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.

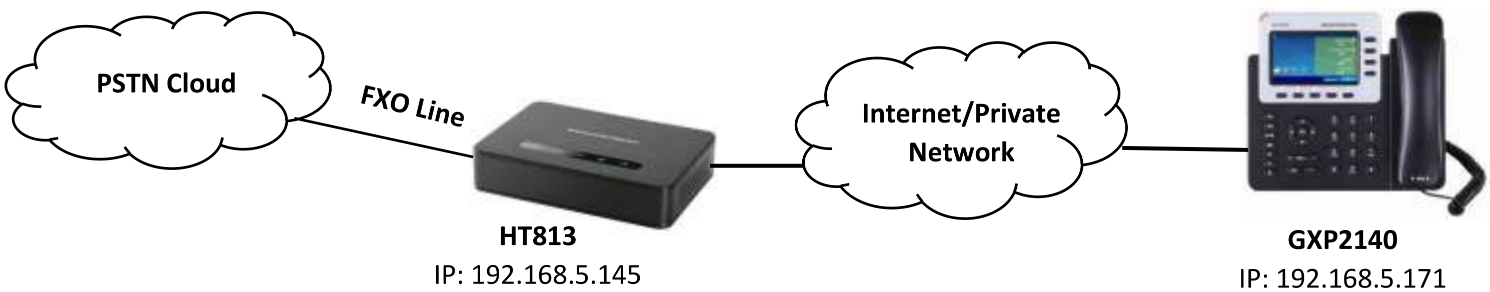

Figure 1: Peering IP Phone with HT813

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.

