

User Manual

HandyTone-286 Rev 3.0

Analog Telephone Adaptor

For Firmware Version 1.0.8.16



Grandstream Networks, Inc.

www.grandstream.com



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1 Welcome

Congratulations on becoming an owner of HandyTone-286. You made an excellent choice and we hope you will enjoy all its capabilities.

Grandstream's award-winning HandyTone-286 is innovative Analog Telephone Adaptor that offers a rich set of functionality and superb sound quality at ultra-affordable price. They are fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market.

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/user_manuals/HandyTone.pdf

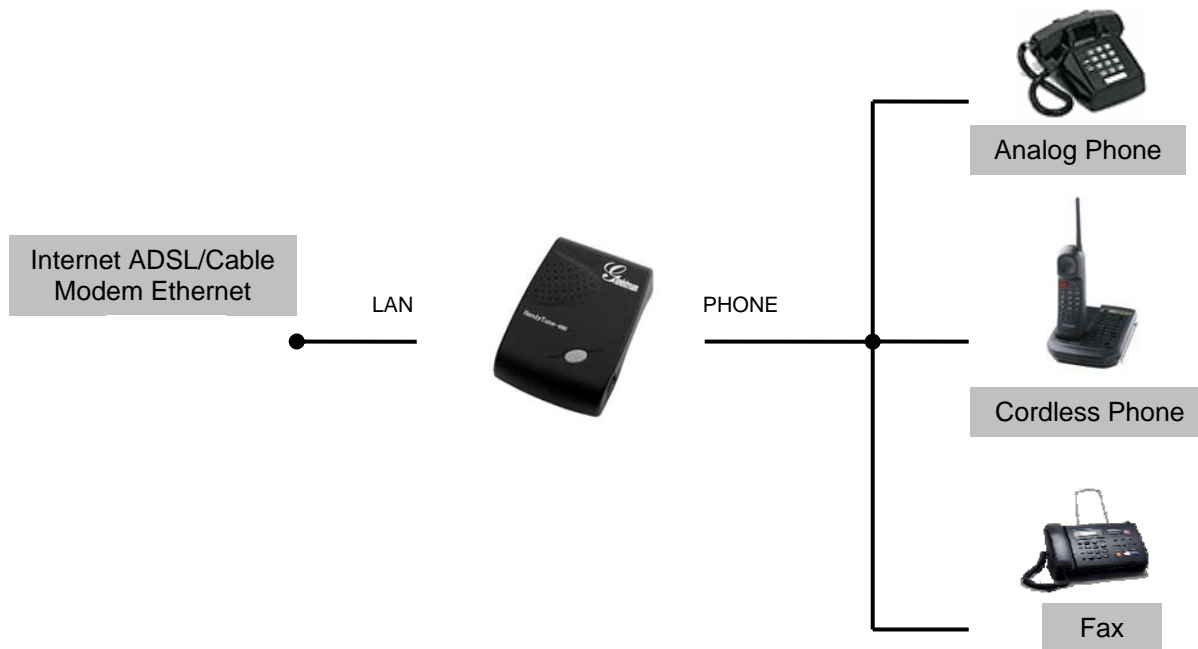


2 Installation

HandyTone-286 is a VoIP Analog Telephone Adaptor designed to work with an ordinary analog telephone. The following photo illustrates the appearance of a HandyTone-286.



Interconnection Diagram of the HandyTone-286:



3 What is Included in the Package

The HandyTone-286 package contains:

- 1) One HandyTone-286
- 2) One universal power adaptor
- 3) One Ethernet cable

3.1 Safety Compliances

The HandyTone-286 is compliant with various safety standards including FCC/CE and C-Tick. Its power adaptor is compliant with UL standard. The HandyTone ATA 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-286.

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.

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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)
- 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 (for PCMU and 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 SIP Session Timer
- Support Syslog (on Rev.2.0)
- Ultra compact (wallet size) and lightweight design, great companion for travelers
- Compact, lightweight Universal Power adaptor

4.2 Hardware Specification

The table below lists the hardware specification of HandyTone-286.

<u>Model</u>	<u>HandyTone-286</u>
LAN interface	1xRJ45 10Base-T
Button	1
LED	GREEN & RED color
Universal Power Adaptor	Input: 100-240VAC Output: +5VDC, 1200mA UL certified
Dimension	65mm (W) 93mm (D) 27mm (H)
Weight	0.57 lbs (0.26kg)
Operating Temperature	32 - 104°F 0 - 40°C
Humidity	10% - 95% (non-condensing)
Compliance	FCC/CE/C-Tick

5 Basic Operations

5.1 Get Familiar with Key Pad and Voice Prompt

HandyTone-286 stores a voice prompt menu for quick browsing and simple configuration. To enter this voice prompt menu, simply press the button on the HandyTone-286 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	“Static IP Mode”, or “Dynamic IP Mode”	Enter “9” to toggle the selection. <i>If user selects “Static IP Mode”, user need configure the all 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.</i>
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
06	“TFTP Server “ + IP address	Same as Menu option 02
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”.)
86	“No Voice Messages”; or “Voice Messages Pending”	If there are voice messages, user can dial “9” and dial pre-configured phone number to retrieve voice message.
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 return to Main Menu

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, it automatically processes them.
- 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

To make a phone or extension number call:

- a) Dial the number directly and wait for 4 seconds (Default “No Key Entry Timeout”). Or
- b) Dial the number 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 the 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 # 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 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 192.168.0.160, the dialing convention is

Voice Prompt with option 47, then 192168000160

followed by pressing the “#” key if it is configured as a send key or wait 4 seconds. In this case, the default destination port 5060 is used if no port is specified.

