User Manual

HandyTone-286

Analog Telephone Adaptor

For Firmware Version 1.0.6.7

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-wining 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.

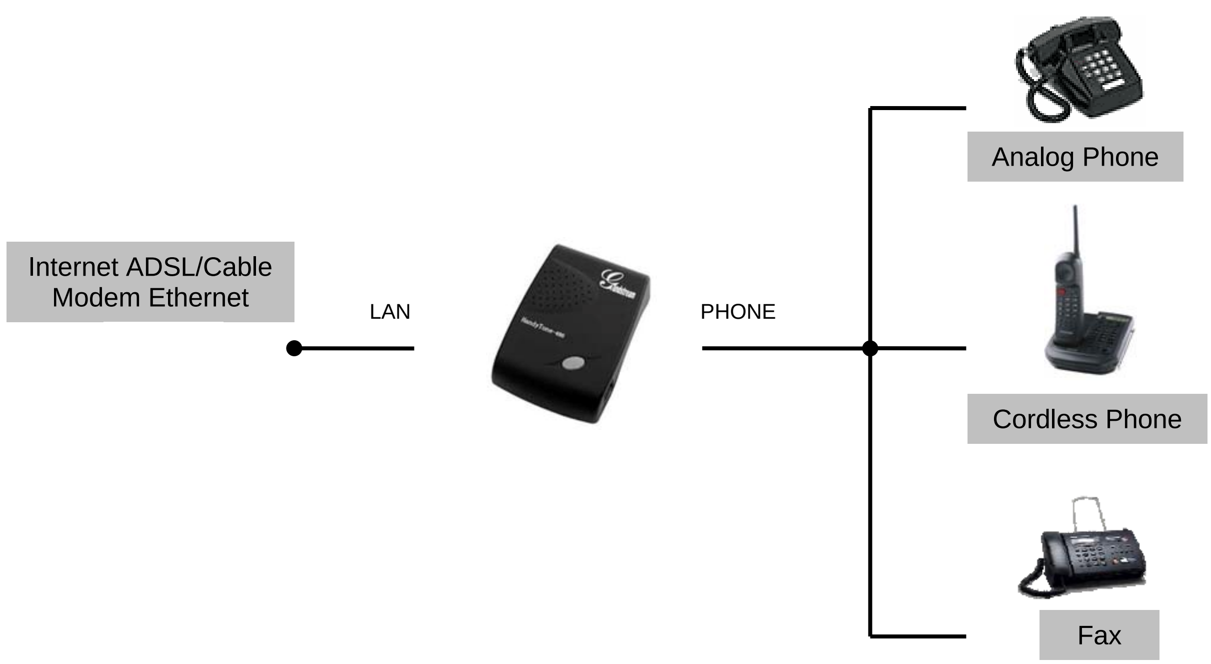
This document is subject to changes without notice. The latest electronic version of this user manual can be downloaded from the following location:

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 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

- 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.
- 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 (40K/32K/24K/16K), as well as G.728, G.729 and iLBC.
- Support Caller ID/name display or block, Call waiting caller ID, Hold, Call Waiting/Flash, Call Transfer, Call Forward, in-band and out-of-band DTMF, Dial Plans, 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 Central configuration files through TFTP or HTTP server
- Support firmware upgrade via TFTP or HTTP with encrypted configuration files.
- Support PSTN pass through, be able to make and receive VoIP or PSTN calls using same connected analogue phone.
- Ultra compact (wallet size) and lightweight design, great companion for travelers.
- Compact, lightweight Universal Power adapter

4.2 Hardware Specification

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

Model	**HandyTone-286**
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	
Operating Temperature	32 - 104^{o}F 0 - 40^{o}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, simple 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
Main Menu	"Enter a Menu Option"	Enter '*' to next option and "#" back to main menu, or Dial 01 – 06, 47, 86 or 99 Menu option
01	"Static IP Mode", or "Dynamic IP Mode"	Dial '9' to toggle the selection. *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.*
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 TFTP server is used to update the firmware of the device.
47	"Direct IP Calling"	When entered, user will be prompted a dial tone, dial the 12-digit IP address to make a direct IP call. (For detail, 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"	Dial '9' to confirm the RESET; or Enter MAC address to restore factory default setting (For detail, 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

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

1. Dial the extension number directly and wait for 4 seconds. (Default "No Key Entry Timeout"). Or
2. Dial the number directly, and press # (assuming that "Use # as dial key" is selected in web configuration).

