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

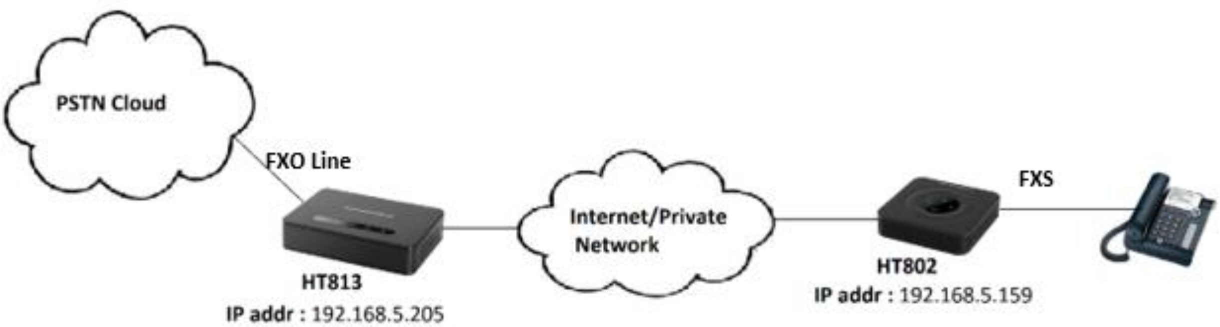


Figure 1: Peering HT8xx with HT813

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 *<IP_Address_of_HT-813>: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.

