

User Manual

HandyTone-286

Analog Telephone Adaptor

Version 1.00

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

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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.



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.

Warning: Please do not attempt to use a different power adaptor. Using other power adaptor may damage the HandyTone ATA.

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

- Support SIP 2.0, TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, TFTP protocols
- Support IETF STUN.
- Interoperable with various 3rd party SIP end user device, Proxy / Registrar Server and gateway products.
- Advanced Digital Signal Processing (DSP) technology to ensure superior audio quality
- Advanced and patent pending adaptive jitter buffer control, packet delay and loss concealment technology
- Support popular vocoders including G.723.1 (5.3K/6.3K), G.729A/B, G.711 (alaw and u-law), G.726 (40K/32K/24K/16K), G.728, and iLBC.
- Support standard voice features such as Call Waiting, Forward, in-band and out-of-band DTMF, Dial Plans.
- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168), and AGC (Automatic Gain Control)
- Support DIGEST authentication (MD5, MD5-sess)
- Provide easy configuration through manual operation (voice prompt along with phone keypad and Web interface) or automated centralized configuration file.
- Support for Layer 2 (802.1Q VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
- Remote software upgrades capability via TFTP

4.2 Hardware Specification

The table below describes the difference among these models.

<u>Model</u>	<u>HandyTone 286</u>
LAN interface	1xRJ45 10Base-T
Button	1
LED	GREEN & RED color
Universal Switching Power Adaptor	Input: 100-240VAC Output: +5VDC, 1200mA, UL certified
Dimension	65mm (W) 93mm (D) 27mm (H)
Weight	
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 has stored a voice prompt menu for quick browsing and simple configuration. To enter this voice prompt menu, simply pick up the phone and 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
Dial '#' to Main Menu	“Enter a Menu Option”	<ul style="list-style-type: none"> ▪ Dial '*' to review the configuration of the device; or ▪ Dial 01 – 06, or 99 menu option; or ▪ Dial '#' to return to the Main Menu
Dial '01' to configure the DHCP Mode	“Static IP Mode”, or “Dynamic IP Mode”	<ul style="list-style-type: none"> ▪ Dial '9' to toggle the selection. <p><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></p>
Dial '02' to configure the IP address of the device	“IP Address” + IP address Such as “IP Address xxx.xxx.xxx.xxx”	<ul style="list-style-type: none"> ▪ Enter 12-digit new IP address if in Static IP Mode. for example the address is “192.168.0.10”, user should dial “192168000010”
Dial '03' to configure the subnet mask	“Subnet mask” + subnet mask	<ul style="list-style-type: none"> ▪ Enter 12-digit new subnet mask address if in Static IP Mode, for example the address is “255.255.255.0”, user should dial “255255255000”.
Dial '04' to configure the default gateway	“Gateway “ + IP address	Enter 12-digit IP address of the default gateway if in Static IP Mode.
Dial '05' to configure the DNS server	“DNS Server” + IP address	Enter 12-digit IP address of the DNS server if in Static IP Mode.
Dial '06' to configure the TFTP server	“TFTP Server “ + IP address	Enter 12-digit IP address of the TFTP server TFTP server is used to update the firmware of the device.
Dial '47' to make a direct IP call	“Direct IP Calling”	When dial '47', user will prompt a dial tone, then dial the 12-digit IP address. For Example, user wants call another SIP phone at “192.168.0.011”. User should dial '47' and dial '192168000011'. (For detail,

		see “4.2.2 Make a Direct IP Call”.)
Dial ‘86’ to check the voice message	“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.
Dial ‘99’ to reset the device	“RESET”	Dial ‘9’ to confirm the RESET; or Enter MAC address to restore factory default setting (For detail, see section 8)
Dial the invalid number or keypad	“Invalid Entry”	Automatically return to Main Menu

Notes:

- Once 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.
- 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

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

- 1) Dial the extension number directly and wait for 5 seconds.
- 2) Dial the numbers directly, and press # (Assuming that “use #” as dial key is selected in web configuration. For detail, see section 5.2 configure Handytone using web browser).

