



# Grandstream Networks, Inc.

HT841/HT881 Series

**User Manual**



# HT841/HT881 – User Manual

Thank you for purchasing Grandstream’s HT841/HT881.

The HT841/HT881 Analog FXO Gateway series is an easy-to-set-up IP communications solution that is suitable for small enterprises, as well as those with virtual and branch locations, who wish to benefit from their broadband network or integrate new IP technology with their existing phone system. This series of enterprise analog FXO gateway can convert SIP/RTP IP calls into conventional PSTN calls, and it is available in two models, the HT841 and HT881, with 4 and 8 FXO ports, respectively. Both models share the same installation process.

This User Manual aims to guide you in effectively operating and managing your HT841/HT881 Analog FXO Gateway, empowering you to utilize its numerous enhanced features, including seamless and swift installation.

## PRODUCT OVERVIEW

### Feature Highlights

The following table contains the major features of the HT841/HT881:

 <p><b>HT841/HT881</b></p>	<ul style="list-style-type: none"> <li>● 2 FXO profiles supporting 2 different SIP Servers.</li> <li>● <b>HT841:</b> 4 FXO RJ11 Ports for PSTN Connections.</li> <li>● <b>HT881:</b> 8 FXO RJ11 Ports for PSTN Connections.</li> <li>● PoE in Support for NET2 port.</li> <li>● 2x 10/100 Mbps dual RJ45 network ports: NET2/NET1.</li> <li>● Advanced security protection with SRTP.</li> <li>● Wide range of caller ID formats.</li> <li>● T.38 Fax for creating Fax-over-IP</li> <li>● Strong AES encryption with security certificate per unit</li> <li>● Failover SIP server automatically switches to secondary server if main server loses connection.</li> <li>● Designed and tested for full interoperability with leading IP-PBXs, soft-switches and SIP-based environments.</li> <li>● Use with Grandstream’s UCM series of IP PBXs for Zero Configuration provisioning</li> <li>● Automated provisioning options include TR-069 and XML config.</li> <li>● Lifeline support (FXS port will be hard-relayed to FXO port) in case of power outage.</li> </ul>
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*HT841/HT881 Feature Highlights*

### HT841/HT881 Technical Specifications

The following table resumes all the technical specifications including the protocols/standards supported, voice codecs, telephony features, languages, and upgrade/provisioning settings for the HT841/HT881.

○ **HT841/HT881**

<b>Interfaces</b>	
<b>Telephone Interfaces</b>	<p>One (1) RJ11 FXS port.</p> <p><b>HT841:</b> Four (4) RJ11 FXO PSTN line port with lifeline support.</p> <p><b>HT881:</b> Eight (8) RJ11 FXO PSTN line port with lifeline support.</p>

<b>Network Interface</b>	Two (2) 10/100 Mbps ports (RJ45) with integrated NAT router
<b>LED Indicators</b>	POWER, FXO, FXS, NET1, NET2.
<b>Factory Reset Button</b>	Yes
<b>Voice, Fax, Modem</b>	
<b>Telephony Features</b>	Caller ID display or block, call waiting, flash, blind or attended transfer, forward, hold, do not disturb, 3-way conference
<b>Voice Codecs</b>	G.711 with Annex I (PLC) and Annex II (VAD/CNG), G.723.1, G.729A/B, G.726, iLBC, OPUS, dynamic jitter buffer, advanced line echo cancellation
<b>Fax over IP</b>	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through
<b>Short/Long Haul Ring Load</b>	3 REN: Up to 1km on 24 AWG
<b>Caller ID</b>	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID
<b>Disconnect Methods</b>	Busy Tone, Polarity Reversal/Wink, Loop Current
<b>Signaling</b>	
<b>Network Protocols</b>	TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, FTP/FTPS, ARP/RARP, ICMP, DNS, DDNS, DHCP, NTP, TFTP, SSH, Telnet, STUN, SIP (RFC3261), SIP over TCP/TLS, SRTP, TR-069
<b>QoS</b>	Layer 2 (802.1Q VLAN, SIP/RTP 802.1p) and Layer 3 (ToS, Diffserv, MPLS).
<b>DTMF Methods</b>	In-audio, RFC2833 and/or SIP INFO
<b>Provisioning and Control</b>	HTTP, HTTPS, SSH, FTP, FTPS, Telnet, SSH, TFTP, TR-069, secure and automated provisioning using AES encryption, syslog
<b>Security</b>	
<b>Media</b>	SRTP
<b>Control</b>	TLS/SIPS/HTTPS
<b>Management</b>	Syslog support, SSH, Telnet remote management using web browser.
<b>Physical</b>	
<b>Universal Power Supply</b>	<b>POE Input:</b> 48 V / 0.5 A <b>DC Input:</b> 12V/1A
<b>Environmental</b>	Operational: 32° – 104°F or 0° – 40°C Storage: 14° – 140°F or -10° – 60°C Humidity: 10 – 90% Non-condensing
<b>Dimensions and Weight</b>	<ul style="list-style-type: none"> <li>● HT841/HT881</li> </ul> <b>Dimensions:</b> 190mm x 100mm x 28mm (L x W x D) <b>Weight:</b> 0.46KG

<b>Compliance</b>	
<b>Compliance</b>	FCC/CE/RCM/IC/UKCA

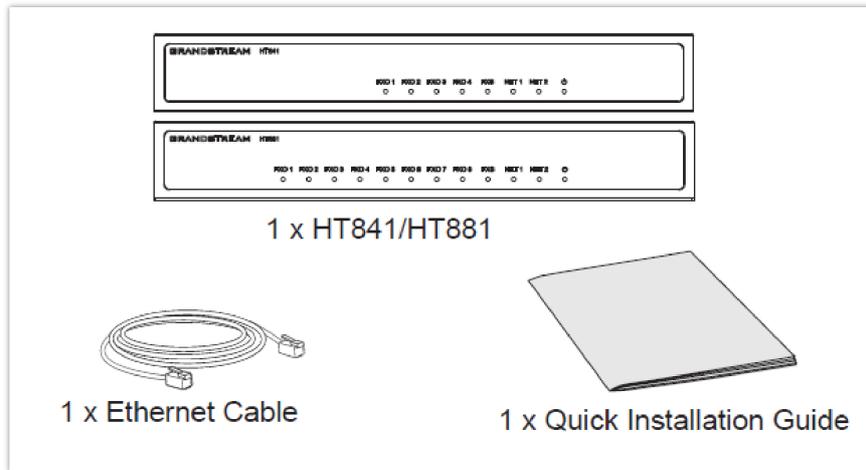
*HT841/HT881 Technical Specifications*

## GETTING STARTED

This chapter provides basic installation instructions including the list of the packaging contents and also information for obtaining the best performance with the HT841/HT881.

### Equipment Packaging

The HT841/HT881 FXO Gateway package contains:

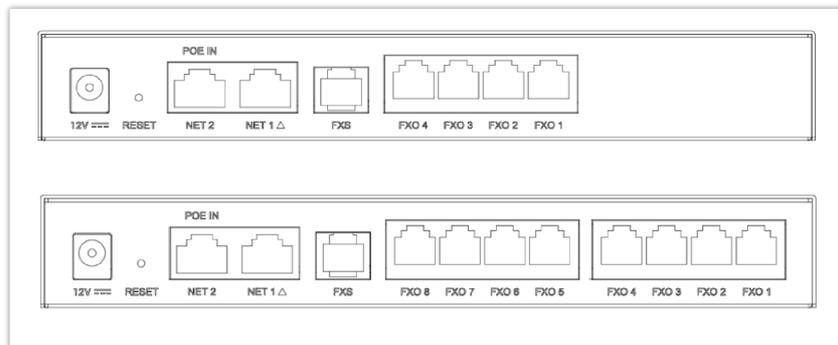


*HT841/HT881 Package Content*

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

### HT841/HT881 Ports Description

The following figure describes the different ports on the back panel of the HT841/HT881.



*HT841/HT881 Ports Description*

Port	Description
DC 12V	Power socket. Used to power HT841/HT881 (12V - 0.5A)
NET 1/NET 2	Network NET 1 / NET 2 port.

	<ul style="list-style-type: none"> <li>• Used to power and connect your HT841/HT881 to local network when using as router.</li> <li>• Used to connect HT841/HT881 to your router or gateway using an Ethernet RJ45 cable. List Item 2</li> </ul>
<b>POE IN</b>	PoE supported port.
<b>FXS</b>	FXS port to connect analog phone / fax machine to HT841/HT881 using RJ11 telephone cable.
<b>FXO 1...4</b> <b>FXO 1...8</b>	FXO ports to be connected to physical PSTN line from a traditional PSTN PBX or PSTN Central Office. HT841 supports 4 FXO ports and HT881 supports 8 FXO Ports.
<b>Reset</b>	Factory reset button. Press for 7 seconds to reset to factory default settings. Quick press will only reboot the unit.

*HT841/HT881 Supported ports*

**Note**

- HT841/HT881 supports switching the working mode of NET1 and NET2 on the Web User interface.
- The default Device Mode for the HT841/881 is bridge. NET1 and NET2 ports function almost identically in bridge mode.

## Connecting HT841/HT881

The HT841/HT881 is designed for easy configuration and easy installation. To connect your HT841/HT881, please follow the steps below:

**Note**

The default Device Mode for the HT841/881 is bridge. NET1 and NET2 ports function almost identically in bridge mode.

## Connecting HT841/HT881 using WAN

When connecting HT841/HT881 using the NET1 port, it will act as a simple DHCP Client.

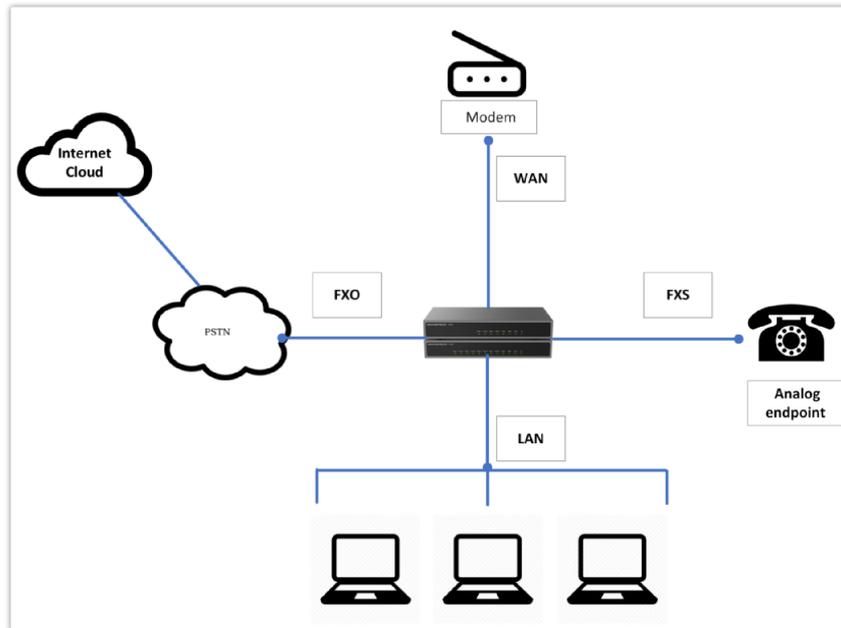
1. Insert a standard RJ11 telephone cable into the FXS port and connect the other end of the telephone cable to a standard touch-tone analog telephone.
2. Connect the NET 1/ NET 2 port of the HT841/HT881 to a router, switch, or modem using an Ethernet cable.
3. Insert the power adapter into the HT841/HT881 and connect it to a wall outlet and make sure to respect the technical specifications of the power adapter used.
4. Power, NET1/NET2, FXO, and FXS LED will be solidly lit when the HT841/HT881 is ready for use.

## Connecting HT841/HT881 using LAN

When connecting the HT841/HT881 using the NET2 port, it will act as a router and DHCP serving addresses, the devices connected with HT841/HT881 LAN will pull DHCP addresses from your HT841/HT881.

1. Insert a standard RJ11 telephone cable into the FXS port and connect the other end of the telephone cable to a standard touch-tone analog telephone.
2. Connect a computer or switch to the NET1/NET2 port of the HT841/HT881 using an Ethernet Cable.
3. Insert the power adapter into the HT841/HT881 and connect it to a wall outlet and make sure to respect the technical specifications of the power adapter used. If the PoE switch is used in step 2, this step could be skipped.
4. Power, NET1/NET2 and FXS, and FXO LED will be solidly lit when the HT841/HT881 is ready for use.

Please make sure to enable NAT Router under Web GUI → Basic Settings → NAT/DHCP Server Information & Configuration → Device Mode.



Connecting the HT841/HT881

### HT841/HT881 LEDs Pattern

There are five (5) LED types that help you manage the status of your HT841/HT881.



HT841/HT881 LED Patterns

LED Lights	Status
<b>Power LED</b>	The Power LED lights up when the HT841/HT881 is powered on and it flashes when the HT841/HT881 is booting up. <b>Note:</b> When powering up the unit with the PSU, the LED light will glow green, whereas it will be orange when using PoE.
<b>NET1 LED</b>	The NET 1 LED lights up when the HT841/HT881 is connected to your network through the NET1 port.
<b>NET2 LED</b>	The NET 2 LED lights up when the HT841/HT881 is connected to your network through the NET2 port.
<b>FXS LED</b>	The FXS LED indicates the status of the FXS port phone on the back panel. <b>OFF</b> – Unregistered <b>ON (Solid Green)</b> – Registered and Available

	<p><b>Blinking every 500 ms</b> – Off-Hook / Busy</p> <p><b>Slow blinking</b> – FXS LED indicates voicemail</p>
<b>FXO LED</b>	<p>The FXO LEDs indicate the status of the FXO ports-phone on the back panel.</p> <p><b>OFF</b> – Unregistered</p> <p><b>ON (Solid Green)</b> – Registered and Available</p> <p><b>Blinking every 500 ms</b> – Off-Hook / Busy</p> <p><b>Slow blinking</b> – FXO LEDs indicate voicemail</p>

#### HT841 HT881 LED Patterns

All LEDs display green when ON. The **Ready** light will only be ON when the network interface is ready and the Web User Interface is accessible.

During a **firmware upgrade** or **configuration download** the following LED pattern will be observed: Power, Ready, and NET1 LEDs will be ON. The FXO port LED will keep flashing during download and then stay OFF while the new files are written. The entire process may take between 20 to 30 minutes. The firmware upgrade is complete when you can login into the web configuration pages.

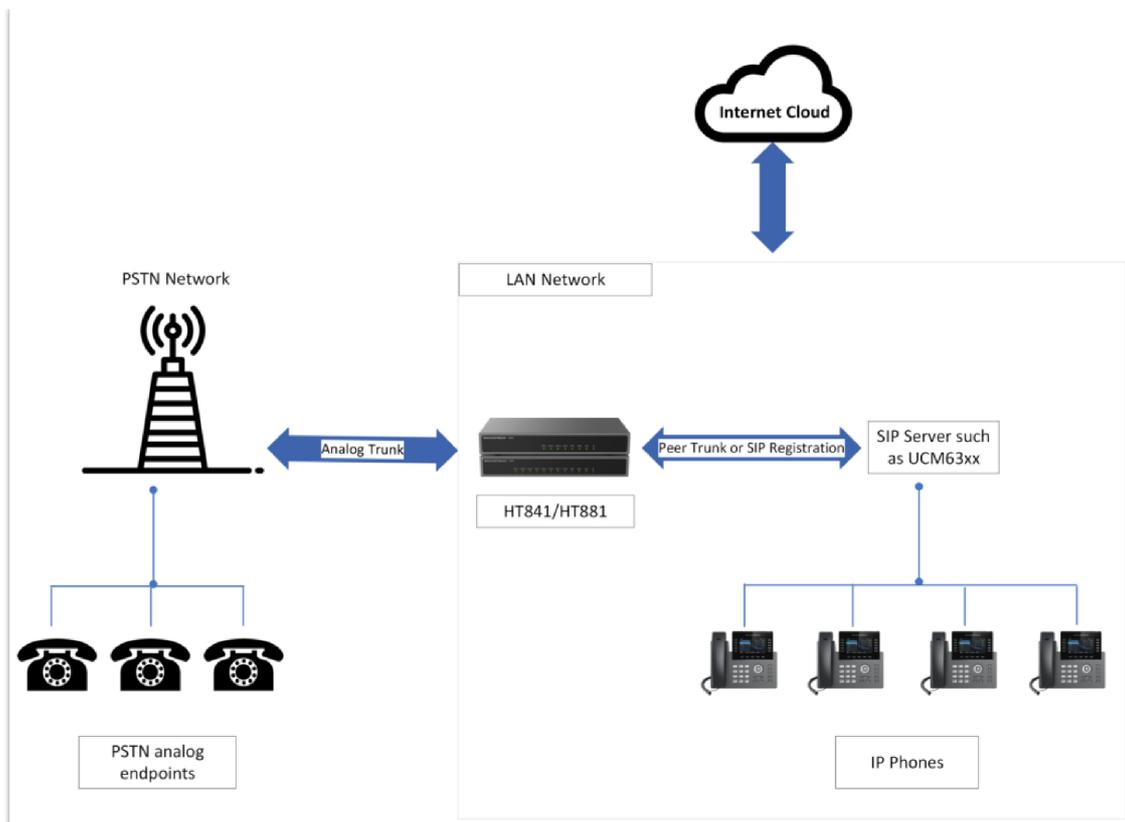
## Application Description

### IP PBX / SIP Server with HT841/HT881

A SIP proxy server such as UCM6xxx can be deployed with the HT841/HT881 series. In this environment, the SIP server handles SIP registration and call control, and the HT841/HT881 processes media conversion between IP and PSTN calls.

There are 2 ways to configure HT841/HT881 when using with a SIP Server:

1. **With SIP accounts configured on the Channels page.** In this case, the HT841/HT881 act like an endpoint requesting registration from the SIP Server. Under the Channels webpage, you will need to fill in information like SIP User ID, Password, etc. Now, when you try to make calls from IP, the call will be routed to the SIP Server which will forward it to one of the SIP accounts on the HT841/HT881, which will then forward it to the PSTN line.
2. **Without SIP accounts.** In this case, you simply have to configure the SIP Server to perform forwarding of the SIP INVITE message with the FXO destination number to the gateway's IP Address. The HT841/HT881 will receive the digits and immediately forward them on the FXO lines to the destination PSTN. Most of the configuration on the Gateway for this case will remain the default, except Stage Method needs to be set to 1, and SIP Server IP Address/DNS name has to be filled.

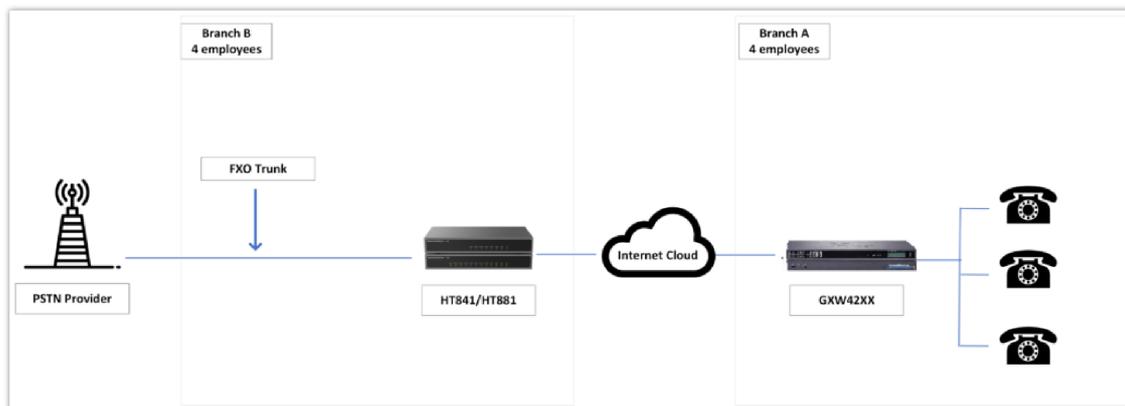


Functional Diagram of IP-PBX & HT841/HT881

For incoming calls from the PSTN analog endpoints to the HT841/HT881, the device will auto-forward each call to a configured IP extension. The SIP Server can then route the call based on its own configuration or IVR system.

