Grandstream Networks, Inc.

HT488
FXS/FXO Port IAD

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WELCOME

Thank you for purchasing Grandstream's HT488, the affordable, feature rich, Analog Telephone Adaptor/IAD. The HT488 is based on SIP standard and features both an FXS and FXO port for Internet data, voice, and PSTN networks. It functions as an "all-in-one" IAD. The HT488 features a high level of integration including, but not limited to an integrated router, an analog telephone FXS interface and a programmable FXO gateway for toggling operations between the IP network (SIP Server platform) and the PSTN network. The HT488 supports the feature of "hop-on/hop-off" calling.

This manual will help you learn how to operate and manage your HT488 Analog Telephone Adaptor/IAD and make the best use of its many features including simple and quick installation, 3-way conferencing, and remote call origination and "hop-on/hop-off" calling using the programmable PSTN FXO port. The HT488 is easy to manage and configure, and is an affordable VoIP solution for both the residential user and the remote user.

This document is subject to changes without notice. The latest electronic version of this user manual can be downloaded from the following location: http://www.grandstream.com/resources.html

SAFETY COMPLIANCES

The HT488 adaptor complies with FCC/CE and various safety standards. The HT488 power adaptor is compliant with UL standard. Only use the universal power adapter provided with the HT488 package. The manufacturer's warranty does not cover damages to the phone caused by unsupported power adaptors.

WARRANTY

If you purchased your HT488 from a reseller, please contact them for replacement, repair or refund. If you purchased the product directly from Grandstream, contact your Grandstream Sales and Service Representative for a RMA (Return Materials Authorization) number before you return the product. Grandstream reserves the right to remedy warranty policy without prior notification.

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.*

- This document is contains links to Grandstream GUI Interfaces. Please remember to download these examples http://www.grandstream.com/user_manuals/GUI/GUI_HT488.rar for your reference.
- This document is subject to change without notice. The latest electronic version of this user manual is available for download from the following location: http://www.grandstream.com/user_manuals/HT4488_User_Manual.pdf
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INSTALLATION

EQUIPMENT PACKAGING

The HT488 ATA package contains:

- One HT488 Main Case
- One Universal Power Adaptor
- One Ethernet Cable

CONNECTING YOUR ATA

HT488 Analog Telephone Adaptor is an all-in-one VoIP integrated device designed to be a total solution for networks providing VoIP services. All of the HT488 VoIP features and functions are available via a regular analog telephone.

