



# Grandstream Networks, Inc.

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HT496

Dual FXS Port

Analog Telephone Adaptor



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## CONFIGURATION GUI INTERFACE EXAMPLES

HT496 User Manual

[http://www.grandstream.com/user\\_manuals/GUI/GUI\\_HT496.rar](http://www.grandstream.com/user_manuals/GUI/GUI_HT496.rar)

1. SCREENSHOT OF ADVANCED USER CONFIGURATION PAGE
2. SCREENSHOT OF AUDIBLE TONE CONFIGURATION PAGE
3. SCREENSHOT OF BASIC SETTINGS CONFIGURATION PAGE
4. SCREENSHOT OF FXS PORT 1 CONFIGURATION LOGIN PAGE
5. SCREENSHOT OF FXS PORT 2 CONFIGURATION PAGE
6. SCREENSHOT OF STATUS CONFIGURATION LOG IN PAGE

## WELCOME

Thank you for purchasing the Grandstream HT496, the affordable, feature rich Analog Telephone Adaptor designed for both the residential and business user.

HT496 is a next generation dual-port SIP IAD for Internet data, voice, and fax. The HT496 is a full feature voice and fax-over IP device that offers a high-level of integration including an integrated router, NAT, DHCP server, dual port FXS telephone gateway, market-leading sound quality, rich functionalities, and a compact and lightweight design. The HT496 is compact, works with any analog or cordless phone and fax machines and offers the simplicity of plug and dial.

**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.*

## SAFETY COMPLIANCES

The HT496 adaptor complies with FCC/CE and various safety standards. The HT496 power adaptor is compliant with UL standard. Only use the universal power adapter provided with the HT496 package. The manufacturer's warranty does not cover damages to the phone caused by unsupported power adaptors.

## WARRANTY

If you purchased your HT496 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.

- This document contains links to the HT496 GUI Interfaces. Please download these examples @ [http://www.grandstream.com/user\\_manuals/GUI/GUI\\_HT496.rar](http://www.grandstream.com/user_manuals/GUI/GUI_HT496.rar) for your reference.
- This document is subject to change without notice. The latest User Manual is available for download @ [http://www.grandstream.com/user\\_manuals/HT496\\_User\\_Manual.pdf](http://www.grandstream.com/user_manuals/HT496_User_Manual.pdf)
- *Reproduction or transmittal of any part, of this manual, in any form or by any means, electronic or print, for any purpose without the express written permission of Grandstream Networks, Inc. is not permitted.*

## INSTALLATION

### EQUIPMENT PACKAGING

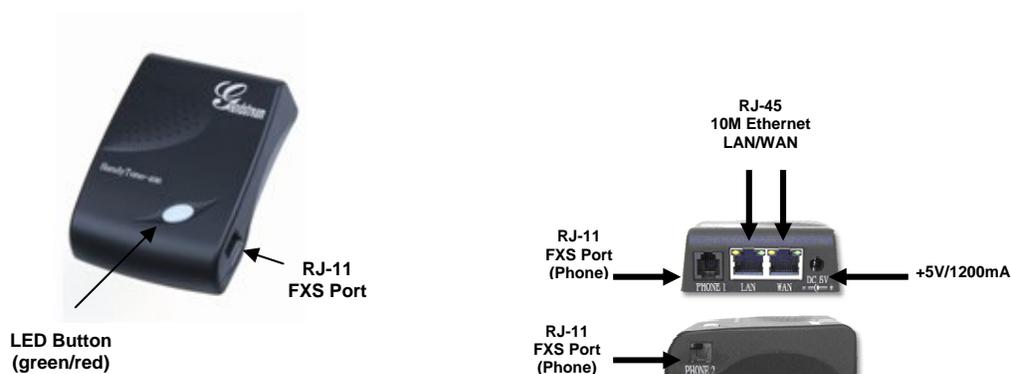
The HT496 ATA package contains:

- One HT496 Main Case
- One Universal Power Adaptor
- One Ethernet Cable

### CONNECTING YOUR ATA

The HT496 is easy to configure. The HT496 has two FXS ports. Each FXS port can have a separate SIP account. This is a key feature of HT496. Both ports can make calls concurrently.