### FXS Gateway with HT841/HT881 [No SIP Server required]

Alternatively, the HT841/HT881 can be used without a SIP Server. You can use it in conjunction with an FXS Gateway (Ex. GXW42xx) and still be able to originate and terminate calls from IP to PSTN and vice versa. All you need to make sure is that the 2 gateways are able to locate each other (they should be on the same LAN or on Public IP addresses).



GXW42xx & HT8x1 Scenario/Toll-Free Calling between Locations

In this diagram, configure the SIP Server field to be the IP Address of the other gateway (i.e. configure the IP address of the FXS gateway to be SIP Server of HT841/HT881 and vice versa). Please be sure you set SIP Registration to No.

**Expected Call Flow:** Analog Phone (GXW42xx) picks up and dials destination PSTN number. The call gets routed to the HT841/HT881 which dials out the digit string onto the FXO Lines, thus reaching the destination PSTN endpoint. In reverse, incoming calls from PSTN endpoints will be routed automatically to the FXS Gateway through HT841/HT881.

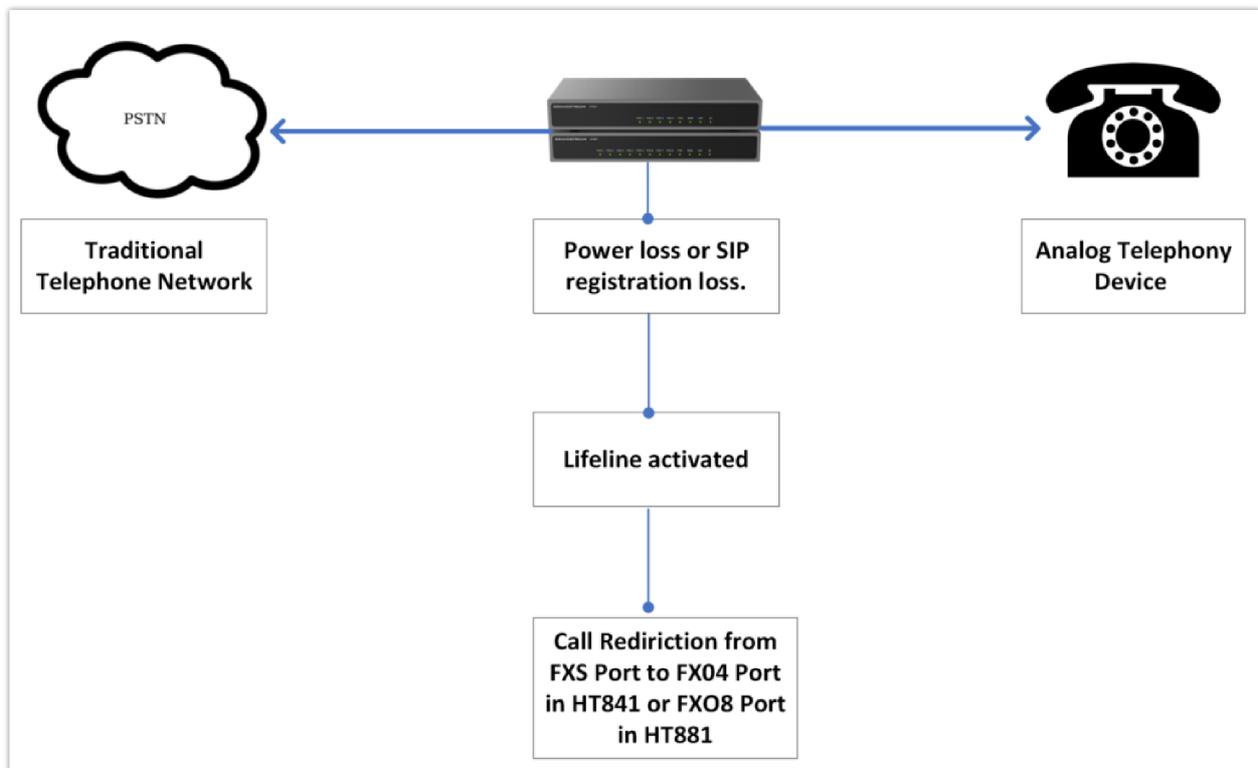
GXW42xx Gateway	HT841/HT881 Gateway
Profile 1	Advanced Settings STUN Server – Blank

SIP Server – Set it to the IP Address of HT841/HT881 SIP Registration – No Outgoing Call without Registration – Yes NAT traversal – No	Use Random Port – No
<b>Advanced Settings</b>  STUN Server – Blank	<b>FXO lines</b>  Wait for Dial Tone – Y or N (whichever works for your PSTN Service Provider) Stage Method – 1 Unconditional Call Forward to VOIP: ch1-8:444; @ch1-8:p1; ch1-8:5060++;  <b>Channels</b> 1-8 5060 Profile 1 Local SIP Listen port (For VOIP to PSTN calls) – 5060++  <b>Profile 1</b>  SIP Server – Set it to IP Address of GXW42xx SIP Registration – No NAT traversal – No

FXS and FXO Gateway Configuration Example

### FXS/FXO Lifeline Implementation

The "Life Line" feature is a telecommunications feature that ensures users can still place or receive a Public Switched Telephone Network (PSTN) call in emergency situations or when there is a disruption in the internet service (SIP registration loss). The feature operates by connecting the PSTN line to an analog phone that is connected to the FXS port.



FXO/FXS Lifeline Support

\*Refer to the following lifeline port info table:

Model	Lifeline Port Mapping	
HT841	FXS	↔ FXO4



The three modes of operation are as follows:

- **Auto Mode (Default):** In Auto mode, the Life Line feature will automatically activate when there is a power loss or when the device loses its registration with the SIP server (which means the VoIP service is unavailable). When this happens, the PSTN line will be seamlessly connected to the analog phone connected to the FXS port. This allows users to make and receive traditional phone calls through the PSTN network, providing a backup communication method during emergencies or internet outages.
- **Always Connected Mode:** In Always Connected mode, the Life Line feature is permanently active, meaning the PSTN line is always connected to the phone connected to the FXS port. In this configuration, VoIP calls will not be allowed. This mode ensures that the device prioritizes the traditional PSTN line for all communications and does not rely on VoIP services.
- **Always Disconnected Mode:** In Always Disconnected mode, the Life Line feature is disabled, and the device will only allow users to make and receive VoIP calls on the FXS port. PSTN calls via the FXS port will not be possible in this configuration. This mode is suitable for situations where the user wants to use VoIP exclusively and does not require the fallback option of PSTN connectivity.

## FXS Telephony Implementation

When the FXS port on an FXO gateway is connected to an analog phone, the following basic telephony operations can typically be performed:

### Placing a Phone Call

To make the outgoing calls using your HT841/HT881

1. Pick up the handset of the connected phone.
2. Dial the number directly and wait for 4 seconds (Default "No Key Entry Timeout"); or
3. Dial the number directly and press # (Use # as dial key" must be configured in web configuration).

#### Examples:

1. Dial an extension directly on the same proxy, (e.g. 1008), and then press the # or wait for 4 seconds.
2. Dial an outside number (e.g. (626) 666-7890), first enter the prefix number (usually 1+ or international code) followed by the phone number. Press # or wait for 4 seconds. Check with your VoIP service provider for further details on prefix numbers.

## Direct IP Calls

Direct IP calling allows two parties, that is, an FXS Port with an analog phone and another VoIP Device, to talk to each other in an ad hoc fashion without a SIP proxy.

#### Elements necessary to completing a Direct IP Call:

- Both HT841/HT881 and other VoIP Devices, have public IP addresses, or
- Both HT841/HT881 and other VoIP Devices are on the same LAN using private IP addresses, or
- Both HT841/HT881 and other VoIP Devices can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

The HT841/HT881 supports two ways to make Direct IP Calling:

#### Using IVR

1. Pick up the analog phone then access the voice menu prompt by dialing "\*\*\*\*"
2. Dial "47" to access the direct IP call menu

3. Enter the IP address after the dial tone and voice prompt "Direct IP Calling"

### Using Star Code

1. Pick up the analog phone then dial "\*\*47"
2. Enter the target IP address.

No dial tone will be played between steps 1 and 2 and destination ports can be specified using "\*" (encoding for ":") followed by the port number.

### Examples of Direct IP Calls:

a) If the target IP address is 192.168.0.160, the dialing convention is **\*47 or Voice Prompt with option 47, then 192\*168\*0\*160**, followed by pressing the "#" key if it is configured as a send key or wait 4 seconds. In this case, the default destination port 5060 is used if no port is specified

b) If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be: **\*47 or Voice Prompt with option 47, then 192\*168\*0\*160\*5062** followed by pressing the "#" key if it is configured as a send key or wait for 4 seconds.

**Note:** When completing a direct IP call, the "Use Random SIP/RTP Port" should be set to "NO".

## CALL FEATURES

The HT841/HT881 supports all the traditional and advanced telephony features, the following table summarizes each key with its corresponding Call Feature.

Key	Call Features
*02	Forcing a Codec (per call) *027110 (PCMU), *027111 (PCMA), *02723 (G723), *02729 (G729), *027201 (iLBC).
*03	Disable LEC (per call) Dial "**03" + " number". No dial tone is played in the middle.
*16	Enable SRTP
*17	Disable SRTP
*30	Block Caller ID (for all subsequent calls)
*31	Send Caller ID (for all subsequent calls)
*47	Direct IP Calling. Dial "**47" + "IP address". No dial tone is played in the middle.
*50	Disable Call Waiting (for all subsequent calls)
*51	Enable Call Waiting (for all subsequent calls)
*67	Block Caller ID (per call). Dial "**67" + " number". No dial tone is played in the middle.
*82	Send Caller ID (per call). Dial "**82" + " number". No dial tone is played in the middle.
*69	Call Return Service: Dial *69 and the phone will dial the last incoming phone number received.
*70	Disable Call Waiting (per call). Dial "**70" + " number". No dial tone is played in the middle.

*71	Enable Call Waiting (per call). Dial “*71” +” number”. No dial tone is played in the middle
*72	Unconditional Call Forward: Dial “*72” and then the forwarding number followed by “#”. Wait for dial tone and hang up. (dial tone indicates successful forward)
*73	Cancel Unconditional Call Forward. To cancel “Unconditional Call Forward”, dial “*73”, wait for dial tone, then hang up.
*74	Enable Paging Call: Dial “*74” and then the destination phone number you want to page.
*78	Enable Do Not Disturb (DND): When enabled all incoming calls are rejected.
*79	Disable Do Not Disturb (DND): When disabled, incoming calls are accepted.
*87	Blind Transfer
*90	Busy Call Forward: Dial “*90” and then the forwarding number followed by “#”. Wait for dial tone then hang up.
*91	Cancel Busy Call Forward. To cancel “Busy Call Forward”, dial “*91”, wait for dial tone, then hang up.
*92	Delayed Call Forward. Dial “*92” and then the forwarding number followed by “#”. Wait for dial tone then hang up.
*93	Cancel Delayed Call Forward. To cancel Delayed Call Forward, dial “*93”, wait for dial tone, then hang up
Flash/Hook	Toggles between active call and incoming call (call waiting tone). If not in conversation, flash/hook will switch to a new channel for a new call.
#	Pressing pound sign will serve as Re-Dial key.

*HT841/HT881 Call Features*

## CONFIGURATION GUIDE

The HT841/HT881 can be configured in one of two ways:

- The IVR voice prompt menu.
- The embedded Web GUI on the HT841/HT881 using PC’s web browser.

### Obtain HT841/HT881 IP Address

HT841/HT881 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 HT841/HT881, you should access the “[Interactive Voice Response Menu](#)” of your adapter via the connected phone and check its IP address mode.

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

1. Use a telephone connected to the FXS port of your HT841/HT881.
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 HT841/HT881 IVR Menu

The HT841/HT881 has a built-in voice prompt menu for simple device configuration which lists actions, commands, menu choices, and descriptions. Connect an analog phone to FXS port. Pick up the handset and dial “\*\*\*\*” to use the IVR menu.

Menu	Voice Prompt	Options
Main Menu	“Enter a Menu Option”	Press “*” for the next menu option Press “#” to return to the main menu Enter 01-05, 07,10, 12-17,47 or 99 menu options
01	“DHCP Mode”, “Static IP Mode” “PPPoE Mode“	Press “9” to toggle the selection If using “Static IP Mode”, configure the IP address information using menus 02 to 05. If using “Dynamic IP Mode”, all IP address information comes from the DHCP server automatically after reboot. If using “PPPoE Mode”, configure PPPoE Username and Password from web GUI to get IP from your ISP.
02	“IP Address “ + IP address	The current WAN IP address is announced. If using “Static IP Mode”, enter 12-digit new IP address. You need to reboot your HT841/HT881 for the new IP address to take Effect.
03	“Subnet “ + IP address	Same as menu 02
04	“Gateway “ + IP address	Same as menu 02
05	“DNS Server “ + IP address	Same as menu 02
07	Preferred Vocoder	Press “9” to move to the next selection in the list: PCM U / PCM A iLBC G-726 G-723 G-729 OPUS
10	“MAC Address”	Announces the MAC address of the unit. <b>Note:</b> The device has two MAC addresses. One for the NET1 port and one for the NET2 port. The device MAC address announced is the address of NET2 port.
12	WAN Port Web Access	Press “9” to toggle between Auto / Enabled / Disabled. Default is Auto.
13	Firmware Server IP Address	Announces current Firmware Server IP address. Enter 12-digit new IP address.
14	Configuration Server IP Address	Announces current Config Server Path IP address. Enter 12-digit new IP address.
15	Upgrade Protocol	Upgrade protocol for firmware and configuration update. Press “9” to toggle between TFTP / HTTP / FTP / FTPS or HTTPS. Default is HTTPS.
16	Firmware Version	Announces Firmware version information.
17	Firmware Upgrade	Firmware upgrade mode. Press “9” to toggle among the following three options: <ul style="list-style-type: none"> <li>• Always check</li> <li>• Check when pre/suffix changes</li> <li>• Never upgrade</li> </ul>
47	“Direct IP Calling”	Enter the target IP address to make a direct IP call, after dial tone. (See “Make a

		Direct IP Call”.)
86	Voice Mail	Access to your voice mails messages.
99	“RESET”	Press “9” to reboot the device. Enter MAC address to restore factory default setting: RESTORE FACTORY DEFAULT SETTINGS
	“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

*HT841/HT881 IVR Menu*

#### Four 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.

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

## Accessing the Web UI

### ○ Via NET1 port

1. You may check your HT841/HT881 IP address using the IVR on the connected phone.

Please see [Obtain the HT841/HT881 IP address via the connected analogue phone](#)

2. Open the web browser on your computer.
3. Enter the HT841/HT881’s IP address in the address bar of the browser.
4. Enter the administrator’s password to access the Web Configuration Menu.

The computer must be connected to the same sub-network as the HT841/HT881. This can be easily done by connecting the computer to the same hub or switch as the HT841/HT881.

### ○ Via NET2 port

1. Power your HT841/HT881 using PSU with the right specifications.
2. Connect your computer or switch directly to your HT841/HT881 LAN port.
3. Open the web browser on your computer.
4. Enter the default LAN IP address (192.168.2.1) in the address bar of the browser.
5. Enter the administrator’s password to access the Web Configuration Menu.
6. Make sure to reboot your device after changing your settings to apply the new configuration.

Please make sure that your computer has a valid IP address on the range 192.168.2.x so you can access the web GUI of your HT841/HT881.

# Web Configuration Pages Definitions

This section describes the options in the HT841/HT881 Web UI.

- **Status:** Displays the system info, network status, account status, and line options.
- **Basic Settings:** Configures the end user level password, IP address modes, web access, time zone settings, and language.
- **Advanced Settings:** Configures networks, upgrading and provisioning, TR-069, security settings, date and time, SNMP, Syslog, audio settings, call settings and call progress tones.
- **FXS Profile:** Configures FXS Profile SIP accounts server settings.
- **FXO Profile (1&2):** Configures FXO Profile SIP accounts server settings, FXO termination.
- **Ports:** Displays the FXS and FXO Ports SIP Settings.

## Status Page Definitions

Status	
<b>MAC Address</b>	Shows device ID in hexadecimal format. This is needed by network administrators for troubleshooting. The MAC address will be used for provisioning and can be found on the label on the original box and on the label located on the bottom panel of the device. <b>Note:</b> The device has two MAC addresses, one for the NET1 port and one for the NET2 port. The MAC address located on the bottom panel of the device is the MAC address of the NET2 port. The MAC address of the WAN port is the MAC address of NET2 port +1. Example: MAC Address: NET1- "00:0B:82:25:AF:32", NET2- "00:0B:82:25:AF:31".
<b>WAN IPv4 Address</b>	Displays assigned IPv4 address.
<b>WAN IPv6 Address</b>	Displays assigned IPv6 address.
<b>Product Model</b>	Displays product model info. Default is HT841/HT881.
<b>Serial Number</b>	Displays the device's serial number.
<b>Hardware Version</b>	Displays the hardware revision information and the part number.
<b>Software Version</b>	<ul style="list-style-type: none"> <li>● <b>Bootloader:</b> Specifies Boot version.</li> <li>● <b>Core:</b> Specifies Core version.</li> <li>● <b>Base:</b> Specifies Base version.</li> <li>● <b>Prog:</b> Specifies Program version. This is the main firmware release number, which is always used for identifying the software system of the HT841/HT881.</li> <li>● <b>CPE:</b> Specifies CPE version. The CPE version is displayed only when HT841/HT881 is connected to an ACS using the TR-069 protocol.</li> </ul>
<b>Software Status</b>	Indicates the current software status of the HT (Running or Stopped).
<b>System Up Time</b>	Indicates actual system time and uptime since last reboot.
<b>CPU Load</b>	Indicates the amount of processing burden placed on the device's central processing unit, providing an insight into system performance and resource utilization.
<b>Network Cable Status</b>	Indicates the connectivity status of the network cable, showing whether it is plugged in or disconnected. For both NET1 and NET2
<b>PPPoE Link Up</b>	Indicates PPPoE connection status.

<b>NAT</b>	Indicates type of NAT when it is configured.
<b>Certificate Type</b>	Indicates the current individual Certificate Generation.
<b>Port Status</b>	Displays relevant information regarding the FXS port and FXO ports and their registration, current status and their appropriate User ID.
<b>Port Options</b>	Displays relevant information regarding the FXS and FXO ports about their DND and call forward features.
<b>CDR File</b>	Refers to the Call Detail Records file that stores information about call activities, including call duration, call time, and call parties, for later analysis and billing purposes. You can Download the CDR file , Preview it or delete it .
<b>SIP File</b>	Refers to the Session Initiation Protocol (SIP) configuration file used to define the device's SIP settings and parameters for establishing and managing VoIP communication.
<b>Provision</b>	Displays provisioning status.
<b>Core Dump</b>	Provides generated core dump file if unit malfunctions. Clean will be displayed if no issues.
<b>System information</b>	Provides essential details about the device, such as model number, firmware version, hardware information, and system status, for diagnostic and management purposes. You can download the system information by clicking "Download"

## Basic Settings Page Definitions

<b>Basic Settings</b>	
<b>New End User Password</b>	Configures user-level password. <b>Note:</b> Must contain 8-30 characters, at least one number, one uppercase, and a lowercase letter.
<b>Confirm End User Password</b>	Re-enter the end user password to confirm the change of user password on the web GUI to avoid typos or mistakes.
<b>New Viewer Password</b>	Configures viewer-level password. <b>Note:</b> Must contain 8-30 characters, at least one number, one uppercase, and a lowercase letter.
<b>Confirm Viewer Password</b>	Re-enter the viewer password to confirm the change of the viewer password on the web GUI to avoid typos or mistakes.
<b>Web/SSH Access</b>	
<b>Web Session Timeout</b>	Configure timer to log out web session during idle. The default is 10 min. The range is 1-60 min.
<b>Web Access Attempt Limit</b>	Configure the attempt limit before the lockout. The default is 5. The range is 1-10.
<b>Lockout Time Interval</b>	If the login attempt failed 5 times, the login would be locked out for the time length. Default 15 mins. Range 0-60 min.
<b>Web Access Mode</b>	Allows users to choose the Web Access Mode between "HTTPS", "HTTP", or "Disabled". If HTTPS is selected, web UI will be accessed using HTTPS. The default is HTTP.