Other functions available during the call are call-waiting/flash, call-transfer, and call-forwarding.

5.2.2 Direct IP calls

Direct IP calling allows two phones to talk to each other in an ad hoc fashion without a SIP proxy. VoIP calls can be made between two phones, if:

- both HandyTone ATA and other VOIP Devices(i.e., another HandyTone ATA or other IP phone) have public IP addresses, or
- both HandyTone ATA and other VOIP Devices(i.e., another HandyTone ATA or other IP phone) are on a same LAN using private or public IP addresses, or
- both HandyTone ATA and other VOIP Devices(i.e., another HandyTone ATA or other IP phone) can be connected through a router using public or private IP addresses.

To make a direct IP calling, 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 HT286, and dials “47” to access the direct IP call menu. User will hear a voice prompt “Direct IP Calling” and a dial tone. Then user dials the 12-digit target IP address to make a call.

If there is a user name part in the target address, such as username@192.168.10.123, encode the user name part (see the following encoding scheme table), followed by *3 (encoding for “@”) and then followed by the 12-digit target IP address. Destination ports can also be specified using *4 (encoding for “:”) followed by the encoded port number.

The follow is a table of the encoding scheme for the most commonly used characters:

F	
00	0
01	1
02	2
03	3
04	4
05	5
06	6
07	7
08	8
09	9
*0	. (dot character)
*1	_ (underscore character)
*2	- (hyphen character)
*3	@
*4	: (column character)
21	A
22	B
23	C
31	D
32	E
33	F
41	G
42	H
43	I
51	J
52	K
53	L
61	M
62	N
63	O
71	P
72	Q
73	R
74	S
81	T
82	U
83	V
91	W

92	X
93	Y
94	Z

The rule of thumb to remember these encoding is: “a” is the first letter on button “1” so its encoding is “11”. “b” is the 2nd letter on button “1” and its encoding is “12”. “c” is the 3rd letter on button “1” and its encoding is “13”. Likewise, “d” is the first letter on button “2” and its encoding is “21”. This pattern and rule applies to all other alphabetic encoding.

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 is it is configured as a send key or wait for more than 5 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 is it is configured as a send key or wait for 5 seconds.

If the target address is john@192.168.1.100:5062, then the dialing convention would be:

Voice Prompt with option 47, then 51634262*3192168001100*45062

followed by pressing the “#” key is it is configured as a send key or wait for 5 seconds.

5.3 Call Features

Following table shows the call features of HandyTone-286.

Dial Key	Call Features
*70	Do not disturb, it will give caller busy tone when called. This setting will be released when user hangs up the phone.
*72	Unconditional Call Forward. To use this feature, dial “*72” and get the dial tone. Then dial the forward number and 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 hang up.
*91	Cancel Busy Call Forward To cancel “Unconditional 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 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 LED Light Pattern Indication

Following are the LED light pattern indication.

RED LED always indicates not normal status	
DHCP Failed or WAN No Cable	flash every 2 seconds (if DHCP is configured)
HandyTone-486 fails to register	flash every 2 seconds (if SIP server is configured)
GREEN LED always for normal working status	
Message Waiting Indication	Button flashes every 2 seconds
RINGING	Button flashes at 1/10 second
RINGING INTERVAL	Button flashes every second

6 Configuration Guide

6.1 Configuring HandyTone IP through Voice Prompt

6.1.1 DHCP Mode

Follow section 4.1 with voice menu option 01 to enable HandyTone to use DHCP.

6.1.2 STATIC IP Mode

Follow section 4.1 with voice menu option 01 to enable HandyTone 286 to use STATIC IP mode, then use option 02, 03, 04 to set up HandyTone's IP, Subnet Mask, Gateway respectively.

6.1.3 TFTP Server Address

Follow section 4.1 with voice menu option 06 to configure the IP address of the TFTP server.

