User Manual

HandyTone-486

Analog Telephone Adaptor

For Firmware Version 1.1.0.37

Grandstream Networks, Inc.

www.grandstream.com

Table of Contents

CONFIGURATION GUI INTERFACE EXAMPLES
HT–486 User Manual
(http://www.grandstream.com/support/ht_series/ht486/documents/ht486_gui.zip)

1. SCREENSHOT OF LOGIN PAGE
2. SCREENSHOT OF STATUS PAGE
3. SCREENSHOT OF BASIC SETTINGS CONFIGURATION PAGE
4. SCREENSHOT OF ADVANCED SETTING1 CONFIGURATION PAGE
5. SCREENSHOT OF ADVANCED SETTING2 CONFIGURATION PAGE
6. SCREENSHOT OF SAVED CONFIGURATION CHANGES
7. SCREENSHOT OF REBOOT PAGE

1 Welcome

Congratulations on becoming an owner of HandyTone-486. You made an excellent choice and we hope you will enjoy all its capabilities.

Grandstream's award-wining HandyTone-486 is an all-in-one VoIP integrated access device that features superb audio quality, rich functionalities, high level of integration, compactness and ultra-affordability. The HandyTone-486 is fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market.

Grandstream HandyTone-486 has been awarded the Best of Show product in 2004 Internet Telephony Conference and Expo.

This document is subject to changes without notice. The latest electronic version of this user manual can be downloaded from the following location:

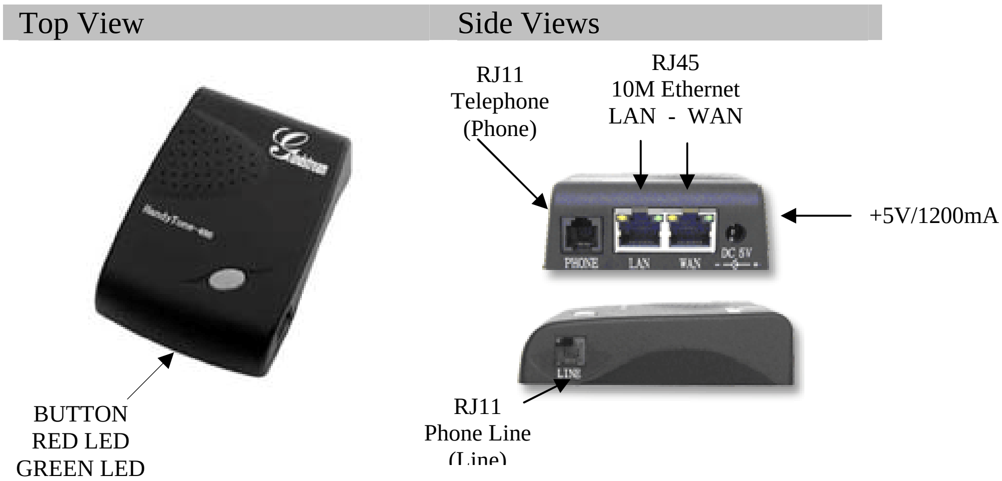
http://www.grandstream.com/support/ht_series/ht486/documents/ht486_usermanual_english.pdf

2 Installation

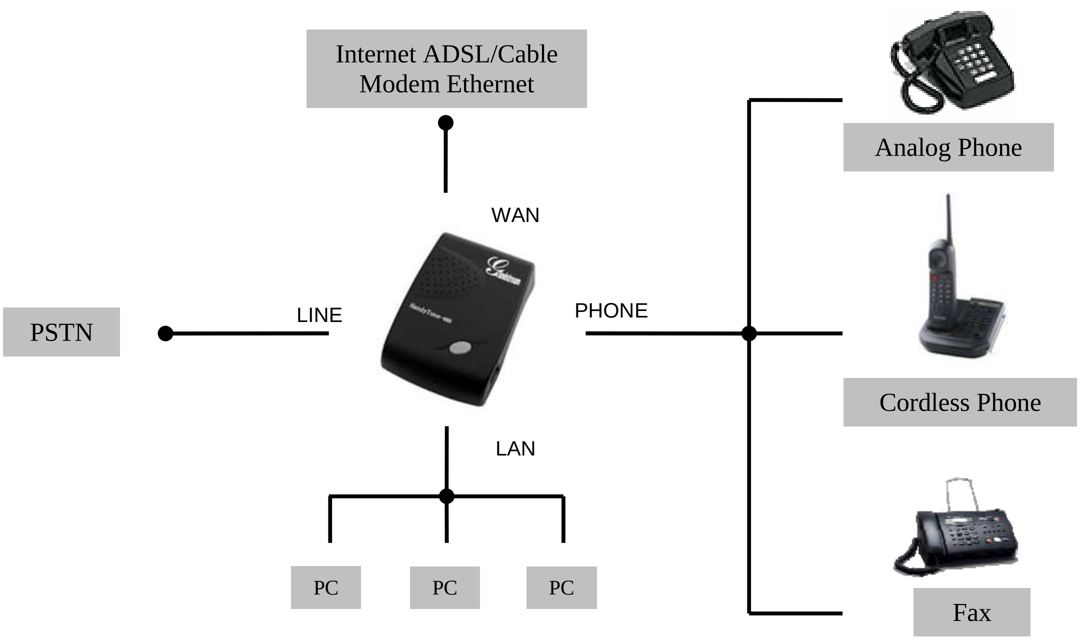
HandyTone-486 Analog Telephone Adaptor is an all-in-one VoIP integrated device designed to be a total solution for networks providing VoIP services.