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 held 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. Then A dials *87 then dials C’s number, and 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 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:

- When Attended Transfer failed, if A hangs up, the HandTone-286 will ring user A 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-286 supports 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 followed by 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.

5.3 Call Features

Following table shows the call features of HandyTone-286.

Key	Call Features
*23	3-way conference code.
*87	Bland transfer
*30	Block Caller ID (for all subsequent calls)
*31	Send Caller ID (for all subsequent calls)
*67	Block Caller ID (per call)
*82	Send Caller ID (per call)
*50	Disable Call Waiting (for all subsequent calls)
*51	Enable Call Waiting (for all subsequent calls)
*70	Disable Call Waiting. (Per Call)
*71	Enable Call Waiting (Per Call)
*72	Unconditional Call Forward. To use this feature, dial “*72” and get the dial tone. Then dial the forward number and “#” for a dial tone, then 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” and get the dial tone. Then dial the forward number and “#” for a dial tone, then 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” and get the dial tone. Then dial the forward number and “#” for a dial tone, then 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 there is a call waiting indication. When in conversation without an incoming call, this action will switch to a new channel for a new call.

5.4 Fax

HandyTone-286 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 always indicates not abnormal status	
DHCP Failed or WAN No Cable	Button flashes every 2 seconds (if DHCP is configured)
HandyTone-486 fails to register	Button flashes every 2 seconds (if SIP server is configured)
Firmware Upgrading	Button flashes every 2 seconds
Device Malfunctions	Red light steady on

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

6 Configuration Guide

6.1 Configuring HandyTone-286 IP through Voice Prompt

6.1.1 DHCP Mode

Follow section 5.1 with voice menu option 01 to enable HandyTone-286 to use DHCP.

6.1.2 STATIC IP Mode

Follow section 5.1 with voice menu option 01 to enable HandyTone-286 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 TFTP Server Address

Follow section 5.1 with voice menu option 06 to configure the IP address of the TFTP server.

6.2 Configuring HandyTone-286 with Web Browser

HandyTone-286 has an embedded Web server that will respond to HTTP GET/POST requests. It also has embedded HTML pages that allows a user to configure the HandyTone-286 through a Web browser such as Microsoft's IE and AOL's Netscape.

6.2.1 Access the Web Configuration Menu

First, get the IP address of the HandyTone-286 through section 5.1 with menu option 02. Then access the HandyTone-286's Web Configuration Menu using the following URI:

http:// HandyTone-IP-Address

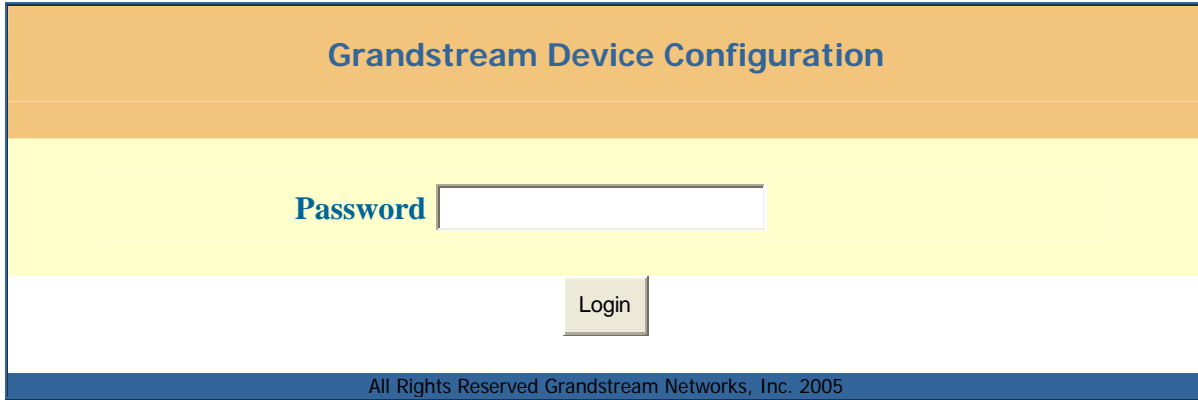
where the *HandyTone-IP-Address* is the IP address of the HandyTone-286.

NOTE:

- To type IP address into browser to get into the configuration page, please strip out the leading "0" 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 IP phone will respond with the following login screen:



The password is case sensitive with 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 SETTING” configuration page.