**FIGURE 1: CONNECTING THE HT496**



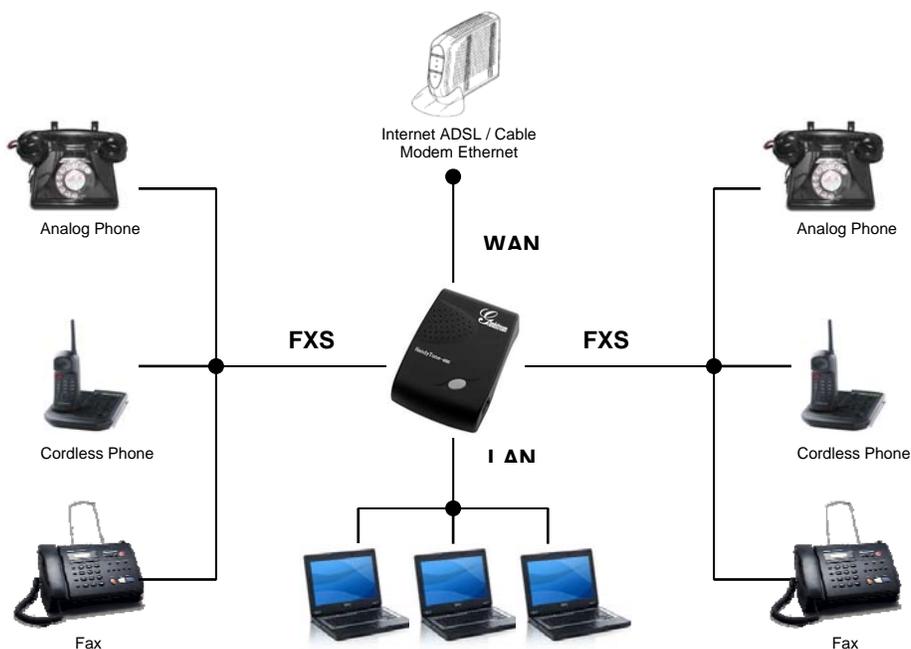
**Table 1: Definitions of the HT496 Connectors**

<b>Power Cable</b>	Power adapter connection
<b>WAN Port (RJ45)</b>	Connect to the internal LAN network or router.
<b>LAN Port (RJ45)</b>	Connect the LAN port with an Ethernet cable to your PC.
<b>PHONE1 (RJ-11)</b>	FXS port to be connected to analog phones / fax machines.
<b>PHONE2 (RJ-11)</b>	FXS port to be connected to analog phones / fax machines.

## FIVE EASY STEPS TO INSTALL THE HT496

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 HT496 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 HT496.
5. Insert the power adapter into the HT496 and connect it to a wall outlet.

**FIGURE 2: CONFIGURING THE HT496**



## PRODUCT OVERVIEW

The HT496 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 H496 is fully compatible with SIP industry standard and is interoperable with most other SIP compliant devices and software. It also supports comprehensive voice codecs.

### KEY FEATURES

Ethernet Ports	DHCP	FXS Port	PSTN Pass – through	Voice Mail Indicator	Voice Codec	Remote Configuration
2 RJ45 (LAN)	Server/ Client	2	No	Yes	iLBC, G.723, G.711, G.729, G.726, T.38	TFTP/HTTP

**TABLE 2: HT496 Technical Specifications**

<b>Lines/SIP Accounts</b>	2 lines / 2 SIP accounts
<b>Protocol Support</b>	SIP 2.0 (RFC 3261), TCP/UDP/IP, RTP/RTCP, HTTP, ARP/RARP, ICMP, DNS, DHCP, NTP, TFTP, PPPoE protocols
<b>Feature Keys</b>	1 button
<b>LAN/WAN Interface</b>	RJ45 10 Mbps
<b>Device Management</b>	Web interface or via secure (AES encrypted) central configuration file for mass deployment, Auto/manual provisioning system, Syslog support Support device configuration via built-in IVR, Web browser or central configuration file through TFTP or HTTP, Support Layer 2 (802.1Q, VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS) NAT-friendly remote software upgrade (via TFTP/HTTP) for deployed devices including behind firewall/NAT
<b>DHCP Server/Client</b>	Yes, NAT Router or Switched Mode
<b>Audio Features</b>	Advanced Digital Signal Processing (DSP), Adaptive jitter buffer control, Packet delay & loss concealment Support configurable Call Progress Tones Support volume amplification, Dynamic negotiation of codec and voice payload length Support for G.723,1 (5.3K/6.3K), G.729A, G.711 $\mu$ /A, G.726, and iLBC codecs In-band and out-of-band DTMF ((in audio, RFC2833, SIP INFO) Silence Suppression, VAD (voice activity detection), CNG (comfort noise generation), ANG (automatic gain control)
<b>Call Handling Features</b>	Caller ID display or block, Call waiting caller ID, Call waiting/flash, Call transfer, hold, forward, mute, 3-way conferencing
<b>Network and Provisioning</b>	Manual or dynamic host configuration protocol (DHCP) network setup; RTP and NAT support traversal via STUN
<b>Fax over IP</b>	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through (pending) Fax Datapump V.17, V.19, V.27ter, V.29 for T.38 fax relay
<b>Security</b>	DIGEST authentication and encryption using MD5 and MD5-sess
<b>Physical Design</b>	Stylish and compact design; small universal power supply, ideal for travel

**TABLE 3: HT496 Hardware Specifications**

<b>LAN Interface</b>	1 x RJ45 10 Mbps
<b>WAN Interface</b>	1 x RJ45 10 Mbps
<b>FXS phone port</b>	2 x FXS
<b>Button</b>	1
<b>LED</b>	Green and red / solid state & blinking state
<b>Universal Switching Power Adaptor</b>	<b>Input:</b> 100-240VAC 50-60 Hz <b>Output:</b> +5VDC, 1200mA <b>UL certified</b>
<b>Dimension</b>	70mm (W) x 130mm (D) x 27mm (H)
<b>Weight</b>	0.6lbs (0.3kg)
<b>Temperature</b>	40 - 130°F / 5 - 45°C
<b>Humidity</b>	10% - 90% (non-condensing)
<b>Compliance</b>	FCC/CE/C-Tick

