Configuring the Grandstream HandyTone 813 (HT-813)

Rev 1.0, 4/12/19

Preface

This guide provides information and hints for configuring the Grandstream HandyTone 813 (HT813) to work with Ignition Voice Alarming. The HT813 is a basic telephony appliance with both an FXS port (for a telephone) and an FXO port (for the phone line). While it is not a complete VOIP server, it does support SIP, and can be successfully used to route calls from Ignition to PSTN, through the FXO port.

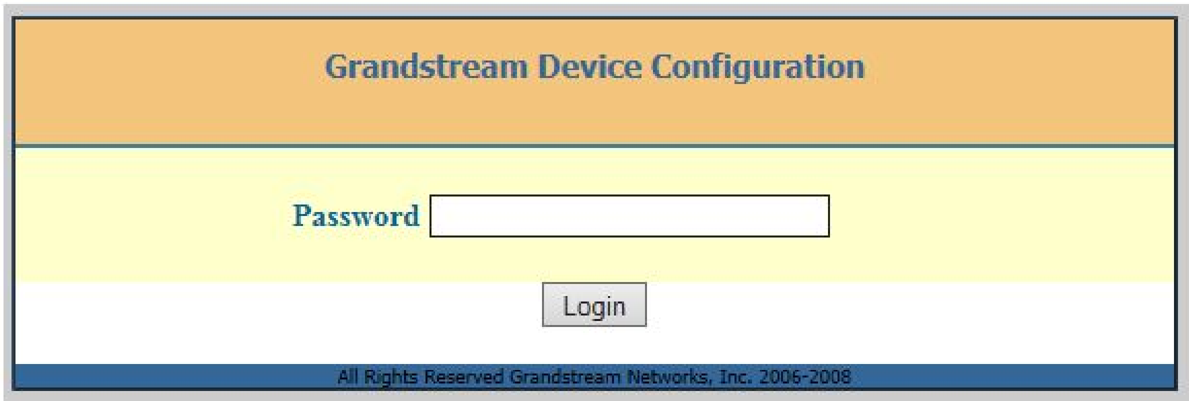
This guide assumes that you have access to the HT813 user manual, available from Grandstream's product website:
http://www.grandstream.com/sites/default/files/Resources/HT813_User_Guide.pdf

Setup

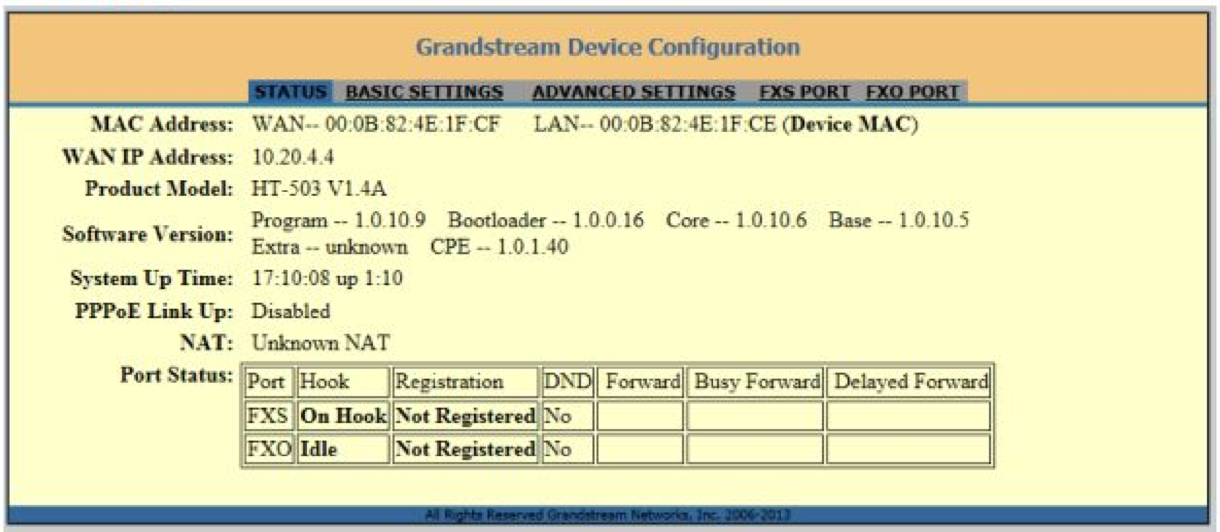
Note: After modifying each page, click "Apply" at the bottom to save the changes. Changes are not automatically saved when moving from page to page.

