Grandstream Networks, Inc.

HT812/HT814/HT818

Analog Telephone Adaptors

Administration Guide

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The latest electronic version of this user manual is available for download here:
http://www.grandstream.com/support

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 User Manual, 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.

GNU GPL INFORMATION

The firmware for the HT81X 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.

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http://www.grandstream.com/support/faq/gnu-general-public-license/gnu-gpl-information-download

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DOCUMENT PURPOSE

This document describes the basic concept and tasks necessary to use and configure your HT81X. It covers also the topic of connecting and configuring the HT81X, making basic operations and the call features. Please visit http://www.grandstream.com/support to download the latest "HT81X User Guide".

This guide covers following topics:

- Product overview
- Getting started
- Configuration guide
- Upgrade and provisioning
- Restore factory default settings

CHANGE LOG

This section documents significant changes from previous versions of admin guide for HT81X. Only major new features or major document updates are listed here. Minor updates for corrections or editing are not documented here.

Firmware Version 1.0.0.5 (Only for HT818)

- This is the initial version for HT818.

Firmware Version 1.0.5.5

- Added feature “Internet Protocol” to choose from “IPv4 Only”, “IPv6 Only”, “Both, prefer IPv4”, “Both, prefer IPv6”. [Internet Protocol]
- Added feature “IPv6 Address” to configure IPv6 Address. [IPv6 Address]

Firmware Version 1.0.3.7

- Added option “Use Actual Ephemeral Port in Contact with TCP/TLS” to force device to use actual ephemeral port. [Use Actual Ephemeral Port in Contact with TCP/TLS]
- Added option “SIP URI Scheme When Using TLS” to choose between ‘SIP’ and ‘SIPS’. [SIP URI Scheme When Using TLS]
- Added Option “Backup Outbound Proxy” to use backup Outbound Proxy if Outbound Proxy registration expires. [Backup Outbound Proxy]
- Added option “Prefer Primary Outbound Proxy” to enable registration through primary outbound proxy if registration expires. [Prefer Primary Outbound Proxy]
- Added option “Enable RTCP” to enable RTCP function through Web UI. [Enable RTCP]
- Added option “Hold Target Before Refer” to enable device to hold before being referred. [Hold Target Before Refer]
- Added Option “Enable Session Timer” to disable session timer. [Enable Session Timer]
- Added feature “Conference URI” to support Conference URI. [Conference URI]
- Added feature “White List for WAN Side” for remote management. [White List for WAN Side]
- Added feature “Black List for WAN Side” for remote management. [Black List for WAN Side]
- Added option “Web Access Mode” to choose between “HTTPS” and “HTTP” to access device Web UI. [Web Access Mode]
- Added feature “HTTPS Web Port” to set HTTPS web port instead of using default HTTPS port. [HTTPS Web Port]
- Added feature “SSH Port” to self-configure SSH port. [SSH Port]
- Added SNMP related features. [Enable SNMP] [SNMP Trap Community] [SNMP Trap Interval] [SNMP Trap IP Address] [SNMP Trap Port] [SNMP Trap Version]

Firmware Version 1.0.3.2

- Added option “DNS SRV use Registered IP” to force DNS SRV to use registered IP instead of use first SRV. [DNS SRV use Registered IP]
- Changed default NTP server from us.pool.ntp to pool.ntp.org. [NTP Server]

Firmware Version 1.0.2.7

- No major changes

Firmware Version 1.0.2.5

- Changed OPUS Payload Type default value to 123 to match other GS products. [OPUS Payload Type]

Firmware Version 1.0.2.3

- Added network check mechanism to enable or disable WAN port web access.
- Added a re-enter box to confirm change user and admin password on web GUI to avoid typo or mistakes. [Confirm End User Password] [Confirm Admin Password]

Firmware Version 1.0.2.1

- This is the initial version for HT812 and HT814.

GUI INTERFACE EXAMPLES

http://www.grandstream.com/sites/default/files/Resources/HT81x_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 Profile Page
6. Screenshots of FXS Ports Page

WELCOME

The HT81X analog telephone adapters provide transparent connectivity for analog phones and faxes to the world of internet voice. Connecting to any analog phone, fax or PBX, the HT81X is an effective and flexible solution for accessing internet-based telephone services and corporate intranet systems across established LAN and internet connections. This Grandstream Handy Tones are a new addition to the popular Handy Tone ATA product family. This manual will help you to learn how to operate and manage your HT81X analog telephone adaptors and make the best use of their many upgraded features including simple and quick installation, 3- way conferencing, direct IP-IP Calling, and new provisioning support among other features. The HT81X are very easy to manage and configure, and they are specifically designed to be an easy to use and affordable VoIP solution for both the residential user and the teleworker.

PRODUCT OVERVIEW

The HT81X are 2/4/8 ports analog telephone adapters (ATA) that allow users to create a high-quality and manageable IP telephony solution for residential and office environments. Their ultra-compact size, voice quality, advanced VoIP functionality, security protection and auto provisioning options enables users to take advantage of VoIP on analog phones and enables service providers to offer high quality IP service. The HT81X are an ideal ATAs for individual use and for large scale commercial IP voice deployments since they permit small and medium businesses to create integrated IP and PSTN telephony systems that efficiently manage communication costs. HT81X's inclusion of an integrated NAT router and dual 10/100/1000Mbps Ethernet WAN and LAN ports enables a shared broadband connection between multiple Ethernet devices as well as the extension of VoIP services to analog phones.

