

Grandstream Networks, Inc.

HT802 Analog Telephone Adaptor

User Guide



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CAUTION

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

WARNING

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



GNU GPL INFORMATION

HT802 firmware contains third-party software licensed under the GNU General Public License (GPL). Grandstream uses software under the specific terms of the GPL. Please see the GNU General Public License (GPL) for the exact terms and conditions of the license.

Grandstream GNU GPL related source code can be downloaded from Grandstream web site from:

http://www.grandstream.com/sites/default/files/Resources/gpl_ht802.tar.gz



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DOCUMENT PURPOSE

This document describes the basic concept and tasks necessary to use and configure your HT802. And it covers the topic of connecting and configuring the HT802, making basic operations and the call features. Please visit <http://www.grandstream.com/support> to download the latest “HT802 User Guide”.

This guide covers following topics:

- [Product overview](#)
- [Getting started](#)
- [Configuration guide](#)
- [Call features.](#)
- [Call operations.](#)
- [Upgrading and provisioning.](#)
- [Restore factory default settings](#)



CHANGE LOG

This section documents significant changes from previous versions of user guide for HT802. Only major new features or major document updates are listed here. Minor updates for corrections or editing are not documented here.

Firmware Version 1.0.1.1

- This is the initial version for HT802.



GUI INTERFACE EXAMPLES

http://www.grandstream.com/sites/default/files/Resources/ht802_web_gui.zip

1. Screenshot of Login Page
2. Screenshots of Status Page
3. Screenshots of Basic Settings Page
4. Screenshots of Advanced Settings Pages
5. Screenshots of FXS Port 1 Pages
6. Screenshots of FXS Port 2 Pages



WELCOME

The HT802 Analog Telephone Adapter provides transparent connectivity for analog phones and faxes to the world of Internet voice. Connecting to any analog phone, fax or PBX, the HT802 is an effective and flexible solution for accessing Internet-based telephone services and corporate intranet systems across established LAN and Internet connections. The Grandstream Handy Tone 802 is a new addition to the popular Handy Tone ATA product family. This manual will help you learn how to operate and manage your Handy Tone 802 Analog Telephone Adaptor and make the best use of its many upgraded features including simple and quick installation, 3- way conferencing, direct IP-IP Calling, and new provisioning support among other features. This HT802 is very easy to manage and configure, and is specifically designed to be an easy to use and affordable VoIP solution for both the residential user and the teleworker



PRODUCT OVERVIEW

The HT802 is a 2 port analog telephone adapter (ATA) that allows users to create a high-quality and manageable IP telephony solution for residential and office environments. Its ultra-compact size, voice quality, advanced VoIP functionality, security protection and auto provisioning options enable users to take advantage of VoIP on analog phones and enables service providers to offer high quality IP service. The HT802 is an ideal ATA for individual use and for large scale commercial IP voice deployments.

Feature Highlights

The following tables contains the major features of the HT802

- Supports 2 SIP profiles through 2 FXS ports and single 10/100Mbps port
- Supports 3-way voice conferencing
- Supports wide range of caller ID formats
- Supports advanced telephony features, including call transfer, call forward, call-waiting, do not disturb, message waiting indication, multi-language prompts, flexible dial plan and more
- Supports T.38 Fax for creating Fax-over-IP
- TLS and SRTP security encryption technology to protect calls and accounts
- Automated provisioning options include TR-069 and XML config files
- Failover SIP server automatically switches to secondary server if main server loses connection
- GR-909 Line Testing Functionalities

HT802 Technical Specifications

The following table resume all the technical specifications including the protocols / standards supported, voice codecs, telephony features, languages and Upgrade/ Provisioning settings for the HT802.

Table 1: HT802 FXS Analog Telephone Adapter Technical Specifications

Interfaces	
Telephone Interfaces	Two (2) RJ11 FXS ports
Network Interface	One (1) 10/100Mbps auto-sensing Ethernet ports (RJ45)
LED Indicators	POWER, INTERNET, PHONE1, PHONE2
Factory Reset Button	Yes
Voice, Fax, Modem	
Telephony Features	Caller ID display or block, call waiting, flash, blind or attended transfer, forward, hold, do not disturb, 3-way conference
Voice Codecs	G.711 with Annex I (PLC) and Annex II (VAD/CNG), G.723.1, G.729A/B, G.726, iLBC, dynamic jitter buffer, advanced line echo cancellation