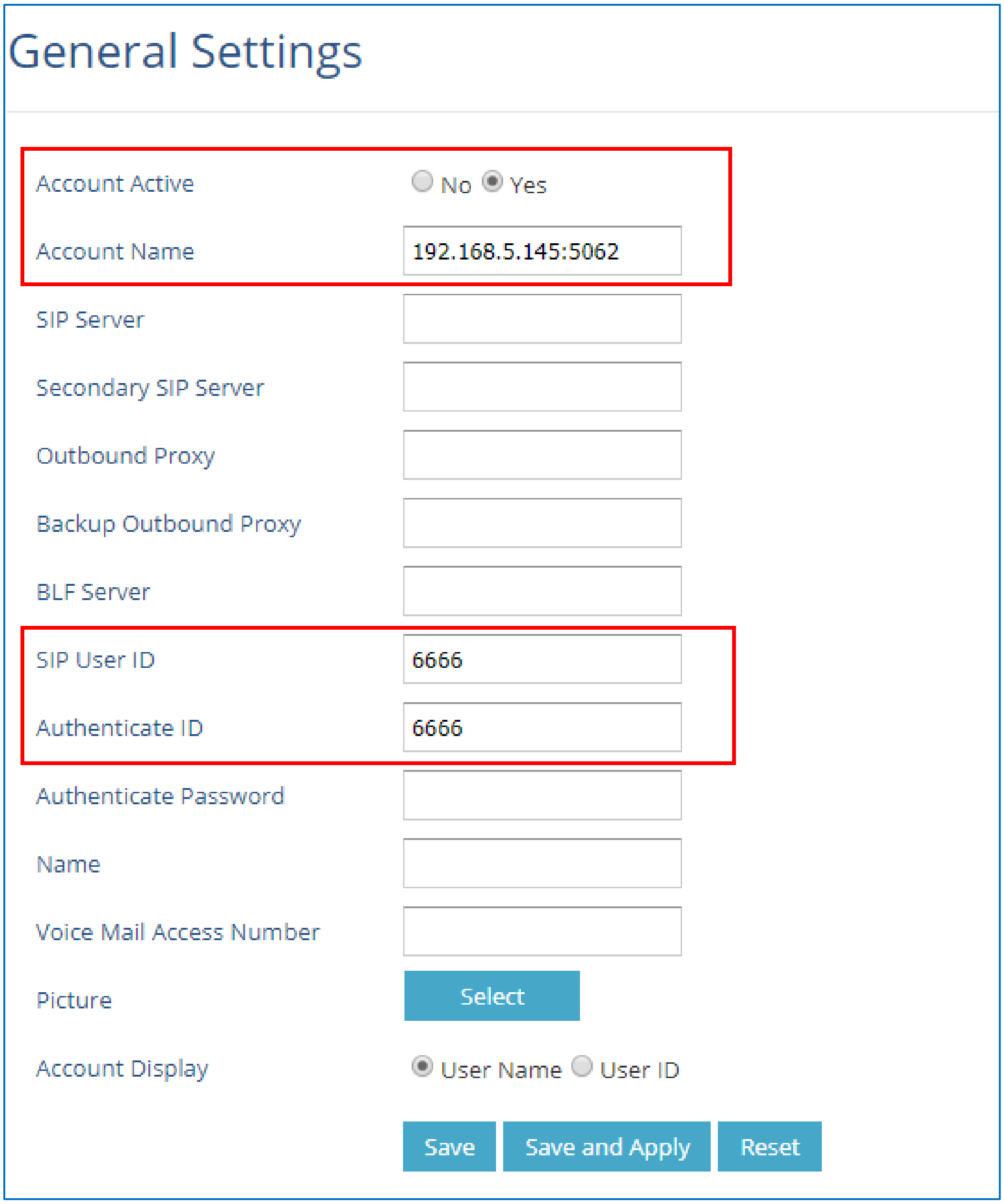

Figure 2: SIP account Configuration

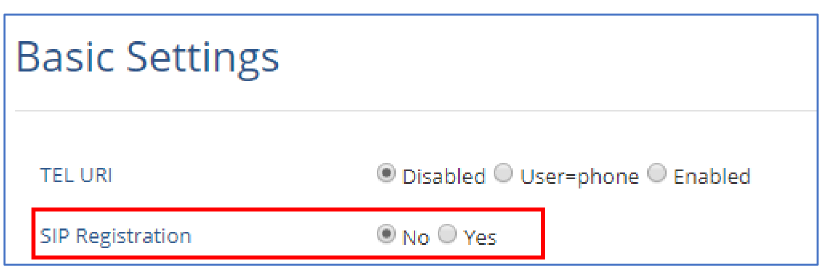

Figure 3: Basic Settings

Notes:

