Grandstream Networks, Inc.

HT–286
Analog Telephone Adaptor

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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.

SAFETY COMPLIANCES

The HT–286 complies with FCC/CE and various safety standards. The HT–286 power adaptor is compliant with UL standard. Only use the universal power adapter provided with the HT–286 package. The manufacturer's warranty does not cover damages to the phone caused by unsupported power adaptors.

WARRANTY

If you purchased your HT–286 from a reseller, please contact the company where you purchased your phone for replacement, repair or refund. If you purchased the product directly from Grandstream, contact your Grandstream Sales and Service Representative for a RMA (Return Materials Authorization) number before you return the product. Grandstream reserves the right to remedy warranty policy without prior notification.

Warning: *Please do not use a different power adaptor with the HT–286 as it may cause damage to the products and void the manufacturer warranty.*

- This document contains links to Grandstream GUI Interfaces. Please download these examples http://www.grandstream.com/support/ht_series/ht286/documents/ht286_gui.zip as your reference.

- This document is subject to change without notice. The latest electronic version of this user manual is available for download @: http://grandstream.com/support/ht_series/general/ht_support.html

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Installation

EQUIPMENT PACKAGING

The HT–286 ATA package contains:

- One HT–286 Main Case
- One Universal Power Adaptor
- One Ethernet Cable

CONNECTING YOUR ATA

The HT–286 is easy to configure. 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.

FIGURE 1: CONNECTING THE HT286

TABLE 1: DEFINITIONS OF THE HT–286 CONNECTORS

Power Cable	Power adapter connection
LAN Port (RJ-45)	Connect the LAN port with an Ethernet cable to your PC.
PHONE	FXS port to be connected to analog phones / fax machines.
BUTTON	Button and two colors led indicator.

THREE EASY STEPS TO INSTALL THE HT–286

1. Connect a standard touch-tone analog telephone (or fax machine) to FXS port.
2. Insert the Ethernet cable into the LAN port of HT–286 and connect the other end of the Ethernet cable to an uplink port (a router or a modem, etc.)
3. Insert the power adapter into the HT–286 and connect it to a wall outlet.

FIGURE 2: HT286 CONNECTION DIAGRAM
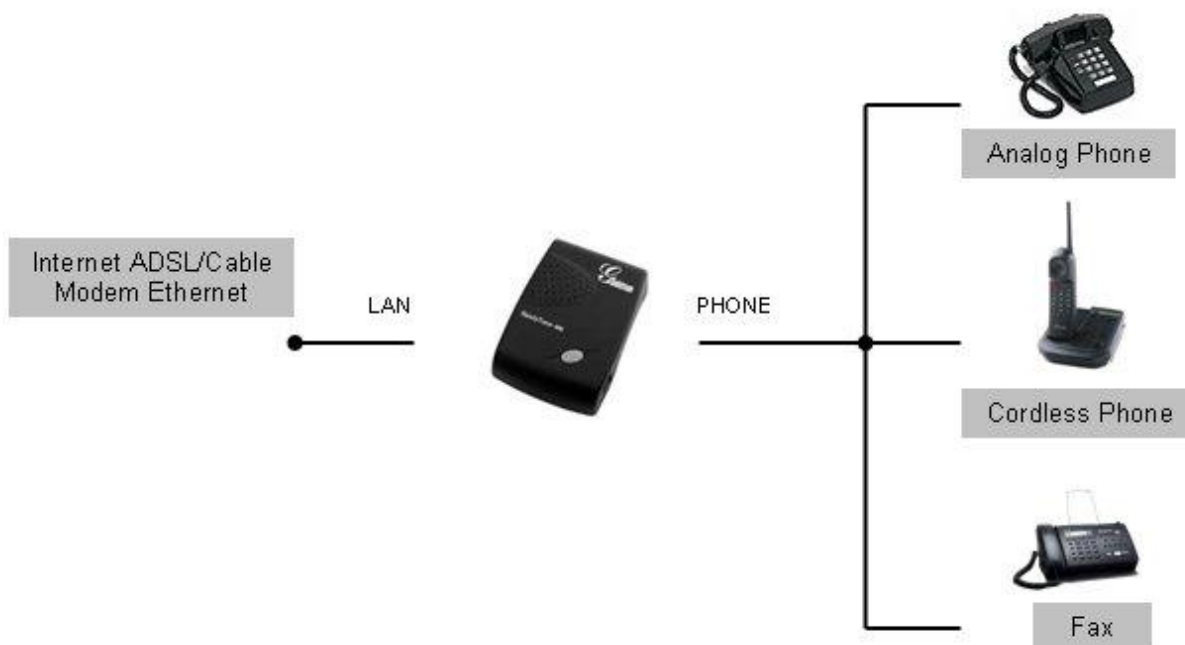
Product Overview

HT-286 supports one FXS port for Internet data, voice, and fax.