Fax over IP	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through
Short/Long Haul Ring Load	2 REN: Up to 1km on 24 AWG
Caller ID	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID
Disconnect Methods	Busy Tone, Polarity Reversal/Wink, Loop Current
Signaling	
Network Protocols	TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP/RARP, ICMP, DNS, DHCP, NTP, TFTP, TELNET, STUN, SIP (RFC3261), SIP over TCP/TLS, SRTP, TR-069
QoS	Layer 3 (ToS, DiffServ, MPLS)
DTMF Method	In-audio, RFC2833 and/or SIP INFO
Provisioning and Control	HTTP, HTTPS, TELNET, TFTP, TR-069, secure and automated provisioning using AES encryption, syslog
Security	
Media	SRTP
Control	TLS/SIPS/HTTPS
Management	Syslog support, telnet, remote management using web browser
Physical	
Universal Power Supply	Input: 100-240VAC, 50-60Hz Output: 5.0VDC/1.0A
Environmental	Operational: 32° – 104°F or 0° – 40°C Storage: 14° – 140°F or -10° – 60°C Humidity: 10 – 90% Non-condensing
Dimension and Weight	100mm x 100mm x 29.5mm, 114g (without package)
Compliance	
Compliance	FCC 15B, AS/NZS CISPR22, AS/NZS60950, EN55022, EN55024, EN60950, EN61000-3-2, EN61000-3-3, UL (Power supply)



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 HT802.

Equipment Packaging

The HT802 ATA package contains:

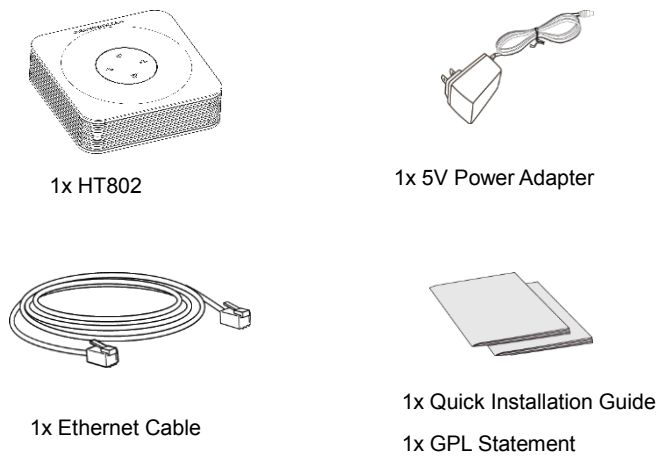


Figure 1: HT802 Package Contents

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

HT802 Ports Description

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

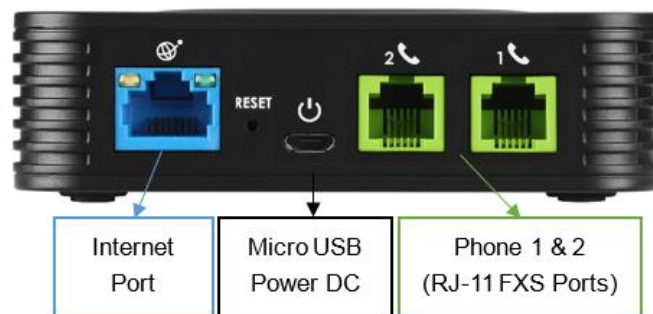


Figure 2: HT802 Back Panel



Table 2: Definition of the HT802 Connectors

Phone 1 & 2	Used to connect the analog phones / fax machines to the phone adapter using an RJ-11 telephone cable.
Internet port	Used to connect the phone adapter to your router or gateway using an Ethernet RJ45 network cable.
Micro USB Power	Used to power the phone adapter (5V – 1A).
Reset	Factory reset button, press for 7 seconds to reset factory default settings.

Connecting HT802

The HT802 are designed for easy configuration and easy installation, to connect your HT802, please follow the steps above:

1. Insert a standard RJ11 telephone cable into the phone port and connect the other end of the telephone cable to a standard touch-tone analog telephone.
2. Insert the Ethernet cable into the internet or LAN port of the HT802 and connect the other end of the Ethernet cable to an uplink port (a router or a modem, etc.)
3. Insert the power adapter into the HT802 and connect it to a wall outlet.
4. The Power, Ethernet, and Phone LEDs will be solidly lit when the Phone Adapter is ready for use.

The HT802 Analog Telephone Adaptor is an all-in-one VoIP integrated device designed to be a total solution for networks providing VoIP services with various VoIP features and functions.



Figure 3: Connecting the HT802

HT802 LEDs Pattern



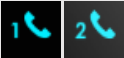
There are four (4) LED buttons that help you manage the status of your Handy Tone 802.