<b>HTTP Web Port</b>	Customizes HTTP port used to access the HT841/HT881 web UI. Default is 80.
<b>HTTPS Web Port</b>	Customizes HTTPS port used to access the HT841/HT881 web UI. Default is 443.
<b>Disable SSH</b>	Enables/disables SSH access. The default is No.
<b>SSH Port</b>	Allows users to self-configure SSH Port number. By default, the port number is 22.
<b>Disable Telnet</b>	Enables/disables the Telnet access. The default is Yes.
<b>Telnet Port</b>	Allows users to self-configure Telnet Port number. By default, the port number is 23.
<b>Security Controls for SSH/Telnet Access</b>	<p>Specifies the security measures and controls applied to the SSH/Telnet protocols to ensure secure and authorized access to the device.</p> <ul style="list-style-type: none"> <li>● <b>Only Allow SSH private IP users to set system Pvalue:</b> This option permits only users with SSH access from private IP addresses to modify specific system-level configuration parameters (system Pvalues).</li> <li>● <b>Allow all SSH users to set system Pvalue:</b> With this setting, all users having SSH access can modify specific system-level configuration parameters (system Pvalues).</li> <li>● <b>Allow all SSH users to set any Pvalue:</b> This allows all users with SSH access to modify any configuration parameter (P-value).</li> <li>● <b>Allow any user to set any Pvalue:</b> Any user, regardless of their privileges, can modify any configuration parameter (Pvalue) through SSH or Telnet access.</li> <li>● <b>Prohibit setting Pvalue:</b> This option blocks all users from modifying any configuration parameter (Pvalue) via SSH or Telnet access.</li> </ul> <p>The Default Value is "Allow any user to set any PValue"</p>
<b>WAN Side Web/SSH Access</b>	<p>Enables/Disables the Web and SSH access through the WAN port. The available options are the following:</p> <ul style="list-style-type: none"> <li>● <b>No:</b> No access to the web or SSH from any IP address on the WAN side.</li> <li>● <b>Yes:</b> Access for the Web GUI and SSH is enabled on the WAN side.</li> <li>● <b>Auto:</b> Only private IP could access the web or SSH on the WAN side.</li> </ul> <p>Default setting is Yes.</p>
<b>White List for WAN Side</b>	<p>If WAN Side Web/SSH Access is set to Yes or Auto. Users can configure the white List for WAN Side to be used for remote management. Multiple IPs are supported and need to be separated by space.</p> <p><b>Example:</b>192.168.5.222 192.168.5.223 192.168.7.0/24</p> <p><b>Note:</b> If both blacklist and whitelist are not empty, the blacklist is processed first, followed by the whitelist.</p>
<b>Black List for WAN Side</b>	<p>If WAN Side Web/SSH Access is set to Yes or Auto. Users can configure the black List for WAN Side to ban WAN side web access. Multiple IPs are supported and need to be separated by space.</p> <p><b>Example:</b>192.168.5.222 192.168.5.223 192.168.7.0/24</p> <p><b>Note:</b> If both blacklist and whitelist are not empty, the blacklist is processed first, followed by the whitelist.</p>
<b>Internet Protocol</b>	<p>Selects one of the following IP protocol modes:</p> <ol style="list-style-type: none"> <li>1. <b>IPv4 Only:</b> Enforce IPv4 protocol only.</li> <li>2. <b>IPv6 Only:</b> Enforce IPv6 protocol only.</li> <li>3. <b>Both, Prefer IPv4:</b> Enable both IPv4 and IPv6 and prefer IPv4.</li> <li>4. <b>Both, prefer IPv6:</b> Enable both IPv4 and IPv6 and prefer IPv6.</li> </ol>

	<b>Note:</b> Make sure to reboot the HT841/HT881 unit for the changes to take effect.
<b>IPv4 Address</b>	Allows users to configure the appropriate network settings on the HT841/HT881 to obtain IPv4 address. Users could select DHCP, Static IP or PPPoE. By default, it is set to DHCP.
<b>Dynamically assigned via DHCP</b>	<p>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.</p> <ul style="list-style-type: none"> <li>● <b>DHCP hostname:</b> Specifies the name of the client. The name may or may not be qualified with the local domain name. This field is optional but may be required by ISP.</li> <li>● <b>DHCP domain name:</b> 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.</li> <li>● <b>DHCP vendor class ID:</b> Exchanges vendor class ID by clients and servers to convey particular configuration or other identification information about a client. Default is HT8XXV2.</li> </ul>
<b>Use PPPoE</b>	<p>Set the PPPoE account settings. If selected, FXO Gateway attempts to establish a PPPoE session if any of the PPPoE fields are set.</p> <ul style="list-style-type: none"> <li>● <b>PPPoE account ID:</b> Defines the PPPoE username. Necessary if ISP requires you to use a PPPoE (Point to Point Protocol over Ethernet) connection.</li> <li>● <b>PPPoE password:</b> Specifies the PPPoE account password.</li> <li>● <b>PPPoE Service Name:</b> Defines PPPoE service name. If your ISP uses a service name for the PPPoE connection, enter the service name here. This field is optional.</li> </ul>
<b>Preferred DNS server</b>	Specifies preferred DNS server to use when DHCP or PPPoE are set. You can set up yo 4 Preferred DNS Servers.
<b>Statically configured as IP address</b>	Configure IP address, subnet Mask, default router IP address, 1st preferred DNS server, 2nd preferred DNS server. These fields are set to zero by default.
<b>IPv6 Address</b>	<p>Allows users to configure the appropriate network settings on the HT841/HT881 to obtain an IPv6 address. Users could select DHCP or Static IP. By default, it is set to DHCP.</p> <ul style="list-style-type: none"> <li>● <b>DHCP:</b> 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.</li> <li>● <b>Static IP mode:</b> configure IP address, 1st and 2nd DNS server, preferred DNS server. These fields are set to zero by default. <ul style="list-style-type: none"> <li>○ <b>Full Static:</b> When enabling the option full static, users need to specify the Static IPv6 and the IPv6 Prefix length.</li> <li>○ <b>Prefix Static:</b> When enabling the option prefix static, users need to specify the IPv6 Prefix (64 bits).</li> </ul> </li> </ul>
<b>Enable Management Interface</b>	<p>Allows activation of a dedicated interface for remote management and configuration of the device.</p> <p>Disabled by Default.</p>
<b>Management Access</b>	<p>defines the allowed methods (protocols) and permissions for remote access and configuration of the device, two options can be selected</p> <ul style="list-style-type: none"> <li>● <b>Management Interface Only:</b> allows remote access and configuration solely through the dedicated management interface</li> <li>● <b>Both Service and Management Interfaces:</b> permits remote access and configuration through both the management interface and the service interface.</li> </ul>
<b>Enable SNMP Through Management Interface</b>	<p>Allows the activation of SNMP (Simple Network Management Protocol) access exclusively through the dedicated management interface for monitoring and managing the device remotely.</p> <p>Disabled by Default</p>

<b>Enable TR069 Through Management Interface</b>	Allows the activation of TR-069 protocol access exclusively through the dedicated management interface for remote device management and provisioning. Disabled by Default
<b>Enable Syslog Through Management Interface</b>	Allows the activation of syslog communication exclusively through the dedicated management interface for remote logging and monitoring of device events. Disabled by Default.
<b>Management Interface IPv4 Address</b>	Sets the specific IPv4 address assigned to the dedicated management interface for remote access and configuration.
<b>802.1Q/VLAN Tag</b>	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
<b>802.1p priority value</b>	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
<b>Dynamically assigned via DHCP</b>	Indicates that the device's network settings, such as IP address, are automatically obtained from a DHCP server rather than being manually configured. It is set as the Default setting for Management Interface IPv4 Address
<b>Statically configured as</b>	Configure IP address, subnet Mask, default router IP address, 1st preferred DNS server, 2nd preferred DNS server.  By default: <b>IP Address:</b> 192.168.100.100 <b>Subnet Mask:</b> 255.255.255.0 <b>Default Router:</b> 192.168.100.1 <b>DNS Server 1:</b> 0.0.0.0 <b>DNS Server 2:</b> 0.0.0.0
<b>Time Zone</b>	Selects time zone to define date/time on the device.
<b>Self-Defined Time Zone</b>	Allows users to define their own time zone when time zone is set to "Using self-defined Time Zone" The syntax is: <b>std offset dst [offset], start [/time], end [/time]</b> Default is set to: <b>MTZ+6MDT+5,M3.2.0,M11.1.0</b> <b>MTZ+6MDT+5</b>  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.  <b>M4.1.0,M11.1.0</b> The 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.
<b>Allow DHCP server to set Time Zone</b>	Obtains time zone setting (offset) from a DHCP server using DHCP Option 2; it will override selected time zone. If set to No, the analogue adapter will use selected time zone even if provided by DHCP server. Default is Yes.
<b>Language</b>	Configures the languages of the voice prompt and web interface. Available languages: English, Chinese, Russian, and Spanish.

NAT/DHCP Server Information & Configuration	
<b>Working Mode of Network Interface</b>	<p>Refers to the configuration of the network interfaces on a device.</p> <ul style="list-style-type: none"> <li>● <b>NET1 is WAN and NET2 is LAN (Default):</b> NET1 is set as the Wide Area Network (WAN) interface, typically used to connect to the internet or external networks, while NET2 is set as the Local Area Network (LAN) interface, used for internal local connections.</li> <li>● <b>NET1 is LAN and NET2 is WAN:</b> the roles of the interfaces are reversed, where NET1 becomes the LAN interface, and NET2 becomes the WAN interface.</li> </ul>
<b>Device Mode</b>	<p>Controls whether the device is working in NAT Router, Bridge, or WAN Only mode.</p> <ul style="list-style-type: none"> <li>● <b>NAT Router:</b> In this mode, the WAN port acts as a DHCP client. LAN port is used as DHCP Base IP; devices connected behind the LAN port will be assigned an IP from HT841/HT881 DHCP Server.</li> <li>● <b>Bridge:</b> In this mode, the WAN port acts as a DHCP client and pass-through to the LAN port; devices connected behind the LAN port will get an IP from your network DHCP server (same as the WAN port).</li> <li>● <b>WAN Only:</b> In this mode, only the WAN port is active. LAN port is not used.</li> </ul> <p>The default mode is Bridge mode. Save the setting and reboot prior to configuring the HT841/HT881.</p>
<b>NAT Maximum Ports</b>	<p>Defines the number of ports that can be managed while in NAT router mode. Range: 0 - 4096, default is 1024. Typically, one port per connection</p>
<b>NAT TCP Timeout</b>	<p>NAT TCP idle timeout in seconds. The connection will be closed after preconfigured, timeout if not refreshed. Range: 0 - 3600</p>
<b>NAT UDP Timeout</b>	<p>NAT UDP idle timeout in seconds. The connection will be closed after preconfigured, timeout if not refreshed. Range: 0 - 3600, default is 300</p>
<b>Uplink Bandwidth</b>	<p>Specifies the maximum uplink bandwidth permitted by the device. This function is disabled by default. The total bandwidth can be set as 128K, 256K, 512K, 1M, 2M, 3M, 4M, 5M, 10M, or 15 M. The primary function of this setting is to limit the uplink bandwidth for the device's internal system, signaling, and NATed traffic. <b>Example:</b> When 512k is configured, there will be at least 512kbps limited for internal system, signaling, and NATed traffic. <b>Note:</b> Voice or RTP stream will never be limited.</p>
<b>Downlink Bandwidth</b>	<p>Specifies the maximum downlink bandwidth permitted by the device. This function is disabled by default. The total bandwidth can be set as 128K, 256K, 512K, 1M, 2M, 3M, 4M, 5M, 10M, or 15 M. The primary function of this setting is to limit the download bandwidth for the device's internal system, signaling, and NATed traffic. <b>Example:</b> if 128 is configured, there will be at least 128kbps limited for internal system, signaling, and NATed traffic. <b>Note:</b> Voice or RTP stream will never be limited.</p>
<b>Enable UPnP Support</b>	<p>When set to Yes, the HT841/HT881 acts as a UPnP gateway for your UPnP-enabled applications. UPnP = Universal Plug and Play. The default is No.</p>
<b>Reply to ICMP on WAN Port</b>	<p>When set to Yes, the HT841/HT881 responds to the PING command from other computers but is also made vulnerable to DOS attacks. The default is No.</p>
<b>Cloned WAN MAC Address</b>	<p>This allows the user to change/set a specific MAC address on the WAN interface. Note: Set in Hex format.</p>
<b>LAN Port VLAN Feature Under Bridge Mode</b>	<p>This feature allows users to configure a different VLAN tag and priority value for the second network port when HT841/881 is configured in bridge mode.</p>

	The priority value range is 0-7, The VLAN tag range is 0-4094. The default VLAN Tag and Priority value are 0.
<b>Enable LAN DHCP</b>	When set to Yes, the device will function as a simple router and the LAN port will provide IP addresses to the internal network. Connect the WAN port to ADSL/Cable modem or any other equipment that provides access to the public Internet. The default setting is Yes.
<b>LAN DHCP Base IP</b>	Base IP Address for a LAN port. The default factory setting is <b>192.168.2.1</b> . <b>Note:</b> When the device detects WAN IP is conflicting with LAN IP, the LAN base IP address will be changed based on the network mask the effective subnet will be increased by 1. For example; 192.168.2.1 will be changed to 192.168.3.1 if the netmask is 255.255.255.0. Then the device will reboot
<b>LAN DHCP Start IP</b>	The default value is 100. The last segment of IP address is assigned to the HT841/HT881 in the LAN Network. Default configuration assigns IP address (to local network devices) starting from 192.168.2.100.
<b>LAN DHCP End IP</b>	Default value is 199. This parameter allows a user to limit the number of local network devices connected to the internal router.
<b>LAN Subnet Mask</b>	Sets the LAN subnet mask. Default value is 255.255.255.0
<b>DHCP IP Lease Time</b>	Default value is 120 hrs. (5 days). The length of time the IP address is assigned to the LAN clients. Value is set in units of hours.
<b>DMZ IP</b>	This function forwards all WAN IP traffic to a specific IP address if no matching port is used by HT841/HT881 or in the defined port forwarding.
<b>Port Forwarding</b>	Forwards a matching (TCP/UDP) port to a specific LAN IP address with a specific (TCP/UDP) port. Up to 8 rules are available.
<b>Reset Type</b>	Gives the administrator the option to restore the default configuration on the HT841/HT881. There are 3 types of factory reset:  <ol style="list-style-type: none"> <li><b>1. ISP Data Reset:</b> All ISP (Internet Service Provider) configurations that may affect the IP address will be reset (including WAN static IP).</li> <li><b>2. VoIP Data Reset:</b> All VoIP-related configurations (mainly everything located on FXS/FXO page).</li> <li><b>3. Full Reset:</b> Both VoIP and ISP-related configuration at the same time.</li> </ol> <b>Note:</b> After choosing the reset type, you will have to click the reset button for it to take effect.

## Advanced Settings Page Definitions

<b>New Admin Password</b>	Defines the administrator level password to access the Advanced Web Configuration page. This field is case sensitive. Only the administrator can configure the “Advanced Settings” page. Password field is purposely left blank for security reasons after clicking update and saved. <b>Note :</b> The Password must contain 8-30 characters, at least one number, one uppercase, and a lowercase letter. Purposely not displayed for security reasons
<b>Confirm Admin Password</b>	Confirms the new admin password.
<b>Disable User Level Web Access</b>	Disable or enable User Level Web Access. Default is No.

<b>Disable Viewer Level Web Access</b>	Disable or enable Viewer Level Web Access Default is No.
<b>Layer 2 QoS</b>	Sets values for 802.1Q/VLAN Tag. Default is 0. Valid range is 0-4094. SIP 802.1p. Default is 0. Valid range is 0-7. RTP 802.1p. Default is 0. Valid range is 0-7.
<b>Blacklist for WAN Side Port</b>	It could be either port range or single port separated by a “,” Example: “5000-6000, 7000 “.
<b>STUN Server is</b>	Configures IP address or domain name of STUN server. Only non-symmetric NAT routers work with STUN.
<b>Keep-alive interval</b>	Sends periodically a blank UDP packet to SIP server to keep the “ping hole” on the NAT router open. Default is 20 seconds.
<b>Use STUN to detect network connectivity</b>	Uses STUN keep-alive to detect WAN side network problems. If keep-alive request does not yield any response for configured number of times (minimum 3), the device will restart the TCP/IP stack. If the STUN server does not respond when the device boots up, the feature is disabled. Default setting is No.
<b>Use DNS to detect network connectivity</b>	Uses DNS to detect WAN side network problems. Default setting is “No”.
<b>Use ARP to detect network connectivity</b>	Uses ARP to check the network connectivity. Default is “Yes”.
<b>Verify host when using HTTPS</b>	Enables / disables the host verification when using HTTPS.
<b>Upgrade via</b>	Selects firmware upgrade/provisioning method: TFTP, HTTP, HTTPS, FTP or FTPS. Default is HTTPS.
<b>Firmware Server Path</b>	Sets IP address or FQDN of firmware server. The URL of the server that hosts the firmware release. <b>Note:</b> You can specify the protocol used in the Firmware Server Path. (example: https://192.168.5.120), this will bypass the "Upgrade Via" method. Default is fm.grandstream.com/gs.
<b>Config Server Path</b>	Sets IP address or FQDN of configuration server. The URL of the server that hosts the configuration file to provision HT8xx. <b>Note:</b> You can specify the protocol used in the Config Server Path. (example: https://192.168.5.120), this will bypass the "Upgrade Via" method. Default is fm.grandstream.com/gs.
<b>XML Config File Password</b>	Decrypts XML configuration file when encrypted. The password used for encrypting the XML configuration file using OpenSSL.
<b>HTTP/HTTPS FTP/FTPS Username</b>	Enters username to authenticate with HTTP/HTTPS FTP/FTPS server.
<b>HTTP/HTTPS FTP/FTPS Password</b>	Enters password to authenticate with HTTP/HTTPS FTP/FTPS server.
<b>Firmware File Prefix</b>	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.
<b>Firmware File Postfix</b>	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.