KEY FEATURES

Ethernet Ports	DHCP	FXS Port	PSTN Pass – through	Voice Mail Indicator	Voice Codec	Remote Configuration
1 RJ45 (LAN)	Client	1	No	Yes	iLBC, G.723, G.711, G.729, G.726, T.38	TFTP/HTTP

TABLE 2: HT–286 TECHNICAL SPECIFICATIONS

Lines/SIP Accounts	1 lines / 1 SIP accounts
Protocol Support	SIP 2.0 (RFC 3261), TCP/UDP/IP, RTP/RTCP, HTTP, ARP/RARP, ICMP, DNS, DHCP, NTP, TFTP, PPPoE protocols
Feature Keys	1 button
LAN Interface	RJ-45 10 Mbps
Device Management	Web interface or via secure (AES encrypted) central configuration file for mass deployment
	Support device configuration via built-in IVR, Web browser or central configuration file through TFTP or HTTP
	Support Layer 2 (802.1Q, VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
	Auto/manual provisioning system
	NAT-friendly remote software upgrade (via TFTP/HTTP) for deployed devices including behind firewall/NAT
	Syslog support (on Rev 2.0)
DHCP Server/Client	Yes, Client
Short and Long Haul	3 RENs: Up to 150 ft on 24 AWG line
Audio Features	Advanced Digital Signal Processing (DSP)
	Dynamic negotiation of codec and voice payload length
	Support for G.723,1 (5.3K/6.3K), G.729A, G.711 µ/A, G.726, and iLBC codecs
	In-band and out-of-band DTMF (in audio, RFC2833, SIP INFO)
	Silence Suppression, VAD (voice activity detection), CNG (comfort noise generation), ANG (automatic gain control), Line Echo Cancellation(G.168)
	Adaptive jitter buffer control
	Packet delay & loss concealment
	Support volume amplification
	Support configurable Call Progress Tones
Call Handling Features	Caller ID display or block, Call waiting caller ID, Call waiting/Flash, Call transfer, hold, forward, mute, 3-way conferencing(on Rev 2.0)

Network and Provisioning	Manual or dynamic host configuration protocol (DHCP) network setup; RTP and NAT support traversal via STUN
Fax over IP	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through (pending), Fax Datapump V.17, V.19, V.27ter, V.29 for T.38 fax relay
Security	DIGEST authentication and encryption using MD5 and MD5-sess
Physical Design	Stylish and compact design; small universal power supply, ideal for travel

TABLE 3: HT–286 HARDWARE SPECIFICATIONS

LAN Interface	1 x RJ45 10 Mbps
FXS phone port	1 x FXS
Button	1
LED	Green and red / solid state & blinking state
Universal Switching Power Adaptor	**Input:** 100-240VAC 50-60 Hz **Output:** +5VDC, 1200mA **UL** certified
Dimension	65mm (W) x 93mm (D) x 27mm (H)
Weight	0.57 lbs (0.26kg)
Temperature	32 - 104oF / 0 - 40oC
Humidity	10% - 90% (non-condensing)
Compliance	FCC/CE/C-Tick

Basic Operations

GET FAMILIAR WITH VOICE PROMPT

HT–286 stores a voice prompt menu (Interactive Voice Response or **IVR**) for quick browsing and simple configuration.

Pick up the handset and press the button on the HT–286 <u>or</u> dial "*******" to use the IVR menu.

