



Grandstream Networks, Inc.

HT- 502

Dual FXS Port

Analog Telephone Adaptor



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CONFIGURATION GUI INTERFACE EXAMPLES

HT-502 USER MANUAL

http://www.grandstream.com/user_manuals/GUI/GUI_HT502.rar

1. SCREENSHOT OF ADVANCED USER CONFIGURATION PAGE
2. SCREENSHOT OF BASIC SETTINGS CONFIGURATION PAGE
3. SCREENSHOT OF FXS PORT 1 CONFIGURATION LOGIN PAGE
4. SCREENSHOT OF FXS PORT 2 CONFIGURATION PAGE
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WELCOME

Thank you for purchasing Grandstream's HT-502, the affordable, feature rich Analog Telephone Adaptor. Grandstream HandyTone-502 is a new addition to the popular HandyTone ATA product family. It features the rich audio quality, a broad range of voice codecs, and functionality of the HT-502, including two (2) FXS ports each with independent SIP accounts.

This manual will help you learn how to operate and manage your HandyTone-502 Analog Telephone Adaptor and make the best use of its many upgraded features including simple and quick installation, 3-way conferencing, and direct IP-IP Calling. This HT-502 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 tele-worker.

SAFETY COMPLIANCES

The HT-502 phone complies with FCC/CE and various safety standards. The HT-502 power adaptor is compliant with UL standard. Only use the universal power adapter provided with the HT-502 package. The manufacturer's warranty does not cover damages to the phone caused by unsupported power adaptors.

WARRANTY

If you purchased your HT-502 from a reseller, please contact the company where you purchased your phone for replacement, repair or refund. If you purchased the product directly from Grandstream, contact your Grandstream Sales and Service Representative for a RMA (Return Materials Authorization) number before you return the product. Grandstream reserves the right to remedy warranty policy without prior notification.

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. Please do not use a different power adaptor with the HT-502 as it may cause damage to the products and void the manufacturer warranty.*

- This document is contains links to Grandstream GUI Interfaces. Please download these examples http://www.grandstream.com/user_manuals/GUI/GUI_HT502.rar for your reference.
- This document is subject to change without notice. The latest electronic version of this user manual is available for download @: http://www.grandstream.com/user_manuals/HT502_User_Manual.pdf
- *Reproduction or transmittal of the entire or any part, in any form or by any means, electronic or print, for any purpose is not permitted without the express written permission of Grandstream Networks, Inc.*

CONFIGURE YOUR HT-502

Configuring your HT-502 and connecting the unit to the VoIP network is very simple. The HT-502 is easy to configure using the embedded GUI pages and the following five (5) steps outlined below. Examples of the GUI Interfaces can be downloaded from: http://www.grandstream.com/user_manuals/GUI/GUI_HT502.rar. Before you begin, please verify the contents of the HT-502 package.

EQUIPMENT PACKAGING

Unpack and check all accessories. Equipment included in the package:

- one 12V universal power adapter
- one Ethernet cable
- one device unit
- one HT502 stand

CONNECT YOUR ATA

First, familiarize yourself with the features of the HT-502. The HT-502 has two FXS ports. Each FXS port can have a separate SIP account. This is a key feature of HT-502. Both ports can make calls concurrently.

FIGURE 1: CONNECTING THE HT-502

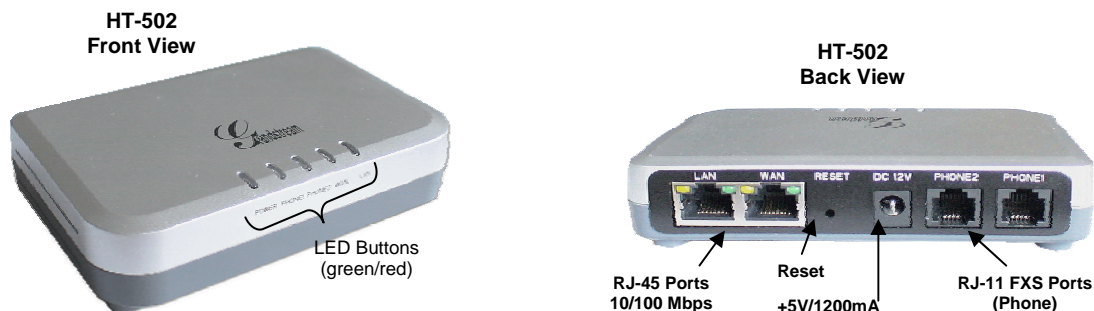


TABLE 1: DEFINITIONS OF THE HT-502 CONNECTORS

Power Cable	Power adapter connection
WAN Port (RJ-45)	Connect to the internal LAN network or router.
LAN Port (RJ-45)	Connect the LAN port with an Ethernet cable to your PC.
RESET	Factory Reset button. Press for 7 seconds to reset factory default settings.
PHONE1 (RJ-11)	FXS port to be connected to analog phones / fax machines.
PHONE2 (RJ-11)	FXS port to be connected to analog phones / fax machines.

There are five (5) LED buttons that help you manage the status of your HandyTone.

TABLE 2: DEFINITIONS OF THE HT-502 LEDs

Power LED	Indicates Power. Remains ON when Power is connected and turned ON.
WAN LED	Indicates LAN (or WAN) port activity.
LAN LED	Indicates PC (or LAN) port activity.
PHONE1 / PHONE2 LED	Indicate status of the respective FXS Ports-PHONE1 / PHONE2 on the back panel Busy – ON (Solid Green) Available – OFF Slow blinking FXS LEDs indicates voicemail for that port.

NOTE: All LEDs display green when ON. Slow blinking of Power, WAN and LAN LED together indicate the product in firmware upgrading or provision state

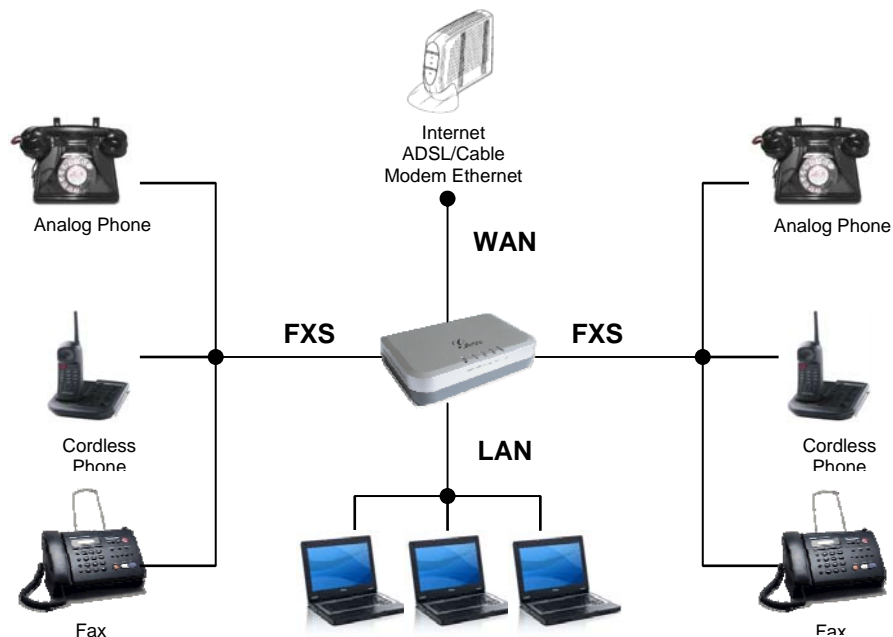
CONFIGURE YOUR ATA

The HT-502 is easy to configure using the embedded GUI pages and the following five (5) steps.

