User Manual

HandyTone-286

Analog Telephone Adaptor

Version 1.00

Grandstream Network, 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.

286 runs the same VoIP stack as BudgeTone-100 series IP telephone's. Additionally, it has G.168 to remove line echo. **For additional specifications about BudgeTone 100 series, please refer to BugeTone-100 series Users Manual at:**

http://www.grandstream.com/user_manuals/budgetone100.pdf

Grandstream Networks, Inc.

2 Installation

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

2.1 What is Included in the Package

The HandyTone 286 package contains:

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

2.2 Safety Compliances

The HandyTone 286 is compliant with various safety standards including FCC/CE. Its power adaptor is compliant with UL standard. The HandyTone ATA should only operate with the universal power adaptor provided in the package.

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

Caution: Changes or modifications to this product not expressly approved by Grandstream, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.

3 Product Overview

3.1 Key Features

• Support SIP 2.0, TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, TFTP protocols
• Support IETF STUN.
• Interoperable with various 3rd party SIP end user device, Proxy / Registrar / Server, and gateway products.
• Advanced Digital Signal Processing (DSP) technology to ensure superior audio quality
• Advanced and patent pending adaptive jitter buffer control, packet delay and loss concealment technology
• Support popular vocoders including G.723.1 (5.3K/6.3K), G.729A/B, G.711 (alaw and u-law), G.726 (40K/32K/24K/16K), as well as G.728.
• Support standard voice features such as Call Waiting[1], Transfer[2], Forward[3], in-band and out-of-band DTMF, Dial Plans
• Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168), and AGC (Automatic Gain Control)
• Support DIGEST authentication (MD5, MD5-sess)
• Provide easy configuration thru manual operation (voce prompt along with the analog phone keypad and Web interface) or automated centralized configuration file.
• Support for Layer 2 (802.1Q VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
• Remote software upgrade capability via TFTP

[1] This feature is yet to implement the current version of HandyTone 286
[2] same as above
[3] same as above

3.2 Hardware Specification

The table below describes the difference among these models.

Model	**HandyTone 286**
LAN interface	1xRJ45 10Base-T
Button	1
LED	GREEN & RED color
Universal Switching Power Adaptor	Input: 100-240VAC 50-60 Hz Output: +5VDC, 1200mA, UL certified
Dimension	65mm (W) 93mm (D) 27mm (H)
Weight	3 oz ± 0.05
Storage Temperature	32 - 104^{o}F 0 - 40^{o}C
Operating Temperature	32 - 104^{o}F 0 - 40^{o}C
Humidity	10% - 95% (non-condensing)
Compliance	FCC/CE

4 Basic Operations

4.1 Get Familiar With Key Pad And Voice Prompt

HandyTone has stored voice prompt menu for quick browsing and simple configuration. To enter this voice prompt menu, simple press the button. After figuring out its IP address, HandyTone can be configured thru web interface just like BT-100 IP telephone.

Menu	Voice Prompt	User's Options
Main Menu	"Enter a Main Option"	Enter '*' to menu_01 Enter 00-06, 99 menu option
01	"DHCP Mode", "Static IP Mode"	Enter '9' to toggle the selection
02	"IP Address " + IP address	It will prompt you with the current IP address. Enter 12 digit new IP address if in Static IP Mode
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
06	"TFTP Server " + IP address	Same as menu 02
47	"Direct IP Calling"	When entered, you will prompt a dialtone, then enter 12 digit IP address This menu can be also entered by pressing the button again
86	"Voice Messages Pending" "No Voice Messages"	Enter 9 to dial pre-configured phone number to retrieve VM
99	"RESET"	Enter '9' to confirm the RESET Enter MAC address to restore factory default setting
	"Invalid Entry"	Automatically return to Main Menu

Notes:
Once button is pressed, it enters voice prompt main menu. If 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

'*' functions similar to '↓' key of BT-100 phone to select the next menu option
'#' returns back to main MENU
'9' is similar to 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

4.2 Make Phone Calls

4.2.1 Calling phone or extension numbers

There are currently two method to make a extension number call:

1) Dial the extension number directly and wait for 5 seconds.
2) You can dial the numbers directly, and press #(assuming that "use #" as dial key is selected in web configuration).

#Note: Currently, Flash, Transfer is not supported yet, Hold is normally a function of

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

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

Examples:
If the target IP address is 192.168.0.160, the dialing convention is
Voice Prompt with option 47, then 192168000160
followed by pressing the "#" key is it is configured as a send key or wait for more than 5 seconds. In this case, the default destination port 5060 is used if no port is specified.

If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be:
Voice Prompt with option 47, then 192168001020*45062
followed by pressing the "#" key is it is configured as a send key or wait for 5 seconds.