Feature Highlights

The following table contains the major features of the HT81X:

Table 1: HT81X Features at a Glance

HT812 / HT814 / HT818 	• Support dual 10/100/1000Mbps Ethernet port, 2 SIP profiles through 2 FXS ports for HT812, 4 FXS port for HT814 and 8 FXS port for HT818 • Support 3-way voice conferencing. • Support wide range of caller ID formats. • Support advanced telephony features, including call transfer, call forward, call-waiting, do not disturb, message waiting indication, multi-language prompts, flexible dial plan and more. • Support 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. • 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. • Strong AES encryption with security certificate per unit • GR-909 Line Testing Functionalities. • Exceptional voice quality with wide-band HD codec

HT81X 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 HT81X.

Table 2: HT81X Technical Specifications

Interfaces	
Telephone Interfaces	Two (2) RJ11 FXS ports for HT812. Four (4) RJ11 FXS ports for HT814. Eight (8) RJ11 FXS ports for HT818
Network Interface	Two (2) 10/100/1000 Mbps Ethernet port (RJ45).
LED Indicators	POWER, LAN, WAN, PHONE1 and PHONE2 for HT812. POWER, LAN, WAN, PHONE1, PHONE2, PHONE3 and PHONE4 for HT814. POWER, LAN, WAN, PHONE1, PHONE2, PHONE3, PHONE4, PHONE 5, PHONE 6, PHONE 7 and PHONE 8 for HT818.
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 24AWG line for HT812. - 2 REN, up to 1km on 24AWG line for HT814 and HT818.
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, ARP/RARP, ICMP, DNS, DHCP, NTP, TFTP, SSH, 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, TFTP, TR-069, secure and automated provisioning using TR069, syslog.
Security	
Media	SRTP.
Control	TLS/SIPS/HTTPS.
Management	Syslog support, SSH, remote management using web browser.
Physical	
Universal Power Supply	Input: 100-240VAC, 50-60Hz Output: 12V/0.5A for HT812. Output: 12V/1A for HT814.

	Output: 12V/1.5A for HT818.
Environmental	Operational: 32º – 104ºF or 0º – 40ºC. Storage: 14º – 140ºF or -10º – 60ºC. Humidity: 10 – 90% Non-condensing.
Dimensions and Weight	Dimension : - 28.5 x 130 x 90 mm (H x W x D) for the HT812/HT814. - 36 x 120 x 180 mm (H x W x D) for the HT818. Weight: - 353.33g for the HT812. - 423.5g for the HT814. - 356g for the HT818.
Compliance	
Compliance	FCC/CE/RCM.

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 HT81X.

Equipment Packaging

The HT81X ATAs packages contain:

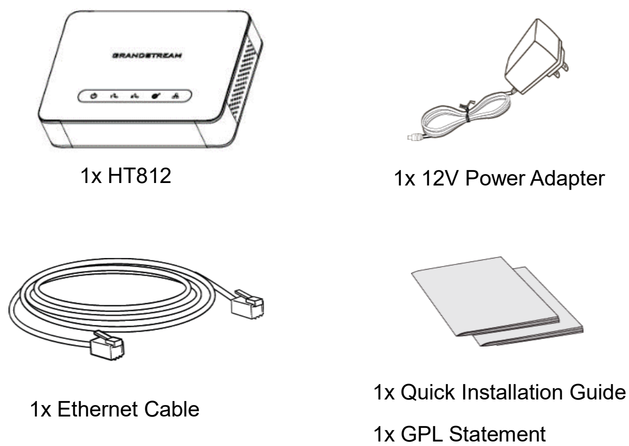


Figure 1: HT812 Package Contents

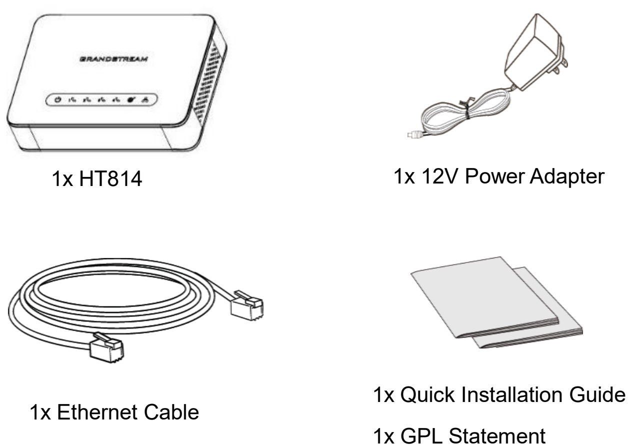
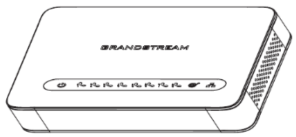
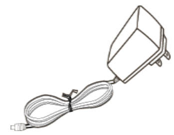


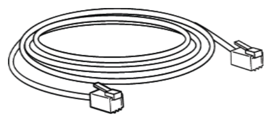
Figure 2: HT814 Package Contents



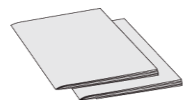
1x HT818



1x 12V Power Adapter



1x Ethernet Cable



1x Quick Installation Guide

1x GPL Statement

Figure 3: HT818 Package Contents

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

HT81X Ports Description

The following figure describes the different ports on the back panel of the HT81X.

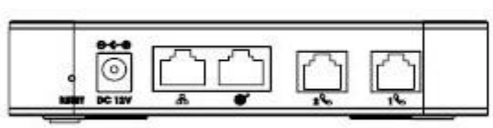


Figure 4: HT812 Back Panel

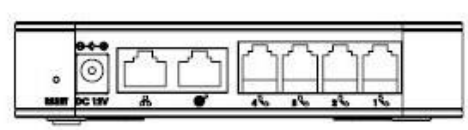


Figure 5: HT814 Back Panel

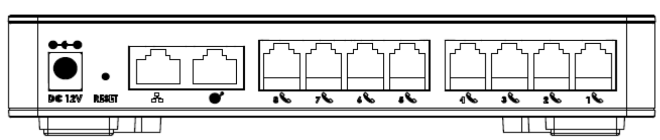


Figure 6: HT818 Back Panel

Table 3: Definition of the HT81X Connectors

Phone 1 & 2 (HT812) **Phone 1- 4 (HT814)** **Phone 1- 8 (HT818)**	Connects the analog phones / fax machines to the phone adapter using an RJ-11 telephone cable.
WAN	Connects the phone adapter to your router, switch or modem using an Ethernet RJ45 network cable.