FIVE EASY STEPS TO CONFIGURE THE HT-502

1. Connect a standard touch-tone analog telephone (or fax machine) to first FXS port.
2. Connect another standard touch-tone analog telephone (or fax machine) to second FXS port.
3. Insert the Ethernet cable into the WAN port of HT-502 and connect the other end of the Ethernet cable to an uplink port (a router or a modem, etc.)
4. Connect a PC to the LAN port of HT-502.
5. Insert the power adapter into the HT-502 and connect it to a wall outlet.

FIGURE 2: CONFIGURING THE HT-502



PRODUCT OVERVIEW

The HT-502 is a full feature voice and fax-over IP device that offers a high-level of integration including dual 10M/100Mbps network ports with integrated router, NAT, DHCP server, dual port FXS telephone gateway, market-leading sound quality, rich functionalities, and a compact and lightweight design. The HT-502 fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market. Moreover, it supports comprehensive voice codecs including G.711 (a/μ-law), G.723.1, G.726 (16/24/32/48 bit rates), G.729A/B/E and iLBC.

KEY FEATURES

Ethernet Ports	DHCP	FXS Port	PSTN Pass – through	Voice Mail Indicator	Voice Codec	Remote Configuration
2 RJ-45 (LAN)	Server/Client	2	No	Yes	iLBC, G.711, G.723, G.726(16/24/32/48), T.38, G.729A/B/E	TFTP/HTTP

TABLE3: HT-502 TECHNICAL SPECIFICATIONS

HT-502 Analog Telephone Adaptor

Telephone Interfaces	2 FXS ports, 2 SIP accounts
Network Interface	Two (2) 10M/100 Mbps, RJ-45
LED Indicators	Power, WAN, LAN, PHONE1 and PHONE2
Reset Button	Factory Reset button.
Voice over Packet Capabilities	Voice Activity Detection (VAD) with CNG (comfort noise generation) and PLC (packet loss concealment), Dynamic Jitter Buffer, Modem detection & auto-switch to G.711, Packetized Voice Protocol Unit (supports RTP/RTCP and AAL2 protocol), G.168 compliant Echo Cancellation, LEC (line echo cancellation) with NLP, Asymmetric RTP stream
Voice Compression	G.711 + Annex I (PLC), Annex II (VAD/CNG format) encoder and decoder, G.723.1A, G.726(ADPCM), G.729A/B/E, iLBC, G.726 provides proprietary VAD, CNG, and signal power estimation, Voice Play Out unit (reordering, fixed and adaptive jitter buffer, clock synchronization), AGC (automatic gain control), Status output, Decoder controlling via voice packet header
DHCP Server/Client	Yes, NAT Router or Switched Mode
Telnet Server	Yes
Fax over IP	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through (pending), Fax Datapump V.17, V.19, V.27ter, V.29 for T.38 fax relay
QoS	Diffserve, TOS, 802.1 P/Q VLAN tagging
IP Transport	RTP/RTCP
DTMF Method	Flexible DTMF transmission method, user interface of In-audio, RFC2833, and/or SIP Info
IP Signaling	SIP (RFC 3261)
Provisioning	TFTP, HTTP, HTTPS (pending)
Control	TLS/SIPS , SIP over TCP/TLS
Management	Syslog support, HTTPS and telnet (pending), remote management using Web browser, Support Layer 2 (802.1Q, VLAN, 802.1p) and Layer 3 QoS (Tos, DiffSery, MPLS), Auto/manual provisioning system
Dial Plan	Yes
UPnP	Yes
Power	<i>Output: 12VDC / Input: 100–240 VAC/50-60 Hz</i>
Environmental	Operational: 32°–104°F or 0°–40°C Storage: 10°–130° F / Humidity: 10–90% Non-condensing
Dimensions (H x W x D)	115mm (L) x 75mm (W) x 27mm (H)
Short and long haul	REN3: Up to150 ft on 24 AWG line
Call Handling Features	Caller ID display or block, Call waiting caller ID, Call waiting/flash, Call transfer, hold, forward, mute, 3-way conferencing, message waiting, Do-Not-Disturb (DND), call-return service
Caller ID	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID
Polarity Reversal / Wink	Yes
EMC	EN55022/EN55024 and FCC part15 Class B
Safety	UL

TABLE 4: HT-502 HARDWARE SPECIFICATIONS

LAN Interface	2 x RJ45 10/100Mbps (integrated router)
LED	5 LEDs (GREEN)
Universal Switching Power Adaptor	Input: 100-240V AC, 50/60Hz, 0.5A Max Output: 12V DC, 1.25A UL certified
Dimension	115mm (L) x 75mm (W) x 27mm (H)
Weight	94 g (0.21lbs)
Temperature	32~104°F / 0~40°C
Humidity	10% - 90% (non-condensing)
Compliance	FCC, CE

BASIC OPERATIONS

BECOMING FAMILIAR WITH THE VOICE PROMPTS

HT-502 stores a voice prompt menu (Interactive Voice Response or **IVR**) for quick browsing and simple configuration. The IVR menu and the LED button work with any of the FXS port. To enter the IVR menu, pick up the handset and dial “***”.

TABLE 5: HT-502 IVR MENU DEFINITIONS

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”	Press “9” to toggle the selection If using “ <i>Static IP Mode</i> ”, configure the IP address information using menus 02 to 05. If using “ <i>Dynamic IP Mode</i> ”, 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 “ <i>Static IP Mode</i> ”, enter 12 digit new IP address.
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.
12	WAN Port Web Access	Press "9" to toggle between enable / disable
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
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> ".)
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

Five Success Tips when using the Voice Prompt

1. "*" shifts down to the next menu option
2. "#" returns to the main menu
3. "9" functions as the ENTER key in many cases to confirm an option
4. 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 192168**0000**26. No decimal is needed).
5. Key entry can not be deleted but the phone may prompt error once it is detected

PLACING A PHONE CALL

PHONE OR EXTENSION NUMBERS

1. Dial the number directly and wait for 4 seconds (Default "No Key Entry Timeout"); or
2. 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:

1. Both HT-502 and other VoIP Device, have public IP addresses, or
2. Both HT-502 and other VoIP Device are on the same LAN using private IP addresses, or
3. Both HT-502 and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

HT-502 supports two ways to make Direct IP Calling:

Using IVR

1. Pick up the analog phone then access the voice menu prompt by dial “****”
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.

Destination ports can be specified using “*” (encoding for “:”) followed by the port number.

Examples:

- 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 Port**” should set to “NO”. You cannot make direct IP calls between FXS1 to FXS2 since they are using same IP.