FIGURE 1: CONNECTING THE HT488

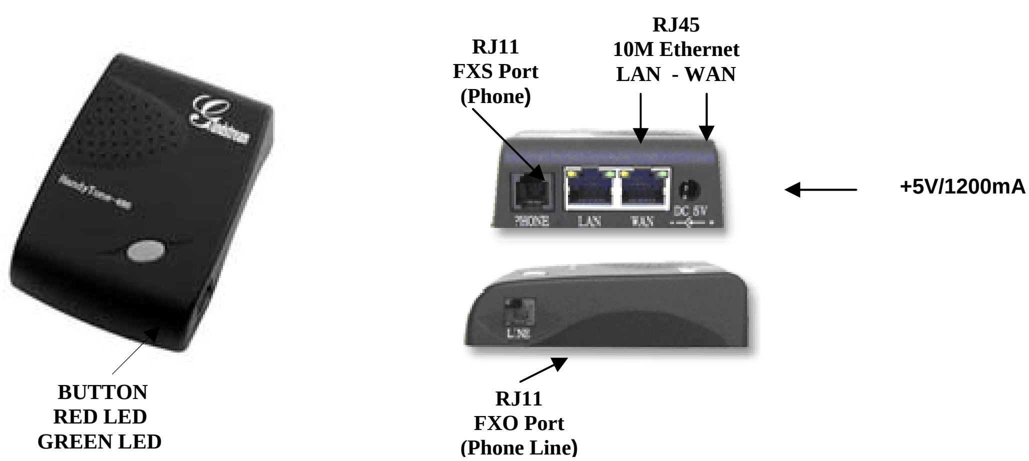


TABLE 1: DEFINITIONS OF THE HT488 CONNECTORS

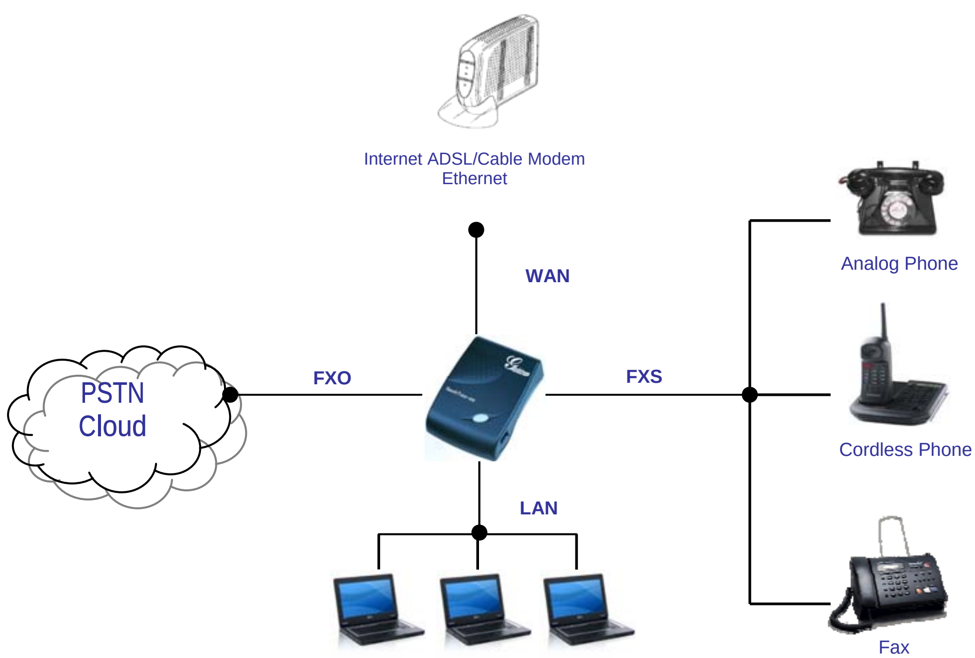
+5V/1.2A	Power adapter connection
LAN Port (RJ-45)	Connect the LAN port with an Ethernet cable to your PC.
WAN Port (RJ-45)	Connect to the internal LAN network or router.
PHONE (RJ-11)	FXS port to be connected to analog phones / fax machines.
LINE (RJ-11)	FXO port should be connected to the PSTN line
BUTTON	Button and two colors led indicator.

FIVE EASY STEPS TO INSTALL THE HT488

The HT488 is designed for easy configuration and easy installation. Detailed configuration instructions are located in the CONFIGURATION section.

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 HT488 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 HT488 if HT488 is used as a router.
5. Insert the power adapter into the HT488 and connect it to a wall outlet.

FIGURE 2: INTERCONNECTION DIAGRAM OF THE HT488



HandyTone488 has one FXS port and one FXO port. The PHONE port next to the LAN port is an FXS port. The LINE port on the side of the HandyTone488 is a FXO port. Both the FXS port and the FXO port can have a separate SIP account. This is a key feature of HandyTone488 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.

PRODUCT OVERVIEW

The HT488 enables IP connectivity for any phone or fax using the FXS port and a web-based GUI for easy configuration and installation. The device supports telephony features including caller ID, call waiting, call transfer, 3-way conferencing, and multi-language voice prompts. It functions as an FXO gateway that enables remote call origination and termination from/to PSTN and supports the feature of “hop-on/hop-off” using the programmable FXO port.

KEY FEATURES & SPECIFICATIONS

Ethernet Ports	DHCP	FXS Port	PSTN Pass – through	Voice Mail Indicator	Voice Codec	Remote Configuration
2 RJ45 (LAN/WAN)	Server/Client	1	Yes	Yes	iLBC, G.723, G.711, G.729, G.726, T.38	TFTP/HTTP

The HT488 supports independent SIP accounts or SIP server platform for each port. From a technical standpoint, the HT488 offers a power-outage survivable life line and fail-over-to-PSTN support, dual 10 Mbps Ethernet ports with an integrated NAT router, and supports a broad range of popular voice codecs. Table 2 and Table 3 summarize the HT488 technical and hardware specifications.