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 the other VoIP device (i.e., another HandyTone ATA or other SIP products) have public IP addresses, or
- Both HandyTone ATA and the other VoIP device (i.e., another HandyTone ATA or other SIP produces) are on the same LAN using private or public IP addresses, or
- Both HandyTone ATA and the other VoIP device (i.e., another HandyTone ATA or other SIP products) can be connected through a router using public or private IP addresses.

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 HT286, 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.

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

Input	Encoding
00	0
01	1
02	2
03	3
04	4
05	5
06	6
07	7
08	8
09	9
*0	. (dot character)
*4	: (column character)

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 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: *Call Feature* has to be set to YES.

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.

5.2.4 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). A and C now in conversation.
3. A can hang up.

Note:

- When intended Transfer failed, if A hangs up, the HandTone-496 will ring user A again to remind A that B is still on the call, by pressing FLASH or Hook again will restore the conversation between A and B.

5.3 Call Features

Following table shows the call features of HandyTone-286.

Key	Call Features
*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 Support

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 indicates abnormal 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 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

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 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 to set up HandyTone-286's IP, Subnet Mask, Gateway 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 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 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

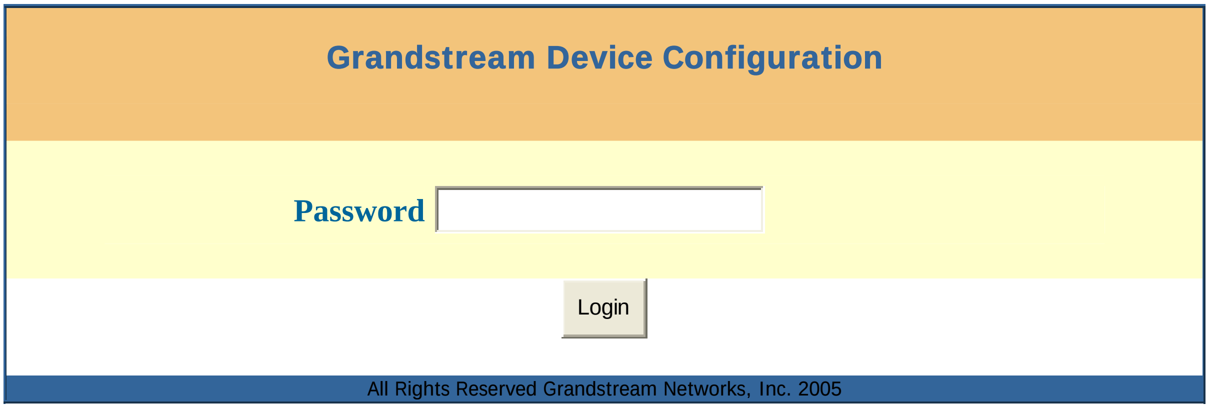
First, get the IP address of the HandyTone-286 through section 5.1 with menu option 02. Then access the HandyTone-286's Web Configuration Menu using the following URI:

http://Phone-IP-Address

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

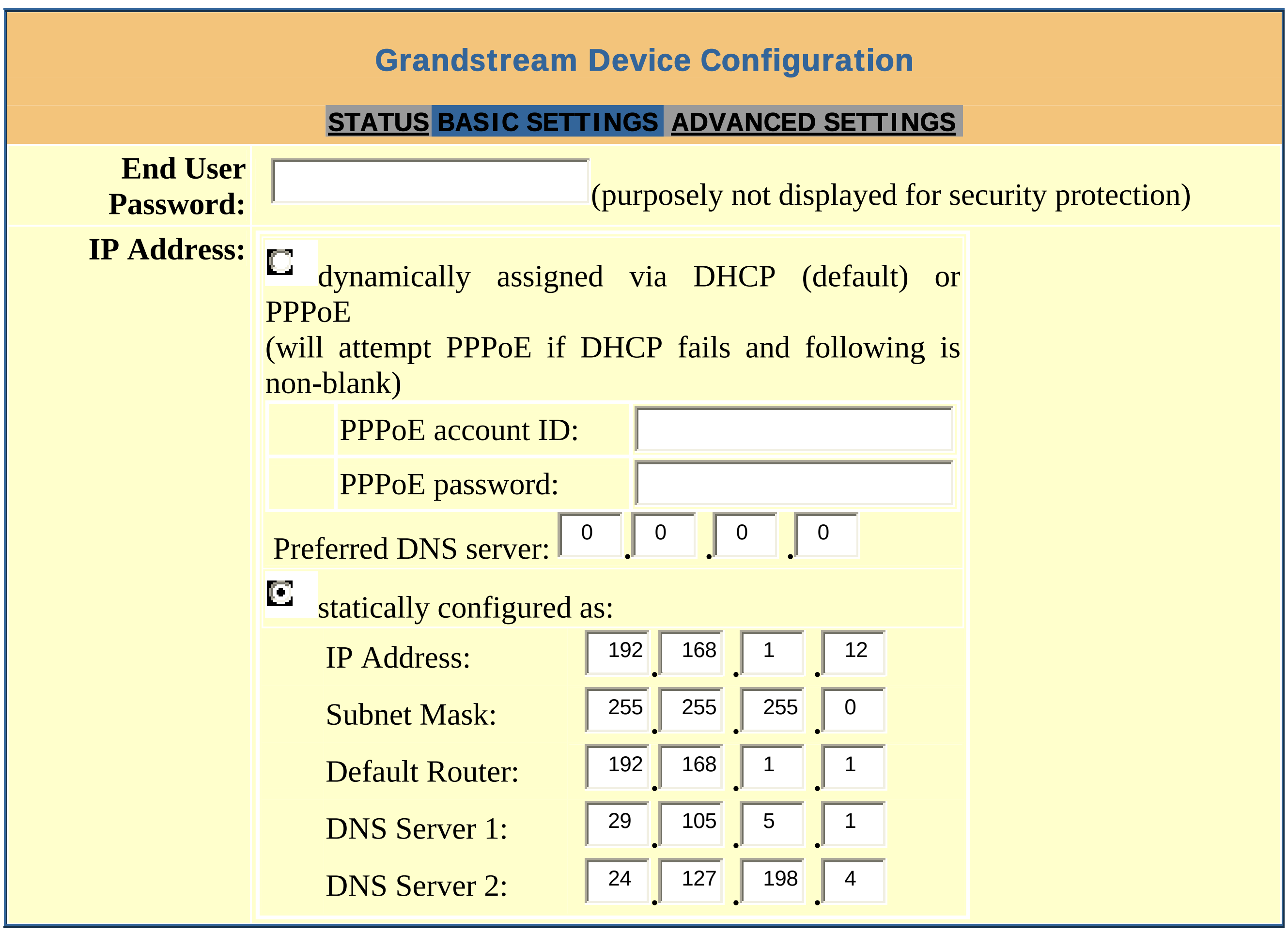
6.2.2 End User Configuration

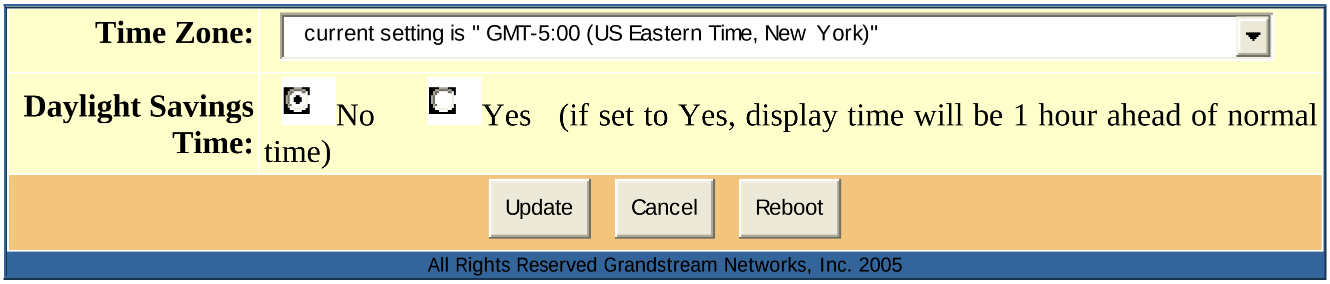
Once this request is entered and sent from a Web browser, the IP phone will respond with the following login screen:



The password is case sensitive with a maximum length of 25 characters. The factory default password for End User is "123" or blank.