TABLE 4: HT–286 IVR MENU DEFINITIONS

Menu	Voice Prompt	Options
Main Menu	"Enter a Menu Option"	Press "*" for the next menu option Press "#" to return to the main menu Enter 01-05, 07,12-17,47 or 99 menu options
01	"DHCP Mode", "Static IP Mode"	Press "**9**" to toggle the selection If using "*Static IP Mode*", configure the IP address information using menus 02 to 05. If using "*Dynamic IP Mode*", all IP address information comes from the DHCP server automatically after reboot.
02	"IP Address " + IP address	The current WAN IP address is announced If using "*Static IP Mode*", enter 12 digit new IP address.
03	"Subnet " + IP address	Same as menu 02
04	"Gateway " + IP address	Same as menu 02
05	"DNS Server " + IP address	Same as menu 02
07	Preferred Vocoder	Press "**9**" to move to the next selection in the list: • PCM U • PCM A • G.723 • G.729 • iLBC • G.726
13	Firmware Server IP Address	Announces current Firmware Server IP address. Enter 12 digit new IP address.
14	Configuration Server IP Address	Announces current Config Server Path IP address. Enter 12 digit new IP address.
15	Upgrade Protocol	Upgrade protocol for firmware and configuration update. Press "**9**" to toggle between **TFTP** / **HTTP**
16	Firmware Version	Firmware version information.
17	Firmware Upgrade	Firmware upgrade mode. Press "**9**" to toggle among the following three options: - always check - check when pre/suffix changes - never upgrade

47	"Direct IP Calling"	Enter a 12 digit IP address to make a direct IP call, after dial tone. (See "*Make a Direct IP Call*".)
99	"RESET"	Press "**9**" to reboot the device; **or**
		Enter encoded MAC address to restore factory default setting (See "*Restoring Factory Settings*")
	"Invalid Entry"	Automatically returns to main menu

TABLE 5: IVR ERROR REPORT

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

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

NOTE:

1. Once the button is pressed, it enters the voice prompt main menu. If the button is pressed again, while it is already in the voice prompt menu, it jumps to "Direct IP Calling" option and a dial tone is prompted

2. "*****" shifts down to the next menu option
   "**#**" returns to the main menu
   "**9**" functions as the ENTER key in many cases to confirm an option

3. All entered digit sequences have known lengths - 2 digits for menu option and 12 digits for IP address. For IP address, add 0 before the digits if the digits are less than 3 (like 192.168.0.26 should be key in like 192168000026, no dot needed while input). Once all of the digits are collected, the input will be processed.

4. Key entry can not be deleted but the phone may prompt error once it is detected

PLACING A PHONE CALL

PHONE OR EXTENSION NUMBERS

1. Dial the number directly and wait for 4 seconds (Default "No Key Entry Timeout"); or

2. Dial the number directly and press # (Use # as dial key" must be configured in web configuration).

Examples:

1. Dial an extension directly on the same proxy, (e.g. 1008), and then press the # or wait for 4 seconds.

2. Dial an outside number (e.g. (626) 666-7890), first enter the prefix number (usually 1+ or international code) followed by the phone number. Press # or wait for 4 seconds. Check with your VoIP service provider for further details on prefix numbers.

DIRECT IP CALLS

Direct IP calling allows two parties with VoIP devices (e.g. a HT–286and another VoIP Device), to talk to each other in an ad hoc fashion without a SIP proxy. Direct IP calls are possible if:

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

To make a Direct IP Call, use either the handset or the speakerphone. Access the IVR using "*******", the **button** on the HT–286or "**47**". After dial tone, enter a 12-digit target IP address to place a call. Destination ports can be specified by using "***4**" (encoding for ":") followed by the port number.

Examples:

a) If the target IP address is 192.168.0.160, enter the 12 digit IP address (e.g. 1921680160) after the voice prompt followed by the "#" key or wait for 4 seconds. The default destination port 5060 is used if no port is specified.

b) If the target IP address/port is 192.168.1.20:5062, enter the 12 digit IP address + ***4**5062 (e.g. 1921680160*45062) after the voice prompt followed by the "#" key or wait for 4 seconds.

CALL HOLD

Press the "flash" button (or hook flash on older models) to place a caller on hold. Press the "flash" button again to release call hold.

CALL WAITING

If call waiting feature is enabled, call waiting tone (3 short beeps) indicates an incoming call. Toggle between incoming call and current call by pressing the "flash" button. First call is placed on hold. Press the "flash" button to toggle between two active calls.