TABLE 2: HT488 TECHNICAL SPECIFICATIONS

Lines/SIP Accounts	2 lines / 2 SIP accounts
Protocol Support	SIP 2.0 (RFC 3261), TCP/UDP/IP, RTP/RTCP, HTTP, ARP/RARP, ICMP, DNS, DHCP, NTP, TFTP, PPPoE protocols
Feature Keys	1 button
LAN/WAN Interface	RJ-45 10 Mbps
Device Management	Web interface or via secure (AES encrypted) central configuration file for mass deployment Support device configuration via built-in IVR, Web browser or central configuration file through TFTP or HTTP Support Layer 2 (802.1Q, VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS) Auto/manual provisioning system NAT-friendly remote software upgrade (via TFTP/HTTP) for deployed devices including behind firewall/NAT Syslog support
DHCP Server/Client	Yes
Audio Features	Advanced Digital Signal Processing (DSP) Dynamic negotiation of codec and voice payload length Support for G.723,1 (5.3K/6.3K), G.729A, G.711 µ/A, G.726, and iLBC codecs In-band and out-of-band DTMF ((in audio, RFC2833, SIP INFO) Silence Suppression, VAD (voice activity detection), CNG (comfort noise generation), ANG (automatic gain control) Adaptive jitter buffer control , Packet delay & loss concealment Support volume amplification, Support configurable Call Progress Tones
Call Handling Features	Caller ID display or block, Call waiting caller ID, Call waiting/flash, Call transfer, hold, forward, mute, 3-way conferencing
Network and Provisioning	Manual or dynamic host configuration protocol (DHCP) network setup; RTP and NAT support traversal via STUN
Fax over IP	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through (pending), Fax Datapump V.17, V.19, V.27ter, V.29 for T.38 fax relay
Security	DIGEST authentication and encryption using MD5 and MD5-sess

TABLE 3: HT488 HARDWARE SPECIFICATION

LAN interface	1xRJ45 10Mbps
WAN interface	1xRJ45 10Mbps
FXS port	1 x FXS
FXO port (PSTN Port)	1x PSTN pass-through and life line port
Button	1
LED	Dual color (green/red)
Universal Switching Power Adaptor	Input: 100-240VAC 50-60 Hz / Output: +5VDC, 1200mA **UL** certified
Dimension	70mm (W) x 130mm (D) x 27mm (H)
Weight	0.6lbs (0.3kg)
Temperature	40 - 130°F / 5 – 45°C
Humidity	10% - 90% (non-condensing)
Compliance	

BASIC OPERATIONS

GET FAMILIAR WITH VOICE PROMPT

HT488 has a stored voice prompt menu for quick browsing and simple configuration. The voice prompt menu and the LED button is designed for the **FXS port ONLY**. Press the button or "***" from the analog phone to enter the IVR menu.