<b>Config File Prefix</b>	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.
<b>Config File Postfix</b>	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.
<b>Enable using tags in URL</b>	Allows users to configure variables on the configuration server path to differentiate the directories on the server.
<b>Always send HTTP Basic Authentication Information</b>	Default is No. If set to Yes, The device will send configured user name and password within HTTP request before server sends authentication challenge.
<b>Allow DHCP Option 66 or 160 to Override Server</b>	Obtains configuration and upgrade server's information using options 66 from DHCP server. Note: If DHCP Option 66 is enabled, the HT8xx 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".
<b>Additional Override DHCP Option</b>	Allows users to enable the Additional Override DHCP Option in Option 150. The default value is "None".
<b>3CX Auto Provision</b>	Sends multicast "SUBSCRIBE" message for provisioning at booting stage, used for PnP (Plug-and-Play) configuration. Default is Yes.
<b>Automatic Upgrade</b>	Specifies when the firmware upgrade process will be initiated; there are 4 options: <b>No:</b> The HT8xx will only do an upgrade once at boot up. <b>Check every X minutes:</b> User needs to specify a period in minutes. <b>Check every day:</b> User needs to specify the start hour and the end hour of the day (0-23). <b>Check every week:</b> User needs to specify "Day of the week (0-6)". (Day of week is starting from Sunday). Default is No.
<b>Randomized Automatic Upgrade</b>	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).
<b>Always Check for New Firmware at Boot up</b>	Configures the HT841/HT881 to always search for the new firmware at boot up. During the boot stage, the HT8xx will contact the firmware upgrade server to search for a new firmware, when available it will start the upgrade process, otherwise it will boot normally.
<b>Check New Firmware only when F/W pre/suffix changes</b>	Configure the HT8xx to search for the new firmware when the firmware prefix / suffix changes. When this option is selected, the HT8xx will check for updates only when the pre/suffix has been changed.
<b>Always Skip the Firmware Check</b>	Configures the HT8xx to skip the firmware check, when this option is selected the HT8xx will always skip searching for a new firmware.
<b>Configuration File Types Allowed</b>	Allows users to configure provision configuration file type in xml file only or all file types.
<b>Download and Process All Available Config Files</b>	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.xml > 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"

<b>Disable SIP NOTIFY Authentication</b>	Disables the SIP NOTIFY Authentication on the phone adapter. If set to “Yes”, the phone adapter will not challenge NOTIFY with 401. The default setting is “No”
<b>Authenticate Conf File</b>	Authenticates configuration before being accepted. This protects the configuration from unauthorized modifications. Default is No.
<b>Validate Server Certificates</b>	This feature allows users to validate server certificates with our trusted list of TLS connections. The device needs to reboot after changing the setting. Default is enabled.
<b>Trusted CA certificates A</b>	Uses the certificate for Authentication if “Check Domain Certificates” is set to “Yes” under “Account”-> “SIP Settings”.
<b>Trusted CA certificates B</b>	Uses the certificate for Authentication if “Check Domain Certificates” is set to “Yes” under “Account”-> “SIP Settings”.
<b>Trusted CA certificates C</b>	Uses the certificate for Authentication if “Check Domain Certificates” is set to “Yes” under “Account”-> “SIP Settings”.
<b>Trusted CA certificates D</b>	Uses the certificate for Authentication if “Check Domain Certificates” is set to “Yes” under “Account”-> “SIP Settings”.
<b>Load CA Certificates</b>	This feature allows user to specify which certificate to trust when performing server authentication. <b>Build-in trusted :</b> (Default) Build-in trusted certificates <b>Custom trusted certificate:</b> Uploaded Certificates <b>All trusted Certificates:</b> Both built-in and uploaded Certificates
<b>SIP TLS Certificate</b>	Specifies SSL certificate used for SIP over TLS is in X.509 format. The HT8xx has built-in private key and SSL certificate. Maximum supported length is 4069.
<b>SIP TLS Private Key</b>	Specifies TLS private key used for SIP over TLS is in X.509 format. Maximum supported length is 4069.
<b>SIP TLS Private Key Password</b>	Specifies SSL Private key password used for SIP Transport in TLS/TCP.
<b>Custom Certificate (Private Key + Certificate)</b>	Allows users to update to the device their own certificate signed by a custom CA certificate to manage client authentication.
<b>Enable TR-069</b>	Sets the phone adapter system to enable the “CPE WAN Management Protocol” (TR-069). The default setting is Yes.
<b>TR-069 firewall rules</b>	Configures the TR-069 firewall rules , Value port range is 1-65535
<b>ACS URL</b>	Specifies URL of TR-069 Auto Configuration Servers (e.g., <a href="http://acs.mycompany.com">http://acs.mycompany.com</a> ), or IP address. Default setting is: “ <a href="https://acs.gdms.cloud">https://acs.gdms.cloud</a> ”
<b>ACS Username</b>	Enters username to authenticate to ACS.
<b>ACS Password</b>	Enters password to authenticate to ACS.
<b>Periodic Inform Enable</b>	Sends periodic inform packets to ACS. Default is Yes.
<b>Periodic Inform Interval</b>	Sets frequency that the inform packets will be sent out to ACS.

	Default is 86400 seconds.
<b>Connection Request Username</b>	Enters username for ACS to connect to the HT8xx.
<b>Connection Request Password</b>	Enters password for ACS to connect to the HT8xx.
<b>Connection Request Port</b>	Configures the TR-069 connection request port. The value range is 0 to 65535.Default is 7547
<b>CPE SSL Certificate</b>	Configures the Cert File for the phone adapter to connect to the ACS via SSL.
<b>CPE SSL Private Key</b>	Specifies the Cert Key for the phone adapter to connect to the ACS via SSL.
<b>Enable SNMP</b>	Default is No.
<b>SNMP Version</b>	Choose between (Version 1, Version 2c, or Version 3).
<b>SNMP Port</b>	Listening Port of SNMP daemon (Default 161).
<b>SNMP Trap IP Address</b>	IP address of trap destination. Up to 3 trap destinations are supported. Users should enter the IP addresses separated with comma (.).
<b>SNMP Trap port</b>	Port of Trap destination (Default 162)
<b>SNMP Trap Version</b>	Choose between (Version 1, Version 2c, or Version 3).
<b>SNMP Trap Interval</b>	Time interval between traps (Default is 5).
<b>SNMPv1/v2c Community</b>	Name of SNMPv1/v2c community.
<b>SNMPv1/v2c Trap Community</b>	Name of SNMPv1/v2c trap community.
<b>SNMPv3 Username</b>	Username for SNMPv3.
<b>SNMPv3 Security Level</b>	<b>noAuthUser:</b> Users with security level noAuthnoPriv and context name as noAuth. <b>authUser:</b> Users with security level authNoPriv and context name as auth. <b>privUser:</b> Users with security level authPriv and context name as priv.
<b>SNMPv3 Authentication Protocol</b>	Select the Authentication Protocol: "None" or "MD5" or "SHA."
<b>SNMPv3 Privacy Protocol</b>	Select the Privacy Protocol: "None" or "AES/AES128" or "DES".
<b>SNMPv3 Authentication Key</b>	Enter the Authentication Key.
<b>SNMPv3 Privacy Key</b>	Enter the Privacy Key.
<b>SNMPv3 Trap Username</b>	Username for SNMPv3 Trap.
<b>SNMPv3 Trap Security Level</b>	<b>noAuthUser:</b> Users with security level noAuthnoPriv and context name as noAuth. <b>authUser:</b> Users with security level authNoPriv and context name as auth. <b>privUser:</b> Users with security level authPriv and context name as priv.
<b>SNMPv3 Trap Authentication Protocol</b>	Select the Authentication Protocol: "None" or "MD5" or "SHA".

<b>SNMPv3 Trap Privacy Protocol</b>	Select the Privacy Protocol: "None" or "AES/AES128" or "DES".
<b>SNMPv3 Trap Authentication Key</b>	Enter the Trap Authentication Key.
<b>SNMPv3 Trap Privacy Key</b>	Enter the Trap Privacy Key.
<b>Enable RADIUS Web Access Control</b>	Enables RADIUS Web Access Control. Default is "No".
<b>Action upon RADIUS Auth Server Error</b>	Choose action upon RADIUS server error. Default is Authenticate Locally (Default Authenticate Locally)
<b>RADIUS Auth Protocol</b>	<p>Configure RADIUS authentication protocol, the available options are:</p> <p><b>PAP (Password Authentication Protocol):</b> PAP sends the username and password in plaintext, making it less secure than other methods.</p> <p><b>CHAP (Challenge Handshake Authentication Protocol):</b> CHAP uses a challenge-response mechanism for authentication, enhancing security.</p> <p><b>MsCHAPv1 (Microsoft Challenge Handshake Authentication Protocol version 1):</b> An older Microsoft-specific CHAP variant.</p> <p><b>MsCHAPv2 (Microsoft Challenge Handshake Authentication Protocol version 2):</b> A more secure and improved Microsoft-specific CHAP variant with mutual authentication support.</p> <p>Default value is "PAP".</p>
<b>RADIUS Auth Server Address</b>	Address of RADIUS Auth server.
<b>RADIUS Auth Server Port</b>	Port of RADIUS Auth server.
<b>RADIUS Shared Secret</b>	Set RADIUS shared secret.
<b>RADIUS VSA Vendor ID</b>	Configure RADIUS VSA Vendor ID to match RADIUS server's configuration. Default is 42397 for Grandstream Networks Inc.
<b>RADIUS VSA Access Level Attribute</b>	Configure RADIUS VSA Access Level Attribute to match RADIUS server's configuration. Incorrect setting would cause Radius authenticate fail.
<b>Enable OpenVPN®</b>	Allow user to enable OpenVPN®. Default is No.
<b>OpenVPN® Server Address</b>	Specify the IP address or FQDN for the OpenVPN® Server.
<b>OpenVPN® Port</b>	Specify the listening port of the OpenVPN® server. Default is 1194
<b>OpenVPN® Interface type</b>	Specify the Interface type of OpenVPN® whether TAP or TUN. Default is TUN.
<b>OpenVPN® Transport</b>	Specify the Transport Type of OpenVPN® whether UDP or TCP. The default is UDP.
<b>Enable OpenVPN® LZO Compression</b>	Enable OpenVPN® LZO Compression. Default is Yes.
<b>OpenVPN® Encryption</b>	Select the OpenVPN® Encryption. Default is BF-CBC 128 bit (default key).
<b>OpenVPN® Digest</b>	Select the OpenVPN® Digest. Default is SHA1.

<b>OpenVPN® CA</b>	Specifies the OpenVPN® CA. Maximum Character Number is 8192.
<b>OpenVPN® Certificate</b>	Specifies the OpenVPN® Certificate. Maximum Character Number is 8192.
<b>OpenVPN® Client Key</b>	Specifies the Client Key. Maximum Character Number is 8192.
<b>OpenVPN® Client Key Password</b>	Configures the OpenVPN® Client Key Password. Maximum Length is 64.
<b>OpenVPN® username</b>	Configure the OpenVPN® username.
<b>OpenVPN® password</b>	Configure the OpenVPN® password
<b>System Ring Cadence</b>	Configuration option is set ring cadence on FXS port for all incoming calls. Syntax: c=on1/off1-on2/off2-on3/off3; (3 cadences maximum) Default is set to c=2000/4000; (US standards)
<b>Call Progress Tones:</b> Dial Tone Ringback Tone Busy Tone Reorder Tone Confirmation Tone Call Waiting Tone Prompt 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: f1=350@-13, f2=440@-13, c=0/0; Syntax: f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3; [...] (Note: freq: 0 – 4000Hz; vol: -30 – 0dBm) * “Conference Party Hang-up Tone” will apply only if the “Special Feature” is set to “MTS”. Special Proceed Indication Tone: This feature allows user to configure the tone played when user goes offhook and there is voicemail on the subscribed mailbox. Need to set ‘MWI Tone’ to ‘Special Proceed Indication Tone’ to use this feature.
<b>Prompt Tone Access Code</b>	Key pattern to get Prompt Tone. Maximum 20 digits.
<b>Lock Keypad Update</b>	Configuration update via keypad (analog phone connected to FXS port keypad using IVR menu) is disabled if set to Yes.
<b>Disable Voice Prompt</b>	Voice prompt is disabled if set to Yes.
<b>Disable Direct IP Call</b>	Direct IP call is disabled if set to Yes.
<b>Play Busy Tone When Account is unregistered</b>	When this feature is set to Yes, device will play busy tone when the FXS port account is not registered, and the attached analog phone is offhook. Disabled by Default.
<b>Life Line Mode</b>	Designates a backup communication route that becomes active when the primary route fails, ensuring continuous connectivity and call reliability. You can set this option to Auto, Always Disconnected, or Always Connected: <ul style="list-style-type: none"><li>● <b>Auto:</b> In case of power loss or loss of SIP registration, the PSTN line will be seamlessly connected to analog phone connected to FXS port.</li><li>● <b>Always Connected:</b> PSTN line will be always connected to the phone connected to FXS port. VoIP calls will not be allowed in this configuration.</li><li>● <b>Always Disconnected:</b> User can only make/receive VoIP calls. PSTN calls will not be possible. Default setting is Auto.</li></ul> Default value is <b>Auto</b>
<b>Blacklist for Incoming Calls</b>	Allow users to block incoming calls from specific list of numbers.

	<p>Maximum allow 10 SIP numbers and each number should be separated by a comma (',') in web UI.</p> <p>Other allowed characters are 0-9, comma (','), asterisk (*), pound sign ('#') and plus sign ('+').</p>
<b>NTP Server</b>	<p>Defines the URL or IP address of the NTP server. The FXO Gateway may obtain the date and time from the server. The default setting is "pool.ntp.org."</p>
<b>Allow DHCP Option 42 to override NTP server</b>	<p>Defines whether DHCP Option 42 should override NTP server or not. When enabled, DHCP Option 42 will override the NTP server if it is set up on the LAN. The default setting is Yes.</p>
<b>DHCP Option 17 Enterprise Number</b>	<p>Configure the DHCP option 17 number. Default is 3561</p>
<b>CDR File Option</b>	<p>By default, the device will split the allowed memory for CDR file into 2 parts. Device will create the first CDR file which is half of the allowed size, when it is full, device will create the second file.</p> <p>When "CDR File Option" is set to Default "Keep", device will keep the call records when both files are full, no more new record will be stored.</p> <p>When this feature is set to "Overwrite", device will clear the first CDR file and start storing again.</p> <p>When the feature is disabled the device will not record any calls.</p> <p>The CDR file output will be available at Status page: [CDR File]</p>
<b>SIP File Option</b>	<p>By default, the device will split the allowed memory for SIP file into 2 parts. Device will create the first SIP file which is half of the allowed size, when it is full, device will create the second file.</p> <p>When "SIP File Option" is set to Default "Keep", device will keep the call records when both files are full, no more new record will be stored.</p> <p>When this feature is set to "Overwrite", device will clear the first SIP file and start storing again.</p> <p>The SIP file output will be available at Status page: [SIP File]</p> <p>Note: "Send SIP Log" must be enabled to be able to capture the trace.</p>
<b>Disable Weak TLS Cipher Suites</b>	<p>Allows users to disable weak ciphers DES/3DES and RC4, Symmetric Encryption SEED, Symmetric Authentication MD5, Protocol Version SSLv2/SSLv3 or Disable All of the Above Weak TLS Ciphers Suites.</p> <p>Default is No.</p>
<b>Minimum TLS Version</b>	<p>This feature allows customer to choose desired Minimum TLS Version.</p> <p>Choices are:</p> <ul style="list-style-type: none"> <li>Unlimited</li> <li>TLS 1.0</li> <li>TLS 1.1</li> <li>TLS 1.2</li> </ul> <p>Default is Unlimited.</p>
<b>Maximum TLS Version</b>	<p>This feature allows customer to choose desired Maximum TLS Version.</p> <p>Choices are:</p> <ul style="list-style-type: none"> <li>Unlimited</li> <li>TLS 1.0</li> <li>TLS 1.1</li> <li>TLS 1.2</li> </ul> <p>Default is Unlimited.</p>
<b>Syslog Protocol</b>	<p>If set to SSL/TLS, the syslog messages will be sent through secured TLS Protocol to syslog server.</p> <p>Default setting is UDP.</p> <p>Note: The CA certificate is required to connect with the TLS server</p> <p>A reboot is required to take effect.</p>

<b>Syslog Server</b>	URL or IP address of syslog server. Note: A reboot is required to take effect.
<b>Syslog Level</b>	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: <ol style="list-style-type: none"> <li>1. product model/version on boot up (INFO level)</li> <li>2. NAT related info (INFO level)</li> <li>3. sent or received SIP message (DEBUG level)</li> <li>4. SIP message summary (INFO level)</li> <li>5. inbound and outbound calls (INFO level)</li> <li>6. registration status change (INFO level)</li> <li>7. negotiated codec (INFO level)</li> <li>8. Ethernet link up (INFO level)</li> <li>9. SLIC chip exception (WARNING and ERROR levels)</li> <li>10. memory exception (ERROR level)</li> </ol> extra syslog style (EXTRA DEBUG level) Note: A reboot is required to take effect.
<b>Send SIP Log</b>	Configures whether the SIP log will be included in the syslog messages. The default setting is No.
<b>Ethernet Capture</b>	Enables the capturing and analysis of Ethernet network traffic for troubleshooting and monitoring purposes.
<b>Automatic Reboot</b>	Default is No. When “Yes, reboot every day at hour” or “Yes, reboot every week at day” or “Yes, reboot every month at day” is checked, user can specify “Hour of the day (0-23)” or “Day of the week (0-6)” or “Day of the month (0-30)”. Default time is Monday 1AM.
<b>Port Record</b>	Allows the recording of port-level call information, such as call duration, caller ID, and call status, for monitoring and logging purposes.
<b>Download Device Configuration</b>	Press Download button to download device configuration file to local computer. The filename is “config.txt”. The file is plain text and not including password fields.
<b>Download Device XML Configuration</b>	Press Download to download device configuration file to local computer. The filename is “config.xml”. The file will not include password fields.
<b>Upload Firmware</b>	Press Upload from local directory button to load the firmware file to the device from your computer. The firmware filename should be “ht8x1fw.bin”
<b>Upload Configuration</b>	Press Upload from local directory button to load configuration file to the device from your computer. The configuration file should be an XML file (for instance: “config.xml”). Note: The field is not mandatory in the document but if available only device with specified MAC address will accept the configuration file.
<b>Export Backup Configuration</b>	Press Download button to export device backup configuration to computer. The output is “cfg<mac>_enc.xml” (where <mac> is the MAC address of the device). The file is encrypted and can be used on same device only.
<b>Restore From Backup Configuration</b>	Press Upload button to restore device configuration from previously exported backup configuration.
<b>E911/HELD Protocol:</b>	
<b>Enable E911</b>	Enable Enhanced 911 call. Default is disabled

<b>HELD Protocol</b>	Configure HELD transfer protocol. HTTP or HTTPS
<b>HELD Synchronization Interval</b>	The valid synchronization interval is between 30 to 1440 minutes. The synchronization is off when the interval is 0.
<b>Location Server</b>	Configure the primary Location Information Server (LIS) address
<b>Location Server Username</b>	Configure the user name of the primary Location Information Server (LIS)
<b>Location Server Password</b>	Configure the password of the primary Location Information Server (LIS)
<b>Secondary Location Server</b>	Configure the secondary Location Information Server (LIS) address
<b>Secondary Location Server Username</b>	Configure the user name of the secondary Location Information Server (LIS)
<b>Secondary Location Server Password</b>	Configure the password of the secondary Location Information Server (LIS)
<b>HELD Location Types</b>	Configure "locationType" element in the location request. "geodetic", "civic" and "location URI"
<b>HELD NAI</b>	Configure "locationType" element in the location request. "geodetic", "civic" and "location URI"
<b>HELD Identity 1-10</b>	HELD Identity
<b>HELD Identity 1-10 Value</b>	HELD Identity value
<b>E911 Emergency Numbers</b>	A user can configure multiple emergency numbers separated with the delimiter symbol ";".
<b>Geolocation-Routing Header</b>	If "Yes", E.911 INVITE message includes the "Geolocation-Routing" header with the value "Yes"
<b>Priority Header</b>	If "Yes", E.911 INVITE message includes the "Priority" header with the value "emergency"

## FXS Profile Settings Page Definitions

<i>FXS Profile</i>	
<b>Profile Active</b>	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.
<b>Primary SIP Server</b>	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 HT841/HT881.
<b>Failover SIP Server</b>	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.
<b>Prefer Primary SIP Server</b>	Selects to prefer primary SIP server. The account will register to primary Server if registration with Failover server expires. Default is No.