LAN	Connects the phone adapter to your PC or switch using an Ethernet RJ45 network cable.
DC Power	Connects the phone adapter to PSU (12V – 0.5A for HT812) , (12V - 1A for HT814) and (12V – 1.5A for HT818).
Reset	Factory reset button. Press for 7 seconds to reset factory default settings.

Connecting HT81X

The HT81X are designed for easy configuration and easy installation, to connect your HT81X, please follow the steps below:

Scenario 1: Connecting the HT81X using WAN Port

When connecting HT81X using the WAN port, they will act as simple DHCP Client.

1. Insert a standard RJ11 telephone cable into the phone ports and connect the other end of the telephone cable to a standard touch-tone analog telephone.
2. Connect the WAN port of the HT81X to a router, switch or modem using an Ethernet cable.
3. Insert the power adapter into the HT81X and connect it to a wall outlet and make sure to respect the technical specifications of the power adapter used.
4. Power, WAN and Phone LEDs will be solidly lit when the HT81X is ready for use.

Scenario 2: Connecting the HT81X using LAN Port

When connecting the HT81X using the LAN port, they will act as a router and DHCP serving addresses, the devices connected with HT81X LAN will pull DHCP addresses from your HT81X.

1. Insert a standard RJ11 telephone cable into the phone ports 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 HT81X using an Ethernet Cable.
3. Insert the power adapter into the HT81X and connect it to a wall outlet and make sure to respect the technical specifications of the power adapter used.
4. Power, LAN and Phone LEDs will be solidly lit when the HT81X is ready for use.

Note: Please make sure to enable NAT Router under Web GUI → Basic Settings →Device Mode.

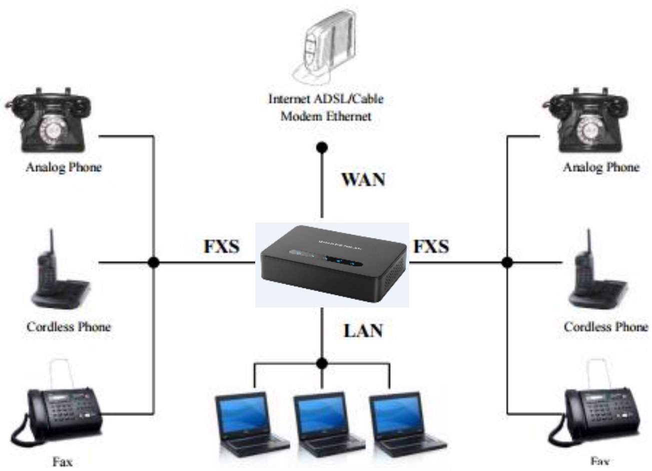


Figure 7: Connecting the HT81X

HT81X LEDs Pattern

There are four (4) LED types that help you manage the status of your HT81X.

Figure 8: HT81X LEDs Pattern

Table 4: HT81X LEDs Pattern Description

LED Lights	Status
Power LED	The Power LED lights up when the HT81X are powered on and it flashes when the HT81X is booting up
WAN LED	The WAN LED lights up when the HT81X are connected to your network through the WAN port.

LAN LED	The LAN LED lights up when the HT81X are connected to your network through the LAN port.
Phone LED 1&2 (HT812) **Phone LED 1- 4 (HT814)** **Phone LED 1- 8 (HT818)**	The phone LEDs indicate status of the respective FXS 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

CONFIGURATION GUIDE

The HT81X can be configured via one of two ways:

- The IVR voice prompt menu.
- The Web GUI embedded on the HT81X using PC's web browser.

Obtain HT81X IP Address via Connected Analogue Phone

HT81X are by default configured to obtain the IP address from DHCP server where the unit is located. In order to know which IP address is assigned to your HT81X, 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 phone ports (FXS) of your HT81X.
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 HT81X Interactive Voice Prompt Response Menu

The HT81X have a built-in voice prompt menu for simple device configuration which lists actions, commands, menu choices, and descriptions. The IVR menu works with any phone connected to the HT81X. 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.
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 HT81X 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 **enable** / **disable**. Default is disabled.
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 / 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 Setting** 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 HT81X embedded Web server responds to HTTP GET/POST requests. Embedded HTML pages allow a user to configure the HT81X 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**

For the initial setup, the Web access is by default enabled when the device is using private IP and disabled when using public IP, and you cannot access the Web UI of your HT81X until it's enabled, the following steps will show you how to enable it via IVR.

1. Power your HT81X using PSU with the right specifications.
2. Connect your analog phone to phone ports (FXS) of your HT81X.
3. Press *** (press the star key three times) to access the IVR menu and wait until you hear "Enter the menu option ".
4. Press 12, the IVR menu will announce that the web access is disabled, press 9 to enable it.
5. Reboot your HT81X to apply the new settings.

Please refer to steps below if your HT81X is connected via WAN port:

1. You may check your HT81X IP address using the IVR on the connected phone.

   Please see Obtain the HT81X IP address via the connected analogue phone

2. Open the web browser on your computer.
3. Enter the HT81X'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 HT81X. This can be easily done by connecting the computer to the same hub or switch as the HT81X.

- **Via LAN port**

Please refer to steps below if your HT81X is connected via LAN port:

1. Power your HT81X using PSU with the right specifications.
2. Connect your computer or switch directly to your HT81X 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 HT81X.

