Grandstream Networks, Inc.

HT813

FXS/FXO Port

Analog Telephone Adaptor

Administration Guide

COPYRIGHT

CAUTION

Changes or modifications to this product not expressly approved by Grandstream, or operation of this product in any way other than as detailed by this Administration guide, could void your manufacturer warranty.

WARNING

Please do not use a different power adaptor with your devices as it may cause damage to the products and void the manufacturer warranty.

HT813 Administration Guide
Version 1.0.5.2

GNU GPL INFORMATION

The firmware for the HT813 contains third-party software licensed under the GNU General Public License (GPL). Grandstream uses software under the specific terms of the GPL. Please see the GNU General Public License (GPL) for the exact terms and conditions of the license.

Grandstream GNU GPL related source code can be downloaded from Grandstream web site from:
http://www.grandstream.com/support/faq/gnu-general-public-license/gnu-gpl-information-download

HT813 Administration Guide
Version 1.0.5.2

Table of Content

HT813 Administration Guide
Version 1.0.5.2

HT813 Administration Guide
Version 1.0.5.2

Table of Tables

Table of Figures

HT813 Administration Guide
Version 1.0.5.2

DOCUMENT PURPOSE

This document describes the basic concept and tasks necessary to use and configure your HT813. It covers also the topic of connecting and configuring the HT813, making basic operations and the call features. Please visit http://www.grandstream.com/support to download the latest "HT813 User Guide".

This guide covers following topics:

- Product overview

- Getting started

- Configuration guide

- Upgrade and provisioning

- Restore factory default settings

HT813 Administration Guide
Version 1.0.5.2

CHANGE LOG

This section documents significant changes from previous versions of admin guide for HT813. Only major new features or major document updates are listed here. Minor updates for corrections or editing are not documented here.

Firmware Version 1.0.5.2

- Added support for unlimited ring without timeout. [Ring Timeout]
- Added feature "Connection Request Port". [Connection Request Port]
- Added feature "SIP User-Agent" for FXS port settings. [SIP User-Agent]
- Added feature "SIP User-Agent" for FXO port settings. [SIP User-Agent]

Firmware Version 1.0.3.12

- Added support for T.38 Fax mode under FXO Port. [Fax Mode]
- Updated "São Paulo" time zone to UTC-3. [Time Zone]
- Added feature "Allow SIP Factory Reset" for FXS port Settings. [Allow SIP Factory Reset]
- Added feature "Allow SIP Factory Reset" for FXO port Settings. [Allow SIP Factory Reset]
- Added feature "Pulse Dialing Standard". [Pulse Dialing Standard]
- Added support for RFC2833 event Count. [RFC2833 Events Count]
- Added support for RFC2833 end event Count. [RFC2833 End Events Count]
- Added feature "Disable Weak TLS Cipher Suites". [Disable Weak TLS Cipher Suites]
- Added feature "Minimum TLS Version". [Minimum TLS Version]
- Added feature "Maximum TLS Version". [Maximum TLS Version]
- Added feature "Syslog Protocol". [Syslog Protocol]

Firmware Version 1.0.1.2

- No major changes.

Firmware Version 1.0.0.8

- This is the initial version for HT813.

HT813 Administration Guide
Version 1.0.5.2

GUI INTERFACE EXAMPLES

http://www.grandstream.com/sites/default/files/Resources/HT813_web_gui.zip

1. Screenshot of Login Page

2. Screenshots of Status Page

3. Screenshots of Basic Settings Page

4. Screenshots of Advanced Settings Page

5. Screenshots of FXS Port Page

6. Screenshots of FXO Port Page

HT813 Administration Guide
Version 1.0.5.2

WELCOME

Thank you for purchasing Grandstream's HT813, It is the first ATA in the HandyTone 81x series offering functions as a true 3-in-1 gateway for PSTN network, analog telephone FXS interface and IP network. It enables remote call origination and termination from/to PSTN. The HT813 is specifically designed to be an easy to use and affordable VoIP solution for both the residential user and the remote user.

This administrator guide will help you learn how to operate and manage your HT813 Analog Telephone Adaptor and make the best use of its many upgraded features including simple and quick installation.

PRODUCT OVERVIEW

The HT813 is an analog telephone adapter (ATA) featuring 1 analog telephone FXS port and 1 PSTN line FXO port. The integration of FXO and FXS ports enables remote call origination and termination to and from the PSTN line. The 1 FXS port allows for extension of a VoIP service to 1 analog phone. HT813's ultra-compact size, voice quality, advanced VoIP functionality, security protection and auto provisioning options enable users to take advantage of VoIP on analog phones and enables service providers to offer high quality IP service.

Feature Highlights

The following table contains the major features of the HT813:

Table 1: HT813 Features at a Glance

HT813	
	• 2 SIP profiles through 1 FXS port and 1 FXO port
	• Dual 100 Mbps LAN and WAN ports
	• 3-way voice conferencing
	• Wide range of caller ID formats
	• Advanced telephony features, including call transfer, call forward, call-waiting, do not disturb, message waiting indication, multi-language prompts, flexible dial plan and more
	• T.38 Fax for creating Fax-over-IP
	• TLS and SRTP security encryption technology to protect calls and accounts
	• Automated provisioning options include TR-069 and XML config files
	• Strong AES encryption with security certificate per unit
	• Failover SIP server automatically switches to secondary server if main server loses connection
	• Use with Grandstream's UCM series of IP PBXs for Zero Configuration provisioning
	• Lifeline support (FXS port will be hard-relayed to FXO port) in case of power outage.

HT813 Technical Specifications

The following table resumes all the technical specifications including the protocols/standards supported, voice codecs, telephony features, languages and upgrade/provisioning settings for the HT813.

Table 2: HT813 Technical Specifications

Interfaces	
Telephone Interfaces	One (1) RJ11 FXS port, One (1) RJ11 FXO PSTN line port with lifeline support
Network Interface	Two (2) 10/100 Mbps ports (RJ45) with integrated NAT router
LED Indicators	POWER, FXO, FXS, WAN, LAN.
Factory Reset Button	Yes
Voice, Fax, Modem	
Telephony Features	Caller ID display or block, call waiting, flash, blind or attended transfer, forward, hold, do not disturb, 3-way conference
Voice Codecs	G.711 with Annex I (PLC) and Annex II (VAD/CNG), G.723.1, G.729A/B, G.726, iLBC, OPUS, dynamic jitter buffer, advanced line echo cancellation
Fax over IP	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through
Short/Long Haul Ring Load	3 REN: Up to 1km on 24 AWG
Caller ID	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID
Disconnect Methods	Busy Tone, Polarity Reversal/Wink, Loop Current
Signaling	
Network Protocols	TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, FTP/FTPS, ARP/RARP, ICMP, DNS, DDNS, DHCP, NTP, TFTP, SSH, Telnet, STUN, SIP (RFC3261), SIP over TCP/TLS, SRTP, TR-069
QoS	Layer 2 (802.1Q VLAN, SIP/RTP 802.1p) and Layer 3 (ToS, Diffserv, MPLS).
DTMF Methods	In-audio, RFC2833 and/or SIP INFO
Provisioning and Control	HTTP, HTTPS, SSH, FTP, FTPS, Telnet, SSH, TFTP, TR-069, secure and automated provisioning using AES encryption, syslog
Security	
Media	SRTP
Control	TLS/SIPS/HTTPS
Management	Syslog support, SSH, Telnet remote management using web browser.

Physical	
Universal Power Supply	Input: 100-240VAC, 50/60Hz
	Output: 12V/0.5A
Environmental	Operational: 32° – 104°F or 0º – 40ºC
	Storage: 14° – 140°F or -10º – 60ºC
	Humidity: 10 – 90% Non-condensing
Dimensions and Weight	130.5 x 90.5 x 29 mm (L x W x D)
	Weight: 0.142Kg
Compliance	
Compliance	FCC/CE/C-TICK/ITU-K.21

HT813 Administration Guide
Version 1.0.5.2

GETTING STARTED

This chapter provides basic installation instructions including the list of the packaging contents and also information for obtaining the best performance with the HT813.

Equipment Packaging

The HT813 ATA package contains:

1x HT813

1x 12V Power Adapter

1x Ethernet Cable

1x Quick Installation Guide
1x GPL Statement
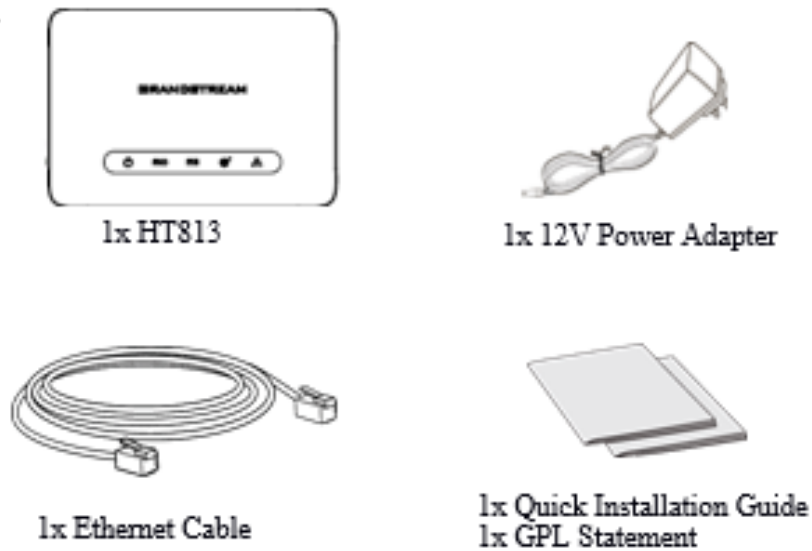
Figure 1: HT813 Package Contents

Note: Check the package before installation. If you find anything missing, contact your system administrator

HT813 Ports Description

The following figure describes the different ports on the back panel of the HT813.
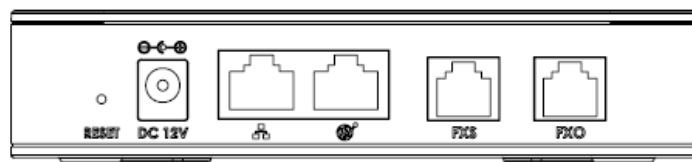
Figure 2: HT813 Back Panel

Table 3: HT813 Connectors Definitions

FXS	Connects the analog phone / fax machine to the ATA using an RJ-11 telephone cable.
FXO	FXO telephone port (PSTN Port) 1x PSTN pass-through and life line port.
	Connects the ATA to your router, switch or modem using an Ethernet RJ45 network cable.
	Connects the ATA to your PC or switch using an Ethernet RJ45 network cable.
DC 12V	Connects the ATA to PSU (Output: 12V/0.5A)
Reset	Factory reset button. Press for 7 seconds to reset factory default. settings. Quick press will only reboot the unit.

Connecting HT813

The HT813 is designed for easy configuration and easy installation. To connect your HT813, please follow the steps below:

Scenario 1: Connecting the HT813 using WAN Port

When connecting HT813 using the WAN port, it will act as simple DHCP Client.

1. Insert a standard RJ11 telephone cable into the FXS port and connect the other end of the telephone cable to a standard touch-tone analog telephone.

2. Connect the WAN port of the HT813 to a router, switch or modem using an Ethernet cable.

3. Insert the power adapter into the HT813 and connect it to a wall outlet and make sure to respect the technical specifications of the power adapter used.

4. Power, WAN and FXS LED will be solidly lit when the HT813 is ready for use.

Scenario 2: Connecting the HT813 using LAN Port

When connecting the HT813 using the LAN port, it will act as a router and DHCP serving addresses, the devices connected with HT813 LAN will pull DHCP addresses from your HT813.

1. Insert a standard RJ11 telephone cable into FXS port and connect the other end of the telephone cable to a standard touch-tone analog telephone.

2. Connect a computer or switch to the LAN port of the HT813 using an Ethernet Cable.

3. Insert the power adapter into the HT813 and connect it to a wall outlet and make sure to respect the technical specifications of the power adapter used.

4. Power, LAN and FXS LED will be solidly lit when the HT813 is ready for use.

Note: Please make sure to enable NAT Router under Web GUI → Basic Settings → Device Mode.
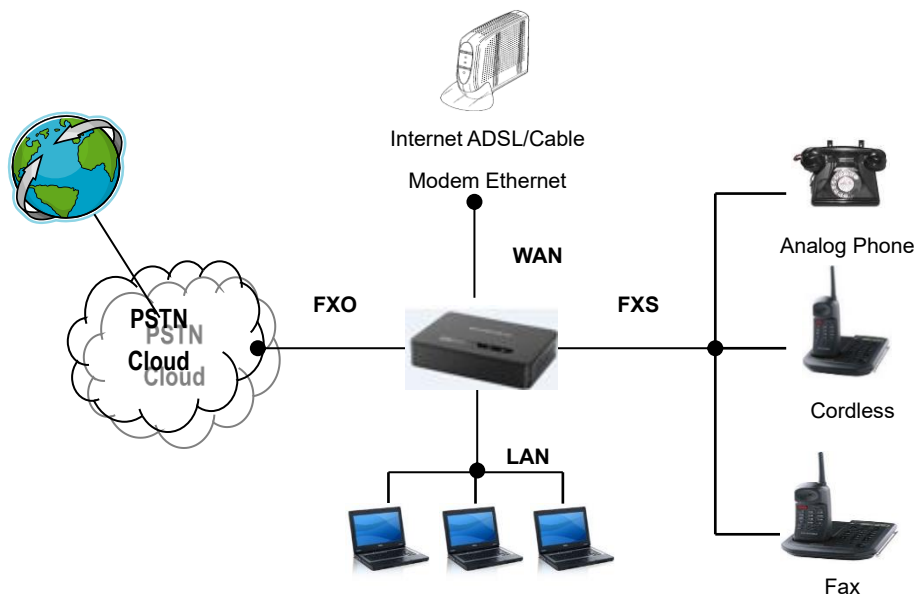
Figure 3: Connecting the HT813

HT813 LEDs Pattern

There are four (4) LED types that help you manage the status of your HT813.

Figure 4: HT813 LEDs Pattern

Table 4: HT813 LEDs Pattern Description

LED Lights	Status
Power LED	The Power LED lights up when the HT813 is powered on and it flashes when the HT813 is booting up.
WAN LED	The WAN LED lights up when the HT813 is connected to your network through the WAN port.
LAN LED	The LAN LED lights up when the HT813 is connected to your network through the LAN port.
FXS/FXO LED	The FXS LEDs indicate status of the respective FXS/FXO port-phone on the back panel • **OFF** - Unregistered • **ON (Solid Blue)** - Registered and Available • **Blinking every 500 ms** - Off-Hook / Busy • **Slow blinking** - FXS LEDs indicates voicemail

HT813 Administration Guide
Version 1.0.5.2

CONFIGURATION GUIDE

The HT813 can be configured via one of two ways:

- The IVR voice prompt menu.
- The Web GUI embedded on the HT813 using PC's web browser.

Obtain HT813 IP Address via Connected Analogue Phone

HT813 is by default configured to obtain the IP address from DHCP server where the unit is located. To know which IP address is assigned to your HT813, you should access to the "Interactive Voice Response Menu" of your adapter via the connected phone and check its IP address mode.

Please refer to the steps below to access the interactive voice response menu:

1. Use a telephone connected to FXS port of your HT813.
2. Press *** (press the star key three times) to access the IVR menu and wait until you hear "Enter the menu option ".
3. Press 02 and the current IP address will be announced.

Understanding HT813 Interactive Voice Prompt Response Menu

The HT813 has a built-in voice prompt menu for simple device configuration which lists actions, commands, menu choices, and descriptions. Connect analog phone to FXS port. Pick up the handset and dial "***" to use the IVR menu.