CALL HOLD

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

Call waiting tone (3 short beeps) indicates an incoming call, if the call waiting feature is enabled. Toggle between incoming call and current call by pressing the “flash” button. First call is placed on hold. Press the “flash” button to toggle between two active calls.

CALL TRANSFER

Blind Transfer

Assume that call Caller A and B are in conversation. A wants to *Blind Transfer* B to C:

3. Caller A presses **FLASH** on the analog phone to hear the dial tone.
4. Caller A dials ***87** then dials caller C's number, and then # (or wait for 4 seconds)
5. Caller A will hear the confirm tone. Then, A can hang up.

NOTE: "*Enable Call Feature*" must be set to "Yes" in web configuration page.

Caller A can place a call on hold and wait for one of three situations:

1. A quick confirmation tone (similar to call waiting tone) followed by a dial-tone. This indicates the transfer is successful (transferee has received a 200 OK from transfer target). At this point, Caller A can either hang up or make another call.
2. A quick busy tone followed by a restored call (on supported platforms only). This means the transferee has received a 4xx response for the INVITE and we will try to recover the call. The busy tone is just to indicate to the transferor that the transfer has failed.
3. Continuous busy tone. The phone has timed out. **Note:** continuous busy tone does not indicate the transfer has been successful, nor does it indicate the transfer has failed. It often means there was a failure to receive second NOTIFY – check firmware for most recent release.

Attended Transfer

Assume that Caller A and B are in conversation. Caller A wants to *Attend Transfer* B to C:

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

NOTE: When Attended Transfer fails and A hangs up, the HT-502 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 HT-502 supports Bellcore style 3-way Conference.

Instructions for 3-way conference:

Assue that call party A and B are in conversation. Caller A(HT502) wants to bring third Caller C into conference:

1. A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
2. A dials C's number then # (or wait for 4 seconds).
3. If C answers the call, then A presses FLASH to bring B, C in the conference.
4. If C does not answer the call, A can press FLASH back to talk to B.
5. If A presses FLASH during conference, C will be dropped out

Call Features

The HT-502 supports all the traditional and advanced telephony features.

TABLE 6: HT-502 CALL FEATURE DEFINITIONS

Key	Call Features
*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. Detail see Direct IP Calling section on page 12.
*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.
*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.

SENDING AND RECEIVING FAX

HT-502 supports fax in two modes: 1) T.38 (Fax over IP) and 2) fax pass through.

T.38 is the preferred method because it is more reliable and works well in most network conditions. If the service provider supports T.38, please use this method by selecting T.38 as fax mode (default). If the service provider does not support T.38, pass-through mode may be used. If you have problems with sending or receiving Fax, toggle the Fax Tone Detection Mode setting.

CONFIGURATION GUIDE

CONFIGURING THE HT-502 VIA VOICE PROMPTS

DHCP MODE

Select voice menu option 01 to enable HT-502 to use DHCP.

STATIC IP MODE

Select voice menu option 01 to enable HT-502 to use 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 voice menu option 15 to choose firmware and configuration upgrade protocol. User can choose between TFTP and HTTP.

FIRMWARE UPGRADE MODE

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

WAN PORT WEB ACCESS

Select voice menu option 12 to enable WAN Port Web Access of the device configuration pages.

CONFIGURING THE HT-502 USING THE WEB BROWSER

HT-502 has an embedded Web server that will respond to HTTP GET/POST requests. It also has embedded HTML pages that allow users to configure the HT-502 through a web browser such as Microsoft's IE and AOL's Netscape.

ACCESS THE WEB CONFIGURATION MENU

The HT-502 HTML configuration menu can be accessed via LAN or WAN port:

From the LAN port:

1. Directly connect a computer to the LAN port.
2. Open a command window on the computer
3. Type in "ipconfig /release", the IP address etc. becomes 0.
4. Type in "ipconfig /renew", the computer gets an IP address in 192.168.2.x segment by default
5. Open a web browser, type in the default gateway IP address. <http://192.168.2.1>. You will see the login page of the device.

From the WAN port:

The WAN port HTML configuration option is disabled by default from factory. To access the HTML configuration menu from the WAN port:

1. Enable the "WAN Port Web Access" option via IVR option 12.
2. Find the WAN IP address of the HT-502 using voice prompt menu option 02.
3. Access the HT-502 Web Configuration page by the following URI via WAN port: <http://HandyTone-IP-Address> (the HT IP-Address is the WAN IP address for the HT-502).

NOTE: If using a web browser to enter the configuration page, strip the leading "0"s because the browser will parse in octet. (i.e. if the IP address is: 192.168.001.014, please type in: 192.168.1.14).

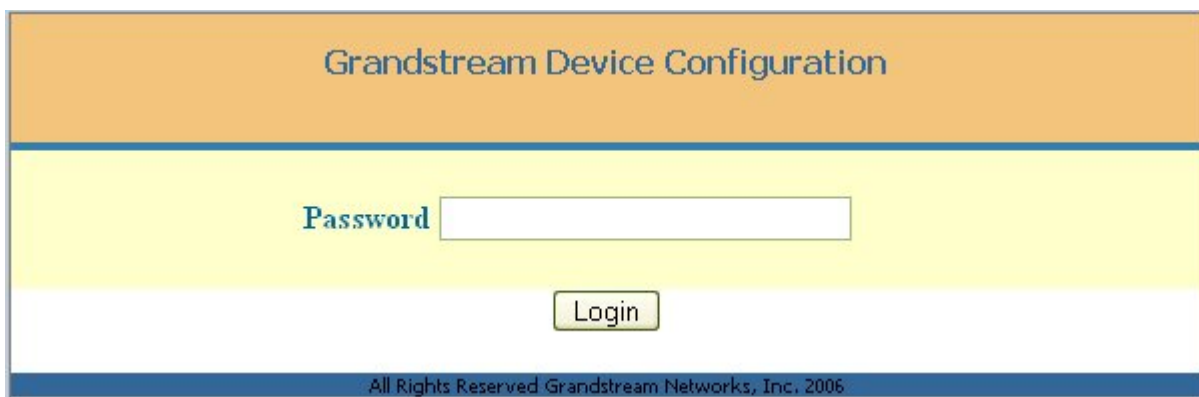
END USER CONFIGURATION

Once the HTTP request is sent from a web browser, the user will see a log in screen. 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	Browse all pages

Only an administrator can access the “*ADVANCED SETTINGS*” configuration page.

FIGURE 3: SCREENSHOT OF CONFIGURATION LOG IN PAGE



NOTE: If you cannot log into the configuration page by using default password, please check with the VoIP service provider. The service provider may have provisioned and configured the device for you. The Basic Configuration Page is the first web GUI the user will see.

Important Settings

The end-user must configure the following settings according to the local environment. **NOTE:** Most settings on the web configuration pages are set to the default values.

NAT SETTINGS

If you plan to keep the gateway within a *private network* behind a firewall, we recommend using STUN Server. The following three (3) 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 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 Profile web pages)
Set this to Yes when gateway is behind firewall on a private network.

