

User Manual

HandyTone-488

Analog Telephone Adaptor

For SW Release Version 1.0.3.18



Grandstream Networks, Inc.

www.grandstream.com

Table of Contents

1	WELCOME	4
2	INSTALLATION	5
3	WHAT IS INCLUDED IN THE PACKAGE	7
3.1	SAFETY COMPLIANCES	7
3.2	WARRANTY	7
4	PRODUCT OVERVIEW	8
4.1	KEY FEATURES.....	8
4.2	HARDWARE SPECIFICATION	9
5	BASIC OPERATIONS	10
5.1	GET FAMILIAR WITH VOICE PROMPT.....	10
5.2	MAKE PHONE CALLS.....	11
5.2.1	<i>Calling phone or extension numbers via VoIP FXS port</i>	11
5.2.2	<i>Direct IP calls</i>	11
5.2.3	<i>Call hold</i>	12
5.2.4	<i>Call waiting</i>	12
5.2.5	<i>Blind Transfer</i>	12
5.2.6	<i>Attended Transfer</i>	13
5.2.7	<i>3-way Conferencing</i>	13
5.2.8	<i>PSTN Pass Through/life line</i>	13
5.2.9	<i>VoIP-to-PSTN Calls</i>	13
5.2.10	<i>PSTN-to-VoIP Calls</i>	14
5.2.11	<i>Route Calls to PSTN</i>	15
5.3	CALL FEATURES	15
5.3.1	<i>Call Features Table</i>	15
5.4	FAX	16
5.5	LED LIGHT PATTERN INDICATION	16
6	CONFIGURATION GUIDE	18
6.1	CONFIGURING HANDYTONE-488 WAN IP THROUGH VOICE PROMPT.....	18
6.1.1	<i>DHCP Mode</i>	18
6.1.2	<i>STATIC IP Mode</i>	18
6.1.3	<i>TFTP Server Address</i>	18
6.2	CONFIGURING HANDYTONE-488 WITH WEB BROWSER	18
6.2.1	<i>Access the Web Configuration Menu</i>	18
6.2.2	<i>End User Configuration</i>	19
6.2.3	<i>Advanced User Configuration</i>	24
	<i>General settings have same meaning as explained in above section for FXS port page. Special settings on FXO port are explained below</i>	33
6.2.4	<i>Saving the Configuration Changes</i>	34
6.2.5	<i>Rebooting the HandyTone-488 from Remote</i>	34

- 6.3 CONFIGURATION THROUGH A CENTRAL SERVER..... 35
- 7 SOFTWARE UPGRADE..... 36**
 - 7.1 FIRMWARE UPGRADE THROUGH TFTP/HTTP..... 36
 - 7.2 CONFIGURATION FILE DOWNLOAD 37
 - 7.3 FIRMWARE AND CONFIGURATION FILE PREFIX AND POSTFIX..... 37
 - 7.4 MANAGING FIRMWARE AND CONFIGURATION FILE DOWNLOAD 37
- 8 RESTORE FACTORY DEFAULT SETTING..... 38**
- 9 GLOSSARY OF TERMS..... 39**

1 Welcome

Congratulations on becoming an owner of HandyTone-488. You made an excellent choice and we hope you enjoy all of its capabilities.

Grandstream's HandyTone-488 is an all-in-one VoIP integrated access device that features superb audio quality, rich functionalities, high level of integration, compactness and ultra-affordability. The HandyTone-488 is fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market.

Grandstream HandyTone-488 is a new addition to the popular HandyTone product family. It is an enhanced model compared to the award-winning HandyTone-486 in that it allows call origination and termination from/to the PSTN network (via FXO port) remotely and automated emergency call routing through PSTN network.

Grandstream HandyTone-488 has been awarded the Best of Show product in 2005 Internet Telephony Conference and Expo.

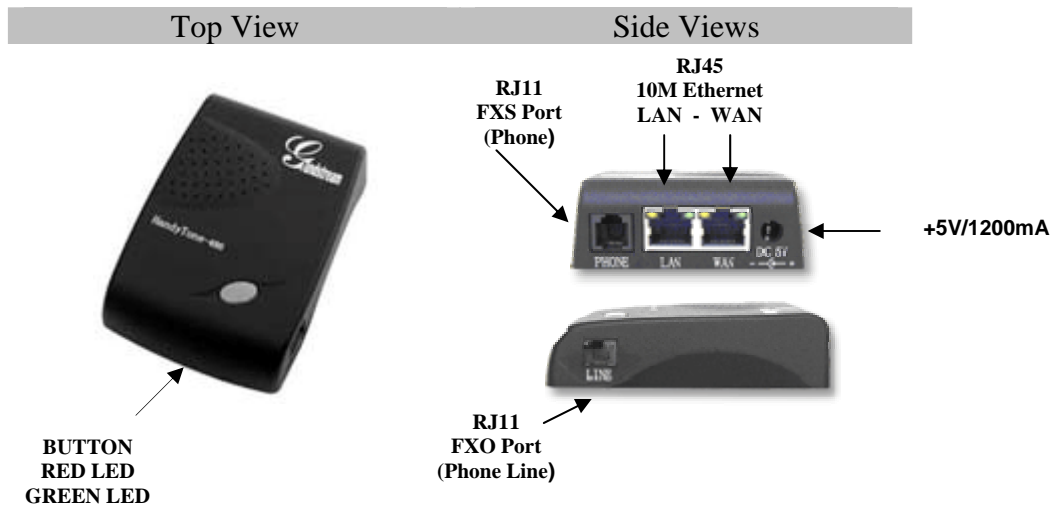


2 Installation

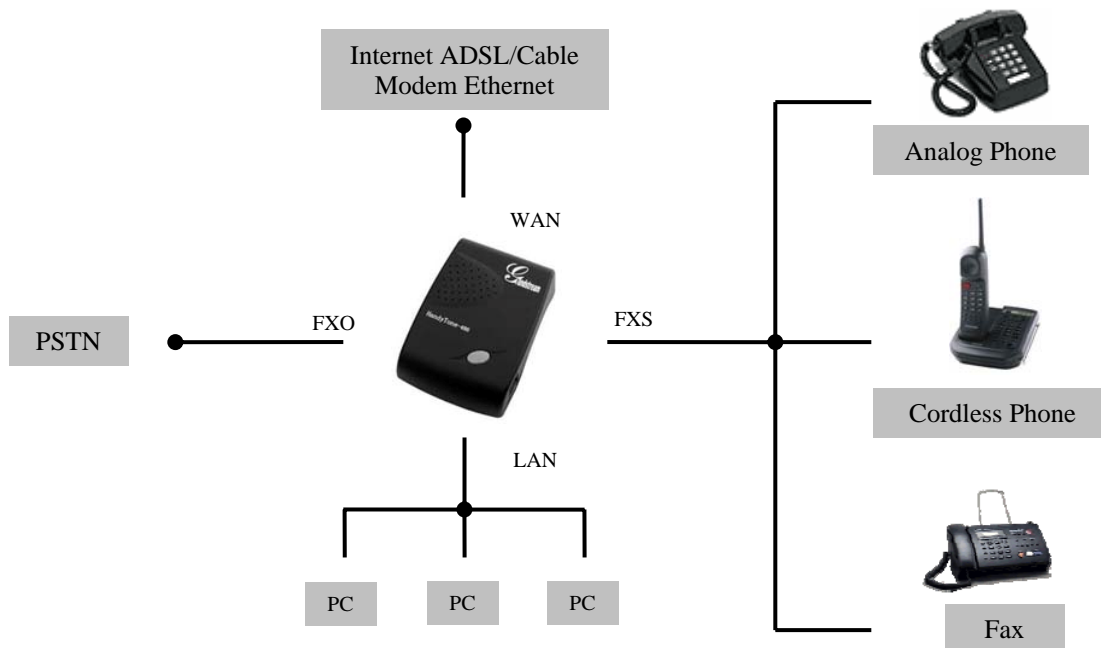
HandyTone-488 Analog Telephone Adaptor is an all-in-one VoIP integrated device designed to be a total solution for networks providing VoIP services.