Web UI Access Level Management

There are two 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

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 HT81X 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.

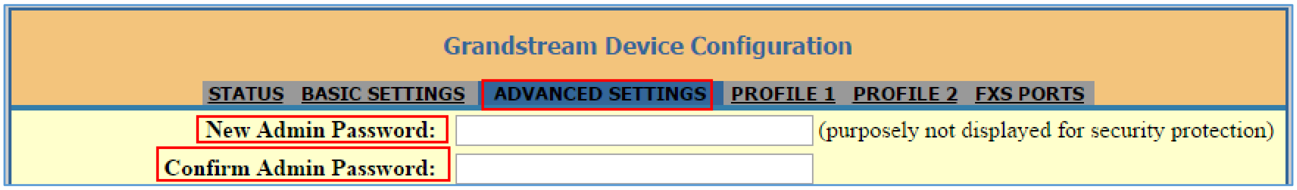


Figure 9: Admin Level Password

Changing User Level Password

1. Access your HT81X 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 **e**nter 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.

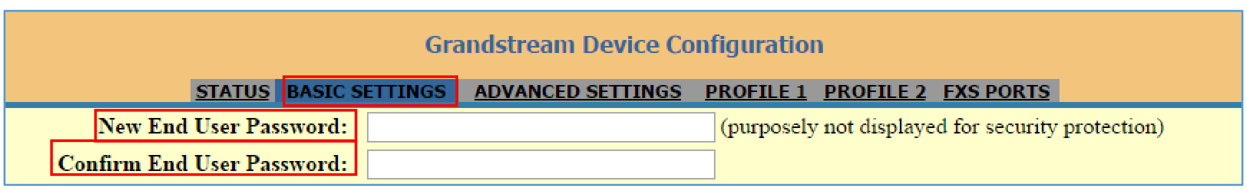


Figure 10: User Level Password

Changing HTTP Web Port

1. Access your HT81X 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 Web Port.**
5. Make sure that the **Web Access Mode** is set to **HTTP**.
6. Change the current port to your desired/new HTTP port. Ports accepted are in range [1-65535].
7. Press **Apply** at the bottom of the page to save your new settings

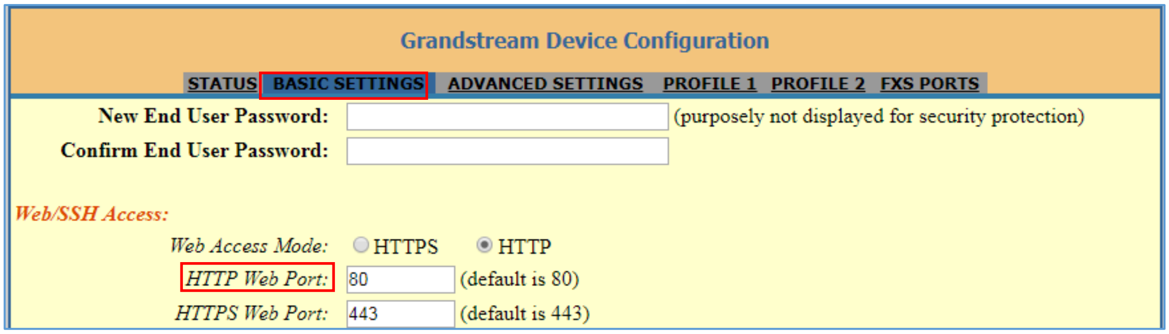


Figure 11: Web HTTP Port

Web Configuration Pages Definitions

This section describes the options in the HT81X Web UI. As mentioned, you can log in as an administrator or an end user.

- **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.
- **Profile (1,2):** Configures the SIP Server, SIP Registration, NAT settings, call features, ring tones.
- **FXS Ports:** Configures SIP accounts settings, Off hook Auto-dial.

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 **HT812** , **HT814** or **HT818**.
Hardware Version	Displays the hardware revision information and the part number.

Software version	• **Program:** Specifies Program version. Current is 1.0.5.5. This is the main firmware release number, which is always used for identifying the software system of the HT81X. • **Bootloader:** Specifies Boot version. Current is 1.0.5.5 • **Core:** Specifies Core version. Current is 1.0.5.1 • **Base:** Specifies Base version. Current is 1.0.5.5 • **CPE:** Specifies CPE version. CPE version is displayed only when HT81X 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's configured.
Port Status	Displays relevant information regarding the FXS ports about their registration, current status and their appropriate User ID.
Port Options	Displays relevant information regarding the FXS 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.
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. By default, "HTTP" is selected.
HTTP Web Port	Customizes HTTP port used to access the HT81X web UI. Default is **80**.

HTTPS Web Port	Customizes HTTPS port used to access the HT81X 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**.
WAN Side Web/SSH Access	Enables / Disables the Web and SSH access through the WAN port. The available options are the following the the default setting is **Auto** : • **No:** WAN side access for the Web GUI and SSH is disabled. • **Yes:** WAN side access for the Web GUI and SSH is enabled. • **Auto:** WAN side access allowed for private IP; rejected for public IP.
White List for WAN Side	Allows users to configure the white List for WAN Side to be used for remote management.
Black List for WAN Side	Allows users to configure the black List for WAN Side to ban WAN side web access.
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 phone for the changes to take effect.
IPv4 Address	Allows users to configure the appropriate network settings on the HT81x to obtain IPv4 address. Users could select "DHCP", "Static IP" or "PPPoE". 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 HT801/HT802 acquires its IP address from the first DHCP server it discovers from the LAN it is connected. • **Use PPPoE:** set the PPPoE account settings. If selected, HT801/HT802 attempt to establish a PPPoE session if any of the PPPoE fields is set. • **Static IP mode:** 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 HT81x 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 HT801/HT802 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).
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**.
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.
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.
Device Mode	Controls whether the device is working in NAT router mode or Bridge mode. Save the setting and reboot prior to configuring the HT81X.