DTMF METHODS

DTMF Settings are in [Profile pages](#).

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

Enable one or more DTMF methods based on your PBX system.

PREFERRED VOCODER (CODEC)

The HT-502 supports a broad range of voice codecs. Under Profile web pages, choose your preferred order of different codecs:

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

TABLE 7: HT-502 BASIC CONFIGURATION SETTINGS DEFINITIONS

End User Password	Password to access the Web Configuration Menu. This field is case sensitive with a maximum length of 25 characters.
Web Port	By default, HTTP uses port 80. This field is for customizable web port.
Telnet Server	Default is set to YES.
IP Address	There are two modes to operate the HT-502: DHCP mode: all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The HT-502 acquires its IP address from the first DHCP server it discovers from the LAN it is connected. <i>Using the PPPoE feature:</i> set the PPPoE account settings. The HT-502 will establish a PPPoE session if any of the PPPoE fields is set. Static IP mode: configure the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields. These fields are set to zero by default.
DHCP hostname	This option specifies the name of the client. This field is optional but may be required by some Internet Service Providers. Default is blank.
DHCP domain	This option specifies the domain name that client should use when resolving hostnames via the Domain Name System. Default is blank.
DHCP vendor class ID	Used by clients and servers to exchange vendor-specific information. Default is HT500.
PPPoE account ID	PPPoE username. Necessary if ISP requires you to use a PPPoE (Point to Point Protocol over Ethernet) connection.
PPPoE password	PPPoE account password.
PPPoE Service Name	This field is optional. If your ISP uses a service name for the PPPoE connection, enter the service name here. Default is blank.
Time Zone	Controls how the date/time is displayed according to the specified time zone.
Self Defined Time Zone	It uses the standard GLIBC TZ syntax. http://ftp.gnu.org/gnu/Manuals/glibc-2.2.5/html_node/TZ-Variable.html#TZ%20Variable .
Language	Languages supported with voice prompt
Device Mode	This parameter controls whether the device is working in NAT router mode or Bridge mode. Save the setting and reboot prior to configuring HT-502.
Enable support UPnP	If set to “Yes”, the HT-502 would act as an UPnP gateway for your UPnP enable applications. UPnP - Universal Plug n Play
Reply to ICMP on WAN port	If set to “Yes”, the HT-502 will respond to the PING command from other computers, but it also is vulnerable to the DOS attack. Default is No .

WAN HTTP/Telnet Access	side	If set to “Yes”, user can access the configuration page through the WAN port, instead of through the “PC” port. <u>Warning</u> : this configuration is less secure than default option. Default is No .
Cloned WAN MAC Addr		This allows you to change/set the MAC address on the WAN interface.
LAN Subnet Mask		Sets the LAN subnet mask. Default value is 255.255.255.0
LAN DHCP Base IP		Base IP for the LAN port which functions as a Gateway for the subnet. Default value is 192.168.2.1 .
DHCP IP Lease Time		Value is set in units of hours. Default value is 120 hrs (5 Days.) The time IP address is assigned to the LAN clients.
DMZ IP		Forward all WAN IP traffic to a specific IP address if no matching port is used by HT-502 or defined in port forwarding.
Port Forwarding		Forwards a matching (TCP/UDP) port to a specific LAN IP address with a specific (TCP/UDP) port.

In addition to the Basic Settings configuration page, end users also have access to the Device Status page.

TABLE 8: HT-502 DEVICE STATUS PAGE DEFINITIONS

MAC Address	The device ID in HEX format. This is needed for ISP troubleshooting. Note there are separate MAC addresses for the WAN side and the LAN side.
WAN IP Address	Shows WAN IP address of HT-502.
Product Model	Contains the product model info.
Software Version	Program: This is the main software release. Boot and Loader are seldom changed.
System Up Time	Shows system up time since the last reboot.
PPPoE Link Up	Indicates whether the PPPoE connection is up if the HT-502 is connected to DSL modem.
NAT	Indicates the type of NAT connection used by the HT-502 via its WAN port. Based on STUN protocol.

Port Status	Shows several information regarding the individual FXS ports.					
Port	Hook	Registration	DND	Forward	Busy Forward	Delayed Forward
FXS1	On Hook	Registered	Yes	613		
FXS2	On Hook	Registered	No		614	

- Both FXS port1 and FXS port2 are registered with this SIP Server.
- FXS Port 1 user has set Do Not Disturb.
- FXS Port 1 user has set his calls to be forwarded unconditionally to ext 613.
- FXS Port 2 user has set his calls to forward to 614 when his phone is busy.

ADVANCED USER CONFIGURATION

Log in to the advanced user configuration page the same way as for the basic configuration page. The password is case sensitive and the factory default password for Advanced User is “*admin*”.

Advanced User configuration includes the end user configuration and the advanced configurations including: a) SIP configuration, b) Codec selection, c) NAT Traversal Setting and d) other miscellaneous configuration. HT-502 each FXS SIP account has its own configuration page. Their configurations are identical.

TABLE 9: HT-502 ADVANCED CONFIGURATION PAGE DEFINITIONS

Admin Password	This contains the password to access the Advanced Web Configuration page. This field is case sensitive. Only the administrator can configure the “Advanced Settings” page. Password field is purposely left blank for security reasons after clicking update and saved. The maximum password length is 25 characters.
Layer 3 QoS	This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv or MPLS. Default value is 48 .
Layer 2 QoS	Value used for layer 2 VLAN tag. Default setting is blank .
STUN Server	IP address or Domain name of the STUN server.
Keep-alive interval	This parameter specifies how often the HT-502 sends a blank UDP packet to the SIP server in order to keep the “hole” on the NAT open. Default is 20 seconds. Minimum value is 20 seconds.
Firmware Upgrade and Provisioning	Enables HT-502 to download firmware or configuration file through either the TFTP or HTTP server.

Via TFTP Server	<p>This is the IP address of the configured TFTP server. If selected and it is non-zero or not blank, the HT-502 retrieves the new configuration file or new code image from the specified TFTP server at boot time. After 5 attempts, the system will timeout and will start the boot process using the existing code image in the Flash memory. If a TFTP server is configured and a new code image is retrieved, the new downloaded image is saved into the Flash memory.</p> <p><u>Note:</u> Please do NOT interrupt the TFTP upgrade process (especially the power supply) as this will damage the device. Depending on the network environment this process can take up to 15 or 20 minutes.</p>
Via HTTP Server	<p>The URL for the HTTP server used for firmware upgrade and configuration via HTTP.</p> <p><i>For example, ttp://provisioning.mycompany.com:6688/Grandstream/1.0.0.34</i> “:6688” is the specific TCP port where the HTTP server is listening; it can be omitted if using default port 80.</p> <p><u>Note:</u> If Auto Upgrade is set to No, HT-502 will only do HTTP download once at boot up.</p>
Firmware Server Path	IP address or domain name of firmware server.
Config Server Path	IP address or domain name of configuration server.
Firmware File Prefix	Default is blank. If configured, HT-502 will request firmware file with the prefix. This setting is useful for ITSPs. End user should keep it blank.
Firmware File Postfix	Default is blank. End user should keep it blank.
Config File Prefix	Default is blank. End user should keep it blank.
Config File Postfix	Default is blank. End user should keep it blank.
Automatic Upgrade	Choose “Yes” to enable automatic upgrade and provisioning. In “Check for new firmware every” field, enter the number of days to enable HT-502 to check the server for firmware upgrade or configuration in the defined period of days. When set to No, HT-502 will only do upgrade once at boot up. “Always check for New Firmware.” Check New Firmware only when F/W pre/suffix changes”
Firmware Key	Used for firmware encryption. Should be 32 digit in hexadecimal representation. End user should keep it blank.
Authenticate Conf File	If set to Yes, config file is authenticated before acceptance. This protects the configuration from an unauthorized change.
Firmware Key	Used for firmware encryption. Should be 32 digit in hexadecimal representation. End user should keep it blank.
Lock Keypad Update	If set to “Yes”, the configuration update via keypad is disabled.
Disable Voice Prompt	Default is No