Figure 4: HT802 LEDs Pattern

Table 3 : HT802 LEDs Pattern Description

Led Lights	Status
Power LED 	The Power LED lights up when the HT802 is powered on and it flashes when the HT802 is booting up.
Internet LED 	The Ethernet LED lights up when the HT802 is connected to your network through the Ethernet port and it flashes when there is data being sent or received.
Phone LED 1&2 	The phone LED 1 & 2 indicate status of the respective FXS Ports-phone on the back panel <ul style="list-style-type: none"> • OFF - Unregistered • ON (Solid Blue) - Registered and Available • Blinking every second - Off-Hook / Busy • Slow blinking - FXS LEDs indicates voicemail



CONFIGURATION GUIDE

The HT802 can be configured via one of two ways:

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

Obtain HT802 IP Address via Connected Analogue Phone

HT802 is by default configured to obtain the IP address from DHCP server where the unit is located. In order to know which IP address is assigned to your HT802, you should access to the "[Interactive Voice Response Menu](#)" of your adapter via the connected phone and check its IP address mode.

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

1. Use a telephone connected to phone 1 or phone 2 ports of your HT802.
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 HT802 Interactive Voice Prompt Response Menu

The HT802 has a built-in voice prompt menu for simple device configuration which lists actions, commands, menu choices, and descriptions. The IVR menu work with any phone connected to the HT802. Pick up the handset and dial "****" to use the IVR menu.

Table 4: Voice Prompt Menu

Menu	Voice Prompt	Options
Main Menu	"Enter a Menu Option"	Press "*" for the next menu option Press "#" to return to the main menu Enter 01-05, 07,10, 13-17,47 or 99 menu options
01	"DHCP Mode", "Static IP 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.
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 the HT802 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: <ul style="list-style-type: none"> • PCM U / PCM A • iLBC • G-726 • G-723 • G-729
10	“MAC Address”	Announces the Mac address of the unit.
13	Firmware Server IP Address	Announces current Firmware Server IP address. Enter 12 digit new IP address.
14	Configuration Server IP Address	Announces current Config Server Path IP address. Enter 12 digit new IP address.
15	Upgrade Protocol	Upgrade protocol for firmware and configuration update. Press “ 9 ” to toggle between TFTP / HTTP / HTTPS
16	Firmware Version	Firmware version information.
17	Firmware Upgrade	Firmware upgrade mode. Press “ 9 ” to toggle among the following three options: <ul style="list-style-type: none"> - always check - check when pre/suffix changes - never upgrade
47	“Direct IP Calling”	Enter the target IP address to make a direct IP call, after dial tone. (See “ <i>Make a Direct IP Call</i> ”.)
86	Voice Mail	Number of Voice Mails
99	“RESET”	Press “ 9 ” to reboot the device Enter MAC address to restore factory default setting (See Restore Factory Default Setting section)
	“Invalid Entry”	Automatically returns to main menu
	“Device not registered”	This prompt will be played immediately after off hook If the device is not register and the option “Outgoing Call without Registration” is in NO

Five success tips when using the voice prompt

- “*” shifts down to the next menu option and “#” returns to the main menu
- “9” functions as the ENTER key in many cases to confirm or toggle an option



- All entered digit sequences have known lengths - 2 digits for menu option and 12 digits for IP address. For IP address, add 0 before the digits if the digits are less than 3 (i.e. - 192.168.0.26 should be key in like 192168000026. No decimal is needed).
- Key entry cannot be deleted but the phone may prompt error once it is detected.

Configuration via Web Browser

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

Accessing the Web UI

1. Connect the computer to the same network as your HT802.
2. Make sure the HT802 is booted up.
3. You may check your HT802 IP address using the IVR on the connected phone. Please see [Obtain HT802 IP Address Via Connected Analogue Phone](#)
4. Open Web browser on your computer.
5. Enter the HT802's IP address in the address bar of the browser.
6. Enter the administrator's password to access the Web Configuration Menu.

Note:

The computer must be connected to the same sub-network as the HT802. This can be easily done by connecting the computer to the same hub or switch as the HT802.

Web UI Access Level Management

There are two default passwords for the login page:

User Level	Password	Web Pages Allowed
End User Level	123	Only Status and Basic Settings
Administrator Level	admin	All pages

The password is case sensitive with maximum length of 25 characters.