TABLE 4: HT488 IVR MENU DEFINITIONS

Menu	Voice Prompt	Options
Main Menu	"Enter a Menu Option"	Press "*****" for the next menu option Press "**#**" to return to the main menu Enter 01-05, 07,12-17,47 or 99 menu options
01	"DHCP Mode", "Static IP Mode"	Press "**9**" to toggle the selection If using "*Static IP Mode*", configure the IP address information using menus 02 to 05. If using "*Dynamic IP Mode*", all IP address information comes from the DHCP server automatically after reboot.
02	"IP Address " + IP address	The current WAN IP address is announced If using "*Static IP Mode*", enter 12 digit new IP address.
03	"Subnet " + IP address	Same as menu 02
04	"Gateway " + IP address	Same as menu 02
05	"DNS Server " + IP address	Same as menu 02
07	Preferred Vocoder	Press "**9**" to move to the next selection in the list: • PCM U / PCM A • G.723 • G.729 • G.726 • iLBC
12	WAN Port Web Access	Press "**9**" to toggle between **enable** / **disable**
13	Firmware Server IP Address	Announces current Firmware Server IP address. Enter 12 digit new IP address.
14	Configuration Server IP Address	Announces current Config Server Path IP address. Enter 12 digit new IP address.
15	Upgrade Protocol	Upgrade protocol for firmware and configuration update. Press "**9**" to toggle between **TFTP** / **HTTP**
16	Firmware Version	Firmware version information.
17	Firmware Upgrade	Firmware upgrade mode. Press "**9**" to toggle among the following three options: - always check - check when pre/suffix changes - never upgrade
47	"Direct IP Calling"	Enter a 12 digit IP address to make a direct IP call, after dial tone. (See "*Make a Direct IP Call*".)
99	"RESET"	Press "**9**" to reboot the device; **or** Enter encoded MAC address to restore factory default setting (See "*Restoring Factory Settings*")
	"Invalid Entry"	Automatically returns to main menu

NOTE:

- "*" 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. For IP address, add 0 before the digits if the digits are less than 3 (like 192.168.0.26 should be key in like 192168000026, no dot needed while input). Once all of the digits are collected, the input will be processed.
- Key entry can not be deleted but the phone may prompt error once it is detected

MAKE PHONE CALLS

CALLING PHONE OR EXTENSION NUMBERS

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 the 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 of North America, then followed by pressing # or wait for 4 seconds.

DIRECT IP CALLS

Direct IP calling allows two parties, that is, a HT 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 HT488 and other VoIP Device(i.e., another HT ATA or Budgetone SIP phone or other VoIP unit) have public IP addresses, or
- Both HT488 and other VoIP Device are on the same LAN using private IP addresses, or
- Both HT488 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 call, first pick up the analog phone or turn on the speakerphone on the analog phone, then access the voice menu prompt by dial "***" or press the button on the HT488, and dials "47" to access the direct IP call menu. User will hear a voice prompt "Direct IP Calling" and a dial tone. Enter a 12-digit target IP address to make a call. 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 is: **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.

NOTE: *When making a direct IP call, the "Use Random Port" should set to "NO".*

CALL HOLD

This function is applicable on FXS port for VoIP calls only. While in conversation, pressing the "flash" button on the attached analogue phone (if the phone has that button) will put the remote end on hold. Pressing the "flash" button again will release the previously held party and the bi-directional media will resume. If no "flash" button, then on-off hook quickly (hook flash) will do the same thing but also risk of losing call if the time is not short enough.

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 toggles between two active calls. The HT488 also provides CWCID (call waiting caller ID) information which includes caller ID information in addition to the special stutter tone. The analog phone must support this feature for it to work on the HT488. Both call waiting functions (call waiting and CWCID) are activated and deactivated from the configuration pages menu.

CALL TRANSFER

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 #
3. A can hang up.

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

Three situations can follow the transfer:

1. A quick confirmation tone (temporarily using the call waiting indication tone) followed by a dial tone. This indicates the transfer is successful (transferee has received a 200 OK from transfer target). At this point, A can either hang up or make another call.
2. A quick busy tone followed by a restored call (on supported platforms only). This means the transferee has received a 4xx response for the INVITE and we will try to recover the call. The busy tone is just to indicate to the transferor that the transfer has failed.
3. Busy tone keeps playing. This means we have failed to receive the second NOTIFY from the transferee and decided to time out. Note: this does not indicate the transfer has been successful, nor does it indicate the transfer has failed. When transferee is a client that does not support the second NOTIFY (such as our own earlier firmware), this will be the case. In bad network scenarios, this could also happen, although the transfer may have been completed successfully.

Attended 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 *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 #.
3. If C answers the call, A and C are in conversation. Then A can hang up to complete transfer.
4. If C does not answer the call, A can press "flash" back to talk to B.

NOTE: When **Attended Transfer** fails and A hangs up, the HT503 will ring user A back again to remind A that party B is still on the call. Party A can pick up the phone to resume a conversation with party B.

3-WAY CONFERENCING

Star Code Style 3-way Conference

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

NOTE: "*Enable Call Feature*" has to be set to YES in FXS PORT in the web configuration page.