Disable Direct IP-IP Calling	Default is No.
NTP server	URI or IP address of the NTP (Network Time Protocol) server. Used by the phone to synchronize the date and time.
Syslog Server	The IP address or URL of System log server. This feature is especially useful for the ITSP (Internet Telephone Service Provider)
Syslog Level	<p>Select the HT-502 to report the log level. Default is NONE. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events:</p> <ol style="list-style-type: none"> 1. product model/version on boot up (INFO level) 2. NAT related info (INFO level) 3. sent or received SIP message (DEBUG level) 4. SIP message summary (INFO level) 5. inbound and outbound calls (INFO level) 6. registration status change (INFO level) 7. negotiated codec (INFO level) 8. Ethernet link up (INFO level) 9. SLIC chip exception (WARNING and ERROR levels) 10. memory exception (ERROR level) <p>The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components:</p> <p>GS_LOG: [device MAC address][error code] error message</p> <p><i>Example:</i> May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up</p>

TABLE 10: HT-502 INDIVIDUAL ACCOUNT SETTINGS DEFINITIONS

Profile Active	When set to Yes the FXS port is activated.
SIP Server	SIP Server's IP address or Domain name provided by VoIP service provider.
Outbound Proxy	IP address or Domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by HT-502 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.
SIP transport	User can select UDP or TCP or TLS.
NAT Traversal (STUN)	<p>This parameter defines whether or not the HT-502 NAT traversal mechanism is activated. If activated (by choosing "Yes") and a STUN server is also specified, then the HT-502 performs according to the STUN client specification. Using this mode, the embedded STUN client will detect if and what type of firewall/NAT is being used. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the HT-502 will use its mapped public IP address and port in all of its SIP and SDP messages.</p> <p>If the NAT Traversal field is set to "Yes" <i>with no specified STUN server</i>, the HT-502 will periodically (every 20 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open.</p>

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.
Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
Authenticate Password	SIP service subscriber's account password.
Name	SIP service subscriber's name for Caller ID display.
Use DNS SRV	Default is No . If set to "Yes" the client will use DNS SRV to look up server.
User ID is Phone Number	If the HT-502 has an assigned PSTN telephone number, this field should be set to "Yes". Otherwise, set it to "No". If "Yes" is set, a "user=phone" parameter will be attached to the "From" header in SIP request.
SIP Registration	Controls whether the HT-502 needs to send REGISTER messages to the proxy server. The default setting is Yes.
Unregister on Reboot	Default is No. If set to Yes, the SIP user's registration information will be cleared on reboot.
Outgoing Call w/o Registration	Default is No . If set to "Yes," user can place outgoing calls even when not registered (if allowed by ITSP) but is unable to receive incoming calls.
Register Expiration	This parameter allows the user to specify the time frequency (in minutes) the HT-502 refreshes its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
Local SIP port	Defines the local SIP port the HT-502 will listen and transmit. The default value for FXS port 1 is 5060. The default value for FXS port 2 is 5062.
Local RTP port	Defines the local RTP-RTCP port pair the HT-502 will listen and transmit. It is the base RTP port for channel 0. When configured, channel 0 uses this port _value for RTP and the port_value+1 for its RTCP; channel 1 uses port_value+2 for RTP and port_value+3 for its RTCP. The default value for FXS port 1 is 5004. The default value for FXS port 2 is 5012.
Use Random Port	This parameter forces the random generation of both the local SIP and RTP ports when set to Yes. This is usually necessary when multiple HT-502 are behind the same NAT.
Refer to Use Target Contact	Default is NO. If set to YES, then for Attended Transfer, the "Refer-To" header uses the transferred target's Contact header information.
DTMF Payload Type	Sets the payload type for DTMF using RFC2833.
DTMF in-audio	Send DTMF as inband (in-audio).
DTMF via RFC2833	Send DTMF via RTP (According to RFC 2833).
DTMF via SIP INFO	Send DTMF via SIP INFO message.
Send Flash Event	Default is No. If set to yes, flash will be sent as DTMF event.
Enable Call Features	Default is Yes . (If Yes, call features using star codes will be supported locally)
Offhook Auto-Dial	This parameter allows users to configure a User ID or extension number to be automatically dialed upon off-hook. <u>Only the user part</u> of a SIP address needs is entered here. The HT-502 will automatically append the "@" and the host portion of the corresponding SIP address.

Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Use NAT IP	NAT IP address used in SIP/SDP message. Default is blank.
Distinctive Ring Tone	Custom Ring Tone 1 to 3 with associate Caller ID: when selected, if Caller ID is configured, then the device will ONLY uses 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.
Disable Call Waiting	Default is No .
Disable Call Waiting Tone	Default is No . This is to disable the stutter Call Waiting Tone when a Call Waiting call arrives. The CWCID will still be displayed.
Ring Timeout	Incoming call will stop ringing when not picked up given a specific period of time.
No Key Entry Timeout	Default is 4 seconds .
Early Dial	<p>Default is No. Use only if proxy supports 484 response. This parameter controls whether the phone will send an early INVITE each time a key is pressed when a user dials a number. If set to "Yes", an INVITE is sent using the dial-number collected thus far; Otherwise, no INVITE is sent until the "(Re-)Dial" button is pressed or after about 5 seconds have elapsed if the user forgets to press the "Re-Dial" button. The "Yes" option should be used ONLY if there is a SIP proxy configured and the proxy server supports 484 Incomplete Address response. Otherwise, the call will likely be rejected by the proxy (with a 404 Not Found error).</p> <p><i><u>This feature is NOT designed to work with and should NOT be enabled for direct IP-to-IP calling.</u></i></p>
Dial Plan Prefix	Sets the prefix added to each dialed number.
Use # as Dial Key	Allows users to configure the "#" key as the "Send" (or "Dial") key. If set to "Yes", "#" will send the number. In this case, this key is essentially equivalent to the "Dial" key. If set to "No", this "#" key can be included as part of number.
Dial Plan	<p>Dial Plan Rules:-</p> <ol style="list-style-type: none"> Accept Digits: 1,2,3,4,5,6,7,8,9,0,*,# Grammar: x - any digit from 0-9; <ol style="list-style-type: none"> xx+ - at least 2 digit number; ^ - exclude; [3-5] - any digit of 3, 4, or 5; [147] - any digit 1, 4, or 7; <2=011> - replace digit 2 with 011 when dialing <ul style="list-style-type: none"> 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 of leading digits 1900 and add prefix 1617 for any dialed 7 digit numbers Example 3: {1xxx[2-9]xxxxxx <2=011>x+} – Allow any length of number with leading digit 2 and 10 digit-numbers of leading digit 1 and leading exchange number between 2 and 9; If leading digit is 2, replace leading digit 2 with 011 before dialing Default: Outgoing - {x+}
Subscribe for MWI	Default is No . When set to "Yes" a SUBSCRIBE for Message Waiting Indication will be sent periodically.