Table 5: Voice Prompt Menu

Menu	Voice Prompt	Options
Main Menu	"Enter a Menu Option"	Press "*" for the next menu option Press "#" to return to the main menu Enter 01-05, 07,10, 12-17,47 or 99 menu options
01	"DHCP Mode", "Static IP Mode" "PPPoE Mode"	Press "9" to toggle the selection If using "Static IP Mode", configure the IP address information using menus 02 to 05. If using "Dynamic IP Mode", all IP address information comes from the DHCP server automatically after reboot. If using "PPPoE Mode", configure PPPoE Username and Password from web GUI to get IP from your ISP.

HT813 Administration Guide
Version 1.0.5.2

02	"IP Address " + IP address	The current WAN IP address is announced
		If using "Static IP Mode", enter 12-digit new IP address.
		You need to reboot your HT813 for the new IP address to take Effect.
03	"Subnet " + IP address	Same as menu 02
04	"Gateway " + IP address	Same as menu 02
05	"DNS Server " + IP address	Same as menu 02
07	Preferred Vocoder	Press "**9**" to move to the next selection in the list:
		• PCM U / PCM A
		• iLBC
		• G-726
		• G-723
		• G-729
		• OPUS
10	"MAC Address"	Announces the MAC address of the unit.
		Note: The device has two MAC addresses. One for the WAN port and one for the LAN port. The device MAC address announced is the address of LAN port.
12	WAN Port Web Access	Press "**9**" to toggle between **Auto** / **Enabled** / **Disabled**. Default is Auto.
13	Firmware Server IP Address	Announces current Firmware Server IP address. Enter 12-digit new IP address.
14	Configuration Server IP Address	Announces current Config Server Path IP address. Enter 12-digit new IP address.
15	Upgrade Protocol	Upgrade protocol for firmware and configuration update. Press "**9**" to toggle between **TFTP / HTTP / FTP / FTPS** or **HTTPS**. Default is HTTPS.
16	Firmware Version	Announces Firmware version information.
17	Firmware Upgrade	Firmware upgrade mode. Press "**9**" to toggle among the following three options:
		• Always check
		• Check when pre/suffix changes
		• Never upgrade

47	"Direct IP Calling"	Enter the target IP address to make a direct IP call, after dial tone. (See "*Make a Direct IP Call*".)
86	Voice Mail	Access to your voice mails messages.
99	"RESET"	Press "**9**" to reboot the device Enter MAC address to restore factory default setting (See **RESTORE FACTORY DEFAULT SETTINGS** section)
	"Invalid Entry"	Automatically returns to main menu
	"Device not registered"	This prompt will be played immediately after off hook If the device is not registered and the option "Outgoing Call without Registration" is in NO

Five success tips when using the voice prompt

- "*" shifts down to the next menu option and "**#**" returns to the main menu

- "9" functions as the ENTER key in many cases to confirm or toggle an option.

- All entered digit sequences have known lengths - 2 digits for menu option and 12 digits for IP address. For IP address, add 0 before the digits if the digits are less than 3 (i.e. - 192.168.0.26 should be key in like 192168000026. No decimal is needed).

- Key entry cannot be deleted but the phone may prompt error once it is detected.

Note: Please make sure to reboot the device after changing network settings (IP Address, Gateway, Subnet…) to apply the new configuration.

Configuration via Web Browser

The HT813 embedded Web server responds to HTTP/HTTPS GET/POST requests. Embedded HTML pages allow a user to configure the HT813 through a web browser such as Google Chrome, Mozilla Firefox and Microsoft's IE.

- **Microsoft Internet Explorer:** version 10 or higher.
- **Google Chrome**: version 58.0.3 or higher.
- **Mozilla Firefox**: version 53.0.2 or higher.
- **Safari**: version 5.1.4 or higher.
- **Opera**: version 44.0.2 or higher.

Accessing the Web UI

- **Via WAN port**

1. You may check your HT813 IP address using the IVR on the connected phone.
   Please see Obtain the HT813 IP address via the connected analogue phone

2. Open the web browser on your computer.

3. Enter the HT813's IP address in the address bar of the browser.

4. Enter the administrator's password to access the Web Configuration Menu.

Note: The computer must be connected to the same sub-network as the HT813. This can be easily done by connecting the computer to the same hub or switch as the HT813.

- **Via LAN port**

1. Power your HT813 using PSU with the right specifications.

2. Connect your computer or switch directly to your HT813 LAN port.

3. Open the web browser on your computer.

4. Enter the default LAN IP address (192.168.2.1) in the address bar of the browser.

5. Enter the administrator's password to access the Web Configuration Menu.

6. Make sure to reboot your device after changing your settings to apply the new configuration.

Note: Please make sure that your computer has a valid IP address on the range 192.168.2.x so you can access the web GUI of your HT813.

Web UI Access Level Management

There are three default passwords for the login page:

User Level	Password	Web Pages Allowed
End User Level	123	Only Status and Basic Settings
Administrator Level	admin	All pages
Viewer Level	viewer	Only checking. Not allowed to modify content.

The password is case sensitive with maximum length of 25 characters. When changing any settings, always submit them by pressing **Update** or **Apply** button on the bottom of the page. After submitting the changes in all the Web GUI pages, if a reboot is required, the web page will prompt the user to reboot by offering a reboot button on the web page.
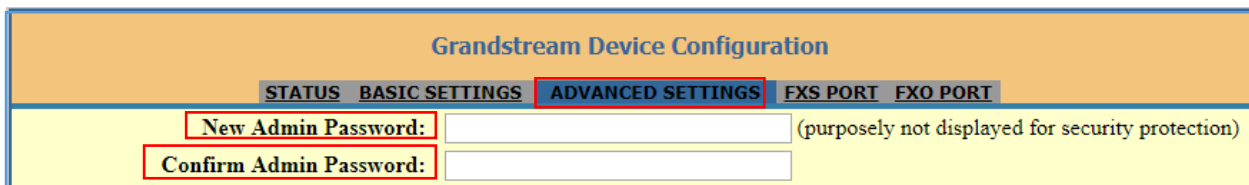
Saving the Configuration Changes

After users make changes to the configuration, pressing **Update** button will save but not apply the changes until **Apply** button is clicked. Users can instead directly press **Apply** button. When a reboot is required to apply changes, the web page will prompt the user to reboot by offering a reboot button on the web page.

Changing Admin Level Password

1. Access your HT813 web UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: admin).
3. Press **Login** to access your settings.
4. Go to **Advanced Settings → New Admin Password** and enter the new admin password.
5. Confirm the new admin password.
6. Press **Apply** at the bottom of the page to save your new settings.
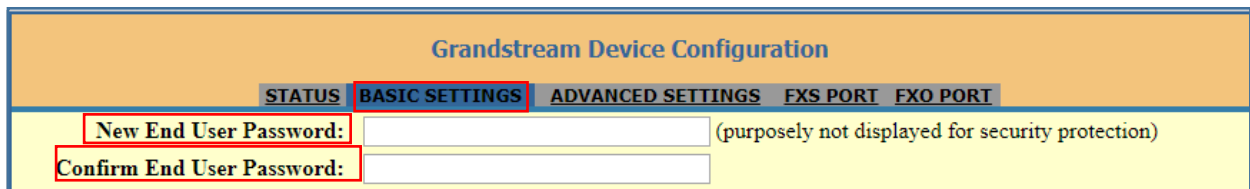
Figure 5: Admin Level Password

Changing User Level Password

1. Access your HT813 web UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: admin).
3. Press **Login** to access your settings.
4. Go to **Basic Settings → New End User Password** and enter the new end-user password.
5. Confirm the new end-user password.
6. Press **Apply** at the bottom of the page to save your new settings.

Figure 6: User Level Password

Changing Viewer Password

1. Access your HT813 web UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: admin).

HT813 Administration Guide
Version 1.0.5.2

3. Press **Login** to access your settings.

4. Go to **Basic Settings** → **New Viewer Password** and **e**nter the new viewer password.

5. Confirm the new viewer password.

6. Press **Apply** at the bottom of the page to save your new settings.

Figure 7: Viewer Level Password

Changing HTTP/HTTPS Web Port

1. Access your HT813 web UI by entering its IP address in your favorite browser.

2. Enter your admin password (default: admin).

3. Press **Login** to access your settings.

4. Go to **Basic Settings** → **HTTP(S) Web Port**.

5. Make sure that the **Web Access Mode** is set to **HTTP(S)**.

6. Change the current port to your new HTTP(S) port. Ports accepted are in range [1-65535].

7. Press **Apply** at the bottom of the page to save your new settings

Figure 8: Web HTTP(S) Port

Web Configuration Pages Definitions

This section describes the options in the HT813 Web UI.

- **STATUS:** Displays the system info, network status, account status, and line options.
- **BASIC SETTINGS:** Configures the end user level password, IP address modes, web access, time zone settings and language.
- **ADVANCED SETTINGS:** Configures networks, upgrading and provisioning, TR-069, security settings, date and time, syslog, audio settings, call settings and call progress tones.
- **FXS PORT:** Configures SIP accounts settings.
- **FXO PORT:** Configures SIP accounts settings.

Status Page Definitions

Table 6: Status Page Definitions

Status	
MAC Address	Shows device ID in hexadecimal format. This is needed by network administrators for troubleshooting. The MAC address will be used for provisioning and can be found on the label on original box and on the label located on the bottom panel of the device. **Note**: The device has two MAC addresses, one for the WAN port and one for the LAN port. The MAC address located on the bottom panel of the device is the MAC address of LAN port. The MAC address of WAN port is MAC address of LAN port +1. Example: MAC Address: WAN - "00:0B:82:25:AF:32", LAN - "00:0B:82:25:AF:31".
WAN IPv4 Address	Displays assigned IPv4 address.
WAN IPv6 Address	Displays assigned IPv6 address.
Product Model	Displays product model info. Default is **HT813.**
Hardware Version	Displays the hardware revision information and the part number.
Software Version	• **Program:** Specifies Program version. This is the main firmware release number, which is always used for identifying the software system of the HT813. • **Bootloader:** Specifies Boot version. • **Core:** Specifies Core version. • **Base:** Specifies Base version. • **CPE:** Specifies CPE version. CPE version is displayed only when HT813 is connected to an ACS using TR-069 protocol.

Software Status	Indicates the current software status of the HT (Running or Stopped).
System Up Time	Indicates actual system time and uptime since last reboot.
PPPoE Link Up	Indicates PPPoE connection status.
NAT	Indicates type of NAT when it is configured.
Port Status	Displays relevant information regarding the FXS and FXO ports about their registration, current status and their appropriate User ID.
Port Options	Displays relevant information regarding the FXS and FXO ports about their DND and call forward features.
Provision	Displays provisioning status.
Core Dump	Provides generated core dump file if unit malfunctions. **Clean** will be displayed if no issues.

Basic Settings Page Definitions

Table 7: Basic Settings Page

Basic Settings	
New End User Password	Configures user level password. Case sensitive and max. Length of 25 characters.
Confirm End User Password	Re-enter the end user password to confirm change user password on web GUI to avoid typo or mistakes.
New Viewer Password	Configures viewer level password. Case sensitive and max. Length of 25 characters
Confirm Viewer Password	Re-enter the viewer password to confirm change viewer password on web GUI to avoid typo or mistakes.
Web/SSH Access	
Web Session Timeout	Configure timer to logout web session during idle. Default is 10 min. Range is 2-60 min.
Web Access Attempt Limit	Configure attempt limit before lockout. Default is 5. Range is 1-10.

Lockout Time Interval	If login attempt failed 5 times, login would be locked out for the time length. Default 15 mins. Range 1-15 min.
Web Access Mode	Allows users to choose the Web Access Mode between "HTTPS" and "HTTP". If "HTTPS" is selected, web UI will be accessed using HTTPS. Default is "HTTP".
HTTP Web Port	Customizes HTTP port used to access the HT813 web UI. Default is **80**.
HTTPS Web Port	Customizes HTTPS port used to access the HT813 web UI. Default is **443**.
Disable SSH	Enables/disables the SSH access. Default is **No** (disabled).
SSH Port	Allows users to self-configure SSH Port number. By default, the port number is **22**.
Disable Telnet	Enables/disables the Telnet access. Default is **Yes** (disabled).
Telnet Port	Allows users to self-configure Telnet Port number. By default, the port number is **23**.
WAN Side Web/SSH Access	Enables/Disables the Web and SSH access through the WAN port. The available options are the following: • **No:** No access to the web or SSH from any IP address on the WAN side. • **Yes:** Access for the Web GUI and SSH is enabled on the WAN side. • **Auto:** Only private IP could access the web or SSH on the WAN side. Default setting is **Auto**.
White List for WAN Side	If "WAN Side Web/SSH Access" is set to "Yes" or "Auto". Users can configure the white List for WAN Side to be used for remote management. Multiple IPs are supported and need to be separated by "space". **Example:** 192.168.5.222 192.168.5.223 192.168.7.0/24 **Note:** If both blacklist and whitelist are not empty, the blacklist is processed first, followed by the whitelist.
Black List for WAN Side	If "WAN Side Web/SSH Access" is set to "Yes" or "Auto". Users can configure the black List for WAN Side to ban WAN side web access.

	Multiple IPs are supported and need to be separated by "space".
	Example: 192.168.5.222 192.168.5.223 192.168.7.0/24
	Note: If both blacklist and whitelist are not empty, the blacklist is processed first, followed by the whitelist.
Internet Protocol	Selects one of the following IP protocol modes: **IPv4 Only:** Enforce IPv4 protocol only. **IPv6 Only:** Enforce IPv6 protocol only**.** **Both, Prefer IPv4:** Enable both IPv4 and IPv6 and prefer IPv4. **Both, prefer IPv6:** Enable both IPv4 and IPv6 and prefer IPv6. **Note**: Make sure to reboot the ATA for the changes to take effect.
IPv4 Address	Allows users to configure the appropriate network settings on the HT813 to obtain IPv4 address. Users could select "DHCP", "Static IP" or "PPPoE". By default, it is set to "DHCP".
Dynamically assigned via DHCP	All the field values for the static IP mode are not used (even though they are still saved in the flash memory.) The ATA acquires its IP address from the first DHCP server it discovers from the LAN it is connected. **DHCP hostname:** Specifies the name of the client. The name may or may not be qualified with the local domain name. This field is optional but may be required by ISP. **DHCP vendor class ID:** Exchanges vendor class ID by clients and servers to convey particular configuration or other identification information about a client. Default is **HT8XX**.
Use PPPoE	Set the PPPoE account settings. If selected, ATA attempt to establish a PPPoE session if any of the PPPoE fields is set. **PPPoE account ID:** Defines the PPPoE username. Necessary if ISP requires you to use a PPPoE (Point to Point Protocol over Ethernet) connection. **PPPoE password:** Specifies the PPPoE account password. **PPPoE Service Name:** Defines PPPoE service name. If your ISP uses a service name for the PPPoE connection, enter the service name here. This field is optional. Default is blank.

Preferred DNS server	Specifies preferred DNS server to use when DHCP or PPPoE are set.
Statically configured as IP address	Configure IP address, subnet Mask, default router IP address, 1st preferred DNS server, 2nd preferred DNS server. These fields are set to zero by default.

IPv6 Address	Allows users to configure the appropriate network settings on the HT813 to obtain IPv6 address. Users could select "DHCP", "Static IP". By default, it is set to "DHCP".
	
	• **DHCP mode:** all the field values for the static IP mode are not used (even though they are still saved in the flash memory.) The ATA acquires its IP address from the first DHCP server it discovers from the LAN it is connected.
	
	• **Static IP mode:** configure IP address, 1st and 2nd DNS server, preferred DNS server. These fields are set to zero by default.
	
