Grandstream Networks, Inc.

HT80x V2 - Administration Guide

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INTRODUCTION

The HT801 V2 and HT802 V2 are analog telephone adapters (ATAs)that allow users to connect analog phones and faxes with a VoIP (Voice over IP) system to create high-quality and manageable IP telephony solutions for residential and office environments. Its ultra-compact size, voice quality, advanced VoIP functionality, security protection, and auto provisioning options enable users to take advantage of VoIP on analog phones. It also allows service providers to offer high-quality IP service to their market. The HT80x V2 series are ideal ATAsfor individual use as well as commercial IP voice deployments worldwide.

This manual will help you learn how to operate and manage your HT801 V2/HT802 V2 analog telephone adapters and leverage their many features, including but not limited to simple and quick installation, 3-way conferencing, direct IP Calling, and automated provisioning.

Once HT801 V2/HT802 V2/HT812 V2/HT814 V2/HT818 V2 is upgraded to firmware 1.0.9.3 or later, downgrading to 1.0.7.5 or earlier is not supported due to CPU frequency adjustments.

PRODUCT OVERVIEW

Feature Highlights

The following table contains major features of HT801 V2 and HT802 V2:

HT801 V2 HT802 V2	1 FXS port on HT801 V2 and 2 FXS ports on HT802 V2. Single 10/100Mbps port on both models. Support 3-way voice conferencing. Wide range of caller ID formats. Advanced telephony features, including Caller ID display or block, blind and attended call transfer, call forward, call-waiting, call hold, do not disturb, flash and more. T.38 Fax for creating Fax-over-IP. TLS and SRTP security encryption technology to protect calls and accounts. Automated provisioning options include TR-069 and XML config files. 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.

HT801 V2/HT802 V2 Features at a Glance

HT801 V2/HT802 V2 Technical Specifications

The following tables summarize all the technical specifications, including the protocols/standards supported, voice codecs, telephony features, languages, and upgrade/provisioning settings for the HT801 V2 and the HT802 V2.

	HT801 V2	HT802 V2
Interfaces		
Telephone Interfaces	One (1) RJ11 FXS port	Two (2) RJ11 FXS ports
Network Interface	One (1) 10/100Mbps auto-sensing Ethernet port (RJ45)	
LED Indicators	POWER, NET, PHONE	POWER, NET, PHONE1, PHONE2
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.722, G.723.1, G.729A/B, G.726-32, 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	5 REN: Up to 1km on 24 AWG	2 REN: Up to 1km on 24 AWG
Caller ID	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID	
Dial Methods	DTMF, Pulse	
Disconnect Methods	Busy Tone, Polarity Reversal/Wink, Loop Current	
Signaling		
Network Protocols	TCP/IP/UDP, RTP/RTCP (RFC1889, 1890), HTTP/HTTPS, ARP/RARP, ICMP, DNS, DHCP, NTP, TFTP, SSH, Telnet, STUN (RFC3489, 5389), SIP (RFC3261), SIP over TCP/TLS, SRTP, SNMP, TR-069, IMS/3GPP, IPoE	
DTMF Methods	In-audio, RFC2833 and/or SIP INFO	
Provisioning and Control	HTTP, HTTPS, SSH, TFTP, TR-069, secure and automated provisioning using AES encryption, syslog	
QoS	Layer 2 (802.1Q VLAN, SIP/RTP 802.1p) and Layer 3 (ToS, DiffServ, MPLS)	
Security		
Media	SRTP	
Control	TLS/SIPS/HTTPS	
Management	Syslog support, SSH, Telnet remote management using web browser	
Physical		
Universal Power Supply	Input: 100-240VAC, 50-60Hz. Output: 5.0VDC/1.0A	
Environmental	Operational: 32 deg - 104 degF or 0o - 40oC. Storage: 14 deg - 140 degF or -10o - 60oC. Humidity: 10 - 90% Non-condensing	
Dimensions and Weight	* 100mm x 100mm x 29.5mm * 102g (without package)	* 100mm x 100mm x 29.5mm * 114g (without package).
Compliance	FCC: Part15B CE: EN55032, EN55024, EN61000-3-2, EN61000-3-3, EN60950-1, RCM: AS/NZS CISPR22, AS/NZS60950.1, S003K.21	FCC 15B, AS/NZS CISPR22, AS/NZS60950, EN55022, EN55024, EN60950, EN61000-3-2, EN61000-3-3, UL (Power supply) K.21

HT801 V2/HT802 V2 Technical Specifications

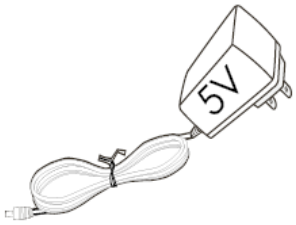
GETTING STARTED

This section provides basic installation instructions, including the list of the packaging contents and information to connect the HT801 V2/HT802 V2.

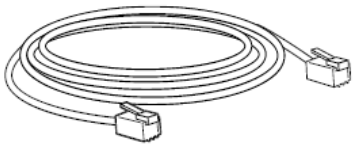
Equipment Packaging

The HT801 V2 ATApackage contains:

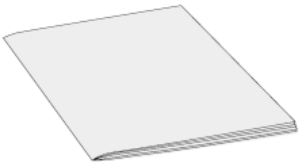
1x HT801 V2



1x 5V Power Adapter



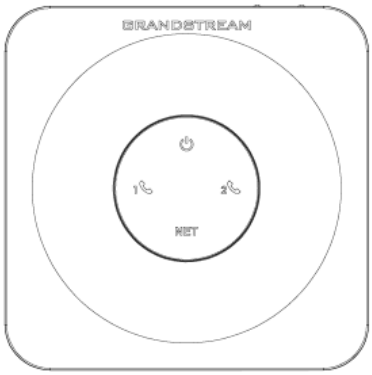
1x Ethernet Cable



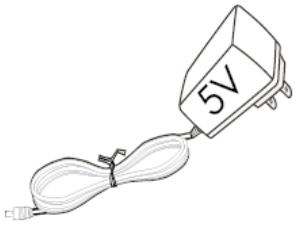
1x Quick Installation Guide

HT801 V2 Package Contents

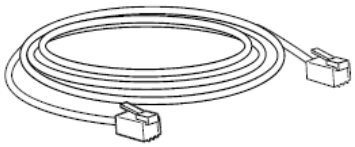
The HT802 V2 ATApackage contains:



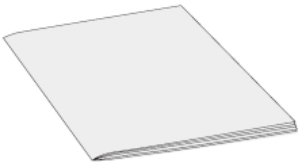
1x HT802 V2



1x 5V Power Adapter



1x Ethernet Cable



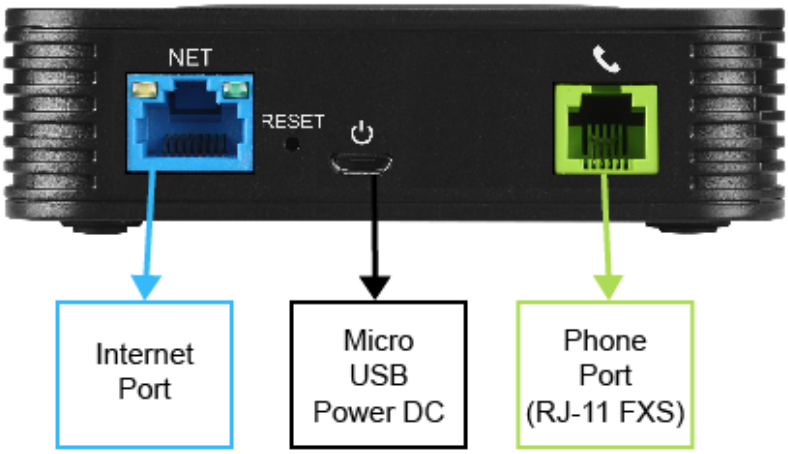
1x Quick Installation Guide

HT802 V2 Package Contents

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

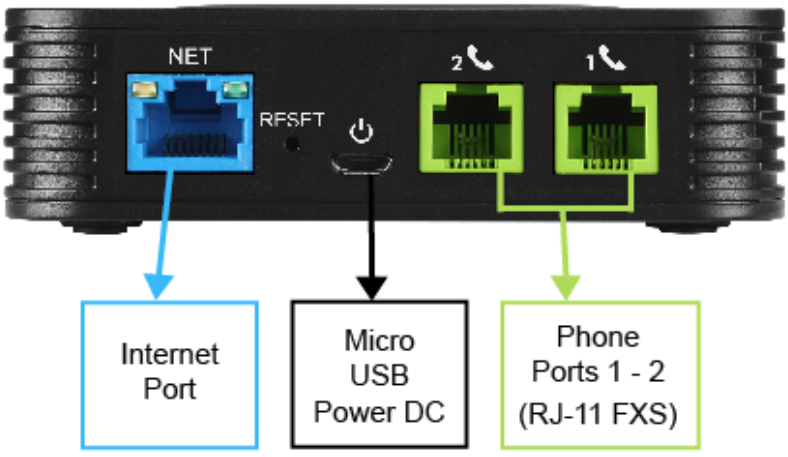
HT801 V2/HT802 V2 Ports Description

The following figure describes the different ports on the back panel of the HT801 V2.



HT801 V2 Ports

The following figure describes the different ports on the back panel of the HT802 V2.



HT802 V2 Ports

Phone port for HT801 V2Phone 1 & 2 ports for HT802 V2	Connects the analog phones/fax machines to the phone adapter using an RJ-11 telephone cable.
NET port	Connects the phone adapter to your router or gateway using an Ethernet RJ45 network cable.
Micro USB DC Power	Connects the phone adapter to PSU (5V - 1A).

Reset	Factory reset button. Press for 7 seconds to reset factory default settings.

HT801 V2/HT802 V2 Ports Definition

Connecting HT801 V2/HT802 V2

In order to connect your HT801 V2 or HT802 V2, please follow the steps mentioned below:

1. Insert a standard RJ11 telephone cable into the phone port and connect the other end of the telephone cable to a standard touch-tone analog telephone.
2. Insert the Ethernet cable into the NET port of the HT801 V2/HT802 V2 and connect the other end of the Ethernet cable to an uplink port (a router or a modem, etc).
3. Insert the power adapter into the HT801 V2/HT802 V2 and connect it to a wall outlet.

Power, NET, and Phone LEDs will be solidly lit once the HT801 V2/HT802 V2 is ready for use.

HT801 V2/HT802 V2 LEDs Pattern

There are 3 LED indications on HT801 V2 and 4 on HT802 V2 as shown in the below figure:

HT801 V2HT802 V2 LEDs Indications

LED Indication	Status
Power LED	The Power LED lights up when the HT801 V2/HT802 V2 is powered on, and it flashes when the unit is booting up
NET LED	The NET LED lights up when the HT801 V2/HT802 V2 is connected to a network through the Ethernet port, and it flashes when there is data being sent or received.
Phone LED for HT801 V2 Phone LED 1&2 for HT802 V2	The phone LEDs 1 & 2 indicate the status of the respective FXS port-phone on the back panelOFF - UnregisteredON (Solid Blue) - Registered and AvailableBlinking every second - Off-Hook / BusySlow blinking - FXS LEDs indicates voicemail

HT801 V2/HT802 V2 LEDs Pattern Description

CONFIGURATION GUIDE

There are two ways to configure HT801 V2/HT802 V2 :

- The IVR voice prompt menu.
- The embedded Web GUI on the HT801 V2/HT802 V2 using the PC's web browser.

Obtain HT80x V2 IP Address via Connected Analog Phone

HT801 V2/HT802 V2 is by default configured to obtain the IP address from the DHCP server where the unit is located. To know which IP address is assigned to your HT801 V2/HT802 V2, you should access the IVR (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 the phone for the HT801 V2 or phone 1 or phone 2 ports of your HT802 V2.
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 HT80x V2 Interactive Voice Prompt Response Menu

The HT801 V2/HT802 V2 has 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 HT801 V2/HT802 V2.

To access the IVR Menu, users need to dial "***".

Menu	Voice Prompt	Options
Main Menu	"Enter a Menu Option"	Press "*" for the next menu optionPress "#" to return to the main menuEnter 01-05, 07,10, 12-17, 20, 47 or 99 menu options
01	"DHCP Mode","Static IP Mode""PPPoE Mode"	Press "9" to toggle the selectionIf 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 HT801 V2/HT802 V2 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-729OPUSG722
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.
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/HTTP /FTP/FTPS
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 changesNever upgrade
20	Certificate Type	Annouces certificate information.
47	"Direct IP Calling"	Enter the target IP address to make a direct IP call, after dial tone.
86	Voice Mail	Access to your voice mails messages.
99	"RESET"	Press "9" to reboot the deviceEnter 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 the menu option and 12 digits for the 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 keyed in like 192168000026. No decimal is needed).
- Key entry cannot be deleted, but the phone may prompt an error once it is detected.
- Dial *98 to announce the extension number of the port.

Configuration via Web Browser

The HT801 V2/HT802 V2 embedded Web server responds to HTTP GET/POST requests. Embedded HTML pages allow a user to configure the HT801/HT802 through a web browser such as Google Chrome, Mozilla Firefox, and Microsoft's IE.

Accessing the Web UI

Please follow the steps mentioned below to access the HT801 V2/HT802 V2 Web UI:

1. Connect the computer to the same network as your HT801 V2/HT802 V2.
2. Make sure the HT801 V2/HT802 V2 is booted up.
3. You may check your HT801 V2/HT802 V2 IP address using the IVR on the connected phone. (To do this, please refer to this section [Obtain HT80x V2 IP Address via Connected Analog Phone])
4. Open the Web browser on your computer.
5. Enter the HT801 V2/HT802 V2's IP address in the address bar of the browser.
6. Enter "admin" under the username as well as the administrator's password to access the Web Configuration Menu (the default password can be found on the sticker located at the back of the unit).

The computer must be connected to the same sub-network as the HT801 V2/HT802 V2. This can be easily

done by connecting the computer to the same hub or switch as the HT801 V2/HT802 V2.

Web UI Access Level Management

There are three access levels for the login page:

User Level	User	Password	Web Pages Allowed
End User Level	user	123	View all pages but can only modify basic settings
Administrator Level	admin	random password located at the back of the unit	Browse all pages and modify all settings
Viewer Level	viewer	viewer	View all pages but no changes allowed.

- The password is case-sensitive and must contain 8-20 characters, at least one number, one uppercase, and one lowercase letter.
- When changing any settings, always submit them by pressing the Update or Apply button at the bottom of the page.
- Some changes require a reboot of the HT80x V2 unit, such as FXS Port settings.
- By default, user and viewer access levels are disabled. In order to enable them, please access **System Settings** -> **Security Settings -> Web Access**.
- After the initial login with the default admin, user, or viewer password, the device will prompt the user to change the password immediately.
- The **P-value P22665** configures whether the default password change is enforced. Setting **P22665 = 1** forces the user to change the default password during the first login, while **P22665 = 0** disables this requirement. The default value is **1**. This parameter is not available in the Web UI and can only be configured in the device's configuration file.

Saving the Configuration Changes

After modifying any configuration parameters, users can save the changes by clicking on the **Save** button. Once the configuration is saved, an **Apply** button will appear at the top of the page to allow users to apply the changes.

Users can also directly click on the **Save and Apply** button for the configuration changes to be saved and applied.

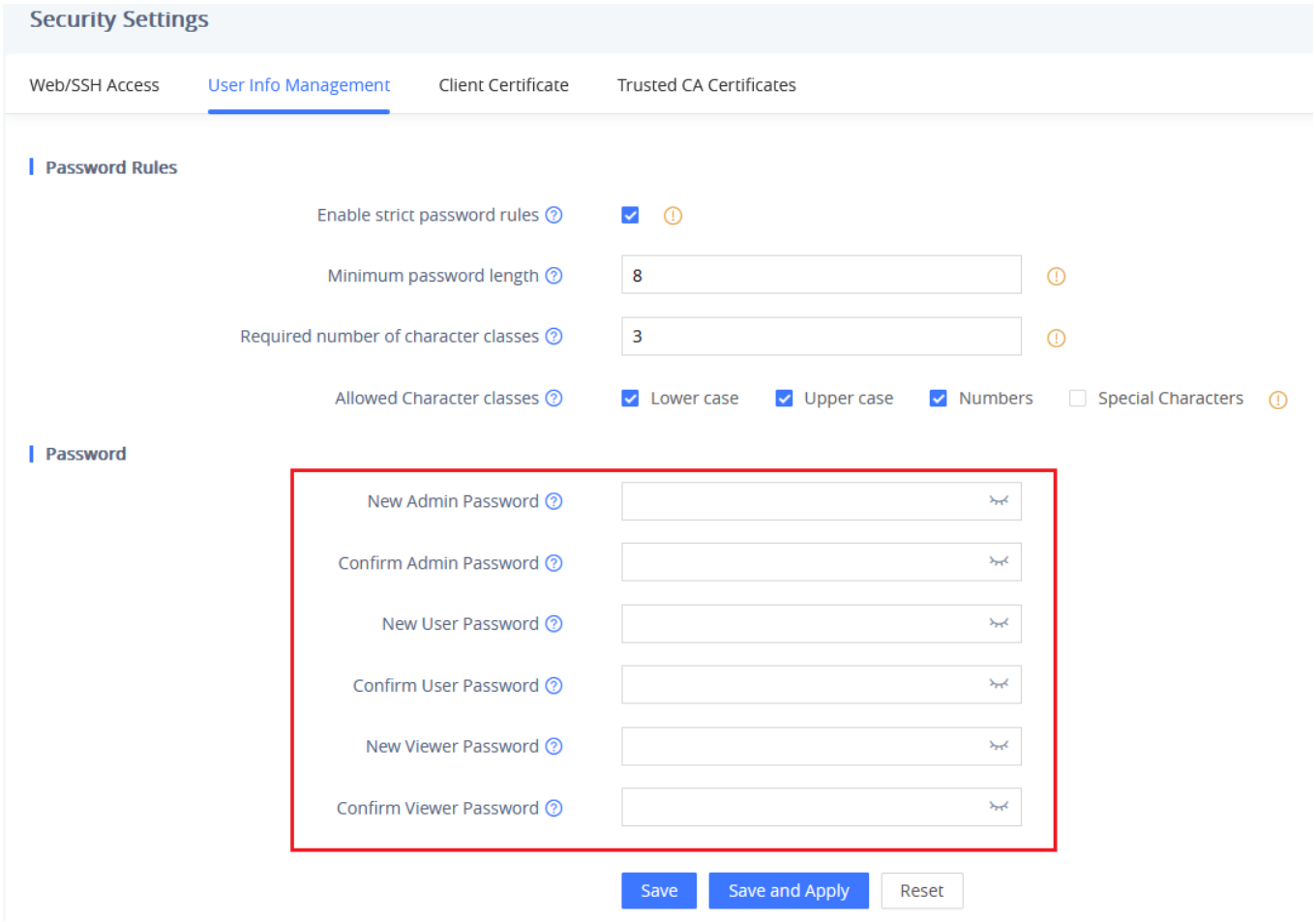
We recommend rebooting or powering cycling the ATAafter applying all the changes.

Changing Admin/User/Viewer Level Password

To change the password for any of the access levels, please refer to the following steps:

1. Access your HT801 V2/HT802 V2 Web UI by entering its IP address on your browser.
2. Enter the access level (admin, user, or viewer) under your username and your default password. (Please refer to this section [Web UI Access Level Management] for the default password).
3. Press **Login** to access your configuration settings and navigate to **System Settings -> Security Settings -> User Info Management -> Password.**
4. **Enter** the new admin/user/viewer password and **confirm** it (The new Password must contain 8-20 characters, at least one number, one uppercase, and a lowercase letter).