<b>Outbound Proxy</b>	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 HT841/HT881 for firewall or NAT penetration in different network environments. If symmetric NAT is detected, STUN will not work and only outbound proxy can correct the problem
<b>Backup Outbound Proxy</b>	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.
<b>Prefer Primary Outbound Proxy</b>	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.
<b>From Domain</b>	allows users to add the actual domain name, it will override the from header.This is an optional configuration.
<b>Allow DHCP Option 120 (override SIP Server)</b>	Configures the HT841/HT881 to collect SIP server address from DHCP option 120. Default is No.
<b>SIP transport</b>	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.
<b>SIP URI Scheme When Using TLS</b>	Specifies if "sip" or "sips" will be used when TLS/TCP is selected for SIP Transport. The default setting is "sips".
<b>Use Actual Ephemeral Port in Contact with TCP/TLS</b>	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.
<b>NAT Traversal</b>	<p>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.</p> <ul style="list-style-type: none"> <li>• <b>No:</b> Disables NAT traversal, potentially causing connection issues for the ATA if it's behind a NAT router.</li> <li>• <b>Keep-alive:</b> Periodically sends signals to maintain the NAT mapping for the FXO Gateway, preventing disconnection.</li> <li>• <b>STUN:</b> Utilizes the STUN protocol to discover and adapt to NAT settings for seamless VoIP communication.</li> <li>• <b>UPnP:</b> Allows the FXO Gateway to dynamically configure NAT port mappings, streamlining VoIP traffic.</li> <li>• <b>VPN:</b> Requires the FXO Gateway to establish a secure VPN connection for VoIP, bypassing NAT restrictions.</li> <li>• <b>Auto:</b> Automatically selects the most suitable NAT traversal method based on network conditions, ensuring reliable FXO Gateway connectivity.</li> </ul> <p>Default setting is No.</p>
<b>DNS Mode</b>	<p>Selects DNS mode to use for the client to look up server. One mode can be chosen.</p> <p><b>A Record (Default):</b> resolves IP Address of target according to domain name.</p> <p><b>SRV:</b> DNS SRV resource records indicate how to find services for various protocols.</p> <p><b>NAPTR/SRV:</b> Naming Authority Pointer according to RFC 2915.</p> <p><b>Use Configured IP:</b> 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.</p>
<b>DNS SRV Failover Mode</b>	<p>Configure the preferred IP mode when DNS Mode is SRV or NAPTR/SRV.</p> <ul style="list-style-type: none"> <li>• 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.</li> <li>• <b>Saved one until DNS TTL</b> SIP request will always be sent to the registered IP address until</li> </ul>

	<p>DNS TTL expires or registered IP address is unreachable.</p> <ul style="list-style-type: none"> <li>• <b>Saved on until no response</b> SIP request will always be sent to the registered IP address only until registered IP address is unreachable.</li> <li>• <b>Failback follows failback expiration time:</b> the primary server regains control only after the failback expiration time, allowing the secondary server to handle requests until then.</li> </ul>
<b>Failback Timer</b>	<p>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).</p>
<b>Register before DNS SRV Failover</b>	<p>This feature is used to control whether the device need 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.</p>
<b>Primary IP</b>	<p>The main IP address used for communication and control.</p>
<b>Backup IP1</b>	<p>The secondary IP address used as a backup for communication in case the Primary IP fails.</p>
<b>Backup IP2</b>	<p>The third IP address used as an additional backup for communication in case both the Primary IP and Backup IP1 fail.</p>
<b>TEL URI</b>	<p>Indicates E.164 number in “From” header by adding “User=Phone” parameter or using “Tel:” in SIP packets, if the HT841/HT881 has an assigned PSTN Number.</p> <p><b>Disabled:</b> Use “SIP User ID” information in the Request-Line and “From” header.</p> <p><b>User=Phone:</b> “User=Phone” parameter will be attached to the Request-Line and “From” header in the SIP request to indicate the E.164 number. If set to “Enable”.</p> <p><b>Enabled:</b> “Tel:” will be used instead of “sip:” in the SIP request.</p> <p>Please consult your carrier before changing this parameter. The default is Disabled.</p>
<b>SIP Registration</b>	<p>Controls whether the HT841/HT881 needs to send REGISTER messages to the proxy server. The default setting is Yes.</p>
<b>Unregister on Reboot</b>	<p>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".</p> <ol style="list-style-type: none"> <li>1. <b>Set to "No":</b> 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.</li> <li>2. <b>Set to "All":</b> 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.</li> <li>3. <b>Set to "Instance":</b> 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.</li> </ol> <p>Default value is set to "No"</p>
<b>Outgoing Call Without Registration</b>	<p>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.</p>
<b>Register Expiration</b>	<p>Refreshes registration periodically with specified SIP proxy (in minutes). Maximum interval is 65535 minutes (about 45 days).</p> <p>Default is 60 minutes (or 1 hour).</p>
<b>Reregister Before Expiration</b>	<p>Sends re-register request after specific time (in seconds) to renew registration before the previous registration expires.</p>

<b>SIP Registration Failure Retry Wait Time</b>	Sends re-register request after specific time (in seconds) when registration process fails. Maximum interval is 3600 seconds (1 hour). Default is 20 seconds.
<b>SIP Registration Failure Retry Wait Time upon 403 Forbidden</b>	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.
<b>MWI Tone</b>	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
<b>Enable SIP OPTIONS/NOTIFY Keep Alive</b>	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.
<b>SIP OPTIONS/NOTIFY Keep Alive Interval</b>	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.
<b>SIP OPTIONS/NOTIFY Keep Alive Max Lost</b>	Defines the Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3.
<b>Layer 3 QoS</b>	Defines Diff-Serv values for SIP and RTP. SIP DSCP (Diff-Serv value in decimal, 0-63, default 26) RTP DSCP (Diff-Serv value in decimal, 0-63, default 46)
<b>Local SIP Port</b>	Defines local port to use by the HT841/HT881 for listening and transmitting SIP packets. The default value for FXS 1 is 5060 and 5062 for FXS 2.
<b>Local RTP Port</b>	Defines the local RTP-RTCP port pair the HT841/HT881 will listen and transmit. It is the HT841/HT881 RTP port for channel 0. The default value for FXS port is 5004
<b>Use Random SIP Port</b>	Controls whether to use configured or random SIP ports. This is usually necessary when multiple HT841/HT881 are behind the same NAT. The default is No.
<b>Use Random RTP Port</b>	Controls whether to use configured or random RTP ports. This is usually necessary when multiple HT841/HT881 are behind the same NAT. The default is No.
<b>Enable RTCP</b>	Allows users to enable RTCP. The default setting is "Yes".
<b>Hold Target Before Refer</b>	Allows user to hold or not hold the phone call before referring. The default setting is Yes.
<b>Refer-To Use Target Contact</b>	Includes target's "Contact" header information in "Refer-To" header when using attended transfer. Default is No.
<b>Transfer on Conference Hang-up</b>	If set to "Yes", when the phone hangs up as the conference initiator, the conference call will be transferred to the other parties so that other parties will remain in the conference call. Default setting is No.
<b>Remove OBP from Route Header</b>	Removes outbound proxy info in "Route" header when sending SIP packets. Default setting is No.
<b>Support SIP Instance ID</b>	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
<b>Validate Incoming SIP Messages</b>	Validates incoming SIP messages. Default is No.
<b>Check SIP User ID for Incoming INVITE</b>	Checks the SIP User ID in the Request URI of the incoming INVITE; if it doesn't match the HT841/HT881 SIP User ID, the call will be rejected. Direct IP calling will also be disabled. The default is No.
<b>Authenticate Incoming INVITE</b>	Challenges the incoming INVITE for authentication with SIP 401 Unauthorized message. Default is No.
<b>Authenticate Server Certificate Domain</b>	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".
<b>Authenticate server certificate chain</b>	Configures whether to validate the server certificate when download the firmware/config file. If it is set to "Yes", the phone will download the firmware/config file only from the legitimate server. The default setting is "No".
<b>Allow Incoming SIP Messages from SIP Proxy Only</b>	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.
<b>Use Privacy Header</b>	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.
<b>Use P-Preferred-Identity Header</b>	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.
<b>Use P-Access-Network-Info Header</b>	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.
<b>Use P-Emergency-Info Header</b>	This feature support of IEEE-48-addr and IEEE-EUI-64 in SIP header for emergency calls.
<b>Use P-Asserted-Identity Header</b>	When this feature is set to Yes, device will send P-Asserted-Identity Header on the SIP Invite. Default setting is No.
<b>Caller ID Fetch Order</b>	Selects the Caller ID display order which need to be respected by the ATA. The available options are: <b>Auto:</b> When set to "Auto", the ATA will look for the caller ID in the order of P-Asserted Identity Header, Remote-Party-ID Header and From Header in the incoming SIP INVITE. <b>Disabled:</b> When set to "Disabled", all incoming calls are displayed with "Unavailable". <b>From Header:</b> When set to "From Header", the ATA will use the FROM header to display the caller ID.
<b>Allow SIP Factory Reset</b>	Allows to reset the devices directly through SIP Notify. If "Allow SIP Factory Reset" is set to "YES" under FXS PORT, then the ATA receives the NOTIFY from the SIP server with Event: reset, the HT should perform a factory reset after the authentication. The authentication in this case can be either with: The admin password if no SIP account is configured on the HT. With the credentials of the SIP account if configured on the ATA.
<b>Maximum Number of SIP Request Retries</b>	This feature allows user to configure the number of SIP retries before failover occurs. (between 1 and 10, default is 2).

<b>SIP T1 Timeout</b>	Defines T1 timeout value. It is an estimate of the round-trip time between the client and server transactions. For example, HT841/HT881 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.
<b>SIP T2 Interval</b>	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.
<b>SIP Timer D</b>	Configures SIP Timer D defined in RFC3261. 0 – 64 seconds. Default is 0.
<b>DTMF Payload Type</b>	Defines payload type for DTMF using RFC2833.
<b>Preferred DTMF method (in order)</b>	Sorts DTMF methods (in-audio, via RTP (RFC2833) or via SIP INFO) by priority.
<b>Inband DTMF Duration</b>	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.
<b>Inband DTMF Tx Gain</b>	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
<b>DSP DTMF Detector Duration Threshold</b>	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.
<b>DSP DTMF Detector Min Level</b>	Defines the minimum level required for DTMF tone to be considered valid. Default value is -25dBm from the valid range -45dBm to 0dBm
<b>DSP DTMF Detector Snr</b>	Defines the maximum signal-to-noise ratio required for DTMF tone to be considered valid. Default value is set to 1, from the range 0-2
<b>DSP DTMF Detector Deviation</b>	Defines the deviation above and below the central frequency. Default value is 25% from the valid range 0-25
<b>DSP DTMF Detector Twist</b>	Defines the maximum value by which one frequency energy level can surpass the another frequency energy level. Default value is 5dB from the valid range 0-12
<b>Disable DTMF Negotiation</b>	Uses above DTMF order without negotiation. Default is No.
<b>Send Hook Flash Event</b>	Default is No. If set to yes, flash will be sent as DTMF event.
<b>Flash Digit Control</b>	When it set to YES it allows the user to perform some call setting when both channels are used while pressing: “Flash + 1” in order to hang up the current call and resume a call that was held. “Flash + 2” in order to hold the current call and resume a call that was held. “Flash + 3” in order to perform 3-way conference. “Flash + 4” in order to perform attended transfer. Note: Please refer to the user guide for detailed steps to perform above operations. More additional Digit events were added on the new firmware 1.0.43.10.
<b>Enable Call Waiting alert-info In 180 Ringing Response</b>	When set to Yes, Alert-Info header will be added in 180 Ringing for Call Waiting case

<b>Callee Flash to 3WC</b>	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.
<b>Off Hook Auto Dial Delay</b>	Specifies the auto-dial delay after off hook.
<b>Proxy-Require</b>	Determines a SIP Extension to notify the SIP server that the HT841/HT881 is behind a NAT/Firewall.
<b>Use NAT IP</b>	Defines NAT IP address used in SIP/SDP messages. It should only be used if required by ITSP.
<b>SIP User-Agent</b>	This feature allows users to configure SIP User Agent. If not configured, device will use the default User Agent header.
<b>SIP User-Agent Postfix</b>	Configures the SIP User-Agent Postfix
<b>Add MAC in User-Agent</b>	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 REGISTER SIP 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.
<b>Use MAC Header</b>	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.
<b>RFC2543 Hold</b>	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.
<b>Disable Call Waiting</b>	Disables receiving a second incoming call when the line is engaged. Default is No.
<b>Disable Call Waiting Caller ID</b>	Disables displaying caller ID when receiving a second incoming call. Default is No.
<b>Disable Call Waiting Tone</b>	Disables playing call waiting tone during active call when receiving a second incoming call. The CWCID will still be displayed. Default is No.
<b>Disable Connected Line ID</b>	Disables displaying the number of the person answering the phone. Default is No.
<b>Disable Receiver Off Hook Tone</b>	Enables / disables the warning to alert that the phone has been left off-hook for an extended period of time. Default is No.
<b>Disable Reminder Ring for On-Hold Call</b>	Enables playing the reminder ring. Default is No.
<b>Disable Reminder Ring for DND</b>	This feature allows user to disable reminder ring when FXS port is on DND mode. Default is Yes.
<b>Disable Visual MWI</b>	Disables use of visual message waiting indicator when there is an unread voicemail message. Default is No.
<b>Visual MWI Type</b>	Configures Visual WMI Type of signal sent to the analog phone to make it turn the lamp ON upon receiving a Voice mail.

	<p>Check the phone's manual to find out what signal is supported, FSK (default) or Neon.</p> <p><b>Note:</b> 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.</p>
<b>Do Not Escape '#' as %23 in SIP URI</b>	<p>Replaces # by %23 in some special situations.</p> <p>Default is No.</p>
<b>Disable Multiple m Line in SDP</b>	<p>Sends only one m line in SDP, regardless of how many m fields are in the incoming SDP.</p> <p>Default is No.</p>
<b>Ring Timeout</b>	<p>Stops ringing when incoming call if not answered within a specific period of time. When set to 0 there will be no ringing timeout.</p> <p>Default is 60 seconds.</p>
<b>Delayed Call Forward Wait Time</b>	<p>Forwards incoming call if not answered within a specific period of time when delayed call forward is activated locally (using *92 code). Default value is 20 seconds.</p>
<b>No Key Entry Timeout</b>	<p>Initiates the call within this time interval if no additional key entry during dialing stage. Default is 4 seconds.</p>
<b>Early Dial</b>	<p>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).</p> <p>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.</p> <p>This feature is NOT designed to work with and should NOT be enabled for direct IP-to-IP calling.</p>
<b>Dial Plan Prefix</b>	<p>Adds specified prefix to dialed number.</p>
<b>Use # as Dial Key</b>	<p>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.</p>
<b>Disable # as Redial Key</b>	<p>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 No.</p>
<b>Dial Plan</b>	<p>Dial Plan Rules:</p> <ol style="list-style-type: none"> <li>1. Accept Digits: <b>1,2,3,4,5,6,7,8,9,0 , * , #, A,a,B,b,C,c,D,d</b></li> <li>2. Grammar: <b>x</b> – any digit from 0-9; <ol style="list-style-type: none"> <li>a. <b>xx+</b> – at least 2 digits number;</li> <li>b. <b>xx</b> – exactly 2 digits number;</li> <li>c. <b>^</b> – exclude;</li> <li>d. <b>.</b> – wildcard, matches one or more characters</li> <li>e. <b>[3-5]</b> – any digit of 3, 4, or 5;</li> <li>f. <b>[147]</b> – any digit 1, 4, or 7;</li> <li>g. <b>&lt;2=011&gt;</b> – replace digit 2 with 011 when dialing</li> <li>h. <b>&lt;=1&gt;</b> – add a leading 1 to all numbers dialed, vice versa will remove a 1 from the number dialed</li> <li>i. <b> </b> – or</li> <li>j. Flag <b>T</b> 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 dial 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.</li> </ol> </li> </ol> <p>Example 1: {[369]11   1617xxxxxxx} – Allow 311, 611, 911, and any 10-digit numbers of leading digits 1617</p> <p>Example 2: {^1900x+   &lt;=1617&gt;xxxxxxx} – Block any number with leading digits 1900 and add prefix 1617 for any dialed 7-digit numbers</p> <p>Example 3: {1xxx[2-9]xxxxxx   &lt;2=011&gt;x+} –</p>

	<p>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.</p> <p>1. Default: Outgoing – { x+   +x+   *x+   *xx*x+ }</p> <p>Example of a simple dial plan used in a Home/Office in the US:  { ^1900x.   &lt;=1617&gt;[2-9]xxxxxx   1[2-9]xx[2-9]xxxxxx   011[2-9]x.   [3469]11 }</p> <p>Explanation of example rule (reading from left to right):  ^1900x. – prevents dialing any number started with 1900  &lt;=1617&gt;[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</p> <p>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.</p>
<b>SUBSCRIBE for MWI</b>	Sends SUBSCRIBE periodically (depends on “Register Expiration” parameter) for message waiting indication. Default is No.
<b>Send Anonymous</b>	Sets “From”, “Privacy” and “P_Asserted_Identity” headers in outgoing INVITE message to “anonymous”, blocking caller ID. Default is No.
<b>Anonymous Call Rejection</b>	Rejects incoming calls with anonymous caller ID with “486 Busy here” message. Default is No.
<b>Special Feature</b>	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.
<b>Enable Session Timer</b>	Disable the session timer when this option is set to “No”. By default, this option is enabled.
<b>Session Expiration</b>	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.
<b>Min-SE</b>	Defines Minimum session expiration (in seconds). Default is 90 seconds.
<b>Caller Request Timer</b>	Uses session timer when making outbound calls if remote party supports it. Valid range is 90-64800 seconds. Default is No.
<b>Callee Request Timer</b>	Uses session timer when receiving inbound calls with session timer request. Default is No.
<b>Force Timer</b>	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.
<b>UAC Specify Refresher</b>	Specifies which end will act as refresher for outgoing calls. Default is Omit. <b>UAC:</b> The HandyTone acts as the refresher. <b>UAS:</b> Callee or proxy server act as the refresher.
<b>UAS Specify Refresher</b>	Specifies which end will act as refresher for incoming calls. Default is Omit.: <b>UAS:</b> The HandyTone acts as the refresher. <b>UAC:</b> Callee or proxy server act as the refresher.