NOTE:

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

- Status Page.

Grandstream Device Configuration	
STATUS BASIC SETTINGS ADVANCED SETTINGS	
MAC Address:	00.0B.82.08.74.D4
IP Address:	10.10.13.195
Product Model:	HT286 REV 3.0
Software Version:	Program-- 1.0.8.16 Bootloader-- 1.0.8.9 HTML-- 1.0.8.16 VOC-- 1.0.0.10
System Up Time:	0 day(s) 0 hour(s) 0 minute(s)
Registered:	Yes
PPPoE Link Up:	disabled
NAT:	detected NAT type is full cone
NAT Mapped IP:	67.153.142.80

NAT Mapped Port:	60946
Total Inbound Calls:	0
Total Outbound Calls:	0
Total Missed Calls:	0
Total Call Time (in minutes):	0
Total SIP Message Sent:	2
Total SIP Message Received:	2
Total RTP Packet Sent:	0
Total RTP Packet Received:	0
Total RTP Packet Loss:	0
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MAC Address	The device ID, in HEX format. This is very important ID for ISP troubleshooting.
IP Address	IP address assigned on the device.
Product Model	This field contains the product model info.
Software Version	Program: This should be consistent with the firmware version. Bootloader: version number for bootloader, it could be lower than program version. HTML: This should be consistent with the firmware version. VOC: This is the codec program, normally not changed.
System Uptime	This shows system up time since last reboot.
Registered	This shows whether the unit is registered to voip service provider's server.
PPPoE Link Up	This shows whether the PPPoE is up if connected to DSL modem
NAT	This shows what kind of NAT the HandyTone-286 is behind.
NAT Mapped IP	WAN side mapped IP if HandyTone-286 is connected to a NAT router.

NAT Mapped Port	WAN side mapped port if HandyTone-286 is connected to a NAT router.
Statistical Status	Self-explainable. Please refer to the page displayed.

- Basic Settings.

Grandstream Device Configuration

STATUS
BASIC SETTINGS
ADVANCED SETTINGS

End User Password:	<input style="width: 90%;" type="password"/> (purposely not displayed for security protection)
IP Address:	<input checked="" type="checkbox"/> dynamically assigned via DHCP (default) or PPPoE (will attempt PPPoE if DHCP fails and following is non-blank)
	PPPoE account ID: <input style="width: 80%;" type="text"/> PPPoE password: <input style="width: 80%;" type="password"/>
	Preferred DNS server: <input style="width: 20px;" type="text"/> 0 <input style="width: 20px;" type="text"/> 0 <input style="width: 20px;" type="text"/> 0 <input style="width: 20px;" type="text"/> 0
	<input type="checkbox"/> statically configured as:
	IP Address: <input style="width: 20px;" type="text"/> 192 <input style="width: 20px;" type="text"/> 168 <input style="width: 20px;" type="text"/> 0 <input style="width: 20px;" type="text"/> 160
	Subnet Mask: <input style="width: 20px;" type="text"/> 0 <input style="width: 20px;" type="text"/> 0 <input style="width: 20px;" type="text"/> 0 <input style="width: 20px;" type="text"/> 0
	Default Router: <input style="width: 20px;" type="text"/> 0 <input style="width: 20px;" type="text"/> 0 <input style="width: 20px;" type="text"/> 0 <input style="width: 20px;" type="text"/> 0
	DNS Server 1: <input style="width: 20px;" type="text"/> 0 <input style="width: 20px;" type="text"/> 0 <input style="width: 20px;" type="text"/> 0 <input style="width: 20px;" type="text"/> 0
	DNS Server 2: <input style="width: 20px;" type="text"/> 0 <input style="width: 20px;" type="text"/> 0 <input style="width: 20px;" type="text"/> 0 <input style="width: 20px;" type="text"/> 0
Time Zone:	<input style="width: 90%;" type="text" value="GMT-5:00 (US Eastern Time, New York)"/> <input style="width: 20px;" type="button" value="v"/>
Daylight Savings Time:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (if set to Yes, display time will be 1 hour ahead of normal time)
<input style="width: 60px;" type="button" value="Update"/>	

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End User Password	This contains the password to access the Web Configuration Menu. This field is case sensitive with max. 25 characters
IP Address	<p>There are 2 modes under which the IP phone can operate:</p> <ul style="list-style-type: none"> - If DHCP mode is enabled, then all the field 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 DHCP server in the network. <p>To use PPPoE feature please set the PPPoE account settings if the HT-286 is connected directly to a DSL modem. The HT-286 will attempt to establish a PPPoE session if any of the PPPoE fields is set.</p> <ul style="list-style-type: none"> - If Static IP mode is selected, then the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (mandatory), DNS Server 2 (optional) fields will need to be configured.
Time Zone	This parameter controls how date/time is displayed 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”, then the displayed time will be 1 hour ahead of normal time.