6.2 Configuring HandyTone with Web Browser

HandyTone 200 series ATA 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 IP phone 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 through section 4.1 with menu option 02. Then access the HandyTone's Web Configuration Menu using the following URI:

<http://Phone-IP-Address>,

where the *Phone-IP-Address* is the IP address of the phone.

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 and the factory default password is lower case ‘*admin*’.

6.2.2 Configuration Menu

After the correct password is entered in the login screen, the embedded Web server inside the IP phone will respond with the Configuration Menu screen, which is explained in details below.

The definitions for all the configuration parameters in the Configuration Menu are:

<i>Information Field</i>	This field shows basic information of the device, including the MAC address, the model, and the software version.
<i>Password</i>	This contains the password to access the Web Configuration Menu. This field is case sensitive.
<i>IP Address</i>	<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 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. - 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 reset to zero by default.
<i>SIP Server</i>	This field contains the URI string or the IP address (and port, if different from 5060) of the SIP proxy server. e.g., the following are some valid examples: sip.my-voip-provider.com, or sip:my-company-sip-server.com, or 192.168.1.200:5066

Outbound Proxy	This field contains the URI string or the IP address (and port, if different from 5060) of the outbound proxy. If there is no outbound proxy, this field SHOULD be left blank. If 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.
SIP User ID is Phone Number	If the IP phone 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 Login ID	This field contains the login ID used for SIP authentication. Typically, this is the account number on an SIP server for this IP phone. It can be the same as or different from the above SIP User ID, depending on whether a separate account ID is used for authentication.
SIP Password	This field contains the password used for SIP authentication. It is used together with the above SIP Login ID

Grandstream HandyTone 286 Configuration

MAC Address: 00.0B.82.00.C5.BA
Product Model: HT286
Software Version: Program--1.0.5.3 Bootloader--1.0.0.18 HTML--1.0.0.36 VOC--1.0.0.6

Admin Password: (password to configure this IP phone)

IP Address: dynamically assigned via DHCP (default) or PPPoE
 (will attempt PPPoE if DHCP fails and following is non-blank)

PPPoE account ID:
 PPPoE password:

Use this DNS server (if specified): . . .

statically configured as:

IP Address:	<input type="text" value="192"/>	<input type="text" value="168"/>	<input type="text" value="0"/>	<input type="text" value="160"/>
Subnet Mask:	<input type="text" value="255"/>	<input type="text" value="255"/>	<input type="text" value="255"/>	<input type="text" value="0"/>
Default Router:	<input type="text" value="192"/>	<input type="text" value="168"/>	<input type="text" value="0"/>	<input type="text" value="1"/>
DNS Server 1:	<input type="text" value="202"/>	<input type="text" value="96"/>	<input type="text" value="134"/>	<input type="text" value="133"/>
DNS Server 2:	<input type="text" value="202"/>	<input type="text" value="96"/>	<input type="text" value="128"/>	<input type="text" value="68"/>