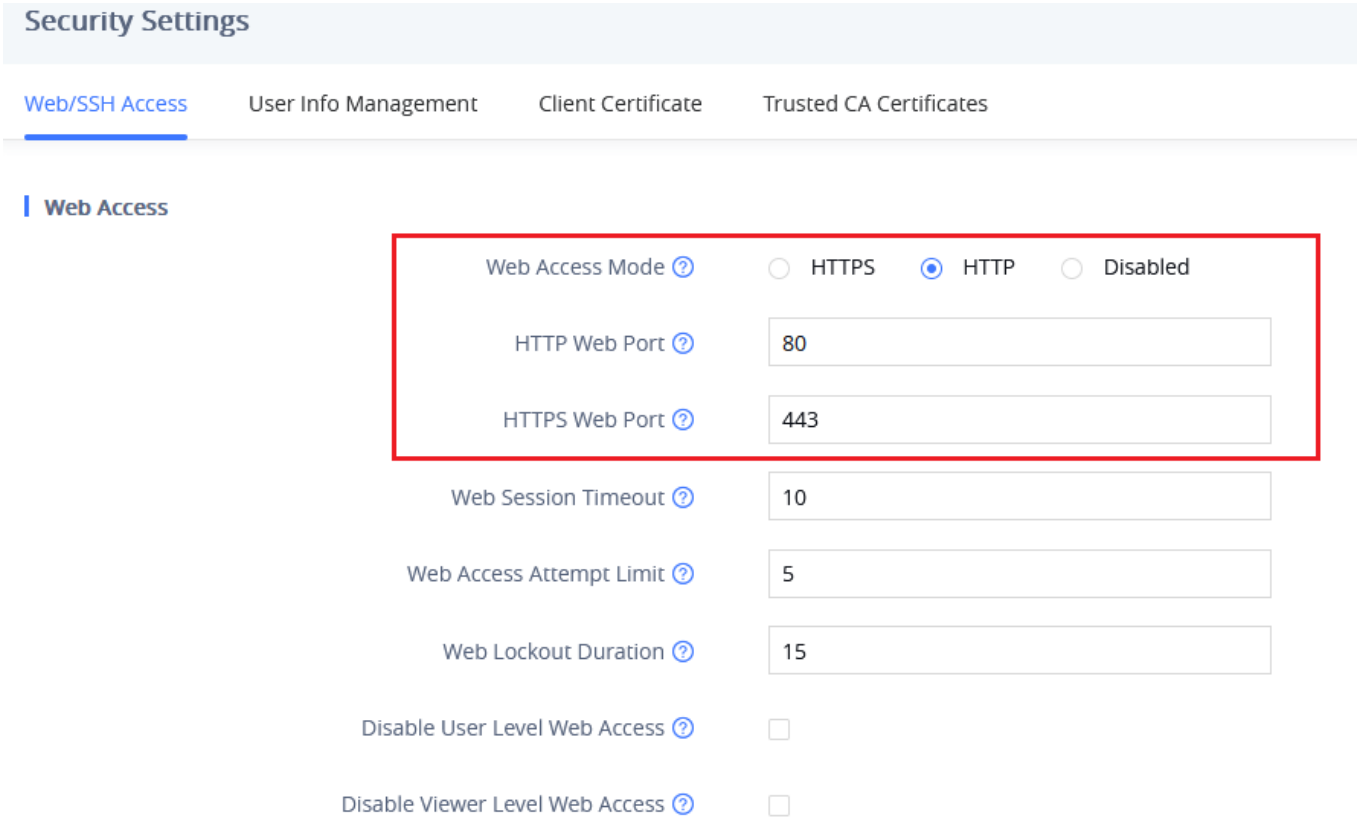
5. **Save** and **apply** the new admin/user/viewer password.



- To allow the User and Viewer accounts to access the Web UI, ensure that **Enable User Level Web Access** and **Enable Viewer Level Web Access** are checked under **System Settings -> Security Settings -> Web/SSH Access -> Web Access**.
- To control the display of User and Viewer Password settings, configure the **P-value P22663**. Setting **P22663 = 1** displays the User and Viewer Password settings on the Web UI, while **P22663 = 0** hides these options. The default value is **1**. This parameter is not available in the Web UI and can only be configured in the device's configuration file.

Changing the HTTP/HTTPS Web Port

1. Access your HT801 V2/HT802 V2 web UI by entering its IP address on your browser.
2. Enter "admin" under the username, then type your admin password.
3. Press **Login** to access your settings and navigate to **System Settings ?? Security Settings ?? Web/SSH Access ?? Web Access**.
4. Choose the desired Web Access Mode and the new HTTP/HTTPS port. The valid range is [1-65535].
5. **Save** and **apply** the new port number.



Web Configuration Pages Definitions

The following sections describe the configuration options on the HT801 V2/HT802 V2 Web UI.

Status Page Definitions

System Info

Product Model	Displays the Product model: HT801V2 or HT802V2.
Serial Number	Displays the device's serial number.
Hardware Version	
Hardware Version	Displays the hardware version.
Part Number	Displays the part number.
Software Version	
Software Version	Program: Displays the current software version running on the device. Bootloader: Displays the bootloader version. Core: Displays the Core version. Base: Displays the Base version. CPE: Displays the Customer Premises Equipment version.
Vendor Default Config Version	
Project	Displays the unique project ID of the custom configuration file.
Vendor Default Config Version	Displays the version of the custom configuration file.
Operation State	
Software Status	Displays whether the software is running correctly on the device.
Mem	Displays the memory usage.
System Up Time	Displays the duration of the system up time.

System Current Time	Displays the HT8xx current time.
CPU Load	Displays the processor load.
Provision	Displays the last status of provisioning.
System Info	Downloads a diagnostic system information file from the device, containing technical details such as firmware version, uptime, system status, and other runtime information useful for troubleshooting. Note: The device periodically saves the current system time to an internal time file. This is included in the System Info download package to assist with troubleshooting time-related issues.
Core Dump	
Core Dump	Displays the core dump file when available and which could be downloaded for troubleshooting purpose.

Network Status

MAC Address	Displays the device's MAC address.
IPv4 Address	Displays the assigned IPv4 address.
IPv6 Address	Displays the assigned IPv6 address.
VPN IPv4 Address	Displays the connected VPN IPv4 address, if available.
VPN IPv6 Address	Displays the connected VPN IPv4 address, if available.
Network Cable Status	Displays the status and speed of the NET port, it is shown as UP if connected, and DOWN if disconnected.
PPPoE Link Up	Indicates active connectivity through PPPoE.
NAT	Displays the type of NAT used.
Certificate Type	Displays the certificate type.

Port Status

Port Status	
Port	Displays the analog port : FXS1, FXS2 (for HT802 V2).
Hook	Shows the status of the port, On Hook, Off Hook, Not connected...
SIP User ID	Displays the SIP user ID registred on the port.
Registration	Displays wether the port is registered or not.
Sip Port	Displays the SIP port used by the analog interface.
Port Options	
Port	Displays the FXS port.
DND	Displays whether the connected analog phone is on DND mode or not.
Forward	Indicates the forwarding number.
Busy Forward	Indicates the busy forwarding number.
Delayed Forward	Indicates the delayed forwarding number.
CID	Indicates the Caller ID.
Call Waiting	Indicates if Call waiting is enabled.
SRTP	Indicates if Secured RTP is enabled.

System Settings Page Definitions

Basic Settings

Enable Voice Prompt	Enable/disables Voice Prompt, it is disabled if set to No. Enabled by default.
Enable Direct IP Call	Enable/diasbles direct IP calling, it is disabled if set to No.Enabled by default.
Blocklist For Incoming Calls	Blocks calls from specific numbers. Use "," to separate numbers.
Lock Keypad Update	The configuration update via keypad is disabled if set to Yes. Default value is "No".
Play Busy Tone When Account is unregistered	If set to Yes, busy tone will be played when user goes offhook from an unregistered account. Default value is "No".
DHCP Option 17 Enterprise Number	Fill in the DHCP Option 17 enterprise number, the default is 3561. Valid range is 0-65535.
ATA Settings	
STUN Server	Defines the IP address or domain name of the STUN server. Only non-symmetric NAT routers work with STUN.
Keep-Alive Interval	Specifies how often the device sends a blank UDP packet to the SIP server in order to keep the "ping hole" on the NAT router to open. Default is 20 seconds. Range is 10-160 seconds.
Use STUN to detect network connectivity	Enable/Disable Using STUN keep-alive to detect WAN side network problems Disable by default.

Time and Language

Time Zone	
NTP Server	This parameter sets the IP address of the NTP server. The device will obtain the date and time from the server.
Secondary NTP Server	This feature allows users to configure the secondary NTP server, by default, no secondary NTP Server is configured.
Allow DHCP Option 42 to override NTP server	If DHCP Option 42 is enabled, the NTP server can be changed. Enabled by default.
Time Zone	Selects the time zone where the phone is located and controls the date/time display, Note: you can set the timezone to be the self defined time.
Self-Defined Time Zone	Allows users to define their own time zone when time zone is set to "Using self-defined Time Zone". The syntax is: std offset dst [offset], start [/time], end [/time] Default is set to: MTZ+6MDT+5,M3.2.0,M11.1.0 MTZ+6MDT+5 This indicates a time zone with 6 hours offset with 1 hour ahead (when daylight saving) which is U.S central time. If it is positive (+) if the local time zone is west of the Prime Meridian (A.K.A: International or Greenwich Meridian) and negative (-) if it is east. M4.1.0,M11.1.0The 1st number indicates Month: 1,2,3.., 12 (for Jan, Feb, .., Dec)The 2nd number indicates the nth iteration of the weekday: (1st Sunday, 3rd Tuesday...)The 3rd number indicates weekday: 0,1,2,..,6 (for Sun, Mon, Tues, ... ,Sat)Therefore, this example is the DST which starts from the Second Sunday of March to the 1st Sunday of November.
Allow DHCP server to set Time Zone	Allows the local server's DHCP option 2 to override the phone's time zone setting.
Local Time	When Sync PC is pressed, the HT8xx will synchronise the date and time from the connected PC.
Language	
IVR Language	Selects IVR voice prompt language type, the supported languages are: English, Chinese, Russian, Spanish.

Ringtone

System Ring Cadence	Cadence on+off value range (0, 16000) milliseconds.
Prompt Tone Access Code	The key pattern to get Prompt Tone. Maximum 20 digits. No default value provided
CPT Settings: Dial Tone Ringback Tone Busy Tone Reorder Tone Confirmation Tone Call Waiting Tone Wait for Dial-Tone Conference Party Hangup Tone Special Proceed Indication Tone Special Condition Tone	Using these settings, users can configure tone frequencies and cadence according to their preference. By default, they are set to North American frequencies. Configure these settings with known values to avoid uncomfortable high pitch sounds. ON is the period of ringing ("On time" in 'ms') while OFF is the period of silence. In order to set a continuous tone, OFF should be zero. Otherwise, it will ring ON ms and a pause of OFF ms and then repeat the pattern. Example configuration for N.A. Dial tone: fxsch1:f1=350@-17,f2=440@-17,c=0/0;fxoch1-4:f1=350@-17,f2=440@-17,c=0/0; Syntax: fxsch1:f1=val[,f2=val[,c=on1/off1[-on2/off2[-on3/off3]]]]; fxoch x-y:f1=val[,f2=val[,c=on1/off1[-on2/off2[-on3/off3]]]] (Frequencies are in (10, 4000) Hz and cadence on and off are in (0, 64000) ms)

Security Settings

Security Settings -> Web/SSH Access	
Basic Settings	
Web Access Mode	Selects the web page access protocol. The available options are: HTTP, HTTPS, Disabled.The default settings is "HTTPS".
HTTP Web Port	Defines the HTTP Web Port. The default setting is "80". The valid is 1-65533.
HTTPS Web Port	Defines the HTTPS Web Port. The default setting is "443". The valid range is 1-65533.
Web Session Timeout	Defines the web session timeout in minutes. The default setting is "10", the valid range is 1-60.
Web Access Attempt Limit	Defines the number of failed web login attempts allowed before web access is locked.The default setting is "5", the valid range is 1-10.
Web Lockout Duration	Defines the duration for which web access is locked in minutes. The default setting is "15", the valid range is 1-60.
Enable User Level Web Access	Enables or disables user-level web access. Disabled by default.
Enable Viewer Level Web Access	Enables or disables viewer-level web access. Disabled by default.
SSH	
Enable SSH	Enables or disables SSH access to the phone. Enabled by default.
SSH Port	Defines the SSH port. This port cannot be the same as the Telnet Port. The default setting is "22". The valid range is 1-65533.
SSH Idle Timeout	Configures SSH session Timeout in seconds. The default setting is "0". The valid range is 0-86400.
Enable Telnet	Enables or disables Telnet access.Disabled by default.
Telnet Port	Defines the Telnet port. This port cannot be the same as the SSH Port. The default setting is "23". The valid range is 1-65535.
Telnet Idle Timeout	Configures Telnet session Timeout in seconds. The default setting is "0". The valid range is 0-86400.

Security Controls for SSH/Telent Access	Specifies security measures and controls that apply to the SSH/Telnet protocol to ensure secure and authorized access to the device, the following options are supported: Only Allow SSH private IP users to set system Pvalue: This option permits only users with SSH access from private IP addresses to modify specific system-level configuration parameters (system Pvalues).Allow all SSH users to set system Pvalue: With this setting, all users having SSH access can modify specific system-level configuration parameters (system Pvalues).Allow all SSH users to set any Pvalue: This allows all users with SSH access to modify any configuration parameter (P-value).Allow any user to set any Pvalue: Any user, regardless of their privileges, can modify any configuration parameter (Pvalue) through SSH or Telnet access.Prohibit setting Pvalue: This option blocks all users from modifying any configuration parameter (Pvalue) via SSH or Telnet access.The Default Value is "Allow any user to set any PValue".
Enable Web/SSH Access	Enables or Disables the Web and SSH access through the WAN port. The available options are the following:No: No access to the web or SSH from any IP address on the WAN side.Yes: Access for the Web GUI and SSH is enabled on the WAN side.Auto: Only private IP could access the web or SSH on the WAN side.The default setting is "Yes".
Access Control	
Allow List for WAN Side	If Enable Web/SSH Access is set to Yes or Auto. Users can configure the white List for WAN Side to be used for remote management.Multiple IPs are supported and need to be separated by space. Example:192.168.5.222 192.168.5.223 192.168.7.0/24 Note: If both blocklist and allowlist are not empty, the blocklist is processed first, followed by the allow list.
Block List for WAN Side	If Enable Web/SSH Access is set to Yes or Auto. Users can configure the black List for WAN Side to ban WAN side web access.Multiple IPs are supported and need to be separated by space. Example: 192.168.5.222 192.168.5.223 192.168.7.0/24 Note: If both blocklist and allowlist are not empty, the blocklist is processed first, followed by the allow list.
Security Settings -> User Info Management	
Password Rules	
Enable strict password rules	Enable or diables strict password rules.
Minimum password length	Defines the Minimum password length, Range: 4-30, default is 8.
Required number of character classes	Sets the minimum number of character classes that a password should contain, composed of allowed combinations of different character classes; Range: 0-4, default is 3.
Allowed Character classes	Defines the Allowed Character classes, which are: Lower case, Upper case, Numbers, Special characters.
Password	
New Admin Password	Defines the new admin password, Must contain 4-30 characters, When strict password rules are enabled, the password needs to comply with the settings in the password rules; Do not display password for security reasons.
Confirm Admin Password	Confirms the entered admin password.
New User Password	Defines the new user password, Must contain 4-30 characters, When strict password rules are enabled, the password needs to comply with the settings in the password rules; Do not display password for security reasons.
Confirm User Password	Confirms the entered user password.
New Viewer Password	Defines the new viewer password, Must contain 4-30 characters, strict password rules are enabled, the password needs to comply with the settings in the password rules; Do not display password for security reasons.When
Confirm Viewer Password	Confirms the entered viewer password.
Security Settings -> Client Certificate	

SIP TLS	
Enable/Disable Weak Cipher Suites	Controls whether the device allows the use of weakened or deprecated TLS cipher suites. You can either allow weak TLS ciphers or selectively disable specific weak encryption, authentication, or key-exchange algorithm. Disabling weak cipher suites is recommended to improve security and comply with modern TLS standards. The available option are : Enable Weak TLS Cipher Suites - Allows the device to use old or weak TLS encryption ciphers.Disable Symmetric Encryption RC4/DES/3DES - Blocks weak symmetric encryption types like RC4 and DES/3DES.Disable Symmetric Encryption SEED - Blocks the SEED symmetric encryption algorithm.Disable Symmetric Encryption IDEA/eNULL - Blocks IDEA encryption and NULL (no-encryption) ciphers.Disable All Of The Above Weak Symmetric Encryption - Blocks all listed weak symmetric encryption algorithms.Disable Symmetric Authentication MD5 - Blocks MD5-based authentication because it is cryptographically weak.Disable All Of The Above Weak Symmetric Authentication - Blocks all weak authentication methods listed.Disable Protocol Version SSLv2/SSLv3 - Disables insecure SSLv2 and SSLv3 protocol versions.Disable All Of The Above Weak Protocol Version - Disables all listed weak TLS/SSL protocol versions.Disable Server Authentication aNULL/aECDH - Blocks cipher suites without proper server authentication.Disable All Of The Above Weak Server Authentication - Blocks all listed weak server-authentication methods.Disable Key Exchange Algorithm kRSA - Disables RSA key-exchange, which is deprecated in modern TLS.Disable All Of The Above Weak Key Exchange Algorithm - Disables all listed weak key-exchange methods.Disable All Of The Above Weak TLS Ciphers Suites - Disables every weak TLS cipher and method included in the list.By default, it is set to "Disable All Of The Above Weak TLS Ciphers Suites"
SIP TLS Certificate	Specifies SSL certificate used for SIP over TLS is in X.509 format. The HT8xx has built-in private key and SSL certificate. The maximum supported length is 8192.
SIP TLS Private Key	Specifies TLS private key used for SIP over TLS is in X.509 format. The maximum supported length is 4069.
SIP TLS Private Key Password	Specifies SSL Private key password used for SIP Transport in TLS/TCP.
TLS Version	
Minimum TLS Version	This feature allows customer to choose desired Minimum TLS Version. The available options are: UnlimitedTLS 1.0TLS 1.1TLS 1.2TLS 1.3The default setting is "TLS 1.2".
Maximum TLS Version	This feature allows customer to choose desired Maximum TLS Version.The available options are:UnlimitedTLS 1.0TLS 1.1TLS 1.2TLS 1.3The default setting is "Unlimited".
Custom Certificate	
Custom Certificate	Allows users to update to the device their own certificate signed by a custom CA certificate to manage client authentication.
Security Settings -> Trusted CA Certificates	
Load CA Certificates	This feature allows user to specify which certificate to trust when performing server authentication. Build-in trusted: (Default) Build-in trusted certificates Custom trusted certificate: Uploaded Certificates All trusted Certificates: Both built-in and uploaded Certificates
Trusted CA Certificates ATrusted CA Certificates BTrusted CA Certificates CTrusted CA Certificates D	These trusted CA certificates will be used for the authentication server TLS certificate
Security Settings -> SCEP	
Basic Settings	
Enable SCEP	Enables or disables the SCEP feature.Disabled by default.
SCEP Server URL	Defines the SCEP server URL (CA or RA). The URL must include the protocol prefix. Currently, only HTTP is supported.
CA FingerPrint Infomation	Defines the fingerprint of the CA certificate.

Challenge Password	Defines the temporary challenge password included in the Certificate Signing Request (CSR) when requesting a new certificate.
Local configuration	
Local Public and Private Key Length	Specifies the length of the local public and private keys.The default setting is "3072 bit".
The Name of Certificate Applicant	Defines the name of the certificate applicant. This field is required.
The Country of Certificate Applicant Located	Selects the country where the certificate applicant is located.
The State of Certificate Applicant Located	Defines the state or province where the certificate applicant is located.
The Locality of Certificate Applicant Located	Defines the locality of where the certificate applicant is located.
The Organization of Certificate Applicant Located	Defines the location of the organization to which the certificate applicant belongs.
The Organization Unit of Certificate Applicant	Defines the organization unit of the certificate applicant, such as a company, government agency, or nonprofit organization.
HTTP Authentication	
CID	Configures the HTTP authentication user ID used to access the CA server.
Password	Configures the HTTP authentication password used to access the CA server.
Certificates and Updates	
CA Certificates	Displays the CA certificate obtained through SCEP. Note: Copy the displayed CA certificate and import it into the trusted CA certificate store on the CA server.
Client Certificate Alternative Name	Configures additional identifiers for the client certificate. These values are added as Subject Alternative Name (SAN) entries on the server and allow the device to be identified using multiple aliases. In the configuration file, different aliases are separated by semicolons (;).The available options are:DNS Name (dNSName): Enter a hostname or domain, for example www.example.com.IP Address (iPAddress): Enter the IPv4 or IPv6 address, for example 192.168.0.1 or 2001:db8::1.Email Address (rfc822Name): Enter an email address, for example user@example.com.X.500 Distinguished Name (directoryName): Enter an X.500 distinguished name, for example /C=US/ST=CA/L=Los Angeles/O=Example Inc./CN=www.example.comURI (uniformResourceIdentifier): Enter a uniform resource identifier, for example https://www.example.com.Kerberos Service Name (krb5PrincipalName): Enter a Kerberos service principal name, for example HTTP/www.example.com.The default setting is "$MAC.local".
Update Interval before Certificate Expiration	Defines the number of days (1-30) before expiration to trigger certificate renewal. The system attempts renewal when the remaining validity is less than the defined interval.Pressing Update Now starts the renewal manually. The default setting is "20".
Scope of Use	Configures where the CA and client certificates are applied. Certificates for selected modules are managed automatically via SCEP, without manual upload or renewal.The available options are: Custom Certificate, 802.1X, OpenVPN, TR069. The default setting is "Custom Certificate".