6.2.3 Advanced User Configuration

To login to the Advanced User Configuration page, follow the instruction in section 6.2.1 to get to the following login page. The password is case sensitive with a maximum length of 25 characters and the factory default password for Advanced User is “admin”.

Grandstream Device Configuration	
STATUS BASIC SETTINGS ADVANCED SETTINGS	
Admin Password:	<input type="text"/> (purposely not displayed for security protection)
SIP Server:	<input type="text" value="sip.mycompany.com"/> (e.g., sip.mycompany.com, or IP address)
Outbound Proxy:	<input type="text"/> (e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID:	<input type="text" value="12345678"/> (the user part of an SIP address)

Authenticate ID:	<input type="text" value="12345678"/> (can be identical to or different from SIP User ID)
Authenticate Password:	<input type="password"/> (purposely not displayed for security protection)
Name:	<input type="text" value="John Doe"/> (optional, e.g., John Doe)
Advanced Options:	
Preferred Vocoder: (in listed order)	choice 1: <input pcmu\""="" type="text" value="current setting is \"/> choice 2: <input pcma\""="" type="text" value="current setting is \"/> choice 3: <input g723\""="" type="text" value="current setting is \"/> choice 4: <input g729\""="" type="text" value="current setting is \"/> choice 5: <input g726-32\""="" type="text" value="current setting is \"/> choice 6: <input ilbc\""="" type="text" value="current setting is \"/> choice 7: <input pcmu\""="" type="text" value="current setting is \"/>
G723 rate:	<input checked="" type="checkbox"/> 6.3kbps encoding rate <input type="checkbox"/> 5.3kbps encoding rate
iLBC frame size:	<input checked="" type="checkbox"/> 20ms <input type="checkbox"/> 30ms
iLBC payload type:	<input type="text" value="97"/> (between 96 and 127, default is 97)
Silence Suppression:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
Voice Frames per TX:	<input type="text" value="2"/> (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
Fax Mode:	<input checked="" type="checkbox"/> T.38 (Auto Detect) <input type="checkbox"/> Pass-Through
Layer 3 QoS:	<input type="text" value="48"/> (Diff-Serv or Precedence value)
Layer 2 QoS:	802.1Q/VLAN Tag <input type="text" value="0"/> 802.1p priority value <input type="text" value="0"/> (0-7)
Allow incoming SIP messages from SIP proxy only:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
Use DNS SRV:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
User ID is phone number:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
SIP Registration:	<input checked="" type="checkbox"/> Yes <input type="checkbox"/> No
Unregister On Reboot:	<input checked="" type="checkbox"/> Yes <input type="checkbox"/> No
Register Expiration:	<input type="text" value="3600"/> (in seconds. default 1 hour, max 45 days)

<i>Early Dial:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (use "Yes" only if proxy supports 484 response)
<i>Dial Plan Prefix:</i>	<input type="text"/> (this prefix string is added to each dialed number)
<i>No Key Entry Timeout:</i>	<input type="text" value="4"/> (in seconds, default is 4 seconds)
<i>Use # as Dial Key:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes (if set to Yes, "#" will function as the Dial key)
<i>local SIP port:</i>	<input type="text" value="5060"/> (default 5060)
<i>local RTP port:</i>	<input type="text" value="5004"/> (1024-65535, default 5004)
<i>Use random port:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
<i>NAT Traversal:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes, STUN server is: <input type="text"/> (URI or IP:port)
<i>keep-alive interval:</i>	<input type="text" value="20"/> (in seconds, default 20 seconds)
<i>Use NAT IP</i>	<input type="text"/> (used in SIP/SDP message if specified)
<i>Proxy-Require:</i>	<input type="text"/>
<i>SUBSCRIBE for MWI:</i>	<input checked="" type="checkbox"/> No, do not send SUBSCRIBE for Message Waiting Indication <input type="checkbox"/> Yes, send periodical SUBSCRIBE for Message Waiting Indication
<i>Offhook Auto-Dial:</i>	<input type="text"/> (User ID/extension to dial automatically when offhook)
<i>Enable Call Features:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally)
<i>Disable Call-Waiting:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
<i>Send DTMF:</i>	<input checked="" type="checkbox"/> in-audio <input type="checkbox"/> via RTP (RFC2833) <input type="checkbox"/> via SIP INFO
<i>DTMF Payload Type:</i>	<input type="text" value="101"/>
<i>Send Flash Event:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (Flash will be sent as a DTMF event if set to Yes)
<i>Onhook Threshold:</i>	<input type="text" value="800 ms"/>
<i>FXS Impedance:</i>	<input type="text" value="600 Ohm (North America)"/>
<i>Caller ID Scheme:</i>	<input type="text" value="Bellcore (North America)"/>
<i>Onhook Voltage:</i>	<input type="text" value="36V"/>
<i>Polarity Reversal:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes (reverse polarity upon call establishment and termination)

