

Grandstream Networks, Inc.

Peering GXW42XX FXS Gateway with HT8x1 FXO Gateways

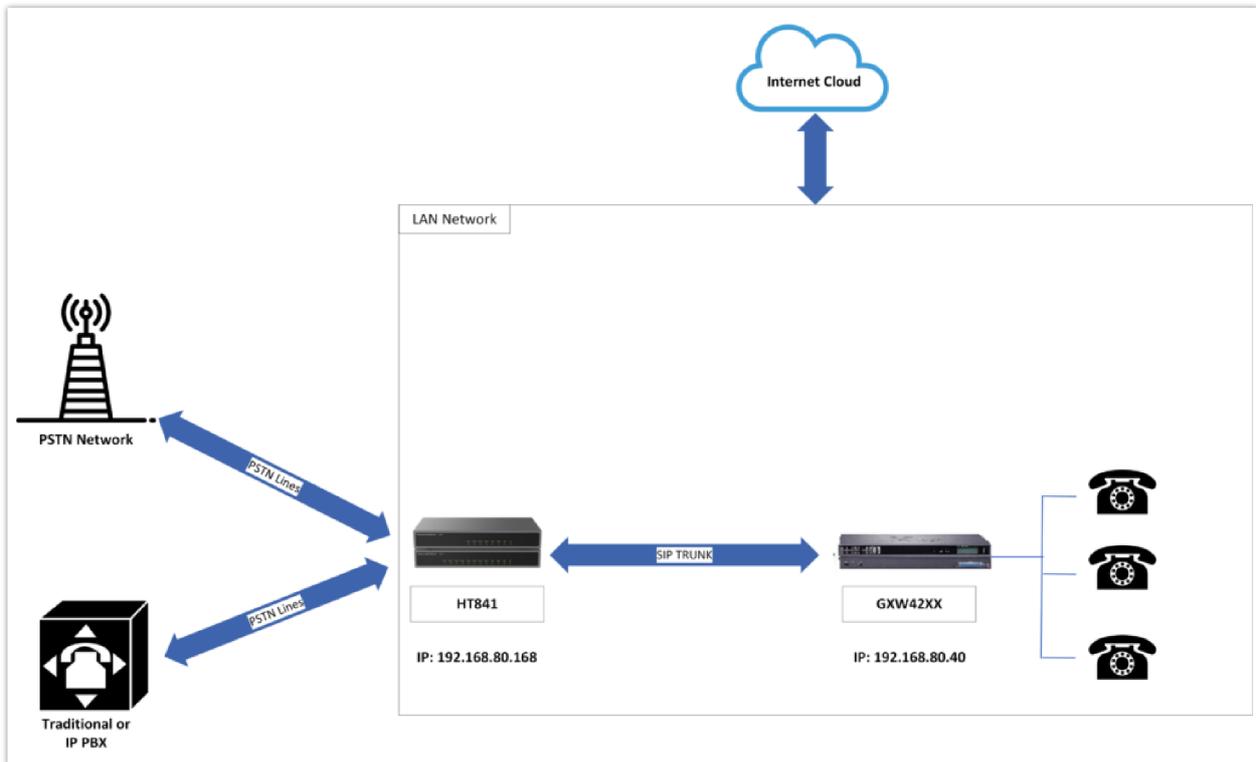


Peering GXW42XX FXS Gateway with HT8x1 FXO Gateways

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 No Yes

STUN server

Number of STUN Response Misses Allowed

Keep-Alive Interval

STUN Settings

Profiles – Profile 1

General Settings:

- SIP server – Set to IP address of HT8x1, followed by the default listening port for FXO port defined on the HT8x1, we will enter the value 192.168.80.168:6062

Profiles

Profile 1 –

General Settings

Network Settings

SIP Settings +

Fax Settings

Audio Settings

Call Settings

Call Features Settings

Ring Tones

Call Waiting Tones

Profile 2 +

Profile 3 +

Profile 4 +

General Settings

Profile Active No Yes

SIP Server

Failover SIP Server

Prefer Primary SIP Server No Yes

Primary Outbound Proxy

Backup Outbound Proxy

Prefer Primary Outbound Proxy No Yes

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SIP Server set up for HT841

Network Settings:

- NAT traversal – No

NAT Settings

NAT Traversal

Use NAT IP

Proxy-Require

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	6062
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

HT841/HT881 CONFIGURATION

HT8x1 – Advanced Settings

- STUN server – Blank

	802.1Q/VLAN Tag	<input type="text" value="0"/>	(0-4094)
<i>Layer 2 QoS:</i>	SIP 802.1p	<input type="text" value="0"/>	(0-7)
	RTP 802.1p	<input type="text" value="0"/>	(0-7)
<i>Black List for WAN Side Port:</i>	<input type="text"/>		
<i>STUN server is:</i>	<input type="text"/> (URI or IP:port)		
<i>Keep-alive Interval:</i>	<input type="text" value="20"/>	(in seconds, default 20 seconds)	
<i>Use STUN to detect network connectivity:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes, total STUN response misses <input type="text" value="3"/> to restart DHCP (minimum=3)		
<i>Use DNS to detect network connectivity:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes		

Stun Server

HT8x1 – FXO Profile 1 – Chaneel 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.