The HandyTone-486 VoIP functionalities are available via a regular analog telephone.

The following photo illustrates the appearance of a HandyTone-486.



Interconnection Diagram of the HandyTone-486:



Following are the steps to install a HandyTone-486:

1. Connect a standard touch-tone analog telephone (or fax machine) to PHONE port.
2. Connect a PSTN telephone line to LINE port (optional).
3. Insert the Ethernet cable into the WAN port of HandyTone-486 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 HandyTone-486.
5. Insert the power adapter into the HandyTone-486 and connect it to a wall outlet.

Please follow the instructions in section 6.2.1 to configure the HandyTone-486.

3 What is Included in the Package

The HandyTone-486 package contains:

1) One HandyTone-486
2) One universal power adaptor
3) One Ethernet cable

3.1 Safety Compliances

The HandyTone-486 is compliant with various safety standards including FCC/CE and C-tick. Its power adaptor is compliant with UL standard. The HandyTone-486 should only operate with the universal power adaptor provided in the package.

3.2 Warranty

Grandstream has a reseller agreement with our reseller customer. End users should contact the company from whom you purchased the product 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.

Grandstream reserves the right to remedy warranty policy without prior notification.

Warning: Please do not attempt to use a different power adaptor. Using other power adaptor may damage the HandyTone-486 and will void the manufacturer warranty.

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.

4 Product Overview

4.1 Key Features

- Supports SIP 2.0(RFC 3261), TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP/RARP, DNS, DHCP (both client and server), NTP, PPPoE, STUN, TFTP, etc.
- Built-in router, NAT, Gateway and DMZ port forwarding. Can also be configured to function as a two Ethernet ports bridge (NAT function is disabled)
- Device bridge mode support
- Powerful digital signal processing (DSP) to ensure superb audio quality; advanced adaptive jitter control and packet loss concealment technology
- Support various codecs including G.711 (PCM a-law and u-law), G.723.1 (5.3K/6.3K), G.726 (32K), as well as G.729A and iLBC
- Support Caller ID/name display or block, Call waiting caller ID, Hold, Call Waiting/Flash, Call Transfer, 3-way conference (on Rev. 2.0), Call Forward, in-band and out-of-band DTMF, etc.
- Support fax pass through (via PCMU or PCMA) and T.38 FoIP (Fax over IP)
- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168), and AGC (Automatic Gain Control)
- Support standard encryption and authentication (DIGEST using MD5 and MD5-sess)
- Support for Layer 2 (802.1Q VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
- Support automated NAT traversal without manual manipulation of firewall/NAT
- Support device configuration via built-in IVR, Web browser or encrypted configuration files through TFTP or HTTP server
- Support firmware upgrade via TFTP or HTTP
- Support PSTN pass through (on Rev.2.0)
- Support SIP Session Timer
- Support Syslog (on Rev.2.0)
- Support volume amplification
- Support configurable Call Progress Tones
- Ultra compact (wallet size) and lightweight design, great companion for travelers
- Compact, lightweight Universal Power adapter

4.2 Hardware Specification

The table below lists the hardware specification of HandyTone-486.

Model	**HandyTone-486**
LAN interface	1xRJ45 10Base-T
WAN interface	1xRJ45 10Base-T
FXS telephone port	1xFXS
PSTN port	1x PSTN pass through or life line port
Button	1
LED	Green and red color
Universal Switching Power Adaptor	Input: 100-240VAC 50-60 Hz Output: +5VDC, 1200mA, **UL** certified
Dimension	70mm (W) 130mm (D) 27mm (H)
Weight	0.6lbs (0.3kg)
Temperature	40 - 130^{o}F 5 – 45^{o}C
Humidity	10% - 90% (non-condensing)
Compliance	

NOTE:

- HandyTone-486 has two hardware revisions. This information can be found on the label at the bottom of the device. The difference between HandyTone-486 Rev.1.0 and HandyTone-486 Rev.2.0 is that a HandyTone-486 Rev.2.0 line port can function as PSTN pass through while a HandyTone-486 (Rev.1.0, old model, no longer shipped) line port is just a life line port and will bridge to PSTN only when the device is out of power.

5 Basic Operations

5.1 Get Familiar with Key Pad and Voice Prompt

HandyTone-486 stores a voice prompt menu (Interactive Voice Response or IVR) for quick browsing and simple configuration. To enter this voice prompt menu, simply press the button on the HandyTone-486 or pick up the phone and dial "***". The following table shows how to use the voice prompt menu to configure the device.

Menu	Voice Prompt	User's Options
Main Menu	"Enter a Menu Option"	Enter "*" for the next menu option Enter "#" to return to the main menu Enter 01 – 06, 47, 86 or 99 Menu option
01	"DHCP Mode", or "Static IP Mode"	Enter '9' to toggle the selection *If user selects "Static IP Mode", user need configure all the IP address information through menu 02 to 05. If user selects "Dynamic IP Mode", the device will retrieve all IP address information from DHCP server automatically when user reboots the device.*
02	"IP Address " + IP address	The current WAN IP address is announced Enter 12-digit new IP address if in Static IP Mode.
03	"Subnet " + IP address	Same as Menu option 02
04	"Gateway " + IP address	Same as Menu option 02
05	"DNS Server " + IP address	Same as Menu option 02
07	Preferred Vocoder	Enter "9" to go to the next selection in the list: - PCM U - PCM A - G-723 - G-729 - iLBC - G-726
12	WAN Port Web Access	Enter "9" to toggle between: - enable - disable
13	Firmware Server IP Address	The current Firmware Server IP address is announced. Enter 12 digit new IP address.
14	Configuration Server IP Address	The current Config Server Path IP address is announced. Enter 12 digit new IP address.
15	Upgrade Protocol	Upgrade protocol for firmware and configuration update. Enter "9" to toggle between: - TFTP - HTTP