## BASIC OPERATIONS

### GET FAMILIAR WITH VOICE PROMPT

HT496 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 the FXS port **1** only. Pick-up the handset and press the button on the HT496 or dial “\*\*\*” to use the IVR menu,

**TABLE 4: HT496 IVR Menu Definitions**

Menu	Voice Prompt	Options
<b>Main Menu</b>	“Enter a Menu Option”	Press “*” for the next menu option Press “#” to return to the main menu Enter 01-05, 07,12-17,47 or 99 menu options
<b>01</b>	“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.
<b>02</b>	“IP Address “ + IP address	The current WAN IP address is announced If using “ <i>Static IP Mode</i> ”, enter 12 digit new IP address.
<b>03</b>	“Subnet “ + IP address	Same as menu 02
<b>04</b>	“Gateway “ + IP address	Same as menu 02
<b>05</b>	“DNS Server “ + IP address	Same as menu 02
<b>07</b>	Preferred Vocoder	Press “9” to move to the next selection in the list: <ul style="list-style-type: none"> <li>• PCM U/A</li> <li>• G.723</li> <li>• iLBC</li> <li>• G.729</li> <li>• G.726</li> </ul>
<b>12</b>	WAN Port Web Access	Press “9” to toggle between <b>enable / disable</b>
<b>13</b>	Firmware Server IP Address	Announces current Firmware Server IP address. Enter 12 digit new IP address.
<b>14</b>	Configuration Server IP Address	Announces current Config Server Path IP address. Enter 12 digit new IP address.
<b>15</b>	Upgrade Protocol	Upgrade protocol for firmware and configuration update. Press “9” to toggle between <b>TFTP / HTTP</b>
<b>16</b>	Firmware Version	Firmware version information.
<b>17</b>	Firmware Upgrade	Firmware upgrade mode. Press “9” to toggle among the following three options: <ul style="list-style-type: none"> <li>- always check</li> <li>- check when pre/suffix changes</li> <li>- never upgrade</li> </ul>
<b>47</b>	“Direct IP Calling”	Enter a 12 digit IP address to make a direct IP call, after dial tone. (See “ <i>Make a Direct IP Call</i> ”.)
<b>99</b>	“RESET”	Press “9” to reboot the device; <b>or</b> Enter encoded MAC address to restore factory default setting (See “ <i>Restoring Factory Settings</i> ”)
	“Invalid Entry”	Automatically returns to main menu

### IVR Definition NOTES:

1. Once the button is pressed, it enters the voice prompt main menu. If the button is pressed again, while it is already in the voice prompt menu, it jumps to “Direct IP Call” option and a dial tone is prompted
2. “\*” shifts down to the next menu option  
“#” returns to the main menu  
“9” functions as the ENTER key in many cases to confirm an option
3. All entered digit sequences have known lengths
  - a. 2 digits for menu option and 12 digits for IP address.
  - b. For IP address, add 0 before the digits if the digits are *less than 3* (example: 192.168.0.26 should be key in like 192168**0000**26 – no period (.) is needed during input).
  - c. Once all of the digits are keyed in, the IP address will be processed.
4. 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 with VoIP devices (e.g. HT496 and another VoIP Device), to talk to each other in an ad hoc fashion without a SIP proxy. Direct IP calls are possible if:

- a) Both VoIP devices use a public IP address; or
- b) Both VoIP devices are on the same LAN using private IP addresses, or
- c) Both VoIP devices can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

To make a Direct IP Call, use either the handset or the speakerphone. Access the IVR using “\*\*\*\*”, the **button** on the HT496 or “**47**”. After dial tone, enter a 12-digit target IP address to place a call. Destination ports can be specified by using “\*4” (encoding for “:”) followed by the port number.

#### Examples:

- a) If the target IP address is 192.168.0.160, enter the 12 digit IP address (e.g. 1921680160) after the voice prompt followed by the “#” key or wait for 4 seconds. 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, enter the 12 digit IP address + \*45062 (e.g. 1921680160\*45062) after the voice prompt followed by the “#” key or wait for 4 seconds.

### CALL HOLD

Press the “flash” button to place a caller on hold. Press the “flash” button again to release call hold.

### CALL WAITING

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

The HT496 supports both **blind** and **attended** transfer:

1. **Blind Transfer:** Press “flash” button (or hook flash on older models), dial \*87, then dial the number to transfer call and press the “#” key (or wait 4 seconds) to complete transfer of active call.

### Expected outcomes:

- a) A quick confirmation tone (call waiting tone) followed by a dial tone. This indicates the transfer is successful. Hang up or place another call.
- b) A quick busy tone followed by a restored call (on supported platforms only). The busy tone indicates the transfer failed.
- c) Continuous busy tone. The phone call 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.