After the correct password is entered in the login screen, the embedded Web server inside the IP phone will respond with the following Basic Settings configuration page, which is explained in details below.





End User Password	This contains the password to access the Web Configuration Menu. This field is case sensitive with max. 25 characters
IP Address	There are 2 modes under which the IP phone can operate: - If DHCP mode is enabled, then all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory) and the IP phone will acquire its IP address from the first DHCP server it discovers on the LAN it attaches to. To use PPPoE feature please set the PPPoE account settings if the 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. In this mode, the WAN side web access is disabled and TFTP upgrade for firmware is not feasible and HTTP upgrade is the only available solution. - 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.
Time Zone	This parameter controls how date/time will be 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.

In addition to the Basic Settings configuration page, end user also has access to the device Status page. The following is a screen shot of the device Status page. Details are explained next.

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS

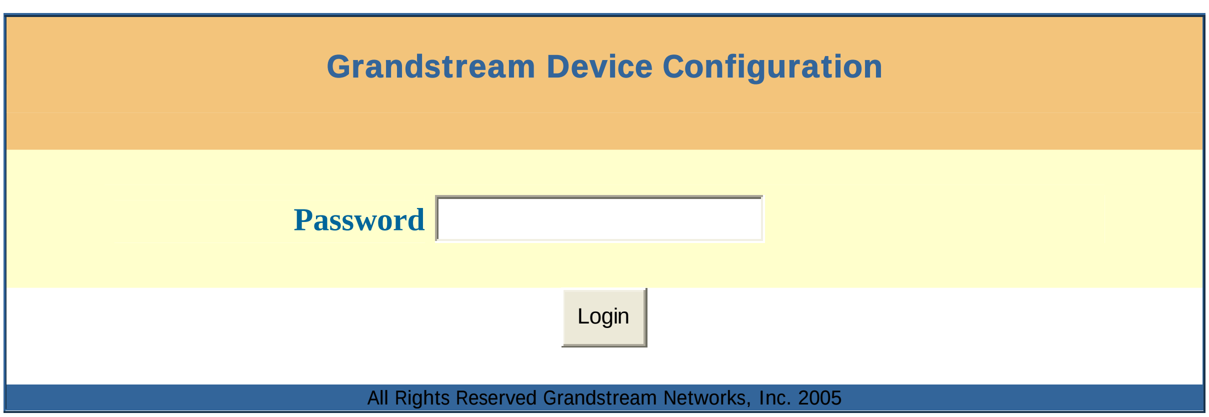
MAC Address:	00.0B.82.01.56.4D
WAN IP Address:	192.168.1.12
Product Model:	HT286
Software Version:	Program-- 1.0.6.7 Bootloader-- 1.0.1.0 HTML-- 1.0.0.49 VOC-- 1.0.0.10
System Up Time:	0 day(s) 0 hour(s) 4 minute(s)
Registered:	Yes
PPPoE Link Up:	disabled
NAT:	detected NAT type is full cone
NAT Mapped IP:	67.153.142.35
NAT Mapped Port:	5060
Total Inbound Calls:	0
Total Outbound Calls:	0
Total Missed Calls:	0
Total Call Time (in minutes):	0
Total SIP Message Sent:	5
Total SIP Message Received:	5
Total RTP Packet Sent:	0
Total RTP Packet Received:	0
Total RTP Packet Loss:	0

MAC Address	The device ID, in HEX format. This is very important ID for ISP troubleshooting.
WAN IP Address	This field shows WAN port IP address.
Product Model	This field contains the product model info.

Software Version	***Program:*** This is the main software release. This number is always used for firmware upgrade. ***Bootloader:*** This is normally not changed. ***HTML:*** This is the user interface, normally not changed. ***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 service provider's server.
PPPoE Link Up	This shows whether the PPPoE is up if connected to DSL modem
NAT	This shows what kind NAT the HandyTone-286 is connected to via its WAN port. It is based on STUN protocol.
NAT Mapped IP	WAN side public IP if connected to LAN of a SOHO router.
NAT Mapped Port	External port detected by STUN.
Statistical Status	Self explainable. Please refer to the page displayed.

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".