CALL TRANSFER

The HT–286 supports both *blind* and *attended* transfer:

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

    Expected outcomes:
    a) A quick confirmation tone (call waiting tone) followed by a dial tone. This indicates the transfer is successful. Hang up or place another call.
    b) A quick busy tone followed by a restored call (on supported platforms only). The busy tone indicates the transfer failed.
    c) Continuous busy tone. The phone call has timed out. **Note:** continuous busy tone does not indicate the transfer has been successful, nor does it indicate the transfer has failed. It often means there was a failure to receive second NOTIFY – check firmware for most recent release.

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

2. **Attended Transfer:**

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

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

NOTE: If Attended Transfer fails, and party A hangs up, the HandTone-496 will ring party A to remind A that party B is still on the line. Party A can pick up the phone to resume conversation with party B.

3-WAY CONFERENCING

HT–286 supports Star Code Style or Bellcore Style 3-way Conference.

Star Code Style 3-way Conference

Assuming that call party A and B are in conversation. A (HT-286) wants to bring C in a conference:

1. A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
2. A dials *23 then C's number then # (or wait for 4 seconds).
3. If C answers the call, then A presses FLASH to bring B, C in the conference.
4. If C does not answer the call, A can press FLASH back to talk to B.
5. If A presses FLASH during conference, C will be dropped out.

Bellcore Style 3-way Conference

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

Assuming that call party A and B are in conversation. A (HT-286) wants to bring C in a conference:

1. A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
2. A dials C's number then # (or wait for 4 seconds).
3. If C answers the call, then A presses FLASH to bring B, C in the conference.
4. If C does not answer the call, A can press FLASH back to talk to B.
5. If A presses FLASH during conference, C will be dropped out.

Call Features

The HT–286 supports all the traditional and advanced telephony features.

TABLE 6: HT–286 CALL FEATURE DEFINITIONS

Key	Call Features
*23	**3-way conference**
*30	**Block Caller ID** (for all subsequent calls)
*31	**Send Caller ID** (for all subsequent calls)
*50	**Disable Call Waiting** (for all subsequent calls)
*51	**Enable Call Waiting** (for all subsequent calls)
*67	**Block Caller ID** (per call)
*70	**Disable Call Waiting** (per call)
*71	**Enable Call Waiting** (per call)
*72	**Unconditional Call Forward.** Dial "*72", wait for dial tone. Dial the forward number and "#" for a dial tone, then hang up.
*73	**Cancel Unconditional Call Forward.** To cancel "Unconditional Call Forward", dial "*73", wait for dial tone, then hang up.
*82	**Send Caller ID** (per call)
*87	**Blind Transfer**
*90	**Busy Call Forward.** Dial "*90" , wait for 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", wait for dial tone, then hang up.
*92	**Delayed Call Forward.** Dial "*92", wait for dial tone. Then dial the forward number and "#" for a dial tone, then hang up.
*93	**Cancel Delayed Call Forward.** To cancel Delayed Call Forward, dial "*93", wait for dial tone, then hang up.
Flash/Hook	Toggles between active call and incoming call (call waiting tone). If not in conversation, flash/hook will switch to a new channel for a new call.

T.38 Fax

HT–286 supports fax in two modes: 1) T.38 (Fax over IP) and 2) fax pass through.

T.38 is the preferred method because it is more reliable and works well in most network conditions. If the service provider supports T.38, please use this method by selecting T.38 as fax mode (default). If the service provider does not support T.38, pass-through mode may be used.

LED Light Pattern Indication

Following tables show the LED light pattern indication.

TABLE 7: HT–286 LED DEFINITIONS

RED LED indicates not normal status	
Button flashes every 1 second	If Ethernet link is down
Button flashes every 2 seconds. (if DHCP is configured)	DHCP Failed or WAN No Cable
Button flashes every 4 seconds. (if SIP server is configured)	HT–286 fails to register
Button flashes every 6 seconds.	Firmware Upgrading
Button flashes briefly	No STUN responses
Red light steady.	Device Malfunctions

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

Configuration Guide

CONFIGURING HT–286 THROUGH VOICE PROMPT

DHCP Mode
Select voice menu option 01 to enable HT–286 to use DHCP.

STATIC IP Mode
Select voice menu option 01 to enable HT–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.