Channel Dialing	
<i>DTMF Digit Length (ms):</i>	<input type="text" value="100"/> (40-127 milliseconds, Default 100 milliseconds)
<i>DTMF Dial Pause (ms):</i>	<input type="text" value="100"/> (40-127 milliseconds, Default 100 milliseconds)
<i>First Digit Timeout (sec):</i>	<input type="text" value="10"/> (1-20 seconds. Default 10 seconds)
<i>Inter-Digit Timeout (sec):</i>	<input type="text" value="4"/> (1-15 seconds. Default 4 seconds)
<i>Wait for Dial-Tone:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes (Default Yes - dial upon dial-tone)
<i>Stage Method (1/2):</i>	<input type="text" value="1"/> (Default 2 - 2 stage dialing)
<i>Min Delay Before Dial PSTN Number:</i>	<input type="text" value="500"/> (default 500ms, range 50 ~ 65000ms)

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

Port	User ID	Sip Server	Sip Destination Port
1	7000	@ 192.168.80.40	: 5060
2		@	:
3		@	:
4		@	:
5		@	:
6		@	:
7		@	:
8		@	:

Unconditional Call Forward to VoIP settings

HT8x1 – FXO Profile 1 – FXO Termination

o 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.

<i>Number of Rings:</i>	<input type="text" value="4"/>	(1-50. Default 4)
	(Number of rings for a PSTN incoming call before FXO port answers to accept VoIP number. Warning: If set to 1, it may affect caller ID detection)	
<i>PSTN Ring Thru FXS:</i>	<input checked="" type="radio"/> No	<input type="radio"/> Yes (Default Yes)
	(If set to yes, all incoming PSTN calls will ring the FXS port after the Ring Thru Delay)	

Number of Rings 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

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS FXS PROFILE FXO PROFILE 1 FXO PROFILE 2 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
 Will register to Primary Server if Failover registration expires
 Will register to Primary Server if Primary Server responds, need to enable SIP

OPTIONS/NOTIFY Keep Alive

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)

From Domain: (Optional, actual domain name, will override the from header)

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 Auto VPN

DNS Mode: A Record SRV NAPTR/SRV Use Configured IP

DNS SRV Failover Mode:

Failback Timer: (in minutes. default 60 minutes, max 45 days)

Register Before DNS SRV Failover: No Yes

Primary IP:

Backup IP1:

Backup IP2:

Tel URI:

SIP Registration: No Yes

Unregister On Reboot: No All Instance

Outgoing Call without Registration: No Yes

Register Expiration: (in minutes. default 1 hour, max 45 days)

Reregister before Expiration: (0-64800. Default 0 second)

SIP Registration Failure Retry Wait Time: (in seconds. Between 1-3600, default is 20)

SIP 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.

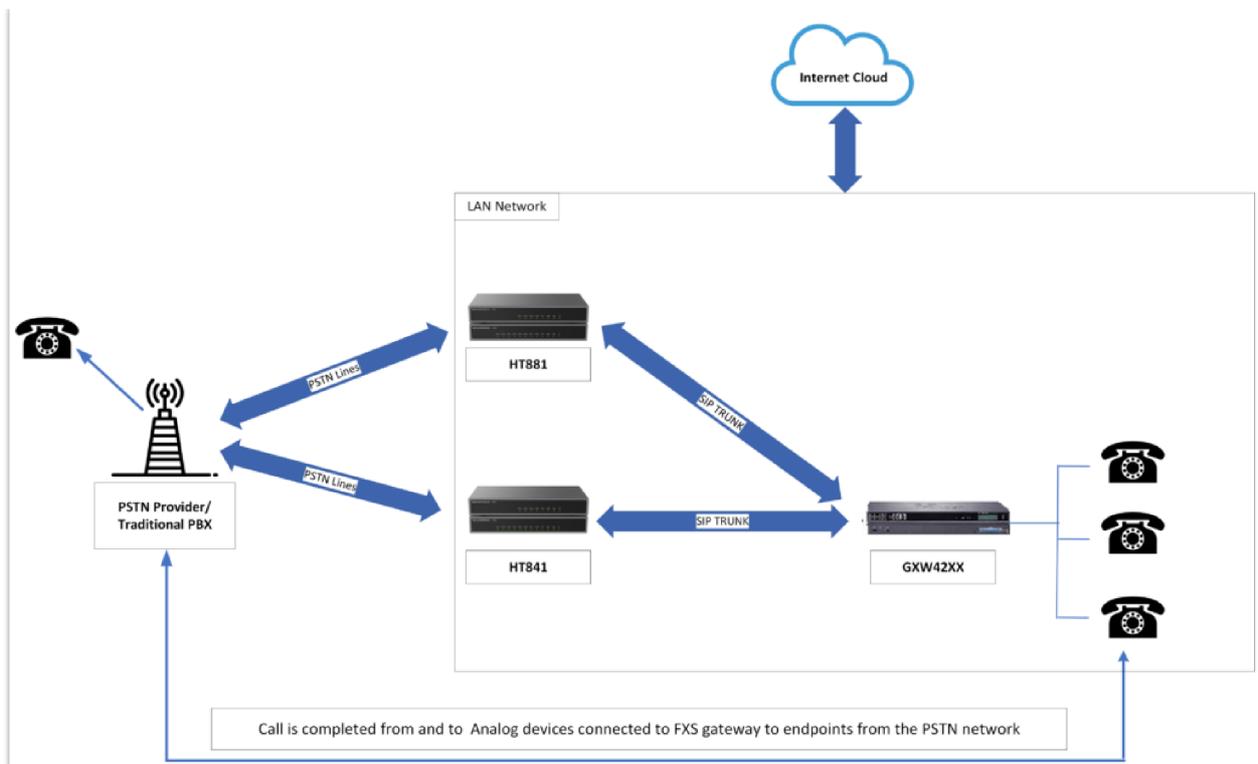


Diagram of the connection results

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+

Supported devices

Need Support?

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