	   - *Full Static*: When enabling the option full static, users need to specify the Static IPv6 and the IPv6 Prefix length.
	
	   - *Prefix Static*: When enabling the option prefix static, users need to specify the IPv6 Prefix (64 bits).

Time Zone	Selects time zone to define date/time on the device.
Self-Defined Time Zone	Allows users to define their own time zone.
Allow DHCP server to set Time Zone	Obtains time zone setting (offset) from a DHCP server using DHCP Option 2; it will override selected time zone. If set to "No", the analogue adapter will use selected time zone even if provided by DHCP server. Default is **Yes**.
Language	Configures the languages of the voice prompt and web interface, except Spanish that it is only in IVR. Available languages: English, Chinese or Spanish IVR.
NAT/DHCP Server Information & Configuration	
Device Mode	Controls whether the device is working in **NAT router**, **Bridge** or **WAN Only** mode. • **NAT Router:** In this mode, WAN port acts as DHCP client. LAN port is used as DHCP Base IP; devices connected behind LAN port will be assigned an IP from HT813 DHCP Server.

	• **Bridge:** In this mode, WAN port acts as DHCP client and pass-through to LAN port; devices connected behind LAN port will get an IP from your network DHCP server (same as WAN port). • **WAN Only:** In this mode, only WAN port is active. LAN port is not used. Default mode is **NAT** Router. Save the setting and reboot prior to configuring the HT813.
NAT Maximum Ports	Defines the number of ports that can be managed while in NAT router mode. Range: 0 – 4096, default is 1024. Typically, one port per connection
NAT TCP Timeout	NAT TCP idle timeout in seconds. Connection will be closed after preconfigured, timeout if not refreshed. Range: 0 - 3600
NAT UDP Timeout	NAT TCP idle timeout in seconds. Connection will be closed after preconfigured, timeout if not refreshed. Range: 0 – 3600, default is 300
Uplink Bandwidth	Specifies the maximum uplink bandwidth permitted by the device. This function is disabled by default. The total bandwidth can be set as: 128K, 256K, 512K, 1M, 2M, 3M, 4M, 5M, 10M or 15M. The primary function of this setting is to limit the uplink bandwidth for the device internal system, signaling and NATed traffic. Example: When 512k is configured, there will be at least 512kbps limited for internal system, signaling and NATed traffic. Voice or RTP stream will never be limited.
Downlink Bandwidth	Specifies the maximum downlink bandwidth permitted by the device. This function is disabled by default. The total bandwidth can be set as: 128K, 256K, 512K, 1M, 2M, 3M, 4M, 5M, 10M or 15M. The primary function of this setting is to limit the download bandwidth for the device internal system, signaling and NATed traffic. Example: if 128 is configured, there will be at least 128kbps limited for internal system, signaling and NATed traffic. Voice or RTP stream will never be limited.
Enable UPnP Support	When set to "Yes", the HT813 acts as an UPnP gateway for your UPnP enabled applications. **UPnP = "Universal Plug and Play".** Default is **No**.
Reply to ICMP on WAN Port	When set to "Yes", the HT813 responds to the PING command from other computers but is also made vulnerable to DOS attacks. Default is **No**.

HT813 Administration Guide
Version 1.0.5.2

Cloned WAN MAC Address	This allows the user to change/set a specific MAC address on the WAN interface. <u>Note:</u> Set in Hex format.
Enable LAN DHCP	When set to "Yes", device will function as a simple router and LAN port will provide IP addresses to internal network. Connect the WAN port to ADSL/Cable modem or any other equipment that provides access to public Internet
LAN DHCP Base IP	Base IP Address for a LAN port. Default factory setting is 192.168.2.1. Note: When the device detects WAN IP is conflicting with LAN IP, the LAN base IP address will be changed based on the network mask -- the effective subnet will be increased by 1. For example; 192.168.2.1 will be changed to 192.168.3.1 if net mask is 255.255.255.0. Then the device will reboot
LAN DHCP Start IP	Default value is 100. The last segment of IP address assigned to the HT813 in the LAN Network. Default configuration assigns IP address (to local network devices) starting from 192.168.2.100.
LAN DHCP End IP	Default value is 199. This parameter allows a user to limit the number of local network devices connected to the internal router. Default configuration assigns IP address (to devices connected to the LAN port) in a range from 192.168.2.100 up to 192.168.2.199.
LAN Subnet Mask	Sets the LAN subnet mask. Default value is 255.255.255.0
DHCP IP Lease Time	Default value is 120 hrs (5 days). The length of time the IP address is assigned to the LAN clients. Value is set in units of hours.
DMZ IP	This function forwards all WAN IP traffic to a specific IP address if no matching port is used by HT813 or in the defined port forwarding.
Port Forwarding	Forwards a matching (TCP/UDP) port to a specific LAN IP address with a specific (TCP/UDP) port. Up to 8 rules are available.
Reset Type	Gives the administrator the option to restore default configuration on the HT813.

	There are 3 types of factory reset:
	• **ISP Data Reset:** All VoIP related configuration (mainly everything located on FXS page).
	• **VoIP Data Reset:** All ISP (Internet Service Provider) configuration which may affect the IP address.
	• **Full Reset:** Both VoIP and ISP related configuration at the same time.
	Note: After choosing reset type, you will have to click the reset button for it to take effect.
PSTN Access Code	Key pattern to use PSTN line. Maximum 5 digits. Default is "*00"
PIN for VoIP-to-PSTN Calls	Maximum 8 digits to authorize calling PSTN numbers from VoIP.
PIN for PSTN-to-VoIP Calls	Maximum 8 digits to authorize calling VoIP terminals from PSTN.
Unconditional Call Forward to PSTN	VoIP calls will be forwarded to the specified PSTN number. Specify PSTN number.
Unconditional Call Forward to VoIP	Incoming PSTN calls will be forwarded to the VoIP number. Specify "User ID", "SIP Server" and "SIP Destination Port"

Advanced Settings Page Definitions

Table 8: Advanced Settings

Advanced Settings	
New Admin Password	Defines the administrator level password to access the Advanced Web Configuration page. This field is case sensitive. Only the administrator can configure the "Advanced Settings" page. Password field is purposely left blank for security reasons after clicking update and saved. The maximum password length is 30 characters.
Confirm Admin Password	Re-enter the admin password to confirm change admin password on web GUI to avoid typo or mistakes.
Layer 2 QoS	Sets values for: **802.1Q/VLAN Tag**. Default is **0**. Valid range is 0-4094. **SIP 802.1p**. Default is **0**. Valid range is 0-7. **RTP 802.1p**. Default is **0**. Valid range is 0-7.

Black List for WAN Side Port	It could be either port range or single port separated by a ",". Example: "5000-6000, 7000 ".
STUN Server	Configures IP address or domain name of STUN server. Only non-symmetric NAT routers work with STUN.
Keep-alive Interval	Sends periodically a blank UDP packet to SIP server in order to keep the "ping hole" on the NAT router open. Default is **20** seconds.
Use STUN to detect network connectivity	Uses STUN keep-alive to detect WAN side network problems. If keep-alive request does not yield any response for configured number of times (minimum 3), the device will restart the TCP/IP stack. If the STUN server does not respond when the device boots up, the feature is disabled. Default setting is **No**.
Use DNS to detect network connectivity	Uses DNS to detect WAN side network problems. Default setting is **No**.
Verify host when using HTTPS	Enables / disables the host verification when using HTTPS.
Firmware Upgrade and Provisioning: **Upgrade via**	Selects firmware upgrade/provisioning method: TFTP, HTTP, HTTPS, FTP, or FTPS. Default is **HTTPS**.
Firmware Server Path	Sets IP address or domain name of firmware server. The URL of the server that hosts the firmware release. Default is **fm.grandstream.com/gs**.
Config Server Path	Sets IP address or domain name of configuration server. The server hosts a copy of the configuration file to be installed on the HT813. Default is **fm.grandstream.com/gs**.
XML Config File Password	Decrypts XML configuration file when encrypted. The password used for encrypting the XML configuration file using OpenSSL.
HTTP/HTTPS/FTP/FTPS User Name	Enters user name to authenticate with HTTP/HTTPS/FTP/FTPS server.
HTTP/HTTPS/FTP/FTPS Password	Enters password to authenticate with HTTP/HTTPS/FTP/FTPS server.
Firmware File Prefix	Checks if firmware file is with matching prefix before downloading it. This field enables user to store different versions of firmware files in one directory on the firmware server.

Firmware File Postfix	Checks if firmware file is with matching postfix before downloading it. This field enables user to store different versions of firmware files in one directory on the firmware server.
Config File Prefix	Checks if configuration files are with matching prefix before downloading them. It allows user to store different configuration files in one directory on the provisioning server.
Config File Postfix	Checks if configuration files are with matching postfix before downloading them. It allows user to store different configuration files in one directory on the provisioning server.
Allow DHCP Option 66 or 160 to Override Server	Obtains configuration and upgrade server's information using options 66 from DHCP server. **Note:** If DHCP Option 66 is enabled, the HT813 will attempt downloading the firmware file from the server URL provided by DHCP, even though **Config Server Path** is left blank
3CX Auto Provision	Sends multicast "SUBSCRIBE" message for provisioning at booting stage, used for PnP (Plug-and-Play) configuration. Default is **Yes**.
Automatic Upgrade	Specifies when the firmware upgrade process will be initiated; there are 4 options: • **No**: The HT813 will only do upgrade once at boot up. • **Check every X minutes**: User needs to specify a period in minutes. • **Check every day**: User needs to specify the start hour and the end hour of the day (0-23). • **Check every week**: User needs to specify "Day of the week (0-6)". (Day of week is starting from Sunday). Default is **No**.
Randomized Automatic Upgrade	Randomized Automatic Upgrade within the range of hours of the day or postpone the upgrade every X minute(s) by random 1 to X minute(s).
Always Check for New Firmware at Boot up	Configures the HT813 to always search for the new firmware at boot up. During the boot stage, the HT813 will contact the firmware upgrade server to search for a new firmware, when available it will start the upgrade process, otherwise it will boot normally.
Check New Firmware only when F/W pre/suffix changes	Configure the HT813 to search for the new firmware when the firmware prefix / suffix changes. When this option is selected, the HT813 will check for updates only when the pre/suffix has been changed.

Always Skip the Firmware Check	Configures the HT813 to skip the firmware check, when this option is selected the HT813 will always skip searching for a new firmware.
Disable SIP NOTIFY Authentication	Disables the SIP NOTIFY Authentication on the ATA adapter. If set to "Yes", the ATA adapter will not challenge NOTIFY with 401. Default is **No**
Authenticate Conf File	Authenticates configuration before being accepted. This protects the configuration from unauthorized modifications. Default is **No**.
Validate Server Certificates	This feature allows users to validate server certificates with our trusted list of TLS connections. Default is enabled. The device needs to reboot after changing the setting.
SIP TLS Certificate	Specifies SSL certificate used for SIP over TLS is in X.509 format. The HT813 has built-in private key and SSL certificate.
SIP TLS Private Key	Specifies TLS private key used for SIP over TLS is in X.509 format.
SIP TLS Private Key Password	Specifies SSL Private key password used for SIP Transport in TLS/TCP.
Custom Certificate (Private Key + Certificate)	Allows users to update to the device their own certificate signed by custom CA certificate to manage client authentication.
Enable TR-069	Sets the ATA adapter system to enable the "CPE WAN Management Protocol" (TR-069). Default setting is **No**.
ACS URL	Specifies URL of TR-069 Auto Configuration Servers (e.g., http://acs.mycompany.com), or IP address.
ACS Username	Enters username to authenticate to ACS.
ACS Password	Enters password to authenticate to ACS.
Periodic Inform Enable	Sends periodic inform packets to ACS. Default is **No**
Periodic Inform Interval	Sets frequency that the inform packets will be sent out to ACS.
Connection Request Username	Enters username for ACS to connect to the HT813.
Connection Request Password	Enters password for ACS to connect to the HT813.

HT813 Administration Guide
Version 1.0.5.2

Connection Request Port	Configures the TR-069 Connection Request Port. The value range is 0 to 65535.Default is 7547
CPE SSL Certificate	Configures the Cert File for the ATA to connect to the ACS via SSL.
CPE SSL Private Key	Specifies the Cert Key for the ATA to connect to the ACS via SSL.
Enable SNMP	Default is **No**.
SNMP Version	Choose between (Version 1, Version 2c, or Version 3).
SNMP Port	Listening Port of SNMP daemon (Default 161).
SNMP Trap IP Address	IP address of trap destination.
Port of Trap port	Port of Trap destination (Default 162)
SNMP Trap Version	Choose between (Version 1, Version 2c, or Version 3).
SNMP Trap Interval	Time interval between traps (Default is 5).
SNMPv1/v2c Community	Name of SNMPv1/v2c community.
SNMPv1/v2c Trap Community	Name of SNMPv1/v2c trap community.
SNMPv3 User Name	User name for SNMPv3.
SNMPv3 Security Level	• **noAuthUser**: Users with security level noAuthnoPriv and context name as noAuth. • **authUser**: Users with security level authNoPriv and context name as auth. • **privUser** : Users with security level authPriv and context name as priv.
SNMPv3 Authentication Protocol	Select the Authentication Protocol: "None" or "MD5" or "SHA."
SNMPv3 Privacy Protocol	Select the Privacy Protocol: "None" or "AES/AES128" or "DES".
SNMPv3 Authentication Key	Enter the Authentication Key.
SNMPv3 Privacy Key	Enter the Privacy Key.

SNMPv3 Trap User Name	User name for SNMPv3 Trap.
SNMPv3 Trap Security Level	**noAuthUser:** Users with security level noAuthnoPriv and context name as noAuth. **authUser:** Users with security level authNoPriv and context name as auth. **privUser:** Users with security level authPriv and context name as priv.
SNMPv3 Trap Authentication Protocol	Select the Authentication Protocol: "None" or "MD5" or "SHA".
SNMPv3 Trap Privacy Protocol	Select the Privacy Protocol: "None" or "AES/AES128" or "DES".
SNMPv3 Trap Authentication Key	Enter the Trap Authentication Key.
SNMPv3 Trap Privacy Key	Enter the Trap Privacy Key.
Enable RADIUS Web Access Control	Default is **No**.
Action upon RADIUS Auth Server Error	Choose action upon RADIUS server error. Default is Authenticate Locally (Default Authenticate Locally)
RADIUS Auth Server Address	Address of RADIUS Auth server.
RADIUS Auth Server Port	Port of RADIUS Auth server.
RADIUS Shared Secret	Set RADIUS shared secret.
RADIUS VSA Vendor ID	Configure RADIUS VSA Vendor ID to match RADIUS server's configuration. Default is 42397 for Grandstream Networks Inc.
RADIUS VSA Access Level Attribute	Configure RADIUS VSA Access Level Attribute to match RADIUS server's configuration. Incorrect setting would cause Radius authenticate fail.
Enable DDNS	Allow users to use DDNS.
DDNS Server	Selects DDNS Server: dyndns.org, freedns.afraid.org, zoneedit.com, no-ip.com, oray.net. Default is dyndns.org.