Send Anonymous	If this parameter is set to "Yes", the "From" header along with Privacy and P_Asserted_Identity headers in outgoing INVITE message will be set to anonymous, blocking Caller ID.
Anonymous Call Rejection	Default is No . If set to Yes, incoming calls with anonymous Caller ID will be rejected with 486 Busy message.
Special Feature	Default is Standard . Choose the selection to meet some special requirements from Softswitch vendors.
Preferred Vocoder	The HT-502 supports up to 5 different Vocoder types including G.711 A-/U-law, G.726 (Supports bit rates 16, 24, 32 and 40), G.723.1, G.729A/B/E and iLBC. The user can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder is entered by choosing the appropriate option in "Choice 1". The last Vocoder is entered by choosing the appropriate option in "Choice 8".
G723 Rate	Defines the encoding rate for G.723 vocoder. By default, 6.3kbps rate is chosen.
iLBC Frame Size	Sets the iLBC frame size in 20ms or 30ms
iLBC Payload type	Defines payload type for iLBC. Default value is 97. The valid range is between 96 and 127.
G726-16 Payload type	Default value is 98. Range is from 96 to 127.
G726 - 24 Payload type	Default value is 99. Range is from 96 to 127.
G726 - 32 Payload type	Default value is 100. Range is from 96 to 127.
G726 - 40 Payload type	Default value is 103. Range is from 96 to 127.
G729E payload type	Default value is 102. Range is from 96 to 127.
VAD	Default is No . VAD allows detecting the absence of audio and conserve bandwidth by preventing the transmission of "silent packets" over the network.
Symmetric RTP	Default is No . When set to Yes the device will change the destination to send RTP packets to the source IP address and port of the inbound RTP packet last received by the device.
Fax Mode	T.38 (Auto Detect) FoIP by default, or Pass-Through (must use codec PCMU/PCMA)
Fax Tone Detection Mode	Default is Callee. This decides whether Caller or Callee sends out the re-INVITE for T.38 or Fax Pass Through.
Jitter Buffer Type	Select either Fixed or Adaptive based on network conditions.
Jitter Buffer Length	Select Low, Medium or High based on network conditions.
SLIC Setting	Dependent on standard phone type (and location)
Caller ID Scheme	<ul style="list-style-type: none"> • Bellcore/Telcordia • ETSI-FSK • ETSI-DTMF • SIN 227 - BT • NTT Japan
Polarity Reversal	Default is No . If set to "Yes", polarity will be reversed upon call establishment and termination.
Loop Current Disconnect	Set it to Yes if the traditional PBX you are using with HT502 uses this method for signaling call termination. Default is No .
Hook Flash Timing	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.

Gain	Handset volume adjustment. <ul style="list-style-type: none"> • RX is for receiving volume, • TX is for transmission volume. Default values are 0dB for both parameters. Loudest volume: +6dB Lowest volume: -6dB.
Call Progress/ Ring Tones	Configure ring or tone frequencies according to preference. By default tones are set to North American frequencies. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds.

SAVING THE CONFIGURATION CHANGES

Click the “Update” button in the Configuration page to save the changes to the HT–502 configuration. The following screen confirms that the changes are saved. Reboot or power cycle the HT–502 to make the changes take effect.

FIGURE 4: SCREENSHOT OF SAVE CONFIGURATION PAGE



REBOOTING THE HT–502 FROM REMOTE

Remotely reboot the HT–502 by clicking the “Reboot” button at the bottom of the configuration page. When finished, re-login to the HT–502 after waiting for about 30 seconds.

CONFIGURATION THROUGH A CENTRAL SERVER

Grandstream HT-502 can be automatically configured from a central provisioning system.

When HT-502 boot up, it will send TFTP or HTTP request to download configuration file, “cfg000b82xxxxx”, where “000b82xxxxx” is the LAN side MAC address of the HT-502

Download the configuration files via TFTP or HTTP from the central server. A service provider or an enterprise with large deployment of HT-502 can easily manage the configuration and service provisioning of individual devices remotely from a central server.

GAPS – GRANDSTREAM AUTOMATED PROVISIONING SYSTEM

Grandstream provides a licensed provisioning system called GAPS (Grandstream Automated Provisioning System) that is used to support automated configuration of HT-502. GAPS uses enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual HT-502 for firmware upgrade, remote reboot, etc.

Grandstream provides GAPS service to VoIP service providers. Use GAPS for either simple redirection or with certain special provisioning settings. At boot-up, Grandstream devices by default point to Grandstream provisioning server GAPS, based on the unique MAC address of each device, GAPS provision the devices with redirection settings so that they will be redirected to customer's TFTP or HTTP server for further provisioning. Grandstream also provide GAPSLite software package which contains our NAT friendly TFTP server and a configuration tool to facilitate the task of generating device configuration files.

The GAPSLite configuration tool is now free to end users. The tool and configuration template are available for download from <http://www.grandstream.com/configurationtool.html>

SOFTWARE UPGRADE

Software upgrade can be done via either TFTP or HTTP. The corresponding configuration settings are in the ADVANCED SETTINGS configuration page.

FIRMWARE UPGRADE THROUGH TFTP/HTTP

To upgrade via TFTP or HTTP, the “Firmware Upgrade and Provisioning upgrade via” field needs to be set to TFTP or HTTP, respectively. “Firmware Server Path” needs to be set to a valid URL of a TFTP or HTTP server, server name can be in either FQDN or IP address format. Here are examples of some valid URL.

e.g. firmware.mycompany.com:6688/Grandstream/1.0.0.39

e.g. 168.75.215.189

NOTES:

- TFTP server in IP address format can be configured via IVR. Please refer to section CONFIGURATION GUIDE for instructions. If TFTP server is in FQDN format, it must be set via web configuration interface.
- End users recommended using our TFTP server. Its address can be found at <http://www.grandstream.com/firmware.html>. Currently, the TFTP server, your HT502 can be upgraded from has an IP address 168.75.215.189. For companies, we recommend to maintain their own TFTP/ HTTP server for upgrade and provisioning procedures.
- Once a “Firmware Server Path” is set, user needs to update the settings and reboot the device. If the configured firmware server is found and a new code image is available, the HT ATA will attempt to retrieve the new image files by downloading them into the HT-502’s SRAM. During this stage, the HT-502’s LEDs will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will then be saved into the Flash. If TFTP/HTTP fails for any reason (e.g., TFTP/HTTP server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the HT-502 will stop the TFTP/HTTP process and simply boot using the existing code image in the flash.
- Firmware upgrade may take as long as 1 to 20 minutes over Internet, or just 20+ seconds if it is performed on a LAN. It is recommended to conduct firmware upgrade in a controlled LAN environment if possible. For users who do not have a local firmware upgrade server, Grandstream provides a NAT-friendly TFTP server on the public Internet for firmware upgrade. Please check the Services section of Grandstream’s Web site to obtain our public TFTP server’s IP address.
- Alternatively, user can download a free TFTP or HTTP server and conduct local firmware upgrade. A free windows version TFTP server is available for download from <http://support.solarwinds.net/updates/New-customerFree.cfm>. Our latest official release can be downloaded from <http://www.grandstream.com/y-firmware.htm>.