16	Firmware Version	Firmware version information.
17	Firmware Upgrade	Firmware upgrade mode. Enter “9” to rotate among the following three options: - always check - check when pre/suffix changes - never upgrade
47	“Direct IP Calling”	When entered, user will be prompted a dial tone, dial a 12-digit IP address to make a direct IP call. (For details, see “4.2.2 Make a Direct IP Call”.)
99	“RESET”	Enter “9” to reboot the device; or Enter MAC address to restore factory default setting (For details, see section 8.)
	“Invalid Entry”	Automatically returns to Main Menu

IVR supports error reporting when the following problems occur. User will hear silence when picking up the handset. After pressing ***, user will hear one or more error codes listed below. User may hear one or more error codes depending on errors detected such as E104E103E. Upon hearing error code, user can press # to get into the IVR main menu.

E101E	Ethernet link down
E102E	No IP address obtained (DHCP or PPPoE mode)
E103E	Device is not registered to SIP server
E104E	Provisioning in action
E105E	No STUN responses

NOTES:

- Once the LED button is pressed, it enters voice prompt main menu. If the button is pressed again while it is already in the voice prompt menu state, it jumps to “Direct IP Calling” option and dial tone plays in this state
- “*” 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
- All entered digit sequences have known lengths - 2 digits for menu option and 12 digits for IP address. Once all digits are accumulated, the device will automatically process them
- For IP address input, omit the dot and enter the digits directly, add 0 for those octets with less than three digits. e.g.: IP: 192.168.1.10, key in: 192168001010
- Key entry cannot be deleted but the phone may prompt error once it is detected

5.2 Make Phone Calls

5.2.1 Calling phone or extension numbers

There are currently two methods to make an extension number call:

a) Dial the numbers directly and wait for 4 seconds (Default "No Key Entry Timeout"). Or

b) Dial the numbers directly, and press # (assuming that "use #" as dial key is selected in web configuration).

Examples:

To dial another extension on the same proxy, such as 1008, simply pick up attached phone, dial 1008 and then press the # or wait for 4 seconds.

To dial a PSTN number such as 6266667890, you might need to enter in some prefix number followed by the phone number. Please check with your VoIP service provider to get the information. If you phone is assigned with a PSTN-like number such as 6265556789, most likely you just follow the rule to dial 16266667890 as if you were calling from a regular analog phone, followed by pressing the # or wait for 4 seconds.

5.2.2 Direct IP calls

Direct IP calling allows two parties, that is, a HandyTone with an analog phone and another VoIP Device, to talk to each other in an ad hoc fashion without a SIP proxy. This kind of VoIP calls can be made between two parties if:

- Both HandyTone ATA and other VoIP Device(i.e., another HandyTone ATA or Budgetone SIP phone or other VoIP unit) have public IP addresses, or
- Both HandyTone ATA and other VoIP Device are on the same LAN using private IP addresses, or
- Both HandyTone ATA and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

To make a direct IP to IP call, first pick up the analog phone or turn on the speakerphone on the analog phone, then access the voice menu prompt by dial "***" or press the button on the HandyTone-286, and dials "47" to access the direct IP call menu. User will hear a voice prompt "Direct IP Calling" and a dial tone. Enter a 12-digit target IP address to make a call. Destination ports can be specified by using "*4" (encoding for ":") followed by the port number.

Examples:

If the target IP address is 192.168.0.10, the dialing convention is
Voice Prompt with option 47, then 192 168 000 010
followed by pressing the "#" key if it is configured as a send key or wait for more than 5 seconds.

If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be:
Voice Prompt with option 47, then 192168001020*45062 followed by pressing the "#" key if it is configured as a send key or wait for 4 seconds.

5.2.3 Call Hold

While in conversation, pressing the "FLASH" button on the attached phone will put the remote end on hold. Pressing the "FLASH" button again will release the previously Hold party and the bi-directional media will resume.

5.2.4 Call Waiting

If call waiting feature is enabled, while the user is in a conversation, he will hear a special stutter tone if there is another incoming call. User can press the flash button to put the current call party on hold and switch to the other call. Pressing flash button toggles between two active calls.

5.2.5 Call Transfer

5.2.5.1 Blind Transfer

Assuming that call party A and B are in conversation. A wants to *Blind 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 dials ***87** followed by C's number, then #(or wait for 4 seconds)
3. A can hang up.

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

A can hold on to the phone and wait for one of the three following behaviors:

- A quick confirmation tone (temporarily using the call waiting indication tone) followed by a dial tone. This indicates the transfer is successful (transferee has received a 200 OK from transfer target). At this point, A can either hang up or make another call.

- 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.

- Busy tone keeps playing. This means we have failed to receive the second NOTIFY from the transferee and decided to time out. Note: this does not indicate the transfer has been successful, nor does it indicate the transfer has failed. When transferee is a client that does not support the second NOTIFY (such as our own earlier firmware), this will be the case. In bad network scenarios, this could also happen, although the transfer may have been completed successfully.