DDNS Username	Enter DDNS Username. 64 characters as Max String Length.
DDNS Password	Enter DDNS Password. 64 characters as Max String Length.
DDNS Hostname	Enter DDNS Hostname. 64 characters as Max String Length.
DDNS Hash	Enter DDNS Hash. 64 characters as Max String Length.
System Ring Cadence	Configuration option is set ring cadence on FXS port for all incoming calls. *Syntax:* **c=on1/off1-on2/off2-on3/off3;** (3 cadences maximum) Default is set to **c=2000/4000;** (US standards)
Call Progress Tones: • **Dial Tone** • **Ringback Tone** • **Busy Tone** • **Reorder Tone** • **Confirmation Tone** • **Call Waiting Tone** • **Prompt Tone**	Using these settings, users can configure tone frequencies and cadence according to their preference. By default, they are set to North American frequencies. Configure these settings with known values to avoid uncomfortable high pitch sounds. ON is the period of ringing ("On time" in 'ms') while OFF is the period of silence. In order to set a continuous tone, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern. Example configuration for N.A. Dialtone: f1=350@-13, f2=440@-13,c=0/0; *Syntax:* **f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3; [...]** (Note: freq: 0 - 4000Hz; vol: -30 - 0dBm)
Prompt Tone Access Code	Key pattern to get Prompt Tone. Maximum 20 digits.
Lock Keypad Update	Configuration update via keypad (analog phone connected to FXS port keypad using IVR menu) is disabled if set to Yes.
Disable Voice Prompt	Voice prompt is disabled if set to Yes.
Disable Direct IP Call	Direct IP call is disabled if set to Yes.
Life Line Mode	Life line feature ensures user can place/receive a PSTN call in an emergency situation. Three modes are supported: • **Auto:** In case of power loss or loss of SIP registration, the PSTN line will be seamlessly connected to analog phone connected to FXS port.

	• **Always Connected:** PSTN line will be always connected to the phone connected to FXS port. VoIP calls will not be allowed in this configuration. • **Always Disconnected:** User can only make/receive VoIP calls. PSTN calls will not be possible. Default setting is **Auto**.
Blacklist for Incoming Calls	Allow users to block incoming calls from specific list of numbers. Maximum allow 10 SIP numbers and each number should be separated by a comma (',') in web UI. Other allowed characters are 0-9, comma (","), asterisk ('*'), pound sign ('#') and plus sign ('+').
NTP Server	Defines the URL or IP address of the NTP server. The ATA may obtain the date and time from the server. The default setting is "pool.ntp.org".
Allow DHCP Option 42 to override NTP server	Defines whether DHCP Option 42 should override NTP server or not. When enabled, DHCP Option 42 will override the NTP server if it's set up on the LAN. The default setting is **Yes**.
DHCP Option 17 Enterprise Number	This option contains vendor-specific option data, much like DHCPv4 option 43. There is an extra difference in that in DHCPv6, this option carries a vendor ID as well, which allows for data from multiple vendors to be provided to the device. Default is **3561**.
Disable Weak TLS Cipher Suites	This feature allows users to disable weak ciphers. The given choices are: 1- Enable Weak TLS Ciphers Suites. 2- Disable Symmetric Encryption RC4/DES/3DES. 3- Disable Symmetric Encryption SEED. 4- Disable All Of The Above Weak Symmetric Encryption. 5- Disable Symmetric Authentication MD5. 6- Disable All Of The Above Weak Symmetric Authentication. 7- Disable Protocol Version SSLv2/SSLv3. Default is **Enable Weak TLS Ciphers Suites**.
Minimum TLS Version	The Feature allows users to choose the Minimum TLS Version. Choices are: 1- Unlimited. 2- TLS 1.0.

	3- TLS 1.1. 4- TLS 1.2. Default is **Unlimited**.
Maximum TLS Version	The Feature allows users to choose the Maximum TLS Version. Choices are: 1- Unlimited. 2- TLS 1.0. 3- TLS 1.1. 4- TLS 1.2. Default is **Unlimited**.
Syslog Protocol	The Feature allow users to customize the Syslog Protocol. The Syslog protocol can be either UDP or SSL/TLS. Default us **UDP**.
Syslog Server	URL or IP address of syslog server.
Syslog Level	Select the HT813 to report the log level. Default is NONE. The level is one of EXTRA DEBUG, DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: 1. product model/version on boot up (INFO level) 2. NAT related info (INFO level) 3. sent or received SIP message (DEBUG level) 4. SIP message summary (INFO level) 5. inbound and outbound calls (INFO level) 6. registration status change (INFO level) 7. negotiated codec (INFO level) 8. Ethernet link up (INFO level) 9. SLIC chip exception (WARNING and ERROR levels) 10. memory exception (ERROR level) 11. extra syslog style (EXTRA DEBUG level)
Send SIP Log	Configures whether the SIP log will be included in the syslog messages. The default setting is **No**.
Automatic Reboot	Default is **No**. When "Yes, reboot every day at hour" or "Yes, reboot every week at day" or "Yes, reboot every month at day" is checked, user can specify "Hour of the day (0-23)" or "Day of the week (0-6)" or "Day of the month (0-30)". Default time is Monday 1AM.

Download Device Configuration	Allows user to download and save a text file containing all the P values of each setting as configured at that point on the unit. For Security Reasons, Passwords won't be Downloaded.
Download Device XML Configuration	Allows user to download and save an XML file containing all the P values of each setting as configured at that point on the unit. For Security Reasons, Passwords won't be Downloaded.
Upload Firmware	Allows the user to upgrade the firmware with a single firmware file by browsing and loading the file from your computer (local directory).
Upload Configuration	Allows to upload configuration file to the device.
Export Backup Configuration	Download backup file to local computer. The backup file is XML and encrypted.
Restore From Backup Configuration	Upload backup file to the ATA to restore your saved configuration

FXS Port Page Definitions

Table 9: FXS Page

Account Active	Activates / Deactivates the accounts. The FXS port configuration will not change if disabled, although the port will not be operational, in this state, there will be no dial tone when picking up the analog phone and making/receiving calls will not be possible.
Primary SIP Server	Configures SIP server IP address or domain name provided by VoIP service provider. This is the primary SIP server used to send/receive SIP messages from/to HT813.
Failover SIP Server	Specifies failover SIP server IP address or domain name provided by VoIP service provider. This server will be used if the primary SIP server becomes unavailable.
Prefer Primary SIP Server	Selects to prefer primary SIP server. The account will register to primary Server if registration with Failover server expires. Default is **No**.
Outbound Proxy	Specifies IP address or domain name of outbound Proxy, or media gateway, or session border controller. Used by HT813 for firewall or NAT penetration in

	different network environments. If symmetric NAT is detected, STUN will not work and only outbound proxy can correct the problem.
Backup Outbound Proxy	Configures the backup outbound proxy to be used when the "Outbound Proxy" registration fails. By default, this field is left empty.
Prefer Primary Outbound Proxy	If the user configures this option to "**Yes**", when registration expires, the device will re-register via primary outbound proxy. By default, this option is disabled.
Allow DHCP Option 120 (override SIP Server)	Configures the HT813 to collect SIP server address from DHCP option 120. Default is **No**.
SIP Transport	Selects transport protocol for SIP packets; UDP or TCP or TLS. Please make sure your SIP Server or network environment supports SIP over the selected transport method. Default is **UDP**.
SIP URI Scheme When Using TLS	Specifies if "sip" or "sips" will be used when TLS/TCP is selected for SIP Transport. The default setting is "sips".
Use Actual Ephemeral Port in Contact with TCP/TLS	Controls the port information in the Via header and Contact header. If set to "No", these port numbers will use the permanent listening port on the phone. Otherwise, it will use the ephemeral port for the connection. The default setting is "No".
NAT Traversal	Indicates type of NAT for each account. This parameter configures whether the NAT traversal mechanism is activated. Users could select the mechanism from No, Keep-alive, STUN, UPnP. Default setting is **No**.
SIP User ID	Defines user account information provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
Authenticate ID	Determines account authenticate ID provided by VoIP service provider (ITSP). Can be identical to or different from "SIP user ID".
Authenticate Password	Specifies account password provided by VoIP service provider (ITSP) to register to SIP servers.
Name	Chooses a name to be associated to user.

DNS Mode	Selects DNS mode to use for the client to look up server. Default is **A Record**. • **A Record:** resolves IP Address of target according to domain name. • **SRV:** DNS SRV resource records indicate how to find services for various protocols. • **NAPTR/SRV**: Naming Authority Pointer according to RFC 2915.
DNS SRV use Registered IP	When this option is set to "Yes", when the HT is registered on second SRV and makes an outbound call, it will try the second SRV (registered IP) first. By default, this option is disabled and the DNS SRV will use first SRV instead of the registered IP.
Tel URI	Indicates E.164 number in "From" header by adding "User=Phone" parameter or using "Tel:" in SIP packets, if the HT813 has an assigned PSTN Number. • **Disabled:** Use "SIP User ID" information in the Request-Line and "From" header. • **User=Phone:** "User=Phone" parameter will be attached to the Request-Line and "From" header in the SIP request to indicate the E.164 number. If set to "Enable". • **Enabled:** "Tel:" will be used instead of "sip:" in the SIP request. Please consult your carrier before changing this parameter. Default is **Disabled**.
SIP Registration	Controls whether the HT813 needs to send REGISTER messages to the proxy server. Default setting is **Yes**.
Unregister on Reboot	Controls whether to clear SIP user's information by sending un-register request to the proxy server. The un-registration is performed by sending a REGISTER message with Contact set to * and Expires=0 parameters to the SIP server. This will unregister the SIP account under the concerned FXS page. Default is **No**.
Outgoing Call Without Registration	Enables the ability to place outgoing calls even if the account is not registered (if allowed by ITSP); device will not be able to receive incoming calls. Default is **No**.
Register Expiration	Refreshes registration periodically with specified SIP proxy (in minutes). Maximum interval is 65535 minutes (about 45 days). Default is **60** minutes (or 1 hour).

Reregister before Expiration	Sends re-register request after specific time (in seconds) to renew registration before the previous registration expires.
SIP Registration Failure Retry Wait Time	Sends re-register request after specific time (in seconds) when registration process fails. Maximum interval is 3600 seconds (1 hour). Default is **20** seconds.
SIP Registration Failure Retry Wait Time upon 403 Forbidden	Sends re-register request after specific time (in seconds) when registration process fails with error 403 Forbidden. Maximum interval is 3600 seconds (1 hour). Default is **1200** seconds.
Enable SIP OPTIONS Keep Alive	Enables SIP OPTIONS to track account registration status so the phone adapter will send periodic OPTIONS message to server to track the connection status with the server. Default setting is **No**.
SIP OPTIONS Keep Alive Interval	Configures the time interval when the phone adapter sends OPTIONS message to SIP server. The default setting is 30 seconds, which means the phone adapter will send an OPTIONS message to the server every 30 seconds. The default range is **1-64800.**
SIP OPTIONS Keep Alive Max Lost	Defines the Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is **3**.
Layer 3 QoS	Defines Diff-Serv values for SIP and RTP. Defaults are: SIP DSCP: **26** RTP DSCP: **46**
Local SIP Port	Defines local port to use by the HT813 for listening and transmitting SIP packets. Default value for FXS is 5060.
Local RTP Port	Defines the local RTP-RTCP port pair the HT813 will listen and transmit. It is the HT813 RTP port for channel 0. The default value for FXS port is **5004.**
Use Random SIP Port	Controls whether to use configured or random SIP ports. This is usually necessary when multiple HT813 are behind the same NAT. Default is **No**.
Use Random RTP Port	Controls whether to use configured or random RTP ports. This is usually necessary when multiple HT813 are behind the same NAT. Default is **No**.
Enable RTCP	Allows users to enable RTCP. Default setting is **Yes**.
Hold Target Before Refer	Allows user to hold the phone call before referring it. If set to No, the call will not be hold before referred. Default is **Yes**.

Refer-To Use Target Contact	Includes target's "Contact" header information in "Refer-To" header when using attended transfer. Default is **No**.
Transfer on Conference Hang-up	If set to "Yes", when the phone hangs up as the conference initiator, the conference call will be transferred to the other parties so that other parties will remain in the conference call. Default setting is **No**.
Disable Bellcore Style 3-Way Conference	Gives the users the possibility of making conference calls by pressing "Flash" key, when it's enabled by dialing *23 +second callee number. Default is **No**
Remove OBP from Route Header	Removes outbound proxy info in "Route" header when sending SIP packets. Default is **No**.
Support SIP Instance ID	Includes "SIP Instance ID" attribute to "Contact" header in REGISTER request as defined in IETF SIP outbound draft. Default is **No**.
Validate Incoming SIP Messages	Validates incoming messages. Default is **No**.
Check SIP User ID for Incoming INVITE	Checks SIP User ID in the Request URI of incoming INVITE; if it doesn't match the HT813 SIP User ID, the call will be rejected. Direct IP calling will also be disabled. Default is **No**.
Authenticate Incoming INVITE	Challenges the incoming INVITE for authentication with SIP 401 Unauthorized message. Default is **No**.
Authenticate server certificate domain	Configures whether to validate the domain certificate when download the firmware/config file. If it is set to "Yes", the phone will download the firmware/config file only from the legitimate server. The default setting is "**No**".
Authenticate server certificate chain	Configures whether to validate the server certificate when download the firmware/config file. If it is set to "Yes", the phone will download the firmware/config file only from the legitimate server. The default setting is "**No**".
Trusted CA Certificates	Uses the certificate for Authentication if "Check Domain Certificates" is set to "Yes" under "Account"→"SIP Settings".
Allow Incoming SIP Messages from SIP Proxy Only	Checks SIP address of the Request URI in the incoming SIP message; if it doesn't match the SIP server address of the account, the call will be rejected. Default is **No**.
Use Privacy Header	Determines if the "Privacy header" will be presented in the SIP INVITE message and if it includes the caller info in this header. If set to Default, it will add Privacy header unless special feature is **Telkom SA** or **CBCOM**. Default is **Default**.

Use P-Preferred-Identity Header	Specifies if the P-Preferred-Identity Header will be presented in the SIP INVITE message. If set to "default", the P-Preferred-Identity Header will be omitted in SIP INVITE message when **Telkom SA** or **CBCOM** is active. If set to "Yes", the P-Preferred-Identity Header will always be presented. If set to "No", it will be omitted. Default setting is: **Default**.
Use P-Access-Network-Info Header	With this feature enabled, device will populate the WAN access node with IEEE802.11a, IEEE-802.11b in P-Access-Network-Info SIP header.
Use P-Emergency-Info Header	This feature support of IEEE-48-addr and IEEE-EUI-64 in SIP header for emergency calls.
SIP REGISTER Contact Header Uses	Specifies which address (LAN or WAN address) the device will detect to use it in SIP Register Contact Header. When set to **LAN**, Contact header will include local IP from ATA in REGISTER messages, while if set to **WAN**, host/port/contact will be updated from SIP 401/403/404/407 Via header "received"/"rport" parameters in REGISTER messages. Default is **LAN Address**.
Caller ID Fetch Order	Selects the Caller ID display order which need to be respected by the HT813. The available options are: ● **Auto**: When set to "Auto", the HT813 will look for the caller ID in the order of P-Asserted Identity Header, Remote-Party-ID Header and From Header in the incoming SIP INVITE. ● **Disabled**: When set to "Disabled", all incoming calls are displayed with "Unavailable". ● **From Header**: When set to "From Header", the HT813 will use the FROM header to display the caller ID.
Allow SIP Factory Reset	This feature allows user to reset the devices directly through SIP Notify. Default is **No**.
SIP T1 Timeout	Defines T1 timeout value. It is an estimate of the round-trip time between the client and server transactions. For example, the HT813 will attempt to send a request to a SIP server. The time it takes between sending out the request to the point of getting a response is the SIP T1 timer. If no response is received the timeout is increased to (2*T1) and then (4*T1). Request re-transmit retries would continue until a maximum amount of time defined by T2. Default is **0.5**

	seconds.
SIP T2 Interval	Identifies maximum retransmission interval for non-INVITE requests and INVITE responses. Retransmitting and doubling of T1 continues until it reaches T2 value. Default is **4** seconds.
SIP Timer D	Configure the SIP Timer D defined in RFC3261. 0 - 64 seconds. Default 0
DTMF Payload Type	Defines payload type for DTMF using RFC2833.
Preferred DTMF method (in order)	Sorts DTMF methods (in-audio, via RTP (RFC2833) or via SIP INFO) by priority.
Disable DTMF Negotiation	Uses above DTMF order without negotiation. Default is **No**.
Generate Continuous RFC2833 Events	When enabled the RFC2833 events are generated until key is released. Default is **No**.
Send Hook Flash Event	Default is **No.** If set to yes, flash will be sent as DTMF event.
Flash Digit Control	When it set to YES it allows the user to perform some call setting when both channels are used while pressing: • **"Flash + 1"** in order to hang up the current call and resume a call that was held. • **"Flash + 2"** in order to hold the current call and resume a call that was held. • **"Flash + 3"** in order to perform 3-way conference. • **"Flash + 4"** in order to perform attended transfer. **Note:** Please refer to the user guide for detailed steps to perform above operations.
Enable Call Features	Enables do not disturb, call forward and other call features via the local feature codes on the base. Otherwise, ITSP feature codes can be used. Default is **Yes**.
Off Hook Auto Dial	Configures a user ID or extension number that is automatically dialed when off-hook. Only the user part of a SIP address needs to be entered. FXS port will automatically append the "@" and the host portion of the corresponding SIP address.
Off Hook Auto Dial Delay	Specifies the auto-dial delay after off hook.