Firmware Server IP Address
Select voice menu option 13 to configure the IP address of the firmware server.

Configuration Server IP Address
Select voice menu option 14 to configure the IP address of the configuration server.

Upgrade Protocol
Select voice menu option 15 to choose firmware and configuration upgrade protocol. User can choose between TFTP and HTTP.

Firmware Upgrade Mode
Select voice menu option 17 to choose firmware upgrade mode among the following three options:
1) always check, 2) check when pre/suffix changes, and 3) never upgrade

CONFIGURING HT–286 WITH WEB BROWSER

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

ACCESS THE WEB CONFIGURATION MENU

HandyTone-286 has embedded HTML pages that allows a user to configure the HandyTone-286 through a Web browser.

1. Find the IP address of the HT–286 using voice prompt menu option 02.

2. Access the HT–286 Web Configuration page by the following URI via WAN port:
    http://HandyTone-IP-Address (the HandyTone-IP-Address is the IP address for the HT–286).

NOTE: If using a web browser to enter the configuration page, strip the leading "0"s because the browser will parse in octet. (i.e. if the IP address is: 192.168.<u>00</u>1.<u>01</u>4, please type in: 192.168.1.14).

END USER CONFIGURATION

Once the HTTP request is entered and sent from a Web browser, the user will see a log in screen. There are two default passwords for the login page:

User Level:	Password:	Web pages allowed:
End User Level	123	Only Status and Basic Settings
Administrator Level	admin	Browse all pages

Only an administrator can access the "*ADVANCED SETTING*" configuration page.

NOTE: If you cannot log into the configuration page by using default password, please check with the VoIP service provider. The service provider may have provisioned and configured the device for you. The Basic Configuration Page is the first web GUI the user will see.

TABLE 8: HT–286 BASIC CONFIGURATION SETTINGS DEFINITIONS

End User Password	Password to access the Web Configuration Menu. This field is case sensitive with a maximum length of 25 characters.
Web Port	By default, HTTP uses port 80. This field is for customizable web port.
IP Address	There are two modes to operate the HT–286: **DHCP mode**: all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The HT–286 acquires its IP address from the first DHCP server it discovers from the LAN it is connected. *Using the PPPoE feature*: set the PPPoE account settings. The HT–286 will establish a PPPoE session if any of the PPPoE fields is set. **Static IP mode:** configure the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields. These fields are set to zero by default.
DHCP hostname	This option specifies the name of the client. This field is optional but may be required by some Internet Service Providers. Default is blank.
DHCP domain	This option specifies the domain name that client should use when resolving hostnames via the Domain Name System. Default is blank.
DHCP vendor class ID	Used by clients and servers to exchange vendor-specific information. Default is blank.
PPPoE account ID	PPPoE username. Necessary if ISP requires you to use a PPPoE (Point to Point Protocol over Ethernet) connection.
PPPoE password	PPPoE account password.
PPPoE Service Name	This field is optional. If your ISP uses a service name for the PPPoE connection, enter the service name here. Default is blank.
Time Zone	Controls how the date/time displays according to the specified time zone.

Daylight Savings Time	Controls whether displayed time is daylight savings time or not. If set to "Yes" and the Optional Rule is empty, then the displayed time will be 1 hour ahead of normal time.
	
	 • The "*Automatic Daylight Saving Time Rule*" has the following syntax:
	     ○ *start-time;end-time;saving*. Both start-time and end-time have the same syntax: *month,day,weekday,hour,minute*
	     ○ month: 1,2,3,..,12 (for Jan, Feb, .., Dec)
	     ○ day: [+|-]1,2,3,..,31
	     ○ weekday: 1, 2, 3, .., 7 (for Mon, Tue, .., Sun), or 0 indicating daylight savings rule is based on the day of the month.
	     ○ hour: hour (0-23),
	     ○ minute: minute (0-59)
	 • If "weekday" is 0, it means the date to start or end daylight savings is the given date. The "day" value must not be negative.
	 • If "weekday" is not zero and "day" is positive, then daylight saving starts on the first "day" of the week (1st Sunday, 3rd Tuesday etc).
	 • If "weekday" is not zero and "day" is negative, then daylight savings starts on the last "day" of the week (last Sunday, 3rd last Tuesday etc).
	 • Daylight savings is in minutes. If preceded by a (-), then subtract the number of minutes.
	