<i>NTP Server:</i>	<input type="text" value="time.nist.gov"/> (URI or IP address)
<i>Send Anonymous:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (caller ID will be blocked if set to Yes)
<i>Anonymous Method:</i>	<input checked="" type="checkbox"/> Use From Header <input type="checkbox"/> Use Privacy Header
<i>Special Feature:</i>	<input type="text" value="Standard"/>
<i>Syslog Server:</i>	<input type="text"/>
<i>Syslog Level:</i>	<input type="text" value="NONE"/>
<i>Session Expiration:</i>	<input type="text" value="180"/> (in seconds. default 180 seconds)
<i>Min-SE:</i>	<input type="text" value="90"/> (in seconds. default and minimum 90 seconds)
<i>Caller Request Timer:</i>	<input type="checkbox"/> Yes <input checked="" type="checkbox"/> No (Request for timer when making outbound calls)
<i>Callee Request Timer:</i>	<input type="checkbox"/> Yes <input checked="" type="checkbox"/> No (When caller supports timer but did not request one)
<i>Force Timer:</i>	<input type="checkbox"/> Yes <input checked="" type="checkbox"/> No (Use timer even when remote party does not support)
<i>UAC Specify Refresher:</i>	<input type="checkbox"/> UAC <input type="checkbox"/> UAS <input checked="" type="checkbox"/> Omit (Recommended)
<i>UAS Specify Refresher:</i>	<input checked="" type="checkbox"/> UAC <input type="checkbox"/> UAS (When UAC did not specify refresher tag)
<i>Force INVITE:</i>	<input type="checkbox"/> Yes <input checked="" type="checkbox"/> No (Always refresh with INVITE instead of UPDATE)
<i>Firmware Upgrade and Provisioning:</i>	Upgrade Via <input type="checkbox"/> TFTP <input checked="" type="checkbox"/> HTTP Firmware Server Path: <input type="text" value="fm.grandstream.com/gs"/> Config Server Path: <input type="text" value="fm.grandstream.com/gs"/> Firmware File Prefix: <input type="text"/> Firmware File Postfix: <input type="text"/> Config File Prefix: <input type="text"/> Config File Postfix: <input type="text"/> Automatic Upgrade: <input type="checkbox"/> No <input checked="" type="checkbox"/> Yes, check for upgrade every <input type="text" value="10080"/> minutes (default 7 days) <input checked="" type="checkbox"/> Always Check for New Firmware <input type="checkbox"/> Check New Firmware only when F/W pre/suffix changes
<i>Firmware Key:</i>	<input type="text"/> (in Hexadecimal Representation)
<i>Authenticate Conf File:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes (cfg file would be authenticated before acceptance if set to Yes)

<i>Lock keypad update:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes (configuration update via keypad is disabled if set to Yes)
<i>Allow conf SIP Account in Basic Settings:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes
<i>Override MTU Size:</i>	<input type="text" value="0"/>
<input type="button" value="Update"/>	
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Admin Password	Administrator password. Only administrator can configure the “Advanced Settings” page. Password field is purposely left blank for security reason after clicking update and saved. The maximum password length is 25 characters.
SIP Server	This field contains the URI string or the IP address. e.g. sip.my-voip-provider.com; 192.168.1.200:5066
Outbound Proxy	This field contains the URI string or the IP of the outbound proxy. If there is no outbound proxy, this field SHOULD be left blank. If it is not blank, all outgoing requests will be sent to this outbound proxy.
SIP User ID	This field contains the user part of the SIP address for this phone. e.g., if the SIP address is: sip:my_user_id@my_provider.com, then the SIP User ID is: my_user_id. Please do NOT include the preceding “sip:” scheme or the host portion of the SIP address in this field. It is given by VoIP service provider.
Authenticate ID	SIP service subscriber’s Authenticate ID used for authentication. It can be identical to or different from SIP User ID and given by VoIP service provider.
Authenticate Password	SIP service subscriber’s account password. It is given by VoIP service provider.
Name	SIP service subscriber’s name which will be used for Caller ID display.
Preferred Vocoder	HandyTone-286 supports up to 7 different vocoder types including G711-ulaw (PCMU), G711-alaw (PCMA), G723, G729A, G726-32, and iLBC. Depending on the product model, some of these vocoders may not be provided in standard release. A user can configure vocoders in a preference order that will be offered in SIP INVITE message.
G723 Rate:	This defines the encoding rate for G723 vocoder. By default, 6.3kbps rate is chosen.
iLBC frame size	This defines the size of the iLBC codec frame. The default setting is 20ms.