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 HT81X acts as an UPnP gateway for your UPnP enabled applications. **UPnP = "Universal Plug and Play"**
Reply to ICMP on WAN Port	Default is No. When set to "Yes", the HT81X responds to the PING command from other computers, but is also made vulnerable to DOS attacks.
Cloned WAN MAC Address	This allows the user to change/set a specific MAC address on the WAN interface. Note: 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 HT81X 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 HT81X 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.
Reset Type	Gives the administrator the option to restore default configuration on the HT81X. 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.

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.
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	Selects firmware upgrade/provisioning method: TFTP, HTTP or 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 HT81X. 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 User Name	Enters user name to authenticate with HTTP/HTTPS server.
HTTP/HTTPS Password	Enters password to authenticate with HTTP/HTTPS 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 to Override Server	Obtains configuration and upgrade server's information using options 66 from DHCP server. **Note:** If DHCP Option 66 is enabled, the HT81X 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 HT81X 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 "Hour of the day (0-23)". • **Check every week**: User needs to specify "Hour of the day (0-23)" and "Day of the week (0-6)". (Day of week is starting from Sunday). Default is **No**.
Always Check for New Firmware at Boot up	Configures the HT81X to always search for the new firmware at boot up. During the boot stage, the HT802 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 HT81X to search for the new firmware when the firmware prefix / suffix changes. When this option is selected, the HT81X will check for updates only when the pre/suffix has been changed.
Always Skip the Firmware Check	Configures the HT81X to skip the firmware check, when this option is selected the HT802 will always skip searching for a new firmware.
Disable SIP NOTIFY Authentication	Disables the SIP NOTIFY Authentication on the phone adapter. If set to “Yes”, the phone 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**.
SIP TLS Certificate	Specifies SSL certificate used for SIP over TLS is in X.509 format. The HT81X 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.
Enable TR-069	Sets the phone 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 HT81X.
Connection Request Password	Enters password for ACS to connect to the HT81X.

CPE SSL Certificate	Configures the Cert File for the phone adapter to connect to the ACS via SSL.
CPE SSL Private Key	Specifies the Cert Key for the phone adapter to connect to the ACS via SSL.
Enable SNMP	Enables / diables the SNMP function. Default is **No**
SNMP Trap Community	Configures the SNMP Trap Community.
SNMP Trap IP Address	Configures the SNMP Trap IP Address.
SNMP Trap Port	Configures the SNMP Trap Port. Default is 162.
SNMP Trap Version	Selects the SNMP Trap Version (Version 1 or Version 2c). Default is Version 2c.
SNMP Trap Interval	Configures the SNMP Trap Interval. Valid interval is (1-1440). Default is 5 minutes.
System Ring Cadence	Sets ring cadences for all incoming calls. Syntax: c=on1/off1-on2/off2-on3/off3;) Default is set to c=2000/4000; (US standards) on1 is the period of ringing ("On time" in "ms") while off1 is the period of silence. Up to three cadences are supported.
Call Progress Tones	Configures tone frequencies according to user preference. By default, the tones are set to North American frequencies. Frequencies should be configured 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 ring, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeats the pattern. • "Dial tone" • "Ring back tone" • "Busy tone" • "Reorder tone" • "Confirmation tone" • "Call-Waiting tone" • "Prompt Tone" Please refer to the document below to determine your local call progress tones: http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf
Prompt Tone Access Code	Specifies the key pattern to get Prompt Tone. Maximum 20 digits

Lock Keypad Update	Locks configuration update via keypad. Default is **No**.
Disable Voice Prompt	Removes ability to use integrated voice prompt menu configuration. Default is **No**.
Disable Direct IP Call	Deactivates Direct IP-to-IP calling function. Default is **No**.
NTP Server	Defines URL or IP address of the NTP (Network Time Protocol). Used by the HT81X to synchronize the date and time. Public NTP servers can be found at http://www.ntp.org . Default is **pool.ntp.org**
Allow DHCP Option 42 to NTP Server	Obtains NTP server address from a DHCP server using DHCP Option 42; it will override configured NTP Server. If set to “No”, the HT81X will use configured NTP server to synchronize time and date even if a NTP server is provided by DHCP server. Default is **Yes**.
Syslog Server	Sets IP address or URL of system log server. The server collects system log information from the HT81X.
Syslog Level	Selects log level; the level is one of DEBUG, INFO, WARNING, ERROR. Syslog messages are sent based on the following events. Default is **NONE**. 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) The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: GS_LOG: [device MAC address] [error code] error message **Example:** May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up
Send SIP Log	Configures the HT81X to send a replicate of the SIP packets on the syslog. Default is **No**

Download Device Configuration	Downloads actual device configuration file in .txt format.
Download Device XML Configuration	Downloads actual device configuration file in .xml format.
Upload Firmware	Allows users to upgrade the firmware with a single firmware file by browsing and loading the file from computer (local directory).
Upload Configuration	Allows users to upload the configuration file by browsing and loading it from computer (local directory).