	 The default value for "Automatic Daylight Saving Time Rule" is US time: "04,01,7,02,00;10,-1,7,02,00;60".
	
	 Example: US/Canada, where daylight savings is applicable: 04,01,7,02,00;10,-1,7,02,00;60
	
	 Daylight savings starts from the first Sunday of April at 2AM and ends the last Sunday of October at 2AM.).

In addition to the Basic Settings configuration page, end users also have access to the Device Status page.

TABLE 9: HT–286 DEVICE STATUS PAGE DEFINITIONS

MAC Address	The device ID in HEX format. This is needed for ISP troubleshooting.
WAN IP Address	Shows WAN IP address of HT–286.
Product Model	Contains the product model info.
Software Version	Program: This is the main software release. Boot and Loader are seldom changed.
System Up Time	Shows system up time since the last reboot.
Registered	Indicates whether the HT–286 is registered to the service provider's server.
PPPoE Link Up	Indicates whether the PPPoE connection is up if the HT–286 is connected to DSL modem.
NAT	Indicates the type of NAT the HT–286 is connected to via its WAN port. Based on STUN protocol.

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.

ADVANCED USER CONFIGURATION

Log in to the advanced user configuration page the same way as for the basic configuration page. The password is case sensitive and the factory default password for Advanced User is "*admin*".

Advanced User configuration includes the end user configuration and the advanced configurations including: a) SIP configuration, b) Codec selection, c) NAT Traversal Setting and d) other miscellaneous configuration.

TABLE 10: HT–286 ADVANCED CONFIGURATION PAGE DEFINITIONS

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.
Home NPA	Local area code for North American Dial Plan.
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	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)
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.
Allow outgoing call without Registration	Default is set to "No".
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.
SIP Registration Failure Retry Wait Time	Retry registration if the process failed. Default is 20 seconds.

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.
Use STUN keep-alive to detect networks connectivity	Use STUN keep-alive to detect WAN side network problems. If keep-alive request does not yield any response for configured number of times, the device will restart the TCP/IP stack. If the STUN server does not respond when the device boots up, the feature is disabled.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
SUBSCRIBE for MWI	Default is "No". When set to "Yes" a SUBSCRIBE for Message Waiting Indication will be sent periodically
Off hook Auto-Dial	This parameter allows the user to configure a User ID or extension number to be automatically dialed upon off hook. 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 using star codes are supported locally.
Use Bell-style 3-way conference	Conference mode, default option is No. If set to yes, the feature code for coference *23 would be disabled.
Disable Call Waiting	Default is No.
Disable Call-Waiting Caller-ID	Default is No.
Send DTMF	This parameter specifies the mechanism to transmit DTMF digit. There are 3 modes supported: in audio which means DTMF is combined in audio signal (not very reliable with low-bit-rate codec), via RTP (RFC2833), or via SIP INFO.
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.
Onhook Threshold	The amount of time the hookflash is pressed that will cause the device to on hook. Default is 800ms.
FXS Impedance	Selects the impedance of the analog telephone connected to the Phone port.

Caller ID Scheme	Select the Caller ID Scheme to suit the standard of different area. • Bellcore (North America) • CID (Canada) • DTMF (Brazil) • DTMF (Denmark) • DTMF (Sweden) • ETSI-FSK (France, Germany, Norway, Taiwan, UK-CCA) • ETSI-DTMF (Finland, Sweden)
Onhook Voltage	The onhook voltage can be selected according to the line voltage depending on the analog phone used. The low power/high power will increase/decrease the output current. Selecting "low power" will make loop current limit = 20mA, selecting "high power" make loop current limit = 32mA. The default selection is 36V (High Power).
Polarity Reversal	Select Polarity Reversal to adapt some call charge/billing system. Default is No.
NTP server	This 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.
Time to ring	This setting allows user to adjust the ring time of the phone. Default is 60 seconds.
Special Features	Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Nortel, Broadsoft, CBCOM etc.
CBCOM Encode	SIP, RT(C)P and T.38 modes, 1.0 and 1.1
CBCOM Encoder 1.1 Key:	Key to be used by CBCOM
Syslog Server	The IP address or URL of System log server. This feature is especially useful for ITSP (Internet Telephone Service Provider)