When changing any settings, always submit them by pressing the **Update** or **Apply** button on the bottom of the page. After submitting the changes in all the Web GUI pages, reboot HT802 to have the changes take effect if necessary; most of the options under the **Advanced Settings** and **FXS Port (x)** pages require reboot

Saving the Configuration Changes

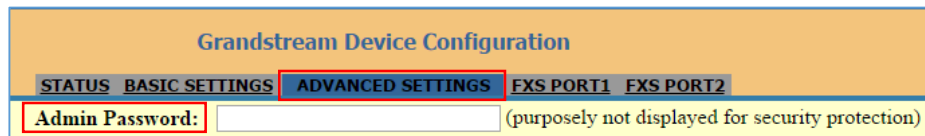
After users makes changes to the configuration, pressing the **Update** button will save but not apply the changes until the **Apply** button is clicked. Users can instead directly press the **Apply** button. We



recommend rebooting or powering cycle the phone after applying all the changes.

Changing Admin Level Password

1. Access your HT802 web UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: admin).
3. Press **Login** to access your settings.
4. Go to **Advanced Settings > Admin Password**.
5. Enter the new admin password
6. Press **Apply** at the bottom of the page to save your new settings.

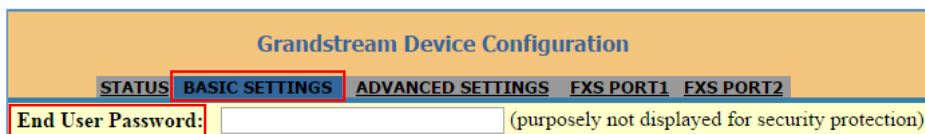


The screenshot shows the 'Grandstream Device Configuration' page with the 'ADVANCED SETTINGS' tab selected. The 'Admin Password' field is highlighted with a red box, and its value is obscured by a grey bar with the text '(purposely not displayed for security protection)'.

Figure 5: Admin Level Password

Changing User Level Password

1. Access your HT802 web UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: admin).
3. Press **Login** to access your settings.
4. Go to **Basic Settings > End User Password**.
5. Enter the new user password
6. Press **Apply** at the bottom of the page to save your new settings.

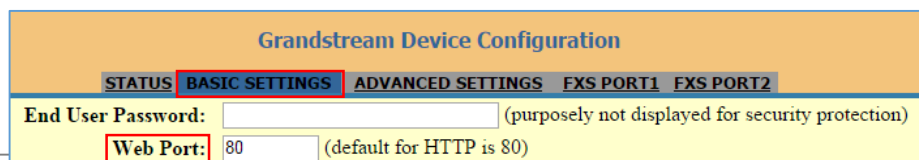


The screenshot shows the 'Grandstream Device Configuration' page with the 'BASIC SETTINGS' tab selected. The 'End User Password' field is highlighted with a red box, and its value is obscured by a grey bar with the text '(purposely not displayed for security protection)'.

Figure 6: User Level Password

Changing HTTP Web Port

1. Access your HT 802 web UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: admin).
3. Press **Login** to access your settings.
4. Go to **Basic Settings > Web Port**.
5. Change the current port to your desired/new HTTP port. Ports accepted are in range [1-65535].
6. Press **Apply** at the bottom of the page to save your new settings



The screenshot shows the 'Grandstream Device Configuration' page with the 'BASIC SETTINGS' tab selected. The 'Web Port' field is highlighted with a red box and contains the value '80'. A note next to it says '(default for HTTP is 80)'. The 'End User Password' field above it is also visible and obscured by a grey bar with the text '(purposely not displayed for security protection)'.



NAT Settings

Figure 7: Web HTTP Port

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

1. STUN Server (under advanced settings webpage) Enter a STUN server IP (or FQDN) that you may have, or look up a free public STUN Server on the internet and enter it on this field. If using Public IP, keep this field blank.
2. Use Random SIP/RTP Ports (under advanced settings webpage) This setting depends on your network settings. Generally, if you have multiple IP devices under the same network, it should be set to Yes. If using a public IP address, set this parameter to No.
3. NAT traversal (under the FXS web page) Set this to Yes when gateway is behind firewall on a private network.

DTMF Methods

The HT802 support the following DTMF mode:

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

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

Preferred Vocoder (Codec)

The HT802 supports following voice codecs. On FXS ports pages, choose the order of your favorite codecs:

- PCMU/A (or G711 μ /a)
- G729 A/B
- G723.1
- G726 (16/24/32/40)
- iLBC

Configuring The HT802 Through Voice Prompts

As mentioned previously, The HT802 has a built-in voice prompt menu for simple device configuration. Please refer to "[Understanding HT802 Interactive Voice Prompt Response Menu](#)" for more information about IVR and how to access its menu.



