and Apply** **Reset****FXO Mapping**

In this part, you can map FXS1 with FXO1 of the HT8x1 to make sure the call is routed through the correct port, the following fields will be filled

- **Map to FXO Port #** → 1: The FXO port 1 will be mapped to FXS 1
- **Map to FXO Gateway IP:** Selects the IP address of the FXO gateway
- **Port:** Selects the FXO port that will be used to route the call, it is going to be the FXO 1 Port

- FXS Ports
- Port Settings
- FXS 1-16
- Advanced Port Settings
- FXS 1-16
- FXO Mapping
- FXS 1-16

FXO Mapping

Port	Map to FXO Port #	Map to FXO Gateway IP	and Port
FXS 1	1	192.168.80.168	6060
FXS 2	1		5060
FXS 3	1		5060
FXS 4	1		5060
FXS 5	1		5060
FXS 6	1		5060
FXS 7	1		5060
FXS 8	1		5060
FXS 9	1		5060
FXS 10	1		5060
FXS 11	1		5060
FXS 12	1		5060
FXS 13	1		5060
FXS 14	1		5060
FXS 15	1		5060
FXS 16	1		5060

FXO Mapping

HT841/HT881 CONFIGURATION

HT8x1 – Ethernet Settings – Advanced Settings

- STUN server – Blank

STUN settings

STUN server ?

Keep-alive Interval ?

Use STUN to detect network connectivity ?
 No
 Yes

STUN Server

HT8x1 – FXO Profile 1 – Channel Dialing

Dialing to PSTN:

- Wait for dial tone – No
- Stage Method – Setting this parameter to 1 will direct the PSTN call from the VOIP endpoint.

FXO PROFILE 1

General Settings

SIP Settings

Codec Settings

Call Settings

FXO Termination

Channel Dialing

DTMF Digit Length (ms) ?	<input type="text" value="100"/>
DTMF Dial Pause (ms) ?	<input type="text" value="100"/>
First Digit Timeout (sec) ?	<input type="text" value="10"/>
Inter-Digit Timeout (sec) ?	<input type="text" value="4"/>
Wait for Dial-Tone ?	<input checked="" type="radio"/> No <input type="radio"/> Yes
Stage Method (1/2) ?	<input type="text" value="1"/>
Min Delay Before Dial PSTN Number ?	<input type="text" value="500"/>

Channel dialing settings

Note

- **Enabled (Wait for Dial Tone):** Gateway waits for a dial tone before dialing. Suitable for lines with dial tones; users dial after hearing it.
- **Disabled (No Wait for Dial Tone):** Gateway doesn't wait for a dial tone. Useful when no dial tone or automated dialing is needed.

HT8x1 – Ports – Unconditional Call Forward to VoIP

Calling to VoIP:

- User ID: 7000
- Sip Server: 192.168.80.40 (IP Address of the FXS Gateway)
- Sip Destination Port: 5060

Unconditional Call Forward to VOIP				
FXO PORTS	CID		Sip Server	Sip Destination Port
1	<input type="text" value="7000"/>	@	<input type="text" value="192.168.80.40"/>	<input type="text" value="5060"/>
2	<input type="text"/>	@	<input type="text"/>	<input type="text"/>
3	<input type="text"/>	@	<input type="text"/>	<input type="text"/>
4	<input type="text"/>	@	<input type="text"/>	<input type="text"/>

Unconditional Call Forward to VoIP settings

HT8x1 – FXO Profile 1 – FXO Termination

- Set the following:
 1. **Number of Rings** → 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.
 2. **PSTN Ring Thru FXS** → No
Disable this option to prevent calls from being routed through the FXS port.

Number of Rings

PSTN Ring Thru FXS No Yes

FXO Termination Settings

HT8x1 – FXO Profile 1 – SIP Settings

General Settings:

- SIP Server: Set it to IP address of GXW42xx

SIP Settings:

- SIP registration – No

Network Settings:

- NAT traversal – No

FXO PROFILE 1

General Settings | SIP Settings | Codec Settings | Call Settings | FXO Termination | Channel Dialing

Account Registration

Profile Active No Yes

Primary SIP Server

Fallover SIP Server

Prefer Primary SIP Server

Outbound Proxy

Backup Outbound Proxy

Prefer Primary Outbound Proxy No Yes

From Domain

Account Registration

FXO PROFILE 1

General Settings | SIP Settings | Codec Settings | Call Settings | FXO Termination | Channel Dialing

