



Grandstream Networks, Inc.

Peering HT8XX with HT813



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OVERVIEW

This document describes basic configuration to peer HT8XX series with HT813. This configuration applies to users seeking to add a HT813 not only as a remote extension but also as an external PSTN trunk.

The document will demonstrate a scenario where you can set up HT8XX series with the HT813.

PEERING HT8XX WITH HT813

A common scenario which involves one HT8XX (ATA) and HT813 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 HT813 in order to configure the Peer Trunk on both sides

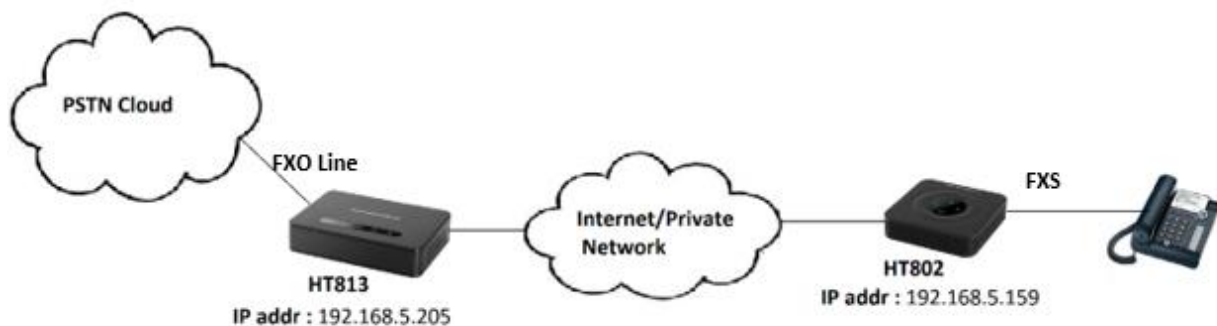


Figure 1: Peering HT8xx with HT813

Note: HT8XX can include HT801/802/814/812/818

HT8XX Configuration

Navigate to web GUI of HT8XX access to "FXS Port", then set the following:

- **Primary SIP Server:** Set to $\langle IP_Address_of_HT-813 \rangle : 5062$, which is in our case: 192.168.5.205:5062 (5062 is the default listening port for FXO on HT813).
- **SIP User ID:** Any Number, in our case it will be 5555.
- **Authenticate ID:** Any Number, in our case it will be 5555.
- **SIP Registration:** No.
- **Outgoing Call without Registration:** Yes.



Grandstream Device Configuration	
STATUS	BASIC SETTINGS
Account Active: <input type="radio"/> No <input checked="" type="radio"/> Yes	
Primary SIP Server: <input type="text" value="192.168.5.205:5062"/> (e.g., sip.mycompany.com, or IP address)	
Failover SIP Server: <input type="text"/> (Optional, used when primary server no response)	
Prefer Primary SIP Server: <input checked="" type="radio"/> No <input type="radio"/> Yes (yes - will register to Primary Server if Failover registration expires)	
Outbound Proxy: <input type="text"/> (e.g., proxy.myprovider.com, or IP address, if any)	
Backup Outbound Proxy: <input type="text"/> (e.g., proxy.myprovider.com, or IP address, if any)	
Prefer Primary Outbound Proxy: <input checked="" type="radio"/> No <input type="radio"/> Yes (yes - will reregister via Primary Outbound Proxy if registration expires)	
Allow DHCP Option 120 (override SIP server): <input checked="" type="radio"/> No <input type="radio"/> Yes	
SIP Transport: <input checked="" type="radio"/> UDP <input type="radio"/> TCP <input type="radio"/> TLS (default is UDP)	
SIP URI Scheme When Using TLS: <input type="radio"/> sip <input checked="" type="radio"/> sips	
Use Actual Ephemeral Port in Contact with TCP/TLS: <input checked="" type="radio"/> No <input type="radio"/> Yes	
NAT Traversal: <input checked="" type="radio"/> No <input type="radio"/> Keep-Alive <input type="radio"/> STUN <input type="radio"/> UPnP	
SIP User ID: <input type="text" value="5555"/> (the user part of an SIP address)	
Authenticate ID: <input type="text" value="5555"/> (can be identical to or different from SIP User ID)	
Authenticate Password: <input type="text"/> (purposely not displayed for security protection)	
Name: <input type="text"/> (optional, e.g., John Doe)	
DNS Mode: <input checked="" type="radio"/> A Record <input type="radio"/> SRV <input type="radio"/> NAPTR/SRV	
DNS SRV use Registered IP: <input checked="" type="radio"/> No <input type="radio"/> Yes	
Tel URI: <input type="text" value="Disabled"/>	
SIP Registration: <input checked="" type="radio"/> No <input type="radio"/> Yes	
Unregister On Reboot: <input checked="" type="radio"/> No <input type="radio"/> Yes	
Outgoing Call without Registration: <input type="radio"/> No <input checked="" type="radio"/> Yes	
Register Expiration: <input type="text" value="60"/> (in minutes. default 1 hour, max 45 days)	
Reregister before Expiration: <input type="text" value="0"/> (0-64800. Default 0 second)	
SIP Registration Failure Retry Wait Time: <input type="text" value="20"/> (in seconds. Between 1-3600, default is 20)	
SIP Registration Failure Retry Wait Time upon 403 Forbidden: <input type="text" value="1200"/> (in seconds. Between 0-3600, default is 1200. 0 means stop retry registration upon 403 response.)	
Enable SIP OPTIONS/NOTIFY Keep Alive: <input checked="" type="radio"/> No <input type="radio"/> OPTIONS <input type="radio"/> NOTIFY	

Figure 2: FXS Port settings

Notes:

- SIP User ID and Authenticate ID should be the same.
- Always set Random Ports to “No”.