If the target address is john@192.168.1.100:5062, then the dialing convention would be:
Voice Prompt with option 47, then 51634262*3192168001100*45062
followed by pressing the "#" key is it is configured as a send key or wait for 5 seconds.

5 Configuration Guide

5.1 Configuring HandyTone IP Through Voice Prompt

5.1.1 DHCP Mode

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

5.1.2 STATIC IP Mode

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

5.2 Configuring HandyTone with Web Browser

HandyTone 200 series ATA has an embedded Web server that will respond to HTTP GET/POST requests. It also has embedded HTML pages that allow a user to configure the HandyTone through a Web browser such as Microsoft's IE and AOL's Netscape.

5.2.1 Access the Web Configuration Menu

First, get the IP address of the HandyTone through section 4.1 with menu option 02. Then the HandyTone's Web Configuration Menu can be accessed by the following URI:
***http://HandyTone-IP-Address*,**
where the ***HandyTone-IP-Address*** is the IP address of the HandyTone ATA.

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

The password is case sensitive and the factory default password is lower case '*admin*'.

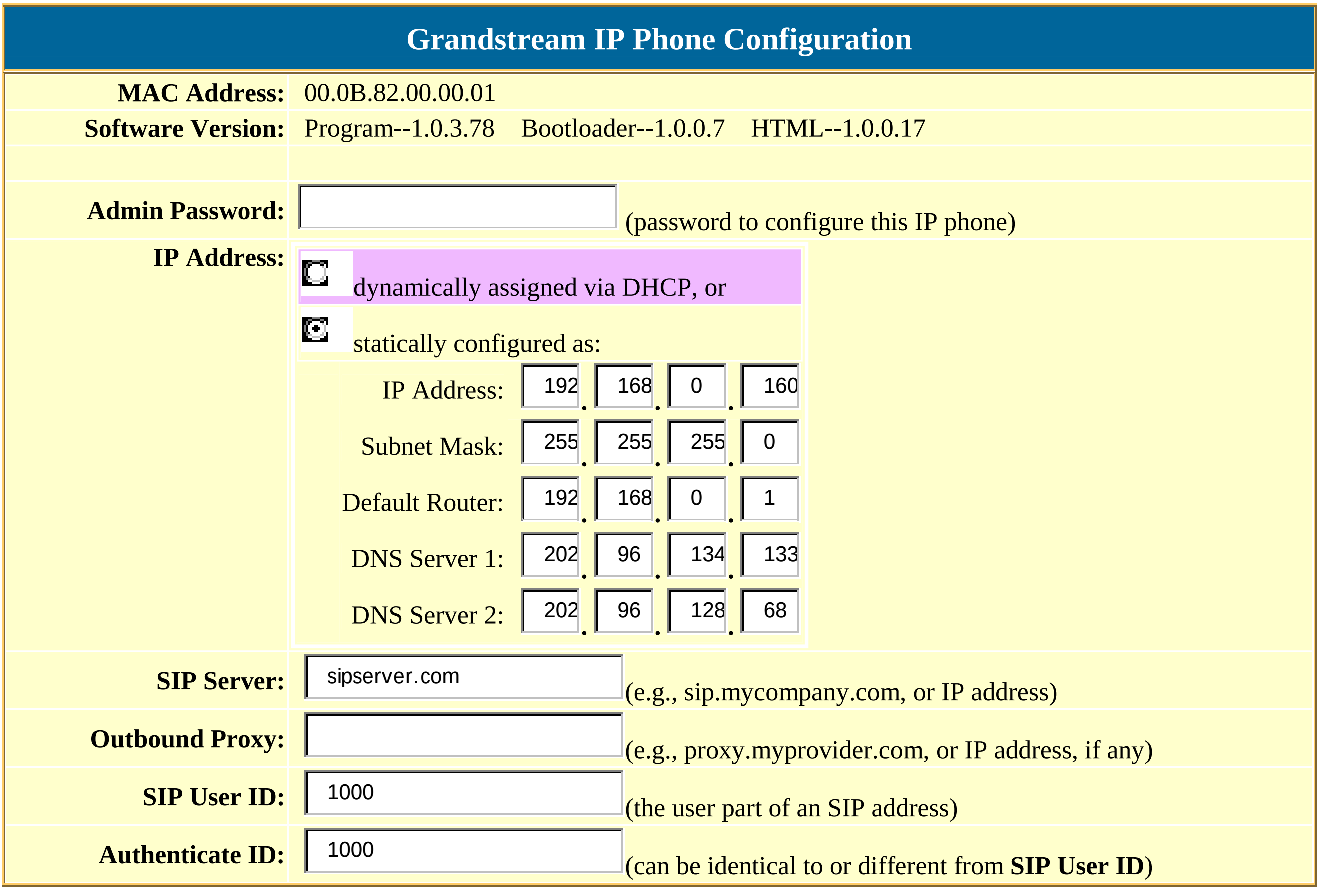
5.2.2 Configuration Menu

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

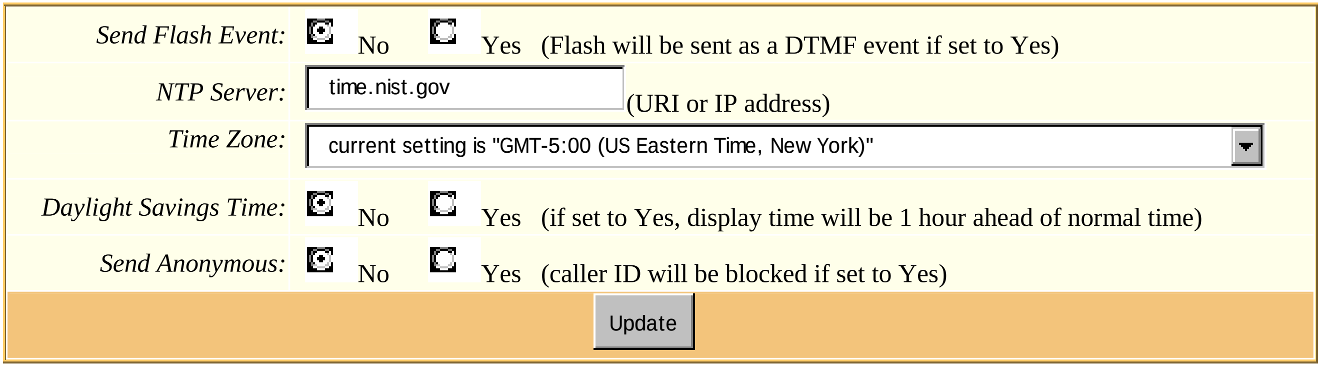
The definitions for all the configuration parameters in the Configuration Menu are:

Password	This contains the password to access the Web Configuration Menu. This field is case sensitive.
IP Address	There are 2 modes under which the HandyTone ATA can operate: - If DHCP mode is enabled, then all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory) and the HandyTone ATA will acquire its IP address from the first DHCP server it discovers on the LAN it attaches to. - If Static IP mode is selected, then the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields will need to be configured. These fields are reset to zero by default.
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 HandyTone ATA. e.g., if the SIP address is: sip:my_user_id@my_provider.com, then the SIP User ID is: my_user_id. Please do NOT include the preceding "sip:" scheme or the host portion of the SIP address in this field.