The HandyTone-488 VoIP functionalities are available via a regular analog telephone.

The following photo illustrates the appearance of a HandyTone-488.



Interconnection Diagram of the HandyTone-488:



HandyTone-488 has one FXS port and one FXO port. The PHONE port next to the LAN port is a FXS port. The LINE port on the side of the HandyTone-488 is a FXO port. Both the FXS port and the FXO port can have a separate SIP account. This is a key feature of HandyTone-488 as it supports simultaneous call on both FXS port and FXO port. Telephone calls can be originated from or terminated on the PSTN network via FXO port remotely.

Following are the steps to install a HandyTone-488:

1. Connect a standard touch-tone analog telephone to the PHONE port.
2. Insert a standard RJ11 telephone cable into the LINE port and connect the other end of the telephone cable to a wall jack.
3. Insert the Ethernet cable into the WAN port of HandyTone-488 and connect the other end of the Ethernet cable to an uplink port (a router or a modem, etc.)
4. Connect a PC to the LAN port of HandyTone-488 if HT488 is used as a router.
5. Insert the power adapter into the HandyTone-488 and connect it to a wall outlet.

Please follow the instructions in section 6.2 to configure the HandyTone-488.

3 What is Included in the Package

The HandyTone-488 package contains:

- 1) One HandyTone-488
- 2) One universal power adaptor
- 3) One Ethernet cable

3.1 Safety Compliances

The HandyTone-488 is compliant with various safety standards including FCC/CE and C-tick. Its power adaptor is compliant with UL standard. The HandyTone-488 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-488 and will void the manufacturer warranty.

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

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4 Product Overview

4.1 Key Features

- Supports SIP 2.0(RFC 3261), TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP/RARP, DNS, DHCP (both client and server), NTP, PPPoE, STUN, TFTP, etc.
- Built-in router, NAT, Gateway and DMZ port forwarding
- Supports call origination and termination from/to the PSTN network (via FXO port)
- Powerful digital signal processing (DSP) to ensure superb audio quality; advanced adaptive jitter control and packet loss concealment technology
- Support various vocoders including G.711 (PCM a-law and u-law), G.723.1 (5.3K/6.3K), G.726 (32K), as well as G.729A, and iLBC.
- Support advanced call features on FXS port, includes Caller ID/Name display or block, Hold, Call Waiting/Flash, Call Transfer, Call Forward, 3-way conference, in-band and out-of-band DTMF, etc.
- Support fax pass through (for PCMU and PCMA) and T.38 FoIP (Fax over IP).
- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168), and AGC (Automatic Gain Control)
- Support standard encryption and authentication (DIGEST using MD5 and MD5-sess)
- Support for Layer 2 (802.1Q VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ)
- Support automated NAT traversal without manual manipulation of firewall/NAT
- Support device configuration via built-in IVR, Web browser or central configuration file through TFTP or HTTP server
- Support firmware upgrade via TFTP or HTTP
- Support PSTN pass through (on Rev.2.0)
- 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-488.

<u>Model</u>	<u>HandyTone-488</u>
LAN interface	1xRJ45 10Base-T
WAN interface	1xRJ45 10Base-T
FXS telephone port	1xFXS
FXO port	1xFXO
Button	1
LED	Green and red color
Universal Switching Power Adaptor	Input: 100-240VAC 50-60 Hz Output: +5VDC, 1200mA, UL certified
Dimension	70mm (W) 130mm (D) 27mm (H)
Weight	0.6lbs (0.3kg)
Temperature	40 - 130°F 5 - 45°C
Humidity	10% - 90% (non-condensing)
Compliance	

5 Basic Operations

5.1 Get Familiar with Voice Prompt

HandyTone-488 has stored a voice prompt menu (Interactive Voice Response or IVR) for quick browsing and simple configuration. To enter this voice prompt menu, simply **press the button or dial “***” from the analog phone**. The following table shows how to use the voice prompt menu to configure the device.

Menu	Voice Prompt	Options
Main Menu	“Enter a Menu Option”	Enter “*” for the next menu option Enter “#” to return to the main menu Enter 01-06, 47, 86, 99 menu option
01	“DHCP Mode”, “Static IP Mode”	Enter “9” to toggle the selection <i>If user selects “Static IP Mode”, user need configure all the IP address information through menu 02 to 05. If user selects “Dynamic IP Mode”, the device will retrieve all IP address information from DHCP server automatically when user reboots the device.</i>
02	“IP Address “ + IP address	The current WAN IP address is announced Enter 12 digit new IP address if in Static IP Mode
03	“Subnet “ + IP address	Same as menu 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 be prompted a dial tone, then enter 12 digit IP address This menu can also be entered by pressing the button again (For details, see “4.2.2 Make a Direct IP Call”.)
99	“RESET”	Enter “9” to reboot the phone Enter encoded MAC address to restore factory default setting
	“Invalid Entry”	Automatically returns to main menu

Notes:

- 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 Call*” option and a dial tone is prompted
- “*” 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 of the digits are collected, they will be processed.
- For IP address input, ignore the dot and key in the digit directly, add 0 before octet with less than three digits. e.g.: IP: 192.168.1.10, key in: 192 168 001 010
- Key entry can not 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 via VoIP FXS port

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

- a) Dial the numbers directly and wait for 4 (default) seconds.
- b) Dial the numbers directly, and press # (assuming that “use #” as dial key is selected in web configuration).

Examples:

To dial another extension on the same proxy, such as 1008, simply pick up attached phone, dial 1008 and then press the # or wait for 4 seconds.

To dial a PSTN number such as 6266667890, you might need to enter in some prefix number followed by the phone number. Please check with your VoIP service provider to get the information. If you phone is assigned with a PSTN-like number such as 6265556789, most likely you just follow the rule to dial 16266667890 as if you were calling from a regular analog phone, followed by pressing the # or wait for 4 seconds.

5.2.2 Direct IP calls

Direct IP calling allows two parties, that is, a HandyTone ATA with an analog phone and another VoIP Device, to talk to each other in an ad hoc fashion without a SIP proxy. This kind of VoIP calls can be made between two parties if:

- both HandyTone ATA and other VoIP Device(i.e., another HandyTone ATA or Budgetone SIP phone or other VoIP unit) have public IP addresses, or
- both HandyTone ATA and other VoIP Device are on the same LAN using private IP addresses, or
- Both HandyTone ATA and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

To make a direct IP to IP call, first pick up the analog phone or turn on the speakerphone on the analog phone, then follow Section 4.1 with voice prompt 47, followed by the 12-digit target IP address. User will hear a voice prompt “Direct IP Calling” and a dial tone. Enter a 12-digit target IP address to make a call. Destination ports can be specified by using “*4” (encoding for “:”) followed by the port number.

Examples:

If the target IP address is 192.168.0.10, the dialing convention is

Voice Prompt with option 47, then 192 168 000 010

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

If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be:

Voice Prompt with option 47, then 192168001020*45062 followed by pressing the “#” key if it is configured as a send key or wait for 4 seconds.

5.2.3 Call hold

This function is applicable on FXS port for VoIP calls only. While in conversation, pressing the “FLASH” button on the phone will put the remote end on hold. Pressing the “FLASH” button again will release the previously Hold state and the bi-directional media resume.

5.2.4 Call waiting

This function is applicable on FXS port for VoIP calls only. If call waiting feature is enabled, while the user is in a conversation, he will hear a special stutter tone if there is another incoming call. User can press the flash button to put the current call party on hold and switch to the other call. Pressing “FLASH” button essentially becomes toggling between two active calls.