HT813 Configuration

On the HT813 web GUI, access to “FXO Port”, then set the following:

- **Primary SIP Server:** Set to *<IP_address_of_HT-802>*, which is in our case: 192.168.5.159.
- **SIP User ID:** Any Number, in our case it will be 6666.
- **Authenticate ID:** Any Number, in our case it will be 6666.
- **SIP Registration:** No
- **Outgoing Call without Registration:** Yes
- **Number of Rings:** 1
- **PSTN Ring Thru FXS:** No
- **Wait for Dial Tone:** No
- **Stage Method:** 1

Grandstream Device Configuration

STATUS
BASIC SETTINGS
ADVANCED SETTINGS
FXS PORT
FXO PORT

Account 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)
SIP Transport: UDP TCP TLS (default is UDP)
NAT Traversal: No Keep-Alive STUN UPnP

SIP User ID: (the user part of an SIP address)
Authenticate ID: (can be identical to or different from SIP User ID)
Authenticate Password: (purposely not displayed for security protection)
Name: (optional, e.g., John Doe)

DNS Mode: A Record SRV NAPTR/SRV
DNS SRV use Registered IP: No Yes
Tel URI:

SIP Registration: No Yes
Unregister On Reboot: No Yes
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 Registration Failure Retry Wait Time upon 403 Forbidden: (in seconds. Between 0-3600, default is 1200. 0 means stop retry registration upon 403 response.)
Enable SIP OPTIONS Keep Alive: No Yes
SIP OPTIONS Keep Alive Interval: (in seconds. Between 1-64800, default is 30)
SIP OPTIONS Keep Alive Max Lost: (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3)

Figure 3: FXO Port settings



AC Termination Model Country-based Impedance-based Auto-Detected
Country-based USA
Impedance-based 600R -- 600 ohms

Number of Rings: 1 (1-50. Default 4)
 (Number of rings for a PSTN incoming call before FXO port answers to accept VoIP number)

PSTN Ring Thru FXS: No Yes (Default Yes)
 (If set to yes, all incoming PSTN calls will ring the FXS port after the Ring Thru Delay)

PSTN Ring Thru Delay (sec): 4 (1-10 seconds. Default 4 seconds)

PSTN Ring Timeout (sec): 6 (2-10 seconds. Default 6 seconds)
 (Used to detect PSTN hangup when FXO port is not answered)

PSTN Idle Wait Timeout between Outgoing Calls: 4 (0-10 seconds. Default 4 seconds)

Channel Dialing

DTMF Digit Length (ms): 100 (40-127 milliseconds, Default 100 milliseconds)

DTMF Dial Pause (ms): 100 (40-127 milliseconds, Default 100 milliseconds)

First Digit Timeout (sec): 10 (1-20 seconds. Default 10 seconds)

Inter-Digit Timeout (sec): 4 (1-15 seconds. Default 4 seconds)

Wait for Dial-Tone: No Yes (Default Yes - dial upon dial-tone)

Stage Method (1/2): 1 (Default 2 - 2 stage dialing)

Min Delay Before Dial PSTN Number: 500 (default 500ms, range 50 ~ 65000ms)

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Figure 4: FXO Port settings

Notes:

- SIP User ID and Authenticate ID Should be the same
- Stage Method 2 doesn't apply for peer to peer. It works when registered with a SIP Server.
- Always set Random Ports to "No".

On the HT813 web GUI, access to "Basic Settings", then set the following:

- **Unconditional Call Forward to VOIP:** Must have a User ID (Could be Any).



<i>PSTN Access Code:</i>	<input type="text" value="*00"/>	(Key pattern to use PSTN line. Maximum 5 digits. Default is "*00")
<i>PIN for VoIP-to-PSTN Calls:</i>	<input type="text"/>	(Maximum 8 digits to authorize calling PSTN numbers from VoIP. No default)
<i>PIN for PSTN-to-VoIP Calls:</i>	<input type="text"/>	(Maximum 8 digits to authorize calling VOIP terminals from PSTN. No default)
<i>Unconditional Call Forward to PSTN:</i>	<input type="text"/>	(VoIP calls will be forwarded to the specified PSTN number)

<i>Unconditional Call Forward to VOIP:</i>	<input type="text" value="7000"/>	@	<input type="text" value="192.168.5.159"/>	:	<input type="text" value="5060"/>
	User ID		Sip Server		Sip Destination Port

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Figure 5: Basic settings configuration

Note:

In order for this setup to work, it is extremely important that both Handy Tones (HT813 and HT8xx) are located on the same LAN OR have Public Static IPs. In short, the Handy Tones should be able to locate each other.