SIP User ID is Phone Number	If the HandyTone ATA has an assigned PSTN telephone number, then this field will be set to "Yes". Otherwise, set it to "No". If "Yes" is set, a "user=phone" parameter will be attached to the "From" header in SIP request.
SIP Login ID	This field contains the login ID used for SIP authentication. Typically, this is the account number on an SIP server for this HandyTone ATA. It can be the same as or different from the above SIP User ID, depending on whether a separate account ID is used for authentication.
SIP Password	This field contains the password used for SIP authentication. It is used together with the above SIP Login ID



Authenticate Password:	
Name:	John Doe (optional, e.g., John Doe)
Advanced Options:	
Preferred Vocoder: (in listed order)	choice1: current setting is "PCMU" choice2: current setting is "PCMA" choice3: current setting is "G723" choice4: current setting is "G729" choice5: current setting is "G726-32" choice6: current setting is "G728"
G723 rate:	6.3kbps encoding rate 5.3kbps encoding rate
Silence Suppression:	No Yes
Voice Frames per TX:	2 (up to 10/20/32/64 frames for G711/G726/G723/other codecs respectively)
IP QoS:	48 (IP Diff-Serv or Precedence value for RTP)
VLAN Tag:	0 (VLAN classification for RTP)
SIP User ID is phone number:	No Yes
Dial Plan:	(dial plan prefix string)
SIP Registration:	Yes No
Register Expiration:	5 (in minutes. default 1 hour, max 45 days)
Early Dial:	No Yes (use "Yes" only if proxy supports 484 response)
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: (URI or IP:port)
TFTP Server:	0 . 0 . 0 . 0 (for remote software upgrade and configuration)
Voice Mail UserID:	(User ID/extension for 3rd party voice mail system)
Offhook Auto-Dial:	(User ID/extension to dial automatically when offhook)
Send DTMF:	in-audio via RTP (RFC2833) via SIP INFO



Preferred Vocoder	The BudgeTone HandyTone ATA supports up to 6 different vocoder types including G711-ulaw, G711-alaw, G723, G729A/B, G726-32 (ADPCM), and G728. 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 6”.
G723 Rate:	This defines the encoding rate for G723 vocoder. By default, 6.3kbps rate is chosen.
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 HandyTone ATA 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.
IP Qos	This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv or MPLS. Default value is 48.
VLAN Tag	This contains the value used for layer 2 VLAN tag. Default setting is blank.
Dial Plan	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.