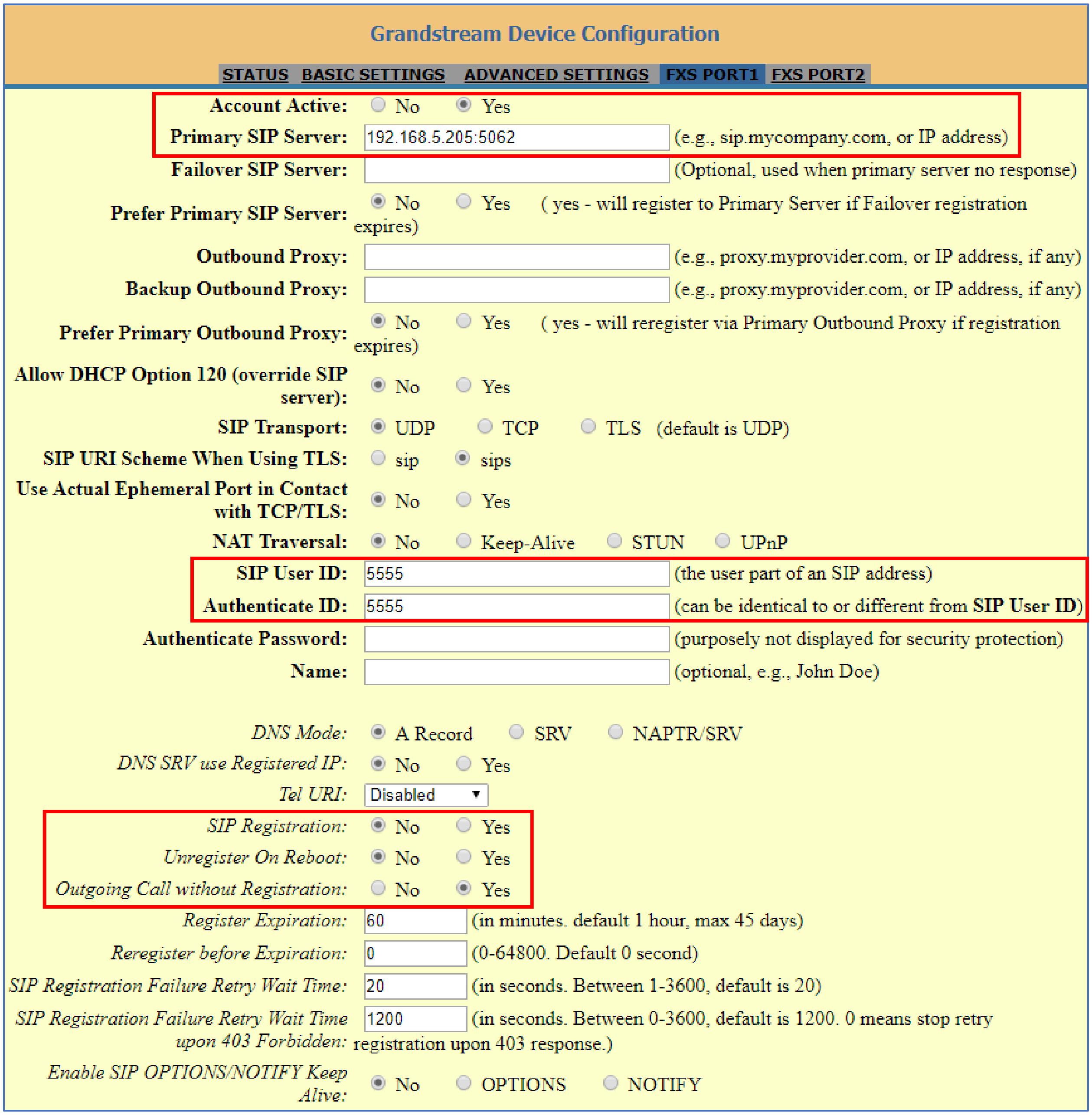


Figure 2: FXS Port