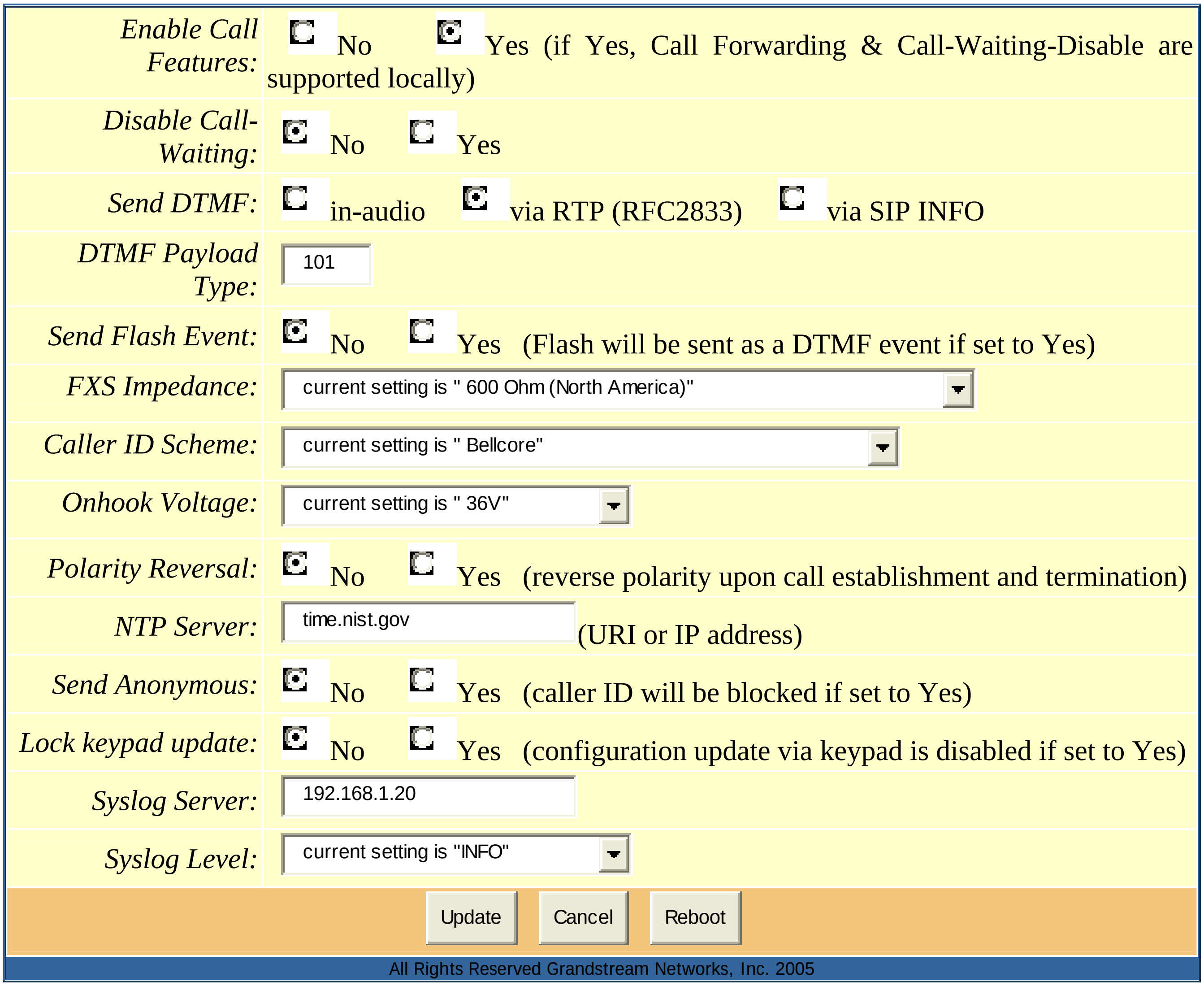
Advanced User configuration page includes not only the end user configuration, but also some advanced configuration such as SIP configuration, Codec selection, NAT Traversal Setting and other miscellaneous configurations. Following is the screen shot of the Advanced configuration page.