Management Settings

TR-069	
Enable TR-069	Enable/disable the "CPE WAN Management Protocol" (TR-069). The default setting is Yes. Note: When configured, users can send a "false ring" signal through the TR-069 protocol for testing pruposes.
ACS URL	Specifies URL of TR-069 Auto Configuration Servers (e.g., http://acs.mycompany.com), or IP address.Default setting is: "https://acs.gdms.cloud"

ACS Username	Enter username to authenticate to ACS.
ACS Password	Enter password to authenticate to ACS.
Periodic Inform Enable	If scheduled connection is enabled, the phone will send connection requests to the ACS server regularly. Default setting is "No". Note: FXS port status can be detected through TR-069.
Periodic Inform Interval	Sets frequency that the inform packets will be sent out to ACS. Default is 300 seconds.
Connection Request Username	Configure the ACS username for connecting the phone.
Connection Request Password	Configure the ACS password for the connected phone.
Connection Request Port	Configures the TR-069 connection request port. The value range is 0 to 65535.Default is 7547
CPE SSL Certificate	Configure the certificate file for the telephony adapter to connect to the ACS over SSL.
CPE SSL Private Key	Specify the certificate key for the phone adapter to connect to the ACS via SSL.
SNMP Settings	
Enable SNMP	Enables the Simple Network Management Protocol. Default is "No"
SNMP Version	Choose between (Version 1, Version 2c, or Version 3).
SNMP Port	Listening Port of SNMP. Valid range is 161 or 1025-65535. (Default is 161).
SNMP Trap IP Address	IP address of trap destination. Up to 3 trap destinations are supported. Users should enter the IP addresses separated with comma (,).
SNMP Trap Port	Port of Trap destination. Valid range is 162 or 1025-65535. (Default 162).
SNMP Trap Version	Choose between (Version 1, Version 2c, or Version 3).
SNMP Trap Interval	Time interval in minutes between traps. (Default is 5).
SNMPv1/v2c Community	Name of SNMPv1/v2c community.
SNMPv1/v2c Trap Community	Name of SNMPv1/v2c trap community.
SNMPv3 User Name	Username for SNMPv3.
SNMPv3 Security Level	noAuthUser: Users with security level noAuthnoPriv and context name as noAuth. authUser: Users with security level authNoPriv and context name as auth. privUser: Users with security level authPriv and context name as priv.
SNMPv3 Authentication Protocol	Select the Authentication Protocol: "None" or "MD5" or "SHA."
SNMPv3 Privacy Protocol	Select the Privacy Protocol: "None" or "AES/AES128" or "DES".
SNMPv3 Authentication Key	Enter the Authentication Key.
SNMPv3 Privacy Key	Enter the Privacy Key.
SNMPv3 Trap User Name	Username for SNMPv3 Trap.
SNMPv3 Trap Security Level	noAuthUser: Users with security level noAuthnoPriv and context name as noAuth. authUser: Users with security level authNoPriv and context name as auth. privUser: Users with security level authPriv and context name as priv.
SNMPv3 Trap Authentication Protocol	Select the Authentication Protocol: "None" or "MD5" or "SHA".
SNMPv3 Trap Privacy Protocol	Select the Privacy Protocol: "None" or "AES/AES128" or "DES".
SNMPv3 Trap Authentication Key	Enter the Trap Authentication Key.
SNMPv3 Trap Privacy Key	Enter the Trap Privacy Key.
MIB	Download Management Information Base (MIB) file.

Management Interface	
Enable Management Interface	Allows activation of a dedicated interface for remote management and configuration of the device. Disabled by Default.
Management Access	Defines the allowed methods (protocols) and permissions for remote access and configuration of the device, two options can be selected: Management Interface Only: allows remote access and configuration solely through the dedicated management interface. Both Service and Management Interfaces: permits remote access and configuration through both the management interface and the service interface.
Enable SNMP Through Management Interface	Allows the activation of SNMP (Simple Network Management Protocol) access exclusively through the dedicated management interface for monitoring and managing the device remotely. Disabled by Default.
Enable TR069 Through Management Interface	Allows the activation of TR-069 protocol access exclusively through the dedicated management interface for remote device management and provisioning. Disabled by Default.
Enable Syslog Through Management Interface	Allows the activation of syslog communication exclusively through the dedicated management interface for remote logging and monitoring of device events. Disabled by Default.
Enable Radius Through Management Interface	Configure routing so that Radius functionality is accessible through management. Disabled by default.
Management Interface IPv4 Address	
802.1Q/VLAN Tag	Allows the tagging of network packets with VLAN information to segment and prioritize network traffic. The Valid Range is 0-4094, Default value is 0.
802.1p priority value	Assigns a priority value to network packets for Quality of Service (QoS) and traffic prioritization purposes. The Valid Range is 0-7, Default value is 0.
IPv4 address type	Configures the address type of the management network port, the options are: Dynamically configured by DHCP. Statically configured as.
IP Address	Configure the IPv4 address of the management network port.
Subnet Mask	Defines the subnet mask.
Default Router	Defines default gateway.
DNS Server 1	Defines DNS server 1 address.
DNS Server 2	Defines DNS server 2 address.

To control the display of TR-069 settings page, configure the **P-value P22664**. Setting **P22664 = 1** displays the TR-069 settings page on the Web UI, while **P22664 = 0** hides it. The default value is **1**. This parameter is not available in the Web UI and can only be configured in the device's configuration file.

RADIUS Settings

Enable RADIUS Web Access Control	Selects whether to enable RADIUS web access control. Default is "No".
Action upon Radius Auth Server Error	Select the RADIUS authentication server error handling method,the user can choose between: Reject Access, or Authentificate locally, the default is "Authenticate Locally".
Enable Bypass Locked State	Once enabled, RADIUS accounts can bypass the webUI locked state caused by local accounts. Disabled by default.
RADIUS Auth Protocol	Configures RADIUS authentication protocol. Users can choose the following option: PAP, CHAP, MsCHAPv1, MsCHAPv2. Default is "PAP".

RADIUS Auth Server Address	Configures RADIUS authentication server address.
RADIUS Auth Server Port	Configure RADIUS authentication server port. Valid range is 0-65535. Default port number is 1812.
RADIUS Shared Secret	Configures the shared secret key.
RADIUS VSA Vendor ID	Configure the RADIUS VSA provider ID to match the RADIUS server's configuration. The default value for Grandstream Networks Inc is 42397
RADIUS VSA Access Level Attribute	Configure the RADIUS VSA access level attributes to match the configuration of the RADIUS server. Incorrect settings will cause Radius authentication to fail.

E911/HELD

E911	
Enable E911	Enable Enhanced 911 call. Default is disabled.
E911 Emergency Numbers	A user can configure multiple emergency numbers separated with the delimiter symbol ";". The default number is 911.
Geolocation-Routing Header	If "Yes", E.911 INVITE message includes the "Geolocation-Routing" header with the value "Yes". Default is disabled.
Priority Header	If "Yes", E.911 INVITE message includes the "Priority" header with the value "emergency". Default is disabled.
Geolocation Header Format	Selects the format used for the SIP Geolocation header when sending emergency calls:Legacy - Sends the Geolocation value as plain text (e.g., Geolocation: 8476BD6B-5FDD-339D-900B-AEAB0AA0429A).RFC6442 - Sends the Geolocation value in RFC6442-compliant format, enclosed in angle brackets (e.g., Geolocation: <8476BD6B-5FDD-339D-900B-AEAB0AA0429A>).Set to "Legacy" by default.
HELD	
HELD Protocol	Configure HELD transfer protocol. HTTP or HTTPS. Default setting is HTTP.
HELD Synchronization Interval	The valid synchronization interval is between 30 to 1440 minutes. The synchronization is off when the interval is 0. The default value is 0.
HELD Location Types	Configure "locationType" element in the location request. "geodetic", "civic" and "location URI".
HELD Use LLDP Information	If set to "Yes", LLDP protocol will be used to gather information about the HT device including information about the ChassisID and PortID. Otherwise, the system defaults to using the MAC address of the gateway and phone. Default setting is "No".
HELD NAI	If 'Yes' is selected, the Network Access Identifier (NAI) is included as the device identity in location requests sent to the Location Information Server (LIS). Default setting is "No".
HELD Identity 1-10	The HELD Identity refers to Hierarchical Mobile IPv6 (HMIPv6) Enhanced Location-based Discovery (HELD) protocol. This protocol is utilized to determine the location of a mobile device when making an emergency call, such as a 911 call, over an IP-based network. You can configure up to 10 HELD identities.
HELD Identity 1-10 Value	Defines the HELD Identity value.
Location Server	
Location Server	Configures the location server (LIS) address.
Location Server Username	Configures Location Server (LIS) username.
Location Server Password	Configures Location Server (LIS) password.
Secondary Location Server	Configures secondary location server (LIS) address.

Secondary Location Server Username	Configures secondary location server (LIS) username.
Secondary Location Server Password	Configures secondary location server (LIS) password.

Network Settings Page Definitions

Ethernet Settings

Ethernet Settings	
General Settings	
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 HT801 V2/HT802 V2 unit for the changes to take effect.
IPv4	
IPv4 address type	Allows users to configure the appropriate network settings on the HT801 V2/HT802 V2 to obtain IPv4 address. Users could select Static IP or PPPoE. By default, it is set to DHCP.
Host Name (Option 12)	Specifies the name of the client. The name may or may not be with the local domain name. This field is optional but may be required by ISP. qualified
DHCP Domain	Allows user to configure DHCP domain name. This option specifies the domain name that the client should use when resolving hostnames via the Domain Name System. This field is optional.
Vendor Class ID (Option 60)	Exchanges vendor class ID by clients and servers to convey particular configuration or other identification information about a client. Default is HT8XXV2.
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.
Preferred DNS server 1-4	Specifies preferred DNS server to use when DHCP or PPPoE are set. You can set up yo 4 Preferred DNS Servers.
IPv6	
IPv6 address type	Configures the IPv6 address of the phone: Dynamically assigned via DHCP: all the field values for the static IP mode are not used (even though they are still saved in the flash memory.) The FXO Gateway acquires its IP address from the first DHCP server it discovers from the LAN it is connected.Statically configured as: configure IP address, 1st and 2nd DNS server, preferred DNS server. These fields are set to zero by default.
Static address type	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).
Static IPv6 Address	When using fully static type IPv6, enter a static IPv6 address.
IPv6 Prefix Length	Enter the static IPv6 address prefix length.
DNS Server 1	Enter DNS server 1 address.
DNS Server 2	Enter DNS server 2 addres.
Preferred DNS Server	Enter preferred DNS server address.
802.1x Settings	
802.1x Mode	Allow users to enable/disable 802.1X Mode. The modes available are:EAP_MD5EAP_TLSEAP-PEAPv0/MSCHAPv2Disabled by default.

802.1x Identity	Enter 802.1X identity information.
802.1x Password	Enter 802.1X MD5 password.
802.1x CA Certificate	Paste down the 802.1X certificate .pem file.
802.1x Client Certificate	Paste down the Client.pem certificate file containing certificate and key. Maximum allowed length is 8192.

DDNS Settings

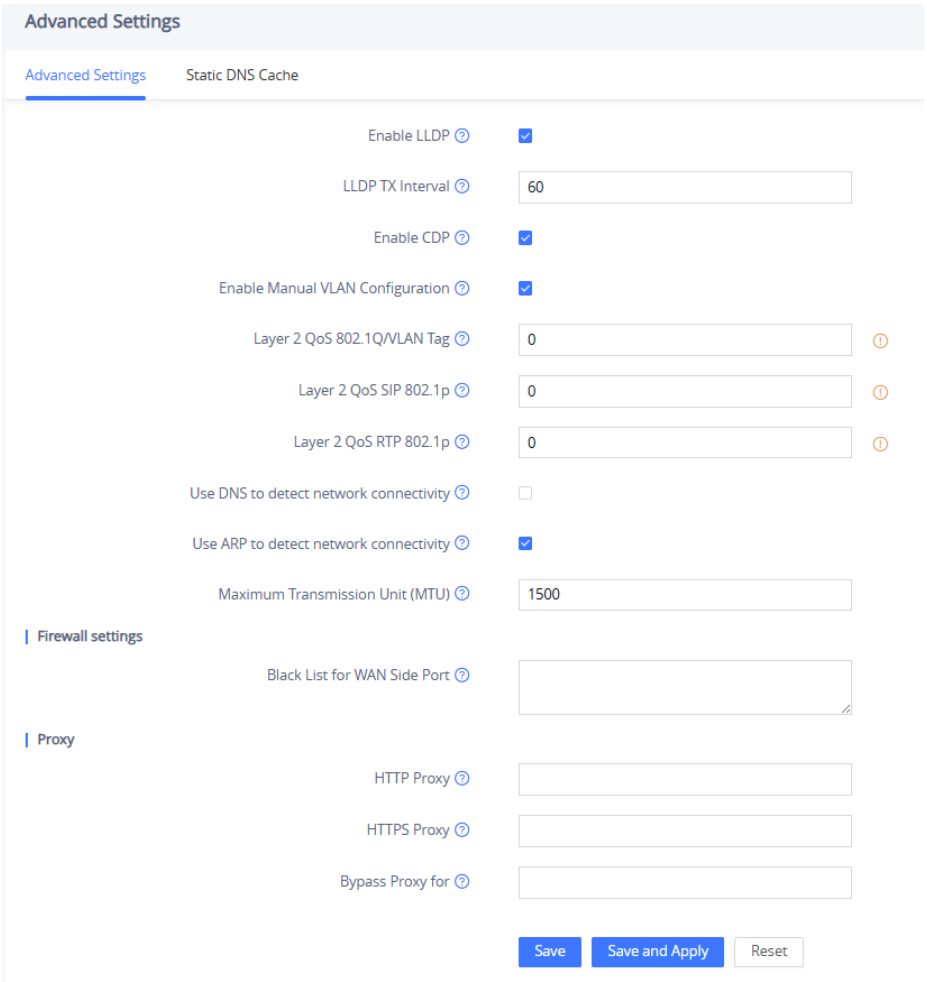
Enable DDNS	Enable DDNS function. Disabled by default
DDNS Server	Select DDNS server used, options are: dyndns.orgfreedns.afraid.orgzoneedit.comno-ip.comoray.net cloudflare.com
DDNS Username	Sets the username of the DDNS server.
DDNS Password	Sets the password of the DDNS server.
DDNS Hostname	Sets the domain name of the DDNS server.
DDNS Hash	Sets the hash value of the DDNS server.

OpenVPN(R) Settings

OpenVPN(R) Settings	
Enable OpenVPN(R)	Enables or disables the OpenVPN(R) feature. Disbaled by default.
OpenVPN(R) Server Address	Specifies the IP address or FQDN of the primary OpenVPN(R) server.
OpenVPN(R) Port	Configures the port of the primary OpenVPN(R) server. The default setting is "1194".
OpenVPN(R) Server Secondary Address	Specifies the IP address or FQDN of the secondary OpenVPN(R) server. If the primary OpenVPN(R) server is unavailable, the device automatically switches to the secondary server to maintain a secure VPN connection.
OpenVPN(R) Secondary Port	Configures the port of the secondary OpenVPN(R) server. The default setting is "1194".
Randomly Select Server	Enables or disables random server selection. When enabled, OpenVPN randomly selects a configured server when initiating a connection. When disabled, servers are used in the order in which they are configured. Disabled by default.
OpenVPN(R) Interface type	Specifies the Interface type of OpenVPN(R). The available options are: TAP or TUN. The default setting is "TUN".
OpenVPN(R) Transport	Specifies the Transport Type of OpenVPN(R). The available options are: UDP or TCP. The default setting is "UDP".
Enable OpenVPN(R) LZO Compression	Enable or disables OpenVPN(R) LZO Compression. Enabled by default.
OpenVPN(R) Encryption	Select the OpenVPN(R) Encryption algorithm. The available options are: BF-CBC 128 bit default key (variable) DES-CBC 64 bit default key (fixed)RC2-CBC 128 bit default key (variable)DES-EDE-CBC 128 bit default key (fixed)DES-EDE3-CBC 192 bit default key (fixed)DESX-CBC 192 bit default key (fixed)RC2-40-CBC 40 bit default key (variable)CAST5-CBC 128 bit default key (variable)RC2-64-CBC 64 bit default key (variable)AES-128-CBC 128 bit default key (fixed)AES-192-CBC 192 bit default key (fixed)AES-256-CBC 256 bit default key (fixed)AES-128-GCM 128 bit default key (fixed)AES-256-GCM 256 bit default key (fixed)CHACHA20-POLY1305 256 bit default key (fixed)The default setting is "BF-CBC 128 bit default key".
OpenVPN(R) Diges	Select the OpenVPN(R) Digest. Default is SHA1.
OpenVPN(R) Username	Configure the OpenVPN(R) username.

OpenVPN(R) Password	Configure the OpenVPN(R) password.
OpenVPN(R) CA	Specifies the OpenVPN(R) CA. The maximum length is 8192.
OpenVPN(R) Certificate	Specifies the OpenVPN(R) Certificate. The maximum length is 8192.
OpenVPN(R) Client Key	Specifies the Client Key. The maximum length is 8192.
OpenVPN(R) Client Key Password	Configures the OpenVPN(R) Client Key Password. The maximum length is 64.
OpenVPN(R) TLS Key	Specifies the static TLS key used by TLS-Auth or TLS-Crypt to authenticate or encrypt the OpenVPN control channel.
OpenVPN(R) TLS Key Type	Selects how the static TLS key is used to protect the OpenVPN control channel. The available options are: none: No static TLS key is used. The VPN relies only on certificate-based TLS authentication.TLS-Auth: Uses the static TLS key for HMAC authentication of the control channel, helping protect against unauthorized or malformed packets, including port scanning and certain denial-of-service (DoS) attacks. TLS-Crypt: Uses the static TLS key to both authenticate and encrypt the control channel, providing greater privacy and protection than TLS-Auth.The default setting is "none".
OpenVPN(R) Additional Options	Defines additional OpenVPN configuration parameters that are appended to the OpenVPN configuration file. Parameters must be separated by semicolons (;). Use this option with caution. Ensure the specified parameters are supported by OpenVPN and do not unintentionally override other configured settings. Example: connect-retry 5;tls-client;keepalive 10 60

Advanced Settings



Advanced Settings	
Enable LLDP	Controls the LLDP (Link Layer Discovery Protocol) service. The default setting is "Enabled".

LLDP TX Interval	Configures LLDP TX Interval (in seconds). Valid range is 1 to 3600. The Default value is 60.
Enable CDP	Enables/Disables CDP "Cisco Discovery Protocol". The default setting is "Enabled".
Enable Manual VLAN Configuration	Allows the device to keep the VLAN configuration without applying it.
Layer 2 QoS 802.1Q/VLAN Tag	Assigns the VLAN Tag of the Layer 2 QoS packets that can be used to identify and distinguish different VLANs. Valid range is 0 to 4094. The default value is 0.
Layer 2 QoS SIP 802.1p	Prioritizes SIP traffic at Layer 2 using the 802.1p standard for Quality of Service (QoS). The valid range 0-7. The default value is 0.
Layer 2 QoS RTP 802.1p	Prioritizes RTP (Real-time Transport Protocol) traffic at Layer 2 using the 802.1p standard for Quality of Service (QoS). The valid range 0-7. The default value is 0.
Use DNS to detect network connectivity	Use DNS to detect WAN side network issues. The default setting is "No".
Use ARP to detect network connectivity	Use ARP to check network connection. The default value is "Yes".
Maximum Transmission Unit (MTU)	Configures the network Maximum Transmission Unit (MTU) in bytes, which defines the largest packet size that can be transmitted. The default is 1500 bytes, with a valid range of 576-1500 bytes.
Firewall settings	
Black List for WAN Side Port	Ports on the blocklist will be disabled on the device.
Proxy	
HTTP Proxy	Specifies the HTTP proxy URL that the phone uses to send HTTP traffic. The proxy server acts as an intermediary, routing packets from the phone to their destination.
HTTPS Proxy	Specifies the HTTPS proxy URL that the phone uses to send traffic. The proxy server acts as an intermediary, routing encrypted packets from the phone to their destination.
Bypass Proxy for	Specifies URL that should be accessed directly instead of through the configured proxy. Traffic matching this entry bypasses the proxy server.
Static DNS Cache	
NAPTR	
NAPTR DNS Cache Name	The domain name to which this resource record refers. Maximum length is 512 characters.
NAPTR DNS Cache Time Interval (s)	(Time to Live) refers to the time interval that the record may be cached before the source of the information should be consulted again. Valid range is from 300 to 65535. Default value is "300".
NAPTR DNS Cache Order	Specifies the order in which the NAPTR record need to be processed, low numbers are processed before high numbers. Valid range is from 0 to 65535. Default value is "0".
NAPTR DNS Cache Preference	Refers to the order in which NAPTR records with the same "Order" values need to be processed. records are processed from lower preference numbers to higher preference numbers. Valid range is from 0 to 65535. Default value is "0".
NAPTR DNS Cache Replacement	This parameter contains the next FQDN to query for NAPTR, SRV, or address records. Maximum length is 64 characters.
NAPTR DNS Cache Service	Specifies the services available in this domain. The replacement field is used to get to this service. It can also specify the protocol used to communicate with the server that offers this service. In SIP, three services are defined along with their resolution services (resolution services are defined after the "+" sign):"SIPS+D2T": Secure SIP, TLS over TCP."SIP+D2T": SIP over TCP."SIPS+D2S": Secure SIP, TLS over SCTP."SIP+D2S": SIP over SCTP."SIP+D2U":SIP over UDP.Maximum length is 64 characters and default value is "SIP+D2U".
SRV	

SRV DNS Cache Name	The domain name of the wanted service. Maximum supported length is 512 characters.
SRV DNS Cache Time Interval (s)	(Time to Live) refers to the time interval that the record may be cached before the source of the information should be consulted again. Valid range is from 300 to 65535. Default value is "300".
SRV DNS Cache Priority	Enables to prioritize target hosts that support the given service. Lowest number will get the highest priority. Target hosts with the same priority will be tried in an order defined by the weight field. Valid range is from 0 to 65535. Default value is "0".
SRV DNS Cache Weight	Specifies the weight for entries with the same priority. Larger weight will get higher priority. Valid range is from 0 to 65535. Default value is "0".
SRV DNS Cache Target	The domain name of the target host. Maximum supported length is 64 characters.
SRV DNS Cache Port	The port on the target host of the specified service. The valid range is 0-65535. Default value is "0".
A	
A DNS Cache Name	The domain name to which the record refers. Maximum supported length is 512 characters.
A DNS Cache Time Interval (s)	(Time to Live) refers to the time interval that the record may be cached before the source of the information should be consulted again. Valid range is from 300 to 65535. Default value is "300".
A DNS Cache IP Address	The IP address associated with the DNS record name.