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

### 2. Attended Transfer:

Assuming that call party A and B are in conversation. A wants to *Attend Transfer B* to C:

- (1) A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone
- (2) A then dial C’s number then # (or wait for 4 seconds).
- (3) If C answers the call, A and C are in conversation. Then A can hang up to complete transfer.
- (4) If C does not answer the call, A can press “flash” back to talk to B.

**NOTE:** If Attended Transfer fails, and party A hangs up, the HandTone496 will ring party A to remind A that party B is still on the line. Party A can pick up the phone to resume conversation with party B.

## 3-WAY CONFERENCING

HT496 supports Star Code Style or Bellcore Style 3-way Conference.

### Star Code Style 3-way Conference

Assuming that call party A and B are in conversation. A (HT496) wants to bring C in a conference:

1. A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
2. A dials \*23 then 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.

### Bellcore Style 3-way Conference

Bellcore style 3-way conference is also supported. To use this method, the user must enable “Use Bell-style 3-way Conference” in FXS1 or FXS2 web configuration.

Assuming that call party A and B are in conversation. A (HT496) wants to bring C in a 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 and C into 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 drops out.

## CALL FEATURES

The HT496 supports all the traditional and advanced telephony features.

**TABLE 5: HT496 Call Feature Definitions**

Key	Call Features
*23	<b>3-way conference</b>
*30	<b>Block Caller ID</b> (for all subsequent calls)
*31	<b>Send Caller ID</b> (for all subsequent calls)
*50	<b>Disable Call Waiting</b> (for all subsequent calls)
*51	<b>Enable Call Waiting</b> (for all subsequent calls)
*67	<b>Block Caller ID</b> (per call)
*70	<b>Disable Call Waiting</b> (per call)
*71	<b>Enable Call Waiting</b> (per call)
*72	<b>Unconditional Call Forward.</b> Dial “*72”, wait for dial tone. Dial the forward number and “#” for a dial tone, then hang up.
*73	<b>Cancel Unconditional Call Forward.</b> To cancel “Unconditional Call Forward”, dial “*73”, wait for dial tone, then hang up.
*82	<b>Send Caller ID</b> (per call)
*87	<b>Blind Transfer</b>
*90	<b>Busy Call Forward.</b> Dial “*90”, wait for dial tone. Then dial the forward number and “#” for a dial tone, then hang up.
*91	<b>Cancel Busy Call Forward.</b> To cancel “Busy Call Forward”, dial “*91”, wait for dial tone, then hang up.
*92	<b>Delayed Call Forward.</b> Dial “*92”, wait for dial tone. Then dial the forward number and “#” for a dial tone, then hang up.
*93	<b>Cancel Delayed Call Forward.</b> To cancel Delayed Call Forward, dial “*93”, wait for dial tone, then hang up.
<b>Flash/Hook</b>	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.

## T.38 FAX

HT496 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.

## LED LIGHT PATTERN INDICATION

Following tables show the LED light pattern indication. The LED shows PHONE1 status only.

**TABLE 6: HT496 LED Definitions**

<b>RED LED</b> always indicates not normal status	
<b>Button flashes every 2 seconds.</b> (if DHCP is configured)	DHCP Failed or WAN No Cable
<b>Button flashes every 2 seconds.</b> (if SIP server is configured)	HT496 fails to register
<b>Button flashes every 2 seconds.</b>	Firmware Upgrading
<b>Red light steady.</b>	Device Malfunctions

<b>GREEN LED</b> mostly indicates normal working status	
<b>Button flashes every 2 seconds.</b>	Message Waiting Indication
<b>Button flashes at 1/10 second.</b>	RINGING
<b>Button flashes every second.</b>	RINGING INTERVAL
<b>Green light steady.</b>	In Conversation

## CONFIGURATION GUIDE

### CONFIGURING HT496 THROUGH VOICE PROMPT

#### DHCP MODE

Select voice menu option 01 to enable HT496 to use DHCP.

#### STATIC IP MODE

Select voice menu option 01 to enable HT496 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 HT496 WITH WEB BROWSER

HT496 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 HT496 through a Web browser such as Microsoft's IE and AOL's Netscape.

### ACCESS THE WEB CONFIGURATION MENU

The HT496 HTML (Web GUI) configuration menu is accessed either from the 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 HT496 using voice prompt menu option 02.
3. Access the HT496 Web Configuration page by the following URI via WAN port:  
<http://HandyTone-IP-Address> (the HandyTone-IP-Address is the WAN IP address for the HT496).