5.2.5 Blind Transfer

This function is applicable on FXS port for VoIP calls only. Assume that call party A and B are in conversation. A wants to *Blind Transfer* B to C:

1. A press “FLASH” on the analog phone to hear the dial tone.
2. Then A dials *87 then dials C’s number, and then #(or wait for 4 seconds)
3. A can hang up.

NOTES: “Enable Call Feature” has to be set to YES in web configuration page. “Send flash event” needs to be set to NO.

A can hold on to the phone and wait one of the three following behaviors:

- A quick confirmation tone (temporarily using the call waiting indication tone) follows by a dial tone. This indicates the transfer has been successful.
- A quick busy tone followed by a restored call (On supported platforms only). This means that B is transferred to C with no success. The busy tone is just to indicate A that the transfer has failed.
- Busy tone keeps playing. This means A have failed to receive the second NOTIFY from B and decide to time out. Note: this does not indicate the transfer has been successful, nor does it indicate the transfer has failed. When B is a client that does not support the second NOTIFY, this will be the case. In bad network scenarios, this could also happen, although the transfer may have been completed successfully.

5.2.6 Attended Transfer

This function is applicable on FXS port for VoIP calls only. 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 to get a dial tone
2. A then dial C’s number followed by # (or wait for 4 seconds).
3. If C answers the call, A and C are in conversation. Then A hang up to complete transfer.
4. If C does not answer the call, A can press “FLASH” back to talk to B.

NOTES:

- *When attended Transfer failed and A hang up, the HandyTone 488 will ring user A back again to remind A that B is still on the call. A can pick up the phone to restore conversation with B.*

5.2.7 3-way Conferencing

This function is applicable on FXS port for VoIP calls only. Assuming that call party A and B are in conversation. A wants to bring C in a conference:

1. A press “FLASH” button on the analog phone to get a dial tone.
2. A dials *23 followed by C’s number and # (or wait for 4 seconds).
3. If C answers the call, then A press “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.

NOTES: “Enable Call Feature” has to be set to YES in web configuration page.

5.2.8 PSTN Pass Through/life line

HandyTone-488 supports PSTN pass through. The user can send and receive PSTN call with attached analog phone.

To receive PSTN calls, simply make phone off hook when the analog phone rings.

To make a PSTN call, simply press the PSTN access code (*00 is default, or any number configured in web configuration page) to switch to the PSTN line and get dial tone, then dial the PSTN number.

When HandyTone-488 is out of power, it will function as a jack. The user will be automatically connected to the PSTN Line.

5.2.9 VoIP-to-PSTN Calls

This function is applicable on FXO port that functions as a bridge between VoIP and PSTN. The user can remotely use PSTN line to initiate a call.

To make a VoIP-to-PSTN call:

1. Dial the FXO SIP account phone number to establish the VoIP session. The caller will hear the ring back tone once. Then the caller hears either a special continuous tone or a dial tone. The special continuous tone is played if the pin code is configured, or the dial tone otherwise.
2. Enter in the pin code that is configurable on the configuration page. The caller will hear the dial tone and get connected to the PSTN line if the pin code is valid, otherwise the continuous tone is played again to prompt caller to enter in the pin code again. The user may try up to 3 times to enter in pin code, if none is valid, HT488 will hang up.
3. After the caller hears dial tone from PSTN line, the caller can start dialing number to make calls.

Note:

- Users can choose whether apply password protection for VoIP-to-PSTN calls or not. A PIN (Pin for PSTN calls) consists of up to 8 numeric digits can be configured through BASIC SETTINGS of the web configuration page. By default, there is no password protection, i.e. there is no authentication required for callers on the use of PSTN line through HT488.
- When a PIN is configured for VOIP-to-PSTN call flow, the VoIP device that calls into the HT488 FXO account needs to configure RFC2833 or SIP Info for DTMF digit transmission.
- Upon hearing the special continuous tone for PIN code input, if the caller don't enter any digit, HT488 will time out and hang up the call in 10 seconds. During any stage of DTMF digits input, a 4 seconds timeout is applied to serve as an end of PIN or destination number input. Users may also use the “#” key to indicate the end of an input.
- On the web configuration page, if the “*Forward to PSTN*” is configured, the second stage dialing is eliminated, i.e., after dialing into the FXO SIP account number, the PSTN number will be called automatically.

5.2.10 PSTN-to-VoIP Calls

This function is applicable on FXO port that functions as a bridge between VoIP and PSTN. The user can make VoIP calls remotely by dialing into FXO Line port on HT488.

To make a PSTN-to-VoIP call:

1. Make an incoming call to the PSTN line on FXO port. The attached analog phone will ring for 4 times by default, this setting is configurable on the configuration page.
2. If no one picks up the phone on FXS port after 4 rings, then the caller hears either a special continuous tone or a dial tone. The continuous tone is played if the pin code is configured, or the dial tone otherwise.
3. Enter in the pin code that is configurable on the configuration page. The caller will hear the dial tone and get bridged to VoIP if the pin code is valid, otherwise the continuous tone is played again to prompt caller to enter in the pin code again. The user may try up to 3 times to enter in pin code, if none is valid, HT488 will hang up.
4. The caller can dial a VoIP number followed by # (or wait for 4 seconds), the VoIP call will be initiated from the SIP account configured on the FXO port.

Note:

- Users can choose whether apply password protection for PSTN-to-VoIP calls or not. A PIN (Pin to VoIP calls) consists of up to 8 numeric digits can be configured through BASIC SETTINGS of the web configuration page. By default, there is no password protection, i.e. there is no authentication required for callers on the use of VoIP SIP account on FXO port.
- Upon hearing the special continuous tone for PIN code input, if the caller don't enter any digit, HT488 will time out and hang up the call in 10 seconds. During any stage of DTMF digits input, a 4 seconds timeout is applied to serve as an end of PIN or destination number input. Users may also use the “#” key to indicate the end of an input.
- On the web configuration page, if the “*Forward to VoIP*” is configured, the second stage dialing is eliminated, i.e., after bridging to VoIP, the configured VoIP number will be called automatically.

5.2.11 Route Calls to PSTN

This function is applicable on FXO port that can access the PSTN network. By default, HT488 is in VoIP mode upon off-hook. If “*Route call to PSTN*” is configured, certain calls will be initiated from FXO PSTN line port. This call feature is especially useful for emergency calls or local telephone calls. To use this feature, users need to specify a prefix or a telephone number in the “*Route call to PSTN*” on BASIC SETTINGS web configuration page. If the dialed digits match one of the specified prefix, outbound calls will be initiated from PSTN line.

For ex, if “*Route call to PSTN*” is configured to be 626, all outgoing calls start with 626 will be initiated from PSTN line.

5.3 Call Features

All the call feature codes are applicable to FXS port for VoIP calls only.

5.3.1 Call Features Table

Following table shows the call features of HandyTone-488.

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

HandyTone-488 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 tables show the LED light pattern indication.

RED LED always indicates not abnormal status	
DHCP Failed or WAN No Cable	Button flashes every 2 seconds (if DHCP is configured)
HandyTone-488 fails to register	Button flashes every 2 seconds (if SIP server is configured)
Firmware Upgrading	Button flashes every 2 seconds
Device Malfunctions	Red light steady on

GREEN LED mostly indicates normal working status	
Message Waiting Indication	Button flashes every 2 seconds

RINGING	Button flashes at 1/10 second
RINGING INTERVAL	Button flashes every second
In Conversation	Green light steady on

6 Configuration Guide

6.1 Configuring HandyTone-488 WAN IP through Voice Prompt

6.1.1 DHCP Mode

Follow section 5.1 with voice menu option 01 to enable HandyTone-488 to use DHCP.

6.1.2 STATIC IP Mode

Follow section 5.1 with voice menu option 01 to enable HandyTone-488 to use STATIC IP mode, then use option 02, 03, 04, 05 to set up HandyTone-488's IP, Subnet Mask, Gateway, and DNS server respectively.