Maintenance Page Definitions

Upgrade

Upgrade -> Firmware	
Upgrade via Manual Upload	
Firmware	Uploads the .bin firmware file.
Upgrade via Network	
Upgrade Via	Selects firmware upgrade/provisioning method: TFTP, HTTP, HTTPS, FTP or FTPS. Default is HTTPS.
Firmware Server Path	Sets IP address or FQDN of firmware server. The URL of the server that hosts the firmware release.Note: You can specify the protocol in the Firmware Server Path. (example: https://192.168.5.120), this will bypass the "Upgrade Via" method. used
HTTP/HTTPS/FTP/FTPS User Name	Enters username to authenticate with HTTP/HTTPS FTP/FTPS server.
HTTP/HTTPS/FTP/FTPS Password	Enters password to authenticate with HTTP/HTTPS FTP/FTPS server.
Firmware File Prefix	Checks if firmware file is with matching prefix before downloading it. This field enables user to store different versions of firmware files in one directory on the firmware server.
Firmware File Postfix	Checks if firmware file is with matching postfix before downloading it. This field enables user to store different versions of firmware files in one directory on the firmware server.
Upgrade -> Config File	
Manual Configuration	
Upload Configuration	Uploads the configuration file in .txt or .xml formats.
Restore From Backup Configuration	Restore From Backup Configuration.
Download Device Configuration	Downloads Device Configuration in .txt format.

Download Device XML Configuration	Downloads Device Configuration in .xml format (The P-values will be sorted in ascending order).
Export Backup Configuration	Export Backup Configuration in .xml format.
Configure via Network	
Config Upgrade via	Selects Config file upload method: TFTP, HTTP, HTTPS, FTP or FTPS. Default is HTTPS.
Config Server Path	Sets IP address or FQDN of configuration server. The URL of the server that hosts the configuration file to provision HT80xV2. Note: You can specify the protocol used in the Config Server Path. (example: https://192.168.5.120), this will bypass the "Upgrade Via" method.
HTTP/HTTPS/FTP/FTPS User Name	Enters username to authenticate with HTTP/HTTPS FTP/FTPS server.
HTTP/HTTPS/FTP/FTPS Password	Enters password to authenticate with HTTP/HTTPS FTP/FTPS server.
XML Config File Password	Decrypts XML configuration file when encrypted. The password used for encrypting the XML configuration file using OpenSSL.
Config File Prefix	Checks if configuration files are with matching prefix before downloading them. This field enables 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. This field enables user to store different configuration files in one directory on the provisioning server.
Upgrade -> Provision	
Allow DHCP Option 66 or 160 to override server	Obtains configuration and upgrade server's information using options 66 from DHCP server.Notes:If DHCP Option 66 is enabled, the HT80x V2 will attempt downloading the firmware file from the server URL provided by DHCP, even though Config Server Path is left blank. The server URL provided by DHCP can include authentication credentials using following format: "username:password@Provisioning_Server_IP"DHCP Option 160 is prioritized over Option 66 to align behavior with other Grandstream devices.
Auto Provision	When enabled, the device sends SUBSCRIBE in multicast mode. If 3CX, UCM and other IPPBX servers are used as SIP servers, the device can be automatically configured. Enabled by default.
Enable using tags in URL	Allows users to configure variables on the configuration server path to differentiate the directories on the server. Disabled by default.
Additional Override DHCP option	Configures additional DHCP Option to be used for firmware/config server instead of the configured firmware/config server or the server from DHCP Option 66 and 160. This option requires a reboot to take effect.
Automatic Upgrade	
Automatic Upgrade	Specifies when the firmware upgrade process will be initiated; there are 4 options: Note: The HT80xV2 will only do an upgrade once at boot up. Check every X minutes: User needs to specify a period in minutes. Check every day: User needs to specify the start hour and the end hour of the day (0-23). Check every week: User needs to specify "Day of the week (0-6)". (Day of week is starting from Sunday). Default is No.
Randomized Automatic Upgrade	Randomized Automatic Upgrade within the range of hours of the day or postpone the upgrade every X minute(s) by random 1 to X minute(s).
Firmware Upgrade and Provisioning	Defines when the device checks the server for new firmware and configuration files:Always Check for New Firmware at Boot up - The device checks for updates every time it restarts.Check New Firmware only when F/W pre/suffix changes - The device checks for updates only if the configured firmware prefix/suffix has been modified.Always Skip the Firmware Check - The device will not check for firmware updates at boot.Default value is set to "Always Skip the Firmware Check".
Config Provision	

Download and Process All Available Config Files	This feature allows users to download and process all available config files. By default, the device will provision the first available config in the order of cfgMAC > cfgMAC.xml > cfgMODEL.xml > and cfg.xml (corresponding to device-specific, model-specific, and global configs). If this option is enabled, the device will inverse the downloading process to cfg.xml > cfgMODEL.xml > cfgMAC.bin > cfgMAC.xml and add cfgMAC_override.xml. The following files will override the files that have already been loaded and processed.The default value is "No".
Config Provision Order	Administrators can define the preferred order in which the device checks for configuration files during startup. The system will sequentially attempt to load the listed files, and try to apply the first successfully retrieved configuration, the files available are:cfgMAC.xml - Configuration file specific to the device's MAC address, using XML format.cfgMAC - Binary file configuration file for a specific MAC address.cfgMODEL.xml - Configuration file specific to the device model.cfg.xml - General configuration file applicable to multiple devices.cfgMAC_override.xml - Override configuration file for a specific MAC address, taking precedence over other MAC-based files.cfgdhcpopt67.xml - Configuration file retrieved via DHCP Option 67, often used in auto-provisioning environments.
Upgrade -> Advanced Settings	
Verify host when using HTTPS	Enables / disables the host verification when using HTTPS. Disabled by default.
Configuration File Types Allowed	Allows users to configure provision configuration file type in xml file only or all file types.
Enable SIP NOTIFY Authentication	The device will challenge NOTIFY with 401 when set to "Yes". Set to "Yes" by default.
Validate Server Certificates	Configures whether to validate the server certificate when downloading the firmware/config file. If set to "Yes", the device will download the firmware/config file only after the server is validated. Disabled by default Note: This option requires a reboot to take effect.

System Diagnosis

Syslog	
Syslog Server	URL or IP address of the syslog server.
Syslog Level	Select the HT8xx to report the log level. Default is NONE. The level is one of EXTRA DEBUG, DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: 1. product model/version on boot up (INFO level) 2. NAT related info (INFO level) 3. sent or received SIP message (DEBUG level) 4. SIP message summary (INFO level) 5. inbound and outbound calls (INFO level) 6. registration status change (INFO level) 7. negotiated codec (INFO level) 8. Ethernet link up (INFO level) 9. SLIC chip exception (WARNING and ERROR levels) 10. memory exception (ERROR level) extra syslog style (EXTRA DEBUG level)
Syslog Protocol	Allow encrypted SSL/TLS transmission to the syslog server if SSL/TLS is selected, the default value is UDP.Notes: The validity of the server CA certificate will be verified.Security-related debug logs are now included in Syslog when debug-level logging is enabled. These logs may contain sensitive diagnostic details about authentication, TLS, and other security events. Ensure Syslog collection is secured appropriately.
Syslog Header Format	Determines how the headers in syslog messages are formatted, the options are: Default: Uses the software's or system's standard header formatRFC3164 Compliant: Adheres to the RFC 3164 standard for syslog headers,
Send SIP Log	Sets whether to include SIP logs in the system logDefault value is "No".
Debug	
Ethernet Capture	Enables the capturing and analysis of Ethernet network traffic for troubleshooting and monitoring purposes.

With secret key information	Allows users to make packet capture including the secret key to decrypt the captured TLS packets. Default value is Yes.
Port Record	Allows the recording of port-level call information, such as call duration, caller ID, and call status, for monitoring and logging purposes.
Core Dump	Provides generated core dump file if unit malfunctions. Clean will be displayed if no issues.
GR909	
Test To Run	Selects the type of test that wll be performed for the specified line, the following tests can be performed: Hazardous Potential TestForeign Electromotive Forces TestResistive Faults TestReceiver Offhook TestRinger Equivalent Number Test
FXS Line	Defines the FXS line under test.
Run Interval	Defines the run Interval in seconds, the default value is 120 seconds, the default range is 0-604800 seconds.

File Management

CDR Records	Displays a list of the last 1000 call records, the ionformation included in the records are the following: UserID ToNumber FromNumber StartTime StartTalkTime EndTime Duration State Direction Clicking the "Download" button will download the last 1000 call records. Clicking the "Delete" button will delete the last 1000 call records.
SIP Messages	The SIP messages tab displays a list of SIP headers that are captured after each triggered call, the information displayed on the tab are similar to the information that can be captured using the wireshark tool, and filtering by SIP protocol. Clicking the "Download" icon will download the whole log of the captured traces. Clicking "Delete" will delete all the captured traces. This tool can be useful for network administrators that want to treoubleshoot the call flows without having to use an external tool, for analyzing the network Note: The SIP PUBLISH method (RFC 6035) enables the gateway to send status updates and event notifications to a SIP server or PBX. This allows for real-time monitoring, presence tracking, and improved call management in VoIP networks.

Device Manager

Reboot	
Automatic restart	This option triggeres an automatic reboot of the unit, the available options are: No Reboot everday: you can specify the reboot hour Reboot everyweek: you can specify the day of the week, and the hour of the day, for the reboot to take place Reboot every month: you can specify the day of the month, and the hour of the day, for the reboot to take place
Restore Factory	
Reset Type	Gives the administrator the option to restore the default configuration on the HT801 V2/HT802 V2. There are 3 types of factory reset: ISP Data Reset: All ISP (Internet Service Provider) configurations that may affect the IP address will be reset (including WAN static IP). VoIP Data Reset: All VoIP-related configurations (mainly everything located on FXS page). Full Reset: Both VoIP and ISP-related configuration at the same time. Note: After choosing the reset type, you will have to click the reset button for it to take effect.

Port Settings Page Definitions

FXS PORT x -> General Settings
Account Registration

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 (Supports both IPv4 and IPv6 addresses) or domain name provided by VoIP service provider. (For example: sip.mycompany.com, IPv4:192.168.5.170, or IPv6: fe80::20b:82ff:fe75:211d). This is the primary SIP server used to send/receive SIP messages from/to HT801 V2/HT802 V2.
Failover SIP Server	Defines failover SIP server IP address (Supports both IPv4 and IPv6 addresses) or domain name provided by VoIP service provider. (For example: sip.mycompany.com, IPv4:192.168.5.170, or IPv6 fe80::20b:82ff:fe75:211d:). This server will be used if 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 (Supports both IPv4 and IPv6 addresses) or domain name of outbound Proxy, or media gateway, or session border controller. (For example: proxy.myprovider.com, IPv4: 192.168.5.170, or IPv6: fe80::20b:82ff:fe75:211d). It's Used by HT801 V2/HT802 V2 firewall or NAT penetration in different network environments. If symmetric NAT is detected, STUN will not work and only outbound for proxy can correct the problem
Backup Outbound Proxy	Configures the backup outbound proxy to be used when the "Outbound Proxy" registration fails. (For example: proxy.myprovider.com, or IP address, if any: IPv4: 192.168.5.170/ IPv6: fe80::20b:82ff:fe75:211d). By default, this field is left empty.
Prefer Primary Outbound Proxy	If the user configures this option to "Yes", when the registration expires, the device will re-register via primary outbound proxy. By default, this option is disabled.
From Domain	Allows users to add the actual domain name, it will override the from header.This is an optional configuration.
Allow DHCP Option 120 (override SIP server)	Configures the HT801 V2/HT802 V2 to collect SIP server address from DHCP option 120. Default is No.
SIP User ID	Defines The user part of a SIP address
Authenticate ID	Defines the Authenticate ID of the User
Authenticate Password	Defines the Authenticate password
Name	Defines the name of the user
Tel URI	If the device has an assigned PSTN telephone number, this field should be set to "User=phone". A "User=phone" parameter will be attached to the Request-Line and "To" header in the SIP request to indicate the E.164 number. If set to "Enabled", "tel:" will be used instead of "sip:" in the SIP request. Disabled by default.
Network Settings	
Layer 3 QoS SIP DSCP	the Diff-Serv value for SIP packets in decimal, the range is 0-63, default value is 26
Layer 3 QoS RTP DSCP	Diff-Serv valuefor RTP packets in decimal, the range is 0-63, default value is 46
DNS Mode	Selects DNS mode to use for the client to look up server. One mode can be chosen. A Record (Default): 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 the SIP server is configured as domain name, device will not send DNS queries, but will use "Primary IP" or "Backup IP" to send SIP message if at least one of them is not empty. It will try to use "Primary IP" first, after 3 tries without any response, it will switch to "Backup IP 1", then "Backup IP 2", and then it will switch back to "Primary IP" after 3 retries.

DNS SRV Failover Mode	Configure the preferred IP mode when DNS Mode is SRV or NAPTR/SRV. * Default SIP request will always be sent to the address with the top priority based on the SRV query result, even if this address is different from the registered IP address. * Saved one until DNS TTL SIP request will always be sent to the registered IP address until DNS TTL expires or registered IP address is unreachable. * on until no response SIP request will always be sent to the registered IP address only until registered IP address is unreachable. * Failback follows failback expiration time: the primary server regains control only after the failback expiration time, allowing the secondary server to handle requests until then.
Failback Timer	When the primary SBC is up, device will send SIP requests to the primary SBC. If at any point device fails over to the secondary SBC, the SIP requests will stay on the failover SBC for the duration of the failback timer. When the timer expires, device will send SIP requests to the primary SBC, (in minutes. Default is 60 minutes, max 45 days).
Maximum Number of SIP Request Retries	This feature allows user to configure the number of SIP retries before failover occurs. (between 1 and 10, default is 2).
Register Before DNS SRV Failover	This feature is used to control whether the device needs to initiate a new registration request (following existing DNS SRV fail-over mode) first and then direct the non-registration SIP request (INVITE) to the new successfully registered server or not.
Primary IP	The main IP address used for communication and control.
Backup IP1	The secondary IP address used as a backup for communication in case the Primary IP fails.
Backup IP2	The third IP address used as an additional backup for communication in case both the Primary IP and Backup IP1 fail.
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, Auto, VPN. No: NAT traversal is disabled; the phone doesn't attempt to handle NAT issues.Keep-alive: Sends periodic packets to keep the NAT mapping open for incoming calls.STUN: Uses a STUN server to discover the public IP address and port, helping traverse NAT.UPnP: Uses Universal Plug and Play to automatically configure the router to allow SIP traffic through NAT.Auto: Automatically selects the best NAT traversal method based on the network environment.VPN: Routes all SIP traffic through a secure Virtual Private Network, bypassing NAT issues.The default setting is "No".
Use NAT IP	Defines NAT IP address used in SIP/SDP messages. It should only be used if required by ITSP.
Proxy-Require	Determines a SIP Extension to notify the SIP server that the HT801 V2/HT802 V2 is behind a NAT/Firewall.
FXS PORT x -> SIP Settings	
SIP Basic Settings	
SIP Registration	Controls whether the HT801 V2/HT802 V2 needs to send REGISTER messages to the proxy server. The default setting is Yes.
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.

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 "Expires=0" parameter to the SIP server. This will unregister the SIP account under the concerned FXS page.Unregister on reboot option can be set to "No", "All" or "Instance".Set to "No": If the "Unregister on reboot" option is set to "No", it means that the SIP user's information will not be cleared when the device reboots. In other words, the SIP account will remain registered with the server even after the device is rebooted.Set to "All": If the "Unregister on reboot" option is set to "All", it means that all SIP accounts associated with the device will be unregistered when the device reboots. This option clears the SIP user's information for all the FXS ports on the device.Set to "Instance": If the "Unregister on reboot" option is set to "Instance", it means that only the SIP account associated with the concerned FXS port will be unregistered when the device reboots. This option clears the SIP user's information for only the specific FXS port that is affected by the reboot.Default value is set to "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).
Re-register 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.
Use Random SIP Registration Failure Retry Wait Time	When enabled, the waiting time to resend a SIP REGISTER request to the SIP server in case of registration failure will be a random number in the following range. Default setting is disabled
Random SIP Registration Failure Retry Wait Time Range	Allows users to set a wait time (in seconds) before attempting to send a SIP registration after each failure. The wait time will be chosen randomly within this range. Valid values range from 60 to 600 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.
Port Voltage Off upon no SIP Registration or SIP Registration Failure	Cuts off voltage to the port if there is no SIP registration or if SIP registration fails, preventing unwanted calls. (in minutes. Between 0-60, default is 0. 0 means port voltage is never turned off)
Delay Time of Port Voltage Off Timer Since Boot	Sets the delay time after boot before the FXO port voltage turns off. It controls the timing for powering down FXO ports on the gateway. The value is in minutes. Between 0-60, default is 0
Enable SIP OPTIONS/NOTIFY Keep Alive	Enables SIP OPTIONS or SIP NOTIFY to track account registration status so the ATA will send periodic OPTIONS/NOTIFY message to server to track the connection status with the server. Default setting is No.
SIP OPTIONS/NOTIFY Keep Alive Interval	Configures the time interval when the ATA send OPTIONS or NOTIFY message to SIP server. The default setting is 30 seconds, which means the ATA will send an OPTIONS/NOTIFY message to the server every 30 seconds. The default range is 1-64800.
SIP OPTIONS/NOTIFY 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.
SUBSCRIBE for MWI	Sends SUBSCRIBE periodically (depends on "Register Expiration" parameter) for message waiting indication. Default is No.
Subscribe Retry Wait Time upon 403 Forbidden	Configures (in minutes) how long it takes to send a SUBSCRIBE request after a "403 Forbiden" response. Valid range is 0-10080. Default is 0, max 7 days.
Local SIP Port	Defines local port to use by the HT801 V2/HT802 V2 for listening and transmitting SIP packets. Default value is 5060

Use Random SIP Port	Controls whether to use configured or random SIP ports. This is usually necessary when multiple HT801 V2/HT802 V2 are behind the same NAT. The default 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.
Remove OBP from Route Header	Removes outbound proxy info in "Route" header when sending SIP packets. Default setting is No.
Support SIP Instance ID	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
Support outbound	Enables/disables outbound support.
Support GRUU	Enables/disables GRUU support.
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. Default is No.
Use Request Routing ID in SIP INVITE Header	This options enables the inclusion of a routing ID in the SIP INVITE header for directing the request to a specific endpoint or route. Note: the HT supports multiple DIDs per FXS port, this allows single port to receive calls for multiple numbers based on the configured Routing ID or other call-matching settings.
Use Privacy Header	Determines if the "Privacy header" will be presented in the SIP INVITE message and if it includes the caller info in this header. If set to Default, it will add Privacy header unless special feature is Telkom SA or CBCOM. Default is Default.
Use P-Preferred-Identity Header	Specifies if the P-Preferred-Identity Header will be presented in the SIP INVITE message. If set to "default", the P-Preferred-Identity Header will be omitted in SIP INVITE message when Telkom SA or CBCOM is active. If set to "Yes", the P-Preferred- Identity Header will always be presented. If set to "No", it will be omitted. Default setting is: Default.
Use P-Access-Network-Info Header	With this feature enabled, device will populate the WAN access node with IEE-802.11a, IEE-802.11b in P-Access-Network-Info SIP header. Default is Yes.
Use P-Emergency-Info Header	This feature support of IEEE-48-addr and IEEE-EUI-64 in SIP header for emergency calls. Default is Yes.
Use P-Asserted-Identity Header	Enables or disables sending the P-Asserted-Identity Header in the SIP INVITE messages.Disabled by default.
Use P-Early-Media Header	Enables or disables sending the P-Early-Media Header in the SIP INVITE messages.Disabled by default.
SIP REGISTER Contact Header Uses	Configures whether the SIP REGISTER Contact Header Uses LAN address or WAN address. Default setting is "LAN Address".
SIP REGISTER Contact Header Uses	SIP REGISTER Contact Header Uses LAN address or WAN address By default, it uses LAN Address
SIP T1 Timeout	Defines T1 timeout value. It is an estimate of the round-trip time between the client and server transactions. For example, HT801 V2/HT802 V2 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 is defined by T2. The 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 the T2 value. The default is 4 seconds.