- **DHCP MODE**

Select voice menu option 01 to allow the HT802 to use DHCP.

- **STATIC IP MODE**

Select voice menu option 01 to allow the HT802 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.

- **FIRMWARE SERVER IP ADDRESS**

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

- **CONFIGURATION SERVER IP ADDRESS**

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

- **UPGRADE PROTOCOL**

Select the menu option 15 to choose firmware and configuration upgrade protocol between TFTP / HTTP.

- **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”.

Register a SIP Account

The HT802 supports 2 FXS ports which can be configured with 2 SIP accounts. Please refer to the following steps in order to register your accounts via web user interface

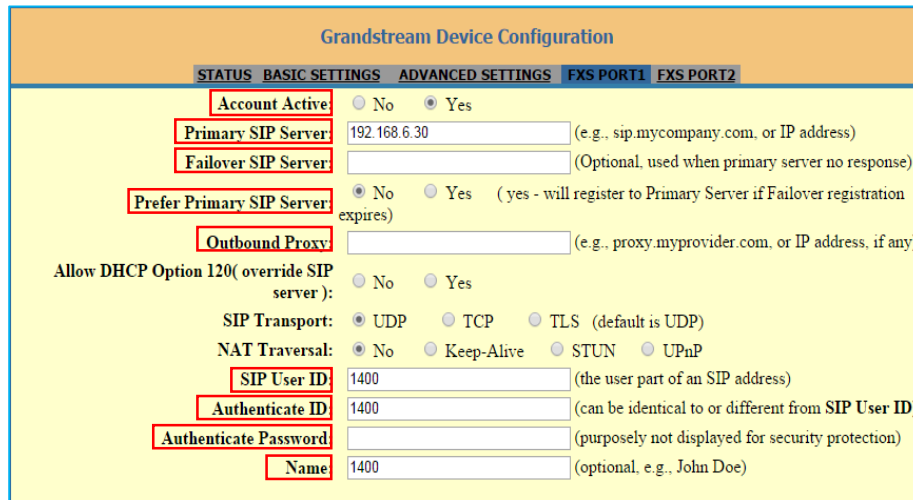
1. Access your HT802 web UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: admin).
3. Press **Login** to access your settings.
4. Go to **FXS Port (1 or 2)** pages.
5. In **FXS Port** tab, set the following:
 - a. **Account Active** to **Yes**.
 - b. **Primary SIP Server** field with your SIP server IP address or FQDN.
 - c. **Failover SIP Server** with your Failover SIP Server IP address or FQDN. Leave empty if not available.
 - d. **Prefer Primary SIP Server** to **No** or **Yes** depending on your configuration. Set to **No** if no Failover SIP Server is defined. If “**Yes**”, account will register to Primary SIP Server when failover registration expires.
 - e. **Outbound Proxy**: Set your Outbound Proxy IP Address or FQDN. Leave empty if not available.
 - f. **SIP User ID**: User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
 - g. **Authenticate ID**: SIP service subscriber’s Authenticate ID used for authentication. Can be identical to or different from SIP User ID.



- h. **Authenticate Password:** SIP service subscriber's account password to register to SIP server of ITSP. For security reasons, the password will field will be shown as empty.
- i. **Name:** Any name to identify this specific user.

For more information, related to above options please refer to [FXS Port Settings](#).

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

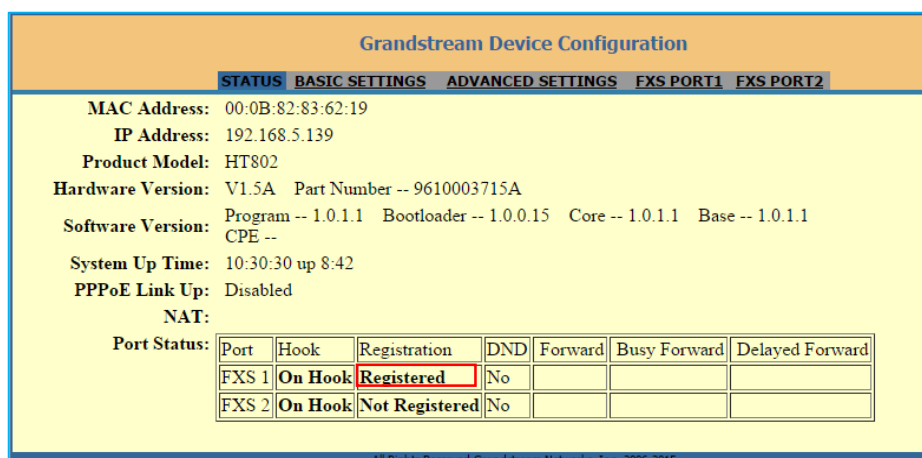


The screenshot shows the 'SIP ACCOUNTS SETTINGS' tab in the Grandstream Device Configuration interface. The settings are as follows:

- Account Active:** Yes
- Primary SIP Server:** 192.168.6.30 (e.g., sip.mycompany.com, or IP address)
- Failover SIP Server:** (Optional, used when primary server no response)
- Prefer Primary SIP Server:** No Yes (yes - will register to Primary Server if Failover registration expires)
- Outbound Proxy:** (e.g., proxy.myprovider.com, or IP address, if any)
- Allow DHCP Option 120(override SIP server):** No Yes
- SIP Transport:** UDP TCP TLS (default is UDP)
- NAT Traversal:** No Keep-Alive STUN UPnP
- SIP User ID:** 1400 (the user part of an SIP address)
- Authenticate ID:** 1400 (can be identical to or different from SIP User ID)
- Authenticate Password:** (purposely not displayed for security protection)
- Name:** 1400 (optional, e.g., John Doe)

Figure 8: SIP Accounts Settings

After applying your configuration, your account will register to your SIP Server, you can verify if it has been correctly registered with your SIP server from your HT802 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).



The screenshot shows the 'STATUS' tab in the Grandstream Device Configuration interface, displaying account status information:

- MAC Address:** 00:0B:82:83:62:19
- IP Address:** 192.168.5.139
- Product Model:** HT802
- Hardware Version:** V1.5A Part Number -- 9610003715A
- Software Version:** Program -- 1.0.1.1 Bootloader -- 1.0.0.15 Core -- 1.0.1.1 Base -- 1.0.1.1 CPE --
- System Up Time:** 10:30:30 up 8:42
- PPPoE Link Up:** Disabled
- NAT:**

Port Status:

Port	Hook	Registration	DND	Forward	Busy Forward	Delayed Forward
FXS 1	On Hook	Registered	No			
FXS 2	On Hook	Not Registered	No			

Figure 9: Account Status

Rebooting The HT802 from Remote

Press the "Reboot" button at the bottom of the configuration menu to reboot the ATA remotely. The web



browser will then display a message window to confirm that reboot is underway. Wait 30 seconds to log in again.



CALL FEATURES

The HT802 supports all the traditional and advanced telephony features.

Table 5: HT802 Call Features

Key	Call features
*02	Forcing a Codec (per call) *027110 (PCMU), *027111 (PCMA), *02723 (G723), *02729 (G729), *0272616 (G726-r16), *0272624 (G724-r24), *0272632 (G726-r32), *0272640 (G726-r40), *027201 (iLBC)
*03	Disable LEC (per call) Dial “*03” +” number “. No dial tone is played in the middle.
*16	Enable SRTP
*17	Disable SRTP
*30	Block Caller ID (for all subsequent calls)
*31	Send Caller ID (for all subsequent calls)
*47	Direct IP Calling . Dial “*47” + “IP address”. No dial tone is played in the middle.
*50	Disable Call Waiting (for all subsequent calls)
*51	Disable Call Waiting (for all subsequent calls)
*67	Block Caller ID (per call) . Dial “*67” +” number”. No dial tone is played in the middle.
*82	Send Caller ID (per call) . Dial “*67” +” number”. No dial tone is played in the middle.
*69	Call Return Service : Dial *69 and the phone will dial the last incoming phone number received.
*70	Disable Call Waiting (per call) . Dial “*70” +” number”. No dial tone is played in the middle.
*71	Enable Call Waiting (per call) . Dial “*71” +” number”. No dial tone is played in the middle
*72	Unconditional Call Forward : Dial “*72” and then the forwarding number followed by “#”. Wait for dial tone and hang up. (dial tone indicates successful forward)
*73	Cancel Unconditional Call Forward . To cancel “Unconditional Call Forward”, dial “*73”, wait for dial tone, then hang up.
*74	Enable Paging Call : Dial “*74” and then the destination phone number you want to page.
*78	Enable Do Not Disturb (DND) : When enabled all incoming calls are rejected.
*79	Disable Do Not Disturb (DND) : When disabled, incoming calls are accepted.



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



CALL OPERATIONS

Placing a phone call

To make the outgoing calls using your HT802

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, a 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 HT802 and other VoIP Device, have public IP addresses, or
- Both HT802 and other VoIP Device are on the same LAN using private IP addresses, or
- Both HT802 and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

The HT802 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.