Profiles Pages Definition

Table 9: Profiles Pages

Profiles (1,2)	
Profile 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 HT81X.
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 HT81X 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 HT81X 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, they 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**.
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. • **Use Configured IP:** If selected, please fill in Primary IP, Backup IP 1 and Backup IP 2 to be used for server look up. Default is **A Record**.
DNS SRV use Registered IP	When the HT81x is registered using the second SRV record, making 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 HT81X 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**.
Use Request Routing ID in SIP INVITE	If set to Yes, device will use the configured [**Request URI Routing ID**] in the SIP INVITE. This option is usually used under a SIP trunk account’s configuration. Default is **No**.
SIP Registration	Controls whether the HT81X 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 send 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 Diff-Serv: **24** | RTP Diff-Serv: **46**
Local SIP Port	Defines local port to use by the HT81X for listening and transmitting SIP packets. Default value for FXS 1 is 5060 and 5062 for FXS 2.
Local RTP Port	Defines the local RTP-RTCP port pair the HT81X will listen and transmit. It is the HT81X 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 HT81X 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 HT81X are behind the same NAT. Default is **No**.
Enable RTCP	Allows users to enable RTCP. The default setting is “Yes”.
Hold Target Before Refer	Allows user to hold or not hold the phone call before referring. The default setting 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 SIP 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 HT81X 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**.
SIP REGISTER Contact Header Uses	Specifies which address (LAN or WAN address) the device will detect to use it in SIP Register Contact Header. Default is **LAN Address**.
Caller ID Fetch Order (Only for HT818)	Selects the Caller ID display order which need to be respected by the HT81X. The available options are: • **Auto**: When set to "Auto", the HT81X 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 HT81X will use the FROM header to display the caller ID.
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 HT81X 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	Overrides the default settings for call control when both channels are in use
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. The HT81X 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 HT81X 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.
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 * (star) will be used.

	For example: if configured as *617, 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**.
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. Default is **60** seconds.
Hunting Group Ring Timeout	If call is not answered within this designated time period, the call will be forwarded to the next member of a Hunt Group. Default value is **20** seconds. **HT814 and HT818 only**.

Hunting Group Type	Specifies Hunting Group Type, either “Linear” or “Circular”. • **Linear** style will sort the call to the lowest numbered available line, this is also called “serial hunting”. • **Circular** style will distribute the calls "round-robin". If a call is assigned to line 1, the next call goes to 2 and the next to 3. The succession throughout each of the lines continues even if one of the previous lines becomes available. When the end of the hunt group is reached, the hunting starts over at the first line. Lines are skipped if they are still busy on a previous call. Default is Circular. **HT814 and HT818 only**.
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 Incomplete Address” responses. 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+} <u>Example of a simple dial plan used in a Home/Office in the US</u>: **{ ^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. An example dial plan will be: **{ *x+ }** which allows the user 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: Broadsoft, CBCOM, RNK, Huawei, China Mobile, ZTE IME, PhonePower, Telkom SA, Vonage, Metaswitch. 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. To turn off Session Timer, select "No" for Caller and Callee Request Timer, and Force Timer. Default is **No**.
UAC Specify Refresher	Specifies which end will act as refresher for outgoing calls. Default is **Omit**.: • **UAC**: The HandyTone acts as the refresher. • **UAS**: Callee or proxy server act as the refresher.
UAS Specify Refresher	Specifies which end will act as refresher for incoming calls. Default is **Omit**.: • **UAS**: The HandyTone acts as the refresher. • **UAC**: Callee or proxy server act as the refresher.
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. The default is null.
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 8 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. The valid range is between 96 and 127. Default is **97**.
OPUS Payload Type	Determines payload type for OPUS. The 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 Detection Mode	Permits 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).
Caller ID Scheme	Selects the caller id scheme, for example : Bellcore/Telcordia, ETSI-FSK …
DTMF Caller ID	Defines the start and stop tones.

Polarity Reversal	Reverses the polarity upon call establishment and termination. Default is **No**.
Loop Current Disconnect	Allows the traditional PBX used with HT81X 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**.
Loop Current Disconnect Duration	Configures the duration of voltage drop described in topic above. HT81X support a duration range from 100 to 10000 ms. Default value is **200**.
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. HT81X support 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. HT81X support 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 (ie. 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	Disables the LEC per call base. Recommended for FAX/Data calls. Default is **No**.
Disable Network Echo Suppressor	Disables the NEC 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)**.
Ring Frequency	Configures ringing frequency for your phone. 15-60 Hz, Default is **20** Hz.
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.

FXS Ports Page Definitions

Table 10: FXS Ports

FXS Ports	
Port	Display the port number
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".
Password	Specifies account password provided by VoIP service provider (ITSP) to register to SIP servers.
Name	Chooses a name to be associated to user.
Profile ID	Defines the profile ID for each port.
Hunting Group	Configures hunting group feature on the specific port. **HT814 and HT818 only**. For example: Port 1, 2, and 3 are members of the same Hunting Group. Port 1 is registered with a SIP account. Ports 2, and 3 are not registered. Ports 2 and 3 will be able to place outbound calls using the SIP account of port 1. Select appropriate value for Hunting Group feature. The original SIP account should be set to Active while the group members should be set to the port number of the Active Port. <u>Example configuration of a Hunting group:</u> FXS Port #1: SIP UserID and Authenticate ID entered, Hunting group set to "Active" FXS Port #2: SIP UserID and Authenticate ID left blank, Hunting Group set to "1" FXS Port #3: SIP UserID and Authenticate ID left blank, Hunting Group set to "1" FXS Port #4: SIP UserID and Authenticate ID entered, Hunting group set to "None" Hunting Group 1 contains ports 1, 2, 3. FXS port 4 is registered but it is not added to the Hunting Group 1
Request URI Routing ID	If configured, device will route the incoming call to designated port by request URI user ID in SIP INVITE.
Enable Port	Enables / Disables the port
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 is entered here. The HT81X will automatically append the "@" and the host portion of the corresponding SIP address.

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 web page) Set this to Yes when gateway is behind firewall on a private network.

DTMF Methods

The HT81X support 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 HT81X support following voice codecs. On Profile pages, choose the order of your favorite codecs:

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

Configuring HT81X Through Voice Prompts

As mentioned previously, The HT81X have a built-in voice prompt menu for simple device configuration.

Please refer to "Understanding HT81X 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 HT81X to use DHCP.