Proxy-Require	Determines a SIP Extension to notify the SIP server that the HT813 is behind a NAT/Firewall.
Use NAT IP	Defines NAT IP address used in SIP/SDP messages. It should only be used if required by ITSP.
Use SIP User Agent Header	Configures the SIP User-Agent Header.
SIP User-Agent	Configures SIP User-Agent. If not configured, device will use the default User Agent Header. The value range is 1024 to Maximum String Length. Default value is Null.
Distinctive Ring Tone	Customizes the Ring Tone 1 to 3 with associate caller ID: when selected, if caller ID is configured, then the device will ONLY use this ring tone when the incoming call is from the Caller ID. System Ring Tone is used for all other calls. When selected but no Caller ID is configured, the selected ring tone will be used for all incoming calls using the FXS port. Distinctive ring tones can be configured not only for matching a whole number, but also for matching prefixes. In this case symbol "x+" will be used.
	For example: If configured as 617x+, Ring Tone 1 will be used in case of call arrived from the area code 617. Any other incoming call will ring using cadence defined in parameter System Ring Cadence located under Advanced Settings Configuration page.
	Note: If server supports Alert-Info header and standard ring tone set (Bellcore) or distinctive ring tone 1-10 is specified, then the ring tone in the Alert-Info header from server will be used. Bellcore rings and tones are independent from custom ring tones. The custom ring tones can also be specified by alert-info header, for example *Alert-Info:; info=ring5*
Disable Call Waiting	Disables receiving a second incoming call when the line is engaged. Default is **No**.
Disable Call Waiting Caller ID	Disables displaying caller ID when receiving a second incoming call. Default is **No**.
Disable Call Waiting Tone	Disables playing call waiting tone during active call when receiving a second incoming call. The CWCID will still be displayed. Default is **No**.
Disable Connected Line ID	Disables displaying the number of the person answering the phone. Default is **No**.

HT813 Administration Guide
Version 1.0.5.2

Disable Receiver Off Hook Tone	Enables / disables the warning to alert that the phone has been left off-hook for an extended period of time. Default is **No**.
Disable Reminder Ring for On-Hold Call	Enables playing the reminder ring. Default is **No**
Disable Visual MWI	Disables use of visual message waiting indicator when there is an unread voicemail message. Default is **No**.
Do Not Escape '#' as %23 in SIP URI	Replaces # by %23 in some special situations. Default is **No**.
Disable Multiple m Line in SDP	Sends only one m line in SDP, regardless of how many m fields are in the incoming SDP. Default is **No**.
Ring Timeout	Stops ringing when incoming call if not answered within a specific period of time. When set to 0, There will be no ringing timeout. Default is **60** seconds.
Delayed Call Forward Wait Timeout	Forwards incoming call if not answered within a specific period of time when delayed call forward is activated locally (using *92 code). Default value is **20** seconds.
No Key Entry Timeout	Initiates the call within this time interval if no additional key entry during dialing stage. Default is **4** seconds.
Early Dial	Sends an early INVITE each time a key is pressed when a user dials a number. Otherwise, only one INVITE is sent after full number is dialed (user presses **Dial Key** or after "no key entry timeout" expires). This option should be used only if there is a SIP proxy is configured and supporting 484 responses (Incomplete Address). Otherwise, the call will likely be rejected by the proxy (with a 404 Not Found error). Default is **No**. *This feature is NOT designed to work with and should NOT be enabled for direct IP-to-IP calling*.
Dial Plan Prefix	Adds specified prefix to dialed number.
Use # as Dial Key	Treats "#" as the "Send" (or "Dial") key. If set to "No", this "#" key can be included as part of the dialed number. Default is **Yes**.

Dial Plan	**Dial Plan Rules**:
	1. Accept Digits: 1,2,3,4,5,6,7,8,9,0 , *, #, A,a,B,b,C,c,D,d
	2. Grammar: **x** - any digit from 0-9;
	a. **xx+** - at least 2 digits number;
	b. **xx** – exactly 2 digits number;
	c. **^** - exclude;
	d. **.** – wildcard, matches one or more characters
	e. **[3-5]** - any digit of 3, 4, or 5;
	f. **[147]** - any digit 1, 4, or 7;
	g. **<2=011>** - replace digit 2 with 011 when dialing
	h. **< =1>** - add a leading 1 to all numbers dialed, vice versa will remove a 1 from the number dialed
	i. **|** - or
	• **Example 1:** {[369]11	1617xxxxxxx} – Allow 311, 611, 911, and any 10 digit numbers of leading digits 1617
	• **Example 2:** {^1900x+	<=1617>xxxxxxx} – Block any number with leading digits 1900 and add prefix 1617 for any dialed 7 digit numbers
	• **Example 3:** {1xxx[2-9]xxxxxx	<2=011>x+} – Allow any length of number with leading digit 2 and 10 digit-numbers of leading digit 1 and leading exchange number between 2 and 9; If leading digit is 2, replace leading digit 2 with 011 before dialing.
	3. Default: Outgoing - {x+}
	Example of a simple dial plan used in a Home/Office in the US:
	**{ ^1900x.	<=1617>[2-9]xxxxxx	1[2-9]xx[2-9]xxxxxx	011[2-9]x.	[3469]11 }**
	Explanation of example rule (reading from left to right):
	• **^1900x. -** prevents dialing any number started with 1900
	• **<=1617>[2-9]xxxxxx** - allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically
	• **1[2-9]xx[2-9]xxxxxx** - allows dialing to any US/Canada Number with 11-digit length
	• **011[2-9]x. -** allows international calls starting with 011
	• **[3469]11 -** allow dialing special and emergency numbers 311, 411, 611 and 911
	Note: In some cases, user wishes to dial strings such as *123 to activate

	voice mail or other application provided by service provider. In this case * should be predefined inside dial plan feature. As an example **{ *x+ }** will allow to dial * followed by any length of numbers.
SUBSCRIBE for MWI	Sends SUBSCRIBE periodically (depends on "Register Expiration" parameter) for message waiting indication. Default is **No**.
Send Anonymous	Sets "From", "Privacy" and "P_Asserted_Identity" headers in outgoing INVITE message to "anonymous", blocking caller ID. Default is **No**.
Anonymous Call Rejection	Rejects incoming calls with anonymous caller ID with "486 Busy here" message. Default is **No**.
Special Feature	Selects Soft switch vendors' special requirements Example of vendors: Standard, Broadsoft, CBCOM, RNK, Huawei, China Mobile, ZTE IMS, PhonePower, TELKOM SA, Vonage, Metaswitch, CenturyLink, MTS. Default is **Standard**.
Enable Session Timer	Disable the session timer when this option is set to "No". By default, this option is enabled.
Session Expiration	Enables SIP sessions to be periodically "refreshed" via a SIP request (UPDATE, or re-INVITE). When the session interval expires, if there is no refresh via an UPDATE or re-INVITE message, the session will be terminated. Session Expiration is the time (in seconds) at which the session is considered timed out, if no successful session refresh transaction occurs beforehand. Default is **180** seconds.
Min-SE	Defines Minimum session expiration (in seconds). Default is **90** seconds.
Caller Request Timer	Uses session timer when making outbound calls if remote party supports it. Default is **No**.
Callee Request Timer	Uses session timer when receiving inbound calls with session timer request. Default is **No**.
Force Timer	Uses session timer even if the remote party does not support this feature. Selecting "No" will enable session timer only when the remote party supports it. Default is **No**. To turn off Session Timer, select "No" for Caller and Callee Request Timer, and Force Timer.
UAC Specify Refresher	Specifies which end will act as refresher for outgoing calls. • **UAC**: The handy tone acts as the refresher. • **UAS**: Callee or proxy server act as the refresher.

	Default is **Omit**.
UAS Specify Refresher	Specifies which end will act as refresher for incoming calls: • **UAS**: The handy tone acts as the refresher. • **UAC**: Callee or proxy server act as the refresher. Default is **Omit**.
Force INVITE	Uses INVITE message to refresh the session timer. Default is **No**.
Enable 100rel	Appends "100rel" attribute to the value of the required header of the initial signaling messages. Default is **No**.
Add Auth Header on Initial REGISTER	Adds "Authentication" header with blank "nonce" attribute in the initial SIP REGISTER request. Default is **No**.
Conference URI	Allows users to manually configure the conference URL.
Use First Matching Vocoder in 200OK SDP	Includes only the first matching vocoder in its 200OK response, otherwise it will include all matching vocoders in same order received in INVITE. Default is **No**.
Preferred Vocoder	Configures vocoders in a preference list (up to 7 preferred vocoders) that will be included with same order in SDP message. Vocoder types are G.711 A-/U-law, G.726-32, G.723, G.729, iLBC and OPUS.
Voice Frames per TX	Transmits a specific number of voice frames per packet. Default is **2**; increases to 10/20/32/64 for G711/G726/G723/other codecs respectively.
G723 Rate	Operates at specified encoding rate for G.723 vocoder. Available encoding rates are 6.3kbps or 5.3kbps. Default is **6.3kbps**.
iLBC Frame Size	Specifies iLBC packet frame size (20ms or 30ms). Default is **20ms**.
Disable OPUS Stereo in SDP	Disables OPUS stereo in SDP. Default is **No**.
iLBC Payload type	Determines payload type for iLBC. Valid range is between 96 and 127. Default is **97**.
OPUS Payload Type	Determines payload type for OPUS. Valid range is between 96 and 127. Default is **123**.

VAD	Allows detecting the absence of audio and conserves bandwidth by preventing the transmission of "silent packets" over the network. Default is **No**.
Symmetric RTP	Changes the destination to send RTP packets to the source IP address and port of the inbound RTP packet last received by the device. Default is **No**.
Fax Mode	Specifies the fax mode: T.38 (Auto Detect) FoIP by default, or Pass-Through (must use codec PCMU/PCMA)
Re-INVITE after Fax Tone Detected	Allows the unit to send out the re-INVITE for T.38 or Fax Pass Through if a fax tone is detected. Default is **Enabled**
Jitter Buffer Type	Selects jitter buffer type (Fixed or Adaptive) based on network conditions.
Jitter Buffer Length	• **High** (initial 200ms, min 40ms, max 600ms) Note: not all vocoders can meet the high requirement. • **Medium** (initial 100ms, min 20ms, max 200ms). • **Low** (initial 50ms, min 10ms, max 100ms).
SRTP Mode	Selects SRTP mode to use ("Disabled", "Enabled but not forced", or "Enabled and forced"). Default is **Disabled.** It uses SDP Security Description to exchange key. Please refer to SDES: https://tools.ietf.org/html/rfc4568 SRTP: https://www.ietf.org/rfc/rfc3711.txt
Crypto Life Time	Adds crypto life time header to SRTP packets. Default is **Yes**.
SLIC Setting	Depends on standard phone type (and location). Available options: • USA 1 (BELLCORE 600 ohms) • USA 2 (BELLCORE 600 ohms + 2.16uF) • AUSTRALIA • CHINA CO • CHINA PBX • EUROPEAN CTR21 • GERMANY • INDIA/NEW ZEALAND • JAPAN CO • JAPAN PBX • STANDARD 900 ohms • UK

Caller ID Scheme	Selects the caller ID scheme. Available options: • Bellcore/Telcordia • ETSI-FSK during ringing • ETSI-FSK prior to ringing with DTAS • ETSI-FSK prior to ringing with LR+DTAS • ETSI-FSK prior to ringing with RP • ETSI-DTMF during ringing • ETSI-DTMF prior to ringing with DTAS • ETSI-DTMF prior to ringing with LR+DTAS • ETSI-DTMF prior to ringing with RP • SIN 227 – BT • NTT JAPAN • DTMF Denmark prior to ringing with no DTAS no LR • DTMF Denmark prior to ringing with LR • DTMF Sweden/Finland prior to ringing with LR • DTMF Brazil • DTMF-FSK Brazil
DTMF Caller ID	Defines the start and stop tones as delimiters for the caller ID. **Start Tone** and **Stop Tone** can be set to "Default", "A", "B", "C", "D" or "#"
Polarity Reversal	Reverses the polarity upon call establishment and termination. Default is **No**.
Loop Current Disconnect	Allows the traditional PBX used with HT813 to apply this method for signaling call termination. Method initiates short voltage drop on the line when remote (VoIP) side disconnects an active call. Default is **No**.
Play busy/reorder tone before Loop Current Disconnect	Allow user to configure if it will play busy/reorder tone before loop current disconnect upon call fail. Default is **No**.
Loop Current Disconnect Duration	Configures the duration of voltage drop described in topic above. HT813 supports a duration range from 100 to 10000 ms. Default value is **200**.
Enable Pulse Dialing	Allow users to enable Pulse Dialing option under FXS Port. Default is **No**.
Pulse Dialing Standard	This feature allows users to use Swedish pulse dialing standard. Default is **General Standard**.
Enable Hook Flash	Enables the FLASH button to be used for terminating calls. Default is **Yes**.
Hook Flash Timing	Defines the time period when the cradle is pressed (Hook Flash) to simulate FLASH. To prevent unwanted activation of the Flash/Hold and automatic phone ring-back, adjust this time value. HT813 supports a range from 40 to

	2000 ms. Default values are **300** minimum and **1100** maximum.
On Hook Timing	Specifies the on-hook time for an on-hook event to be validated. HT813 supports a range from 40 to 2000 ms. Default value is **400**.
Gain	Adjusts the voice path volume. • **Rx** is a gain level for signals transmitted by FXS • **Tx** is a gain level for signals received by FXS. Default = 0dB for both parameters. Loudest volume: +6dB Lowest volume: -6dB. User can adjust volume of call using the Rx gain level parameter and the Tx gain level parameter located on the FXS port configuration page. If call volume is too low when using the FXS port (i.e. the ATA is at user site), adjust volume using the Rx gain level parameter under the FXS port configuration page. If voice volume is too low at the other end, user may increase the far end volume using the Tx gain level parameter under the FXS port configuration page.
Disable Line Echo Canceller (LEC)	Disables the LEC will per call base. Recommended for FAX/Data calls. Default is **No**.
Disable Network Echo Suppressor	Disables the NEC will per call base. Recommended for FAX/Data calls. Default is **No**.
Outgoing Call Duration Limit	Defines the call duration limit for the outgoing calls, Default is **0 (No limit)**.
Enable High Ring Power	Configures a high ringing voltage output for the FXS port of HT813.
RFC2833 Events Count	This feature allows users to customize the count of RFC2833 events. Supported range is 2-10. Default is **8**.
RFC2833 End Events Count	This feature allows users to customize the count of RFC2833 end events. Supported range is 2-10. Default is **3**.
Ring tones	Configures the ring tone cadence preferences. User has 10 choices. The configuration, completed in Distinctive Ring Tones block in the same page, applies to ring tones cadences configured here.