5.2.5.2 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 followed by # (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:

- When Attended Transfer failed and A hang up, HandyTone- 486 will ring user A back again to remind A that B is still on the call. A can pick up the phone to restore conversation with B.

5.2.6 3-way Conferencing

HandyTone-496 supports 3-way conference in two styles: star code style or Bellcore style.

5.2.6.1 Star Code Style 3-way Conference

Assuming that call party A and B are in conversation. A 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 press "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.

5.2.6.2 Bellcore Style 3-way Conference

Bellcore style 3-way conference is also supported. To do this, user needs to enable "Use Bell-style 3-way Conference" in FXS1 or FXS2 web configuration.

Assuming that call party A and B are in conversation. A 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 press "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.

5.2.7 PSTN Pass Through/Life line

HandyTone-486 Rev. 2.0 supports PSTN pass through. User can send and receive PSTN calls with the attached analog phone.

To receive PSTN calls, simply make phone off hook when the analog phone rings.

To make a PSTN call, press the PSTN access code (*00 by default, or any number configured in web configuration page) to switch to PSTN line and get a dial tone, then dial the PSTN number.

When HandyTone-486 is out of power, it will function as a jack. User can use the same analog phone for PSTN calls.

5.3 Call Features

5.3.1 Call Features Table (star code)

Following table shows the call features (* code) of HandyTone-486.

Key	Call Features
*23	3-way conference
*87	Blind Transfer
*30	Block CallerID (for all-config change)
*31	Send CallerID (for all-config change)
*67	Block CallerID (per call)
*82	Send CallerID (per call)
*50	Disable Call Waiting (for all-config change)
*51	Enable Call Waiting (for all-config change)
*70	Disable Call Waiting. (Per Call)
*71	Enable Call Waiting (Per Call)
*72	Unconditional Call Forward. To use this feature, dial "*72", wait for the dial tone. Then dial the forward number ended with #, wait for dial tone, hang up.
*73	Cancel Unconditional Call Forward To cancel "Unconditional Call Forward", dial "*73" and get the dial tone, then hang up.
*90	Busy Call Forward To use this feature, dial "*90", wait for the dial tone. Then dial the forward number ended with #, wait for dial tone, hang up.
*91	Cancel Busy Call Forward To cancel "Busy Call Forward", dial "*91" and get the dial tone, then hang up
*92	Delayed Call Forward To use this feature, dial "*92", wait for the dial tone. Then dial the forward number ended with #, wait for dial tone, hang up.
*93	Cancel Delayed Call Forward To cancel this Forward, dial "*93" and get the dial tone, then hang up
Flash/Hook	When in conversation, this action will switch to the new incoming call if user heard the call waiting sound. When in conversation and no incoming call heard, this action will switch to a new channel for a new call.

5.4 FAX

HandyTone-486 supports FAX in two modes: T.38 (Fax over IP) and 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 Fax mode to be T.38. If the service provider does not support T.38, pass-through mode may be used. To send or receive faxes in fax pass through mode, users will need to select all the Preferred Codecs to be PCMU/PCMA.

5.5 LED Light Pattern Indication

Following are the LED light pattern indications.

RED LED indicates abnormal status	
Button flashes every 1 second	Ethernet link is down
Button flashes every 2 seconds (if DHCP is configured)	DHCP Failed or WAN No Cable
Button flashes every 4 seconds (if SIP server is configured)	HT–486 fails to register
Button flashes every 6 seconds	Firmware Upgrading
Button flashes briefly	No STUN responses
Red light steady.	Device Malfunctions

Green LED indicates normal status	
Button flashes every 2 second	Message Waiting Indication
Button flashes every 1/10 second	Ringing
Button flashes every second	Ringing interval
Green light steady	In conversation

6 Configuration Guide

6.1 Configuring HandyTone-486 through Voice Prompt

6.1.1 DHCP Mode

Follow section 5.1 with voice menu option 01 to enable HandyTone-486 to use DHCP.

6.1.2 STATIC IP Mode

Follow section 5.1 with voice menu option 01 to enable HandyTone-486 to use STATIC IP mode, then use option 02, 03, 04, 05 to set up IP address, Subnet Mask, Gateway, and DNS server respectively.

6.1.3 Firmware Server IP Address

Follow section 5.1 with voice menu option 13 to configure the IP address of the firmware server.

6.1.4 Configuration Server IP Address

Follow section 5.1 with voice menu option 14 to configure the IP address of the configuration server.

6.1.5 Upgrade Protocol

Follow section 5.1 with voice menu option 15 to choose firmware and configuration upgrade protocol. User can choose between TFTP and HTTP.

6.1.6 Firmware Upgrade Mode

Follow section 5.1 with voice menu option 17 to choose firmware upgrade mode among the following three options:

- always check
- check when pre/suffix changes
- never upgrade

6.1.7 WAN Port Web Access

Follow section 5.1 with voice menu option 12 to enable WAN Port Wed Access of the device configuration pages.

6.2 Configuring HandyTone-486 with Web Browser

HandyTone-486 has an embedded Web server that will respond to HTTP GET/POST requests. It also has embedded HTML pages that allow a user to configure the HandyTone-486 through a Web browser such as Microsoft's IE and AOL's Netscape.