**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 SETTING*” 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.

**TABLE 7: HT496 Basic Configuration Settings Definitions**

<b>End User Password</b>	Password to access the Web Configuration Menu. This field is case sensitive with a maximum length of 25 characters.
<b>Web Port</b>	By default, HTTP uses port 80. This field is for customizable web port.
<b>IP Address</b>	There are two modes to operate the HT496: <b>DHCP mode:</b> all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The HT496 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 HT496 will establish a PPPoE session if any of the PPPoE fields is set. <b>Static IP mode:</b> 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.
<b>DHCP hostname</b>	This option specifies the name of the client. This field is optional but may be required by some Internet Service Providers. Default is blank.
<b>DHCP domain</b>	This option specifies the domain name that client should use when resolving hostnames via the Domain Name System. Default is blank.
<b>DHCP vendor class ID</b>	Used by clients and servers to exchange vendor-specific information. Default is blank.
<b>PPPoE account ID</b>	PPPoE username. Necessary if ISP requires you to use a PPPoE (Point to Point Protocol over Ethernet) connection.
<b>PPPoE password</b>	PPPoE account password.
<b>PPPoE Service Name</b>	This field is optional. If your ISP uses a service name for the PPPoE connection, enter the service name here. Default is blank.
<b>Time Zone</b>	Controls how the date/time displays according to the specified time zone.

<b>Daylight Savings Time</b>	<p>Controls whether displayed time is daylight savings time or not. If set to “Yes” and the Optional Rule is empty, then the displayed time will be 1 hour ahead of normal time.</p> <ul style="list-style-type: none"> <li>The “Automatic Daylight Saving Time Rule” has the following syntax: <ul style="list-style-type: none"> <li><i>start-time;end-time;saving</i>. Both start-time and end-time have the same syntax: <i>month,day,weekday,hour,minute</i></li> <li>month: 1,2,3,...,12 (for Jan, Feb, ..., Dec)</li> <li>day: [+ -]1,2,3,...,31</li> <li>weekday: 1, 2, 3, ..., 7 (for Mon, Tue, ..., Sun), or 0 indicating daylight savings rule is based on the day of the month.</li> <li>hour: hour (0-23),</li> <li>minute: minute (0-59)</li> </ul> </li> <li>If “weekday” is 0, it means the date to start or end daylight savings is the given date. The “day” value must not be negative.</li> <li>If “weekday” is not zero and “day” is positive, then daylight saving starts on the first “day” of the week (1st Sunday, 3rd Tuesday etc).</li> <li>If “weekday” is not zero and “day” is negative, then daylight savings starts on the last “day” of the week (last Sunday, 3rd last Tuesday etc).</li> <li>Daylight savings is in minutes. If preceded by a (-), then subtract the number of minutes.</li> </ul> <p>The default value for “Automatic Daylight Saving Time Rule” is US time: “04,01,7,02,00;10,-1,7,02,00;60”.</p> <p><b>Example:</b> US/Canada, where daylight savings is applicable: 04,01,7,02,00;10,-1,7,02,00;60 Daylight savings starts from the first Sunday of April at 2AM and ends the last Sunday of October at 2AM.).</p>
<b>Device Mode</b>	Enables use of the device as a router or a bridge.
<b>WAN side HTTP/Telnet Access</b>	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 <b>No</b> .
<b>Reply to ICMP on WAN port</b>	If set to “Yes”, the HT496 will respond to the PING command from other computers, but it also is vulnerable to the DOS attack. Default is <b>No</b> .
<b>Cloned WAN MAC Address:</b>	Set a specific MAC address. Set in <b>Hex</b> format.
<b>LAN Subnet Mask</b>	Sets the LAN subnet mask. Default value is 255.255.255.0.
<b>LAN DHCP Base IP:</b>	Base IP for the LAN port which functions as a gateway for the subnet. Default value is 192.168.2.1.
<b>DHCP IP Lease Time:</b>	Value is set in number of hours. Default value is 120hr (5 Days.) The amount of time IP address are assigned to LAN clients.
<b>DMZ IP:</b>	Forward all WAN IP traffic to a specific IP address if no matching port is used by HT496 or defined in port forwarding.
<b>Port Forwarding:</b>	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: HT496 Device Status Page Definitions**

<b>MAC Address</b>	The device ID in HEX format. This is needed for ISP troubleshooting.
<b>WAN IP Address</b>	Shows WAN IP address of HT496.
<b>Product Model</b>	Contains the product model info.
<b>Software Version</b>	Program: This is the main software release. Boot and Loader are seldom changed.
<b>System Up Time</b>	Shows system up time since the last reboot.
<b>Registered</b>	Indicates whether the HT496 is registered to the service provider's server.
<b>PPPoE Link Up</b>	Indicates whether the PPPoE connection is up if the HT496 is connected to DSL modem.
<b>NAT</b>	Indicates the type of NAT the HT496 is connected to via its WAN port. Based on STUN protocol.

## 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. HT496 each FXS SIP account has its own configuration page. Their configurations are identical.