SIP Timer B	Configures SIP Timer B defined in RFC3261.0-128 sec. Default 0, which means using 64*T1.When this SIP Timer is set to 0, its value is calculated as:SIP Timer B = SIP T1 Timeout x 64E.g., If the SIP T1 Timeout is set to its default value of 0.5 seconds, then:SIP Timer B = 0.5 x 64 = 32 seconds
SIP Timer D	Configures SIP Timer D defined in RFC3261. 0-64 seconds. Default is 0.
SIP Timer F	Configures SIP Timer F defined in RFC3261.0-128 sec. Default 0, which means using 64*T1.When this SIP Timer is set to 0, its value is calculated as:SIP Timer F = SIP T1 Timeout x 64E.g., If the SIP T1 Timeout is set to its default value of 0.5 seconds, then:SIP Timer F = 0.5 x 64 = 32 seconds
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.
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.
Enable Call Waiting Alert-Info in 180 Ringing Response	Activates the Call Waiting feature by including Call Waiting Alert-Info in the 180 Ringing response sent to the caller during an incoming call. Default value is no.
SIP User-Agent	This feature allows users to configure SIP User Agent. If not configured, device will use the default User Agent header.
SIP User-Agent Postfix	Configures the SIP User-Agent Postfix
Add MAC in User-Agent	This feature allows users to configure "User-Agent" in the SIP header field, when this feature is set to"No", "User-Agent" does not carry a MAC, when this feature is set "Yes except REGISTER", "User-Agent" in REGISTERSIP header field does not carry MAC but other SIP packet header fields carries MAC, when this feature is set to "Yes to all SIP", "User-Agent" in the SIP packet header field will carries the MAC.
Use MAC Header	This feature allows users to configure MAC Header in the SIP packet header field, when this feature is set to "No" ,the MAC header field is not carried in the SIP packet header field, when this feature is set to "REGISTER Only", the register packet header field carries the MAC header field but the remaining SIP packets do not carry MAC header fields, when this feature is set to "Yes to all SIP", the MAC header field is carried in the SIP data packet header field.
CSeq Tracking Mode	Defines how the device tracks SIP CSeq values.per-call - CSeq is tracked independently for each call.per-dialog - CSeq is tracked across the entire SIP dialog and increments within the same dialog.Use per-dialog if the SIP platform expects continuous CSeq sequencing within a dialog (per RFC behavior); per-call is typically more tolerant with certain SIP servers and proxies. It is set to "per-call" by default.
Session Timer	
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. Valid range is 90-64800 seconds. 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. Valid range is 90-64800 seconds. 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 device 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 UAC: UAS: The device 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.
When To Restart Session After Re-INVITE received	Allows users to delay posting Media Change Event, it can be set to "Immediately" or to "After replying 200OK" The default value is "Immediately".
FXS PORT x -> Codec Settings	
DTMF Settings	
Preferred DTMF	Sorts DTMF methods (in-audio, via RTP (RFC2833) or via SIP INFO) by priority.
Force DTMF to be sent via SIP INFO simultaneously	Configures whether or not to force DTMF to be sent via SIP INFO simultaneously. Default setting is No.
Inband DTMF Duration	Allows users to config the Inband DTMF Duration and inter-duration. The default Duration is 100 ms. Valid range: 40-2000 ms. The default Inter-duration is 50 ms. Valid range: 40-2000 ms.
DTMF Payload Type	Defines payload type for DTMF using RFC2833.
Enable Multiple Sampling Rates in SDP telephone-event	This option allows the device to advertise multiple supported sampling rates for DTMF events in the SDP (Session Description Protocol), which enhances compatibility with different VoIP systems by ensuring proper DTMF signal interpretation across varying codec sampling rates. When the option is enabled, the HT includes multiple supported sampling rates for DTMF (telephone-event) in the SDP negotiation, which allows better interoperability with VoIP systems using different codec rates. When disabled, only a single default sampling rate is advertised, which may lead to DTMF tone mismatches if the remote system expects a different rate.
Inband DTMF Tx Gain	Adjusts the transmit gain for inband Dual-Tone Multi-Frequency (DTMF) tones during call signaling. The Valid range is -12-12 db, default is 0
DSP DTMF Detector Duration Threshold	Allows users to config the DSP DTMF Detector Duration Threshold. The default Duration is 30 ms. Valid range: 20-2000 ms. The default Inter-duration is 30 ms. Valid range: 20-2000 ms.
DSP DTMF Detector Min Level	determines the minimum level of DTMF tones that the DSP DTMF detector can reliably detect, range is -45-0 dBm, default is -25
DSP DTMF Detector Snr	Sets the signal-to-noise ratio threshold for the DSP DTMF detector, determining the sensitivity to distinguish DTMF tones from background noise in the audio signal. Range: 0-12, default is 1
DSP DTMF Detector Deviation	Specifies the allowed frequency deviation from standard DTMF tones that the DSP DTMF detector can accommodate. Range: 0-25, default is 25
DSP DTMF Detector Twist	Defines the maximum deviation from ideal tone frequencies allowed for reliable DTMF detection by the DSP. Range: 0-12dB, default is 5
Enable DTMF Negotiation	Uses DTMF order without negotiation. Default is No.
RFC2833 Events Count	This feature allows users to customize the count of RFC2833 events. Default is 8.
RFC2833 End Events Count	This feature allows users to customize the count of RFC2833 end events. Default is 3.
Vocoder Settings	

Vocoder Settings (in listed order)	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.
G.723 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.
Enable OPUS Stereo in SDP	Enabkes OPUS stereo in SDP. Default is Yes.
iLBC Payload Type	Determines payload type for iLBC. Valid range is between 96 and 127. Default is 97.
Opus Payload Type	Determines payload type for OPUS. Valid range is between 96 and 127. Default is 123.
Enable Audio RED with FEC	Enables the use of audio redundancy (RED) with forward error correction (FEC) to enhance audio quality and resilience against packet loss in real-time communication systems such as VoIP
Audio FEC Payload Type	Specifies the payload type used for transmitting audio forward error correction (FEC) packets in a VoIP system.
Audio RED Payload Type	Defines the payload type assigned to audio redundancy (RED) packets in a VoIP system,
Silence Suppression	If set to "Yes", when the device detects that there is no voice flow in the call, the device will send out a small number of VAD packets (instead of voice packets) Disabled by default.
Use First Matching Vocoder in 200OK SDP	When enabled, the gateway will use the first matching code in 200OK SDP to make calls Disabled by default.
Fax Mode	Specifies the fax mode: T.38 (Auto Detect) FoIP by default, or Pass-Through. If using Pass-through mode, select preference codec as PCMU or PCMA.
T.38 Max Bit Rate	Selects the maximum T.38 bit rate. Lowering the maximum fax bit rate may help improve the fax success rate. Only effective when the fax machine is fax receipt. Available options are: 4800bps, 9600bps, 14400bps. Default value is 9600bps.
Enable T.38 TEP	Enables or disables T.38 Talker Echo Protection (TEP).When enabled, TEP is applied during T.38 fax calls to help reduce echo interference. However, it may affect transmission reliability on some connections. Enable this feature only if echo is impacting fax quality.Disabled by default.
Re-INVITE After Fax Tone Detected	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
Re-INVITE Upon CNG Count	This feature enables users to initiate a re-invite request for fax transmission when the fax machine is the sender. The feature is controlled by an adjustable setting, where a value of 0 disables the feature, and a value of 1 or greater allows the ATA to automatically initiate the re-invite request once the CNG (Calling Tone) count threshold is met. The valid setting range for this feature is between 0 and 6. The default setting is 0.
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).
Crypto Life Time	Adds crypto life time header to SRTP packets. Default is Yes.
RTP Setting	
Local RTP Port	Defines the local RTP-RTCP port pair the HT801 V2/HT802 V2 will listen and transmit. It is the HT801 V2/HT802 V2 RTP port for channel 0. The default value for FXS port is 5004

Use Random RTP Port	Controls whether to use configured or random RTP ports. This is usually necessary when multiple HT801 V2/HT802 V2 are behind the same NAT. The default is Yes.
Random RTP Port Range	Set Random RTP Port Range:1024-65535, Min - Max.
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.
Enable RTCP	Allows users to enable RTCP. The default setting is "Yes".
RTP/RTCP Keep Alive On Hold	Enables or disables RTP/RTCP keep-alive packets during call hold to maintain connectivity and prevent session timeouts. Disabled by default.
SRTP Mode	Selects SRTP mode to use:Disabled: All calls use standard RTP with no media encryption. Recommended when interoperability with legacy or non-secure SIP servers is required.Enabled but not forced: The ATA offers SRTP during call setup; if the peer supports it, media is encrypted. If not, it automatically falls back to RTP to keep the call working.Enabled and forced: The ATA mandates SRTP; if the remote party cannot negotiate SRTP, the call will not be established. Ensures fully encrypted audio for compliant peers.Follow SIP Transport: Automatically aligns media encryption with signaling security: when SIP uses TLS, SRTP is enabled; when SIP uses UDP or TCP, RTP is used instead.Default is DisabledSRTP uses SDP Security Description to exchange key. Please refer to the below links for more informationSDES: https://tools.ietf.org/html/rfc4568?SRTP: https://www.ietf.org/rfc/rfc3711.txt?
SRTP Key Length	Allows users to select supported SRTP Key Length. the available values are :AES 128&256 bit AES 128 bitAES 256 bit
VQ RTCP-XR Collector Name	Configures the host name of the central report collector that accepts voice quality reports contained in SIP PUBLISH messages.
VQ RTCP-XR Collector Address	Configures the IP address of the central report collector that accepts voice quality reports contained in SIP PUBLISH messages.
VQ RTCP-XR Collector Port	Configures the port of the central report collector that accepts voice quality reports contained in SIP PUBLISH messages. The default setting is "5060".
FXS PORT x -> Analog Signal Line Configuration	
Analog Signal Line Configuration	
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	Configures start and stop tone for DTMF caller ID scheme.
Polarity Reversal	Reverses the polarity upon call establishment and termination. Default is No.
Loop Current Disconnect	Allows the traditional PBX used with HT801 V2/HT802 V2 to apply this method for signaling call termination. The method initiates a short voltage drop on the line when the remote (VoIP) side disconnects an active call. The default is No.
Play Busy/Reorder Tone Before Loop Current Disconnect	Enables or disables playing the busy/reorder tone before loop current disconnect upon call fail. Disabled by default
Busy/Reorder Tone Delay Timer After Loop Current Disconnect	Configures the Busy/reorder tone delay timer after the loop current disconnects. The default setting is "25". The valid range is 0-30.
Loop Current Disconnect Duration	Configures the duration of voltage drop described in the topic above. HT801 V2/HT802 V2 support a duration range from 100 to 10000ms. The default value is 200.
Enable Pulse Dialing	Allow users to enable Pulse Dialing option under FXS Port. Default is No.
Pulse Dialing Standard	Allows users to use Swedish pulse dialing standard or New Zealand pulse dialing standard. Default is General Standard.

Enable Hook Flash	Enables the FLASH button to be used for terminating calls. Default is Yes.
Hook Flash Timing	Defines the time period when the cradle is pressed (Hook Flash) to simulate FLASH. To prevent unwanted activation of the Flash/Hold and automatic phone ring-back, adjust this time value. HT801 V2/HT802 V2 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. HT801 V2/HT802 V2 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 (LEC)	Disables the LEC per call base. Recommended for Fax/Data calls. Default is No.
Ring Frequency	Chooses an appropriate ringing frequency. Default is 20Hz.
Ring Power	Selects ringing power settings, the options are: 45Vrms, 50Vrms, and 55Vrms , the higher the value, the more powerful the ring is. Default is 45Vrms.
OnHook DC Feed Current	This feature is used to adjust DC feed current.
FXS PORT x -> Call Settings	
Dial Setting	
Off-hook Auto Dial	Configures the user ID, extension or IP address to dial automatically when offhook.
Offhook Auto-Dial Delay	Sets the delay time before automatic dialing begins after going off-hook on an FXO port. Range is 0-60 seconds, default is 0
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.

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 j. Flag T when adding a "T" at the end of the dial plan, the phone will wait for 3 seconds before dialing out. This gives users more flexibility on their plan setup. E.g. with dial plan 1XXT, phone will wait for 3 seconds to let user dial more than just 3 digits if needed. Originally the phone will dial out immediately after dialing the third digit. 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. 1. Default: Outgoing - { x+ | +x+ | *x+ | *xx*x+ } Example of a simple dial plan used in a Home/Office in the US: { ^1900x. | <=1617>[2-9]xxxxxx | 1[2-9]xx[2-9]xxxxxx | 011[2-9]x. | [3469]11 } Explanation of example rule (reading from left to right): ^1900x. - prevents dialing any number started with 1900 <=1617>[2-9]xxxxxx - allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically 1[2-9]xx[2-9]xxxxxx - allows dialing to any US/Canada Number with 11 digits length 011[2-9]x. - allows international calls starting with 011 [3469]11 - allow dialing special and emergency numbers 311, 411, 611 and 911 Note: In some cases, user wishes to dial strings such as *123 to activate voice mail or other application provided by service provider. In this case * should be predefined inside dial plan feature. An example dial plan will be: { *x+ } which allows the user to dial * followed by any length of numbers.
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.
Enable # as Redial Key	Disables # to act as Redial key. If set to "Yes" and feature "Use # as Dial Key" set to Yes, the # key will act as dial key but not as redial key. Default is Yes .
General	
RFC2543 Hold	Toggles between RFC2543 hold and RFC3261 hold. RFC2543 hold allows to disable the hold music sent to the other side, in this case IP address (0.0.0.0) it will be sent in SDP instead of the IP address of the unit . RFC3261 (a line) will play the hold music to the other side.
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.
Send Anonymous	Sets "From", "Privacy" and "P_Asserted_Identity" headers in outgoing INVITE message to "anonymous", blocking caller ID. Default is No. Note: When the caller is marked as anonymous, the current phone Caller ID will be set as "private caller", and when is marked as unavailable, the current phone caller ID is marked as "unknown caller"
Anonymous Call Rejection	Enables or disables anonymous call rejection. When enabled, anonymous calls are rejected.
Outgoing Call Duration Limit	Configures the maximum duration of outgoing calls in minutes. The default setting is "0", which disables the limit. The valid range is 0-180.
Incoming Call Duration Limit	Configures the maximum duration of incoming calls in minutes. The default setting is "0", which disables the limit. The valid range is 0-180.
Reminder Tone	
Enable Receiver Offhook Tone	Enables / disables the warning to alert that the phone has been left off-hook for an extended period of time. Default is No.
Enable Reminder Ring for On-Hold Call	Enables playing the reminder ring. Default is No.

Enable Reminder Ring for DND	This feature allows user to disable reminder ring when FXS port is on DND mode. Default is Yes.
Enable Visual MWI	Disables use of visual message waiting indicator when there is an unread voicemail message. Default is No.
Visual MWI Type	Configures Visual WMI Type of signal sent to the analog phone to make it turn the lamp ON upon receiving a Voice mail. Check the phone's manual to find out what signal is supported, FSK (default) or Neon. Note: Some phones (depending on the model of the analog phone) when this feature is set to NEON it might auto ring (short beeps) when there is a voice mail available for that FXS port where it is connected.
MWI Tone	When set to Default, device will play Stutter Dial Tone when there is voicemail, if set to Special Proceed Indication Tone, device will play the configured special proceed indication tone upon user offhook when there is voicemail
Hook Flash	
Transfer on Conference Hangup	Determines whether a call is automatically transferred to a specified destination after a conference call ends. Disabled by Default
Ringing Transfer	Allows users to trigger the transfer once the user hangs up the call, Default setting is "No".
No Answer Timeout	Defines the timeout (in seconds) before the call is forwarded on no answer. Valid range is 1 to 120.
Send Hook Flash Event	Default is No. If set to yes, flash will be sent as DTMF event.
Flash Digit Control	Enables or disables overriding the default call control settings when both channels are in use.The available options are: Yes, No and Yes with Attended Transfer Upon Onhook. When set to Yes, the following flash key sequences are available: "Flash + 1" in order to hang up the current call and resume a call that was held. "Flash + 2" in order to hold the current call and resume a call that was held. "Flash + 3" in order to perform 3-way conference. "Flash + 4" in order to perform attended transfer.
Callee Flash to 3WC	When this feature is set to Yes, device would be able to set up the 3-way conference call even when device is the callee in the second call. Default is No.
Ring	
Ringback Tone at No Early Media	Enables or disbales playing a ringback tone when no early media RTP packets are received. Disabled by default.
Ring Timeout	Stops ringing when incoming call if not answered within a specific period of time. When set to 0 there will be no ringing timeout. Default is 60 seconds.
Call Display	
Caller ID Display	When set to "Auto", the device will look for the Caller ID in the order of P-Asserted Identity, Remote-Party-ID and From header field in the incoming SIP INVITE. When set to "Disabled", all incoming calls are displayed with "Unavailable". When set to "From Header", the device will look for the Caller ID in the From header field of the incoming SIP INVITE.
Caller ID Transmission Mode	Selects the FSK Caller ID message structure. The selected format must match the requirements of the connected analog phone or the regional Caller ID standard. The available options are:Multiple Data Message Format (MDMF): Sends Caller ID information using multiple data fields, supporting additional information such as caller name, date/time. Requires phone support. Single Data Message Format (SDMF): Sends Caller ID information in a simplified format, typically containing only the calling number and date/time. Offers strong compatibility. The default setting is "Multiple Data Message Format".
Enable Unknown Caller ID	Enables the Unknown Caller ID information Enabled by default
Replace Beginning '+' with 00 in Caller ID	Replace Beginning '+' with 00 in Caller ID Disbaled by default.