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

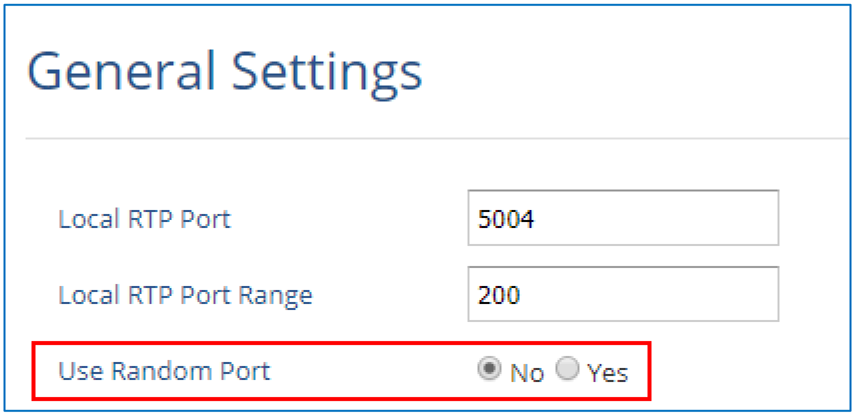

Figure 4: Disable Use Random Port

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

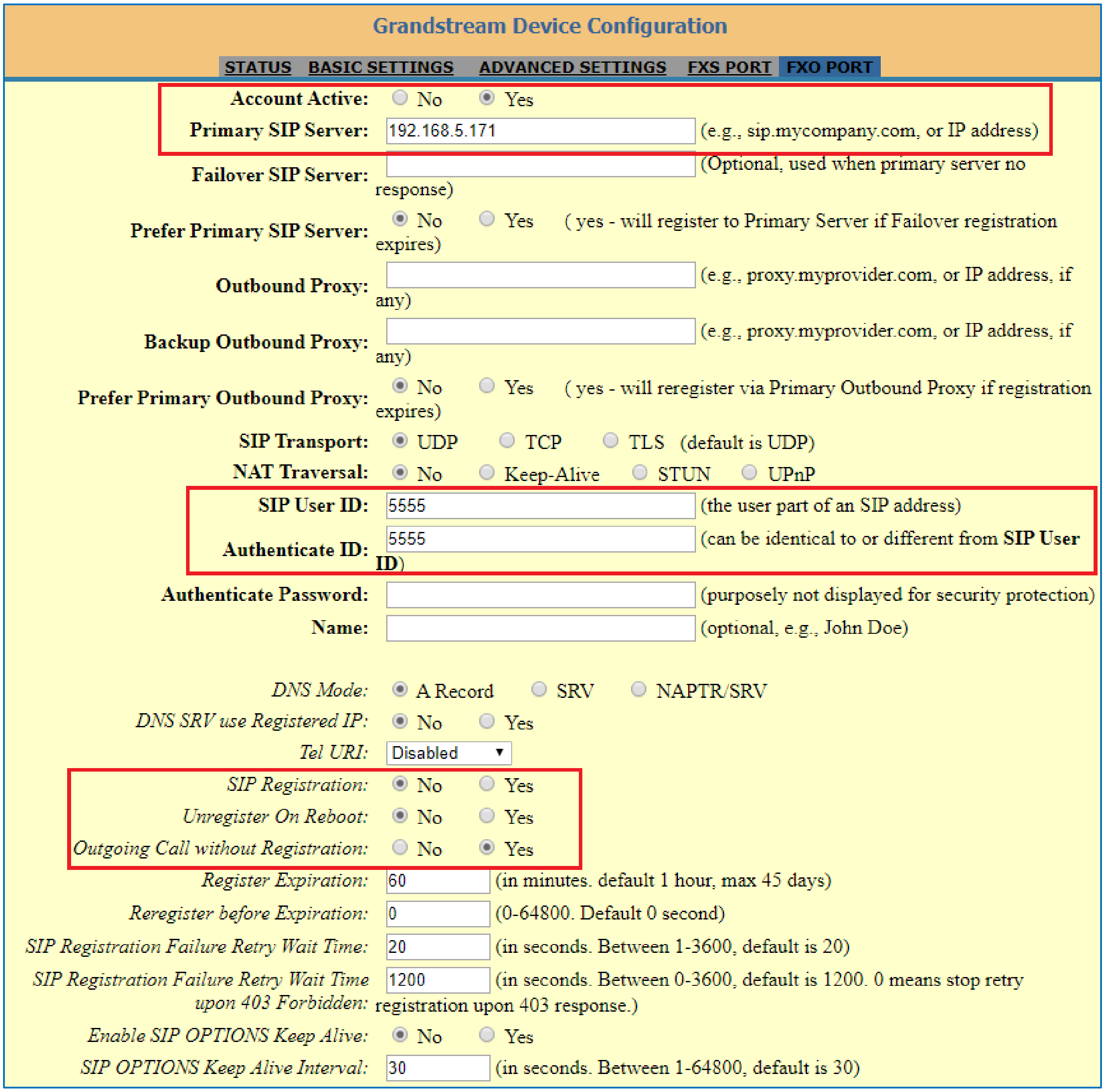

Figure 5: FXO Port settings