6.2.1 Access the Web Configuration Menu

HandyTone-486's web configuration page can be accessed via LAN or WAN port:

- From the LAN port:
    - Directly connect a computer to the LAN port.
    - Open a command window on the computer
    - Type in "ipconfig /release", the IP address etc. become 0.0.0.0.
    - Type in "ipconfig /renew", the computer gets an IP address in 192.168.2.1 by default
    - Open a web browser, type in the default gateway IP address. You will see the login page of the device.

http://192.168.2.1

- 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, first enable the "WAN side HTTP access" option via IVR option 12. With the WAN side HTTP access enabled, then get the WAN IP address of the HandyTone-486 through section 5.1 with menu option 02. The HandyTone-486's Web Configuration page can be accessed by the following URI via WAN port:

http://HandyTone-IP-Address

where the ***HandyTone-IP-Address*** is the WAN IP address of the HandyTone-486.

NOTE:

- To type IP address into browser to get into the configuration page, please strip out the leading "0"s as the browser will parse in octet. e.g.: if the IP address is: 192.168.001.014, please type in: 192.168.1.14.

6.2.2 End User Configuration

Once this request is entered and sent from a Web browser, the HandyTone-486 will respond with a login screen.

The password is case sensitive with a maximum length of 25 characters. The factory default password for End User and administrator is “123” and “admin” respectively. Only administrator can get access to “ADVANCED SETTINGS” configuration page.

NOTE:

- If you can not log into the configuration page by using the default password, please check with your VoIP service provider. Most likely, the service provider has provisioned the device and configured for you and changed the default password.

After the correct password is entered into the login screen, the embedded Web server inside the device will respond with a BASIC SETTINGS configuration page.

End User Password	This field contains the password to access the Web Configuration Menu. The password is case sensitive with a maximum of 25 characters.
Web Port	This is the device’s internal HTTP server port. Default is 80.
IP Address	There are 2 modes under which the HandyTone ATA can operate: - If DHCP mode is enabled, then all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory) and the IP phone will acquire its IP address from the first DHCP server it discovers on the LAN it attaches to. To use PPPoE feature please set the PPPoE account settings if the HandyTone ATA is connected directly to a DSL modem. The HandyTone ATA will attempt to establish a PPPoE session if any of the PPPoE fields are set. - If Static IP mode is selected, then the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields will need to be configured. 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	This option is used by clients and servers to exchange vendor-specific information. Default is blank.
PPPoE account ID	PPPoE username. Fill this field if your ISP requires you to use a PPPoE (Point to Point Protocol over Ethernet) connection.
PPPoE password	PPPoE account password.

Time Zone	This parameters decides how the displayed date/time will be adjusted according to the specified time zone.
Daylight Savings Time	This parameter controls whether the displayed time will be 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. The "Automatic Daylight Saving Time Rule" shall have the following syntax: start-time;end-time;saving Both start-time and end-time have the same syntax: month,day,weekday,hour,minute month: 1,2,3,..,12 (for Jan, Feb, .., Dec) day: [+|-]1,2,3,..,31 weekday: 1, 2, 3, .., 7 (for Mon, Tue, .., Sun), or 0 which means the daylight saving rule is not based on week days but based on the day of the month. hour: hour (0-23), minute: minute (0-59) If "weekday" is 0, it means the date to start or end daylight saving is at exactly the given date. In that case, the "day" value must not be negative. If "weekday" is not zero and "day" is positive, then the daylight saving starts on the first "day"th iteration of the weekday (1st Sunday, 3rd Tuesday etc). If "weekday" us not zero and "day" is negative, then the daylight saving starts on the last "day"th iteration of the weekday (last Sunday, 3rd last Tuesday etc). The saving is in the unit of minutes. The saving time may also be preceded by a negative (-) sign if subtraction is desired instead of addition. The default value for "Automatic Daylight Saving Time Rule" shall be set to "04,01,7,02,00;10,-1,7,02,00;60" which is the rule for US. Examples US/Canada where daylight saving time is applicable: 04,01,7,02,00;10,-1,7,02,00;60 This means the daylight saving time starts from the first Sunday of April at 2AM and ends the last Sunday of October at 2AM. The saving is 60 minutes (1hour).
PSTN Access Code	Default is "*00". User can switch the phone to PSTN line connected to the Line port of ATA and make outgoing calls.
Device Mode	Default is NAT router mode. HandyTone-486 Rev.2.0 can be configured in Bridge mode so the device functions as a bridge.
WAN Side HTTP Access	Default is "No". The access to configuration page via WAN port is disabled. This setting has no effect if the device is in Bridge mode.

Reply to ICMP on WAN port	Unit will not respond to PING from WAN side if set to “No”.
Cloned WAN MAC Address:	Allow user to set a specific MAC address. Set in Hex format.
LAN Subnet Mask	Sets the LAN subnet mask. Default value is 255.255.255.0. If bridge mode is selected, the LAN settings have no effect.
LAN DHCP Base IP:	Base IP for the LAN port, which functions as a gateway for its LAN. Default value is 192.168.2.1
DHCP IP Lease Time	The amount of time that a given IP address will be valid for a LAN client. Value is set in units of hours. Default value is 120hr (5 Days).
DMZ IP:	Forward all WAN IP traffic to a specific IP address if no matching port is used by HandyTone-486 itself or in the defined port forwarding.
Port Forwarding:	Allow user to forward 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.

MAC Address	The device ID, in HEX format. This is very important ID for ISP troubleshooting.
WAN IP Address	This field shows WAN port IP address.
Product Model	This field contains the product model info, such as HT486 Rev:2.0
Software Version	**Program:** This is the main software release. This number is always used for firmware upgrade. Current release is 1.0.8.32. **Bootloader:** current version is 1.0.8.11. **HTML:** current version 1.0.8.32. **VOC:** current version is 1.0.0.12
System Uptime	This shows how long the device has been up since the last reboot.
Registered	This shows whether the unit is registered to service provider’s server or not.
PPPoE Link Up	This field shows whether the PPPoE connection is up if the HandyTone ATA is connected to DSL modem.
NAT	This shows what kind NAT the HandyTone ATA is connected to via its WAN port. It is based on STUN protocol.