- **STATIC IP MODE**

  Select voice menu option 01 to enable HT81X 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 HT81X 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, HTTP and 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 HT81X can be automatically configured from a central provisioning system. When HT81X boots up, it will send TFTP or HTTP/HTTPS requests to download configuration files, "cfg000b82xxxxxx" and "cfg00082xxxxxx.xml", where "000b82xxxxxx" is the LAN MAC address of the HT81X. 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 or HTTP/HTTPS from the central server. A service provider or an enterprise with large deployment of HT81X 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 uses 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 they 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 HT81X support 2 profiles which can be configured with 2 SIP accounts. Please refer to the following steps in order to register your accounts via web user interface

1. Access your HT81X 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 **Profile (1** or **2)** pages.
5. In **Profile** tab, 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.
6. After configuring the SIP server and activating the profiles, you should access to **FXS Ports** page to register your accounts. In **FXS Ports** tab, set the following:
    a. **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.
    b. **Authenticate ID**: SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
    c. **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.

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

e. Set **Enable Port** to **Yes.**

For more information, related to above options please refer to Profile(s) settings and FXS Port Settings.

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

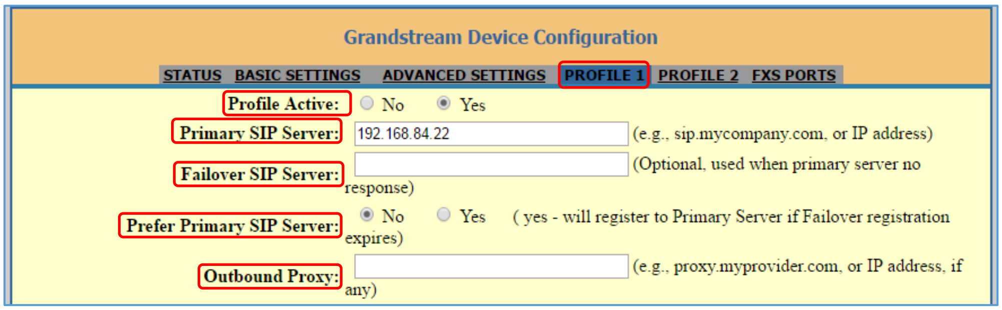


Figure 12: SIP Profiles Settings

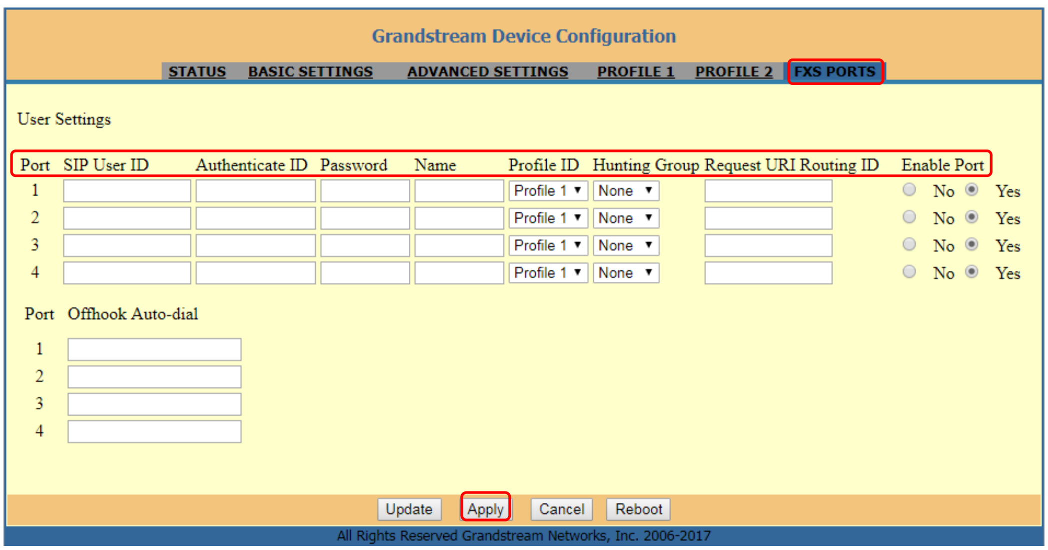


Figure 13: SIP Accounts settings

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 HT81X 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).