Note: NO dial tone will be played between step 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 direct IP call, the “**Use Random SIP/RTP Port**” should set to “NO”. You cannot make direct IP calls between FXS1 to FXS2 since they are using same IP.

Call hold

You can place a call on hold by pressing the “flash” button on the analog phone (if the phone has that button).

Press the “flash” button again to release the previously held Caller and resume conversation. If no “flash” button is available, use “hook flash” (toggle on-off hook quickly). You may drop a call using hook flash.

Call waiting

The call waiting tone (3 short beeps) indicates an incoming call, if the call waiting feature is enabled.

To toggle between incoming call and current call, you need to press the “flash” button the first call is placed on hold.

Press the “flash” button to toggle between the active calls.

Call transfer

Blind transfer

Assume that the call is established between phone A and B are in conversation. The phone A wants to *blind transfer* phone B to phone C:

1. On the phone A presses FLASH to hear the dial tone.
2. The phone A dials *87 then dials caller C’s number, and then # (or wait for 4 seconds)
3. The phone A will hear the dial tone. Then, A can hang up.

Note: “Enable Call Feature” must be set to “Yes” in web configuration page.

Attended transfer

Assume that the call is established between phone A and B are in conversation. The phone A wants to *attend transfer* phone B to phone C:



1. On the phone A presses FLASH to hear the dial tone.
2. Phone A dials the phone C's number followed by # (or wait for 4 seconds).
3. If phone C answers the call, phones A and C are in conversation. Then A can hang up to complete transfer.
4. If phone C does not answer the call, phone A can press "flash" to resume call with phone B.

Note: When attended transfer fails and A hangs up, the HT802 will ring back user A to remind A that B is still on the call. A can pick up the phone to resume conversation with B.

3-Way conferencing

The HT802 supports Bellcore style 3-way Conference. To perform the 3-way conference, we assume that the call is established between phone A and B are in conversation. Phone A(HT802) wants to bring third phone C into conference:

1. Phone A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
2. Phone A dials C's number then # (or wait for 4 seconds).
3. If phone C answers the call, then A presses FLASH to bring B, C in the conference.
4. If phone C does not answer the call, phone A can press FLASH back to talk to phone B.
5. If phone A presses FLASH during conference, the phone C will be dropped out.
6. If phone A hangs up, the conference will be terminated for all three parties when configuration "Transfer on Conference Hang up" is set to "No". If the configuration is set to "Yes", A will transfer B to C so that B and C can continue the conversation.

Call Return

In order to call back to the latest incoming number.

1. Pick up the handset of the connected phone (Off-hook).
2. After hearing the dial tone, input "*69", and follow by "#" or wait for time out.
3. Your phone will automatically call back to the latest incoming number.

Note: All star code (*XX) related features mentioned above are supported by ATA default settings. If your service provider provides different feature codes, please contact them for instructions.



UPGRADING AND PROVISIONING

The HT802 can be upgraded via TFTP/HTTP/HTTPS by configuring the URL/IP Address for the TFTP/HTTP/HTTPS server and selecting a download method. Configure a valid URL for TFTP or HTTP/HTTPS; the server name can be FQDN or IP address.

Examples of valid URLs:

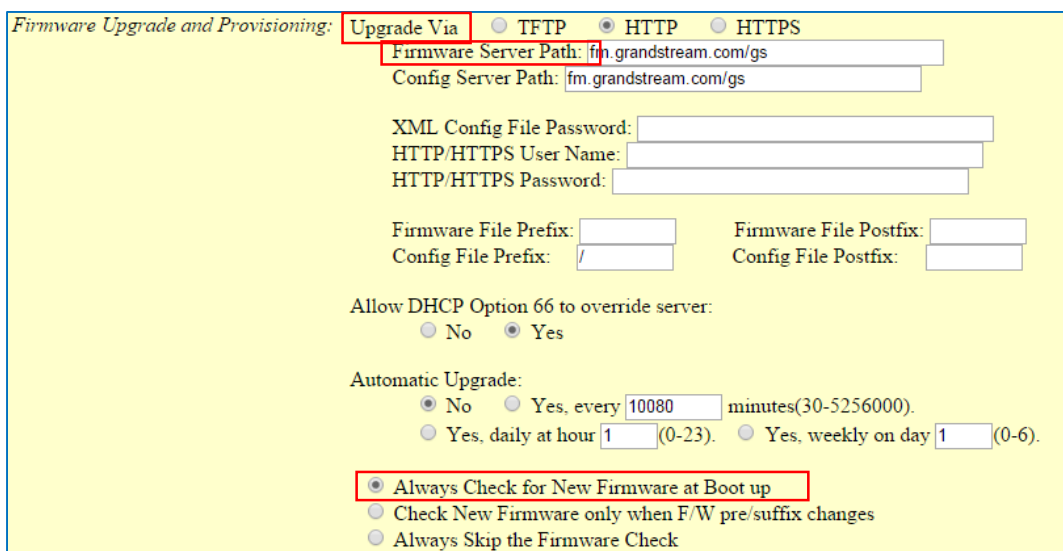
firmware.grandstream.com

fw.ipvideotalk.com/gs

Firmware Upgrade procedure

Please follow below steps in order to upgrade the firmware version of your HT802:

1. Access your HT802 UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: admin).
3. Press **Login** to access your settings.
4. Go to **Advanced Settings > Firmware Upgrade and Provisioning** page, and enter the IP address or the FQDN for the upgrade server in "**Firmware Server Path**" field and choose to upgrade via **TFTP** or **HTTP/HTTPS**.
5. Make sure to check "**Always Check for New Firmware**".
6. Update the change by clicking the "**Apply**" button at the bottom of the page. Then "**Reboot**" or power cycle the HT802 to update the new firmware.



Firmware Upgrade and Provisioning: Upgrade Via TFTP HTTP HTTPS

Firmware Server Path: fw.grandstream.com/gs

Config Server Path: fw.grandstream.com/gs

XML Config File Password:

HTTP/HTTPS User Name:

HTTP/HTTPS Password:

Firmware File Prefix: Firmware File Postfix:

Config File Prefix: Config File Postfix:

Allow DHCP Option 66 to override server:
 No Yes

Automatic Upgrade:
 No Yes, every 10080 minutes(30-5256000).
 Yes, daily at hour 1 (0-23). Yes, weekly on day 1 (0-6).

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

Figure 10: Firmware Upgrade Page



Upgrading via Local Directory:

1. Download the firmware file from Grandstream web site
2. Unzip it and copy the file in to a folder in your PC
3. From the HT802 web interface (Advanced Settings page) you can browse your hard drive and select the folder you previously saved the file (HT802fw.bin)
4. Click "Upload Firmware" and wait few minutes until the new program is loaded.

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

Upgrading via Local TFTP/HTTP Servers

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

<http://www.grandstream.com/support/firmware>

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

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

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

Instructions for local firmware upgrade via TFTP:

1. Unzip the firmware files and put all of them in the root directory of the TFTP server.
2. Connect the PC running the TFTP server and the phone to the same LAN segment.
3. Launch the TFTP server and go to the File menu->Configure->Security to change the TFTP server's default setting from "**Receive Only**" to "**Transmit Only**" for the firmware upgrade.
4. Start the TFTP server and configure the TFTP server in the phone's web configuration interface.
5. Configure the Firmware Server Path to the IP address of the PC.
6. Save and Apply the changes and reboot the HT802.

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

Firmware and Configuration File Prefix and Postfix

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



Managing Firmware and Configuration File Download

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

Configuration File Download

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

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

When the HT802 boots up or reboots, it will send a request to download a file named "cfgxxxxxxxxxxx" followed by a configuration XML file named "cfgxxxxxxxxxxx.xml", where "xxxxxxxxxxx" is the MAC address of the phone, i.e., "cfg000b820102ab" and "cfg000b820102ab.xml". If the download of "cfgxxxxxxxxxxx.xml" file is not successful, the provision program will download a generic cfg.xml file. The configuration file name should be in lower case letters.

For more details on XML provisioning, please refer to:

http://www.grandstream.com/sites/default/files/Resources/gs_provisioning_guide.pdf



RESTORE FACTORY DEFAULT SETTINGS

Warning:

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

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

Using the Reset Button

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

1. Unplug the Ethernet cable.
2. Locate the reset hole on the back panel of your HT802.
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:
 - 0-9: 0-9
 - A: 22 (press the “2” key twice, “A” will show on the LCD)
 - B: 222



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

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

Reset from Web Interface (Reset Type)

1. Access your HT802 UI by entering its IP address in your favorite browser.
2. Enter your admin password (default: admin).
3. Press **Login** to access your settings.
4. Go to **Basic Settings > Reset Type**
5. Press **Reset** button (after selecting the reset type).

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

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

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

Note:

- Factory Reset will be disabled if the “Lock keypad update” is set to “Yes”.
- If the HT802 was previously locked by your local service provider, pressing the RESET button will only restart the unit. The device will not return to factory default settings.



EXPERIENCING HT802

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

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

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

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