**TABLE 9: HT496 Advanced Configuration Page Definitions**

<b>Admin Password</b>	This contains the password to access the Advanced Web Configuration page. This field is case sensitive.
<b>Home NPA</b>	Local area code for North American Dial Plan.
<b>Layer 3 QoS</b>	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.
<b>Layer 2 QoS</b>	This setting includes two fields. The 802.1Q/VLAN Tag contains the value used for layer 2 VLAN tag. Default setting is blank. The 802.1p priority value contains the value of the priority value.
<b>STUN Server</b>	STUN server's IP or FQDN
<b>Keep-alive interval</b>	This parameter specifies how often the HT496 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 10 seconds.
<b>Use NAT IP:</b>	NAT IP address used in SIP/SDP message. Default is blank.
<b>Firmware Upgrade and Provisioning</b>	Default method is HTTP. Firmware upgrade may take up to 20 minutes depending on network environment. Do not interrupt the firmware upgrading process.
<b>Firmware Server Path</b>	IP address or domain name of firmware server.
<b>Config Server Path</b>	IP address or domain name of configuration server.
<b>Firmware File Prefix</b>	Default is blank. If configured, HT496 will request firmware file with the prefix. This setting is useful for ITSPs. End user should keep it blank.
<b>Firmware File Postfix</b>	Default is blank. End user should keep it blank.
<b>Config File Prefix</b>	Default is blank. End user should keep it blank.
<b>Config File Postfix</b>	Default is blank. End user should keep it blank.
<b>Automatic Upgrade</b>	Default is "NO". When set to No, HT496 will only do upgrade once at boot up. "Check every day ", "Check every week", "Always check for New Firmware at Boot up." "Check New Firmware only when F/W pre/suffix changes"
<b>Firmware Key</b>	Used for firmware encryption. Should be 32 digit in Hexadecimal Representation. End user should keep it blank.
<b>On-hook Threshold</b>	The time hook-flash is pressed that causes the device to on-hook. Default is 800ms.
<b>FXS Impedance</b>	Selects the impedance of the analog telephone connected to the RJ-11 FXS port. The following information may be useful for end user configuration: 600Ohm – North America 270+750Ohm    150nF – Most of Europe 220+820Ohm    120nF – Australia, New Zealand 220+820Ohm    115nF – Austria, Bulgaria, Germany, Slovakia, South Africa 370+620Ohm    310nF – UK, India

<b>Caller ID Scheme</b>	<p>Select the Caller ID Scheme to suit the standard of different area.</p> <ul style="list-style-type: none"> <li>• Bellcore (North America)</li> <li>• CID-Canada</li> <li>• DTMF-Brazil</li> <li>• DTMF-Sewden</li> <li>• DTMF (Denmark)</li> <li>• ETSI-DTMF (Finland, Sweden)</li> <li>• ETSI-FSK (France, Germany, Norway, Taiwan, UK-CCA)</li> </ul>
<b>On-hook Voltage</b>	<p>Select the on-hook voltage to suit different area or PBX. Here is the voltage the FXS port will supply to connected analog Phone or PBX.</p>
<b>Polarity Reversal</b>	<p>Default is No. If set to "Yes", polarity will be reversed upon call establishment and termination.</p>
<b>NTP server</b>	<p>URI or IP address of the NTP (Network Time Protocol) server. Used by the HT496 to display the current date/time.</p>
<b>Syslog Server</b>	<p>The IP address or URL of System log server. This feature is especially useful for ITSP (Internet Telephone Service Provider)</p>
<b>Syslog Level</b>	<p>Select the ATA 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"> <li>1. product model/version on boot up (INFO level)</li> <li>2. NAT related info (INFO level)</li> <li>3. Sent or received SIP message (DEBUG level)</li> <li>4. SIP message summary (INFO level)</li> <li>5. inbound and outbound calls (INFO level)</li> <li>6. registration status change (INFO level)</li> <li>7. negotiated codec (INFO level)</li> <li>8. Ethernet link up (INFO level)</li> <li>9. SLIC chip exception (WARNING and ERROR levels)</li> <li>10. memory exception (ERROR level)</li> </ol> <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  <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: HT496 Individual Account Settings Definitions**