iLBC payload type	This defines the iLBC payload type. The default setting is 97. The valid range is between 96 and 127.
Silence Suppression	This controls the silence suppression/VAD feature of G723 and G729. If set to “Yes”, when a silence is detected, small quantity of VAD packets (instead of audio packets) will be sent during the period of no talking. If set to “No”, this feature is disabled.
Voice Frames per TX	<p>This field contains the number of voice frames to be 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 vocoder in the above vocoder Preference List or the actual used payload type negotiated between the 2 conversation parties at run time.</p> <p>e.g., if the first vocoder 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 vocoder chosen is G729 or G711 or G726, then the “ptime” value in the SDP message of an INVITE request will be 20ms.</p> <p>If the configured voice frames per TX exceeds the maximum allowed value, the phone will use and save the maximum allowed value for the corresponding first vocoder choice. The maximum value for PCM is 10(x10ms) frames; for G726, it is 20 (x10ms) frames; for G723, it is 32 (x30ms) frames; for G729/G728, 64 (x10ms) and 64 (x2.5ms) frames respectively.</p>
Fax Mode	T.38 (Auto Detect) FoIP by default, or Pass-Through (must use codec PCMU/PCMA)
Layer 3 QoS	This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv. Default value is 48
Layer 2 QoS	This setting includes two fields. The 802.1Q/VLAN Tag contains the value used for layer 2 VLAN tag. Default setting is blank. And 802.1p priority value contains the value of the priority value.
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 is No.
Use DNS SRV	This parameter controls whether the IP phone supports the DNS SRV route function.
User ID is phone number	If the HandyTone-286 has an assigned PSTN telephone number, then this field will be set to “Yes”. Otherwise, set it to “No”. If “Yes” is set, a “user=phone” parameter will be attached to the “From” header in SIP request.

SIP Registration	This parameter controls whether the IP phone 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.
Registration Expiration	This parameter allows the user to specify the time frequency (in minutes) the phone will refresh its registration with the specified registrar. The default interval is 3600 seconds (or 1 hour). The maximum interval is 45 days.
Early Dial	This parameter controls whether the phone will attempt to send an early INVITE each time a key is pressed when a user dials a number. If set to “Yes”, an INVITE is sent using the dial-number collected thus far; Otherwise, no INVITE is sent until the “(Re-)Dial” button is pressed or after about 5 seconds have elapsed if the user forgets to press the “(Re-)Dial” button. The “Yes” option should be used ONLY if there is a SIP proxy configured and the proxy server supports 484 Incomplete Address response. Otherwise, the call will most likely be rejected by the proxy (with a 404 Not Found error). Please note that this feature is NOT designed to work with and should NOT be enabled for direct IP-to-IP calling.
Dial Plan Prefix	This value contains the dial plan prefix string (typically an ASCII numeric string). If it is not blank, then this string will added to the dialed number.
No Key Entry Timeout	Default is 4 seconds.
Use # as Send Key	This parameter allows the 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. In this case, this key is essentially equivalent to the “(Re)Dial” key. If set to “No”, this # key will then be included as part of the dial string to be sent out.
Local SIP port	This parameter defines the local SIP port the IP phone will listen and transmit on. The default value is 5060.
Local RTP port	This parameter defines the local RTP-RTCP port pair the IP phone will listen and transmit on. 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	This parameter, when set to Yes, will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple IP phones are behind the same NAT.

NAT Traversal	This parameter defines whether the phone NAT traversal mechanism will be activated or not. If activated (by choosing “Yes”) and a STUN server is also specified, then the phone will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the phone will attempt to detect if and what type of firewall/NAT it is behind by sending appropriate request to the specified STUN server. If this field is set to “Yes” with no specified STUN server, then the phone will only periodically (every 20 seconds by default) send a blank UDP packet (with no payload data) to the SIP server to keep the mapped port open on the NAT.
keep-alive interval	The HandyTone-286 sends a UDP package to the SIP server periodically in order to keep the port open on the router. This parameter defines the interval time that HT286 send the UDP package. The default setting is 20 second.
Use NAT IP	NAT IP address is used in SIP/SDP message. Default is blank.
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 the 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 phone will automatically append the “@” and the host portion of the corresponding SIP address.
Enable Call Feature	Default is YES. If set to “Yes”, call features are supported locally, such as call-waiting, transfer, 3-way conference, etc.
Disable Call Waiting	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.
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 DTMF event.
FXS Impedance	Selects the impedance of the analog telephone connected to the Phone port.

