



Grandstream Networks, Inc.

Peering IP Phone with HT813



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OVERVIEW

This document describes basic configuration to peer an IP Phone 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 GXP/GRP series with the HT813.

PEERING IP PHONE WITH HT813

A common scenario which involves one IP Phone 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 GXP Phone, then on the HT813 in order to configure the Peer Trunk on both sides.

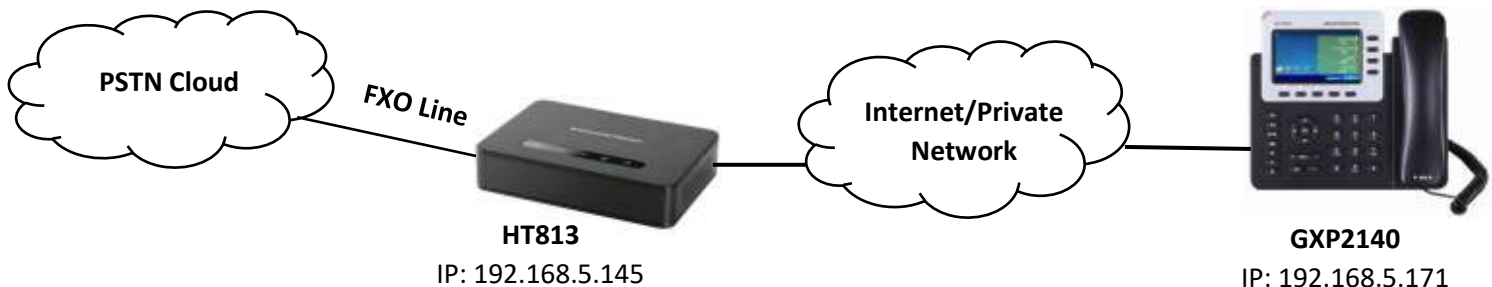


Figure 1: Peering IP Phone with HT813

Note: We will be using a GXP2140 as example in this document.

GXP IP Phone Configuration

Navigate to web GUI of GXP access to Accounts → Account 1 → General Settings, then set the following:

- **Primary SIP Server:** Set to *<IP_Address_of_HT-813>:5062*, which is in our case: 192.168.5.145:5062 (5062 is the default listening port for FXO on HT813).
- **SIP User ID:** Any Number, in our case it will be 6666.
- **Authenticate ID:** Any Number, in our case it will be 6666.

Under Accounts → Account 1 → SIP Settings → Basic Settings:

- **SIP Registration:** No.



General Settings

Account Active	<input type="radio"/> No <input checked="" type="radio"/> Yes
Account Name	<input type="text" value="192.168.5.145:5062"/>
SIP Server	<input type="text"/>
Secondary SIP Server	<input type="text"/>
Outbound Proxy	<input type="text"/>
Backup Outbound Proxy	<input type="text"/>
BLF Server	<input type="text"/>
SIP User ID	<input type="text" value="6666"/>
Authenticate ID	<input type="text" value="6666"/>
Authenticate Password	<input type="text"/>
Name	<input type="text"/>
Voice Mail Access Number	<input type="text"/>
Picture	<input type="button" value="Select"/>
Account Display	<input checked="" type="radio"/> User Name <input type="radio"/> User ID

Figure 2: SIP account Configuration

Basic Settings

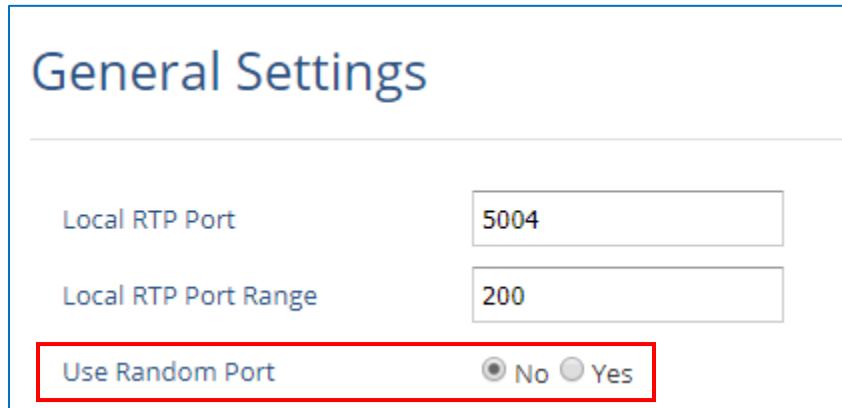
TEL URI	<input checked="" type="radio"/> Disabled <input type="radio"/> User=phone <input type="radio"/> Enabled
SIP Registration	<input checked="" type="radio"/> No <input type="radio"/> Yes

Figure 3: Basic Settings



Notes:

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



The screenshot shows the 'General Settings' page. It contains three input fields: 'Local RTP Port' with the value '5004', 'Local RTP Port Range' with the value '200', and 'Use Random Port' with radio buttons for 'No' (selected) and 'Yes'. A red rectangular box highlights the 'Use Random Port' section.

Figure 4: Disable Use Random Port

HT813 Configuration

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

- **Primary SIP Server:** Set to *<IP_address_of_GXPphone>*, which is in our case: 192.168.5.171
- **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
- **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)

Figure 5: 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)

Update Apply Cancel Reboot

Figure 6: 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>	<table border="1"> <thead> <tr> <th>User ID</th> <th>Sip Server</th> <th>Sip Destination Port</th> </tr> </thead> <tbody> <tr> <td><input type="text" value="6000"/></td> <td>@ <input type="text" value="192.168.5.171"/></td> <td>: <input type="text" value="5060"/></td> </tr> </tbody> </table>	User ID	Sip Server	Sip Destination Port	<input type="text" value="6000"/>	@ <input type="text" value="192.168.5.171"/>	: <input type="text" value="5060"/>	
User ID	Sip Server	Sip Destination Port						
<input type="text" value="6000"/>	@ <input type="text" value="192.168.5.171"/>	: <input type="text" value="5060"/>						

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Figure 7: Basic settings configuration

Notes:

- 5060 is the default listening port for Account1 on GXP2140.
- In order for this setup to work, it is extremely important that both the Handy Tone HT813 and the IP phone are located on the same LAN OR have Public Static IPs. In short, the Handy Tones should be able to locate each other.