<b>SIP Server</b>	This field contains the URI string or the IP address (and port, if different from 5060) of the SIP proxy server. e.g., the following are some valid examples: sip.my-voip-provider.com, or sip:my-company-sip-server.com, or 192.168.1.200:5066
<b>Outbound Proxy</b>	This field contains the URI string or the IP address (and port, if different from 5060) of the outbound proxy. If there is no outbound proxy, this field SHOULD be blank. If not blank, all outgoing requests are sent to this outbound proxy.
<b>SIP User ID</b>	This field contains the user part of the SIP address for this phone. e.g., if the SIP address is <b>sip:my_user_id@my_provider.com</b> , then the SIP User ID is: <b>my_user_id</b> . Do NOT include the preceding "sip:" scheme or the host portion of the SIP address in this field.
<b>Authenticate ID</b>	SIP service subscriber's Authenticate ID. Auth ID can be identical to the SIP User ID.
<b>Authenticate Password</b>	SIP service subscriber's account password.
<b>Name</b>	SIP service subscriber's name for Caller ID display.
<b>Use DNS SRV</b>	Default is No. If set to Yes the client will use DNS SRV for server lookup.
<b>User ID is Phone Number</b>	If the HT496 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.
<b>SIP Registration</b>	Controls whether the HT496 needs to send REGISTER messages to the proxy server. The default setting is Yes.
<b>Unregister on Reboot</b>	Default is No. If set to Yes, the SIP user's registration information will be cleared on reboot.
<b>Register Expiration</b>	This parameter allows the user to specify the time frequency (in minutes) the HT496 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).
<b>Local SIP port</b>	Defines the local SIP port the HT496 will listen and transmit. The default value for FXS port 1 is 5060. The default value for FXS port 2 is 5062.
<b>Local RTP port</b>	Defines the local RTP-RTCP port pair the HT496 will listen and transmit. It is the base RTP port for channel 0. When configured, <b>channel 0</b> uses this port _value for RTP and the port_value+1 for its RTCP; <b>channel 1</b> 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 5008.
<b>Use Random Port</b>	This parameter forces the random generation of both the local SIP and RTP ports when set to Yes. This is usually necessary when multiple HT496 are behind the same NAT.
<b>SIP Registration Failure Retry Wait Time</b>	In case sip server fail from some reason, the HT496 will try to re-register not according to standard SIP timers, but according to preconfigured interval. Recommended to leave default value 20sec. Designed by special request of big service providers.
<b>DTMF Payload Type</b>	This parameter sets the payload type for DTMF using RFC2833.
<b>Send DTMF</b>	This parameter specifies the mechanism to transmit DTMF digit. There are 3 modes supported: <b>1)</b> in audio which means DTMF is combined in audio signal (not very reliable with low-bit-rate codec), <b>2)</b> via RTP (RFC2833), or <b>3)</b> via SIP INFO. Multiple selections of DTMF method are supported.
<b>Send Flash Event</b>	Default is No. If set to yes, flash will be sent as DTMF event.

<b>Enable Call Features</b>	Default is Yes. Advanced call features and feature codes functions are supported locally.
<b>Use Bell-style 3-way Conference</b>	If this parameter is set to “Yes”, user will be able to make Bellcore style 3-way conference. *23 will be disabled.
<b>Off-hook Auto-Dial</b>	This parameter automatically configures and dials User ID or extension number upon off-hook. <u>Only the user part</u> of a SIP address needs is entered here. The HT496 will automatically append the “@” and the host portion of the corresponding SIP address.
<b>Proxy-Require</b>	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
<b>Disable Call Waiting</b>	Default is No. User can use * code to use this feature per call basis.
<b>Disable Call Waiting Caller ID (CWCID)</b>	Default is No. If set to YES, user will not be able to see CWCID information. If set to No, also requires support of this feature by analog phone connected to FXS port.
<b>NAT Traversal (STUN)</b>	<p>This parameter defines whether the HT496 NAT traversal mechanism is activated or not. If activated (by choosing “Yes”) and a STUN server is also specified, then the HT496 performs according to the STUN client specification. Under this mode, the embedded STUN client will detect if and what type of firewall/NAT is used. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the HT496 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 HT496 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>
<b>No Key Entry Timeout</b>	Default is 4 seconds. While dealing, the number will considered as completed if during this timeout no additional entry has been done.
<b>Preferred Vocoder</b>	<p>The HT496 supports up to 6 different Vocoder types including G.711 A-/U-law , G.723.1, G.726, G.729A/B, iLBC.</p> <p>Users can configure Vocoders in a preference list included with the same preference order in SDP message. The first Vocoder in this list is the appropriate option in “Choice 1”. The last Vocoder in this list is the appropriate option in “Choice 6”.</p>
<b>Voice Frames per TX</b>	<p>Number of voice frames transmitted in a single packet. User should be aware of the requested packet time (used in SDP message). This parameter is associated with the first vocoder in the Vocoder Preference List or the actual payload type negotiated between the 2 conversation parties at run time.</p> <p><b>e.g.</b> if the first vocoder is configured as G.723 and the “Voice Frames per TX” is set to 2, then the “ptime” value in the SDP message of an INVITE request will be 60ms because each G.723 voice frame contains 30ms of audio.</p> <p>If the configured voice frames per TX exceeds the maximum allowed value, the HT496 will use and save the maximum allowed value for the corresponding first vocoder choice. The maximum value for PCM is 10(x10ms) frames:</p> <ul style="list-style-type: none"> <li>• for G.726, it is 20 (x10ms) frames; .</li> <li>• for G.723, it is 32 (x30ms) frames;</li> <li>• for G.729/G.728, 64 (x10ms) and 64 (x2.5ms) frames respectively.</li> </ul>
<b>G723 Rate:</b>	Default is 6.3kbps rate. Defines the encoding rate for G723 vocoder.
<b>iLBC frame size:</b>	Sets the iLBC frame size in 20ms or 30ms
<b>iLBC payload type:</b>	Defines payload time for iLBC. Default value is 97. Valid range: 96 - 127.
<b>Silence Suppression</b>	This controls the silence suppression/VAD feature of G.723. If set to “Yes”, a small quantity of VAD packets (instead of audio packets) will be sent during the a period of silence. If set to “No”, feature is disabled.
<b>Fax Mode</b>	T.38 (Auto Detect) FoIP (fax over IP) by default, or Pass-Through ( <u>must use codec PCMU/PCMA</u> )