1) Connect to the web interface
Follow the instructions in the user manual section titled "ACCESS THE WEB CONFIGURATION MENU".

When connected to the "LAN" port, the default ip address is: **192.168.2.1**
The default password is: **admin**

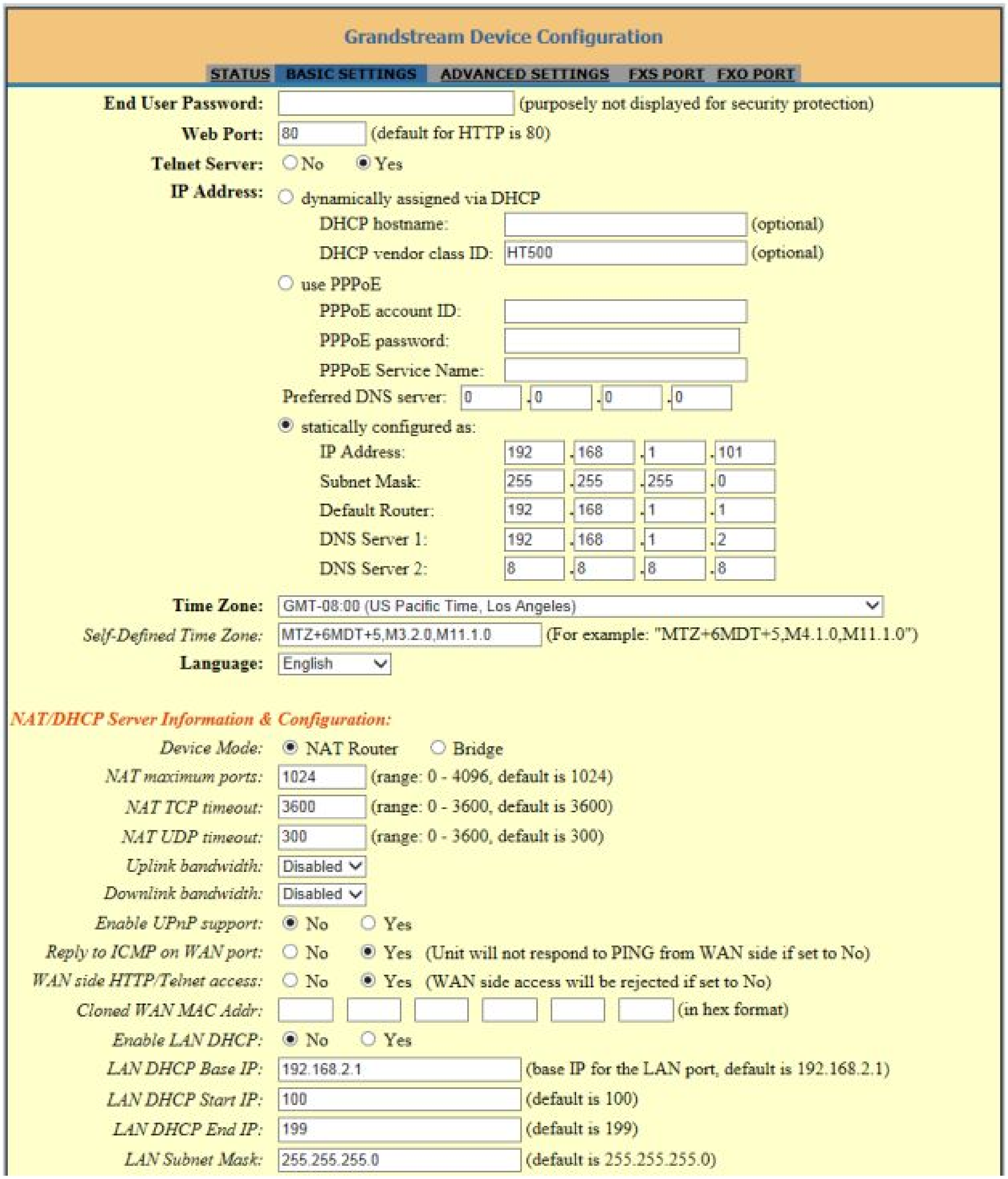


The initial login page



The status page, after logging in

2) Configure basic network settings
You will likely need to change the IP address of the box to fit with your network scheme. Although Ignition will likely connect over the local network, this guide assumes you will connect the network to the WAN port.