Number of Beginning Digits to Strip from Caller ID	Range is between 0 and 10, default is 0
Escape '#' as %23 in SIP URI	Replaces # by %23 for some special situations.
Enable Connected Line ID	If set to "No", Connected Line ID will not be displayed even received
FXS PORT x -> Advanced Settings	
Advanced Settings	
Special Feature	Selects Soft switch vendors' special requirements Examples of vendors: BroadSoft, CBCOM, RNK, Huawei, China Mobile, ZTE IMS, PhonePower, TELKOM SA, Vonage, Metaswitch, CenturyLink, MTS, Oi_BR, Telefonica, GIBTELECOM. The default is Standard,
Conference URI	Configures the conference URI when using BroadSoft N-way calling feature.
Security Settings	
Allow SIP Reset	Allows ressting the unit using the SIP requests.
Validate Incoming SIP Message	Checks the authenticity and integrity of incoming SIP messages for enhanced security.
Check SIP User ID for Incoming INVITE	Checks the SIP User ID in the Request URI of the incoming INVITE; if it doesn't match the HT801 V2/HT802 V2 SIP User ID, the call will be rejected. Direct IP calling will also be disabled. The default is No.
Authenticate Incoming INVITE	Challenges the incoming INVITE for authentication with SIP 401 Unauthorized message. Default is No.
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.
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".
Trusted Domain Name List	Supports setting specific domain names and wildcard domain names, wildcard domain names such as "*.grandstream.com", use "," to separate domain names
Authenticate Server Certificate chain	Verifies certificate when communication method is TCP/TLS
FXS PORT x -> Call Features Settings	
Enable Call Features	When enabled, Do No Disturb, Call Forward and other call features can be used via the local feature codes on the phone. Otherwise, the ITSP feature codes will be used. Enable All will override all individual features enable setting. Default is Yes
Reset Call Features	Allows users to reset all call features configuration. Default is No
SRTP	
SRTP Feature	Allow users to customize the SRTP feature codes. Default is Yes
Enable SRTP per call	Enable SRTP: Default is 16
Disable SRTP per call	Disable SRTP: Default is 17
SRTP per call Feature	- Enable SRTP per call: Default is 18 - Disable SRTP per call: Default is 19
Enable SRTP per call	Set the function code to enable SRTP (the default value is 18), which is only valid for the current call
Disable SRTP per call	Set the function code to disable SRTP (the default value is 18), which is only valid for the current call
CID	
CID Feature	Allow users to customize the CID feature codes. Default is Yes
Enable CID	Enable CID: Default is 31

Disable CID	Disable CID: Default is 30
CID per call Feature	Whether to display the user ID, it is only valid for the current call
Enable CID per call	Enable CID per call: Default is 82
Disable CID per call	Disable CID per call: Default is 67
Calling Feature	
Direct IP Calling Feature	Allow users to customize the Direct IP feature code. Default is Yes
Direct IP Calling	Direct IP Calling: Default is 47
CW Feature	Allow users to customize the call waiting feature codes. Default is Yes
Enable CW	Enable CW: Default is 51
Disable CW	Disable CW: Default is 50
CW per call Feature	Enable or cancel the call waiting function, only the current call is valid
Enable CW per call	Enable CW per call: Default is 71
Disable CW per call	Disable CW per call: Default is 70
Call Return Settings	
Call Return Feature	Allow users to customize the Call Return feature code. Default is Yes
Call Return	Call return: Default is 69
Forward	
Unconditional Forward Feature	Allow users to customize the Unconditional Forward feature codes. Default is Yes
Enable Unconditional Forward	Set the function code to enable unconditional call forwarding (default value is 72)
Disable Unconditional Forward	Set the function code to cancel unconditional call forwarding (default value is 73)
Busy Forward Feature	Whether to enable the call forwarding function when busy
Enable Busy Forward	Set the function code to enable call forwarding when busy (default value is 90)
Disable Busy Forward	Set the function code to cancel call forwarding when busy (default value is 91)
Delayed Forward Feature	Whether to enable the function of transferring calls without answering
Enable Delayed Forward	Set the function code to enable call transfer without answering (default value is 92)
Disable Delayed Forward	Set the function code for canceling unanswered call transfer (default value is 93)
Paging Settings	
Paging Feature	Allow users to customize the Paging feature code. Default is Yes
Paging	Paging: Default is 74
DND	
DND Feature	Allow users to customize the CW feature codes. Default is Yes
Enable DND	Enable DND: Default is 78
Disable DND	Disable DND: Default is 79
Blind Transfer Feature	
Blind Transfer Feature	Allow users to customize the Blind Transfer feature code. Default is Yes
Enable Blind Transfer	Enable Blind Transfer: Default is 87

LEC	
Disable LEC per call Feature	Whether to disable the LEC (Line Echo Cancellation) function of the current call (only the current call is prohibited)
Disable LEC per call	Set the function code to disable the LEC call function for the current call (default value is 03)
Off-hook Auto Dial	
Set Offhook Auto-Dial Feature	Configures whether to Set Offhook Auto-Dial Feature.
Enable set Offhook Auto-Dial	Enable to set Offhook Auto-Dial (default value is 77).
Registration Id	
Play registration id Feature	Whether it is displayed as a registration ID.
Enable playing registration id	Set to enable the function code to display the registration ID (default value is 98).
Style 3-Way Conference	
Disable Bellcore Style 3-Way Conference	Whether to set the function code for the three-party conference will only take effect after closing the Bellcore three-party conference.
Star Code 3WC Feature	Allows users to initiate a three-way conference call by dialing a predefined star code.
Star Code 3WC	Set the function code of the three-party conference (default value is 23).
Codec	
Forced Codec Feature	Whether to enable the function of forcing the speech encoding type
Forced Codec	Sets the function code to enable forced voice coding type calls (default value is 02)
PCMU Codec Feature	Whether to force PCMU speech encoding
PCMU Codec	Sets the function code to enable forced PCMU voice encoding calls (default value is 7110)
PCMA Codec Feature	enables the PCMU codec for audio encoding and decoding in VoIP communications,
PCMA Codec	Set the function code to enable forced PCMA voice encoding (default value is 7111)
G723 Codec Feature	Whether to force G723 speech encoding
G723 Codec	Set the function code to enable forced G723 speech encoding (default value is 723)
G729 Codec Feature	Whether to force G729 speech encoding
G729 Codec	Set the function code to enable forced G729 speech encoding (default value is 729)
iLBC Codec Feature	Whether to force iLBC voice encoding
iLBC Codec	Set the function code to enable forced iLBC voice encoding (default value is 7201)
G722 Codec Feature	Whether to force G722 speech encoding
G722 Codec	Set the function code to enable forced G722 codec voice encoding (The default value is 722)
FXS PORT x -> Ringtone	
Custom Ring Tone	
Custom Ring Tone 1-3	If the callee number matches, a custom ringtone will be used
Custom Ring Tone 1-3 will be used when the caller is	Set the outgoing call number using custom call ring tone
Ring Tone	

Ring Tone 1-10	Define the Ringtone(s) frequencies, Syntax: c=on1/ off1[- on2/ off2[- on3/ off3]]; (melody on/off time must be a multiple of 1ms) Default value: c=2000/4000;
Distinctive Call Waiting Tone	
Call Waiting Tone 1-3	If the caller number matches, a custom call waiting tone will be used
Call Waiting Tone 1-3 used if incoming caller ID is	Set the incoming call number using custom call waiting tone
Call Waiting Tone	
Call Waiting Tone 1-10	Defines the call waiting tone cadence Syntax: f1=val, f2=val[, c=on1/ off1[- on2/ off2[- on3/ off3]]]; (f is in Hz, on and off are calculated in 1ms. on is the ringing time , off is the silent time) Default value: f1=440@-13,c=300/10000;

NAT Settings

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

1. STUN Server (under the 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 a public IP, keep this field blank.
2. Use random SIP/RTP ports (under the 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 the gateway is behind a firewall on a private network.

DTMF Methods

The HT80x V2 support the following DTMF modes:

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

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

Preferred Vocoder (Codec)

The HT80x V2 supports the following voice codecs. On Profile pages, choose the order of your favorite codecs:

- PCMU/A (or G711u/a)
- G729 A/B
- G723.1
- G726
- iLBC
- OPUS
- G722

Configuring HT80x V2 through Voice Prompts

As mentioned previously, the HT80x V2 has a built-in voice prompt menu for simple device configuration. Please refer to "Understanding HT80x V2 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 HT80x V2 to use DHCP.

- **STATICIP MODE**

Select voice menu option 01 to enable HT80x V2 to use STATICIP mode, then use options 02, 03, 04, and 05 to set up IP address, Subnet Mask, Gateway, and DNS server, respectively.

- **PPPOE MODE**

Select voice menu option 01 to allow the HT80x V2 to enable the PPPoE mode. PPPoE Username and Password should be configured from the 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 menu option 15 to choose the firmware and configuration upgrade protocol between TFTP, HTTP, HTTPS, FTP, and FTPS.

- **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 the WAN port. Press 9 in this menu to toggle between enable/disable. The default is disabled.

Configuration through a Central Server

The HT80x V2 can be automatically configured from a central provisioning system. When HT80x V2 boots up, it will send TFTP, HTTP/HTTPS, or FTP/FTPS requests to download configuration files, "cfgec74d7xxxxxx" and "cfgec74d7xxxxxx.xml", where "ec74d7xxxxxx" is the LAN MAC address of the HT80x V2. If the download of "cfgxxxxxxxxxxxx.xml" is not successful, the provision program will issue a request for a generic configuration file "cfg<Model>.xml" followed by a request to cfg.xml. The configuration file name should be in lower-case letters. The configuration data can be downloaded via TFTP, HTTP/HTTPS, or FTP/FTPS from the central server. A service provider or an enterprise with a large deployment of HT80x V2 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),

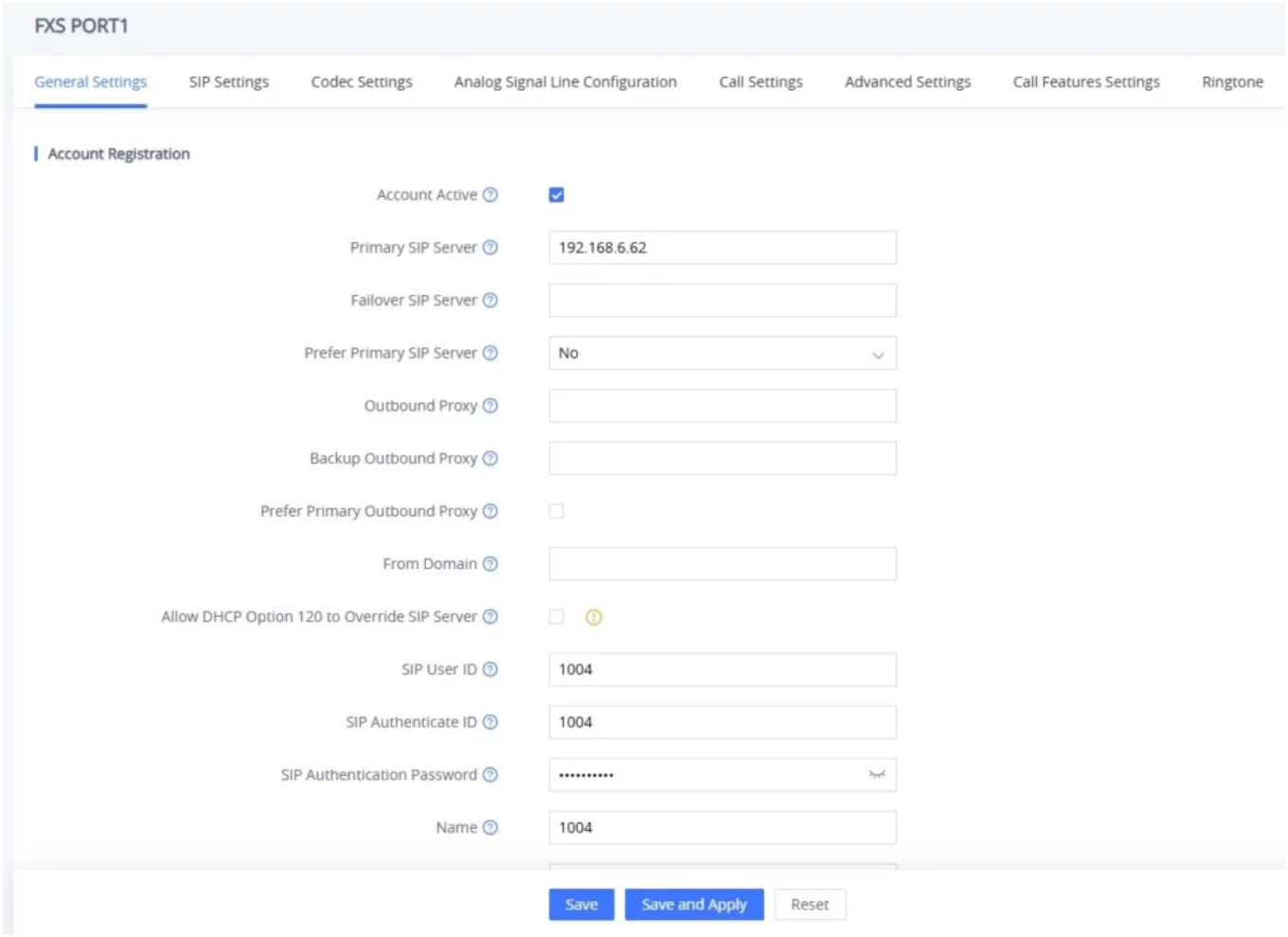
to support the automated configuration of Grandstream devices. GAPS uses HTTPS and other communication protocols to communicate with each individual Grandstream device for firmware upgrades, remote reboot, etc. Grandstream provides GAPS service to VoIP service providers. Use GAPS for either simple redirection or with certain special provisioning settings. At boot-up, Grandstream devices by default point to the Grandstream provisioning server GAPS, based on the unique MAC address of each device. GAPS provisions the devices with redirection settings so that they will be redirected to the customer's TFTP, HTTP/HTTPS, or FTP/FTPS server for further provisioning. Grandstream also provides configuration tools (Windows and Linux/Unix versions) 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 https://www.grandstream.com/support/tools

Register a SIP Account

The HT80x V2 supports two SIP profiles, allowing you to configure up to two SIP servers. Follow the steps below to register your SIP accounts using the Web UI:

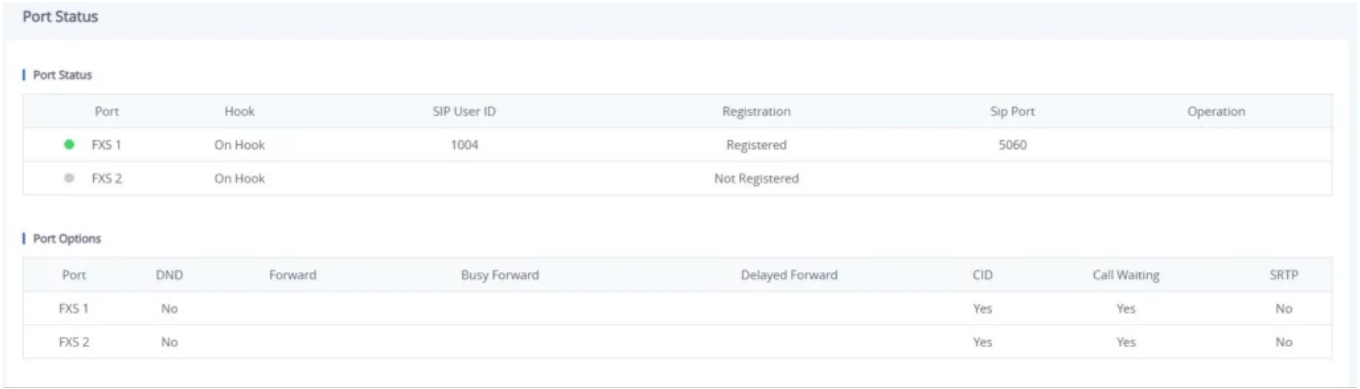
1. Access the HT80x V2 Web UI using the administrator credentials.
2. Navigate to **Port Settings -> FXS PORT x -> General Settings -> Account Registration**.
3. Enable **Account Active**.
4. In the **Primary SIP Server** field, enter the IP address or FQDN of the SIP server.
5. In the **Failover SIP Server** field, enter the IP address or FQDN of the failover SIP server. Leave this field empty if a failover server is not available.
6. In the **Outbound Proxy** field, enter the IP address or FQDN of the outbound proxy. Leave this field empty if an outbound proxy is not required.
7. In the **SIP User ID** field, enter the SIP user ID provided by the VoIP service provider (ITSP). This is typically a phone number or a numeric identifier.
8. In the **Authenticate ID** field, enter the SIP authentication ID provided by the ITSP. This value may be the same as or different from the **SIP User ID**.
9. In the **Authenticate Password** field, enter the SIP account password provided by the ITSP. For security reasons, the password is displayed as blank after it is saved.
10. In the **Name** field, enter a name to identify the account.
11. Click **Save and Apply** at the bottom of the page to save the configuration.



HT80x V2 SIP Registration

After applying the configuration, the account attempts to register with the SIP server. To verify the registration status, navigate to **Status -> Port Status -> Registration** in the HT80x V2 Web UI.

If the status displays **Registered**, the account has successfully registered with the SIP server. If the status displays **Not Registered**, verify the account settings or contact your VoIP service provider (ITSP) for assistance.



HT80x V2 Port Status

Call Features

The HT80x V2 supports all the traditional and advanced telephony features listed in the table below.

Key	Call Features

*02	Forcing a Codec (per call) *027110 (PCMU), *027111 (PCMA), *02723 (G723), *02729 (G729), *027201 (iLBC), *02722 (G722).
*03	Disable LEC (per call) Dial "*03" +" number".No dial tone is played in the middle.
*16	Enable SRTP
*17	Disable SRTP
*18	Enable SRTP per call. Meaning it's only valid for the current call.
*19	Disable SRTP per call. Meaning it's only valid for the current call.
*30	Block Caller ID (for all subsequent calls)
*31	Send Caller ID (for all subsequent calls)
*47	Direct IP Calling. Dial "*47" + "IP address". No dial tone is played in the middle.
*50	Disable Call Waiting (for all subsequent calls)
*51	Enable Call Waiting (for all subsequent calls)
*67	Block Caller ID (per call). Dial "*67" +" number". No dial tone is played in the middle.
*82	Send Caller ID (per call). Dial "*82" +" number". No dial tone is played in the middle.
*69	Call Return Service: Dial *69 and the phone will dial the last incoming phone number received.
*70	Disable Call Waiting (per call). Dial "*70" +" number". No dial tone is played in the middle.
*71	Enable Call Waiting (per call). Dial "*71" +" number". No dial tone is played in the middle
*72	Unconditional Call Forward: Dial "*72" and then the forwarding number followed by "#". Wait for dial tone and hang up. (dial tone indicates successful forward)
*73	Cancel Unconditional Call Forward. To cancel "Unconditional Call Forward", dial "*73", wait for dial tone, then hang up.
*74	Enable Paging Call: Dial "*74" and then the destination phone number you want to page.
*77	Enable set Offhook Auto-Dial.
*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
*98	Enable to play the registration ID.
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.

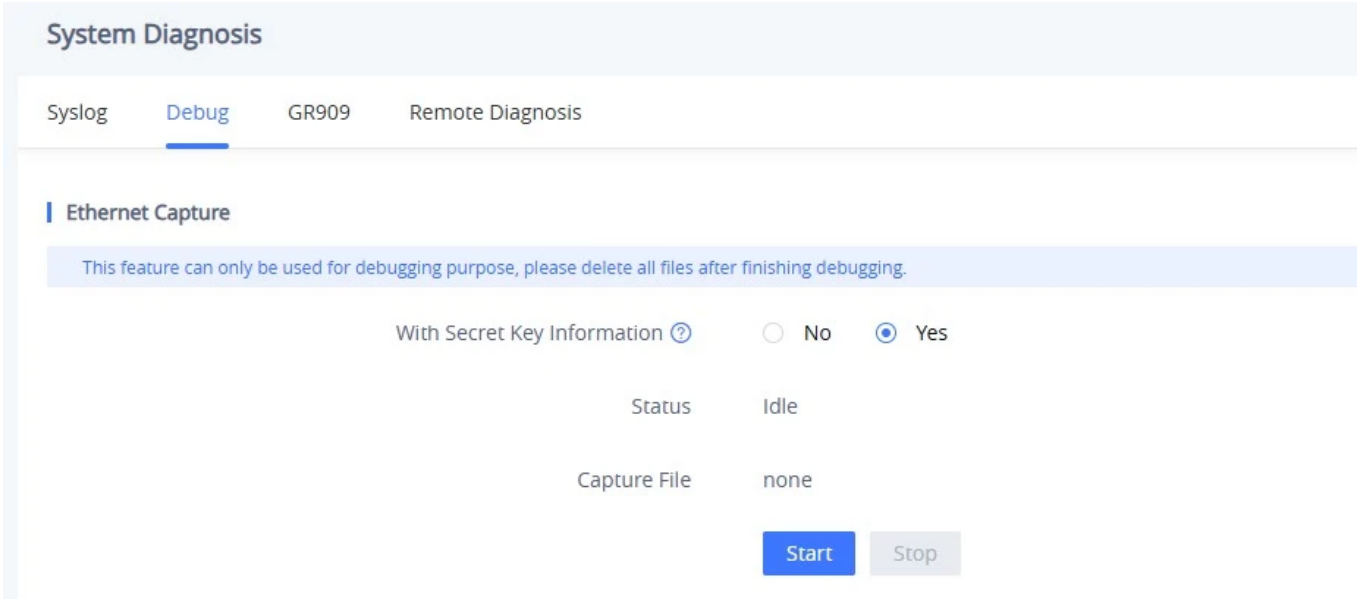
Ethernet Capture

Ethernet Capture records network traffic exchanged between the HT80x V2 and other devices, allowing administrators to analyze communication for troubleshooting and diagnostics.

When the HT80x V2 communicates with a SIP server over **TLS**, Ethernet Capture can also collect the **secret key information** generated during the TLS handshake. The downloaded capture package includes the packet capture file and an **sslkeylogfile.txt** file containing the client's TLS session keys. These keys can be used to decrypt the captured TLS traffic in supported network analysis tools, such as Wireshark.