FXO Port Page Definitions

Table 10: FXO Page

Account Active	Activates / Deactivates the accounts. The FXO port configuration will not change if

	disabled, although the port will not be operational, in this state, there will be no dial tone when picking up the analog phone and making/receiving calls will not be possible.
Primary SIP Server	Configures SIP server IP address or domain name provided by VoIP service provider. This is the primary SIP server used to send/receive SIP messages from/to HT813.
Failover SIP Server	Specifies failover SIP server IP address or domain name provided by VoIP service provider. This server will be used if the primary SIP server becomes unavailable.
Prefer Primary SIP Server	Selects to prefer primary SIP server. The account will register to primary Server if registration with Failover server expires. Default is **No**.
Outbound Proxy	Specifies IP address or domain name of outbound Proxy, or media gateway, or session border controller. Used by HT813 for firewall or NAT penetration in different network environments. If symmetric NAT is detected, STUN will not work and only outbound proxy can correct the problem.
Backup Outbound Proxy	Configures the backup outbound proxy to be used when the "Outbound Proxy" registration fails. By default, this field is left empty.
Prefer Primary Outbound Proxy	If the user configures this option to "**Yes**", when registration expires, the device will re-register via primary outbound proxy. By default, this option is disabled.
SIP Transport	Selects transport protocol for SIP packets; UDP or TCP or TLS. Please make sure your SIP Server or network environment supports SIP over the selected transport method. Default is **UDP**.
NAT Traversal	Indicates type of NAT for each account. This parameter configures whether the NAT traversal mechanism is activated. Users could select the mechanism from No, Keep-alive, STUN, UPnP. Default setting is **No**.
SIP User ID	Defines user account information provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
Authenticate ID	Determines account authenticate ID provided by VoIP service provider (ITSP). Can be identical to or different from "SIP user ID".
Authenticate Password	Specifies account password provided by VoIP service provider (ITSP) to register to SIP servers.
Name	Chooses a name to be associated to user.
DNS Mode	Selects DNS mode to use for the client to look up server. One mode can be chosen. • **A Record:** resolves IP Address of target according to domain name.

	- **SRV:** DNS SRV resource records indicate how to find services for various protocols. - **NAPTR/SRV**: Naming Authority Pointer according to RFC 2915. Default is **A Record**.
DNS SRV use Registered IP	When this option is set to "Yes", when the HT is registered on second SRV and makes an outbound call, it will try the second SRV (registered IP) first. By default, this option is disabled and the DNS SRV will use first SRV instead of the registered IP.
Tel URI	Indicates E.164 number in "From" header by adding "User=Phone" parameter or using "Tel:" in SIP packets, if the HT813 has an assigned PSTN Number. - **Disabled:** Use "SIP User ID" information in the Request-Line and "From" header. - **User=Phone:** "User=Phone" parameter will be attached to the Request-Line and "From" header in the SIP request to indicate the E.164 number. If set to "Enable". - **Enabled:** "Tel:" will be used instead of "sip:" in the SIP request. Please consult your carrier before changing this parameter. Default is **Disabled**.
SIP Registration	Controls whether the HT813 needs to send REGISTER messages to the proxy server. Default setting is **Yes**.
Unregister on Reboot	Controls whether to clear SIP user's information by sending un-register request to the proxy server. The un-registration is performed by sending a REGISTER message with Contact set to * and Expires=0 parameters to the SIP server. This will unregister the SIP account under the concerned FXO page. Default is **No**.
Outgoing Call Without Registration	Enables the ability to place outgoing calls even if the account is not registered (if allowed by ITSP); device will not be able to receive incoming calls. Default is **No**.
Register Expiration	Refreshes registration periodically with specified SIP proxy (in minutes). Maximum interval is 65535 minutes (about 45 days). Default is **60** minutes (or 1 hour).
Reregister Before Expiration	Sends re-register request after specific time (in seconds) to renew registration before the previous registration expires.
SIP Registration Failure Retry Wait Time	Sends re-register request after specific time (in seconds) when registration process fails. Maximum interval is 3600 seconds (1 hour). Default is **20** seconds.

SIP Registration Failure Retry Wait Time upon 403 Forbidden	Sends re-register request after specific time (in seconds) when registration process fails with error 403 Forbidden. Maximum interval is 3600 seconds (1 hour). Default is **1220** seconds.
Enable SIP OPTIONS Keep Alive	Enables SIP OPTIONS to track account registration status so the phone adapter will send periodic OPTIONS message to server to track the connection status with the server. Default setting is **No**.
SIP OPTIONS Keep Alive Interval	Configures the time interval when the phone adapter sends OPTIONS message to SIP server. The default setting is 30 seconds, which means the phone adapter will send an OPTIONS message to the server every 30 seconds. The default range is **1-64800.**
SIP OPTIONS Keep Alive Max Lost	Defines the Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is **3**.
Layer 3 QoS	Defines Diff-Serv values for SIP and RTP. Defaults are: SIP DSCP: **26** RTP DSCP: **46**
Local SIP Port	Defines local port to use by the HT813 for listening and transmitting SIP packets. Default value for FXO port is **5062**.
Local RTP Port	Defines the local RTP-RTCP port pair the HT813 will listen and transmit. It is the HT813 RTP port for channel 0. The default value for FXS port is **5012.**
Use Random SIP Port	Controls whether to use configured or random SIP ports. This is usually necessary when multiple HT813 are behind the same NAT. Default is **No**.
Use Random RTP Port	Controls whether to use configured or random RTP ports. This is usually necessary when multiple HT813 are behind the same NAT. Default is **No**.
Enable RTCP	Allows users to enable RTCP. Default setting is **Yes**.
Remove OBP from Route Header	Removes outbound proxy info in "Route" header when sending SIP packets. Default is **No**.
Support SIP Instance ID	Includes "SIP Instance ID" attribute to "Contact" header in REGISTER request as defined in IETF SIP outbound draft. Default is **No**.
Validate Incoming SIP Message	Validates incoming messages. Default is **No**.
Check SIP User ID for Incoming	Checks SIP User ID in the Request URI of incoming INVITE; if it doesn't match the HT813 SIP User ID, the call will be rejected. Direct IP calling will also be disabled.

INVITE	Default is **No**.
Authenticate Incoming INVITE	Challenges the incoming INVITE for authentication with SIP 401 Unauthorized message. Default is **No**.
Authenticate server certificate domain	Configures whether to validate the domain certificate when download the firmware/config file. If it is set to "Yes", the phone will download the firmware/config file only from the legitimate server. The default setting is "**No**".
Authenticate server certificate chain	Configures whether to validate the server certificate when download the firmware/config file. If it is set to "Yes", the phone will download the firmware/config file only from the legitimate server. The default setting is "**No**".
Trusted CA Certificates	Uses the certificate for Authentication if "Check Domain Certificates" is set to "Yes" under "Account"→"SIP Settings".
Allow Incoming SIP Messages from SIP Proxy Only	Checks SIP address of the Request URI in the incoming SIP message; if it doesn't match the SIP server address of the account, the call will be rejected. Default is **No**.
Use Privacy Header	Determines if the "Privacy header" will be presented in the SIP INVITE message and if it includes the caller info in this header. If set to Default, it will add Privacy header unless special feature is **Telkom SA** or **CBCOM**. Default is **Default**.
Use P-Preferred-Identity Header	Specifies if the P-Preferred-Identity Header will be presented in the SIP INVITE message. If set to "default", the P-Preferred-Identity Header will be omitted in SIP INVITE message when **Telkom SA** or **CBCO** is active. If set to "Yes", the P-Preferred-Identity Header will always be presented. If set to "No", it will be omitted. Default setting is: **Default**.
Use P-Access-Network-Info Header	With this feature enabled, device will populate the WAN access node with IEEE802.11a, IEEE-802.11b in P-Access-Network-Info SIP header.
Use P-Emergency-Info Header	This feature support of IEEE-48-addr and IEEE-EUI-64 in SIP header for emergency calls.
SIP REGISTER Contact Header Uses	Specifies which address (LAN or WAN address) the device will detect to use it in SIP Register Contact Header. When set to **LAN**, Contact header will include local IP from ATA in REGISTER messages, while if set to **WAN**, host/port/contact will be updated from SIP 401/403/404/407 Via header "received"/"rport" parameters in REGISTER messages. Default is **LAN Address**.
Allow SIP Factory	This feature allows user to reset the devices directly through SIP Notify. Default is **No**.

Reset	
SIP T1 Timeout	Defines T1 timeout value. It is an estimate of the round-trip time between the client and server transactions. For example, the HT813 will attempt to send a request to a SIP server. The time it takes between sending out the request to the point of getting a response is the SIP T1 timer. If no response is received the timeout is increased to (2*T1) and then (4*T1). Request re-transmit retries would continue until a maximum amount of time defined by T2. Default is **0.5** seconds.
SIP T2 Interval	Identifies maximum retransmission interval for non-INVITE requests and INVITE responses. Retransmitting and doubling of T1 continues until it reaches T2 value. Default is **4** seconds.
SIP Timer D	Configure the SIP Timer D defined in RFC3261. 0 - 64 seconds. Default 0
DTMF Payload Type	Defines payload type for DTMF using RFC2833.
Preferred DTMF method (in order)	Sorts DTMF methods (in-audio, via RTP (RFC2833) or via SIP INFO) by priority.
Disable DTMF Negotiation	Uses above DTMF order without negotiation. Default is **No**.
Generate Continuous RFC2833 Events	When enabled the RFC2833 events are generated until key is released. Default is **No**.
Flash Digit Control	When it set to YES it allows the user to perform some call setting when both channels are used while pressing: • **"Flash + 1"** in order to hang up the current call and resume a call that was held. • **"Flash + 2"** in order to hold the current call and resume a call that was held. • **"Flash + 3"** in order to perform 3-way conference. • **"Flash + 4"** in order to perform attended transfer. **Note:** Please refer to the user guide for detailed steps to perform above operations.
Proxy-Require	Determines a SIP Extension to notify the SIP server that the HT813 is behind a NAT/Firewall.
Use NAT IP	Defines NAT IP address used in SIP/SDP messages. It should only be used if required by ITSP.

Use SIP User-Agent Header	Configures the SIP User-Agent Header.
SIP User-Agent	Configures SIP User-Agent. If not configured, device will use the default User Agent Header. The value range is 1024 to Maximum String Length. Default value is Null.
Do Not Escape '#' as %23 in SIP URI	Replaces # by %23 in some special situations. Default is **No**.
Disable Multiple m Line in SDP	Sends only one m line in SDP, regardless of how many m fields are in the incoming SDP. Default is **No**.
Ring Timeout	Stops ringing when incoming call if not answered within a specific period of time. Default is **60** seconds. When configure the Ring Timeout to 0, will have no ring timeout.
Early Dial	Sends an early INVITE each time a key is pressed when a user dials a number. Otherwise, only one INVITE is sent after full number is dialed (user presses **Dial Key** or after "no key entry timeout" expires). This option should be used only if there is a SIP proxy is configured and supporting 484 responses (Incomplete Address). Otherwise, the call will likely be rejected by the proxy (with a 404 Not Found error). Default is **No**. *This feature is NOT designed to work with and should NOT be enabled for direct IP-to-IP calling*.
Dial Plan Prefix	Adds specified prefix to dialed number.
Use # as Dial Key	Treats "#" as the "Send" (or "Dial") key. If set to "No", this "#" key can be included as part of the dialed number. Default is **Yes**.

Dial Plan	**Dial Plan Rules**:
	 1. Accept Digits: 1,2,3,4,5,6,7,8,9,0 , *, #, A,a,B,b,C,c,D,d
	 2. Grammar: **x** - any digit from 0-9;
	     a. **xx+** - at least 2 digits number;
	     b. **xx** – exactly 2 digits number;
	     c. **^** - exclude;
	     d. **.** – wildcard, matches one or more characters
	     e. **[3-5]** - any digit of 3, 4, or 5;
	     f. **[147]** - any digit 1, 4, or 7;
	     g. **<2=011>** - replace digit 2 with 011 when dialing
	     h. **< =1>** - add a leading 1 to all numbers dialed, vice versa will remove a 1 from

the number dialed

  i. **|** - or

- **Example 1:** {[369]11 | 1617xxxxxxx} –

  Allow 311, 611, 911, and any 10 digit numbers of leading digits 1617

- **Example 2:** {^1900x+ | <=1617>xxxxxxx} –

  Block any number with leading digits 1900 and add prefix 1617 for any dialed 7 digit numbers

- **Example 3:** {1xxx[2-9]xxxxxx | <2=011>x+} –

  Allow any length of number with leading digit 2 and 10 digit-numbers of leading digit 1 and leading exchange number between 2 and 9; If leading digit is 2, replace leading digit 2 with 011 before dialing.

3. Default: Outgoing - {x+}

Example of a simple dial plan used in a Home/Office in the US:

{ ^1900x. | <=1617>[2-9]xxxxxx | 1[2-9]xx[2-9]xxxxxx | 011[2-9]x. | [3469]11 }

Explanation of example rule (reading from left to right):

- **^1900x. -** prevents dialing any number started with 1900
- **<=1617>[2-9]xxxxxx** **-** allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically
- **1[2-9]xx[2-9]xxxxxx** **-** allows dialing to any US/Canada Number with 11 digits length
- **011[2-9]x. -** allows international calls starting with 011
- **[3469]11 -** allow dialing special and emergency numbers 311, 411, 611 and 911

Note: In some cases, user wishes to dial strings such as *123 to activate voice mail or other application provided by service provider. In this case * should be predefined inside dial plan feature. As an example **{ *x+ }** will allow to dial * followed by any length of numbers.