Bellcore Style 3-way Conference

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

Assuming that call party A and B are in conversation. A (HT488) wants to bring C in a conference:

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

NOTE: *A is the call initiator for both calls with B and C*.

PSTN PASS THROUGH

HT488 supports PSTN pass through on FXS port 1. User can make and receive PSTN calls with attached analog phone in FXS PORT.

- 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 number.

When the HT488 is out of power, it will function as a jack. The user will be automatically connected to the PSTN Line.

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 use 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.
4. The user can hit the # key to identify the end of the pin code or wait 4 seconds for a new dial tone and then dialing the PSTN number.

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.

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 (default configuration), 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 use 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.

ROUTE CALLS TO PSTN

The FXO port enables access to the PSTN network. By default, the HT503 is in VoIP mode at off-hook. If "*Route call to PSTN*" is configured, certain calls will be initiated from the 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"* in the BASIC SETTINGS web configuration page. If the dialed digits match the specified prefix, outbound calls will be initiated from PSTN line.

Note: The route to PSTN feature is only applicable to a phone connected to the FXS Port. The configuration is done using the "dial plan" feature under the FXS tab. An example of the configuration is **{L: 911x+}** This shows that only calls that start with 911 are immediately forwarded to the PSTN line. All other numbers will not be routed to the PSTN. An normal # would be: **{L: 617x+|x+}** or **{x+| L: 617x+}**

For example, if "*Route call to PSTN*" is configured as 626, all outgoing calls starting with 626 will be initiated from the PSTN line.

FORWARD CALLS TO PSTN

Any VOIP call may be forwarded to a specified PSTN number if the call is not answered after a pre configured numbers of rings. By default "*Number of Rings*" parameter has value 4.

For example, if the end-user has configured a cell phone number in the field "*Forward to PSTN*" under BASIC SETTINGS configuration page, all calls will be forwarded to the cell phone number after 4 rings.

FORWARD CALLS TO VOIP

By default, each incoming PSTN call is received over the FXS port. The end-user may forward such a call to any preconfigured VoIP extension, in case the call is not answered in a certain number of rings. The Default value of the parameter "Number of Rings" is 4. If during 4 rings, the incoming from the PSTN call is not answered, the call will be forwarded to another VoIP number previously configured in the field:"*Forward to VoIP*". This parameter can also be found under BASIC SETTINGS configuration page.

ONE STAGE DIALING

This feature is applicable for VoIP to PSTN calls. Any VoIP extension may dial directly to a local PSTN number if the *one-stage dialing* feature is activated. This feature is configured under the FXO Configuration page and requires SIP Server configuration and support. The special dial plan feature must be activated in the SIP Server. An outbound call will be sent directly to the assigned FXO port account, where there the HT488 will initiate a call to the local CO. The RequestURI header in the INVITE message contains the phone number used to initiate the call to the local CO.

FAX SUPPORT

HT488 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 (default). 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 must select all the Preferred Codecs to be PCMU/PCMA (G.711-u/a).

CALL FEATURES

TABLE 5: HT488 CALL FEATURE DEFINITIONS

Key	Call Features
*23	3-Way Conferencing. Refer to above section for procedure to perform 3-Way Calling.
*30	**Block Caller ID** (for all subsequent calls)
*31	**Send Caller ID** (for all subsequent calls)
*67	**Block Caller ID** (per call). Dial "*67" + " number ". **No** dial tone will be played in the middle.
*82	**Send Caller ID** (per call). Dial "*82" + " number ". **No** dial tone will be played in the middle.
*50	**Disable Call Waiting** (for all-config change)
*51	**Enable Call Waiting** (for all-config change)
*70	**Disable Call Waiting** (Per Call)
*71	**Enable Call Waiting** (Per Call)
*72	**Unconditional Call Forward.** To use this feature, dial "*72", wait for the dial tone. Then dial the forward number ended with #, wait for dial tone, hang up.
*73	**Cancel Unconditional Call Forward** To cancel "Unconditional Call Forward", dial "*73" and get the dial tone, then hang up.
*87	**Blind Transfer** Refer to section above for procedure to perform Blind Transfer.
*90	**Busy Call Forward** To use this feature, dial "*90", wait for the dial tone. Then dial the forward number ended with #, wait for dial tone, 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", wait for the dial tone. Then dial the forward number ended with #, wait for dial tone, 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 beep. When in conversation and there is no call waiting, this action will switch to a new channel for a new call.