The following settings generally need to be modified:

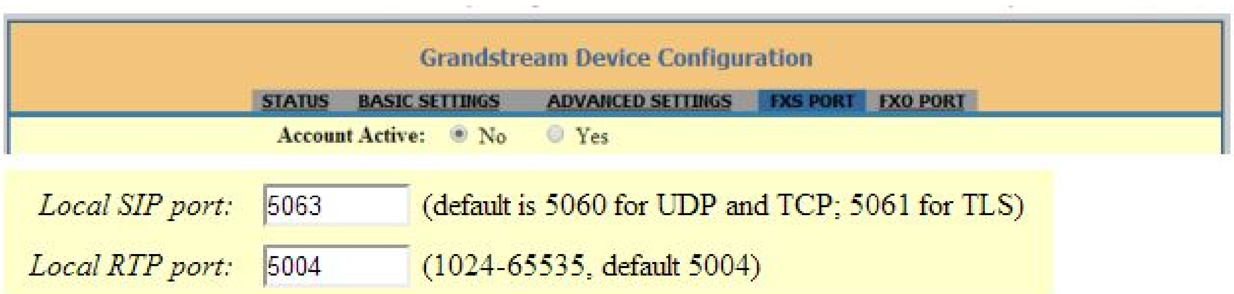
1. IP Address - **Statically Configured**
    a. Set relevant fields as necessary for your network.
2. Wan side HTTP Access - **Yes**, unless you intend to perform all configuration while connected to the LAN port.

3. Enable LAN DHCP - **No**, although the LAN port likely won't be used, it is likely not desirable to have the box provide DHCP services if connected to the network.

There are no settings on the Advanced Settings panel that must be changed.

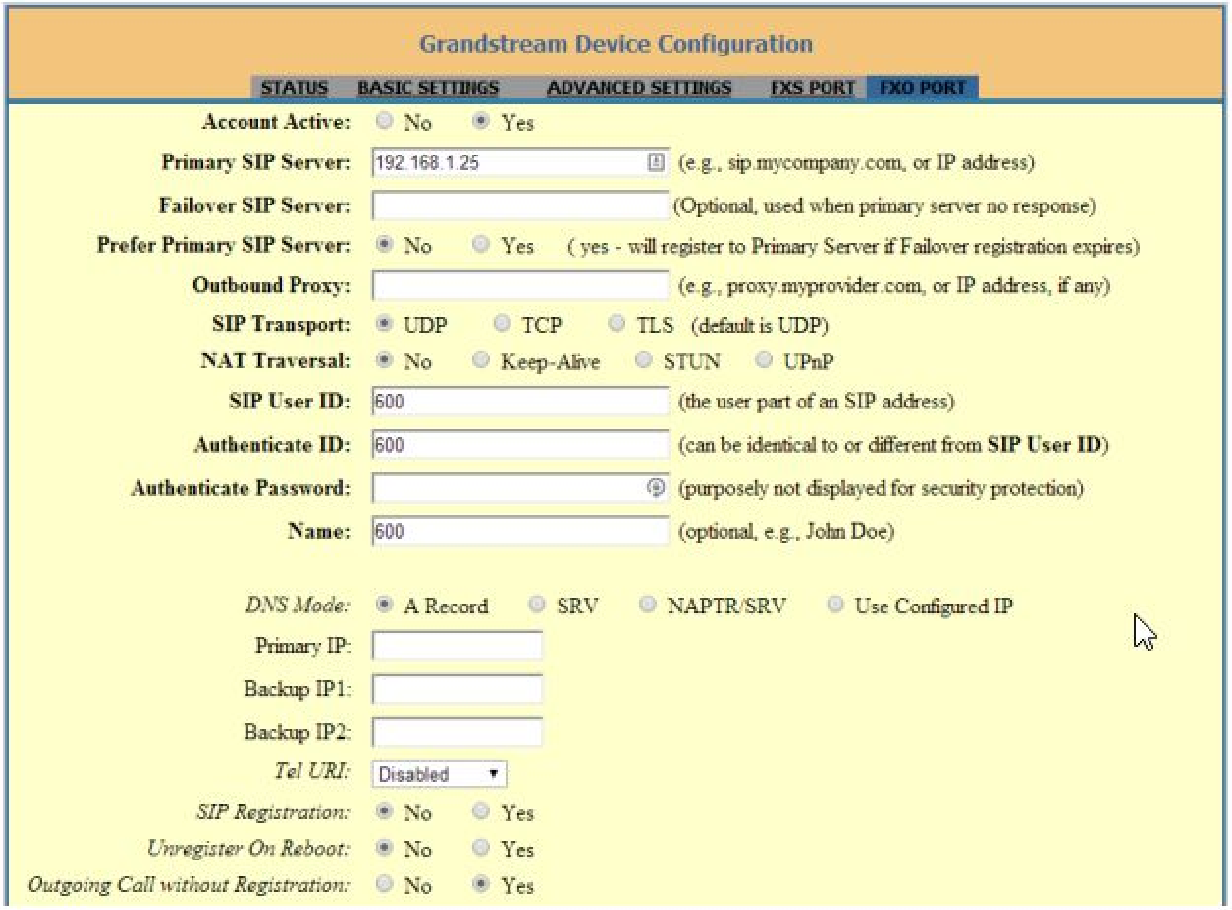
3) Configure the FXS Port settings
Although the FXS port will not be used, there are several settings that must be changed.