Directions to Download TFTP Server:

1. Unzip the file and put all of them under the root directory of the TFTP server.
2. Put the PC running the TFTP server and the HT–502 device in the same LAN segment.
3. Please go to File -> 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, in the phone's web configuration page
5. Configure the Firmware Server Path with the IP address of the PC
6. Update the change and reboot the unit

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

CONFIGURATION FILE DOWNLOAD

Grandstream SIP Device can be configured via Web Interface as well as via Configuration File through TFTP or HTTP. “Config Server Path” is the TFTP or HTTP server path for configuration file. It needs to be set to a valid URL, either in FQDN or IP address format. The “Config Server Path” can be 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 “Admin Password” in the ADVANCED SETTINGS page. For a detailed parameter list, please refer to the corresponding firmware release configuration template.

When Grandstream Device boots up or reboots, it will issue request for configuration file named “cfgxxxxxxxxxx”, where “xxxxxxxxxx” is the LAN side MAC address of the device, i.e., “cfg000b820102ab”. The configuration file name should be in lower cases.

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 to “Yes”, Service Provider can use P193 (Auto Check Interval, in minutes, default and minimum is 60 minutes) to have the devices periodically check with either Firmware Server or Config Server, whenever they are defined. 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.

AutomaticUpgrade:

No Check every day Check every week

RESTORE FACTORY DEFAULT SETTING

WARNING! Restoring the Factory Default Setting will DELETE all configuration information of the phone. Please BACKUP or PRINT out all the settings before you approach to following steps. Grandstream will not take any responsibility if you lose all the parameters of setting and cannot connect to your VoIP service provider.

There are two ways to reset the device.

RESET VIA THE RESET BUTTON

1. Locate a needle-sized hole on the back panel of the HT-502 unit next to the power connection.
2. Insert a pin in this hole, and press for about 7 seconds. The back LEDs for LAN and WAN will be solid on to indicate the reset.
3. Take out the pin. All unit settings are restored to factory settings.

RESET VIA IVR

1. Find the MAC address of the device. It is a 12 digits HEX number located on the bottom of the unit.
2. Encode the MAC address. Please use the following mapping:
0-9: 0-9
A: 22
B: 222
C: 2222
D: 33
E: 333
F: 3333

For example, if the MAC address is 000b8200e395, it should be encoded as "0002228200333395".
3. To perform factory reset:
 - Pick up the headset and dial "****" for voice prompt.
 - Enter "99" and get the voice prompt "Reset".
 - Enter the encoded MAC address of the device.
 - Wait for 15 seconds. The device will reboot automatically and restore to factory default setting.

NOTE:

1. Factory Reset will be disabled if the "**Lock keypad update**" is set to "Yes".
2. Please be aware by default the HT-502 WAN side HTTP access is disabled. After a factory reset, the device's web configuration page can be accessed only from its LAN port.

GLOSSARY OF TERMS

ADSL Asymmetric Digital Subscriber Line: Modems attached to twisted pair copper wiring that transmit from 1.5 Mbps to 9 Mbps downstream (to the subscriber) and from 16 kbps to 800 kbps upstream, depending on line distance.

AGC Automatic Gain Control is an [electronic system](#) found in many types of devices. Its purpose is to control the [gain](#) of a system in order to maintain some measure of performance over a changing range of real world conditions.

ARP Address Resolution Protocol is a protocol used by the [Internet Protocol \(IP\) \[RFC826\]](#), specifically IPv4, to map [IP network addresses](#) to the hardware addresses used by a data link protocol. The protocol operates below the network layer as a part of the interface between the OSI network and OSI link layer. It is used when [IPv4 is used over Ethernet](#)

ATA Analogue Telephone Adapter. Covert analogue telephone to be used in data network for VoIP, like Grandstream HT series products.

CODEC Abbreviation for Coder-Decoder. It's an analog-to-digital (A/D) and digital-to-analog (D/A) converter for translating the signals from the outside world to digital, and back again.

CNG Comfort Noise Generator, generate artificial background [noise](#) used in [radio](#) and [wireless](#) communications to fill the [silent](#) time in a transmission resulting from [voice activity detection](#).

DATAGRAM A data packet carrying its own address information so it can be independently routed from its source to the destination computer

DECIMATE To discard portions of a signal in order to reduce the amount of information to be encoded or compressed. Lossy compression algorithms ordinarily decimate while sub-sampling.

DECT Digital Enhanced Cordless Telecommunications: A standard developed by the European Telecommunication Standard Institute from 1988, governing pan-European digital mobile telephony. DECT covers wireless PBXs, telepoint, residential cordless telephones, wireless access to the public switched telephone network, Closed User Groups (CUGs), Local Area Networks, and wireless local loop. The DECT Common Interface radio standard is a multi-carrier time division multiple access, time division duplex (MC-TDMA-TDD) radio transmission technique using ten radio frequency channels from 1880 to 1930 MHz, each divided into 24 time slots of 10ms, and twelve full-duplex accesses per carrier, for a total of 120 possible combinations. A DECT base station (an RFP, Radio Fixed Part) can transmit all 12 possible accesses (time slots) simultaneously by using different frequencies or using only one frequency. All signaling information is transmitted from the RFP within a multi-frame (16 frames). Voice signals are digitally encoded into a 32 Kbit/s signal using Adaptive Differential Pulse Code Modulation.

DNS Short for *Domain Name System* (or *Service* or *Server*), an [Internet](#) service that translates [domain names](#) into IP addresses

DID Direct Inward Dialing. The ability for an outside caller to dial to a PBX extension without going through an attendant or auto-attendant.

DSP Digital Signal Processor. A specialized CPU used for digital signal processing. Grandstream products all have DSP chips built inside.

DTMF Dual Tone Multi Frequency. The standard tone-pairs used on telephone terminals for dialing using in-band signaling. The standards define 16 tone-pairs (0-9, #, * and A-F) although most terminals support only 12 of them (0-9, * and #).

FQDN Fully Qualified Domain Name. A FQDN consists of a host and domain name, including top-level domain. For example, www.grandstream.com is a fully qualified domain name. www is the host, Grandstream is the second-level domain, and and.com is the top level domain.