<b>Force INVITE</b>	Uses INVITE message to refresh the session timer. Default is No.
<b>When To Restart Session After Re-INVITE received</b>	Allows users to delay posting Media Change Event, it can be set to "Immediately" or to "After replying 200OK" The default value is "Immediately".
<b>Enable 100rel</b>	Appends "100rel" attribute to the value of the required header of the initial signaling messages. Default is No.
<b>Add Auth Header on Initial REGISTER</b>	Adds "Authentication" header with blank "nonce" attribute in the initial SIP REGISTER request. Default is No.
<b>Enable Call Waiting Alert-Info in 180 Ringing Response</b>	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.
<b>Conference URI</b>	Allows users to manually configure the conference URL. The default is null.
<b>Use First Matching Vocoder in 200OK SDP</b>	Includes only the first matching vocoder in its 200OK response, otherwise it will include all matching vocoders in same order received in INVITE. Default is No.
<b>Preferred Vocoder</b>	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
<b>Voice Frames per TX</b>	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.
<b>G723 Rate</b>	Operates at specified encoding rate for G.723 vocoder. Available encoding rates are 6.3kbps or 5.3kbps. Default is 6.3kbps.
<b>iLBC Frame Size</b>	Specifies iLBC packet frame size (20ms or 30ms). Default is 20ms.
<b>Disable OPUS Stereo in SDP</b>	Disables OPUS stereo in SDP. Default is No.
<b>iLBC Payload Type</b>	Determines payload type for iLBC. Valid range is between 96 and 127. Default is 97.
<b>OPUS Payload Type</b>	Determines payload type for OPUS. Valid range is between 96 and 127. Default is 123.
<b>VAD</b>	Allows detecting the absence of audio and conserves bandwidth by preventing the transmission of "silent packets" over the network. Default is No.
<b>Symmetric RTP</b>	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.
<b>Fax Mode</b>	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.
<b>Re-Invite after Fax Tone Detected</b>	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

<b>Jitter Buffer Type</b>	Selects jitter buffer type (Fixed or Adaptive) based on network conditions.
<b>Jitter Buffer Length</b>	<b>High</b> (initial 200ms, min 40ms, max 600ms) Note: not all vocoders can meet the high requirement. <b>Medium</b> (initial 100ms, min 20ms, max 200ms). <b>Low</b> (initial 50ms, min 10ms, max 100ms).
<b>SRTP Mode</b>	Selects SRTP mode to use (“Disabled”, “Enabled but not forced”, or “Enabled and forced”). Default is Disabled It uses SDP Security Description to exchange key. Please refer to <b>SDES</b> : <a href="https://tools.ietf.org/html/rfc4568">https://tools.ietf.org/html/rfc4568</a> <b>SRTP</b> : <a href="https://www.ietf.org/rfc/rfc3711.txt">https://www.ietf.org/rfc/rfc3711.txt</a>
<b>SRTP Key Length</b>	Allows users to select supported SRTP Key Length. the available values are :  1. AES 128&256 bit 2. AES 128 bit 3. AES 256 bit  Set to AES 128&256 bit By Default.
<b>Crypto Life Time</b>	Adds crypto life time header to SRTP packets. Default is Yes.
<b>SLIC Setting</b>	Depends on standard phone type (and location).
<b>Caller ID Scheme</b>	Selects the caller id scheme, for example: Bellcore/Telcordia, ETSI-FSK ...
<b>Disable Unknown Caller ID</b>	Disable analog phone’s caller ID when receiving a call with “Anonymous”, “unavailable” or “unknown” in FROM header and without “Display info”. Note: This relies also on analog phone’s design, some phones will still display “unknown” with this feature enabled. Default is No.
<b>Replace Beginning ‘+’ with 00 in Caller ID</b>	When this feature is set to Yes, device will replace the “+” sign at the beginning of a number in the FROM header. Default is No.
<b>Number of Beginning Digits to Strip from Caller ID</b>	Removes a specified number of digits from the beginning of the Caller ID information received before displaying it to the called party. The range is between 0 and 10, default is 0
<b>Polarity Reversal</b>	Reverses the polarity upon call establishment and termination. Default is No.
<b>Loop Current Disconnect</b>	Allows the traditional PBX used with HT841/HT881 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.
<b>Play busy/reorder tone before Loop Current Disconnect</b>	Allow user to configure if it will play busy/reorder tone before loop current disconnect upon call fail. Default is No.
<b>Loop Current Disconnect Duration</b>	Configures the duration of voltage drop described in the topic above. HT841/HT881 support a duration range from 100 to 10000ms. The default value is 200.
<b>Enable Pulse Dialing</b>	Allow users to enable Pulse Dialing option under FXS Port. Default is No.
<b>Pulse Dialing Standard</b>	Allows users to use Swedish pulse dialing standard or New Zealand pulse dialing standard. Default is General Standard.

<b>Enable Hook Flash</b>	Enables the FLASH button to be used for terminating calls. Default is Yes.
<b>Hook Flash Timing</b>	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. HT841/HT881 support a range from 40 to 2000 ms. Default values are 300 minimum and 1100 maximum.
<b>On Hook Timing</b>	Specifies the on-hook time for an on-hook event to be validated. HT841/HT881 support a range from 40 to 2000 ms. Default value is 400.
<b>Gain</b>	Adjusts the voice path volume. <ul style="list-style-type: none"> <li>• Rx is a gain level for signals transmitted by FXS</li> <li>• Tx is a gain level for signals received by FXS.</li> </ul> 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.
<b>Disable Line Echo Canceller</b>	Disables the LEC per call base. Recommended for Fax/Data calls. Default is No.
<b>Disable Network Echo Suppressor</b>	Disables the NEC per call base. Recommended for Fax/Data calls. Default is No.
<b>Outgoing Call Duration Limit</b>	Defines the call duration limit for the outgoing calls. Default is 0 (No limit).
<b>Ring Frequency</b>	Customizes ring frequency. Valid options: 20Hz – 25Hz. Default is 20Hz.
<b>Enable High Ring Power</b>	Configures a high ringing voltage output for the ATA.
<b>OnHOOK DC Feed Current</b>	This feature is used to adjust DC feed current.
<b>RFC2833 Events Count</b>	This feature allows users to customize the count of RFC2833 events. Default is 8.
<b>RFC2833 End Events Count</b>	This feature allows users to customize the count of RFC2833 end events. Default is 3.
<b>PSTN Access Code</b>	Defines Key pattern to use PSTN line. Maximum 5 digits. Default is "*00"
<b>Distinctive Ring Tone</b>	Customizes the Ring Tone 1 to 3 with associate caller ID: when selected, if caller ID is configured, then the device will ONLY use this ring tone when the incoming call is from the Caller ID. System Ring Tone is used for all other calls. When selected but no Caller ID is configured, the selected ring tone will be used for all incoming calls using the FXS port. Distinctive ring tones can be configured not only for matching a whole number, but also for matching prefixes. In this case the symbol "x+" will be used. For example: if configured as 617x+, Ring Tone 1 will be used in case of call arrived from the area code 617. Any other incoming call will ring using cadence defined in parameter System Ring Cadence located under Advanced Settings Configuration page. Note: If server supports Alert-Info header and standard ring tone set (Bellcore) or distinctive ring tone 1-10 is specified, then the ring tone in the Alert-Info header from server will be used. Bellcore rings and tones are independent from custom ring tones. The custom ring tones can also be specified by alert-info header, for example <i>Alert-Info: &lt;127.0.0.1&gt;; info=ring5</i>

<b>Ring tones</b>	Configures the ring tone cadence preferences. User has 10 choices. The configuration, completed in Distinctive Ring Tones block in the same page, applies to ring tones cadences configured here.
<b>Distinctive Call Waiting Tone</b>	Customizes the Call Waiting Tone 1 to 10 with associate caller ID: when selected, if caller ID is configured, then the device will ONLY use this call waiting tone when the incoming call waiting is from the Caller ID. When selected but no Caller ID is configured, the selected call waiting tone will be used for all incoming waiting calls using the FXS port. Distinctive Call Waiting Tones can be configured not only for matching a whole number, but also for matching prefixes. In this case symbol “x+” will be used. For example: If configured as 617x+, Call Waiting Tone 1 will be used in case of waiting call arrived from the area code 617. Any other incoming call waiting will be using cadence defined in parameter Call Waiting Tone located under Advanced Settings Configuration page.
<b>Call Waiting Tones</b>	This feature allows user to customize call waiting tone. User has 10 choices. <i>Syntax: f1=val[,f2=val[,c=on1/off1[-on2/off2[-on3/off3]]]]];</i> (Frequencies are in (300, 3400) Hz and cadence on and off are in (0, 64000) ms) Note: The configuration, completed in Distinctive Call Waiting Tones block in the same page, applies to call waiting cadences configured here. Default is f1=440@-13,c=300/10000;
<i>Call Features Settings</i>	
<b>Enable Call Features</b>	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
<b>Reset Call Features</b>	Allows users to reset all call features configuration. Default is No
<b>SRTP Feature</b>	Allow users to customize the SRTP feature codes. Default is Yes – Enable SRTP: Default is 16 – Disable SRTP: Default is 17
<b>SRTP per call Feature</b>	– Enable SRTP per call: Default is 18 – Disable SRTP per call: Default is 19
<b>CID Feature</b>	Allow users to customize the CID feature codes. Default is Yes – Enable CID: Default is 31 – Disable CID: Default is 30
<b>CID per call Feature</b>	– Enable CID per call: Default is 82 – Disable CID per call: Default is 67
<b>Direct IP Calling Feature</b>	Allow users to customize the Direct IP feature code. Default is Yes – Direct IP Calling: Default is 47
<b>CW Feature</b>	Allow users to customize the CW feature codes. Default is Yes – Enable CW: Default is 51 – Disable CW: Default is 50
<b>CW per call Feature</b>	– Enable CW per call: Default is 71 – Disable CW per call: Default is 70
<b>Call Return Feature</b>	Allow users to customize the Call Return feature code. Default is Yes – Call return: Default is 69
<b>Unconditional Forward Feature</b>	Allow users to customize the Unconditional Forward feature codes.

	<p>Default is Yes</p> <ul style="list-style-type: none"> <li>– Enable Unconditional Forward: Default is 72</li> <li>– Disable Unconditional Forward: Default is 73</li> </ul>
<b>Busy Forward Feature</b>	<p>Allow users to customize the Busy Forward feature codes. Default is Yes</p> <ul style="list-style-type: none"> <li>– Enable Busy Forward: Default is 90</li> <li>– Disable Busy Forward: Default is 91</li> </ul>
<b>Delayed Forward Feature</b>	<p>Allow users to customize the Delayed Forward feature codes. Default is Yes</p> <ul style="list-style-type: none"> <li>– Enable Delayed Forward: Default is 92</li> <li>– Disable Delayed Forward: Default is 93</li> </ul>
<b>Paging Feature</b>	<p>Allow users to customize the Paging feature code. Default is Yes</p> <ul style="list-style-type: none"> <li>– Paging: Default is 74</li> </ul>
<b>DND Feature</b>	<p>Allow users to customize the CW feature codes. Default is Yes</p> <ul style="list-style-type: none"> <li>– Enable DND: Default is 78</li> <li>– Disable DND: Default is 79</li> </ul>
<b>Blind Transfer Feature</b>	<p>Allow users to customize the Blind Transfer feature code. Default is Yes</p> <ul style="list-style-type: none"> <li>– Enable Blind Transfer: Default is 87</li> </ul>
<b>Disable LEC per call Feature</b>	<p>Default is Yes</p> <ul style="list-style-type: none"> <li>– Disable LEC per call: Default is 03</li> </ul>
<b>Disable Bellcore Style 3-Way Conference</b>	<p>Default is No</p>
<b>Star Code 3WC Feature</b>	<p>Default is Yes</p> <ul style="list-style-type: none"> <li>– Star Code 3WC: Default is 23</li> </ul>
<b>Forced Codec Feature</b>	<p>Allow users to customize the Forced Codec feature code. Default is Yes</p> <ul style="list-style-type: none"> <li>– Forced Codec: Default is 02</li> </ul>
<b>PCMU Codec Feature</b>	<p>Default is Yes</p> <ul style="list-style-type: none"> <li>– PCMU Codec: Default is 7110</li> </ul>
<b>PCMA Codec Feature</b>	<p>Default is Yes</p> <ul style="list-style-type: none"> <li>– PCMA Codec: Default is 7111</li> </ul>
<b>G723 Codec Feature</b>	<p>Default is Yes</p> <ul style="list-style-type: none"> <li>– G723 Codec: Default is 723</li> </ul>
<b>G729 Codec Feature</b>	<p>Default is Yes</p> <ul style="list-style-type: none"> <li>– G729 Codec: Default is 729</li> </ul>
<b>iLBC Codec Feature</b>	<p>Default is Yes</p> <ul style="list-style-type: none"> <li>– iLBC Codec: Default is 7201</li> </ul>
<b>G722 Codec Feature</b>	<p>Default is Yes.</p> <ul style="list-style-type: none"> <li>– G722 Codec:: Default is 722</li> </ul>

## FXO Profile Settings Page Definitions

*FXO Profile (1, 2)*

<b>Profile Active</b>	Activates / Deactivates the accounts. The FXO port configuration will not change if disabled, although the port will not be operational, in this state, there will be no dial tone when picking up the analog phone and making/receiving calls will not be possible.
<b>Primary SIP Server</b>	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 HT841/HT881.
<b>Failover SIP Server</b>	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.
<b>Prefer Primary SIP Server</b>	Selects to prefer primary SIP server. The account will register to primary Server if registration with Failover server expires. Default is No.
<b>Outbound Proxy</b>	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 HT841/881 for firewall or NAT penetration in different network environments. If symmetric NAT is detected, STUN will not work and only outbound proxy can correct the problem
<b>Backup Outbound Proxy</b>	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.
<b>Prefer Primary Outbound Proxy</b>	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.
<b>From Domain</b>	allows users to add the actual domain name, it will override the from header.This is an optional configuration.
<b>SIP Transport</b>	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.
<b>SIP URI Scheme When Using TLS</b>	Specifies if "sip" or "sips" will be used when TLS/TCP is selected for SIP Transport. The default setting is "sips".
<b>Use Actual Ephemeral Port in Contact with TCP/TLS</b>	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.
<b>NAT Traversal</b>	<p>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, VPN, and Auto.</p> <ul style="list-style-type: none"> <li>● <b>No:</b> Disables NAT traversal, potentially causing connection issues for the ATA if it's behind a NAT router.</li> <li>● <b>Keep-alive:</b> Periodically sends signals to maintain the NAT mapping for the FXO Gateway, preventing disconnection.</li> <li>● <b>STUN:</b> Utilizes the STUN protocol to discover and adapt to NAT settings for seamless VoIP communication.</li> <li>● <b>UPnP:</b> Allows the FXO Gateway to dynamically configure NAT port mappings, streamlining VoIP traffic.</li> <li>● <b>VPN:</b> Requires the FXO Gateway to establish a secure VPN connection for VoIP, bypassing NAT restrictions.</li> <li>● <b>Auto:</b> Automatically selects the most suitable NAT traversal method based on network conditions, ensuring reliable FXO Gateway connectivity.</li> </ul>

	Default setting is No.
<b>DNS Mode</b>	<p>Selects DNS mode to use for the client to look up server. One mode can be chosen.</p> <p><b>A Record (Default):</b> resolves IP Address of target according to domain name.</p> <p><b>SRV:</b> DNS SRV resource records indicate how to find services for various protocols.</p> <p><b>NAPTR/SRV:</b> Naming Authority Pointer according to RFC 2915.</p> <p><b>Use Configured IP:</b> 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.</p>
<b>DNS SRV Failover Mode</b>	<p>Configure the preferred IP mode when DNS Mode is SRV or NAPTR/SRV.</p> <ul style="list-style-type: none"> <li>• 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.</li> <li>• 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</li> <li>• Saved on until no response SIP request will always be sent to the registered IP address only until registered IP address is unreachable.</li> </ul>
<b>Failback Timer</b>	<p>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).</p>
<b>Register before DNS SRV Failover</b>	<p>This feature is used to control whether the device need 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.</p>
<b>Primary IP</b>	The main IP address used for communication and control.
<b>Backup IP1</b>	The secondary IP address used as a backup for communication in case the Primary IP fails.
<b>Backup IP2</b>	the third IP address used as an additional backup for communication in case both the Primary IP and Backup IP1 fail.
<b>TEL URI</b>	<p>Indicates E.164 number in the “From” header by adding the “User=Phone” parameter or using “Tel:” in SIP packets, if the HT841/HT881 has an assigned PSTN Number.</p> <p><b>Disabled:</b> Use “SIP User ID” information in the Request-Line and “From” header.</p> <p><b>User=Phone:</b> “User=Phone” parameter will be attached to the Request-Line and “From” header in the SIP request to indicate the E.164 number. If set to “Enable”.</p> <p><b>Enabled:</b> “Tel:” will be used instead of “sip:” in the SIP request.</p> <p>Please consult your carrier before changing this parameter. The default is Disabled.</p>
<b>SIP Registration</b>	<p>Controls whether the HT841/HT881 needs to send REGISTER messages to the proxy server. The default setting is Yes.</p>
<b>Unregister on Reboot</b>	<p>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 FXO page. Unregister on reboot option can be set to "No", "All" or "Instance".</p> <ol style="list-style-type: none"> <li>1. <b>Set to "No":</b> 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.</li> <li>2. <b>Set to "All":</b> 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 FXO ports on the device.</li> <li>3. <b>Set to "Instance":</b> If the "Unregister on reboot" option is set to "Instance", it means that only the SIP account associated with the concerned FXO port will be unregistered when</li> </ol>

	<p>the device reboots. This option clears the SIP user's information for only the specific FXO port that is affected by the reboot.</p> <p>Default value is set to "No"</p>
<b>Outgoing Call Without Registration</b>	<p>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.</p>
<b>Register Expiration</b>	<p>Refreshes registration periodically with specified SIP proxy (in minutes). Maximum interval is 65535 minutes (about 45 days).</p> <p>Default is 60 minutes (or 1 hour).</p>
<b>Reregister Before Expiration</b>	<p>Sends re-register request after specific time (in seconds) to renew registration before the previous registration expires.</p>
<b>SIP Registration Failure Retry Wait Time</b>	<p>Sends re-register request after specific time (in seconds) when registration process fails. Maximum interval is 3600 seconds (1 hour).</p> <p>Default is 20 seconds.</p>
<b>SIP Registration Failure Retry Wait Time upon 403 Forbidden</b>	<p>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.</p>
<b>Port Voltage Off upon no SIP Registration or SIP Registration Failure</b>	<p>Cuts off voltage to the port if there is no SIP registration or if SIP registration fails, preventing unwanted calls.</p> <p>(in minutes. Between 0-60, default is 0. 0 means port voltage is never turned off)</p>
<b>Enable SIP OPTIONS/NOTIFY Keep Alive</b>	<p>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.</p> <p>Default setting is No.</p>
<b>SIP OPTIONS/NOTIFY Keep Alive Interval</b>	<p>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.</p>
<b>SIP OPTIONS/NOTIFY Keep Alive Max Lost</b>	<p>Defines the Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3.</p>
<b>Layer 3 QoS</b>	<p>Defines Diff-Serv values for SIP and RTP.</p> <p>SIP DSCP (Diff-Serv value in decimal, 0-63, default 26)</p> <p>RTP DSCP (Diff-Serv value in decimal, 0-63, default 46)</p>
<b>Local SIP Port</b>	<p>Defines local port to use by the HT841/HT881 for listening and transmitting SIP packets.</p> <p>Default value for FXO 1 is 5060 and 5062 for FXO 2.</p>
<b>Local RTP Port</b>	<p>Defines the local RTP-RTCP port pair the HT841/881 will listen and transmit. It is the HT841/881 RTP port for channel 0.</p> <p>The default value for FXO port is 5004</p>
<b>Use Random SIP Port</b>	<p>Controls whether to use configured or random SIP ports. This is usually necessary when multiple HT841x/HT881 are behind the same NAT.</p> <p>The default is No.</p>
<b>Use Random RTP Port</b>	<p>Controls whether to use configured or random RTP ports. This is usually necessary when multiple HT841/HT881 are behind the same NAT.</p> <p>The default is No.</p>
<b>Enable RTCP</b>	<p>Allows users to enable RTCP to monitor RTP delivery. The default setting is "Yes".</p>

<b>Remove OBP from Route Header</b>	Removes outbound proxy info in “Route” header when sending SIP packets. Default is No.
<b>Support SIP Instance ID</b>	Includes “SIP Instance ID” attribute to “Contact” header in REGISTER request as defined in IETF SIP outbound draft. Default is No.
<b>Validate Incoming SIP Message</b>	Checks the authenticity and integrity of incoming SIP messages for enhanced security.
<b>Check SIP User ID for Incoming INVITE</b>	Checks the SIP User ID in the Request URI of the incoming INVITE; if it doesn’t match the HT841/HT881 SIP User ID, the call will be rejected. Direct IP calling will also be disabled. The default is No.
<b>Authenticate Incoming INVITE</b>	Challenges the incoming INVITE for authentication with SIP 401 Unauthorized message. Default is No.
<b>Authenticate Server Certificate Domain</b>	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”.
<b>Authenticate server certificate chain</b>	Configures whether to validate the server certificate when download the firmware/config file. If it is set to “Yes”, the phone will download the firmware/config file only from the legitimate server. The default setting is “No”.
<b>Allow Incoming SIP Messages from SIP Proxy Only</b>	Restricts the acceptance of SIP messages to only those originating from the designated SIP proxy, enhancing network security. If Set to Yes , there will no direct IP Calling Default value is No
<b>Use Privacy Header</b>	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.
<b>Use P-Preferred-Identity Header</b>	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.
<b>Use P-Access-Network-Info Header</b>	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.
<b>Use P-Emergency-Info Header</b>	This feature support of IEEE-48-addr and IEEE-EUI-64 in SIP header for emergency calls.
<b>Use P-Asserted-Identity Header</b>	When this feature is set to Yes, device will send P-Asserted-Identity Header on the SIP Invite. Default setting is No.
<b>Allow SIP Factory Reset</b>	Allows to reset the devices directly through SIP Notify. If “Allow SIP Factory Reset” is set to “YES” under FXO PORT, then the ATA receives the NOTIFY from the SIP server with Event: reset, the HT should perform a factory reset after the authentication. The authentication in this case can be either with: The admin password if no SIP account is configured on the HT. With the credentials of the SIP account if configured on the ATA.
<b>Maximum Number of SIP Request Retries</b>	This feature allows user to configure the number of SIP retries before failover occurs. (between 1 and 10, default is 2).
<b>SIP T1 Timeout</b>	Defines T1 timeout value. It is an estimate of the round-trip time between the client and server transactions. For example, HT841/HT881 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.
<b>SIP T2 Interval</b>	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.
<b>SIP Timer D</b>	Configures SIP Timer D defined in RFC3261. 0 – 64 seconds. Default is 0.
<b>DTMF Payload Type</b>	Defines payload type for DTMF using RFC2833.
<b>Preferred DTMF method (in order)</b>	Sorts DTMF methods (in-audio, via RTP (RFC2833) or via SIP INFO) by priority.
<b>Inband DTMF Duration</b>	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.
<b>Inband DTMF Tx Gain</b>	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
<b>DSP DTMF Detector Duration Threshold</b>	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.
<b>DSP DTMF Detector Min Level</b>	Defines the minimum level required for DTMF tone to be considered valid. Default value is -25dBm from the valid range -45dBm to 0dBm
<b>DSP DTMF Detector Snr</b>	Defines the maximum signal-to-noise ratio required for DTMF tone to be considered valid. Default value is set to 1, from the range 0-2
<b>DSP DTMF Detector Deviation</b>	Defines the deviation above and below the central frequency. Default value is 25% from the valid range 0-25
<b>DSP DTMF Detector Twist</b>	Defines the maximum value by which one frequency energy level can surpass the another frequency energy level. Default value is 5dB from the valid range 0-12
<b>Disable DTMF Negotiation</b>	Uses DTMF order without negotiation. Default is No.
<b>Flash Digit Control</b>	When it set to YES it allows the user to perform some call setting when both channels are used while pressing: “Flash + 1” in order to hang up the current call and resume a call that was held. “Flash + 2” in order to hold the current call and resume a call that was held. “Flash + 3” in order to perform 3-way conference. “Flash + 4” in order to perform attended transfer. Note: Please refer to the user guide for detailed steps to perform above operations. More additional Digit events were added on the new firmware 1.0.43.10.
<b>Proxy-Require</b>	Determines a SIP Extension to notify the SIP server that the HT841/HT881 is behind a NAT/Firewall.
<b>Use NAT IP</b>	Defines NAT IP address used in SIP/SDP messages. It should only be used if required by ITSP.
<b>SIP User-Agent</b>	This feature allows users to configure SIP User Agent. If not configured, device will use the default User Agent header.