6.1.3 TFTP Server Address

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

6.2 Configuring HandyTone-488 with Web Browser

HandyTone series ATA 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 HandyTone-488 through a Web browser such as Microsoft's IE and AOL's Netscape.

6.2.1 Access the Web Configuration Menu

The HandyTone-488 HTML configuration page can be accessed via LAN or WAN port:

- From the LAN port:
 - Directly connect a computer to the LAN port.
 - Open a command window on the computer
 - Type in "ipconfig /release", the IP address etc becomes 0.
 - Type in "ipconfig /renew", the computer gets an IP address in 192.168.2.x segment by default
 - Open a web browser, type in the default gateway IP address. You will see the log in page of the device.

<http://192.168.2.1>

- From the WAN port:
 - Follow section 5.1 to find out the WAN side IP address.
 - Open a web browser, type in the WAN side IP address.

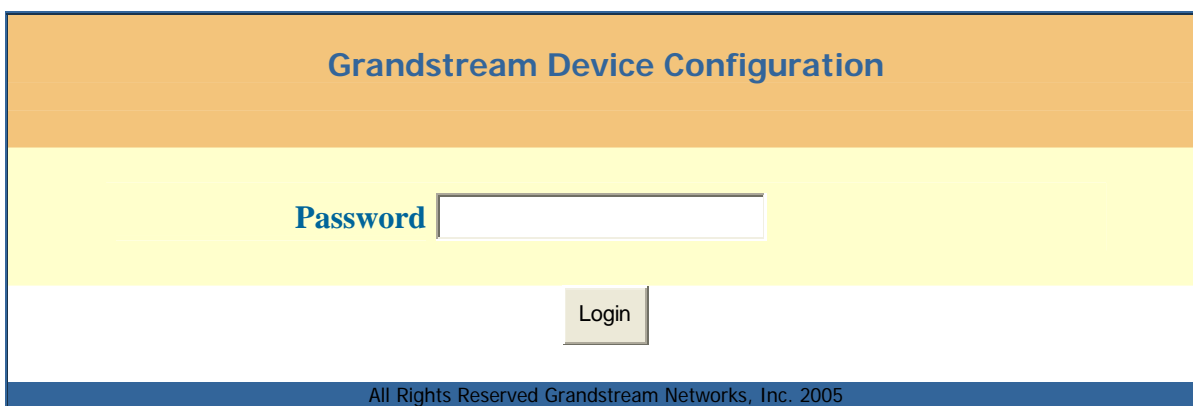
<http://HandyTone-WAN-IP-Address>,

Note:

- *WAN side HTTP access is by default disabled for security reason. You can enable it on the configuration page by setting “WAN side HTTP access” to be YES. The very first time to access the configuration page is always from LAN port. The instructions are listed above.*
- *The IVR announces 12 digits IP address, you need to strip out the leading “0” in IP address. For ex. IP address: 192.168.001.014, you need to type in <http://192.168.1.14> in the web browser.*

6.2.2 End User Configuration

- ❖ Log in page.



Grandstream Device Configuration

Password

Login

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The password is case sensitive with maximum length of 25 characters. The factory default password for End User and administrator is “123” and “admin” respectively. Only administrator can get access to “ADVANCED SETTING” configuration page.

Note:

- *If you can not log into the configuration page by using default password, please check with the voip service provider. Most likely the service provider has provisioned the device and changed the login password.*

- ❖ Status Page

Grandstream Device Configuration					
	STATUS	BASIC SETTINGS	ADVANCED SETTINGS	FXS PORT	FXO PORT
MAC Address:	00.0B.82.03.DA.36				
WAN IP Address:	10.10.11.225				
Product Model:	HT488				
Software Version:	Program-- 1.0.3.18 Bootloader-- 1.0.8.9 HTML-- 1.0.3.18 VOC-- 1.0.0.10				
System Up Time:	0 day(s) 1 hour(s) 47 minute(s)				
Registered:	Yes				
PPPoE Link Up:	disabled				
NAT:	detected NAT type is full cone				
All Rights Reserved Grandstream Networks, Inc. 2005					

MAC Address	The unique 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, such as HT488
Software Version	Program: This is the main software release. <i>This number is always used for firmware upgrade.</i> Current release is 1.0.3.18. Bootloader: current version is 1.0.8.9. HTML: current version 1.0.3.18 VOC: current version is 1.0.0.10
System Uptime	This shows system up time since last reboot.
Registered	Whether the unit is registered to the SIP server.
PPPoE Link Up	This shows whether the PPPoE link is up if connected to DSL modem
NAT	This shows what kind of NAT the HandyTone is behind. NAT detection mechanism is based on STUN protocol.

❖ Basic settings page

Grandstream Device Configuration										
STATUS	BASIC SETTINGS	ADVANCED SETTINGS	FXS PORT	FXO PORT						
End User Password:	<input type="text"/> (purposely not displayed for security protection)									
IP Address:	<input checked="" type="checkbox"/> dynamically assigned via DHCP (default) or PPPoE (will attempt PPPoE if DHCP fails and following is non-blank) <table border="1"><tr><td>PPPoE account ID:</td><td><input type="text"/></td></tr><tr><td>PPPoE password:</td><td><input type="text"/></td></tr><tr><td>PPPoE Service Name:</td><td><input type="text"/></td></tr></table> Preferred DNS server: <input type="text"/> 0 <input type="text"/> 0 <input type="text"/> 0 <input type="text"/> 0 <input type="checkbox"/> statically configured as: IP Address: <input type="text"/> 192 <input type="text"/> 168 <input type="text"/> 0 <input type="text"/> 160 Subnet Mask: <input type="text"/> 0 <input type="text"/> 0 <input type="text"/> 0 <input type="text"/> 0 Default Router: <input type="text"/> 0 <input type="text"/> 0 <input type="text"/> 0 <input type="text"/> 0 DNS Server 1: <input type="text"/> 0 <input type="text"/> 0 <input type="text"/> 0 <input type="text"/> 0 DNS Server 2: <input type="text"/> 0 <input type="text"/> 0 <input type="text"/> 0 <input type="text"/> 0				PPPoE account ID:	<input type="text"/>	PPPoE password:	<input type="text"/>	PPPoE Service Name:	<input type="text"/>
PPPoE account ID:	<input type="text"/>									
PPPoE password:	<input type="text"/>									
PPPoE Service Name:	<input type="text"/>									
Time Zone:	<input type="text"/> GMT-5:00 (US Eastern Time, New York) <input type="button" value="v"/>									
Daylight Savings Time:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (if set to Yes, display time will be 1 hour ahead of normal time)									
NAT/DHCP Server Information & Configuration:										
<i>Cloned WAN MAC Addr:</i>	<input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> (in hex format)									
<i>LAN Subnet Mask:</i>	<input type="text"/> 255.255.255.0 (default is 255.255.255.0)									
<i>LAN DHCP Base IP:</i>	<input type="text"/> 192.168.2.1 (base IP for the LAN port, default is 192.168.2.1)									
<i>DHCP IP Lease Time:</i>	<input type="text"/> 120 (in units of hours, default is 120 hours or 5 days)									
<i>DMZ IP:</i>	<input type="text"/>									
<i>Port</i>	WAN port <input type="text"/> 0	LAN IP <input type="text"/>	LAN port <input type="text"/> 0	Protocol <input type="text"/> UDP Only <input type="button" value="v"/>						