SIP Basic Settings

SIP Registration No Yes

SIP Transport UDP TCP TLS

Unregister On Reboot No All Instance

Outgoing Call without Registration No Yes

SIP Settings

Register Before DNS SRV Failover No Yes

Primary IP

Backup IP1

Backup IP2

NAT Traversal

NAT Traversal Settings

Results

After the configuration is complete between the GXW42xx FXS Gateway and HT841/HT881 FXO Gateways, users from inside the LAN can use their analog phones connected to the GXW42xx FXS gateways to reach outside PSTN lines, without the need to deploy any SIP server, it only requires to set up peer trunk between the GXW42xx FXS Gateway and the HT841/HT881 FXO Gateways.

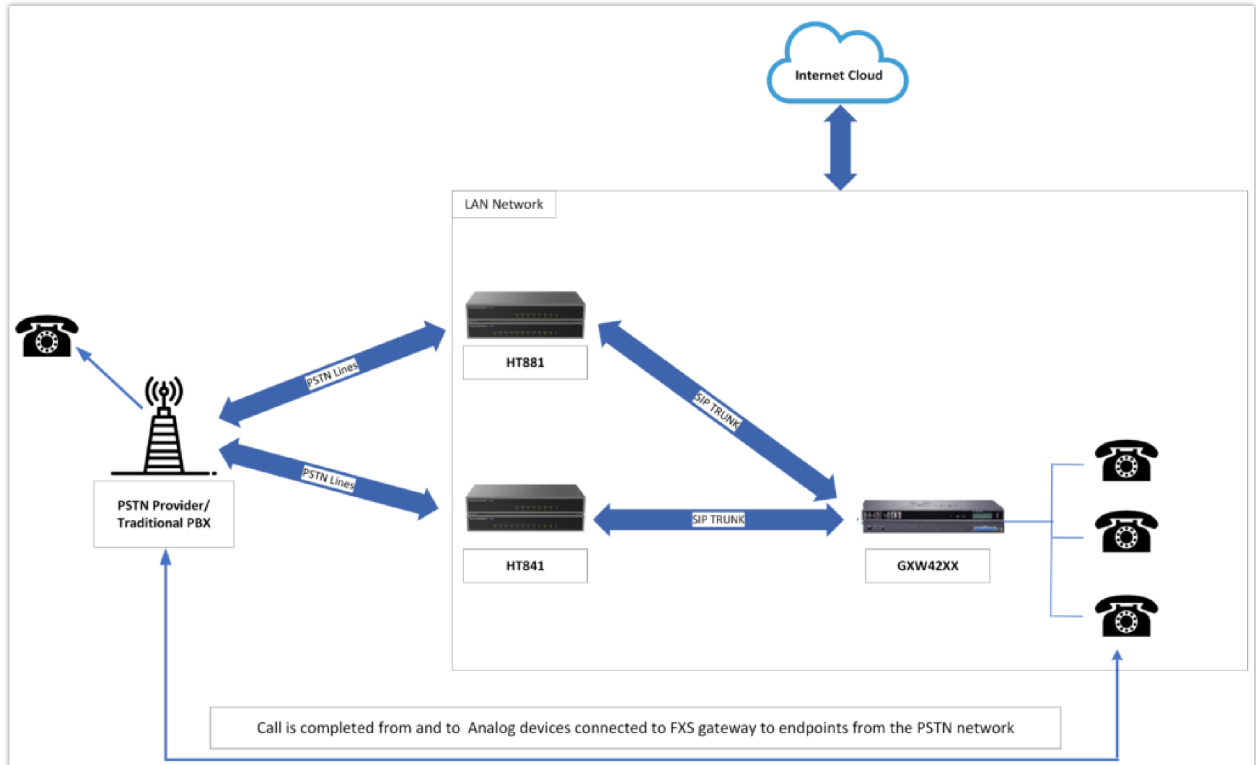


Diagram of the connection results

Supported Devices

Device	Firmware Required
HT841	1.0.1.2+
HT881	1.0.1.2+
GXW4216 v1	1.0.23.7+
GXW4224 v1	1.0.23.7+
GXW4232 v1	1.0.23.7+
GXW4248 v1	1.0.23.7+
GXW4216 v2	1.0.21.2+
GXW4224 v2	1.0.21.2+
GXW4232 v2	1.0.21.2+
GXW4248 v2	1.0.21.2+

Need Support?

Can't find the answer you're looking for? Don't worry we're here to help!

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