<b>SIP User-Agent Postfix</b>	Configures the SIP User-Agent Postfix
<b>Add MAC in User-Agent</b>	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 REGISTER SIP 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.
<b>Use MAC Header</b>	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.
<b>Do Not Escape '#' as %23 in SIP URI</b>	Replaces # by %23 in some special situations. Default is No.
<b>Disable Multiple m Line in SDP</b>	Sends only one m line in SDP, regardless of how many m fields are in the incoming SDP. Default is No.
<b>Early Dial</b>	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.
<b>Dial Plan Prefix</b>	Adds specified prefix to dialed number.
<b>Use # as Dial Key</b>	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.
<b>Disable # as Redial Key</b>	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 No.
<b>Dial Plan</b>	Dial Plan Rules: 1. Accept Digits: <b>1,2,3,4,5,6,7,8,9,0 , *, #, A,a,B,b,C,c,D,d</b> 2. Grammar: <b>x</b> – any digit from 0-9; a. <b>xx+</b> – at least 2 digits number; b. <b>xx</b> – exactly 2 digits number; c. <b>^</b> – exclude; d. <b>.</b> – wildcard, matches one or more characters e. <b>[3-5]</b> – any digit of 3, 4, or 5; f. <b>[147]</b> – any digit 1, 4, or 7; g. <b>&lt;2=011&gt;</b> – replace digit 2 with 011 when dialing h. <b>&lt;=1&gt;</b> – add a leading 1 to all numbers dialed, vice versa will remove a 1 from the number dialed i. <b> </b> – or j. Flag <b>T</b> 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 dial 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+} –

	<p>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.</p> <p>1. Default: Outgoing – { x+   +x+   *x+   *xx*x+ }</p> <p>Example of a simple dial plan used in a Home/Office in the US:  { ^1900x.   &lt;=1617&gt;[2-9]xxxxxx   1[2-9]xx[2-9]xxxxxx   011[2-9]x.   [3469]11 }</p> <p>Explanation of example rule (reading from left to right):  ^1900x. – prevents dialing any number started with 1900  &lt;=1617&gt;[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</p> <p>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.</p>
<b>Anonymous Call Rejection</b>	Rejects incoming calls with anonymous caller ID with “486 Busy here” message. Default is No.
<b>Special Feature</b>	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.
<b>Enable Session Timer</b>	Disable the session timer when this option is set to “No”. By default, this option is enabled.
<b>Session Expiration</b>	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.
<b>Min-SE</b>	Defines Minimum session expiration (in seconds). Default is 90 seconds.
<b>Caller Request Timer</b>	Uses session timer when making outbound calls if remote party supports it. Valid range is 90-64800 seconds. Default is No.
<b>Callee Request Timer</b>	Uses session timer when receiving inbound calls with session timer request. Default is No.
<b>Force Timer</b>	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.
<b>UAC Specify Refresher</b>	Specifies which end will act as refresher for outgoing calls. Default is Omit. <b>UAC:</b> The HandyTone acts as the refresher. <b>UAS:</b> Callee or proxy server act as the refresher.
<b>UAS Specify Refresher</b>	Specifies which end will act as refresher for incoming calls. Default is Omit.: <b>UAS:</b> The HandyTone acts as the refresher. <b>UAC:</b> Callee or proxy server act as the refresher.
<b>Force INVITE</b>	Uses INVITE message to refresh the session timer. Default is No.
<b>When To Restart Session After Re-INVITE received</b>	Allows users to delay posting Media Change Event, it can be set to “Immediately” or to “After replying 200OK” The default value is “Immediately”.

<b>Enable 100rel</b>	Appends "100rel" attribute to the value of the required header of the initial signaling messages. Default is No.
<b>Add Auth Header on Initial REGISTER</b>	Adds "Authentication" header with blank "nonce" attribute in the initial SIP REGISTER request. Default is No.
<b>Use First Matching Vocoder in 200OK SDP</b>	Includes only the first matching vocoder in its 200OK response, otherwise it will include all matching vocoders in same order received in INVITE. Default is No.
<b>Preferred Vocoder</b>	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
<b>Voice Frames per TX</b>	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.
<b>G723 Rate</b>	Operates at specified encoding rate for G.723 vocoder. Available encoding rates are 6.3kbps or 5.3kbps. Default is 6.3kbps.
<b>iLBC Frame Size</b>	Specifies iLBC packet frame size (20ms or 30ms). Default is 20ms.
<b>Disable OPUS Stereo in SDP</b>	Disables OPUS stereo in SDP. Default is No.
<b>iLBC Payload Type</b>	Determines payload type for iLBC. Valid range is between 96 and 127. Default is 97.
<b>OPUS Payload Type</b>	Determines payload type for OPUS. Valid range is between 96 and 127. Default is 123.
<b>VAD</b>	Allows detecting the absence of audio and conserves bandwidth by preventing the transmission of "silent packets" over the network. Default is No.
<b>Symmetric RTP</b>	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.
<b>Fax Mode</b>	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.
<b>Re-Invite after Fax Tone Detected</b>	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
<b>Jitter Buffer Type</b>	Selects jitter buffer type (Fixed or Adaptive) based on network conditions.
<b>Jitter Buffer Length</b>	<b>High</b> (initial 200ms, min 40ms, max 600ms) Note: not all vocoders can meet the high requirement. <b>Medium</b> (initial 100ms, min 20ms, max 200ms). <b>Low</b> (initial 50ms, min 10ms, max 100ms).
<b>SRTP Mode</b>	Selects SRTP mode to use ("Disabled", "Enabled but not forced", or "Enabled and forced"). Default is Disabled It uses SDP Security Description to exchange key. Please refer to <b>SDES</b> : <a href="https://tools.ietf.org/html/rfc4568">https://tools.ietf.org/html/rfc4568</a> <b>SRTP</b> : <a href="https://www.ietf.org/rfc/rfc3711.txt">https://www.ietf.org/rfc/rfc3711.txt</a>

<b>SRTP Key Length</b>	<p>Allows users to select supported SRTP Key Length. the available values are :</p> <ol style="list-style-type: none"> <li>1. AES 128&amp;256 bit</li> <li>2. AES 128 bit</li> <li>3. AES 256 bit</li> </ol> <p>Set to AES 128&amp;256 bit By Default.</p>
<b>Crypto Life Time</b>	Adds crypto life time header to SRTP packets. Default is Yes.
<b>Caller ID Scheme</b>	Selects the caller id scheme, for example: Bellcore/Telcordia, ETSI-FSK ...
<b>FSK Caller ID Minimum RX Level (dB)</b>	<p>Sets the minimum signal level required to detect and receive FSK Caller ID information accurately.</p> <p>The valid range is -96 - 0dB. Default -40dB</p>
<b>FSK Caller ID Seizure Bits</b>	<p>Represents the number of bits used to detect the start of a Caller ID transmission.</p> <p>The valid range is 0 - 800 bits. Default 70</p>
<b>FSK Caller ID Mark Bits</b>	<p>Represents the number of bits used to identify the caller's mark frequency during Caller ID transmission.</p> <p>The Valid range is 1 - 800 bits. Default 40</p>
<b>Caller ID Transport Type</b>	<p>Refers to the method used to transmit Caller ID information, which can be:</p> <ul style="list-style-type: none"> <li>• <b>Relay via SIP Form:</b> Caller ID information is transmitted using the "From" field in the SIP header.</li> <li>• <b>Relay via P-SIP Asserted identity:</b> Caller ID information is transmitted using the "P-Asserted-Identity" header in the SIP message.</li> <li>• <b>Relay via P-SIP Preferred identity:</b> Caller ID information is transmitted using the "P-Preferred-Identity" header in the SIP message.</li> <li>• <b>Send Anonymous:</b> Caller ID information is transmitted using the "P-Asserted-Identity" header in the SIP message.</li> <li>• <b>Disable:</b> Caller ID information is transmitted using the "P-Asserted-Identity" header in the SIP message.</li> </ul> <p>Set to <b>Relay via SIP Form</b> by Default.</p>
<b>Send Hook Flash To PSTN</b>	Refers to the capability of sending a signal to the Public Switched Telephone Network (PSTN) to simulate a hook flash during a call, typically used to access certain telephony features or services.
<b>Hook Flash Duration (ms)</b>	Refers to the time duration, measured in milliseconds, for which the hook flash signal is transmitted to the Public Switched Telephone Network (PSTN) during a call, typically used to trigger specific telephony features or services.
<b>Disable Unknown Caller ID</b>	<p>Disable analog phone's caller ID when receiving a call with "Anonymous", "unavailable" or "unknown" in FROM header and without "Display info".</p> <p>Note: This relies also on analog phone's design, some phones will still display "unknown" with this feature enabled. Default is No.</p>
<b>Replace Beginning '+' with 00 in Caller ID</b>	When this feature is set to Yes, device will replace the "+" sign at the beginning of a number in the FROM header. Default is No.
<b>Number of Beginning Digits to Strip from Caller ID</b>	Removes a specified number of digits from the beginning of the Caller ID information received before displaying it to the called party.
<b>Gain</b>	<p>Adjusts the voice path volume.</p> <ul style="list-style-type: none"> <li>• Rx is a gain level for signals transmitted by FXO</li> <li>• Tx is a gain level for signals received by FXO.</li> </ul> <p>Default = 0dB for both parameters. Loudest volume: +6dB Lowest volume: -6dB.</p>

	<p>User can adjust volume of call using the Rx gain level parameter and the Tx gain level parameter located on the FXO port configuration page.</p> <p>If call volume is too low when using the FXO port (ie. the ATA is at user site), adjust volume using the Rx gain level parameter under the FXO port configuration page.</p> <p>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 FXO port configuration PAGE.</p>
<b>Disable Line Echo Canceller</b>	<p>Disables the LEC per call base. Recommended for Fax/Data calls.</p> <p>Default is No.</p>
<b>Outgoing Call Duration Limit</b>	<p>Defines the call duration limit for the outgoing calls. Default is 0 (No limit).</p>
<b>Incoming Call Duration Limit</b>	<p>Defines the call duration limit for the incoming calls. Default is 0 (No limit).</p>
<b>RFC2833 Events Count</b>	<p>This feature allows users to customize the count of RFC2833 events. Default is 8.</p>
<b>RFC2833 End Events Count</b>	<p>This feature allows users to customize the count of RFC2833 end events. Default is 3.</p>
<b>FXO Termination</b>	
<b>Enable Current Disconnect</b>	<p>This value should be used in case the PSTN provider uses line power drop to indicate call completion to the end point. In this case the HT841/HT881 will search for a power drop.</p>
<b>Current Disconnect Threshold (ms)</b>	<p>This is a preconfigured value of duration for a line power drop used by specific service providers. For example, for a configured value of 500ms the device will ignore any random voltage drops on the line if duration of such drop is less than 500ms and the call will NOT be considered as terminated. This is useful to prevent unnecessary call drops in some low quality PSTN lines. Default is 100 ms. Range from 50 to 800 ms.</p>
<b>Enable PSTN Disconnect Tone Detection</b>	<p>If set to Yes, arrived Busy Tone is used as the disconnect signal.</p> <p>Default is "No"</p>
<b>Enable Polarity Reversal</b>	<p>This should be set to Yes only if the FXO lines are subscribed to PR service from PSTN Service provider. It is merely a PR detect feature. Default is No.</p> <p><b>Note:</b> If there is no PR service from provider on the FXO line, and this setting is configured to Yes, calls will not be successful.</p>
<b>Polarity On Answer Delay</b>	<p>Allows users to configure the Polarity On Answer Delay.</p> <p>Valid range is 200-30000ms, Default value is 1000ms</p>
<b>AC Termination Model</b>	<p>You can select the AC termination by Country or by Impedance.</p> <p>Default is Country-based.</p>
<b>Country-Based</b>	<p>15 Countries are selectable in this version of the F/W.</p> <p>USA AUSTRIA AUSTRALIA/NEW ZEALAND BELGIUM CHINA FINLAND FRANCE GERMANY GREECE ITALY JAPAN NORWAY SPAIN SWEDEN UK</p> <p>Default is USA</p>

<b>Impedance-Based</b>	<p>Select the Impedance used by the PSTN service provider.</p> <p><b>600R – 600 ohms</b>  <b>600C – 600 ohms + 2.16uF</b>  <b>900R – 900 ohms</b>  <b>900C – 900 ohms + 2.16uF</b>  <b>COMPLEX1 – 220 ohms + (820 ohms    115nF)</b>  <b>COMPLEX2 – 270 ohms + (750 ohms    150nF)</b>  <b>COMPLEX3 – 370 ohms + (620 ohms    310nF)</b>  <b>COMPLEX4 – 600R, 270 ohms + (750 ohms    150nF)</b>  <b>COMPLEX5 – 320 ohms + (1050 ohms    230nF)</b>  <b>COMPLEX6 – 350 ohms + (1000 ohms    210nF)</b>  <b>COMPLEX7 – 200 ohms + (680 ohms    100nF)</b>  <b>COMPLEX8 – 370 ohms + (820 ohms    110nF)</b>  <b>COMPLEX9 – 275 ohms + (780 ohms    115nF)</b>  <b>COMPLEX10 – 120 ohms + (820 ohms    110nF)</b>  Default is 600R – 600 ohms</p>
<b>Number of Rings</b>	The FXO port will ring the number of times configured in this field before sending the call to the VoIP side. The supported range is 1-50. Default is 4.
<b>PSTN Ring Thru FXO</b>	If Yes, the phone connected to the FXO port will ring a configured number of times (see above). If not, the phone connected to the FXO port will not ring.
<b>PSTN Ring Thru Delay (sec)</b>	If the PSTN Ring Thru Delay is set to Yes, all incoming PSTN calls through FXO will ring the phone connected to the FXO port, after this delay or after caller id is detected (whichever comes first).
<b>PSTN Ring Timeout (sec)</b>	Range is 2-10 seconds. Default is 6 seconds. Option is used to detect PSTN hang up when FXO port is not answered.
<b>PSTN Idle Wait Timeout between Outgoing Calls</b>	Used to customize timeout value between PSTN outgoing calls. Range is 0-10 seconds. Default is 4 seconds.
<b>PIN for VoIP-to-PSTN Calls</b>	Requires users to enter a Personal Identification Number (PIN) before making calls from VoIP to the traditional Public Switched Telephone Network (PSTN) to enhance security and prevent unauthorized usage.
<b>PIN for PSTN-to-VoIP Calls</b>	requires users to enter a Personal Identification Number (PIN) before initiating calls from the traditional Public Switched Telephone Network (PSTN) to the VoIP network, ensuring secure access and preventing unauthorized usage.
<b>Channel Dialing</b>	
<b>DTMF Digit Length (ms)</b>	<p>Digit length and Dial Pause are port digit dialing configurations; FXO needs to dial out digits for VoIP to PSTN 1 stage calls, and unconditional call forward to PSTN, and route to PSTN. Digit Length is the play time for each digit.</p> <p>Note: In order to receive the caller ID information, the delay should be set to a value larger than the delay required to complete the PSTN caller ID delivery.</p>
<b>DTMF Dial Pause (ms)</b>	Dial pause is the time between 2 digits for the same scenario as explained above.
<b>First Digit Timeout (sec)</b>	Used for PSTN to VoIP calls. PSTN users need to enter the FIRST digit within the first digit timeout period. Otherwise the call will be dropped.
<b>Inter-Digit Timeout (sec)</b>	When dialing from the PSTN to VoIP, subsequent digits must be input within the period of inter-digit timeout. Otherwise the dial plan thinks it is the end of the digit input.
<b>Wait for Dial Tone</b>	Wait for Dial tone is used for one stage VoIP to PSTN calls. If set to Yes, the device will first obtain a PSTN line and a dial tone from a central office. After obtaining the dial tone, the digits

	dialed will be sent to the central office.
<b>Stage Method (1/2)</b>	This configuration is applicable for VoIP to PSTN calls and indicates one or two stage dialing methods.
<b>Min Delay Before Dial PSTN Number</b>	The time to wait before HT841/HT881 initiates the call via PSTN line. Default 500ms, range is from 50 to 65000ms.

## Ports Settings Page Definitions

<b>FXS Port Settings</b>	
<b>Port</b>	Display the port number
<b>SIP User ID</b>	Defines user account information provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
<b>Authenticate ID</b>	Determines account authenticate ID provided by VoIP service provider (ITSP). Can be identical to or different from "SIP user ID".
<b>Password</b>	Specifies account password provided by VoIP service provider (ITSP) to register to SIP servers.
<b>Name</b>	Chooses a name to be associated to user.
<b>Profile ID</b>	Defines the profile ID for each port.
<b>Enable Port</b>	Enables / Disables the port
<b>Off hook Auto-Dial</b>	Configures a User ID or extension number that is automatically dialed when off-hook. Only the user part of a SIP address needs is entered here. The HT841/HT881 will automatically append the "@" and the host portion of the corresponding SIP address.
<b>FXO Ports Settings</b>	
<b>Port</b>	Display the port number
<b>SIP User ID</b>	Defines user account information provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
<b>Authenticate ID</b>	Determines account authenticate ID provided by VoIP service provider (ITSP). Can be identical to or different from "SIP user ID".
<b>Password</b>	Specifies account password provided by VoIP service provider (ITSP) to register to SIP servers.
<b>Name</b>	Chooses a name to be associated to user.
<b>Profile ID</b>	Defines the profile ID for each port.
<b>Enable Port</b>	Enables / Disables the port
<b>Unconditional Call Forward to PSTN</b>	Enables all incoming calls from the VoIP network to be automatically forwarded to the traditional Public Switched Telephone Network (PSTN) without any conditions or restrictions. You can Configure Unconditional Call Forward to PSTN on eight FXO ports

<b>Unconditional Call Forward to VOIP</b>	allows all incoming calls from the traditional Public Switched Telephone Network (PSTN) to be automatically forwarded to a VoIP network without any conditions or restrictions. You can Configure Unconditional Call Forward to PSTN on eight FXO ports
<b>Port</b>	Displays the port number.
<b>Sip Server</b>	Displays the SIP Server.
<b>Sip Destination Port</b>	Displays the destination port.