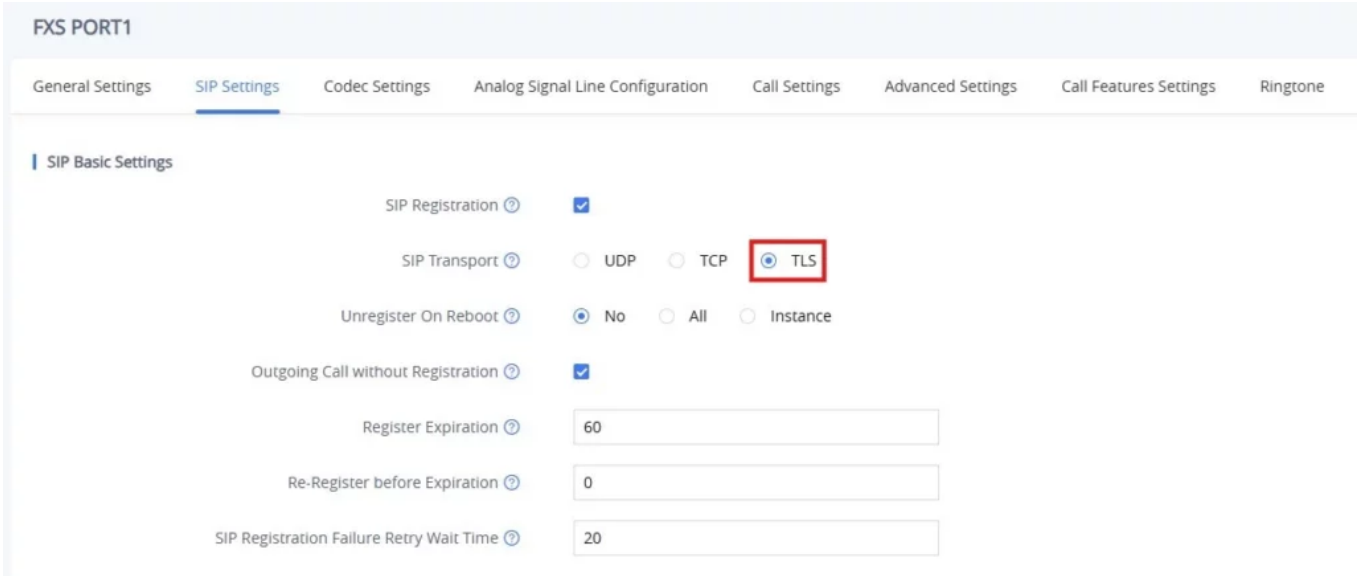
To capture the secret key information, follow the steps below:

1. Initiate packet capture between the HT80x V2 and the server by navigating to **Maintenance -> System Diagnosis -> Debug**. Ensure **With Secret Key Information** is set to **"Yes"**, and press **Start**.



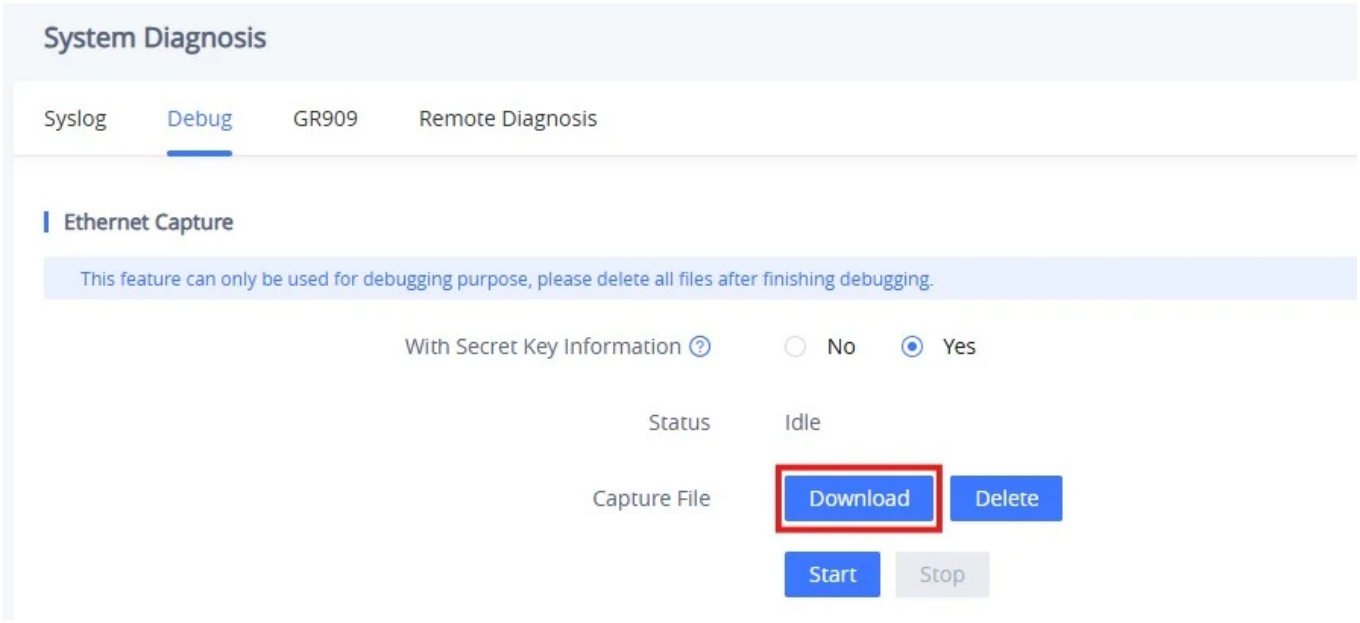
HT80x V2 Ethernet Capture

1. Access **Port Settings -> FXS PORT x -> SIP Settings -> SIP Basic Settings** and set **SIP Transport** to **TLS**. The HT80x account will then re-register with the SIP server using the **TLS** transport protocol.



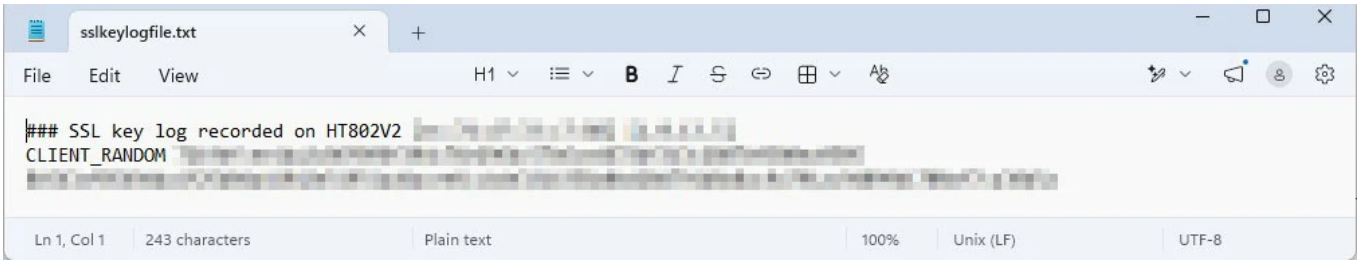
HT80x V2 TLS SIP Transport

1. After the HT80x resends the SIP REGISTER request to the server, press **Stop** to end the packet capture, then press **Download** when the capture file is available.



HT80x V2 Downloading Ethernet Capture

1. After the file is downloaded, extract it to obtain the packet capture file and the sslkeylogfile.txt file containing the secret key information.



HT80x V2 Secret Key Information

Please use Information capture only when authorized, since extracting secret key information without proper authorization is considered unethical.

SCEP Configuration

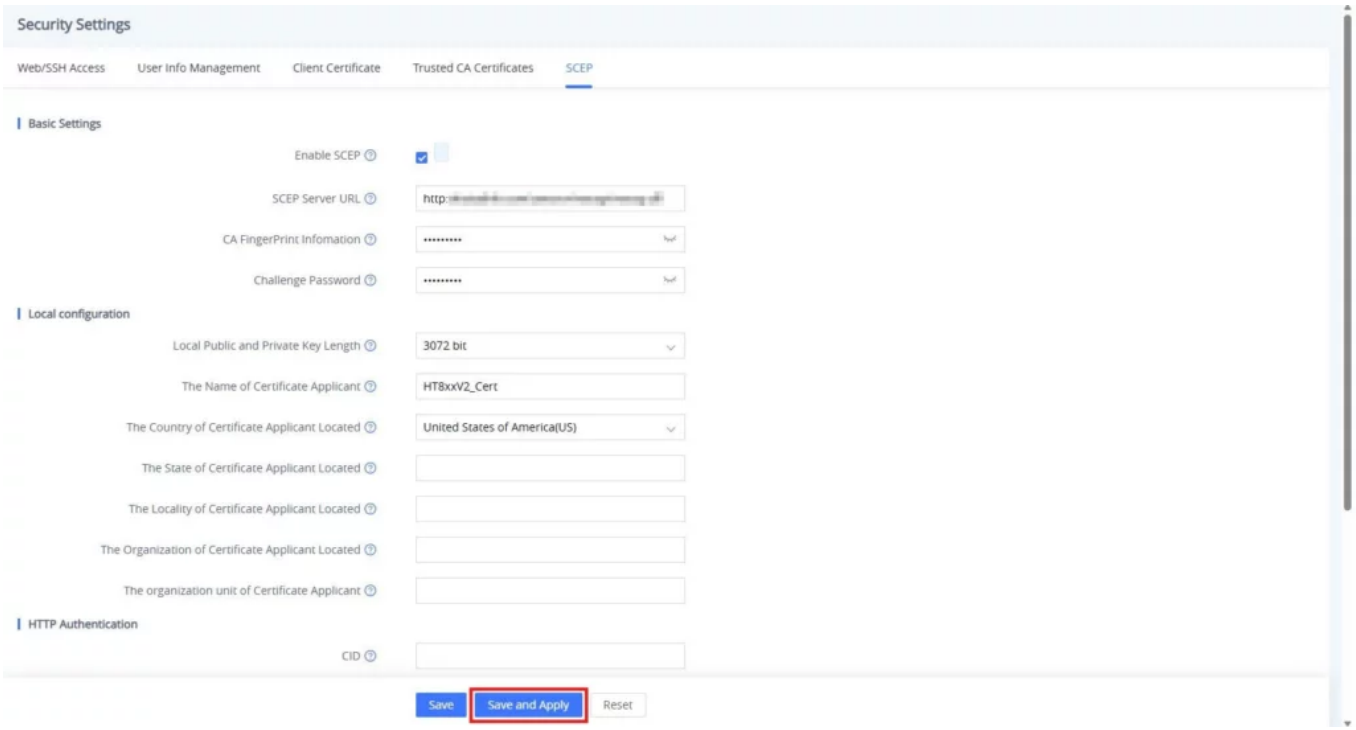
To configure SCEP settings, navigate to **System Settings -> Security Settings -> SCEP** and follow the steps below:

- **Enable SCEP:** Enable SCEP service.
- **SCEP Server URL:** Enter the SCEP server enrollment URL provided by your Certificate Authority (CA) or Registration Authority (RA). This URL is used by the device to request certificates using the SCEP protocol.
- **CA Fingerprint Information:** Enter the fingerprint (thumbprint) of the issuing CA certificate. This field is optional and can be used to verify the identity of the CA before certificate enrollment.
- **Challenge Password:** Enter the challenge password required by the SCEP server to authorize certificate enrollment. If required, this password is provided by your CA or RA administrator.
- **Local Public and Private Key Length:** Select the key size used to generate the device's public and private key pair. Ensure that the selected key length complies with your CA's certificate policy. The default setting is "3072 bit".

- **The Name of Certificate Applicant:** Enter the Common Name (CN) for the certificate applicant. This value is required and is included in the certificate request.
- **The Country of Certificate Applicant Located:** Enter the country where the certificate applicant is located. This field is optional and may be required by your CA policy.
- **The State of Certificate Applicant Located:** Enter the state or province where the certificate applicant is located. This field is optional and may be required by your CA policy.
- **The Locality of Certificate Applicant Located:** Enter the city or locality where the certificate applicant is located. This field is optional and may be required by your CA policy.
- **The Organization of Certificate Applicant Located:** Enter the name of the organization to which the certificate applicant belongs. This field is optional and may be required by your CA policy.
- **The Organizational Unit of Certificate Applicant:** Enter the organizational unit (OU) of the certificate applicant, such as a department or division. This field is optional and depends on the CA policy.
- **User ID:** Enter the user ID required to authenticate with the SCEP server, if applicable.
- **Password:** Enter the password required to authenticate with the SCEP server, if applicable.
- **CA Certificate:** View the CA certificate obtained through SCEP.
- **Client Certificate:** View the client certificate issued through SCEP.
- **Client Certificate Alternative Name:** Configure additional identifiers for the client certificate. These values are included as **Subject Alternative Name (SAN)** entries in the certificate. The supported identifiers are:

    - **DNS Name (dNSName):** Enter a hostname or domain, for example www.example.com.
    - **IP Address (iPAddress):** Enter the IPv4 or IPv6 address, for example 192.168.0.1 or 2001:db8::1.
    - **Email Address (rfc822Name):** Enter an email address, for example user@example.com.
    - **X.500 Distinguished Name (directoryName):** Enter an X.500 distinguished name, for example /C=US/ST=CA/L=Los Angeles/O=Example Inc./CN=www.example.com
    - **URI (uniformResourceIdentifier):** Enter a uniform resource identifier, for example https://www.example.com.
    - **Kerberos Service Name (krb5PrincipalName):** Enter a Kerberos service principal name, for example HTTP/www.example.com.
    - The default setting is **$MAC.local**, which uses the device's MAC address as the default identifier in the certificate request.

- **Update Interval Before Certificate Expiration:** Define the number of days (**1-30**) before certificate expiration that the device attempts to renew the certificate automatically. The default setting is "20".
- **Scope of Use:** Select the modules that use SCEP-managed certificates. By default, certificates are used for **Custom Certificate (SSL/TLS)**. Additional modules, such as **802.1X**, **OpenVPN**, and **TR-069**, can also be configured to use certificates managed through SCEP.
- Press **Save and Apply** to complete the configuration. If the configuration is successful, the Status indicator next to Enable SCEP displays Success in green.



HT80x V2 SCEP Configuration

Rebooting HT80x V2 Remotely

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

UPGRADING AND PROVISIONING

The HT80x V2 can be upgraded via TFTP/HTTP/HTTPS/FTP/FTPS by configuring the URL/IP Address for the TFTP/HTTP/HTTPS/FTP/FTPS server and selecting a download method. Configure a valid URL for TFTP, HTTP/HTTPS, or FTP/FTPS; the server name can be an FQDN or an IP address.

Examples of valid URLs:

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

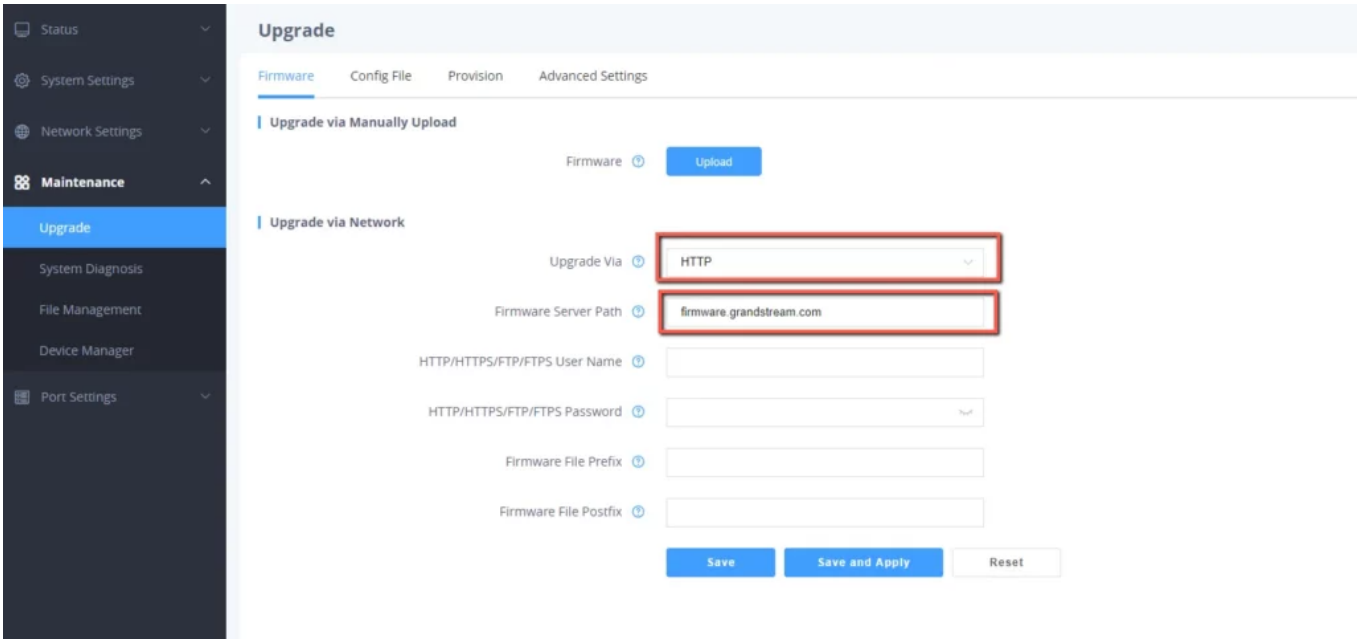
Once HT801 V2/HT802 V2/HT812 V2/HT814 V2/HT818 V2 is upgraded to firmware 1.0.9.3 or later, downgrading to 1.0.7.5 or earlier is not supported due to CPU frequency adjustments.

Firmware Upgrade procedure

Please follow the steps below to upgrade the firmware version of your HT80x V2:

1. Access your HT80x V2 UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: found on the sticker on the back of the unit).
3. Press **Login** to access your settings.

4. Go to **Maintenance** > **Upgrade > Firmware** page and enter the IP address or the FQDN for the upgrade server in the "Firmware Server Path" field and choose to upgrade via TFTP, HTTP/HTTPS, **or FTP/FTPS**.
5. Update the change by clicking the "Save and Apply" button at the bottom of the page. The,n "**Reboot**" or power cycle the HT80x V2 to update the new firmware.



Firmware Upgrade Page

Upgrading via Local Directory

1. Download the firmware file from the Grandstream website
2. Unzip it and copy the file into a folder on your PC
3. From the HT80x V2 web interface **Maintenance** > **Upgrade > Firmware,** you can browse your hard drive and select the folder you previously saved the file.
4. Click "Upload" and wait a few minutes until the new program is loaded.

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

The filename in URL for the firmware upgrade has been disabled.

Upgrading via Local TFTP/HTTP/HTTPS/FTP/FTPS Servers

For users who would like to use remote upgrading without a local TFTP/HTTP/HTTPS/FTP/FTPS server, Grandstream offers a NAT-friendlyHTTP server. This enables users to download the latest software upgrades for their devices via this server. Please refer to the webpage:

https://www.grandstream.com/support/firmware

Alternatively, users can download, for example, a free TFTP or HTTP server and conduct a local firmware upgrade. A free Windows version of TFTP server is available for download from:

https://www.solarwinds.com/products/freetools/free_tftp_server.aspx

https://pjo2.github.io/tftpd64/

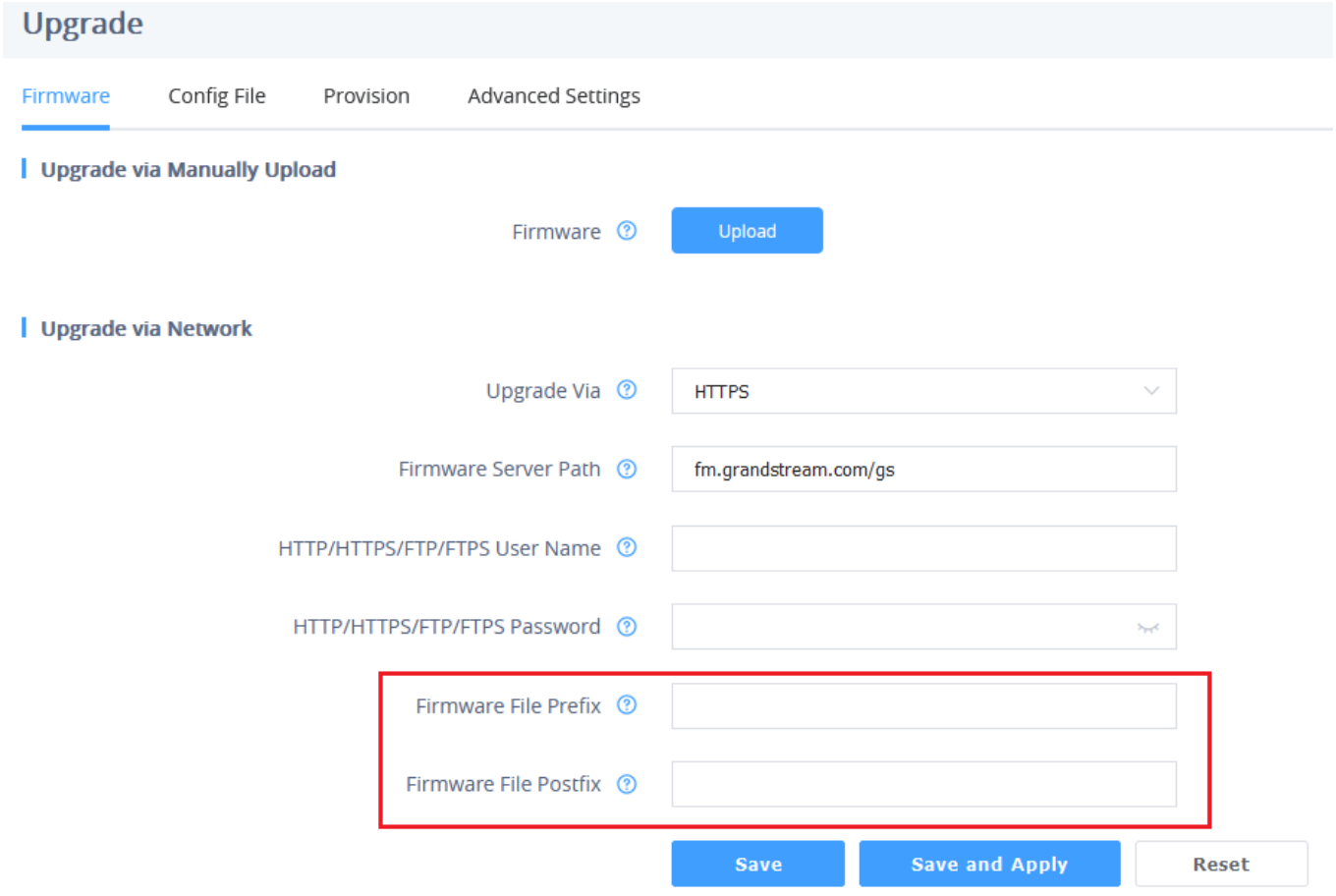
Instructions for local firmware upgrade via TFTP:

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 HT80x V2.