<b>Early Dial</b>	<p>This parameter controls whether the early INVITE each time a key is pressed. If set to "Yes", an INVITE is sent for each dialed number. If set to NO, INVITE is sent after the "(Re-)Dial" button is pressed or after 5 seconds have elapsed.</p> <p>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 is rejected.</p> <p><u>This feature is NOT designed to work with and should NOT be enabled for direct IP-to-IP calling.</u></p>
<b>Allow incoming SIP messages from SIP proxy only</b>	Default is set to "No". If set to "YES", each incoming call will be checked first if it arrived from authorized proxy. Useful for security installation to prevent incoming calls from not authorized sources.
<b>Allow outgoing call without Registration</b>	Default is set to "No". If set to "YES", any outgoing call will be generated to the address specified in SIP Server field. Generation outgoing call in this case does not requires registration with any SIP Server.
<b>Dial Plan Prefix</b>	Sets the prefix added to each dialed number
<b>Use # as Dial Key</b>	If set to "Yes", the "#" key as the "Send" (or "Dial") key. If set to "No", "#" key is part of the dial string.
<b>Subscribe for MWI:</b>	Default is No. When set to "Yes" a SUBSCRIBE for Message Waiting Indication will be sent periodically.
<b>Send Anonymous</b>	Blocks Caller ID display if set to "Yes".
<b>Lock keypad update</b>	If this parameter is set to "Yes", the configuration update via keypad is disabled.
<b>Refer-To Uses Target Contact</b>	Used for Attended transfer Feature. The "Refer-To" header uses the transferred target's "Contact" header information.
<b>Special Feature</b>	Default is Standard. Choose this selection to meet special requirements from Soft Switch vendors.
<b>Volume Amplification</b>	Adjusts handset volume. RX is for receiving volume, TX is for transmission volume. Default values are 0dB for both parameters. +6dB generates the highest volume and -6dB generates the lowest volume.

**TABLE 11: HT496 AUDIBLE TONES SETTINGS PAGE DEFINITIONS**

<b>Audible Tones</b>	<p>Using these settings, user can configure ring or tone frequencies according to their preference. By default they are set to North American frequencies.</p> <p>Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. <b>ON</b> is the period of ringing ("On time" in 'ms') while <b>OFF</b> is the period of silence.</p> <p>To set a continuous ring, OFF should be zero. Otherwise, it will ring ON ms and a pause of OFF ms and then repeat the pattern.</p>
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## SAVING THE CONFIGURATION CHANGES

After making a change, click the “Update” button in the Configuration page. The HT496 will display the following screen to confirming changes. Reboot or power cycle the HT496 to enable the changes.

**FIGURE 4: SCREENSHOT OF SAVE CONFIGURATION PAGE**



## REBOOTING THE HT496 FROM REMOTE

The administrator of the HT496 can remotely reboot the HT496 by clicking the “Reboot” button at the bottom of the configuration page. When finished, re-login to the HT496 after waiting about 30 seconds.

## CONFIGURATION THROUGH A CENTRAL SERVER

Grandstream HandyTone ATAs can be configured automatically from a central provisioning system.

When HandyTone ATA boot up, it will send TFTP or HTTP request to download configuration file, “cfg000b82xxxxx”, where “000b82xxxxx” is the MAC address of the HandyTone ATA.

The configuration files can be downloaded via TFTP or HTTP from the central server. A service provider or an enterprise with large deployment of HandyTone ATA can easily manage the configuration and service provisioning of individual devices remotely from a central server.

Grandstream provides a licensed provisioning system called GAPSLITE that supports automated configuration of HandyTone ATA. GAPSLITE (Grandstream Automated Provisioning System) uses enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual HandyTone ATA for firmware upgrade, remote reboot, etc.

Grandstream provide GAPS (Grandstream Automated Provisioning System) service to VoIP service providers. It could be either simple redirection or with certain special provisioning settings. Initially upon booting 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/1.0.3.86

e.g. 168.75.215.190

#### 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://grandstream.com/y-firmware.htm>. Currently, the TFTP server, your HT496 can be upgraded from has an IP address 168.75.215.190. 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 ATA’s SRAM. During this stage, the HT ATA’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 ATA 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 around 20 seconds if performed on a LAN. We recommend that firmware upgrades be completed 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>. The latest official release is available @ <http://www.grandstream.com/y-firmware.html>.
- User can also choose to download the free HTTP server from <http://httpd.apache.org/> or use Microsoft IIS web server.

#### Directions to download a free 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 HT496 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

## CONFIGURATION FILE DOWNLOAD

The HT496 can be configured via Web GUI as well as through the 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 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.

## 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.

### 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 "00022228200333395".

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. By default, the HT496 WAN side HTTP access is disabled.
3. After a factory reset, the device's web configuration page accessible only from its LAN port.