Syslog Level	Select the ATA to report the log level. Default is NONE. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: • product model/version on boot up (INFO level) • NAT related info (INFO level) • sent or received SIP message (DEBUG level) • SIP message summary (INFO level) • inbound and outbound calls (INFO level) • registration status change (INFO level) • negotiated codec (INFO level) • Ethernet link up (INFO level) • SLIC chip exception (WARNING and ERROR levels) • memory exception (ERROR level) The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: GS_LOG: [device MAC address][error code] error message Here is an example: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up
Session Expiration	Grandstream implemented SIP Session Timer. The session timer extension enables SIP sessions to be periodically "refreshed" via a re-INVITE request. Once the session interval expires, if there is no refresh via a re-INVITE message, the session will be terminated. Session Expiration is the time (in seconds) at which the session is considered timed out, if no successful session refresh transaction occurs beforehand. The default value is 180 seconds.
Min-SE	The minimum session expiration (in seconds). The default value is 90 seconds.
Caller Request Timer	If selecting "Yes" the phone will use session timer when it makes outbound calls if remote party supports session timer.
Callee Request Timer	If selecting "Yes" the phone will use session timer when it receives inbound calls with session timer request.
Force Timer	If selecting "Yes" the phone will use session timer even if the remote party does not support this feature. Selecting "No" will allow the phone to enable session timer only when the remote party support this feature. To turn off Session Timer, select "No" for Caller Request Timer, Callee Request Timer, and Force Timer.
UAC Specify Refresher	As a Caller, select UAC to use the phone as the refresher, or UAS to use the Callee or proxy server as the refresher.
UAS Specify Refresher	As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the phone as the refresher.
Force INVITE	Session Timer can be refreshed using INVITE method or UPDATE method. Select "Yes" to use INVITE method to refresh the session timer.

Firmware Upgrade and Provisioning	Default 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, HT286 rev. 3.0 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	Choose "Yes" to enable automatic upgrade and provisioning and input the number, in minutes, you want the HT to check for an update. When set to No, HT286 will only do upgrade once at boot up. "Always check for New Firmware at Boot up" will check for new firmware every time the device reboots. "Check New Firmware only when F/W pre/suffix changes" will check for updates only when the pre/suffix has been changed.
Firmware Key	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", except for IVR MENU items 1 to 5, 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.
Volume Amplification	Handset volume adjustment. RX is for receiving volume, TX is for transmission volume. Default values are 0dB for both parameters. +6dB generates the highest volume and -6dB generates the lowest volume.
Powerline Ring Tone	This setting allows user to configure the ringing frequencies and cadences.
Call Progress Tones	Using these settings, users can configure various call progress tone frequencies and cadences according to their country standard. By default they are set to North American standard. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. ON is the period of ringing ("On time" in 'ms') while OFF is the period of silence. In order to set a continuous ring, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern. Up to three cadences are supported.
Disable Line Echo Canceller (LEC):	Default is No. If set to Yes, echo canceller is not used.

Disable voice prompt:	Default is No. If set to Yes, voice prompt is disabled. Only short IVR will be heard when pressing the led button.
Enable Toll Application:	Default is No. Set to Yes if connecting to a toll device or phone.

SAVING THE CONFIGURATION CHANGES

After making a change, click the "Update" button in the Configuration page. The HT–286 will display a screen to confirming changes. Reboot or power cycle the HT–286 to enable the changes.

REBOOTING THE HT–286 FROM REMOTE

The administrator of the HT–286 can remotely reboot the HT–286 by clicking the "Reboot" button at the bottom of the configuration page. When finished, re-login to the HT–286 after waiting about 30 seconds.

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 has a provisioning system called GAPS (Grandstream Automated Provisioning System) that is used to support automated configuration of Grandstream devices. GAPS uses enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual Grandstream device for firmware upgrade, remote reboot, etc.

Grandstream provides GAPS service to VoIP service providers. Use GAPS for either simple redirection or with certain special provisioning settings. At boot-up, Grandstream devices by default point to Grandstream provisioning server GAPS, based on the unique MAC address of each device, GAPS provision the devices with redirection settings so that they will be redirected to customer's TFTP or HTTP/HTTPS server for further provisioning.