SIP Server: (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="1234"/>	(the user part of an SIP address)
Authenticate ID:	<input type="text" value="1234"/>	(can be identical to or different from SIP User ID)
Authenticate Password:	<input type="password" value="*****"/>	
Name:	<input type="text"/>	(optional, e.g., John Doe)
Advanced Options:		
Preferred Vocoder: (in listed order)	choice 1: <input type="text" value="PCMU"/>	
	choice 2: <input type="text" value="PCMA"/>	
	choice 3: <input type="text" value="G.723.1"/>	
	choice 4: <input type="text" value="G.729A/B"/>	
	choice 5: <input type="text" value="iLBC"/>	
	choice 6: <input type="text" value="G.726-32"/>	
	choice 7: <input type="text" value="G.728"/>	
G723 rate:	<input checked="" type="radio"/> 6.3kbps encoding rate <input type="radio"/> 5.3kbps encoding rate	
iLBC frame size:	<input checked="" type="radio"/> 20ms <input type="radio"/> 30ms	
iLBC payload type:	<input type="text" value="98"/>	(between 96 and 127, default is 98)
Silence Suppression:	<input checked="" type="radio"/> No <input type="radio"/> Yes	
Voice Frames per TX:	<input type="text" value="2"/>	(up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
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)
Use DNS SRV:	<input checked="" type="radio"/> No <input type="radio"/> Yes	
User ID is phone number:	<input checked="" type="radio"/> No <input type="radio"/> Yes	
SIP Registration:	<input checked="" type="radio"/> Yes <input type="radio"/> No	
Unregister On Reboot:	<input checked="" type="radio"/> Yes <input type="radio"/> No	
Register Expiration:	<input type="text" value="60"/>	(in minutes, default 1 hour, max 45 days)
Early Dial:	<input checked="" type="radio"/> No <input type="radio"/> Yes (use "Yes" only if proxy supports 484 response)	
Dial Plan Prefix:	<input type="text"/>	(this prefix string is added to each dialed number)
No Key Entry Timeout:	<input type="text" value="4"/>	(in seconds, default is 4 seconds)
Use # as Dial Key:	<input type="radio"/> No <input checked="" type="radio"/> Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)	
local SIP port:	<input type="text" value="5060"/>	(default 5060)
local RTP port:	<input type="text" value="5004"/>	(1024-65535, default 5004)
Use random port:	<input type="radio"/> No <input checked="" type="radio"/> Yes	
NAT Traversal:	<input checked="" type="radio"/> No <input type="radio"/> Yes, STUN server is: <input type="text"/> (URI or IP:port)	
keep-alive interval:	<input type="text" value="30"/>	(in seconds, default 20 seconds)
Use NAT IP	<input type="text"/>	(if specified, this IP address is used in SIP/SDP message)
TFTP Server:	<input type="text" value="4"/> . <input type="text" value="3"/> . <input type="text" value="153"/> . <input type="text" value="50"/>	(for remote software upgrade and configuration)
SUBSCRIBE for MWI:	<input checked="" type="radio"/> No, do not send SUBSCRIBE for Message Waiting Indication <input type="radio"/> Yes, send periodical SUBSCRIBE for Message Waiting Indication	
Offhook Auto-Dial:	<input type="text"/>	(User ID/extension to dial automatically when offhook)
Enable Call Features:	<input type="radio"/> No <input checked="" type="radio"/> Yes (if Yes, Call Forwarding & Do-Not-Disturb are supported locally)	
Disable Call-Waiting:	<input checked="" type="radio"/> No <input type="radio"/> Yes	
Send DTMF:	<input checked="" type="radio"/> in-audio <input type="radio"/> via RTP (RFC2833) <input type="radio"/> via SIP INFO	
DTMF Payload Type:	<input type="text" value="101"/>	
Send Flash Event:	<input checked="" type="radio"/> No <input type="radio"/> Yes (Flash will be sent as a DTMF event if set to Yes)	
FXS Impedance:	<input type="text" value="current setting is '600 Ohm (North America)"/>	
NTP Server:	<input type="text" value="ntp01.sipphone.com"/>	(URI or IP address)
Time Zone:	<input type="text" value="current setting is 'GMT-5:00 (US Eastern Time, New York)"/>	
Daylight Savings Time:	<input checked="" type="radio"/> No <input type="radio"/> Yes (if set to Yes, display time will be 1 hour ahead of normal time)	
Send Anonymous:	<input checked="" type="radio"/> No <input type="radio"/> Yes (caller ID will be blocked if set to Yes)	
Lock keypad update:	<input checked="" type="radio"/> No <input type="radio"/> Yes (configuration update via keypad is disabled if set to Yes)	
<input type="button" value="Update"/> <input type="button" value="Cancel"/> <input type="button" value="Reboot"/>		

