

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**

Grandstream Device Configuration

Password

All Rights Reserved Grandstream Networks, Inc. 2006-2008

The initial login page

Grandstream Device Configuration

STATUS
BASIC SETTINGS
ADVANCED SETTINGS
FXS PORT
FXO PORT

MAC Address: WAN-- 00:0B:82:4E:1F:CF LAN-- 00:0B:82:4E:1F:CE (**Device MAC**)
WAN IP Address: 10.20.4.4
Product Model: HT-503 V1.4A
Software Version: Program -- 1.0.10.9 Bootloader -- 1.0.0.16 Core -- 1.0.10.6 Base -- 1.0.10.5
 Extra -- unknown CPE -- 1.0.1.40
System Up Time: 17:10:08 up 1:10
PPPoE Link Up: Disabled
 NAT: Unknown NAT
Port Status:

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

All Rights Reserved Grandstream Networks, Inc. 2006-2011

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.

Grandstream Device Configuration									
STATUS		BASIC SETTINGS		ADVANCED SETTINGS		FXS PORT		FXO PORT	
End User Password:		(purposely not displayed for security protection)							
Web Port:		80 (default for HTTP is 80)							
Telnet Server:		☐ No ☒ Yes							
IP Address:		☐ dynamically assigned via DHCP DHCP hostname: (optional) DHCP vendor class ID: HT500 (optional)							
		☐ use PPPoE PPPoE account ID: PPPoE password: PPPoE Service Name: Preferred DNS server: 0 0 0 0							
		☒ statically configured as: IP Address: 192 168 1 101 Subnet Mask: 255 255 255 0 Default Router: 192 168 1 1 DNS Server 1: 192 168 1 2 DNS Server 2: 8 8 8 8							
Time Zone:		GMT-08:00 (US Pacific Time, Los Angeles) ▼							
Self-Defined Time Zone:		MTZ+6MDT+5,M3.2.0,M11.1.0 (For example: "MTZ+6MDT+5,M4.1.0,M11.1.0")							
Language:		English ▼							
NAT/DHCP Server Information & Configuration:									
Device Mode:		☒ NAT Router ☐ Bridge							
NAT maximum ports:		1024 (range: 0 - 4096, default is 1024)							
NAT TCP timeout:		3600 (range: 0 - 3600, default is 3600)							
NAT UDP timeout:		300 (range: 0 - 3600, default is 300)							
Uplink bandwidth:		Disabled ▼							
Downlink bandwidth:		Disabled ▼							
Enable UPnP support:		☒ No ☐ Yes							
Reply to ICMP on WAN port:		☐ No ☒ Yes (Unit will not respond to PING from WAN side if set to No)							
WAN side HTTP/Telnet access:		☐ No ☒ Yes (WAN side access will be rejected if set to No)							
Cloned WAN MAC Addr:		(in hex format)							
Enable LAN DHCP:		☒ No ☐ Yes							
LAN DHCP Base IP:		192.168.2.1 (base IP for the LAN port, default is 192.168.2.1)							
LAN DHCP Start IP:		100 (default is 100)							
LAN DHCP End IP:		199 (default is 199)							
LAN Subnet Mask:		255.255.255.0 (default is 255.255.255.0)							

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.

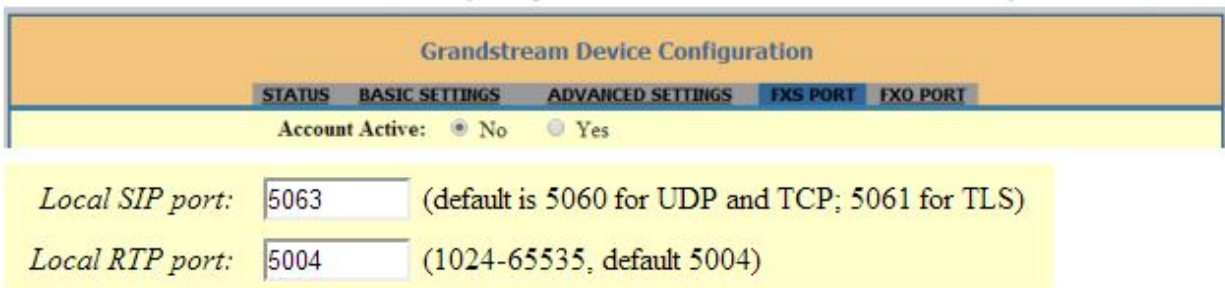
3. Enable LAN DHCP - **No**, although the LAN port likely won't be used, it is likely not desirable to have the box provide DHCP services if connected to the network.

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)



The screenshot shows the 'Grandstream Device Configuration' web interface. At the top, there is a navigation bar with tabs: STATUS, BASIC SETTINGS, ADVANCED SETTINGS, FXS PORT, and FXO PORT. The 'FXS PORT' tab is currently selected. Below the tabs, the 'Account Active' setting is shown with radio buttons for 'No' (selected) and 'Yes'. Below this, the 'Local SIP port' is set to '5063' with a note '(default is 5060 for UDP and TCP; 5061 for TLS)'. The 'Local RTP port' is set to '5004' with a note '(1024-65535, default 5004)'. The entire configuration area has a yellow background.

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
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:	Priority 1: RFC2833
(in listed order)	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

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

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

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:

FXO Termination

Enable Current Disconnect: ☐ No ☒ Yes (Default Yes. If set to yes, enter threshold below)

Current Disconnect Threshold (ms): (50-800 milliseconds. Default 100 milliseconds)

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*

Edit Alarm Notification Profile

Main	
Name	Grandstream
Description	
Enabled	☒ (default: true)

VOIP Gateway Settings	
Gateway Address	192.168.1.101 The ip address or domain name of your SIP gateway.
Username/Account	
Change Password?	☐ Check this box to change the existing password.
Password	
Password	 Re-type password for verification.
Outbound Proxy	 The proxy to use, if any.

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.