## IMPORTANT WEB CONFIGURATIONS

The HT841/HT881 embedded Web server responds to HTTP/HTTPS GET/POST requests. Embedded HTML pages allow a user to configure the HT841/HT881 through a web browser such as Google Chrome, Mozilla Firefox and Microsoft's IE.

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

### Web UI Access Level Management

There are three default passwords for the login page:

User Level	Password	Web Pages Allowed
<b>Administrator Level</b>	admin password is found on a sticker on the back of the unit	All pages.
<b>End User Level</b>	123	Only Status and Basic Settings
<b>Viewer Level</b>	viewer	Only checking. Not allowed to modify content.

*HT841/HT881 Access Level Types*

#### Note

The password is case sensitive with maximum length of 30 characters. When changing any settings, always submit them by pressing **Update** or **Apply** button on the bottom of the page. After submitting the changes in all the Web GUI pages, if a reboot is required, the web page will prompt the user to reboot by offering a reboot button on the web page.

### Saving the Configuration Changes

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

### Changing Admin Level Password

1. Access your HT841/HT881 web UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: The password found on the sticker on the back of the unit).

3. Press **Login** to access your settings.
4. Go to **Advanced Settings** → **New Admin Password** and enter the new admin password. ( Must contain 8-30 characters, at least one number, one uppercase and lowercase letter. Purposely not displayed for security protection.)
5. Confirm the new admin password.
6. Press **Apply** at the bottom of the page to save your new settings.

Grandstream Device Configuration						
STATUS	BASIC SETTINGS	ADVANCED SETTINGS	FXS PROFILE	FXO PROFILE 1	FXO PROFILE 2	PORTS
<b>New Admin Password:</b>	<input type="text"/>	(Must contain 8-30 characters, at least one number, one uppercase and lowercase letter. Purposely not displayed for security protection.)				
<b>Confirm Admin Password:</b>	<input type="text"/>					

*HT841/HT881 Admin Password change*

## Changing User Level Password

1. Access your HT841/HT881 web UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: The password found on the sticker on the back of the unit).
3. Press **Login** to access your settings.
4. Go to **Basic Settings** → **New End User Password** and enter the new end-user password. (Must contain 8-30 characters, at least one number, one uppercase and lowercase letter. Purposely not displayed for security protection.)
5. Confirm the new end-user password.
6. Press **Apply** at the bottom of the page to save your new settings.

Grandstream Device Configuration						
STATUS	BASIC SETTINGS	ADVANCED SETTINGS	FXS PROFILE	FXO PROFILE 1	FXO PROFILE 2	PORTS
<b>New End User Password:</b>	<input type="text"/>	(Must contain 8-30 characters, at least one number, one uppercase and lowercase letter. Purposely not displayed for security protection.)				
<b>Confirm End User Password:</b>	<input type="text"/>					

*HT841/HT881 User Password change*

## Changing Viewer Password

1. Access your HT841/HT881 web UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: The password found on the sticker on the back of the unit).
3. Press **Login** to access your settings.
4. Go to **Basic Settings** → **New Viewer Password** and enter the new viewer password. (Must contain 8-30 characters, at least one number, one uppercase and lowercase letter. Purposely not displayed for security protection.)
5. Confirm the new viewer password.
6. Press **Apply** at the bottom of the page to save your new settings.

<b>New Viewer Password:</b>	<input type="text"/>	(Must contain 8-30 characters, at least one number, one uppercase and lowercase letter. Purposely not displayed for security protection.)
<b>Confirm Viewer Password:</b>	<input type="text"/>	

*HT841/HT881 Viewer Password change*

## Changing HTTP/HTTPS Web Port

1. Access your HT841/HT881 web UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: The password found on the sticker on the back of the unit).

3. Press **Login** to access your settings.
4. Go to **Basic Settings** → **HTTP(S) Web Port**.
5. Make sure that the **Web Access Mode** is set to **HTTPS**.
6. Change the current port to your new HTTP(S) port. Ports accepted are in range [1-65535].
7. Press **Apply** at the bottom of the page to save your new settings

**Web/SSH Access:**

Web Session Timeout:  (1-60, default 10 minutes.)

Web Access Attempt Limit:  (1-10, default 5.)

Web Lockout Duration:  (0-60, default 15 minutes.)

Web Access Mode:  **HTTPS**  HTTP  Disabled

HTTP Web Port:  (default is 80)

HTTPS Web Port:  (default is 443)

Disable SSH:  No  Yes

SSH Port:  (default is 22. Cannot be the same as Telnet Port.)

Disable Telnet:  No  Yes

Telnet Port:  (default is 23. Cannot be the same as SSH Port.)

Security Controls for SSH/Telnet Access:

WAN Side Web/SSH Access:  No  Yes  Auto (WAN side access allowed for private IP; rejected for public IP)

HT841/HT881 HTTPS Web Port

## Configuring HT841/HT881 Through Voice Prompts

As mentioned previously, The HT841/HT881 has a built-in voice prompt menu for simple device configuration. Please refer to "Understanding HT841/HT881 Interactive Voice Prompt Response Menu" for more information about IVR and how to access its menu.

### o DHCP MODE

Select voice menu option 01 to allow the HT841/HT881 to use DHCP.

### o STATIC IP MODE

Select voice menu option 01 to allow the HT841/HT881 to enable the STATIC IP mode, then use option 02, 03, 04, 05 to set up IP address, Subnet Mask, Gateway and DNS server respectively.

### o PPPOE MODE

Select voice menu option 01 to allow the HT841/HT881 to enable the PPPoE mode. PPPoE Username and Password should be configured from web GUI.

### o FIRMWARE SERVER IP ADDRESS

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

### o CONFIGURATION SERVER IP ADDRESS

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

### o UPGRADE PROTOCOL

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

### o FIRMWARE UPGRADE MODE

Select voice menu option 17 to choose firmware upgrade mode among the following three options:

"Always check, check when pre/suffix changes, and never upgrade".

- **WAN PORT WEB ACCESS**

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

## Register a SIP Account

The HT841 supports 1 SIP account for FXS port, and 4 accounts for FXO ports, While the HT881 supports 1 SIP account for FXS port, and 8 accounts for FXO ports.

Please refer to the following steps in order to register your accounts via web user interface.

- Access your HT841/HT881 web UI by entering its IP address in your favorite browser.
- Enter your admin password (default: found on the sticker on the back of the unit).
- Press **Login** to access your settings.
- Go to **FXS Profile** (same steps for **FXO Profile 1&2**) web pages and set the following:
  1. **Account Active** to **Yes**.
  2. **Primary SIP Server** field with your SIP server IP address or FQDN.
  3. **Failover SIP Server** with your Failover SIP Server IP address or FQDN. Leave empty if not available.
  4. **Prefer Primary SIP Server** to **No** or **Yes** depending on your configuration. Set to **No** if no Failover SIP Server is defined. If "**Yes**", account will register to Primary SIP Server when failover registration expires.
  5. **Outbound Proxy**: Set your Outbound Proxy IP Address or FQDN. Leave empty if not available.

The screenshot shows the 'Grandstream Device Configuration' web interface. The 'FXS PROFILE' tab is selected. The configuration options are as follows:

- Profile Active:**  No  Yes
- Primary SIP Server:**  (e.g., sip.mycompany.com, or IP address)
- Failover SIP Server:**  (Optional, used when primary server no response)
- Prefer Primary SIP Server:**  No  
 Will register to Primary Server if Failover registration expires  
 Will register to Primary Server if Primary Server responds, need to enable SIP OPTIONS/NOTIFY Keep Alive
- Outbound Proxy:**  (e.g., proxy.myprovider.com, or IP address, if any)
- Backup Outbound Proxy:**  (e.g., proxy.myprovider.com, or IP address, if any)
- Prefer Primary Outbound Proxy:**  No  Yes (yes - will reregister via Primary Outbound Proxy if registration expires)
- From Domain:**  (Optional, actual domain name, will override the from header)
- Allow DHCP Option 120 (override SIP server):**  No  Yes
- SIP Transport:**  UDP  TCP  TLS (default is UDP)

*Configure SIP Server*

- Go to **Ports** web page and set the following
  1. **SIP User ID:** Under **Ports** Web Page , Enter the SIP User ID User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
  2. **Authenticate ID:** SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
  3. **Authenticate Password:** SIP service subscriber's account password to register to SIP server of ITSP. For security reasons, the password will field will be shown as empty.
  4. **Name:** Any name to identify this specific user.
- Press **Apply** at the bottom of the page to save your configuration.

### Grandstream Device Configuration

[STATUS](#) [BASIC SETTINGS](#) [ADVANCED SETTINGS](#) [FXS PROFILE](#) [FXO PROFILE 1](#) [FXO PROFILE 2](#) [PORTS](#)

FXS Port Settings

Port	SIP User ID	Authenticate ID	Password	Name	Profile ID	Enable Port
1	<input type="text" value="1005"/>	<input type="text" value="1005"/>	<input type="text" value="*****"/>	<input type="text" value="1005"/>	<input type="text" value="FXS Profile"/>	<input type="radio"/> No <input checked="" type="radio"/> Yes

Port Offhook Auto-dial

1	<input type="text"/>
---	----------------------

*Register SIP Account*

After applying your configuration, your account will register to your SIP Server, you can verify if it has been correctly registered with your SIP server from your HT841/HT881 web interface under **Status → Port Status → Registration** (If it displays **Registered**, it means that your account is fully registered, otherwise it will display **Not Registered** so in this case you must double check the settings or contact your provider).

### Grandstream Device Configuration

[STATUS](#) [BASIC SETTINGS](#) [ADVANCED SETTINGS](#) [FXS PROFILE](#) [FXO PROFILE 1](#) [FXO PROFILE 2](#) [PORTS](#)

**MAC Address:** WAN -- C0:74:AD:23:C4:9E LAN -- C0:74:AD:23:C4:9D (Device MAC)  
**WAN IPv4 Address:** 192.168.125.201  
**WAN IPv6 Address:**  
**VPN IPv4 Address:**  
**VPN IPv6 Address:**  
**Product Model:** HT881  
**Serial Number:**  
**Hardware Version:** V1.0A Part Number -- 9660005710A  
**Software Version:** Boot 0.30.0.3  
                                   Core 1.0.0.3  
                                   Base 1.0.0.3  
                                   Prog 1.0.0.3  
                                   CPE  
**Software Status:** Running Mem: 41236  
**System Up Time:** 11:01:06 up 13 days  
**CPU Load:** 20%  
**Network Cable Status:** WAN -- Up 100Mbps Full LAN -- Down 10Mbps Half  
**PPPoE Link Up:** Disabled  
**NAT:** Unknown NAT  
**Certificate Type:** General Certificate  
**Port Status:**

Port	Hook	User ID	Registration
FXS	On Hook	1005	Not Registered
FXO 1	Not Connected		Not Registered
FXO 2	Not Connected		Not Registered
FXO 3	Not Connected		Not Registered
FXO 4	Not Connected		Not Registered
FXO 5	Not Connected		Not Registered
FXO 6	Not Connected		Not Registered
FXO 7	Not Connected		Not Registered
FXO 8	Not Connected		Not Registered

**Port Options:**

Port	DND	Forward	Busy Forward	Delayed Forward	CID	Call Waiting	SRTP
FXS	No				Yes	Yes	No

*Account Registered*

## Rebooting HT841/HT881 from Remote

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

# UPGRADING AND PROVISIONING

The HT841/HT881 can be upgraded via FTP/FTPS/TFTP/HTTP/HTTPS by configuring the URL/IP Address for the FTP/FTPS/TFTP/HTTP/HTTPS server and selecting a download method. Configure a valid URL for TFTP or FTP/FTPS or HTTP/HTTPS (default is HTTPS); the server name can be FQDN or IP address.

## Examples of valid URLs:

firmware.grandstream.com

fw.ipvideotalk.com/gs

## Firmware Upgrade procedure

Please follow below steps in order to upgrade the firmware version of your HT841/HT881:

1. Access your HT881/HT841 UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: Found on a sticker on the back of the unit).
3. Press **Login** to access your settings.
4. Go to **Advanced Settings** → **Firmware Upgrade and Provisioning** page, and enter the IP address or the FQDN for the upgrade server in "**Firmware Server Path**" field and choose to upgrade via **TFTP , FTP/FTPS or HTTP/HTTPS**.
5. Make sure to check "**Always Check for New Firmware**".
6. Update the change by clicking the "**Apply**" button at the bottom of the page. Then "**Reboot**" or power cycle the HT841/HT881 to update the new firmware.

*Firmware Upgrade and Provisioning:*

Upgrade Via  TFTP  HTTP  HTTPS  FTP  FTPS

Firmware Server Path:

Config Server Path:

XML Config File Password:

HTTP/HTTPS/FTP/FTPS User Name:

HTTP/HTTPS/FTP/FTPS Password:

Firmware File Prefix:  Firmware File Postfix:

Config File Prefix:  Config File Postfix:

Enable using tags in URL:  
 No  Yes

Always send HTTP Basic Authentication Information:  
 No  Yes

Allow DHCP Option 66 or 160 to override server:  
 No  Yes

Additional Override DHCP Option:

3CX Auto Provision:  
 No  Yes

Automatic Upgrade:  
 No  
 Yes, every  minutes(30-5256000).  
 Yes, daily at start hour  (0-23), at end hour  (0-23).  
 Yes, weekly on day  (0-6).

Randomized Automatic Upgrade:  No  Yes

Always Check for New Firmware at Boot up  
 Check New Firmware only when F/W pre/suffix changes  
 Always Skip the Firmware Check

## Upgrading via Local Directory

1. Download the firmware file from Grandstream web site
2. Unzip it and copy the file in to a folder in your PC
3. From the HT841/HT881 web interface (Advanced Settings page) you can browse your hard drive and select the folder you previously saved the file.
4. Click "Upload Firmware" and wait few minutes until the new program is loaded.

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

## Upgrading via Local TFTP/HTTP Servers

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

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

[http://www.solarwinds.com/products/freetools/free\\_tftp\\_server.aspx](http://www.solarwinds.com/products/freetools/free_tftp_server.aspx)

<http://tftpd32.jounin.net/>.

Instructions for local firmware upgrade via TFTP:

1. Unzip the firmware files and put all of them in the root directory of the TFTP server.
2. Connect the PC running the TFTP server and the phone to the same LAN segment.
3. Launch the TFTP server and go to the File menu->Configure->Security to change the TFTP server's default setting from "**Receive Only**" to "**Transmit Only**" for the firmware upgrade.
4. Start the TFTP server and configure the TFTP server in the phone's web configuration interface.
5. Configure the Firmware Server Path to the IP address of the PC.
6. Save and Apply the changes and reboot the HT841/HT881.

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

## Firmware and Configuration File Prefix and Postfix

Firmware Prefix and Postfix allows device to download the firmware name with the matching Prefix and Postfix. This makes it the possible to store all of firmware with different version in one single directory. Similarly, Config File Prefix and Postfix allows device to download the configuration file with the matching Prefix and Postfix. Thus, multiple configuration files for the same device can be stored in one directory.

In addition, when the field "Check New Firmware only when F/W pre/suffix changes" is set to "Yes", the device will only issue firmware upgrade request if there are changes in the firmware Prefix or Postfix.

## Managing Firmware and Configuration File Download

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

## Configuration File Download

Grandstream SIP Devices can be configured via the Web Interface as well as via a Configuration File (binary or XML) through TFTP or FTP/FTPS or HTTP/HTTPS. The **Config Server Path** is the TFTP or FTP/FTPS or HTTP/HTTPS server path for the configuration file. It needs to be set to a valid URL, either in FQDN or IP address format. The **Config Server Path** can be the same or different from the **Firmware Server Path**.

A configuration parameter is associated with each particular field in the web configuration page. A parameter consists of a Capital letter P and 2 to 3 (Could be extended to 4 in the future) digit numeric numbers. i.e., P2 is associated with the "New Password" in the Web GUI->Maintenance->Web/SSH Access page->Admin Password. For a detailed parameter list, please refer to the corresponding firmware release configuration template.

When the HT841/HT881 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 "cfgxxxxxxxxxxxx.xml" file is not successful, the provision program will download a generic cfg.xml file. The configuration file name should be in lower case letters.

For more details on XML provisioning, please refer to: <https://documentation.grandstream.com/knowledge-base/sip-device-provisioning-guide/>

## RESTORE FACTORY DEFAULT SETTINGS

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

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

### Using the Reset Button

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

1. Unplug the Ethernet cable.
2. Locate the reset hole on the back panel of your HT841/HT881.
3. Insert a pin in this hole, and press for about 7 seconds.
4. Take out the pin. All unit settings are restored to factory settings

### Using the IVR Command

Reset default factory settings using the IVR prompt:

1. Dial "\*\*\*\*" for voice prompt.
2. Enter "99" and wait for "reset" voice prompt.
3. Enter the encoded MAC address (Look below on how to encode MAC address).
4. Wait 15 seconds and device will automatically reboot and restore factory settings.

### Encode the MAC Address

1. Locate the MAC address of the device. It is the 12-digit HEX number on the bottom of the unit.
2. Key in the MAC address. Use the following mapping:

Key	Mapping
-----	---------

<b>0-9</b>	0-9
<b>A</b>	22 (press the “2” key twice, “A” will show on the LCD)
<b>B</b>	222
<b>C</b>	2222
<b>D</b>	33 (press the “3” key twice, “D” will show on the LCD)
<b>E</b>	333
<b>F</b>	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 HT841/HT881 UI by entering its IP address in your favorite browser.
  2. Enter your admin password (default: Found on a sticker on the back of the unit).
  3. Press **Login** to access your settings.
  4. Go to **Basic Settings** → **Reset Type**
  5. Press **Reset** button (after selecting the reset type).
    - **Full Reset:** This will make a full reset
    - **ISP Data:** This will reset only the basic settings, like IP mode, PPPoE and Web port
    - **VOIP Data:** This will reset only the data related with a service provider like SIP server, sip user ID, provisioning and others.
- Factory Reset will be disabled if the “Lock keypad update” is set to “Yes”.
- If the HT841/HT881 was previously locked by your local service provider, pressing the RESET button will only restart the unit. The device will not return to factory default settings.

## CHANGE LOG

This section documents significant changes from previous versions of the user guide for HT841/HT881. Only major new features or major document updates are listed here. Minor updates for corrections or editing are not documented here.

### Firmware Version 1.0.3.4

- Added support “Polarity On Answer Delay”. [[Polarity On Answer Delay](#)]
- Updated config file download request starts from cfgMAC.xml [[Configuration File Download](#)]
- Added new value “4 – Auto” in NAT Traversal. [[NAT Traversal](#)]
- Added support “AVS DTMF Detector Parameter” in both FXS and FXO profiles. [[DSP DTMF Detector Min Level](#)] [[DSP DTMF Detector SNR](#)] [[DSP DTMF Detector Deviation](#)] [[DSP DTMF Detector Twist](#)]
- Added support “Radius auth protocol”. [[Radius auth protocol](#)]
- Removed “Ring Timeout” from FXO page.

### Firmware Version 1.0.1.2

- This is the Initial Release.
- 

### **Need Support?**

Can't find the answer you're looking for? Don't worry we're here to help!

[CONTACT SUPPORT](#)