LED Light Pattern Indication

TABLE 6: HT488 LED DEFINITIONS

RED LED always indicates not normal status	
DHCP Failed or WAN No Cable	Button flashes every 2 seconds (if DHCP is configured)
HT488 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 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

CONFIGURATION GUIDE

CONFIGURING HT488 THROUGH VOICE PROMPT

DHCP MODE

Follow Table 3 with voice menu option 01 to enable HT488 to use DHCP.

STATIC IP MODE

Follow Table 3 with voice menu option 01 to enable HT488 to use STATIC IP mode, then use Option 02, 03, 04 to set up the HT488's IP, Subnet Mask, Gateway respectively.

TFTP SERVER ADDRESS

Follow Table 3 with voice menu option 06 to configure the IP address of the TFTP server.

FIRMWARE SERVER IP ADDRESS

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

CONFIGURATION SERVER IP ADDRESS

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

UPGRADE PROTOCOL

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

FIRMWARE UPGRADE MODE

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

CONFIGURING HT488 WITH WEB BROWSER

HT488 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 HT488 through a Web browser such as Microsoft's IE, AOL's Netscape or Mozilla Firefox installed on Windows or Unix OS. (Macintosh OS does not included).

Access the Web Configuration Menu

The HT488 HTML configuration page can be accessed via LAN or WAN ports.

- **FROM THE LAN PORT:**
  1. Directly connect a computer to the LAN port
  2. Open a command window on the computer
  3. Type in "ipconfig /release", the IP address etc becomes 0
  4. Type in "ipconfig /renew", the computer gets an IP address in 192.168.2.x segment by default
  5. Open a web browser, type in the default IP address of the LAN port. **http://192.168.2.1**. You will see the log in page of the device.

- **FROM THE WAN PORT:**
  1. Follow table 4 to find the WAN side IP address.
  2. Open a web browser, type in the WAN side IP address – for example: **http://HT503-WAN-IP-Address**

Note:

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

END USER CONFIGURATION

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

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

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

Once this HTTP request is entered and sent from a Web browser, the HT488 will respond with the following login screen:

FIGURE 3: SCREENSHOT OF CONFIGURATION LOG-IN PAGE

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 the “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 VoIP service provider has provisioned the device and configured for you therefore the password has already been changed.

After a correct password is entered in the login screen, the embedded Web server inside the HT488 will respond with the Configuration pages which are explained in details below.

TABLE 7: HT488 DEVICE STATUS PAGE DEFINITIONS

MAC Address	The device ID, in HEX format. This is very important ID for ISP troubleshooting.
IP Address	This field shows IP address of the HT488.
Product Model	This field contains the product model info, such as HT488.
Software Version	***Program:*** This is the main software release. *This number is always used for firmware upgrade.* Current release is 1.0.3.86 ***Bootloader:*** current version is 1.1.0.1. ***HTML:*** current version 1.0.3.86. ***VOC:*** current version is 1.0.0.13
System Uptime	This shows system up time since last reboot.
Registered	Whether the unit is registered to service provider’s server.
PPPoE Link Up	This shows whether the PPPoE is up if connected to DSL modem
NAT	This shows what kind NAT the HT386 is connected to. It is based on STUN protocol. If the detected NAT is symmetric NAT, STUN will not work and Outbound Proxy needed to make HT386 functioning correctly.