Early Dial	This parameter controls whether the HandyTone ATA 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.
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.
SIP Registration	This parameter controls whether the HandyTone ATA needs to send REGISTER messages to the proxy server. The default setting is "Yes".
Registration Interval	This parameter allows the user to specify the time frequency (in minutes) the HandyTone ATA will refresh its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
Local SIP port	This parameter defines the local SIP port the HandyTone ATA will listen and transmit on. The default value is 5060.
Local RTP port	This parameter defines the local RTP-RTCP port pair the HandyTone ATA 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 HandyTone ATAs are behind the same NAT.

NAT Traversal	This parameter defines whether the HandyTone ATA NAT traversal mechanism will be activated or not. If activated (by choosing "Yes") and a STUN server is also specified, then the HandyTone ATA will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the HandyTone ATA 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 HandyTone ATA 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 HandyTone ATA will periodically (every 10 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open.
TFTP Server	This is the IP address of the configured tftp server. If it is non-zero or not blank, the HandyTone ATA will attempt to retrieve new configuration file or new code image 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.
Voice Mail User ID [4]	This parameter defines the User ID (or extension number) of a 3rd party voice mail system where the user may have an account. By defining this Voice Mail extension, when the user presses the "Message" button on the phone, an INVITE message will be sent to that extension to allow the user to retrieve messages.
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 HandyTone ATA will automatically append the "@" and the host portion of the corresponding SIP address.

[4] This feature is for Budgetone 100 series IP phone.

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.
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.
NTP server	This parameter defines the URI or IP address of the NTP server which the HandyTone ATA will use to display the current date/time.
Time Zone	This parameter controls how the displayed date/time will be adjusted according to the specified time zone.
Daylight Savings Time	This parameter controls whether the displayed time will be daylight savings time or not. If set to Yes, then the displayed time will be 1 hour ahead of normal 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.

5.2.3 Saving the Configuration Changes

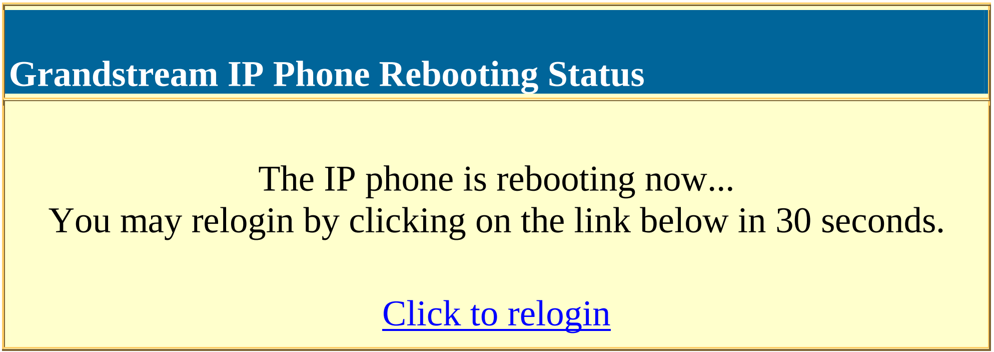
Once a change is made, the user should press the "Update" button in the Configuration Menu. The HandyTone ATA will then display the following screen to confirm that the changes have been saved.

Grandstream IP Phone Configuration Update Status
Your configuration changes have been saved. They will take effect on next reboot. Back to Home Page

The user is recommended to power cycle the HandyTone ATA after seeing the above message.

5.2.4 Rebooting the HandyTone ATA from remotely

The administrator of the HandyTone ATA can remotely reboot the HandyTone ATA 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 HandyTone ATA after waiting for about 30 seconds.

5.3 Configuration through a Central Server

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

With GAPS, a service provider or an enterprise with large deployment of HandyTone ATAs can easily manage the configuration and service provisioning of individual devices remotely and automatically via a central server. GAPS uses enhanced (NAT friendly) tftp and other communication protocols to communicate with each individual HandyTone ATA even if the HandyTone ATA is behind a NAT.

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

For details on how GAPS works, please refe

6 Software Upgrade

6.1 Upgrade with TFTP

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

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

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

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

TFTP checking is only performed during the initial power up. If the configured tftp server is found and a new code image is available, the HandyTone ATA will attempt to retrieve the new image files by downloading them into the HandyTone ATA's SRAM. During this stage, the HandyTone ATA's LEDs will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will then be saved into the Flash. If TFTP fails for any reason (e.g., TFTP server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the HandyTone ATA will stop the TFTP process and simply boot using the existing code image in the flash.

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