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

Firmware and Configuration File Prefix and Postfix

Firmware Prefix and Postfix allow the device to download the firmware name with the matching Prefix and Postfix. This makes it possible to store all the firmware with different versions in one single directory. Similarly, Config File Prefix and Postfix allow the 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.

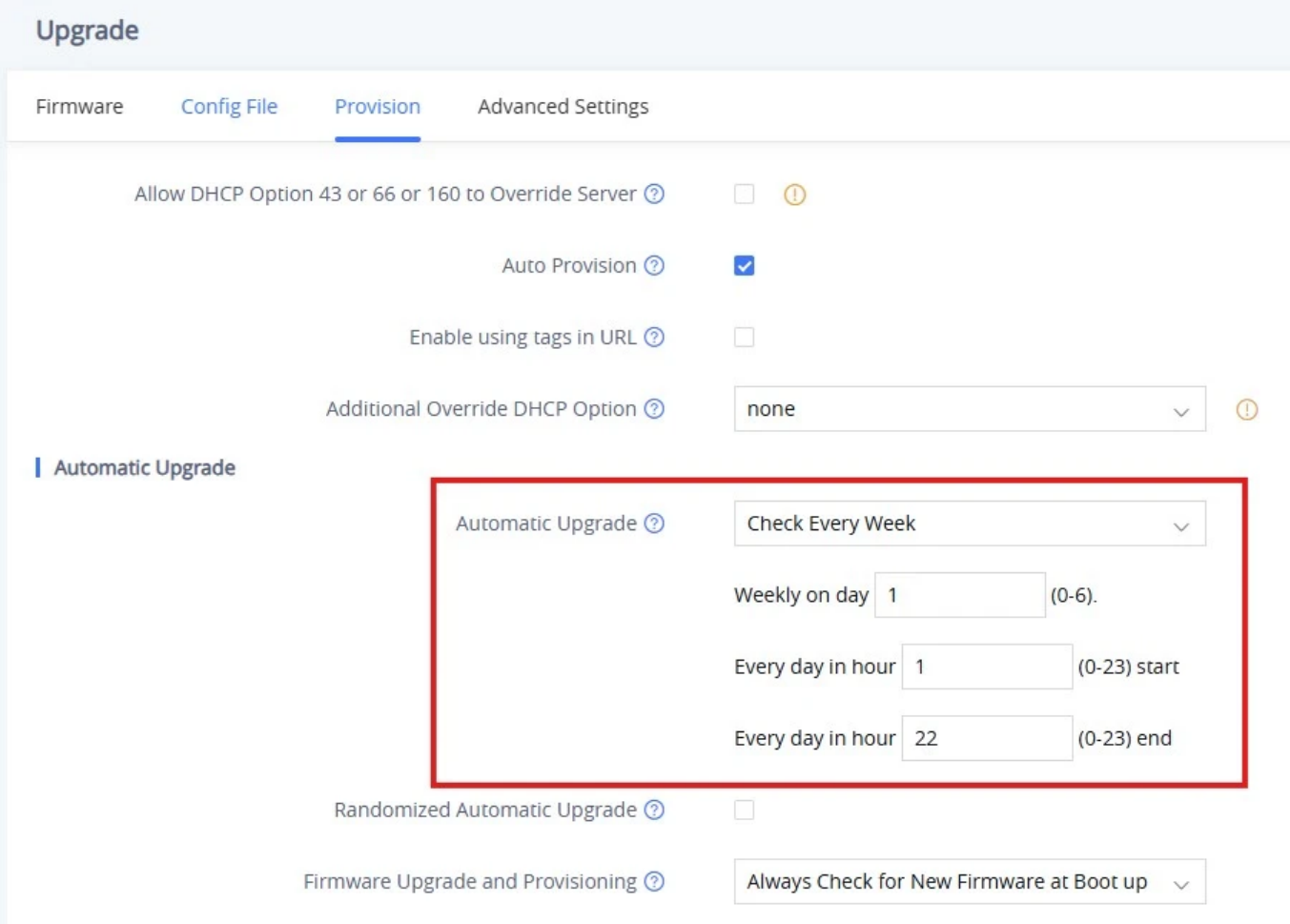


Firmware File Prefix and Postfix

Managing Automatic Upgrade

When "Automatic Upgrade" is set "**Check**", the auto check will be done in the minute specified in this field. If set to "**Every day in hour (0-23)"**, the 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 to check if there are any new changes that 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



HT80x V2 Managing Automatic Upgrade

Managing Device using GDMS Cloud

The GDMS Device Management System is a cloud solution from Grandstream that facilitates the provisioning and management of Grandstream devices, including the HT8xx ATAs.

The platform enables users to centralize the control and management of multiple HT8xx devices, deployed across various locations, on a single interface. It provides functionalities to upgrade, add, configure, and monitor HT8xx devices from one centralized dashboard.

For more details on the supported HT8xx models and their deployment to the GDMS platform, refer to the following guide: GDMS Unified Communications - User Guide.

If you don't have a GDMS account yet, visit the following website to create one: GDMS.Cloud.

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, HTTP/HTTPS, or FTP/FTPS. The **Config Server Path** is the TFTP, HTTP/HTTPS, or FTP/FTPS 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 1 to 5 (Could be extended in the future) digit numbers. i.e., P30 is associated with the "NTP Server" in the **Web GUI > System Settings > Time and Language > Time Zone > NTP Server**. For a detailed parameter list, please refer to the corresponding firmware release configuration

template.

When the HT80x V2 boots up or reboots, it will send a request to download the xml file named "cfgxxxxxxxxxxxx.xml" followed by the binary file named "cfgxxxxxxxxxxxx", where "xxxxxxxxxxxx" is the MAC address of the phone, i.e., "cfg000b820102ab" and "cfg000b820102ab.xml". If the download of the "cfgxxxxxxxxxxxx.xml" file is not successful, the provision program will download a generic cfg<Model>.xml file and then download cfg.xml. The configuration file name should be in lower-case letters.

HT80x V2 supports DHCP option 67, allowing the provision of a custom name for the provisioning file. If DHCP option 67 is used, the following file download sequence will be applied:

Step 1: <option 67 bootfile>

Step 2: cfg<MAC>.xml ->cfg<MAC>->cfg<Model>.xml ->cfg.xml

Notes:

1. The Only acceptable Config file formats to be used when provisioning are XML or binary.

2. Make sure the MAC header on the Config file is the provisioned device's MAC address or you can remove the header completely.

3. When the <option 67 bootfile> is downloaded from the server, the cfg<MAC>.xml is not requested.

For more details on XML provisioning, please refer to:

https://documentation.grandstream.com/knowledge-base/sip-device-provisioning-guide/

The filename in URL for config provision has been disabled.

Configuration and Firmware Upgrade through Resync SIP NOTIFY

HT80x V2 supports triggering firmware and configuration upgrade using Resync SIP NOTIFY. The **event:resync** NOTIFY allows the device to re-synchronize its configuration by checking the provisioning server to download any updates.

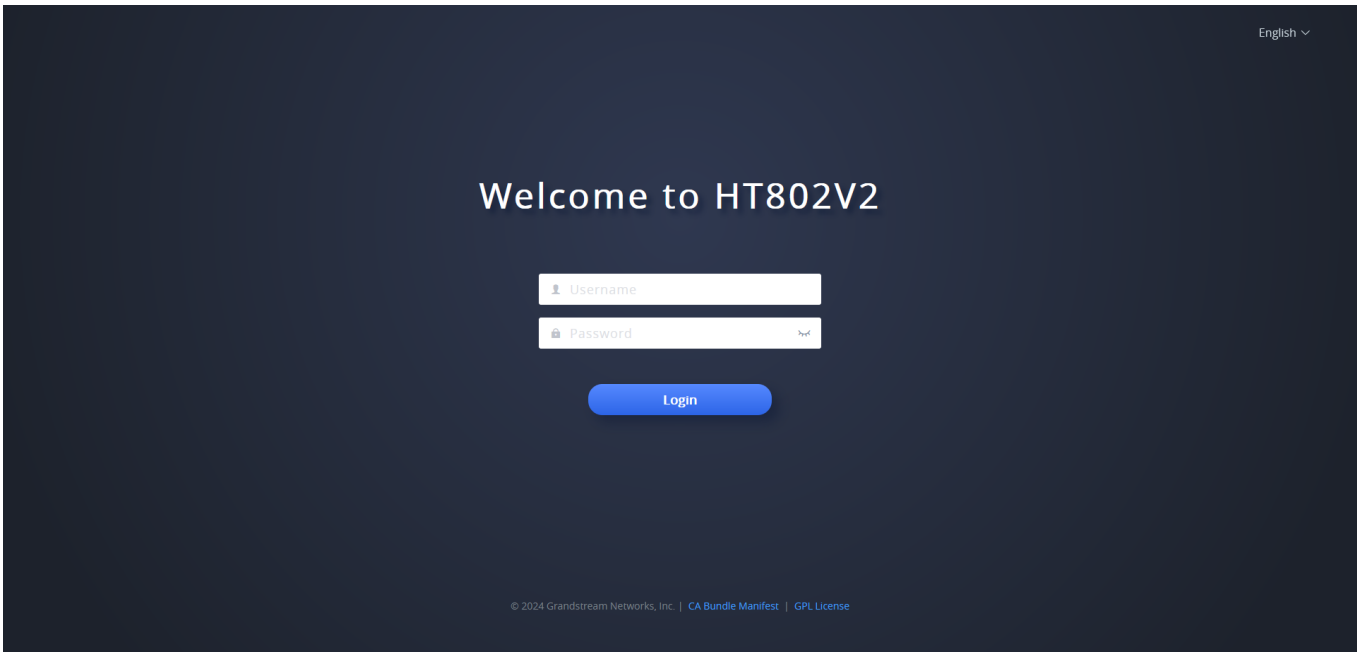
Below is an example of a resync SIP NOTIFY:

```
NOTIFY sip:device@domain.com SIP/2.0
Via: SIP/2.0/UDP 10.0.0.5:5060;branch=z9hG4bK-d4f2-9a75
Max-Forwards: 70
From: "Provisioning Server" <sip:provisioning@domain.com>;tag=prov-server
To: <sip:device@domain.com>;tag=device-987654
Call-ID: 987654xyz321
CSeq: 12345 NOTIFY
Event: resync
Content-Type: application/simple-message-summary
Content-Length: 45
```

CA Bundle Manifest

CA (Certification Authority) Bundle Manifest is a collection of trusted security certificates used to verify the authenticity of websites or services. It helps ensure secure and encrypted connections by confirming the identity of the remote server and protecting against unauthorized access and data interception.

You can access the CA Bundle Manifest by clicking on the bottom corner of the web page, as displayed in the screenshot below:



This will redirect you to a web page displaying a list of certification names and details, with their corresponding expiration dates:

	Certificate Name	Expiration
1	GlobalSign Root CA	Jan 28 12:00:00 2028 GMT
2	Entrust.net Certification Authority (2048)	Jul 24 14:15:12 2029 GMT
3	Baltimore CyberTrust Root	May 12 23:59:00 2025 GMT
4	Entrust Root Certification Authority	Nov 27 20:53:42 2026 GMT
5	AAA Certificate Services	Dec 31 23:59:59 2028 GMT
6	QuoVadis Root CA 2	Nov 24 18:23:33 2031 GMT
7	QuoVadis Root CA 3	Nov 24 19:06:44 2031 GMT
8	Security Communication RootCA1	Sep 30 04:20:49 2023 GMT
9	XRamp Global Certification Authority	Jan 1 05:37:19 2035 GMT
10	Go Daddy Class 2 Certification Authority	Jun 29 17:06:20 2034 GMT
11	Starfield Class 2 Certification Authority	Jun 29 17:39:16 2034 GMT
12	DigiCert Assured ID Root CA	Nov 10 00:00:00 2031 GMT
13	DigiCert Global Root CA	Nov 10 00:00:00 2031 GMT
14	DigiCert High Assurance EV Root CA	Nov 10 00:00:00 2031 GMT
15	SwissSign Gold CA - G2	Oct 25 08:30:35 2036 GMT
16	SwissSign Silver CA - G2	Oct 25 08:32:46 2036 GMT
17	SecureTrust CA	Dec 31 19:40:55 2029 GMT
18	Secure Global CA	Dec 31 19:52:06 2029 GMT
19	COMODO Certification Authority	Dec 31 23:59:59 2029 GMT
20	Network Solutions Certificate Authority	Dec 31 23:59:59 2029 GMT
21	COMODO ECC Certification Authority	Jan 18 23:59:59 2038 GMT
22	Certigna	Jun 29 15:13:05 2027 GMT

RESTORE FACTORY DEFAULT SETTINGS

Restoring the Factory Default Settings will delete all configuration information on the phone. Please back up 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 HT80x V2.
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 the "reset" voice prompt.
3. Enter the encoded MAC address (Look below on how to encode the MAC address).
4. Wait 15 seconds, and the 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:

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

MAC Address Key Mapping

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

Reset from Web Interface (Reset Type)

1. Access your HT80x V2 UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: found on the sticker on the back of the unit).
3. Press **Login** to access your settings.
4. Go to **Maintenance** > **Restore Factory > Reset Type**.
5. Press the **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 to a service provider, like SIP server, SIP user ID, provisioning, and others.

- Factory Reset from the phone will be disabled if the "Lock keypad update" is set to "Yes".
- If the HT80x V2 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.

Reset using SIP NOTIFY

1. Access your HT80x V2 UI by entering its IP address in your favorite browser.
2. Go to **Port Settings > Profile #** page.
3. Set "Allow SIP Factory Reset" to "Yes". (Default is No)
4. Once a SIP NOTIFY with "event: reset" is received, the ATAwill perform a factory reset.

Received SIP NOTIFY will be first challenged for authentication purposes before taking factory reset action. The authentication can be done either using admin credentials (if no SIP account is configured) or using SIP account credentials.

CHANGE LOG

This section documents significant changes from previous versions of the admin guide for HT80x V2. Only major new features or major document updates are listed here. Minor updates for corrections or editing are not documented here.

Firmware Version 1.0.15.1

- Added the ability to configure the VLAN settings via TR069. [Enable Manual VLAN Configuration]

Firmware Version 1.0.13.5

- Added support for displaying Vendor Default Config Version information on the System Info page. [Vendor Default Config Version]
- Added support for SCEP protocol. [SCEP] [SCEP Configuration]
- Added support for direct IP calls in Off-Hook Auto Dial. [Off-Hook Auto Dial]
- Added support for the options "AES-128-GCM", "AES-256-GCM", and "CHACHA20-POLY1305" in OpenVPN(R) Encryption. [OpenVPN(R) Encryption]
- Renamed the setting "WAN Side Web/SSH Access" to "Enable Web/SSH Access". [Enable Web/SSH Access]
- Added support for T.38 Talker Echo Protection (TEP). [Enable T.38 TEP]
- Added support for the option "Yes with Attended Transfer Upon Onhook" in Flash Digit Control. [Flash Digit Control]
- Added support for Use P-Early-Media Header. [Use P-Early-Media Header]
- Added support for Ringback Tone at No Early Media. [Ringback Tone at No Early Media]
- Added support for Caller ID Transmission Mode. [Caller ID Transmission Mode]
- Added support for Busy/Reorder Tone Delay Timer After Loop Current Disconnect. [Busy/Reorder Tone

Delay Timer After Loop Current Disconnect]

Firmware Version 1.0.11.4

- Added support for HTTP/HTTPS Proxy setting. [HTTP/HTTPS Proxy]

Firmware Version 1.0.9.4

- Added P-values to configure settings not available in the Web UI. [Force Default Password Change] [User/Viewer Password Setting Display] [TR-069 Settings Page Display]

Firmware Version 1.0.9.3

- Added "TLS 1.3" option for "Minimum TLS Version" [Minimum TLS Version]
- Added "TLS 1.3" option for "Maximum TLS Version" [Maximum TLS Version]
- Added ability to configure B-timer [SIP Timer B]
- Added ability to configure F-timer [SIP Timer F]

Firmware Version 1.0.7.5

- Adjusted DHCP Option 160 to a higher priority than Option 66 on HT8xx v2 to align with other GS devices. [Allow DHCP Option 66 or 160 to override server]
- Added support for the E911 SIP Geolocation Header Format. [Geolocation Header Format]
- Added Cloudflare as a DDNS provider in the DDNS Server field. [Cloudflare]
- Added support for the Vonage OEM ID. [Vonage]
- Updated the default value of "Web Access Mode" to "0 - HTTPS." [Web Access Mode]
- Changed the default value of "Use Random SIP Port" (to "1 - Yes." [Use Random SIP Port]
- Changed the default value of "Use Random RTP Port" to "1 - Yes." [Use Random RTP Port]
- Changed the default value of "Enable/Disable Weak Cipher Suites" to "8 - Disable All Of The Above Weak TLS Cipher Suites." [Enable/Disable Weak Cipher Suites]
- Changed the default value of "Minimum TLS Version" to "12 - TLS 1.2." [Minimum TLS Version]
- Added support for debug info containing security-related logs in Syslog. [Security-related debug logs]
- Added support for "CSeq Tracking Mode". [CSeq Tracking Mode]
- Added support for "Radius bypass locked state". [Enable Bypass Locked State]
- Added support for "Maximum Transmission Unit (MTU)". [Maximum Transmission Unit (MTU)]
- Added support for enabling TR069 through the management interface. [Enable TR069 Through Management Interface]
- Added support for enabling RADIUS through the management interface. [Enable Radius Through Management Interface]
- Updated the Custom Order for Config Files (P8501) config file download request sequence to start from cfgMAC.xml. [Config Provision Order]
- Added the default value as: cfg$mac.xml;cfg$mac;cfg$product.xml;cfg.xml. [Config Provision Order]
- Added the ability to periodically save system time to a time file. [System Info]
- Added support for OpenVPN TLS Key. [OpenVPN(R) TLS Key] [OpenVPN(R) Additional Options] [OpenVPN(R) TLS Key Type]

Firmware Version 1.0.5.10

- Added the option "Follow SIP Transport" for SRTP Mode settings. [SRTP Mode]

Firmware Version 1.0.5.9

- Updated the default value of Web Access Mode to HTTPS. [Web Access Mode]

Firmware Version 1.0.5.7

- No major changes.

Firmware Version 1.0.5.5

- No major changes.

Firmware Version 1.0.5.4

- No major changes.

Firmware Version 1.0.5.3

- Added support OpenVPN(R) failover option. [OpenVPN(R) Server Secondary]
- Added support for SIP PUBLISH method (RFC 6035). [RFC 6035]
- Added support for Config Provision Order in Config Provision. [Config Provision Order]
- Added support for Multiple DIDs per FXS Port. [Use Request Routing ID in SIP INVITE Header]
- Added support for Enable Multiple Sampling Rates in SDP telephone-event. [Enable Multiple Sampling Rates in SDP telephone-event]

Firmware Version 1.0.3.10

- No major changes.

Firmware Version 1.0.3.8

- Added support to initiate a Re-Invite for fax transmission when the fax machine is the sender. [Re-INVITE Upon CNG Count]

Firmware Version 1.0.3.5

- Added support for 55Vrms Ring Voltage (For New Zealand/Australia) SLIC. (P4234/4235 Ring Power. 0 - 45Vrms default, 3 - 50Vrms, 4 - 55Vrms). [Ring Power]
- Added "HELD Use LLDP Information". [HELD Use LLDP Information]
- Added support to enable LLDP. [Enable LLDP]
- Added "LLDP TX Interval". [LLDP TX Interval]
- Added support to enable CDP. [Enable CDP]
- Added "Use Random SIP Registration Failure Retry Wait Time". [Use Random SIP Registration Failure Retry Wait Time]
- Added the ability to configure minimum and maximum values for the registration retry timer. [Random SIP Registration Failure Retry Wait Time Range]
- Added support to configure a static DNS SRV record. [Static DNS Cache]
- Added support for firmware upgrade via resync SIP Notify. [Configuration and Firmware Upgrade through Resync SIP NOTIFY]

Firmware Version 1.0.1.16

- This is the initial Firmware.