<i>Preferred Vocoder</i>	<p>The BudgeTone IP phone supports up to 7 different vocoder types including G711-ulaw (PCMU), G711-alaw (PCMA), G723, G729A/B, G726-32 (ADPCM), G728, and iLBC. Depending on the product model, some of these vocoders may not be provided in standard release.</p> <p>A user can configure vocoders in a preference list that will be included with the same preference order in SDP message. The first vocoder in this list can be entered by choosing the appropriate option in “Choice 1”. Similarly, the last vocoder in this list can be entered by choosing the appropriate option in “Choice 7”.</p>
<i>G723 Rate:</i>	This defines the encoding rate for G723 vocoder. By default, 6.3kbps rate is chosen.
<i>iLBC frame size</i>	This defines the size of the iLBC codec frame. The default setting is 20ms.
<i>iLBC payload type</i>	This defines the iLBC payload type. The default setting is 96.
<i>Silence Suppression</i>	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.

<i>Voice Frames per TX</i>	<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>
<i>Layer 3 QoS</i>	<p>This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv. Default value is 48</p>
<i>Layer 2 QoS</i>	<p>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.</p>
<i>Use DNS SRV</i>	<p>This parameter controls whether the IP phone supports the DNS SRV route function.</p>
<i>User ID is phone number</i>	<p>If the HandyTone ATA 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.</p>
<i>SIP Registration</i>	<p>This parameter controls whether the IP phone needs to send REGISTER messages to the proxy server. The default setting is “Yes”.</p>
<i>Unregister On Reboot</i>	<p>Default is No. If set to yes, the SIP user will be unregistered on reboot.</p>

<i>Registration Expiration</i>	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 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
<i>Early Dial</i>	<p>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.</p> <p>The “Yes” option should be used ONLY if there is a SIP proxy configured and the proxy server supports 484 Incomplete Address response. Otherwise, the call will most likely be rejected by the proxy (with a 404 Not Found error).</p> <p>Please note that this feature is NOT designed to work with and should NOT be enabled for direct IP-to-IP calling.</p>
<i>Dial Plan Prefix</i>	This value contains the dial plan prefix string (typically an ASCII numeric string). If it is not blank, then this string will be used as a prefix to the target URI string in the “To” header field of an INVITE message.
<i>Use # as Send Key</i>	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.
<i>Local SIP port</i>	This parameter defines the local SIP port the IP phone will listen and transmit on. The default value is 5060.
<i>Local RTP port</i>	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.
<i>Use Random Port</i>	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.

<i>NAT Traversal</i>	<p>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 through communication with the specified STUN server. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the phone will attempt to use its mapped public IP address and port in all the SIP and SDP messages it sends out.</p> <p>If this field is set to “Yes” with no specified STUN server, then the phone will periodically (every 10 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the “hole” on the NAT open.</p>
<i>keep-alive interval</i>	<p>The HT286 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.</p>
<i>Use NAT IP</i>	<p>NAT IP address used in SIP/SDP message. Default is blank.</p>
<i>TFTP Server</i>	<p>This is the IP address of the configured tftp server. If it is non-zero or not blank, the IP phone will attempt to retrieve new configuration file or new code image (update) from the specified tftp server at boot time. It will make up to 3 attempts before timeout and then it will start the boot process using the existing code image in the Flash memory. If a tftp server is configured and a new code image is retrieved, the new downloaded image will be verified and then saved into the Flash memory.</p>
<i>SUBSCRIBE for MWI</i>	<p>Default is NO. When set to Yes a SUBSCRIBE for Message Waiting Indication will be sent periodically</p>
<i>Offhook Auto-Dial</i>	<p>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.</p>
<i>Enable Call Feature</i>	<p>Default is No. If set to Yes, Call Forwarding & Do-Not-Disturb are supported locally.</p>

<i>Disable Call Waiting</i>	Default is Not.
<i>Send DTMF</i>	This parameter controls the way DTMF events are transmitted. There are 3 ways: 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.
<i>Send Flash Event</i>	This parameter allows the user to control whether to send an SIP NOTIFY message indicating the Flash event, or just to switch to the voice channel when the user presses the Flash key.
<i>NTP server</i>	This parameter defines the URI or IP address of the NTP server which the IP phone will use to display the current date/time.
<i>Time Zone</i>	This parameter controls how the displayed date/time will be adjusted according to the specified time zone.
<i>Daylight Savings Time</i>	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.
<i>Send Anonymous</i>	If this parameter is set to “Yes”, the “From” header in outgoing INVITE message will be set to anonymous, essentially blocking the Caller ID from displaying.
<i>Lock keypad update</i>	If this parameter is set to “Yes”, the configuration update via keypad is disable.