Grandstream Device Configuration

STATUS BASIC SETTINGS **ADVANCED SETTINGS**

Admin Password:	(purposely not displayed for security protection)
SIP Server:	sip.mycompany.com (e.g., sip.mycompany.com, or IP address)
Outbound Proxy:	(e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID:	3125250 (the user part of an SIP address)
Authenticate ID:	3125250 (can be identical to or different from **SIP User ID**)
Authenticate Password:	(purposely not displayed for security protection)
Name:	(optional, e.g., John Doe)
Advanced Options:	
Preferred Vocoder: (in listed order)	choice 1: current setting is " PCMU" choice 2: current setting is " PCMA" choice 3: current setting is " G723" choice 4: current setting is " G729" choice 5: current setting is " G726-32" choice 6: current setting is " G728" choice 7: current setting is " iLBC"
G723 rate:	6.3kbps encoding rate 5.3kbps encoding rate
iLBC frame size:	20ms 30ms
iLBC payload type:	97 (between 96 and 127, default is 97)
Silence Suppression:	No Yes
Voice Frames per TX:	2 (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
Fax Mode:	T.38 (Auto Detect) Pass-Through
Layer 3 QoS:	48 (Diff-Serv or Precedence value)

Layer 2 QoS:	802.1Q/VLAN Tag [0] 802.1p priority value [0] (0-7)
Use DNS SRV:	(•) No ( ) Yes
User ID is phone number:	(•) No ( ) Yes
SIP Registration:	(•) Yes ( ) No
Unregister On Reboot:	( ) Yes (•) No
Register Expiration:	[60] (in minutes. default 1 hour, max 45 days)
Early Dial:	(•) No ( ) Yes (use "Yes" only if proxy supports 484 response)
Dial Plan Prefix:	[ ] (this prefix string is added to each dialed number)
No Key Entry Timeout:	[4] (in seconds, default is 4 seconds)
Use # as Dial Key:	( ) No (•) Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
local SIP port:	[5060] (default 5060)
local RTP port:	[5004] (1024-65535, default 5004)
Use random port:	(•) No ( ) Yes
NAT Traversal:	( ) No (•) Yes, STUN server is: [stun.mycompany.com] (URI or IP:port)
keep-alive interval:	[20] (in seconds, default 20 seconds)
Use NAT IP	[ ] (if specified, this IP address is used in SIP/SDP message)
Proxy-Require:	[ ] (if specified, the content will appear in Proxy-Require header)
Firmware Upgrade:	( ) Via TFTP Server [192].[168].[1].[30] (•) Via HTTP Server [192.168.1.20] Automatic HTTP Upgrade: (•) No ( ) Yes, check for upgrade every [7] days (default 7 days)
SUBSCRIBE for MWI:	(•) No, do not send SUBSCRIBE for Message Waiting Indication ( ) Yes, send periodical SUBSCRIBE for Message Waiting Indication
Offhook Auto-Dial:	[ ] (User ID/extension to dial automatically when offhook)



Enable Call Features:	No Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally)
Disable Call-Waiting:	No Yes
Send DTMF:	in-audio via RTP (RFC2833) via SIP INFO
DTMF Payload Type:	101
Send Flash Event:	No Yes (Flash will be sent as a DTMF event if set to Yes)
FXS Impedance:	current setting is " 600 Ohm (North America)"
Caller ID Scheme:	current setting is " Bellcore"
Onhook Voltage:	current setting is " 36V"
Polarity Reversal:	No Yes (reverse polarity upon call establishment and termination)
NTP Server:	time.nist.gov (URI or IP address)
Send Anonymous:	No Yes (caller ID will be blocked if set to Yes)
Lock keypad update:	No Yes (configuration update via keypad is disabled if set to Yes)
Syslog Server:	192.168.1.20
Syslog Level:	current setting is "INFO"
	Update Cancel Reboot

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 (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	User account information, provided by VoIP service provider (ITSP), usually has the form of digit similar to phone number or actually a phone number.
Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
Authenticate Password	SIP service subscriber's account password for GXP-2000 to register to (SIP) servers of ITSP.
Name	SIP service subscriber's name which will be used for Caller ID display.
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.
Preferred Vocoder	HandyTone-286 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. 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".
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.
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.
Use DNS SRV	This parameter controls whether the IP phone supports the DNS SRV route function.