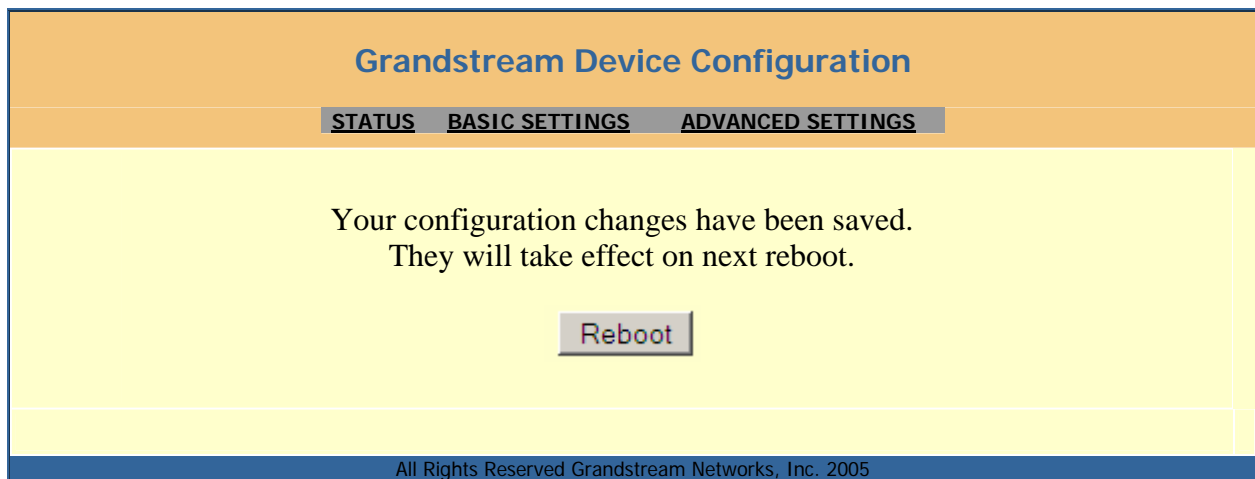
Caller ID Scheme	Select the Caller ID Scheme to suit the standard of different area. <ul style="list-style-type: none"> • Bellcore (North America) • CID (Canada) • DTMF (Brazil) • DTMF (Denmark) • DTMF (Sweden) • ETSI-FSK (France, Germany, Norway, Taiwan, UK-CCA) • ETSI-DTMF (Finland, Sweden)
Onhook Voltage	Select the onhook voltage to suit different area or PBX.
Polarity Reversal	Select Polarity Reversal to adapt some call charge/billing system. Default is No.
NTP server	This parameter defines the URI or IP address of the NTP server which the IP phone will use to display the current date/time.
Send Anonymous	If this parameter is set to “Yes”, the device is employing the mechanism to block its ID.
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.
Special Features	Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Nortel, Broadsoft, etc.
Syslog Server	The IP address or URL of System log server. This feature is especially useful for ITSP (Internet Telephone Service Provider)

<p>Syslog Level</p>	<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> <ul style="list-style-type: none"> • 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) <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 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</p>
<p>Session Expiration</p>	<p>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.</p> <p>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.</p>
<p>Min-SE</p>	<p>The minimum session expiration (in seconds). The default value is 90 seconds.</p>
<p>Caller Request Timer</p>	<p>If selecting “Yes” the phone will use session timer when it makes outbound calls if remote party supports session timer.</p>
<p>Callee Request Timer</p>	<p>If selecting “Yes” the phone will use session timer when it receives inbound calls with session timer request.</p>

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 HTTP. Firmware upgrading may take up to 10 minutes depends 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 blank. If it is configured, HT486 rev. 2.0 will request the firmware file with the prefix. 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	Default is YES.
Firmware Key	32 digit in Hexadecimal Representation. Useful for ITSP to encrypt firmware. End user should keep it blank.
Authenticate Conf File	Default NO. End user should use default setting.
Lock keypad update	If this parameter is set to “Yes”, the configuration update via keypad is disabled.
Allow conf SIP Account in Basic Settings	If set to Yes, these four fields: SIP User ID, Authenticate ID, Authenticate Password and Name will be included in Basic Settings configuration page.
Override MTU size	Override the MTU size.

6.2.4 Saving the Configuration Changes

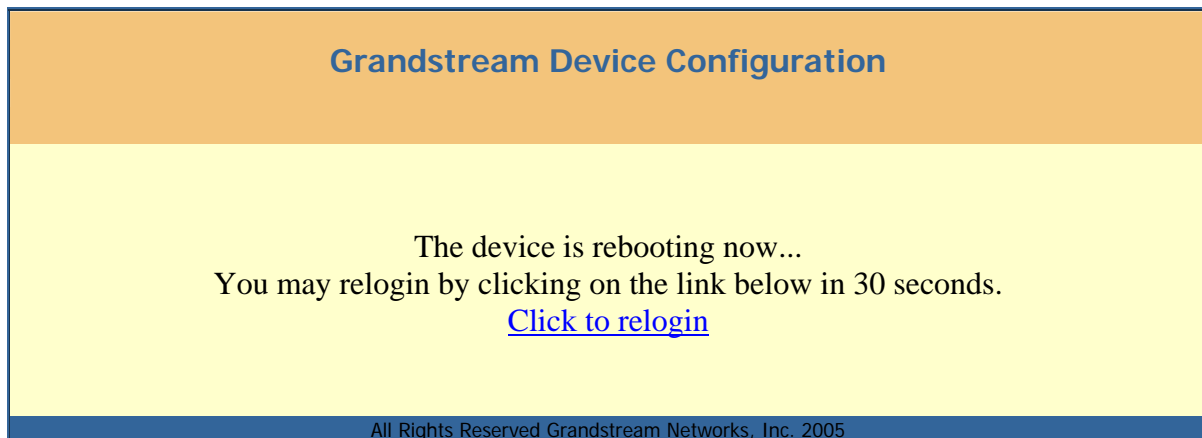
Once a change is made, the user should press the “Update” button in the Configuration Menu. The IP phone will then display the following screen to confirm that the changes have been saved.