settings

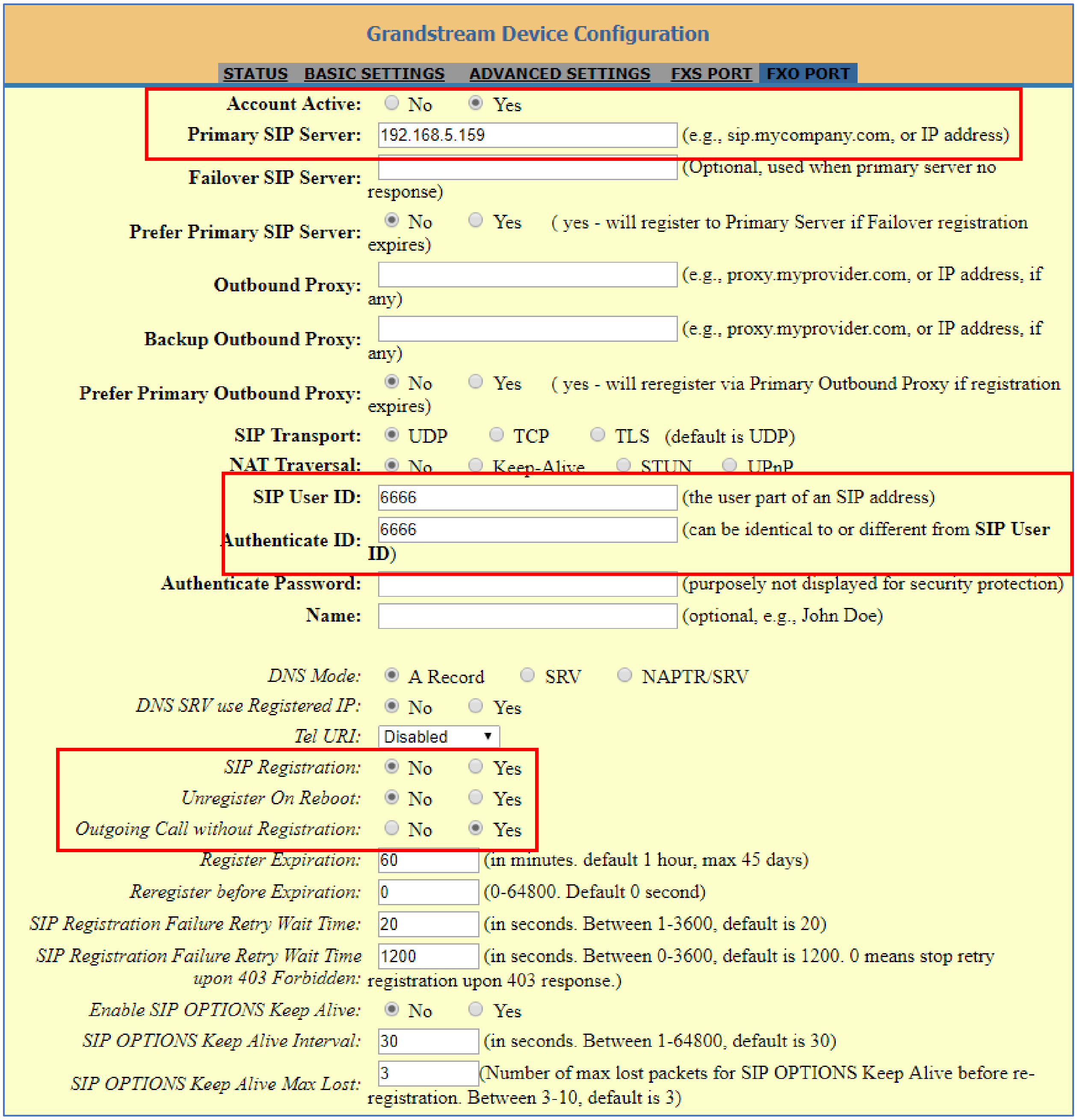
Notes:

- 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:	192.168.5.159 (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:	6666 (the user part of an SIP address)
Authenticate ID:	6666 (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:	Disabled
SIP Registration:	◉ No ○ Yes
Unregister On Reboot:	◉ No ○ Yes
Outgoing Call without Registration:	○ No ◉ Yes
Register Expiration:	60 (in minutes. default 1 hour, max 45 days)
Reregister before Expiration:	0 (0-64800. Default 0 second)
SIP Registration Failure Retry Wait Time:	20 (in seconds. Between 1-3600, default is 20)
SIP Registration Failure Retry Wait Time upon 403 Forbidden:	1200 (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:	30 (in seconds. Between 1-64800, default is 30)
SIP OPTIONS Keep Alive Max Lost:	3 (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3)

Figure 3: FXO Port settings

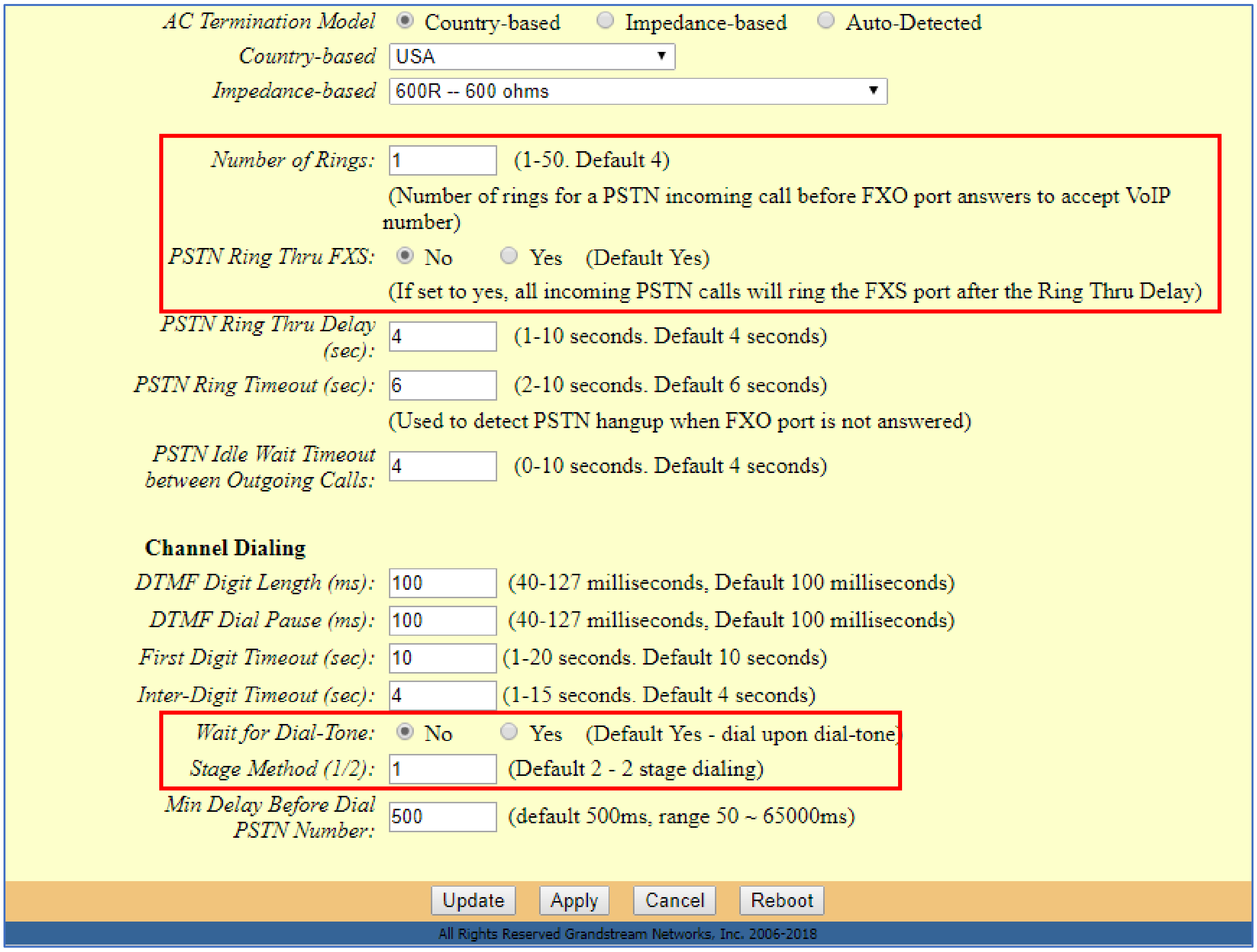


Figure 4: FXO Port settings

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).

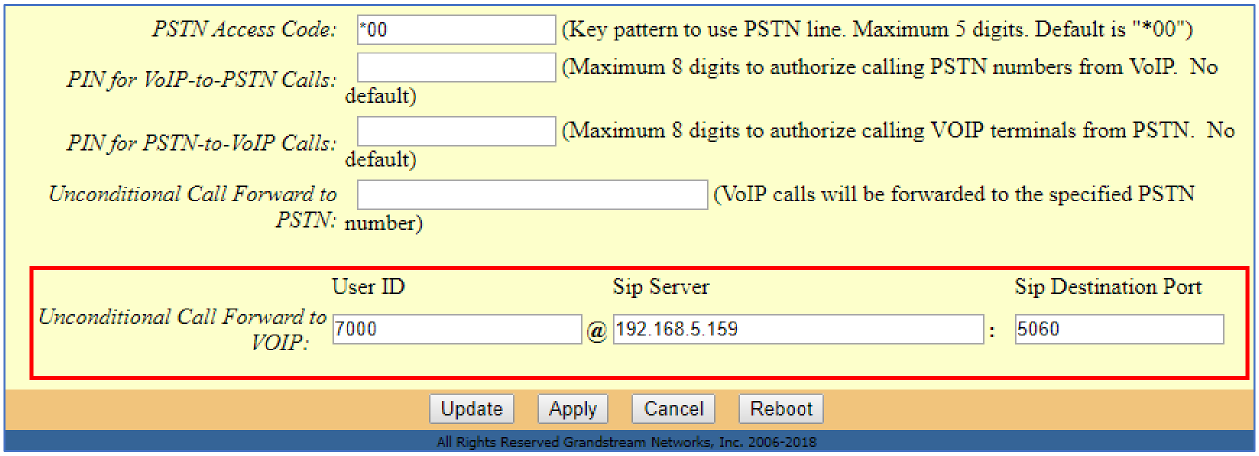


Figure 5: Basic settings configuration

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.