1. Account Active - **No**
2. Local SIP Port **- 5063** (Or anything besides 5060, which will be used by the FXO port)



4) Configure the FXO Port settings
This is where the bulk of the configuration takes place. If these settings are not correct, the default action is to forward calls to the FXS port on the box (in other words, when Ignition attempts to make a call, you will see the FXS port status go to "Ringing" on the HT813 status page).

The HT813 expects to be connected to a SIP server. While Ignition is not a sip server, the box can be configured to point to it, as it will not actually try to connect with the "SIP Registration" option set to "No" (see below).



Grandstream Device Configuration

STATUS | BASIC SETTINGS | ADVANCED SETTINGS | FXS PORT | FXO PORT

Account Active: ○ No ● Yes

Primary SIP Server: 192.168.1.25 (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)

SIP Transport: ● UDP ○ TCP ○ TLS (default is UDP)

NAT Traversal: ● No ○ Keep-Alive ○ STUN ○ UPnP

SIP User ID: 600 (the user part of an SIP address)

Authenticate ID: 600 (can be identical to or different from **SIP User ID**)

Authenticate Password: (purposely not displayed for security protection)

Name: 600 (optional, e.g., John Doe)

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

Primary IP:

Backup IP1:

Backup IP2:

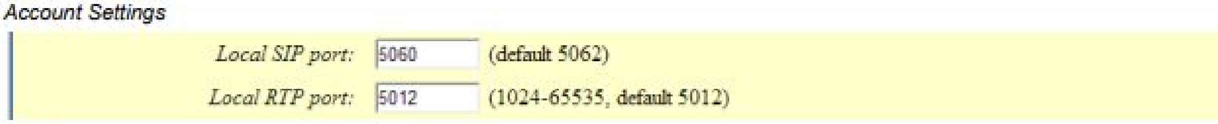
Tel URI: Disabled

SIP Registration: ● No ○ Yes

Unregister On Reboot: ● No ○ Yes

Outgoing Call without Registration: ○ No ● Yes

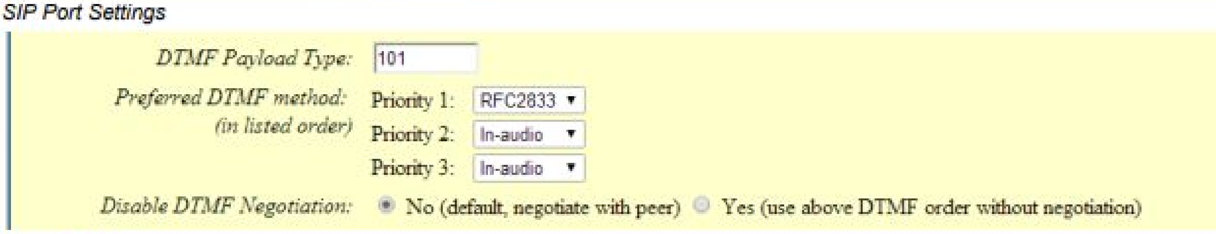
Account Settings



Local SIP port: 5060 (default 5062)