Voice Frames per TX	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. 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. 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.
Fax Mode	T.38 (Auto Detect) FoIP by default, or Pass-Through (must use codec PCMU/PCMA)
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 SIP user's registration information will be cleared on reboot.
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 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 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 be used as a prefix to the target URI string in the “To” header field of an INVITE message.
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.
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 used in SIP/SDP message. Default is blank.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
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 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. If this field is set to “Yes” with no specified STUN server, then the phone will periodically (every 20 seconds by default) send a blank UDP packet (with no payload data) to the SIP server to keep the “hole” on the NAT open.
Firmware Upgrade	This radio button will enable HandyTone-286 to download firmware or configuration file through either TFTP or HTTP.

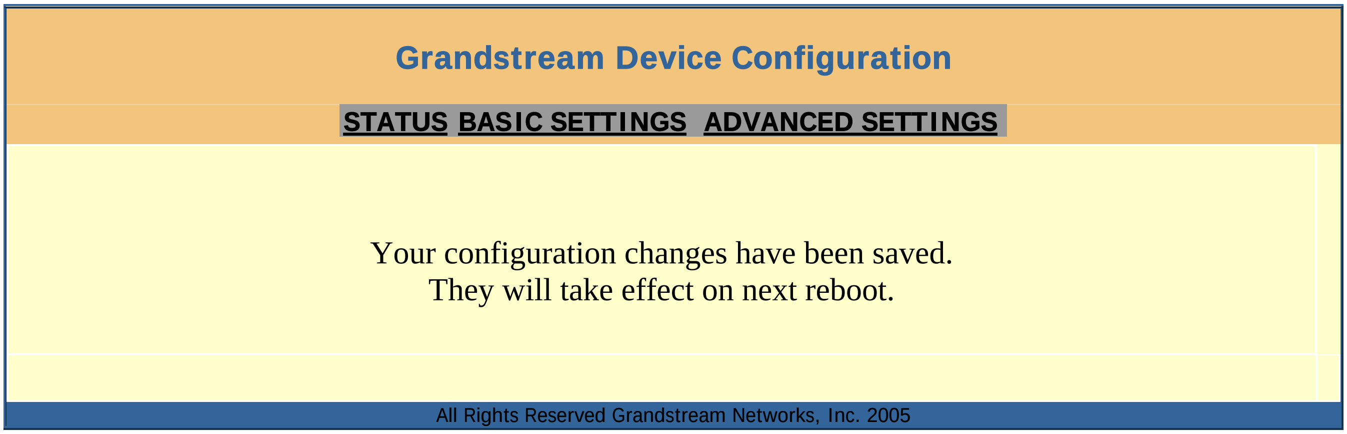
Via TFTP Server	This is the IP address of the configured tftp server. If it is non-zero or not blank, the IP phone will attempt to retrieve new configuration file or new code image (update) from the specified tftp server at boot time. It will make up to 3 attempts before timeout and then it will start 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. Note: Please do NOT interrupt the TFTP upgrade process (especially the power supply) as this will damage the device. Depending on the network environment this process can take up to 15 or 20 minutes.
Via HTTP Server	The URL for the HTTP server used for firmware upgrade and configuration via HTTP. For example, http://provisioning.mycompany.com:6688/Grandstream/1.0.5.16 Here ":6688" is the specific TCP port that the HTTP server is listening at, it can be omitted if using default port 80. Note: If Auto Upgrade is set to "No", HandyTone-286 will only do HTTP download once at boot up.
Automatic HTTP Upgrade	Choose "Yes" to enable automatic HTTP upgrade and provisioning. In "Check for new firmware every" field, enter the number of days to enable HandyTone-286 to check the HTTP server for firmware upgrade or configuration in the defined period of days. When set to "No", HandyTone-286 will only do HTTP upgrade once at boot up.
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 No. If set to Yes, Call Forwarding & Do-Not-Disturb are supported locally.
Disable Call Waiting	Default is No.
Send DTMF	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.
DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833

Send Flash Event	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.
FXS Impedance	Selects the impedance of the analog telephone connected to the Phone port.
Caller ID Scheme	Select the Caller ID Scheme to suit the standard of different area. • Bellcore (North America) • ETSI-FSK (France, Germany, Norway, Taiwan, UK-CCA) • ETSI-DTMF (Finland, Sweden) • DTMF (Denmark)
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 “From” header in outgoing INVITE message will be set to anonymous, essentially blocking the Caller ID from displaying.
Lock keypad update	If this parameter is set to “Yes”, the configuration update via keypad is disabled.
Syslog Server	The IP address or URL of System log server. This feature is especially useful for ITSP (Internet Telephone Service Provider)