<i>Forwarding:</i>	WAN port	<input type="text" value="0"/>	LAN IP	<input type="text"/>	LAN port	<input type="text" value="0"/>	Protocol	UDP Only <input type="button" value="v"/>
	WAN port	<input type="text" value="0"/>	LAN IP	<input type="text"/>	LAN port	<input type="text" value="0"/>	Protocol	UDP Only <input type="button" value="v"/>
	WAN port	<input type="text" value="0"/>	LAN IP	<input type="text"/>	LAN port	<input type="text" value="0"/>	Protocol	UDP Only <input type="button" value="v"/>
	WAN port	<input type="text" value="0"/>	LAN IP	<input type="text"/>	LAN port	<input type="text" value="0"/>	Protocol	UDP Only <input type="button" value="v"/>
	WAN port	<input type="text" value="0"/>	LAN IP	<input type="text"/>	LAN port	<input type="text" value="0"/>	Protocol	UDP Only <input type="button" value="v"/>
	WAN port	<input type="text" value="0"/>	LAN IP	<input type="text"/>	LAN port	<input type="text" value="0"/>	Protocol	UDP Only <input type="button" value="v"/>
	WAN port	<input type="text" value="0"/>	LAN IP	<input type="text"/>	LAN port	<input type="text" value="0"/>	Protocol	UDP Only <input type="button" value="v"/>
<i>Number of Rings:</i>	<input type="text" value="4"/>	(number of phone rings before a PSTN incoming call is forwarded, default 4)						
<i>PSTN access code:</i>	<input type="text"/> (key pattern to use PSTN line, default is "*00")							
<i>PIN for PSTN Calls:</i>	<input type="text"/> (Enter digits to authorize calling PSTN numbers from VOIP, no default)							
<i>PIN for VOIP Calls:</i>	<input type="text"/> (Enter digits to authorize calling VOIP terminals from PSTN, no default)							
<i>Route Call to PSTN:</i>	Outbound calls will be routed to PSTN port when dialed digits match one of the following:							
	<input type="text"/>							
	<input type="text"/>							
	<input type="text"/>							
	<input type="text"/>							
<i>Forward to PSTN:</i>	<input type="text"/> (VoIP calls will be forwarded to the specified PSTN number if ring no answer)							
<i>Forward to VoIP:</i>	<input type="text"/> (PSTN calls will be forwarded to the specified VoIP number)							
<input type="button" value="Update"/>								

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End User Password

This contains the password to access the Web Configuration Menu. This field is case sensitive with max. 25 characters

IP Address	<p>This setting is for the WAN port.</p> <ul style="list-style-type: none"> - If DHCP mode is enabled, then all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The HandyTone ATA will acquire its IP address from DHCP in the network. <p>PPPoE settings is usually for DSL/ADSL modem users. The HandyTone will attempt to establish a PPPoE session if PPPoE account is set.</p> <ul style="list-style-type: none"> - If Static IP mode is selected, the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (mandatory), DNS Server 2 (optional) fields need to be configured.
Time Zone	Displayed date/time will be adjusted according to the specified time zone.
Daylight Savings Time	Default NO. If set to Yes, then the displayed time will be 1 hour ahead of normal time.
Cloned WAN MAC Address:	Allow the user to set a specific MAC address. Set in Hex format
LAN Subnet Mask	Sets the LAN subnet mask. Default value is 255.255.255.0
LAN DHCP Base IP:	Base IP for the LAN port, which functions as default gateway for its LAN. Default value is 192.168.2.1
DHCP IP Lease Time:	Value is set in units of hours. Default value is 120hr (5 Days.) The time IP address are assigned to the LAN clients
DMZ IP:	Forward all WAN IP traffic to a specific IP address if no matching port is used by HandyTone-488 itself or in the defined port forwarding.
Port Forwarding:	Allow users to forward a matching (TCP/UDP) port to a specific LAN IP address with a specific (TCP/UDP) port.
Number of rings	Default is 4. It specifies number of phone rings before a PSTN incoming call is bridged to VoIP
PSTN access code	The code to access the PSTN line. Default is “*00”.
PIN for PSTN calls	PIN code to bridge from VoIP to PSTN
PIN for VoIP calls	PIN code to bridge from PSTN to VoIP
Route Call to PSTN	If the dialed digits match one of the specified prefix here, outbound calls will be initiated from PSTN line. This field is especially useful for emergency calls.

Forward to PSTN	Calls are unconditionally forwarded to the specified PSTN phone number for all incoming VoIP calls on FXO port.
Forward to VoIP	Calls are unconditionally forwarded to the specified VoIP phone number for all incoming PSTN calls.

6.2.3 Advanced User Configuration

To login to the Advanced User Configuration page, please follow the instructions in section 6.2.1 to get to the following login page. The password is case sensitive and the factory default password for Advanced User is “admin”.

❖ Advanced settings

Grandstream Device Configuration	
	STATUS BASIC SETTINGS ADVANCED SETTINGS FXS PORT FXO PORT
Admin Password:	<input type="text"/> (purposely not displayed for security protection)
<i>Layer 3 QoS:</i>	<input type="text" value="48"/> (Diff-Serv or Precedence value)
<i>Layer 2 QoS:</i>	802.1Q/VLAN Tag <input type="text" value="0"/> 802.1p priority value <input type="text" value="0"/> (0-7)
<i>No Key Entry Timeout:</i>	<input type="text" value="4"/> (in seconds, default is 4 seconds)
<i>NAT Traversal:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes, STUN server is: <input type="text"/> (URI or IP:port)
<i>keep-alive interval:</i>	<input type="text" value="20"/> (in seconds, default 20 seconds)
<i>Use NAT IP:</i>	<input type="text"/> (used in SIP/SDP message if specified)
<i>Firmware Upgrade and Provisioning:</i>	Upgrade Via <input type="checkbox"/> TFTP <input checked="" type="checkbox"/> HTTP Firmware Server Path: <input type="text" value="fm.grandstream.com/gs"/> Config Server Path: <input type="text" value="fm.grandstream.com/gs"/> Firmware File Prefix: <input type="text"/> Firmware File Postfix: <input type="text"/> Config File Prefix: <input type="text"/> Config File Postfix: <input type="text"/> Automatic Upgrade: <input checked="" type="checkbox"/> No <input type="checkbox"/> Yes, check for upgrade every <input type="text" value="10080"/> minutes (default 7 days)

	<input checked="" type="checkbox"/> Always Check for New Firmware <input type="checkbox"/> Check New Firmware only when F/W pre/suffix changes
<i>Firmware Key:</i>	<input type="text"/> (in Hexadecimal Representation)
<i>NTP Server:</i>	<input type="text" value="time.nist.gov"/> (URI or IP address)
<i>WAN side http access:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes (WAN side access to http server will be rejected if set to No)
<input type="button" value="Update"/>	
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Admin Password	Administrator password. Only administrator can configure the “Advanced Settings”, FXS port and FXO port page. Password field is purposely blanked for security reason after clicking UPDATE button. The maximum password length is 25 characters.
Layer 3 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.
Layer 2 QoS	Layer 2 QoS settings. Default setting is blank. VLAN supported equipment is required if user needs to change these settings.
No Key Entry Timeout	This parameter allows 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. If set to “No”, the “#” key will then be included as part of the dialed string to be sent out.
NAT Traversal	Defines whether the NAT traversal mechanism is activated. It should be set to YES if the device is behind NAT router. If outbound proxy is not configured, STUN server needs to be set to activate STUN detection mechanism. Usually ITSP will provide these settings. If this field is set to “Yes”, then the device will periodically (every <i>Keep-alive interval</i>) send a dummy UDP packet to the SIP server to pinhole the NAT.
Keep-alive interval	Default is 20 seconds. The interval of sending dummy UDP packet to keep NAT “pin hole” open.
Use NAT IP:	NAT IP address used in SIP/SDP message. Default is blank.
Firmware Upgrade and Provisioning	Default method is HTTP. Firmware upgrade may take up to 10 minutes depending 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 is blank. If configured, HT488 will request the firmware file with the prefix. This setting is useful for ITSPs. End user should keep it blank.
Firmware File Postfix	Default is blank. End user should keep it blank.
Config File Prefix	Default is blank. End user should keep it blank.
Config File Postfix	Default is blank. End user should keep it blank.
Automatic Upgrade	Default is “Yes”.
Firmware Key	For firmware encryption. It should be 32 digit in Hexadecimal Representation. End user should keep it blank.
NTP server	URI or IP address of the NTP (Network Time Protocol) server, which the HandyTone ATA will use to synchronize the date/time.
WAN side http access	Default is “No”. The access to configuration page via WAN port is disabled. Need to change to “Yes” if user wants WAN side HTTP access to the ATA.