Local RTP port: 5012 (1024-65535, default 5012)

SIP Port Settings

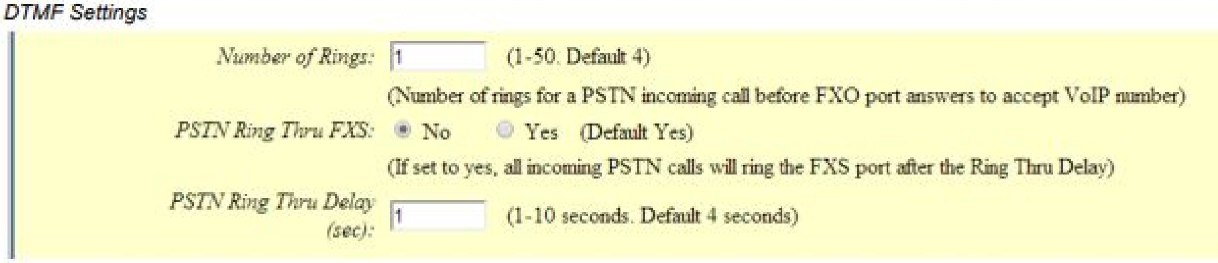


DTMF Payload Type: 101

Preferred DTMF method: (in listed order) Priority 1: RFC2833 Priority 2: In-audio Priority 3: In-audio

Disable DTMF Negotiation: ● No (default, negotiate with peer) ○ Yes (use above DTMF order without negotiation)

DTMF Settings



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): 1 (1-10 seconds. Default 4 seconds)

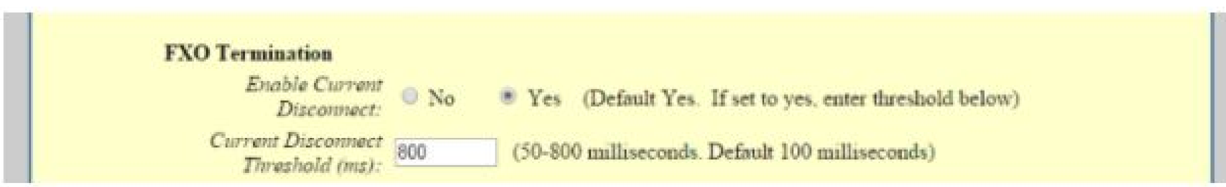
FXO Termination Setting

Channel Dialing Settings

1. Account Active - **Yes**
2. Primary SIP Server - **The IP address of your Ignition gateway** (Note: since the box will not actually register with Ignition, in a redundant setup it should be sufficient to simply enter the master address)
3. SIP User ID, Authenticate ID, Name - **Any simple name/id**
4. SIP Registration - **No**
5. Outgoing Call without Registration - **Yes**
6. Local SIP port - **5060**
7. DTMF Payload Type **- 101**
8. Preferred DTMF method, Priority 1 - **RFC2833**
9. Number of Rings (FXO Termination) - **1**
10. PSTN Ring Thru FXS - **No**
11. PSTN Ring Thru Delay - **1**
12. Wait for DialTone (Channel Dialing) **- No**
13. Stage Method - **1**