SUBSCRIBE for MWI	Sends SUBSCRIBE periodically (depends on "Register Expiration" parameter) for message waiting indication. Default is **No**.
Anonymous Call Rejection	Rejects incoming calls with anonymous caller ID with "486 Busy here" message. Default is **No**.
Special Feature	Selects Soft switch vendors' special requirements Example of vendors: Standard, Broadsoft, CBCOM, RNK, Huawei, China Mobile, ZTE IMS, PhonePower, TELKOM SA, Vonage, Metaswitch, CenturyLink, MTS. Default is **Standard**.
Session Expiration	Enables SIP sessions to be periodically "refreshed" via a SIP request (UPDATE, or re-INVITE). When the session interval expires, if there is no refresh via an UPDATE or re-

	INVITE message, the session will be terminated. Session Expiration is the time (in seconds) at which the session is considered timed out, if no successful session refresh transaction occurs beforehand. Default is **180** seconds.
Min-SE	Defines Minimum session expiration (in seconds). Default is **90** seconds.
Caller Request Timer	Uses session timer when making outbound calls if remote party supports it. Default is **No**.
Callee Request Timer	Uses session timer when receiving inbound calls with session timer request. Default is **No**.
Force Timer	Uses session timer even if the remote party does not support this feature. Selecting "No" will enable session timer only when the remote party supports it. Default is **No**. To turn off Session Timer, select "No" for Caller and Callee Request Timer, and Force Timer.
UAC Specify Refresher	Specifies which end will act as refresher for outgoing calls. • **UAC**: The handy tone acts as the refresher. • **UAS**: Callee or proxy server act as the refresher. Default is **Omit**.
UAS Specify Refresher	Specifies which end will act as refresher for incoming calls: • **UAS**: The handy tone acts as the refresher. • **UAC**: Callee or proxy server act as the refresher. Default is **Omit**.
Force INVITE	Always refresh with INVITE instead of UPDATE Default is **No**.
INVITE Ring-No-Answer Timeout (sec)	Between 5-300 seconds. Default 40 seconds.
Enable 100rel	Appends "100rel" attribute to the value of the required header of the initial signaling messages. Default is **No**.
Add Auth Header on Initial REGISTER	Adds "Authentication" header with blank "nonce" attribute in the initial SIP REGISTER request. Default is **No**.
Use First Matching Vocoder in 200OK SDP	Includes only the first matching vocoder in its 200OK response, otherwise it will include all matching vocoders in same order received in INVITE. Default is **No**.
Preferred Vocoder	Configures vocoders in a preference list (up to 7 preferred vocoders) that will be

	included with same order in SDP message. Vocoder types are G.711 A-/U-law, G.726-32, G.723, G.729, iLBC and OPUS.
Voice Frames per TX	Transmits a specific number of voice frames per packet. Default is **2**; increases to 10/20/32/64 for G711/G726/G723/other codecs respectively.
G723 Rate	Operates at specified encoding rate for G.723 vocoder. Available encoding rates are 6.3kbps or 5.3kbps. Default is **6.3kbps**.
iLBC Frame Size	Specifies iLBC packet frame size (20ms or 30ms). Default is **20ms**.
Disable OPUS Stereo in SDP	Disables OPUS stereo in SDP. Default is **No**.
iLBC Payload type	Determines payload type for iLBC. Valid range is between 96 and 127. Default is **97**.
OPUS Payload Type	Determines payload type for OPUS. Valid range is between 96 and 127. Default is **123**.
VAD	Allows detecting the absence of audio and conserves bandwidth by preventing the transmission of "silent packets" over the network. Default is **No**.
Symmetric RTP	Changes the destination to send RTP packets to the source IP address and port of the inbound RTP packet last received by the device. Default is **No**.
Fax Mode	Specifies the fax mode: T.38 (Auto Detect) FoIP by default, or Pass-Through (must use codec PCMU/PCMA)
Re-Invite after Fax Tone Detected	Allows the unit to send out the re-INVITE for T.38 or Fax Pass Through if a fax tone is detected. Default is **Enabled**
Jitter Buffer Type	Selects jitter buffer type (Fixed or Adaptive) based on network conditions.
Jitter Buffer Length	• **High** (initial 200ms, min 40ms, max 600ms) Note: not all vocoders can meet the high requirement. • **Medium** (initial 100ms, min 20ms, max 200ms). • **Low** (initial 50ms, min 10ms, max 100ms).
SRTP Mode	Selects SRTP mode to use ("Disabled", "Enabled but not forced", or "Enabled and forced"). Default is **Disabled.** It uses SDP Security Description to exchange key. Please refer to SDES: https://tools.ietf.org/html/rfc4568 SRTP: https://www.ietf.org/rfc/rfc3711.txt
Crypto Life Time	Adds crypto life time header to SRTP packets. Default is **Yes**.

Caller ID Scheme	Selects the caller ID scheme. Available options: • Bellcore/Telcordia • ETSI-FSK during ringing • ETSI-FSK prior to ringing with DTAS • ETSI-FSK prior to ringing with LR+DTAS • ETSI-FSK prior to ringing with RP • ETSI-DTMF during ringing • ETSI-DTMF prior to ringing with DTAS • ETSI-DTMF prior to ringing with LR+DTAS • ETSI-DTMF prior to ringing with RP • SIN 227 – BT • NTT JAPAN • DTMF Denmark prior to ringing with no DTAS no LR • DTMF Denmark prior to ringing with LR • DTMF Sweden/Finland prior to ringing with LR • DTMF Brazil • DTMF-FSK Brazil
DTMF Caller ID	Defines the start and stop tones as delimiters for the caller ID. **Start Tone** and **Stop Tone** can be set to "Default", "A", "B", "C", "D" or "#"
FSK Caller ID Minimum RX Level (dB)	An adjustable value for the Caller ID signal to help this device to recognize Caller ID from different networks. Range: -96 to -0dB. Default **-40**dB.
FSK Caller ID Seizure Bits	Range is from 0 to 800 bits. Default **70**.
FSK Caller ID Mark Bits	Range is from 0 to 800 bits. Default **40**.
Caller ID Transport Type	According to customer's choice CID information will be transferred from PSTN network to VoIP network using following rules: • **Relay via SIP From** - PSTN CID is in the SIP From field • **Relay via P-Asserted-Identity** - SIP From field uses the pre-configured account user Id. PSTN CID is in the P-Asserted-Identity field • **Relay via P-Preferred-Identity** - PSTN CID is in the P-Preferred-Identity field • **Send anonymous** - SIP From field uses "anonymous". PSTN CID is put in the P-Asserted-Identity field • **Disable** - PSTN CID will not be sent. SIP From field uses the pre-configured

	account user ID
Send Hook Flash to PSTN	If Yes, hook flash will be sent to PSTN upon receiving flash event from RFC2833 or SIP INFO.
Hook Flash Duration (ms)	The time period when the cradle is pressed (Hook Flash) to simulate a FLASH. Adjust this time value to prevent unwanted activation of the Flash/Hold and automatic phone ring-back.
Gain	Voice path volume adjustment. • RX is a gain level for signals transmitted by FXO (FXO-To-VoIP volume) • TX is a gain level for signals received by FXO (FXO-To-PSTN volume). Default = 0dB for both parameters. Loudest volume: +6dB; Lowest volume: -6dB. User can adjust volume of call on either end using the Rx Gain Level parameter and the TX Gain Level parameter located on the FXO Port Configuration page. These parameters affect call volume ONLY for calls placed to/from PSTN and VoIP networks. If call volume is too low when using VoIP extension, adjust volume using the Rx Gain Level parameter under the FXO Port Configuration page. If voice volume is too low at the other end (PSTN side), user may increase the far end volume using the TX Gain Level parameter under the FXO Port Configuration page.
Disable Line Echo Canceller (LEC)	Disables the LEC will per call base. Recommended for FAX/Data calls. Default is **No**.
Disable Network Echo Suppressor	Disables the NEC will per call base. Recommended for FAX/Data calls. Default is **No**.
Outgoing Call Duration Limit	Defines the call duration limit for the outgoing calls, Default is **0 (No limit)**.
FXO Termination	
Enable Current Disconnect	This value should be used in case the PSTN provider uses line power drop to indicate call completion to the end point. In this case the HT813 will search for a power drop for a preconfigured time frame to disconnect such calls from a VoIP extension. Default is **Yes**.
Current Disconnect Threshold (ms)	This is a preconfigured value of duration for a line power drop used by specific service providers. For example, for a configured value of 500ms the device will ignore any random voltage drops on the line if duration of such drop is less than 500ms and the call will NOT be considered as terminated. This is useful to prevent unnecessary call drops in some low quality PSTN lines. Default is **100** ms. Range from 50 to 800 ms.
Enable PSTN	If set to Yes, arrived Busy Tone is used as the disconnect signal.

Disconnect Tone Detection	
PSTN Disconnect Tone	In certain countries, the central office will send a special busy tone to indicate when a call is disconnected from the remote side. User can pre-configure this tone on the ATA. The user should know the frequency values and cadences of these tones. Here is an example for the syntax for a busy tone in the U.S.A: Syntax: **f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3;** Note: freq: 0 - 4000Hz; vol: -30 - 0dBm Default: Busy Tone - f1=480@-32, f2=620@-32, c=500/500; Note: Maximum supported cadences are 3
Enable Polarity Reversal	This should be set to Yes only if the FXO lines are subscribed to PR service from PSTN Service provider. It is merely a PR detect feature. Default is No. **Note:** If there is no PR service from provider on the FXO line, and this setting is configured to Yes, calls will not be successful.
AC Termination Model	You can select the AC termination by Country or by Impedance. Default is **Country-based**.
Country-Based	15 Countries are selectable in this version of the F/W. • USA • AUSTRIA • AUSTRALIA/NEW ZEALAND • BELGIUM • CHINA • FINLAND • FRANCE • GERMANY • GREECE • ITALY • JAPAN • NORWAY • SPAIN • SWEDEN • UK Default is USA
Impedance-Based	Select the Impedance used by the PSTN service provider. • 600R – 600 ohms • 600C – 600 ohms + 2.16uF • 900R – 900 ohms

	• 900C – 900 ohms + 2.16uF • COMPLEX1 – 220 ohms + (820 ohms || 115nF) • COMPLEX2 – 270 ohms + (750 ohms || 150nF) • COMPLEX3 – 370 ohms + (620 ohms || 310nF) • COMPLEX4 – 600R, 270 ohms + (750 ohms || 150nF) • COMPLEX5 – 320 ohms + (1050 ohms || 230nF) • COMPLEX6 – 350 ohms + (1000 ohms || 210nF) • COMPLEX7 – 200 ohms + (680 ohms || 100nF) • COMPLEX8 – 370 ohms + (820 ohms || 110nF) • COMPLEX9 – 275 ohms + (780 ohms || 115nF) • COMPLEX10 – 120 ohms + (820 ohms || 110nF) Default is 600R – 600 ohms
Number of Rings	The FXO port will ring the number of times configured in this field before sending the call to the VoIP side. The supported range is 1-50. Default is **4**.
PSTN Ring Thru FXS	If Yes, the phone connected to the FXS port will ring a configured amount of times (see above). If not, the phone connected to the FXS port will not ring.
PSTN Ring Thru Delay (sec)	If the PSTN Ring Thru Delay is set to Yes, all incoming PSTN calls through FXO will ring the phone connected to the FXS port, after this delay or after caller id is detected (whichever comes first).
PSTN Ring Timeout (sec)	Range is 2-10 seconds. Default is 6 seconds. Option is used to detect PSTN hang up when FXO port is not answered.
PSTN Idle Wait Timeout between Outgoing Calls	Used to customize timeout value between PSTN outgoing calls. Range is 0-10 seconds. Default is 4 seconds.
Channel Dialing	
DTMF Digit Length (ms)	Digit length and Dial Pause are port digit dialing configurations; FXO needs to dial out digits for VoIP to PSTN 1 stage calls, and unconditional call forward to PSTN, and route to PSTN. Digit Length is the play time for each digit. Note: In order to receive the caller ID information, the delay should be set to a value larger than the delay required to complete the PSTN caller ID delivery.
DTMF Dial Pause (ms)	Dial pause is the time between 2 digits for the same scenario as explained above.
First Digit Timeout	Used for PSTN to VoIP calls. PSTN users need to enter the FIRST digit within the first

(sec)	digit timeout period. Otherwise the call will be dropped.
Inter-Digit Timeout (sec)	When dialing from the PSTN to VoIP, subsequent digits must be input within the period of inter-digit timeout. Otherwise the dial plan thinks it is the end of the digit input.
Wait for Dial Tone	Wait for Dial tone is used for one stage VoIP to PSTN calls. If set to Yes, the device will first obtain a PSTN line and a dial tone from a central office. After obtaining the dial tone, the digits dialed will be sent to the central office.
Stage Method (1/2)	This configuration is applicable for VoIP to PSTN calls and indicates one or two stage dialing methods.
Min Delay Before Dial PSTN Number	The time to wait before HT813 initiates the call via PSTN line. Default 500ms, range is from 50 to 65000ms.

Important Settings

NAT Settings

If you plan to keep the Handy Tone within a private network behind a firewall, we recommend using STUN Server. The following three settings are useful in the STUN Server scenario:

1. STUN Server (under advanced settings webpage) enter a STUN server IP (or FQDN) that you may have or look up a free public STUN server on the internet and enter it on this field. If using public IP, keep this field blank.

2. Use random SIP/RTP ports (under advanced settings webpage), this setting depends on your network settings. Generally, if you have multiple IP devices under the same network, it should be set to Yes. If using a public IP address, set this parameter to No.

3. NAT traversal (under the FXS and FXO web page) Set this to Yes when gateway is behind firewall on a private network.

DTMF Methods

The HT813 supports the following DTMF mode:

- DTMF in-audio
- DTMF via RTP (RFC2833)
- DTMF via SIP INFO

Set priority of DTMF methods according to your preference. This setting should be based on your server DTMF setting.

Preferred Vocoder (Codec)

The HT813 supports following voice codecs. On FXS/FXO page, choose the order of your favorite codecs:

- PCMU/A (or G711μ/a)
- G729 A/B
- G723.1
- G726
- iLBC
- OPUS

Configuring HT813 Through Voice Prompts

As mentioned previously, The HT813 has a built-in voice prompt menu for simple device configuration. Please refer to "Understanding HT813 Interactive Voice Prompt Response Menu" for more information about IVR and how to access its menu.

- **DHCP MODE**

Select voice menu option 01 to enable HT813 to use DHCP.

- **STATIC IP MODE**

Select voice menu option 01 to enable HT813 to use STATIC IP mode, then use option 02, 03, 04, 05 to set up IP address, Subnet Mask, Gateway and DNS server respectively.

- **PPPOE MODE**

Select voice menu option 01 to allow the HT813 to enable the PPPoE mode. PPPoE Username and Password should be configured from web GUI.

- **FIRMWARE SERVER IP ADDRESS**

Select voice menu option 13 to configure the IP address of the firmware server.

- **CONFIGURATION SERVER IP ADDRESS**

Select voice menu option 14 to configure the IP address of the configuration server.

- **UPGRADE PROTOCOL**

Select the menu option 15 to choose firmware and configuration upgrade protocol between TFTP, FTP, FTPS, HTTP and HTTPS. Default is HTTPS.

- **FIRMWARE UPGRADE MODE**

Select voice menu option 17 to choose firmware upgrade mode among the following three options:
1) Always check, 2) check when pre/suffix changes, and 3) never upgrade.

- **WAN PORT WEB ACCESS**

Select voice menu option 12 to enable/disable web access from WAN port. Press 9 in this menu to toggle between enable / disable. Default is disabled.

Configuration through a Central Server

The HT813 can be automatically configured from a central provisioning system.

When HT813 boots up, it will send TFTP, FTP/FTPS or HTTP/HTTPS requests to download configuration files, "cfg000b82xxxxxx" and "cfg00082xxxxxx.xml", where "000b82xxxxxx" is the LAN MAC address of the HT813. If the download of "cfgxxxxxxxxxxxx.xml" is not successful, the provision program will issue request a generic configuration file "cfg.xml". Configuration file name should be in lower case letters. The configuration data can be downloaded via TFTP, FTP/FTPS or HTTP/HTTPS from the central server. A service provider or an enterprise with large deployment of HT813 can easily manage the configuration and service provisioning of individual devices remotely from a central server.

Grandstream provides a central provisioning system GAPS (Grandstream Automated Provisioning System) to support automated configuration of Grandstream devices. GAPS use enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual Grandstream device for firmware upgrade, remote reboot, etc. Grandstream provides GAPS service to VoIP service providers. Use GAPS for either simple redirection or with certain special provisioning settings. At boot-up, Grandstream devices by default point to Grandstream provisioning server GAPS, based on the unique MAC address of each device, GAPS provision the devices with redirection settings so that it will be redirected to customer's TFTP or HTTP/HTTPS server for further provisioning. Grandstream also provides configuration tools (Windows and Linux/Unix version) to facilitate the task of generating device configuration files.