❖ FXS port page

Grandstream Device Configuration	
	STATUS BASIC SETTINGS ADVANCED SETTINGS FXS PORT FXO PORT
SIP Server:	<input type="text" value="sip.mycomany1.com"/> (e.g., sip.mycompany.com, or IP address)
Outbound Proxy:	<input type="text"/> (e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID:	<input type="text" value="123456789"/> (the user part of an SIP address)
Authenticate ID:	<input type="text" value="123456789"/> (can be identical to or different from SIP User ID)
Authenticate Password:	<input type="text"/> (purposely not displayed for security protection)
Name:	<input type="text" value="John Doe-1"/> (optional, e.g., John Doe)
Use DNS SRV:	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes
User ID is phone	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes

<i>number:</i>	
<i>SIP Registration:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes
<i>Unregister On Reboot:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes
<i>Register Expiration:</i>	<input type="text" value="60"/> (in minutes. default 1 hour, max 45 days)
<i>local SIP port:</i>	<input type="text" value="5060"/> (default 5060)
<i>local RTP port:</i>	<input type="text" value="5004"/> (1024-65535, default 5004)
<i>Use random port:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
<i>DTMF Payload Type:</i>	<input type="text" value="101"/>
<i>Send DTMF:</i>	<input type="checkbox"/> in-audio <input checked="" type="checkbox"/> via RTP (RFC2833) <input type="checkbox"/> via SIP INFO
<i>Send Flash Event:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (Flash will be sent as a DTMF event if set to Yes)
<i>Enable Call Features:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally)
<i>Offhook Auto-Dial:</i>	<input type="text"/> (User ID/extension to dial automatically when offhook)
<i>Proxy-Require:</i>	<input type="text"/>
<i>Disable Call-Waiting:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
<i>Preferred Vocoder: (in listed order)</i>	choice 1: <input pcmu"="" type="text" value="current setting is "/> <input type="button" value="v"/> choice 2: <input pcma"="" type="text" value="current setting is "/> <input type="button" value="v"/> choice 3: <input g723"="" type="text" value="current setting is "/> <input type="button" value="v"/> choice 4: <input g729"="" type="text" value="current setting is "/> <input type="button" value="v"/> choice 5: <input g726-32"="" type="text" value="current setting is "/> <input type="button" value="v"/> choice 6: <input ilbc"="" type="text" value="current setting is "/> <input type="button" value="v"/>
<i>Voice Frames per TX:</i>	<input type="text" value="2"/> (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
<i>G723 rate:</i>	<input checked="" type="checkbox"/> 6.3kbps encoding rate <input type="checkbox"/> 5.3kbps encoding rate
<i>iLBC frame size:</i>	<input checked="" type="checkbox"/> 20ms <input type="checkbox"/> 30ms
<i>iLBC payload type:</i>	<input type="text" value="97"/> (between 96 and 127, default is 97)
<i>Silence Suppression:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes

<i>Fax Mode:</i>	<input checked="" type="checkbox"/> T.38 (Auto Detect) <input type="checkbox"/> Pass-Through
<i>Early Dial:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (use "Yes" only if proxy supports 484 response)
<i>Dial Plan Prefix:</i>	<input type="text"/> (this prefix string is added to each dialed number)
<i>Use # as Dial Key:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
<i>SUBSCRIBE for MWI:</i>	<input checked="" type="checkbox"/> No, do not send SUBSCRIBE for Message Waiting Indication <input type="checkbox"/> Yes, send periodical SUBSCRIBE for Message Waiting Indication
<i>Send Anonymous:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes (caller ID will be blocked if set to Yes)
<i>Lock keypad update:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (configuration update via keypad is disabled if set to Yes)
<i>Special Feature:</i>	<input type="text" value="Standard"/>
<i>FXS Impedance:</i>	<input type="text" value="600 Ohm (North America)"/>
<i>Caller ID Scheme:</i>	<input type="text" value="Bellcore (North America)"/>
<i>Onhook Voltage:</i>	<input type="text" value="36V"/>
<i>Polarity Reversal:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (reverse polarity upon call establishment and termination)
<input type="button" value="Update"/>	

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SIP Server	SIP Server's Domain name or IP address provided by VoIP service provider.
Outbound Proxy	IP address or Domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by ATA for firewall or NAT penetration in different network environment. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy will provide solution for it. This information is provided by VoIP service provider.
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	ID used for authentication, usually same as SIP user ID, but could be different and decided by ITSP.
Authenticate Password	Account information, password for ATA to register to (SIP) servers of ITSP.

Name	SIP service subscriber's name which will be used for Caller ID display
Use DNS SRV:	Default is No. If set to Yes the client will use DNS SRV to lookup for the SIP server.
User ID is Phone Number	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 HandyTone ATA 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 all previous bindings. Use only if proxy supports this remove bindings request.
Register Expiration	This parameter allows the user to specify the time frequency (in minutes) the HandyTone ATA refreshes 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. The default value for FXS port is 5060.
Local RTP port	This parameter defines the local RTP-RTCP port pair the HandyTone ATA will listen and transmit. 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 for FXS port is 5004.
Use Random Port	Default No. If set to Yes, the device will pick randomly-generated SIP and RTP ports. This is usually necessary when multiple HandyTone ATAs are behind the same NAT.
DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833
Send DTMF	This parameter specify 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.
Send Flash Event	Default is NO. If set to yes, flash will be sent as DTMF event.
Enable Call Features	Default is Yes. Advance call features and feature codes functions are supported locally.

Offhook Auto-Dial	This parameter allows users 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. Note: Please write down the IP address of the ATA if you use this feature as it will prevent you to access the IVR and the only way to access the device configuration is via the web configuration page.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Disable Call Waiting	Default is No.
Preferred Vocoder	The HandyTone ATA supports 6 different Vocoder types including G.711 A-/U-law, G.723.1, G.726-32, G.729A, iLBC. Users can configure Vocoders in a preference list that will be included with the same preference order in SDP message.
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.
G723 Rate:	This defines the encoding rate for G723 vocoder. Default setting is 6.3kbps.
iLBC frame size:	This sets the iLBC size in 20ms or 30ms
iLBC payload type:	This defines payload type for iLBC. Default value 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.
Fax Mode	T.38 (Auto Detect) FoIP by default, or fax Pass-Through.

Early Dial	Default is No. Use only if proxy supports 484 response
Dial Plan Prefix	Sets the prefix added to each dialed number
Use # as Send Key	This parameter allows users to configure the “#” key to be used as the “Send” (or “Dial”) key. If set to “Yes”, pressing this key will immediately trigger the sending of dialed string collected so far. If set to “No”, this “#” key will then be included as part of the dial string to be sent out.
Subscribe for MWI:	Default is No. When set to “Yes” a SUBSCRIBE for Message Waiting Indication will be sent periodically.
Send Anonymous	If this parameter is set to “Yes”, user ID will be sent as 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.
Special Features	Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Nortel, Broadsoft, etc.
FXS Impedance	Selects the impedance of the analog telephone connected to the Phone port.
Caller ID Scheme	Select the Caller ID Scheme to suit the standard of different area. <ul style="list-style-type: none"> • Bellcore (North America) • CID - Canada • DTMF (Brazil) • DTMF (Denmark) • ETSI-DTMF (Finland, Sweden) • ETSI-FSK (France, Germany, Norway, Taiwan, UK-CCA)
Onhook Voltage	Select the onhook voltage to suit the analog phone.
Polarity Reversal	Select Polarity Reversal to adapt some call charge/billing system. Default is No.