NAT Mapped IP	WAN side public IP if connected to LAN of a SOHO router.
NAT Mapped Port	WAN side SIP port if connected to LAN of a SOHO router.
Other Statistical Status of ATA	Self explainable, please see the page displayed.

6.2.3 Advanced User Configuration

To login to the Advanced User Configuration page, please follow the instructions in section 6.2.1 to get to the following login page. The password is case sensitive and the factory default password for Advanced User is “admin”.

Advanced User configuration includes not only the end user configuration, but also advanced configuration such as SIP configuration, Codec selection, NAT Traversal Setting and other miscellaneous configuration.

Admin Password	Administrator password. Only administrator can configure the “Advanced Settings” page. Password field is purposely left blank for security reason after user clicks UPDATE button. This field is case sensitive and the maximum password length is 25 characters.
SIP Server	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 ATA for firewall or NAT penetration in different network environment. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy will provide solution for it.
SIP User ID	User account information, provided by VoIP service provider (ITSP), usually has the form of digit similar to phone number or actually a phone number.
Authenticate ID	ID used for authentication, usually same as SIP user ID, but could be different and decided by ITSP.
Authentication Password	Account information, password for ATA to register to (SIP) servers of ITSP.
Name	SIP service subscriber’s name which will be used for Caller ID display.
Home NPA	Local area code for North American Dial Plan.

Preferred Vocoder	The HandyTone ATA supports 6 different codec types including G.711 A/U law，G.723.1, G.726, G.729A/B, iLBC. A user can configure Codecs in a preference list that will be included with the same preference order in SDP message.
G723 Rate:	Encoding rate for G723 codec. By default, 6.3kbps rate is set.
iLBC frame size:	iLBC packet frame size. Default is 20ms. For Asterisk PBX, 30ms might be required.
iLBC payload type:	Payload type for iLBC. Default value is 97. The valid range is between 96 and 127.
Silence Suppression	This enables/disables the silence suppression/VAD feature of G723. If set to “Yes”, when a silence is detected, small quantity of VAD packets (instead of audio packets) will be sent during this period. If set to “No”, this feature is disabled.
Voice Frames per TX	This field contains the number of voice frames transmitted in a single packet. When setting this value, the user should be aware of the requested packet time (used in SDP message) as a result of configuring this parameter. This parameter is associated with the first codec in the above codec Preference List or the actual used payload type negotiated between the 2 conversation parties at run time. e.g., if the first codec is configured as G723 and the “Voice Frames per TX” is set to be 2, then the “ptime” value in the SDP message of an INVITE request will be 60ms because each G723 voice frame contains 30ms of audio. Similarly, if this field is set to be 2 and if the first codec chosen is G729 or G711 or G726, then the “ptime” value in the SDP message of an INVITE request will be 20ms. If the configured voice frames per TX exceeds the maximum allowed value, the HandyTone ATA will use and save the maximum allowed value for the corresponding first codec choice. The maximum value for PCM is 10(x10ms) frames. For G726, it is 20 (x10ms) frames. And 32 (x30ms) and 64 (x10ms) frames for G723 and G729 respectively.
Fax Mode	T.38 (Auto Detect) FoIP by default, or Fax Pass-Through.
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 (VoIP)	Layer 2 QoS settings for VoIP traffic. Default setting is blank. VLAN supported equipment is required if user needs to change these settings.
Layer 2 QoS (PC)	Layer 2 QoS settings for LAN port device traffic. Default setting is blank. VLAN supported equipment is required if user needs to change these settings.

Allow incoming SIP messages from SIP proxy only	If set to "Yes", the device will ignore any SIP message that does not come from the IP address (Source IP in the IP header) that it is registered to. Default setting is "No".
Use DNS SRV:	Default is "No". If set to "Yes" the client will use DNS SRV to lookup for the server.
User ID is Phone Number	If set to yes, a "user=phone" parameter will be attached to the "From" header in SIP request.
SIP Registration	This parameter decides whether the HandyTone ATA 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 device will first send registration request to remove previous bindings.
Register Expiration	This parameter allows the user to specify the time frequency (in minutes) the HandyTone ATA will refresh its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
Early Dial	Default is "No". Use only if proxy supports 484 response.
Allow outgoing call without Registration	Default is "No". When set to "Yes", the device still has a dial tone and allows user make outgoing calls even if the device is not registered to the SIP proxy.
Dial Plan Prefix	Sets the prefix added to each dialed number.
No Key Entry timeout	Default is 4 seconds.
Use # as Send Key	This parameter allows user to configure the "#" key to be used as the "Send" (or "Dial") key. Once set to "Yes", pressing this key will immediately trigger the sending of dialed string collected so far. If set to "No", the "#" key will then be included as part of the dialed string to be sent out.
Local SIP port	This parameter defines the local SIP port the HandyTone ATA listens and transmits. The default value is 5060.
Local RTP port	This parameter defines the local RTP-RTCP port pair the HandyTone ATA listens and transmits. It is the base RTP port for channel 0. When configured, channel 0 will use this port_value for RTP and the port_value+1 for its RTCP; channel 1 will use port_value+2 for RTP and port_value+3 for its RTCP. The default value is 5004.
Use Random Port	Default "No". If set to "Yes", the device will pick randomly-generated SIP and RTP ports. This is usually necessary when multiple HandyTone ATAs are behind the same NAT.