The Grandstream configuration tools are free to end users. The configuration tools and configuration templates are available for download from http://www.grandstream.com/support/tools

Register a SIP Account

The HT813 supports 2 SIP accounts. Please refer to the following steps in order to register your accounts via web user interface

1. Access your HT813 web UI by entering its IP address in your favorite browser.

2. Enter your admin password (default: admin).

3. Press **Login** to access your settings.

4. Go to **FXS** (same steps for **FXO**) web pages and set the following:

   a. **Account Active** to **Yes.**

   b. **Primary SIP Server** field with your SIP server IP address or FQDN.

   c. **Failover SIP Server** with your Failover SIP Server IP address or FQDN. Leave empty if not available.

d. **Prefer Primary SIP Server** to **No** or **Yes** depending on your configuration. Set to **No** if no Failover SIP Server is defined. If "**Yes**", account will register to Primary SIP Server when failover registration expires.

e. **Outbound Proxy:** Set your Outbound Proxy IP Address or FQDN. Leave empty if not available.

f. **SIP User ID**: User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.

g. **Authenticate ID**: SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.

h. **Authenticate Password**: SIP service subscriber's account password to register to SIP server of ITSP. For security reasons, the password will field will be shown as empty.

i. **Name**: Any name to identify this specific user.

5. Press **Apply** at the bottom of the page to save your configuration.

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS FXS PORT FXO PORT	
Account Active:	○ No ● Yes
Primary SIP Server:	192.168.85.53 (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)
Allow DHCP Option 120 (override SIP server):	● No ○ Yes
SIP Transport:	● UDP ○ TCP ○ TLS (default is UDP)
SIP URI Scheme When Using TLS:	○ sip ● sips
Use Actual Ephemeral Port in Contact with TCP/TLS:	● No ○ Yes
NAT Traversal:	● No ○ Keep-Alive ○ STUN ○ UPnP
SIP User ID:	6017 (the user part of an SIP address)
Authenticate ID:	6017 (can be identical to or different from SIP User ID)
Authenticate Password:	(purposely not displayed for security protection)
Name:	(optional, e.g., John Doe)

Figure 9: FXS Port Settings

After applying your configuration, your account will register to your SIP Server, you can verify if it has been correctly registered with your SIP server from your HT813 web interface under **Status → Port Status → Registration** (If it displays **Registered**, it means that your account is fully registered, otherwise it will display **Not Registered** so in this case you must double check the settings or contact your provider).

Figure 10: Accounts Status

Rebooting HT813 from Remote

Press "Reboot" button at the bottom of the configuration menu to reboot the ATA remotely. The web browser will then display a message window to confirm that reboot is underway. Wait 30 seconds to log in again.

HT813 Administration Guide
Version 1.0.5.2

Call Features

The HT813 supports all the traditional and advanced telephony features.

Table 11: HT813 Call Features

Key	Call features
*02	**Forcing a Codec** (per call) *027110 (PCMU), *027111 (PCMA), *02723 (G723), *02729 (G729), *027201 (iLBC).
*03	**Disable LEC** (per call) Dial "*03" +" number". **No** dial tone is played in the middle.
*16	**Enable SRTP**
*17	**Disable SRTP**
*30	**Block Caller ID** (for all subsequent calls)
*31	**Send Caller ID** (for all subsequent calls)
*47	**Direct IP Calling**. Dial "*47" + "IP address". **No** dial tone is played in the middle.
*50	**Disable Call Waiting** (for all subsequent calls)
*51	**Enable Call Waiting** (for all subsequent calls)
*67	**Block Caller ID (per call)**. Dial "*67" +" number". **No** dial tone is played in the middle.
*82	**Send Caller ID (per call)**. Dial "*82" +" number". **No** dial tone is played in the middle.
*69	**Call Return Service:** Dial *69 and the phone will dial the last incoming phone number received.
*70	**Disable Call Waiting (per call).** Dial "*70" +" number". **No** dial tone is played in the middle.
*71	**Enable Call Waiting (per call).** Dial "*71" +" number". **No** dial tone is played in the middle
*72	**Unconditional Call Forward:** Dial "*72" and then the forwarding number followed by "#". Wait for dial tone and hang up. (dial tone indicates successful forward)
*73	**Cancel Unconditional Call Forward**. To cancel "Unconditional Call Forward", dial "*73", wait for dial tone, then hang up.
*74	**Enable Paging Call**: Dial "*74" and then the destination phone number you want to page.
*78	**Enable Do Not Disturb (DND):** When enabled all incoming calls are rejected.
*79	**Disable Do Not Disturb (DND):** When disabled, incoming calls are accepted.
*87	**Blind Transfer**

*90	**Busy Call Forward:** Dial "*90" and then the forwarding number followed by "#". Wait for dial tone then hang up.
*91	**Cancel Busy Call Forward**. To cancel "Busy Call Forward", dial "*91", wait for dial tone, then hang up.
*92	**Delayed Call Forward.** Dial "*92" and then the forwarding number followed by "#". Wait for dial tone then hang up.
*93	**Cancel Delayed Call Forward.** To cancel Delayed Call Forward, dial "*93", wait for dial tone, then hang up
Flash/ Hook	Toggles between active call and incoming call (call waiting tone). If not in conversation, flash/hook will switch to a new channel for a new call.
#	Pressing pound sign will serve as Re-Dial key.

UPGRADING AND PROVISIONING

The HT813 can be upgraded via TFTP/FTP/FTPS/HTTP/HTTPS by configuring the URL/IP Address for the TFTPFTP/FTPS/HTTP/HTTPS server and selecting a download method. Configure a valid URL for TFTP or FTP/FTPS or HTTP/HTTPS (default is HTTPS); the server name can be FQDN or IP address.
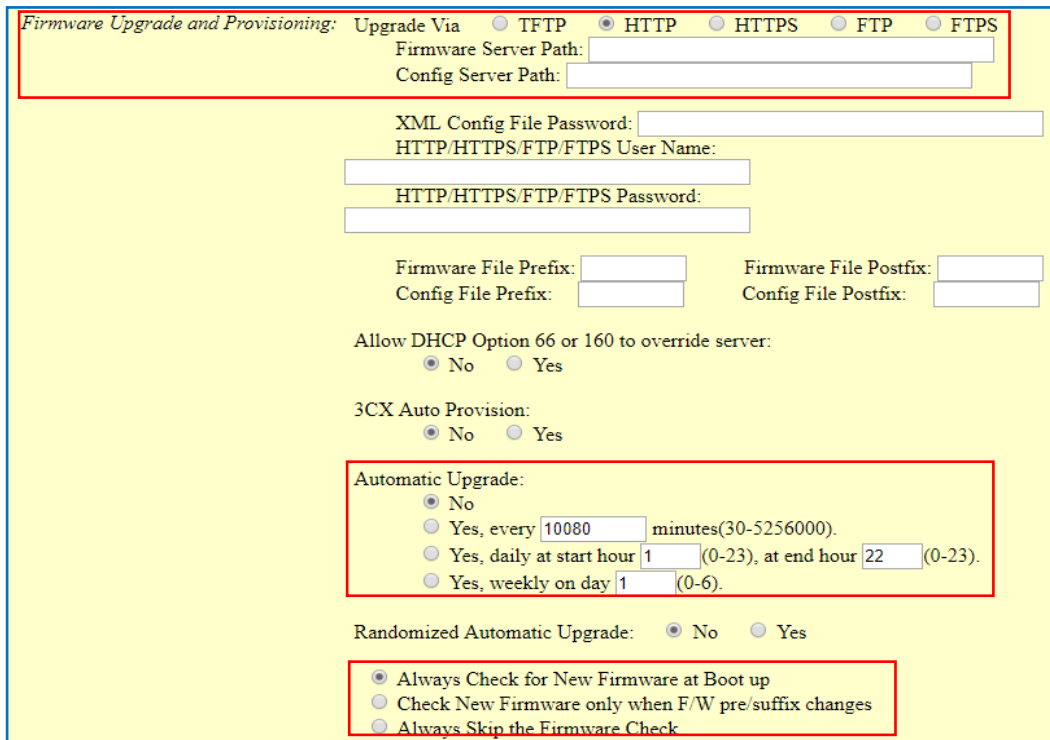
Examples of valid URLs:

firmware.grandstream.com or fw.ipvideotalk.com/gs

Firmware Upgrade procedure

Please follow below steps in order to upgrade the firmware version of your HT813:

1. Access your HT813 UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: admin).
3. Press **Login** to access your settings.
4. Go to **Advanced Settings → Firmware Upgrade and Provisioning** page and enter the IP address or the FQDN for the upgrade server in "**Firmware Server Path**" field and choose to upgrade via **TFTP** or **HTTP/HTTPS or FTP/FTPS**.
5. Make sure to check "**Always Check for New Firmware**".
6. Update the change by clicking the "Apply" button at the bottom of the page. Then "**Reboot**" or power cycle the HT813 to update the new firmware.

Figure 11: Firmware Upgrade Page

Upgrading via Local Directory

1. Download the firmware file from Grandstream web site
2. Unzip it and copy the file in to a folder in your PC
3. From the HT813 web interface (Advanced Settings page) you can browse your hard drive and select the folder you previously saved the file (HT8xfw.bin)
4. Click "Upload Firmware" and wait few minutes until the new program is loaded.

Note: Always check the status page to see that the program version has changed.

Upgrading via Local TFTP/HTTP Servers

For users that would like to use remote upgrading without a local TFTP/FTP/HTTP server, Grandstream offers a NAT-friendly HTTP server. This enables users to download the latest software upgrades for their devices via this server. Please refer to the webpage:
http://www.grandstream.com/support/firmware

Alternatively, users can download a free TFTP or HTTP server and conduct a local firmware upgrade. A free window version TFTP server is available for download from:
http://www.solarwinds.com/products/freetools/free_tftp_server.aspx
http://tftpd32.jounin.net/.

Instructions for local firmware upgrade via TFTP:
1. Unzip the firmware files and put all of them in the root directory of the TFTP server.
2. Connect the PC running the TFTP server and the phone to the same LAN segment.
3. Launch the TFTP server and go to the File menu->Configure->Security to change the TFTP server's default setting from "**Receive Only**" to "**Transmit Only**" for the firmware upgrade.
4. Start the TFTP server and configure the TFTP server in the phone's web configuration interface.
5. Configure the Firmware Server Path to the IP address of the PC.
6. Save and Apply the changes and reboot the HT813.

End users can also choose to download a free HTTP server from http://httpd.apache.org/ or use Microsoft IIS web server.

Firmware and Configuration File Prefix and Postfix

Firmware Prefix and Postfix allows device to download the firmware name with the matching Prefix and Postfix. This makes it the possible to store all of the firmware with different version in one single directory. Similarly, Config File Prefix and Postfix allows device to download the configuration file with the matching Prefix and Postfix. Thus, multiple configuration files for the same device can be stored in one directory. In addition, when the field "Check New Firmware only when F/W pre/suffix changes" is set to "Yes", the device will only issue firmware upgrade request if there are changes in the firmware Prefix or Postfix.

Managing Firmware and Configuration File Download

When "Automatic Upgrade" is set "**Yes, every**" the auto check will be done in the minute specified in this field. If set to "**daily at hour (0-23)**", Service Provider can use P193 (Auto Check Interval) to have the devices do a daily check at the hour set in this field with either Firmware Server or Config Server. If set to "**weekly on day (0-6)**" the auto check will be done on the day specified in this field. This allows the device to periodically check if there are any new changes need to be taken on a scheduled time. By defining different intervals in P193 for different devices, Server Provider can spread the Firmware or Configuration File download in minutes to reduce the Firmware or Provisioning Server load at any given time

Configuration File Download

Grandstream SIP Devices can be configured via the Web Interface as well as via a Configuration File (binary or XML) through TFTP, FTP/FTPS or HTTP/HTTPS. The **Config Server Path** is the TFTP or HTTP/HTTPS server path for the configuration file. It needs to be set to a valid URL, either in FQDN or IP address format. The **Config Server Path** can be the same or different from the **Firmware Server Path**.
A configuration parameter is associated with each particular field in the web configuration page. A parameter consists of a Capital letter P and 2 to 3 (Could be extended to 4 in the future) digit numeric numbers. i.e., P2 is associated with the "New Password" in the Web GUI->Maintenance->Web/SSH Access page->Admin Password. For a detailed parameter list, please refer to the corresponding firmware release configuration template.

When the HT813 boots up or reboots, it will send a request to download a file named "cfgxxxxxxxxxxxx" followed by a configuration XML file named "cfgxxxxxxxxxxxx.xml", where "xxxxxxxxxxxx" is the MAC address of the phone, i.e., "cfg000b820102ab" and "cfg000b820102ab.xml". If the download of "cfgxxxxxxxxxxxx.xml" file is not successful, the provision program will download a generic cfg.xml file. The configuration file name should be in lower case letters.

For more details on XML provisioning, please refer to:
http://www.grandstream.com/sites/default/files/Resources/gs_provisioning_guide.pdf

Notes:

1. Only XML or binary config file formats are accepted.
2. The MAC header in XML config file should be the device MAC or needs to be removed completely.

Figure 12: XML Config File - MAC Header

RESTORE FACTORY DEFAULT SETTINGS

⚠️ **Warning:**

Restoring the Factory Default Settings will delete all configuration information on the phone. Please backup or print all the settings before you restore to the factory default settings. Grandstream is not responsible for restoring lost parameters and cannot connect your device to your VoIP service provider.

There are three (3) methods for resetting your unit:

Using the Reset Button

To reset default factory settings using the reset button please follow the steps above:

1. Unplug the Ethernet cable.

2. Locate the reset hole on the back panel of your HT813.

3. Insert a pin in this hole, and press for about 7 seconds.

4. Take out the pin. All unit settings are restored to factory settings

Using the IVR Command

Reset default factory settings using the IVR prompt:

1. Dial "***" for voice prompt.

2. Enter "99" and wait for "reset" voice prompt.

3. Enter the encoded MAC address (Look below on how to encode MAC address).

4. Wait 15 seconds and device will automatically reboot and restore factory settings.

Encode the MAC Address

1. Locate the MAC address of the device. It is the 12-digit HEX number on the bottom of the unit.

2. Key in the MAC address. Use the following mapping:

<div align="center">

Table 12: MAC Address Key Mapping

</div>

Key	Mapping
0-9	0-9
A	22 (press the "2" key twice, "A" will show on the LCD)
B	222

C	2222
D	33 (press the "3" key twice, "D" will show on the LCD)
E	333
F	3333

For example: if the MAC address is 000b8200e395, it should be keyed in as "0002228200333395"

Reset from Web Interface (Reset Type)

1. Access your HT813 UI by entering its IP address in your favorite browser.

2. Enter your admin password (default: admin).

3. Press **Login** to access your settings.

4. Go to **Basic Settings** → **Reset Type**

5. Press **Reset** button (after selecting the reset type).

• **Full Reset:** This will make a full reset

• **ISP Data:** This will reset only the basic settings, like IP mode, PPPoE and Web port

• **VoIP Data Reset:** This will reset only the data related with a service provider like SIP server, sip user ID, provisioning and others.

Note:

- Factory Reset will be disabled if the "Lock keypad update" is set to "Yes".
- If the HT813 was previously locked by your local service provider, pressing the RESET button will only restart the unit. The device will not return to factory default settings.

EXPERIENCING THE HT813

Please visit our website: http://www.grandstream.com to receive the most up- to-date updates on firmware releases, additional features, FAQs, documentation and news on new products.

We encourage you to browse our product related documentation, FAQs and User and Developer Forum for answers to your general questions.   If you has purchased our products through a Grandstream Certified Partner or Reseller, please contact them directly for immediate support.

Our technical support staff is trained and ready to answer all of your questions. Contact a technical support member or submit a trouble ticket online to receive in-depth support.

Thank you again for purchasing Grandstream analogue telephone adapter, it will be sure to bring convenience to both your business and personal life.