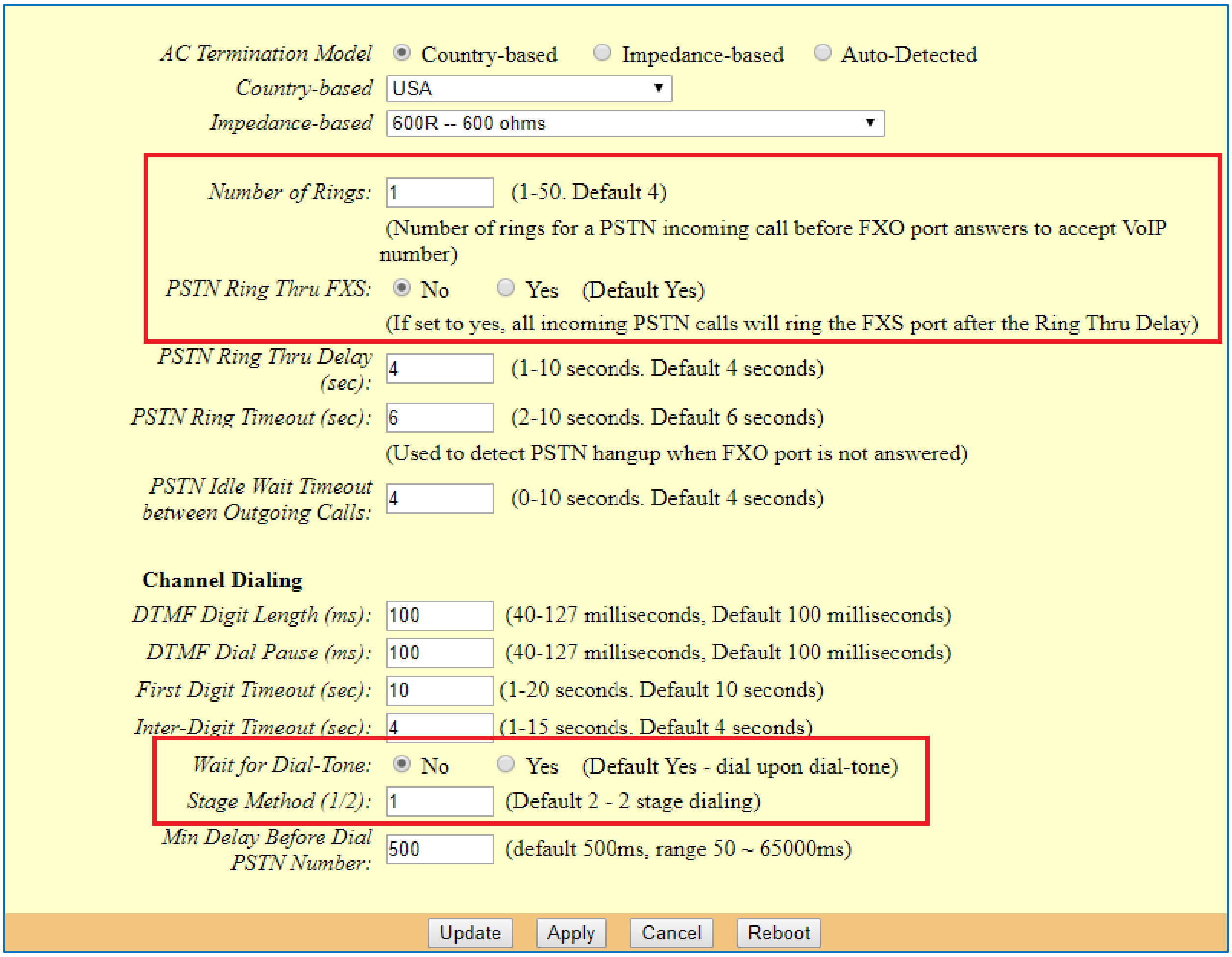

Figure 6: FXO Port settings

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).

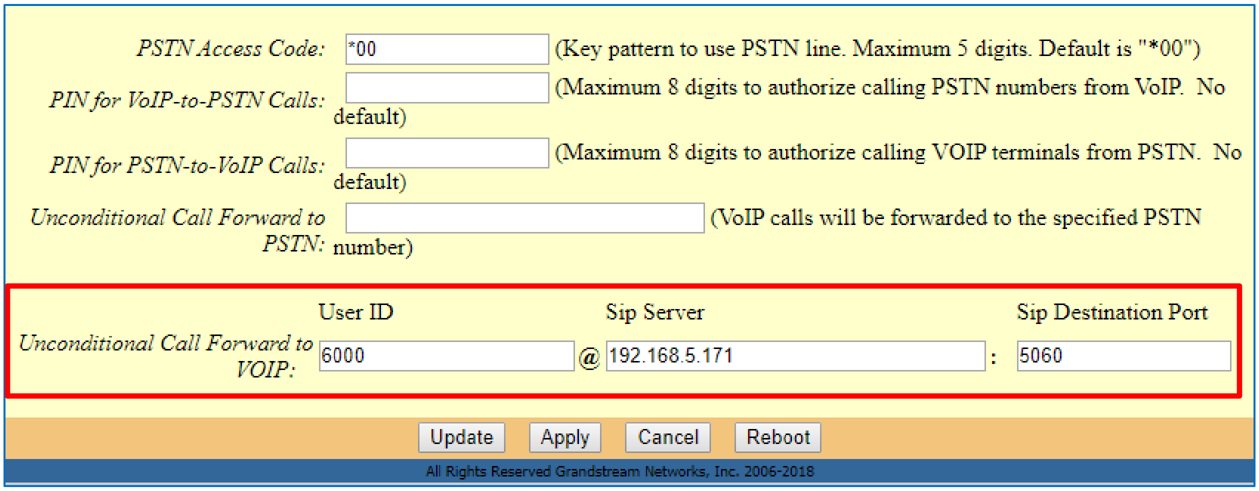

Figure 7: Basic settings configuration

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.