Software Upgrade

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

FIRMWARE UPGRADE THROUGH TFTP/HTTP

To upgrade via TFTP or HTTP, the "Firmware Upgrade and Provisioning upgrade via" field needs to be set to TFTP or HTTP, respectively. "Firmware Server Path" needs to be set to a valid URL of a TFTP or HTTP server, server name can be in either FQDN or IP address format. Here are examples of some valid URL.

      e.g. firmware.mycompany.com:6688/Grandstream/1.1.0.45
      e.g. 72.172.83.110

NOTES:

- Firmware upgrade server in IP address format can be configured via IVR. Please refer to the CONFIGURATION GUIDE section for instructions. If the server is in FQDN format, it must be set via the web configuration interface.

- Grandstream recommends end-user use the Grandstream HTTP server. Its address can be found at http://www.grandstream.com/firmware.html. Currently the HTTP firmware server IP address is 72.172.83.110. For large companies, we recommend to maintain their own TFTP/ HTTP server for upgrade and provisioning procedures.

- Once a "Firmware Server Path" is set, user needs to update the settings and reboot the device. If the configured firmware server is found and a new code image is available, the HT ATA will attempt to retrieve the new image files by downloading them into the HT ATA's SRAM. During this stage, the HT ATA's LEDs will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will then be saved into the Flash. If TFTP/HTTP fails for any reason (e.g., TFTP/HTTP server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the HT ATA will stop the TFTP/HTTP process and simply boot using the existing code image in the flash.

- Firmware upgrade may take as long as 1 to 20 minutes over Internet, or 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.

- Grandstream's latest firmware is available http://www.grandstream.com/firmware.html.
  Oversea users are strongly recommended to download the binary files and upgrade firmware locally in a controlled LAN environment.

- Alternatively, user can download a free TFTP or HTTP server and conduct local firmware upgrade. A free windows version TFTP server is available for download from http://support.solarwinds.net/updates/New-customerFree.cfm.

Instructions for firmware upgrade using a local TFTP Server:

1. Unzip the file and put all of them under the root directory of the TFTP server.
2. Put the PC running the TFTP server and the HT286 device in the same LAN segment.
3. Please go to File -> Configure -> Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade.
4. Start the TFTP server, in the phone's web configuration page
5. Configure the Firmware Server Path with the IP address of the PC
6. Update the change and reboot the unit

End users can also choose to download the free HTTP server from http://httpd.apache.org/ or use Microsoft IIS web server.

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.

FIRMWARE AND CONFIGURATION FILE PREFIX AND POSTFIX

Firmware Prefix and Postfix allows device to download the firmware name with the matching Prefix and Postfix. This makes it the possible to store ALL of the firmware with different version in one single directory. Similarly, Config File Prefix and Postfix allows device to download the configuration file with the matching Prefix and Postfix. Thus multiple configuration files for the same device can be stored in one directory.

In addition, when the field "Check New Firmware only when F/W pre/suffix changes" is set to "Yes", the device will only issue firmware upgrade request if there are changes in the firmware Prefix or Postfix.

MANAGING FIRMWARE AND CONFIGURATION FILE DOWNLOAD

When "Automatic Upgrade" is set to "Yes", Service Provider can use P193 (Auto Check Interval, in minutes, default and minimum is 60 minutes) to have the devices periodically check with either Firmware Server or Config Server, whenever they are defined. This allows the device periodically check if there are any new changes need to be taken on a scheduled time. By defining different intervals in P193 for different devices, Server Provider can spread the Firmware or Configuration File download in minutes to reduce the Firmware or Provisioning Server load at any given time.

Restore Factory Default Setting

WARNING! **Restoring the Factory Default Setting will DELETE all configuration information of the phone. Please BACKUP or PRINT out all the settings before you approach to following steps. Grandstream will not take any responsibility if you lose all the parameters of setting and cannot connect to your VoIP service provider.**

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:

- Pick up the headset. Dial "***" or press the LED button for voice prompt.
- Enter "99" and get the voice prompt "Reset".
- Enter the encoded MAC address of the device.
- Wait for 15 seconds. The device will reboot automatically and restore to factory default setting.

NOTE: Factory Reset will be disabled if the "Lock keypad update" or "Disable voice prompt" is set to "Yes".