❖ FXO port page

Grandstream Device Configuration					
	<u>STATUS</u>	<u>BASIC SETTINGS</u>	<u>ADVANCED SETTINGS</u>	<u>FXS PORT</u>	<u>FXO PORT</u>
SIP Server:		<input type="text" value="sip.mycompany2.com"/>			(e.g., sip.mycompany.com, or IP address)
Outbound Proxy:		<input type="text"/>			(e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID:		<input type="text" value="987654321"/>			(the user part of an SIP address)

Authenticate ID:	<input type="text" value="987654321"/>	(can be identical to or different from SIP User ID)
Authenticate Password:	<input type="password"/>	(purposely not displayed for security protection)
Name:	<input type="text" value="John Doe-2"/>	(optional, e.g., John Doe)
Use DNS SRV:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes	
User ID is phone number:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes	
SIP Registration:	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes	
Unregister On Reboot:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes	
Register Expiration:	<input type="text" value="10"/>	(in minutes. default 1 hour, max 45 days)
local SIP port:	<input type="text" value="5062"/>	(default 5062)
local RTP port:	<input type="text" value="5008"/>	(1024-65535, default 5008)
Use random port:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes	
DTMF Payload Type:	<input type="text" value="101"/>	
Send DTMF:	<input type="checkbox"/> in-audio <input checked="" type="checkbox"/> via RTP(RFC2833) <input type="checkbox"/> via SIP INFO	
Send Flash Event:	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes	(Flash will be sent as a DTMF event if set to Yes)
Proxy-Require:	<input type="text"/>	
Preferred Vocoder: (in listed order)	choice 1: <input pcmu"="" type="text" value="current setting is "/> <input type="button" value="v"/> choice 2: <input pcma"="" type="text" value="current setting is "/> <input type="button" value="v"/> choice 3: <input g723"="" type="text" value="current setting is "/> <input type="button" value="v"/> choice 4: <input g729"="" type="text" value="current setting is "/> <input type="button" value="v"/> choice 5: <input g726-32"="" type="text" value="current setting is "/> <input type="button" value="v"/> choice 6: <input ilbc"="" type="text" value="current setting is "/> <input type="button" value="v"/>	
Voice Frames per TX:	<input type="text" value="2"/>	(up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
G723 rate:	<input checked="" type="checkbox"/> 6.3kbps encoding rate <input type="checkbox"/> 5.3kbps encoding rate	
iLBC frame size:	<input checked="" type="checkbox"/> 20ms <input type="checkbox"/> 30ms	

<i>iLBC payload type:</i>	<input type="text" value="97"/> (between 96 and 127, default is 97)
<i>Silence Suppression:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
<i>Fax Mode:</i>	<input checked="" type="checkbox"/> T.38 (Auto Detect) <input type="checkbox"/> Pass-Through
<i>Early Dial:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (use "Yes" only if proxy supports 484 response)
<i>Dial Plan Prefix:</i>	<input type="text"/> (this prefix string is added to each dialed number)
<i>Use # as Dial Key:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
<i>PSTN AC Termination:</i>	<input type="text" value="320 Ohm + (1050 Ohm 230 nF)"/> impedance
<i>PSTN Disconnect Tone :</i>	Frequency: f1 <input type="text" value="480"/> f2 <input type="text" value="620"/> (Hz, 0 - inactive, default is 480Hz + 620Hz)
<i>PSTN Disconnect Tone Cadence:</i>	Choice 1: On <input type="text" value="0"/> Off <input type="text" value="0"/> (ms, 0 - disabled)
	Choice 2: On <input type="text" value="0"/> Off <input type="text" value="0"/> (ms, 0 - disabled)
	Choice 3: On <input type="text" value="0"/> Off <input type="text" value="0"/> (ms, 0 - disabled)
<i>PSTN Silence Timeout:</i>	<input type="text" value="60"/> (sec, terminate call after long silence detected, default is 60 sec, max 65536)
<input type="button" value="Update"/>	

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General settings have same meaning as explained in above section for FXS port page. Special settings on FXO port are explained below.

Local SIP port	The default value for FXO port is 5062.
Local RTP port	The default value for FXO port is 5008.
PSTN AC Termination	Selects the impedance of the analog PSTN line connected to the Line port.
PSTN Disconnect Tone	This configuration should be configured by the VoIP service provider. Some country use single frequency tone to signal PSTN disconnection, some country use double frequency tone.
PSTN Disconnect Tone Cadence	This setting can be configured to suit the telephone company's standard in different country.

PSTN Silence Timeout	Terminate call after long silence detected. Default setting is 60 sec, max 65536
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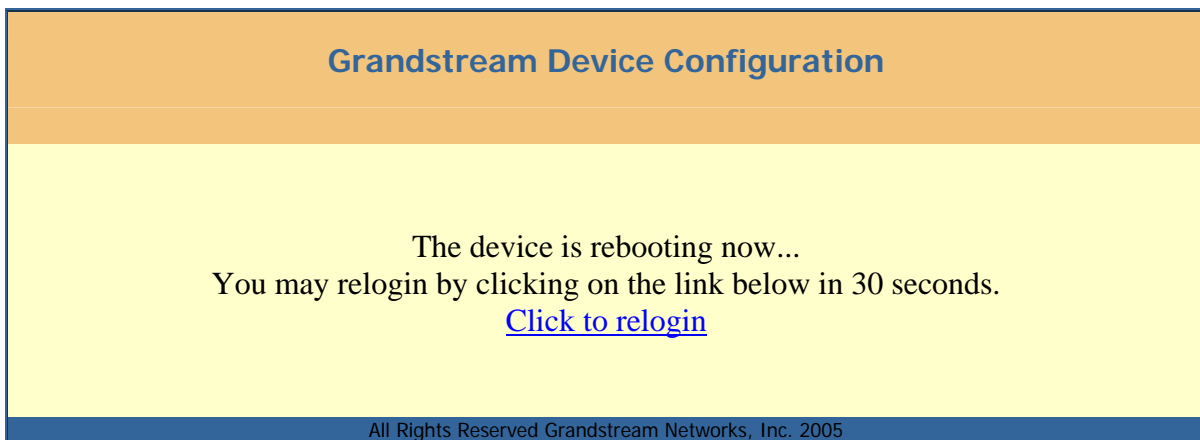
6.2.4 Saving the Configuration Changes

Once a change is made, users should click on the “Update” button on the Configuration page. The HandyTone ATA will then display the following screen to confirm that the changes have been saved.



6.2.5 Rebooting the HandyTone-488 from Remote

User can then power cycle the device or reboot the HandyTone ATA by clicking on the “Reboot” button at the bottom of the configuration page. Once done, the following screen will be displayed to indicate that rebooting is underway.



Note:

- DO NOT INTERRUPT THE BOOTING UP PROCESS, OR THE DEVICE COULD BE DAMAGED.

6.3 Configuration through a Central Server

Grandstream HandyTone ATAs 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 files can be downloaded via TFTP or HTTP from the central server. A service provider or an enterprise with large deployment of HandyTone ATA can easily manage the configuration and service provisioning of individual devices remotely from a central server.

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

Grandstream provide GAPS (Grandstream Automated Provisioning System) service to VoIP service providers. It could be either simple redirection or with certain special provisioning settings. Initially upon booting up, Grandstream devices by default point to Grandstream provisioning server GAPS, based on the unique MAC address of each device, GAPS provision the devices with redirection settings so that they will be redirected to customer’s TFTP or HTTP server for further provisioning. Grandstream also provide GAPSLite software package which contains our NAT friendly TFTP server and a configuration tool to facilitate the task of generating device configuration files.