FXS Foreign eXchange Office. An FXS device can be an analog phone, answering machine, fax, or anything that handles a call from the telephone company like AT&T. They should also operate the same way when connected to an FXS interface.

- An FXS interface will accept calls from FXS or PSTN interfaces. All countries and regions have their own standards.
- FXS is complimentary to FXS (and the PSTN).

FXS Foreign eXchange Station. An FXS device has hardware to generate the ring signal to the FXS extension (usually an analog phone).

- An FXS device will allow any FXS device to operate as if it were connected to the phone company. This makes your PBX the POTS+PSTN for the phone.
- The FXS Interface connects to FXS devices (by an FXS interface, of course).

DHCP The *Dynamic Host Configuration Protocol* (DHCP) is an Internet protocol for automating the configuration of computers that use TCP/IP. DHCP can be used to automatically assign IP addresses, to deliver TCP/IP stack configuration parameters such as the subnet mask and default router, and to provide other configuration information such as the addresses for printer, time and news servers.

ECHO CANCELLATION Echo Cancellation is used in [telephony](#) to describe the process of removing [echo](#) from a voice communication in order to improve voice quality on a [telephone call](#). In addition to improving quality, this process improves [bandwidth](#) savings achieved through [silence suppression](#) by preventing echo from traveling across a [network](#). There are **two types** of echo of relevance in telephony: acoustic echo and hybrid echo. [Speech compression](#) techniques and [digital processing](#) delay often contribute to echo generation in [telephone networks](#).

H.323 A suite of standards for multimedia conferences on traditional packet-switched networks.

HTTP Hyper Text Transfer Protocol; the World Wide Web protocol that performs the request and retrieve functions of a server

IP Internet Protocol. A packet-based protocol for delivering data across networks.

IP-PBX IP-based Private Branch Exchange

IP Telephony (Internet Protocol telephony, also known as Voice over IP Telephony) A general term for the technologies that use the Internet Protocol's packet-switched connections to exchange voice, fax, and other forms of information that have traditionally been carried over the dedicated circuit-switched connections of the public switched telephone network (PSTN). The basic steps involved in originating an IP Telephony call are conversion of the analog voice signal to digital format and compression/translation of the signal into Internet protocol (IP) packets for transmission over the Internet or other packet-switched networks; the process is reversed at the receiving end. The terms IP Telephony and Internet Telephony are often used to mean the same; however, they are not 100 per cent interchangeable, since Internet is only a subcase of packet-switched networks. For users who have free or fixed-price Internet access, IP Telephony software essentially provides free telephone calls anywhere in the world. However, the challenge of IP Telephony is maintaining the quality of service expected by subscribers. Session border controllers resolve this issue by providing quality assurance comparable to legacy telephone systems.

IVR IVR is a software application that accepts a combination of voice telephone input and touch-tone keypad selection and provides appropriate responses in the form of voice, fax, callback, e-mail and perhaps other media.

MTU A Maximum Transmission Unit (MTU) is the largest size [packet](#) or [frame](#), specified in [octets](#) (eight-bit bytes), that can be sent in a packet- or frame-based network such as the Internet. The maximum for Ethernet is 1500 byte.

NAT Network Address Translation

NTP Network Time Protocol, a protocol to exchange and synchronize time over networks The port used is UDP 123 Grandstream products using NTP to get time from Internet

OBP/SBC Outbound Proxy or another name Session Border Controller. A device used in [VoIP](#) networks. OBP/SBCs are put into the signaling and media path between calling and called Caller. The OBP/SBC acts as if it was the called VoIP phone and places a second call to the called Caller. The effect of this behavior is that not only the signaling traffic, but also the media traffic (voice, video etc) crosses the OBP/SBC. Without an OBP/SBC, the media traffic travels directly between the VoIP phones. Private OBP/SBCs are used along with [firewalls](#) to enable VoIP calls to and from a protected enterprise network. Public VoIP service providers use OBP/SBCs to allow the use of VoIP protocols from private networks with [internet](#) connections using [NAT](#).

PPPoE Point-to-Point Protocol over Ethernet is a network protocol for encapsulating PPP frames in Ethernet frames. It is used mainly with cable modem and DSL services.

PSTN Public Switched Telephone Network. The phone service we use for every ordinary phone call, or called POT (Plain Old Telephone), or circuit switched network.

RTCP Real-time Transport Control Protocol, defined in [RFC 3550](#), a sister protocol of the [Real-time Transport Protocol](#) (RTP), It partners RTP in the delivery and packaging of multimedia data, but does not transport any data itself. It is used periodically to transmit control packets to participants in a streaming multimedia session. The primary function of RTCP is to provide feedback on the quality of service being provided by RTP.

RTP Real-time Transport Protocol defines a standardized packet format for delivering audio and video over the Internet. It was developed by the Audio-Video Transport Working Group of the [IETF](#) and first published in 1996 as [RFC 1889](#)

SDP Session Description Protocol is a format for describing [streaming media](#) initialization parameters. It has been published by the [IETF](#) as [RFC 2327](#).

SIP Session Initiation Protocol, An IP telephony signaling protocol developed by the IETF (RFC3261). SIP is a text-based protocol suitable for integrated voice-data applications. SIP is designed for voice transmission and uses fewer resources and is considerably less complex than H.323. All Grandstream products are SIP based

STUN Simple Traversal of UDP over NATs is a [network protocol](#) allowing clients behind [NAT](#) (or multiple NATs) to find out its public address, the type of NAT it is behind and the internet side port associated by the NAT with a particular local port. This information is used to set up UDP communication between two hosts that are both behind NAT routers. The protocol is defined in [RFC 3489](#). STUN will usually work well with non-symmetric NAT routers.

TCP Transmission Control Protocol is one of the core protocols of the [Internet protocol suite](#). Using TCP, applications on networked hosts can create *connections* to one another, over which they can exchange data or [packets](#). The protocol guarantees reliable and in-order delivery of sender to receiver data.

TFTP Trivial File Transfer Protocol, is a very simple [file transfer protocol](#), with the functionality of a very basic form of [FTP](#); It uses [UDP](#) (port 69) as its [transport protocol](#).

UDP User Datagram Protocol (UDP) is one of the core protocols of the [Internet protocol suite](#). Using UDP, programs on networked computers can send short messages known as [datagrams](#) to one another. UDP does not provide the reliability and ordering guarantees that [TCP](#) does; datagrams may arrive out of order or go missing without notice. However, as a result, UDP is faster and more efficient for many lightweight or time-sensitive purposes.

VAD Voice Activity Detection or Voice Activity Detector is an algorithm used in [speech processing](#) wherein, the presence or absence of human speech is detected from the audio samples.

VLAN A virtual [LAN](#), known as a VLAN, is a logically-independent [network](#). Several VLANs can co-exist on a single physical [switch](#). It is usually refer to the [IEEE 802.1Q](#) tagging protocol.

VoIP Voice over the Internet. VoIP encompasses many protocols. All the protocols do some form of signaling of call capabilities and transport of voice data from one point to another. e.g.: SIP, H.323, etc.