Syslog Level	Select the ATA to report the log level. Default is NONE. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: • product model/version on boot up (INFO level) • NAT related info (INFO level) • sent or received SIP message (DEBUG level) • SIP message summary (INFO level) • inbound and outbound calls (INFO level) • registration status change (INFO level) • negotiated codec (INFO level) • Ethernet link up (INFO level) • SLIC chip exception (WARNING and ERROR levels) • memory exception (ERROR level) The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: GS_LOG: [device MAC address][error code] error message Here is an example: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up

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-488 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-286 can be automatically configured from a central provisioning system.

When HandyTone-286 boots up, it will send TFTP or HTTP request to download configuration files, there are two configuration files, one is "cfg.txt" and the other is "cfg000b82xxxxxx", where

"000b82xxxxxx" is the MAC address of the HandyTone-286. For more information regarding configuration file format, please refer to the related Grandstream documentation.

The configuration files can be downloaded via TFTP or HTTP from the central server. A service provider or an enterprise with large deployment of HandyTone-286 can easily manage the configuration and service provisioning of individual devices remotely and automatically from a central server. 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-286 for firmware upgrade, remote reboot, etc.

Grandstream provides a licensed provisioning system called GAPS that can be used to support automated configuration of HandyTone-286. To enable this feature on the HandyTone-286, a user just needs to enter the IP address of the GAPS server in the TFTP server field of the configuration screen, or enter the HTTP provisioning Server URL in the HTTP Upgrade Server field. Then reboot the HandyTone-286.

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

7 Software Upgrade with TFTP

7.1 Upgrade through HTTP

To upgrade software, HandyTone ATAs can be configured with an HTTP server where the new code image file is located. For example, following URL in the HTTP Upgrade Server:

http://firmware.mycompany.com:6688/Grandstream/1.0.6.2

Where, firmware.mycompany.com is the FQDN of the HTTP server, ":6688" is the TCP port the HTTP server listening to, "/Grandstream/1.0.6.2" is the RELATIVE directory to the root dir in HTTP web server. Thus, you can put different firmware into different directory as well.

NOTES:

- For firmware version 1.0.5.21 and below, "Auto Upgrade" field must be set to "Yes" to enable HTTP upgrade. In addition, the ATA will check the HTTP server in the number of days that is defined in "Check for new firmware every" field.

- For firmware version 1.0.5.22 and above, if "Auto Upgrade" field is set to "No", HTTP upgrade will be performed only once during boot up. If it is set to "Yes", the device will check the HTTP server in the number of days that is defined in "Check for new firmware every" field.

7.2 Upgrade through 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 to 20 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 Services section of Grandstream's Web site to obtain this TFTP server IP address.

NOTES:

- When HandyTone ATA boot up, it will send TFTP or HTTP request to download configuration files, there are two configuration files, one is "cfg.txt" and the other is "cfg000b82xxxxxx", where "000b82xxxxxx" is the MAC address of the HandyTone ATA. These two files are for initial automatically provisioning purpose only, for normal TFTP or HTTP firmware upgrade, the following error messages in a TFTP or HTTP server log can be ignored.

```
TFTP Error from [IP ADRESS] requesting cfg000b82023dd4 : File does not exist
TFTP Error from [IP ADRESS] requesting cfg.txt : File does not exist
```

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. The MAC address of the device is located on the bottom of the device. It is a 12 digits hex number.

- **Step 2:** Encode the MAC address to decimal digits. Please use the following mapping:

  0-9: 0-9
  A: 22
  B: 222
  C: 2222
  D: 33
  E: 333
  F: 3333

  For example, for MAC address 000b8200e395, the
  user should encode it as "0002228200333395".

- **Step 3:** Access the voice menu by pressing *** or the LED button, then dial "99" and get the voice prompt "RESET"

- **Step 4:** Key in the encoded MAC address decimal digits after hear the IVR prompt. Once the correct encoded MAC address is entered, the device will reboot automatically and restore the factory default setting.