SIP Registration Failure Retry Wait Time	Retry registration if the process failed. Default is 20 seconds.
NAT Traversal	This setting decides whether the NAT traversal mechanism is activated. It should be set to "Yes" if the device is behind a NAT router. If no outbound proxy is configured, a STUN server needs to be set to activate STUN detection mechanism. Usually ITSP will provide these settings. If this field is set to "Yes", then the device will periodically (every ***Keep-alive interval***) send a dummy UDP packet to the SIP server to pinhole the NAT.
Keep-alive interval	Default is 20 seconds. The minimum value allowed is 10 seconds. This is the interval of sending dummy UDP packet to keep the NAT "pin hole" open.
Use NAT IP:	If configured, the NAT IP address will be used in SIP/SDP message. Default is blank.
Use STUN keep-alive to detect networks connectivity	Use STUN keep-alive to detect WAN side network problems. If keep-alive request does not yield any response for configured number of times, the device will restart the TCP/IP stack. If the STUN server does not respond when the device boots up, the feature is disabled.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Subscribe for MWI:	Default is "No". When set to "Yes" a SUBSCRIBE for Message Waiting Indication will be sent periodically.
Offhook Auto-Dial	This parameter allows a user to configure a User ID or extension number to be automatically dialed upon offhook. Please note that only the user part of a SIP address needs to be entered here. The HandyTone ATA will automatically append the "@" and the host portion of the corresponding SIP address. Note: Please write down the IP address of the ATA if you use this feature as it will prevent you to access the IVR and the only way to access the device configuration is via the web configuration page.
Enable Call Features	Default is "Yes". Advanced call features using start codes are supported locally.
Use Bell-style 3-way Conference	If this parameter is set to "Yes", user will be able to make Bellcore style 3-way conference. *23 will be disabled.
Disable Call Waiting	Default is "No". User can use star codes to enable/disable call waiting.
Disable Call-Waiting Caller-ID	Default is "No".

Send DTMF	This parameter specifies the mechanism to transmit DTMF digit. There are 3 modes supported: in audio which means DTMF is combined in audio signal (not very reliable with low-bit-rate codec), via RTP (RFC2833), or via SIP INFO. Multiple selections of DTMF method are supported.
DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833
Send Flash Event	Default is “No”. If set to “Yes”, flash will be sent as a DTMF event.
Onhook Threshold	The amount of time the hookflash is pressed that will cause the device to onhook. Default is 800ms.
FXS Impedance	Selects the impedance of the analog telephone connected to the Phone port.
Caller ID Scheme	Select the Caller ID Scheme to suit the standard of different area. • Bellcore (North America) • CID-Canada • DTMF-Brazil • DTMF-Sewden • DTMF (Denmark) • ETSI-DTMF (Finland, Sweden) • ETSI-FSK (France, Germany, Norway, Taiwan, UK-CCA)
Onhook Voltage	The onhook voltage can be selected according to the line voltage depending on the analog phone used. The low power/high power will increase/decrease the output current. Selecting "low power" will make loop current limit = 20mA, selecting "high power" make loop current limit = 32mA. The default selection is 36V (High Power).
Polarity Reversal	Select Polarity Reversal to adapt some call charge/billing system. Default is “No”.
NTP server	This is the URI or IP address of the NTP (Network Time Protocol) server, which the HandyTone ATA will use to synchronize the date/time.
Send Anonymous	If this parameter is set to “Yes”, user ID will be sent as anonymous, essentially blocks the Caller ID from displaying.
Anonymous Method	If it is set to “Use from header”. Callers’ SIP user ID will be sent as anonymous, essentially block the Caller ID from displaying. If it is set to “User privacy header”, the SIP INVITE message contains a “privacy” header, and the server blocks the caller ID from the called party.
Time to Ring	The duration of ringing when a call is not answered. Default is 60 seconds.
Special Features	Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Nortel, Broadsoft, etc.

CBCOM Encode	SIP, RT(C)P and T.38 modes, 1.0 and 1.1
CBCOM Encoder 1.1 Key	Key to be used by CBCOM
Syslog Server	The IP address or URL of System log server. This feature is especially useful for ITSP (Internet Telephone Service Provider).
Syslog Level	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: • product model/version on boot up (INFO level) • NAT related info (INFO level) • sent or received SIP message (DEBUG level) • SIP message summary (INFO level) • inbound and outbound calls (INFO level) • registration status change (INFO level) • negotiated codec (INFO level) • Ethernet link up (INFO level) • SLIC chip exception (WARNING and ERROR levels) • memory exception (ERROR level) The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: GS_LOG: [device MAC address][error code] error message Here is an example: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up
Session Expiration	Grandstream implemented SIP Session Timer. The session timer extension enables SIP sessions to be periodically “refreshed” via a re-INVITE request. Once the session interval expires, if there is no refresh via a re-INVITE message, the session will be terminated. Session Expiration is the time (in seconds) at which the session is considered timed out, if no successful session refresh transaction occurs beforehand. The default value is 180 seconds.
Min-SE	The minimum session expiration (in seconds). The default value is 90 seconds.