The GAPSLite configuration tool is now free to end users. The tool and configuration template are available for download from http://www.grandstream.com/DOWNLOAD/Configuration_Tool/.

For details on how GAPS works, please contact Grandstream and refer to the documentation of GAPS product provided.

7 Software Upgrade

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

7.1 Firmware Upgrade through TFTP/HTTP

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

e.g. `firmware.mycompany.com:6688/Grandstream/1.0.3.18`

e.g. `168.75.215.189`

NOTES:

- TFTP server in IP address format can be configured via IVR. Please refer to section 6.1.3 for instructions. If TFTP server is in FQDN format, it must be set via web configuration interface.
- Once a “Firmware Server Path” is set, user needs to update the settings and reboot the device. If the configured firmware server is found and a new code image is available, the HandyTone ATA will attempt to retrieve the new image files by downloading them into the HandyTone ATA’s SRAM. During this stage, the HandyTone ATA’s LEDs will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will then be saved into the Flash. If TFTP/HTTP fails for any reason (e.g., TFTP/HTTP server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the HandyTone ATA will stop the TFTP/HTTP process and simply boot using the existing code image in the flash.
- Firmware upgrade may take as long as 1 to 20 minutes over Internet, or just 20+ seconds if it is performed on a LAN. It is recommended to conduct firmware upgrade in a controlled LAN environment if possible. For users who do not have a local firmware upgrade server, Grandstream provides a NAT-friendly TFTP server on the public Internet for firmware upgrade. Please check the Services section of Grandstream’s Web site to obtain our public TFTP server’s IP address.
- Alternatively, user can download a free TFTP or HTTP server and conduct local firmware upgrade. A free windows version TFTP server is available for download from <http://support.solarwinds.net/updates/New-customerFree.cfm>. Our latest official release can be downloaded from <http://www.grandstream.com/y-firmware.htm>. Unzip the file and put all of them under the root directory of the TFTP server. Put the PC running the TFTP server and the HandyTone ATA in the same LAN segment. Please go to File -> Configure -> Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade. Start the TFTP server, in the HandyTone ATA’s web configuration page, configure the Firmware Server Path with the IP address of the PC, update the change and reboot the unit. Please be advised that our client will pull out firmware from the WAN side, if

the TFTP server is connected to the device's LAN port, the firmware upgrade will not work by design.

7.2 Configuration File Download

Grandstream SIP Device can be configured via Web Interface as well as via Configuration File through TFTP or HTTP. "Config Server Path" is the TFTP or HTTP server path for configuration file. It needs to be set to a valid URL, either in FQDN or IP address format. The "Config Server Path" can be same or different from the "Firmware Server Path".

A configuration parameter is associated with each particular field in the web configuration page. A parameter consists of a Capital letter P and 2 to 3 (Could be extended to 4 in the future) digit numeric numbers. i.e., P2 is associated with "Admin Password" in the ADVANCED SETTINGS page. For a detailed parameter list, please refer to the corresponding firmware release configuration template.

When Grandstream Device boots up or reboots, it will issue request for configuration file named "cfgxxxxxxxxxxxx", where "xxxxxxxxxxxx" is the MAC address of the device, i.e., "cfg000b820102ab". The configuration file name should be in lower cases.

7.3 Firmware and Configuration File Prefix and Postfix

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

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

7.4 Managing Firmware and Configuration File Download

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

8 Restore Factory Default Setting

Warning !!!

Restore the Factory Default Setting will DELETE all configuration information of the device. Please backup or print out all the settings before you approach to following steps. Grandstream will not take any responsibility if you lose all the parameters of setting and cannot connect to your service provider.

Please disconnect network cable and power cycle the unit before trying to reset the unit to factory default. The steps are as follows:

Step 1:

Find the MAC Address of the device. It is a 12 digits HEX number located on the bottom of the unit.

Step 2:

Encode the MAC address. Please use the following mapping:

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

For example, if the MAC address is 000b8200e395, it should be encoded as "0002228200333395".

Step 3:

To perform factory reset:

- a. Press "*" or the LED button for voice prompt.
- b. Enter "99" and get the voice prompt "Reset".
- c. Enter the encoded MAC address of the device.
- d. Wait for 15 seconds.

The device will reboot automatically and restore to factory default setting.

NOTES:

- Please be aware by default the HandyTone-488 WAN side HTTP access is disabled. After a factory reset, the device's web configuration page can be accessed only from its LAN port, please refer to instructions in section 6.2.1 for details.

9 Glossary of Terms

ADSL

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

AGC

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

ARP

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

ATA

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

CODEC

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

CNG

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

DATAGRAM

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

DECIMATE

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

DECT

Digital Enhanced Cordless Telecommunications: A standard developed by the European Telecommunication Standard Institute from 1988, governing pan-European digital mobile telephony. DECT covers wireless PBXs, telepoint, residential cordless telephones, wireless access to the public switched telephone network, Closed User Groups (CUGs), Local Area Networks, and wireless local loop. The DECT Common Interface radio standard is a multicarrier time division multiple access, time division duplex (MC-TDMA-TDD) radio transmission technique using ten radio frequency channels from 1880 to 1930 MHz, each

divided into 24 time slots of 10ms, and twelve full-duplex accesses per carrier, for a total of 120 possible combinations. A DECT base station (an RFP, Radio Fixed Part) can transmit all 12 possible accesses (time slots) simultaneously by using different frequencies or using only one frequency. All signaling information is transmitted from the RFP within a multiframe (16 frames). Voice signals are digitally encoded into a 32 kbit/s signal using Adaptive Differential Pulse Code Modulation.

DNS

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

DID

Direct Inward Dialing

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

DSP

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

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

Grandstream products all have DSP chips built inside.

DTMF

Dual Tone Multi Frequency

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

FQDN

Fully Qualified Domain Name

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

FXO

Foreign eXchange Office

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

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

FXO is complimentary to FXS (and the PSTN).

FXS

Foreign eXchange Station

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

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

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

DHCP

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

ECHO CANCELLATION

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

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

H.323

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

HTTP

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

IP

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

IP-PBX

IP-based Private Branch Exchange

IP Telephony

(Internet Protocol telephony, also known as Voice over IP Telephony) A general term for the technologies that use the Internet Protocol's packet-switched connections to exchange voice, fax, and other forms of information that have traditionally been carried over the dedicated circuit-switched connections of the public switched telephone network (PSTN). The basic steps involved in originating an IP Telephony call are conversion of the analog voice signal to digital

format and compression/translation of the signal into Internet protocol (IP) packets for transmission over the Internet or other packet-switched networks; the process is reversed at the receiving end. The terms IP Telephony and Internet Telephony are often used to mean the same; however, they are not 100 per cent interchangeable, since Internet is only a subcase of packet-switched networks. For users who have free or fixed-price Internet access, IP Telephony software essentially provides free telephone calls anywhere in the world. However, the challenge of IP Telephony is maintaining the quality of service expected by subscribers. Session border controllers resolve this issue by providing quality assurance comparable to legacy telephone systems.

IVR

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

MTU

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

NAT

Network Address Translation

NTP

[Network Time Protocol](#), a protocol to exchange and synchronize time over networks
The port used is UDP 123
Grandstream products using NTP to get time from Internet

OBP/SBC

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

PPPoE

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

PSTN

Public Switched Telephone Network

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

RTCP

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

RTP

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

SDP

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

SIP

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

All Grandstream products are SIP based

STUN

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

TCP

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

TFTP

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

UDP

User Datagram Protocol (UDP) is one of the core protocols of the [Internet protocol suite](#). Using UDP, programs on networked computers can send short messages known as [datagrams](#) to one

another. UDP does not provide the reliability and ordering guarantees that [TCP](#) does; datagrams may arrive out of order or go missing without notice. However, as a result, UDP is faster and more efficient for many lightweight or time-sensitive purposes.

VAD

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

VLAN

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

VoIP

Voice over IP

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