6.2.3 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.

Grandstream IP Phone Configuration Update Status

Your configuration changes have been saved.
They will take effect on next reboot.

[Back to Home Page](#)

6.2.4 Rebooting the phone from remotely

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.

Grandstream IP Phone Rebooting Status

The IP phone is rebooting now...
You may relogin by clicking on the link below in 30 seconds.

[Click to relogin](#)

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

*The user is recommended to power cycle HT286 after seeing the above message.

6.3 Configuration through a Central Server

Grandstream IP phones can be automatically configured via a central provisioning system called Grandstream Automated Provisioning System (GAPS).

With GAPS, a service provider or an enterprise with large deployment of IP phones can easily manage the configuration and service provisioning of individual devices remotely and automatically via a central server. GAPS uses enhanced (NAT friendly) tftp and

other communication protocols to communicate with each individual IP phone even if the phone is behind a NAT.

GAPS must be used to support automated configuration of an IP phone. To enable this feature on the phone, the user just needs to enter the IP address of GAPS server in the tftp server field of the configuration screen. Then power cycle the phone.

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

7 Software Upgrade with TFTP

To upgrade software, HandyTone ATAs can be configured with a TFTP server where the new code image is located. The TFTP upgrade can work in either static IP or DHCP mode using private or public IP address. It is recommended that the TFTP server must have either public IP address or be on the same LAN with the HandyTone ATA.

There are 2 ways to set up the TFTP server to upgrade the firmware, namely through voice menu prompt or via the HandyTone ATA's Web configuration interface. To configure the TFTP server via voice prompt, follow section 4.1 with option 06, once set up the tftp ip address, power cycle the ATA, the firmware will be fetched once the ATA boot up.

To configure the TFTP server via the Web configuration interface, open up your browser to point at the IP address of the HandyTone ATA. Input the admin password to enter the configuration screen. From there, enter the TFTP server address in the designated field towards the bottom of the configuration screen.

Once the TFTP server is configured, power cycle the HandyTone ATA.

TFTP checking is only performed during the initial power up. If the configured tftp 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 fails for any reason (e.g., TFTP 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 process and simply boot using the existing code image in the flash.

TFTP may take as long as 1—2 minutes over Internet, or just 20+ seconds if it is performed on a LAN. It is generally recommended to conduct TFTP upgrade in a controlled LAN environment if possible. For users who do not have local TFTP server, Grandstream provides a NAT-friendly TFTP server on the public Internet for users to download the latest firmware upgrade automatically. Please check the Service or Support section of Grandstream's Web site to obtain this TFTP server IP address.

8 Restore Factory Default Setting

Warning: Restore the Factory Default Setting will delete all configuration information of the device.

Step one: Find the Mac Address of the device. The Mac address of the device is located on the bottom of the device. It is a 12 digit number.

Step two: Encode the Mac address. The encode rule is:

“2” is the first letter on the button “2” so its encoding is “2”.

“A” is the second letter on button “2” so its encoding is “22”.

“B” is the third letter on button “2” and its encoding is “222”.

“C” is the fourth letter on button “2” and its encoding is “2222”.

For example, the Mac address is 000b8200e395,
User should encode it as “0002228200333395”.

Step three: Access the voice menu, then dial “99” and get the voice prompt “RESET”

Step four: Dial in the encode of the Mac address. Once the correct encode Mac address dial in, the device will reboot automatically and restore the factory default setting.