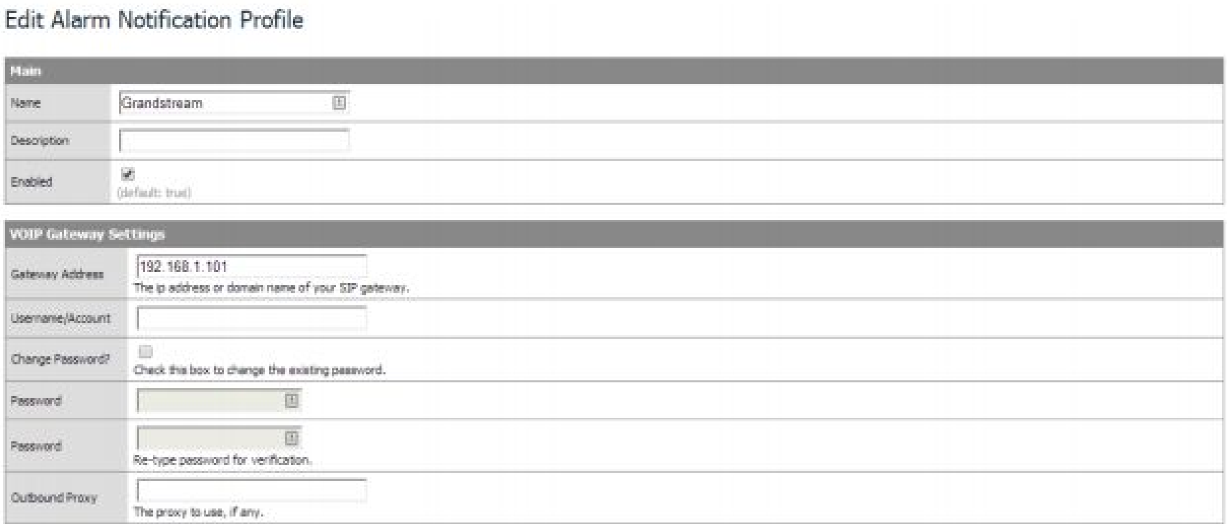
SIP Register Contact Header uses - Default is LAN but it needs to correspond to the interface Ignition is plugged into (LAN/WAN).

Note: It has been observed that in some environments, the Current Disconnect Threshold (ms) setting must be set higher in order for calls to be made successfully. If everything appears to be fine but calls are not placed, try increasing the value to 800:



5) Configure Ignition

1. Create a new voice notification profile by going to Gateway Configuration>Alarming>Notification>Create new profile, and selecting “VOIP”
2. Enter the IP address assigned to the box
3. *Leave the username and password blank*



Testing and Troubleshooting

To place a test call, you must first configure all of the parts of Ignition normally required for alarm notification. This includes:

1. Create a user in your User Source
2. Assign the user a phone number (**NB**: The phone number must be valid for the phone line, as it will be dialed directly. It will usually include the long distance country code, and should not include dashes. For example: 15551235432)
3. Create an OnCall Roster, assign the user
4. Create an alarm pipeline, with a notification block set to use the VOIP profile, and the correct OnCall Roster.
5. Create an alarm set to use the alarm pipeline as the “active pipeline”.

When a call is placed, you will be able to monitor its progress from Status>Voice Alarming in the gateway.

In the HT813, you should see the status of the FXO port change:

Port Status:

Port	Hook	Registration	DND	Forward	Busy Forward	Delayed Forward
FXS	**On Hook**	**Not Registered**	No			
FXO	**Idle**	**Not Registered**	No			

Normal, non-call status

Port Status:

Port	Hook	Registration	DND	Forward	Busy Forward	Delayed Forward
FXS	**On Hook**	**Not Registered**	No			
FXO	**In Use**	**Not Registered**	No			

Status during call

Troubleshooting Notes

- If the status of the FXS port becomes "Ringing", the call is not being routed correctly. Verify that all of the settings described in this document are set, and have been correctly Saved.
- In Ignition, you may need to increase the "Answer Timeout" in order to allow more time for the call to be initialized and established.
- In Ignition, all SIP messages are logged under the logger "Alarming.Notification.Voice.CallManager.Agent", at the Debug level. By enabling this logger in Configuration>System>Console>Levels, you will see the messages sent and received in the console. Primarily, note the messages that begin with "[network, SENT..." or "[network, RECEIVED...". If you do not see any "RECEIVED" messages, it is likely that the HT813 is not able to communicate with Ignition due to network or firewall settings. Also verify that port 5060 is not already in use on the Ignition system.
- If there is an issue with dial tones being ignored during the call for acknowledgement, first check on the Grandstream device for the setting "Symmetric RTP" and enable it. Then ensure that all the RTP ports between devices are the same.