Users are recommended to power cycle the HandyTone-286 after seeing the above message.

6.2.5 Rebooting the HandyTone-286 from Remote

The administrator of the phone can remotely reboot the phone by pressing the “Reboot” button at the bottom of the configuration menu. Once done, the following screen will be displayed to indicate that rebooting is underway.



At this point, the user can relogin to the phone after waiting for about 30 seconds.

6.3 Configuration through a Central Server

Grandstream HandyTone ATA 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 file can be loaded into devices via TFTP or HTTP from the central provisioning server, so the service provider or an enterprise with large deployment of HandyTone ATAs can easily manage the configuration and service provision to individual devices remotely.

Grandstream provides a licensed provisioning system called GAPS that can be used to support automated configuration of HandyTone ATA. GAPS (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 templates can be downloaded from http://www.grandstream.com/DOWNLOAD/Configuration_Tool/.

For details on how GAPS works, please refer to the documentation of GAPS product.

7 Software Upgrade with TFTP

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 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/1.0.8.16`

e.g. `168.75.215.189`

NOTES:

- TFTP server in IP address format can be configured via IVR. Please refer to section 6.1.3 for instructions. If TFTP server is in FQDN format, it must be set via web configuration interface.
- 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 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>. Our latest official release can be downloaded from <http://www.grandstream.com/y-firmware.htm>. Unzip the file and put all of them under the root directory of the TFTP server. Put the PC running the TFTP server and the HandyTone ATA in the same LAN segment. Please go to File -> Configure -> Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade. Start the TFTP server, in the HandyTone ATA’s web configuration page, configure the Firmware Server Path with the IP address of the PC, 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.

9 Glossary of Terms

ADSL

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

AGC

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

ARP

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

ATA

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

CODEC

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

CNG

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

DATAGRAM

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

DECIMATE

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

DECT

Digital Enhanced Cordless Telecommunications: A standard developed by the European Telecommunication Standard Institute from 1988, governing pan-European digital mobile telephony. DECT covers wireless PBXs, telepoint, residential cordless telephones, wireless access to the public switched telephone network, Closed User Groups (CUGs), Local Area

Networks, and wireless local loop. The DECT Common Interface radio standard is a multicarrier time division multiple access, time division duplex (MC-TDMA-TDD) radio transmission technique using ten radio frequency channels from 1880 to 1930 MHz, each divided into 24 time slots of 10ms, and twelve full-duplex accesses per carrier, for a total of 120 possible combinations. A DECT base station (an RFP, Radio Fixed Part) can transmit all 12 possible accesses (time slots) simultaneously by using different frequencies or using only one frequency. All signaling information is transmitted from the RFP within a multiframe (16 frames). Voice signals are digitally encoded into a 32 kbit/s signal using Adaptive Differential Pulse Code Modulation.

DNS

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

DID

Direct Inward Dialing

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

DSP

Digital Signal Processing. Using computers to process signals such as sound, video, and other analog signals which have been converted to digital form.

Digital Signal Processor. A specialized CPU used for digital signal processing.

Grandstream products all have DSP chips built inside.

DTMF

Dual Tone Multi Frequency

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

FQDN

Fully Qualified Domain Name

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

FXO

Foreign eXchange Office

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

An FXO interface will accept calls from FXS or PSTN interfaces. All countries and regions have their own standards.

FXO is complimentary to FXS (and the PSTN).

FXS

Foreign eXchange Station

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

An FXS device will allow any FXO device to operate as if it were connected to the phone company. This makes your PBX the POTS+PSTN for the phone.

The FXS Interface connects to FXO devices (by an FXO interface, of course).

DHCP

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

ECHO CANCELLATION

Echo Cancellation is used in [telephony](#) to describe the process of removing [echo](#) from a voice communication in order to improve voice quality on a [telephone call](#). In addition to improving quality, this process improves [bandwidth](#) savings achieved through [silence suppression](#) by preventing echo from traveling across a [network](#).

There are two types of echo of relevance in telephony: acoustic echo and hybrid echo. [Speech compression](#) techniques and [digital processing](#) delay often contribute to echo generation in [telephone networks](#).

H.323

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

HTTP

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

IP

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

IP-PBX

IP-based Private Branch Exchange

IP Telephony

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

IVR

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

MTU

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

NAT

Network Address Translation

NTP

Network Time Protocol, a protocol to exchange and synchronize time over networks

The port used is UDP 123

Grandstream products using NTP to get time from Internet

OBP/SBC

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

OBP/SBCs to allow the use of VoIP protocols from private networks with [internet](#) connections using [NAT](#).

PPPoE

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

PSTN

Public Switched Telephone Network

i.e. the phone service we use for every ordinary phone call, or called POT (Plain Old Telephone), or circuit switched network.

RTCP

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

RTP

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

SDP

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

SIP

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

All Grandstream products are SIP based

STUN

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

TCP

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

TFTP

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

UDP

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

VAD

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

VLAN

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

VoIP

Voice over IP

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