Caller Request Timer	If selecting "Yes" the phone will use session timer when it makes outbound calls if remote party supports session timer.
Callee Request Timer	If selecting "Yes" the phone will use session timer when it receives inbound calls with session timer request.
Force Timer	If selecting "Yes" the phone will use session timer even if the remote party does not support this feature. Selecting "No" will allow the phone to enable session timer only when the remote party support this feature. To turn off Session Timer, select "No" for Caller Request Timer, Callee Request Timer, and Force Timer.
UAC Specify Refresher	As a Caller, select UAC to use the phone as the refresher, or UAS to use the Callee or proxy server as the refresher.
UAS Specify Refresher	As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the phone as the refresher.
Force INVITE	Session Timer can be refreshed using INVITE method or UPDATE method. Select "Yes" to use INVITE method to refresh the session timer.
Firmware Upgrade and Provisioning	Default method is HTTP. Firmware upgrade may take up to 10 minutes depending on network environment. Do not interrupt the firmware upgrading process.
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, HT486 rev. 2.0 will request the 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 and input the number, in minutes, you want the HT to check for an update. When set to No, HT502 will only do upgrade once at boot up. "Always check for New Firmware at Boot up" will check for new firmware every time the device reboots. "Check New Firmware only when F/W pre/suffix changes" will check for updates only when the pre/suffix has been changed.
Firmware Key	For firmware encryption. It should be 32 digit in Hexadecimal Representation. End user should keep it blank.

Authenticate Conf File	Default is "No". If set to "Yes", configuration file would be authenticated before acceptance. End user should use default setting.
Lock keypad update	If this parameter is set to "Yes", except for IVR MENU items 1 to 5, the configuration update via keypad is disabled.
Allow conf SIP Account in Basic Settings	Default "No". If set to "Yes", user ID, authentication IP, authentication password and display name can be configured in BASIC SETTINGS page.
Volume Amplification	Handset volume adjustment. 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.
Powerline Ring Tone	This setting allows user to configure the ringing frequencies and cadences.
Call Progress Tones	Using these settings, users can configure various call progress tone frequencies and cadences according to their country standard. By default they are set to North American standard. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. ON is the period of ringing ("On time" in 'ms') while OFF is the period of silence. In order to set a continuous ring, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern. Up to three cadences are supported.
Disable Line Echo Canceller (LEC):	Default is No. If set to Yes, echo canceller is not used.

6.2.4 Saving the Configuration Changes

Once a change is made, users should click on the "Update" button in the Configuration page. The HandyTone-486 will then display a screen to confirm that the changes have been saved.

6.2.5 Rebooting the HandyTone-486 from remote

The user/administrator of the HandyTone-486 can remotely reboot the HandyTone-486 by pressing the "Reboot" button at the bottom of the configuration page. Once done, a screen will be displayed to indicate that rebooting is underway.

6.3 Configuration through a Central Server

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

When HandyTone ATA boot up, it will send TFTP or HTTP request to download configuration file, “cfg000b82xxxxxx”, where “000b82xxxxxx” 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 has a provisioning system called GAPS (Grandstream Automated Provisioning System) that is used to support automated configuration of Grandstream devices. GAPS uses enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual Grandstream device 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/support/configurationtool.html.

7 Software Upgrade

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

7.1 Firmware Upgrade through TFTP/HTTP

To upgrade via TFTP or HTTP, the "Firmware Upgrade and Provisioning upgrade via" field (IVR option 17) 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/1.1.0.37
e.g. 72.172.83.110

NOTES:

- Firmware server in IP address format can be configured via IVR. Please refer to section 5.1 for instructions. If firmware server is in FQDN format, it must be set via web configuration interface.
- Grandstream recommends end-user use the Grandstream HTTP server. Its address can be found at http://www.grandstream.com/firmware.html. Currently the HTTP firmware server IP address is 72.172.83.110. For large companies, we recommend to maintain their own TFTP/HTTP server for upgrade and provisioning procedures.
- Once a "Firmware Server Path" and the upgrade protocol are 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 HandyTone ATA will attempt to retrieve the new image files by downloading them into the HandyTone ATA's SRAM. During this stage, the HandyTone 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 HandyTone 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 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.

Instructions 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 HT486 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

Please be advised that our client will pull out firmware from the WAN side, if the TFTP server is connected to the device's LAN port, the firmware upgrade will not work by design.

7.2 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 "cfgxxxxxxxxxxxx", where "xxxxxxxxxxxx" is the MAC address of the device, i.e., "cfg000b820102ab". The configuration file name should be in lower cases.

7.3 Firmware and Configuration File Prefix and Postfix

Starting from firmware version 1.0.7.11 for HandyTone-486 Rev 2.0, adding prefix and postfix for both firmware and configuration file is supported.

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.

7.4 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.

8 Restore Factory Default Setting

Warning !!!

Restore the Factory Default Setting will DELETE all configuration information of the device. 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 service provider.

Please disconnect network cable and power cycle the unit before trying to reset the unit to factory default. The steps are as follows:

Step 1:

Find the MAC Address of the device. It is a 12 digits HEX number located on the bottom of the unit.

Step 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".

Step 3:

To perform factory reset:

a. Press "***" or the LED button for voice prompt.
b. Enter "99" and get the voice prompt "Reset".
c. Enter the encoded MAC address of the device.
d. Wait for 15 seconds.

The device will reboot automatically and restore to factory default setting.

NOTE:

- Please be aware by default the HandyTone-486 WAN side HTTP access is disabled. After a factory reset, the device's web configuration page can be accessed only from its LAN port, please refer to instructions in section 6.2.1 for details.