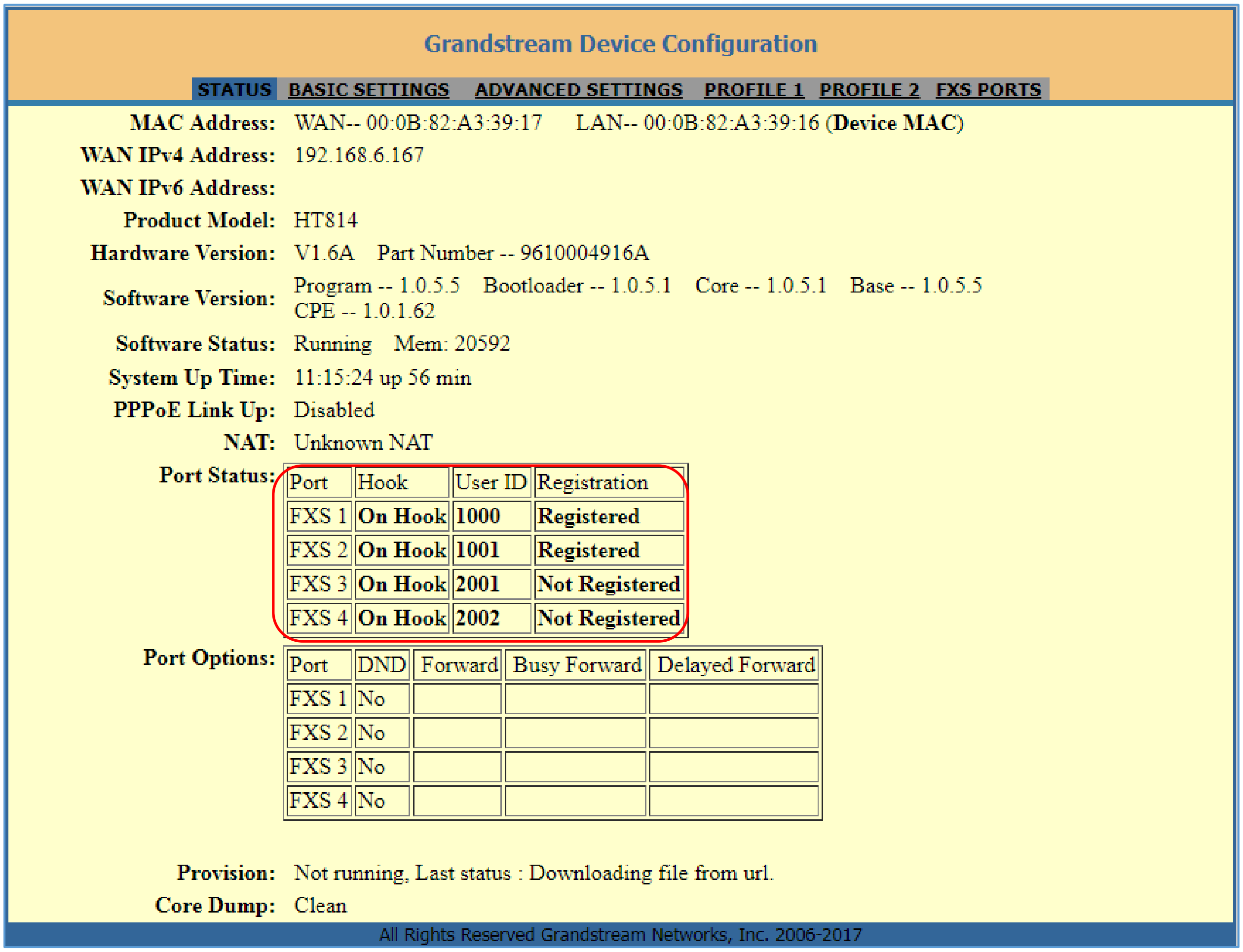


Figure 14: Accounts Status

Call Features

The HT81X support all the traditional and advanced telephony features.

Table 11: HT81X 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	**Disable 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 "*67" +" 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.

Rebooting HT81X 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.

UPGRADING AND PROVISIONING

The HT81X can be upgraded via TFTP/HTTP/HTTPS by configuring the URL/IP Address for the TFTP/HTTP/HTTPS server and selecting a download method. Configure a valid URL for TFTP or HTTP/HTTPS; the server name can be FQDN or IP address.

Examples of valid URLs:

firmware.grandstream.com

fw.ipvideotalk.com/gs

Firmware Upgrade procedure

Please follow below steps in order to upgrade the firmware version of your HT81X:

1. Access your HT81X 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**.
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 HT81X to update the new firmware.

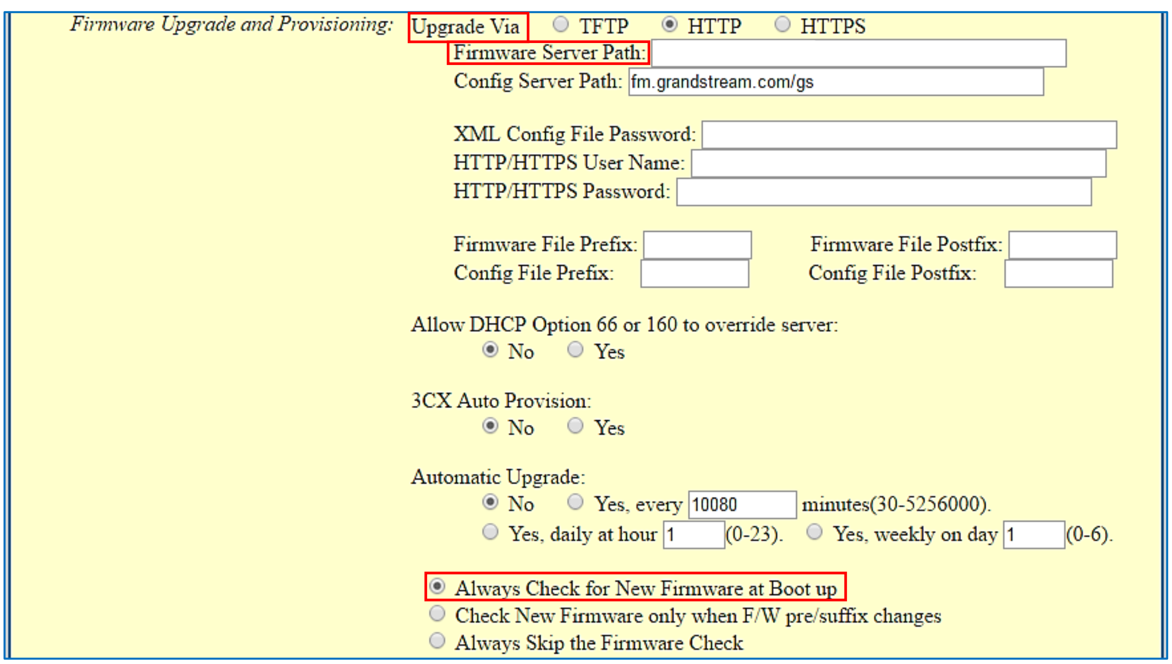


Figure 15: Firmware Upgrade Page

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 HT81X 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/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 HT81X.

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 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 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 HT81X 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

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 HT81X.
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:

Table 12: MAC Address Key Mapping

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 HT81X 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 HT81X were 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 HT81X

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 have 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.