TABLE 8: HT488 BASIC SETTINGS PAGE DEFINITIONS

End User Password	This contains the password for end user to access the Web Configuration Menu. User can put new password here. This field is case sensitive with maximum of 25 characters
Web Port	This is the device's internal HTTP server port. Default is 80.
IP Address	- 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 HT386 will acquire its IP address from DHCP in the network. PPPoE settings is usually for DSL/ADSL modem users. The HT will attempt to establish a PPPoE session if PPPoE account is set. - 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.
DHCP hostname	This option specifies the name of the client. This field is optional but may be required by some Internet Service Providers. Default is blank.
DHCP domain	This option specifies the domain name that client should use when resolving hostnames via the Domain Name System. Default is blank.
DHCP vendor class ID	This option is used by clients and servers to exchange vendor-specific information. Default is blank.
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" and the Optional Rule is empty, then the displayed time will be 1 hour ahead of normal time. The "Automatic Daylight Saving Time Rule" shall have the following syntax: start-time;end-time;saving Both start-time and end-time have the same syntax: month,day,weekday,hour,minute month: 1,2,3,..,12 (for Jan, Feb, .., Dec) day: [+|-]1,2,3,..,31 weekday: 1, 2, 3, .., 7 (for Mon, Tue, .., Sun), or 0 which means the daylight saving rule is not based on week days but based on the day of the month. hour: hour (0-23), minute: minute (0-59) If "weekday" is 0, it means the date to start or end daylight saving is at exactly the given date. In that case, the "day" value must not be negative. If "weekday" is not zero and "day" is positive, then the daylight saving starts on the first "day"th iteration of the weekday (1st Sunday, 3rd Tuesday etc). If "weekday" us not zero and "day" is negative, then the daylight saving starts on the last "day"th iteration of the weekday (last Sunday, 3rd last Tuesday etc). The saving is in the unit of minutes. The saving time may also be preceded by a negative (-) sign if subtraction is desired instead of addition. The default value for "Automatic Daylight Saving Time Rule" shall be set to "04,01,7,02,00;10,-1,7,02,00;60" which is the rule for US. Examples US/Canada where daylight saving time is applicable:

	04,01,7,02,00;10,-1,7,02,00;60 This means the daylight saving time starts from the first Sunday of April at 2AM and ends the last Sunday of October at 2AM. The saving is 60 minutes (1hour).
Device Mode	This parameter controls whether the device is working in NAT router mode or Bridge mode. Save the setting and reboot prior to configuring the HT488.
Reply to ICMP on WAN Port	When set to "Yes", the HT488 responds to the PING command from other computers, but is also made vulnerable to DOS attacks. Default is **No**.
WAN Side HTTP/Telnet Access	When set to "Yes", the user can access the web configuration pages through the WAN port, instead of through the PC port. Warning: this configuration is less secure than the default option. Default is **No**.
Cloned WAN MAC Address:	This allows the user to change/set a specific MAC address on the WAN interface. Note: 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
DMZ IP:	Forward all WAN IP traffic to a specific IP address if no matching port is used by HandyTone488 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.
FXO One Stage Dialing	This configuration is applicable for VoIP to PSTN calls and indicates one or two stage dialing methods.

ADVANCED CONFIGURATION AND FXS/FXO PORTS PARAMETERS

To login to the Advanced Setting and FXS port configuration pages, administrator password is required. The default administrator password is "admin". User can change the administrator password here. The password is case sensitive and the maximum length is 25 characters.

TABLE 9: HT488 ADVANCED SETTINGS PAGE DEFINITIONS

Admin Password	Administrator password. Only administrator can configure the “Advanced Settings” page. Password field is purposely blanked for security reason after clicking update and saved. The maximum password length is 25 characters.
Home NPA	Local area code for North American Dial Plan.
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. Other VLAN supported equipments required if configured these settings.
No Key Entry timeout	Default is 4 seconds. User can short or extend that depends on digits dialed
STUN Server	IP address or Domain name of the STUN server.
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.
Authenticate Conf File	This protects the configuration from an unauthorized change. If set to “Yes, the configuration file is authenticated before acceptance.
NTP server	URI or IP address of the NTP (Network Time Protocol) server, which the HT386 will use to synchronize the date/time.
Syslog Server	The IP address or URL of syslog server, especially useful for ITSP (Internet Telephone Service Provider)
Syslog Level	Select the ATA to report the log level. Default is NONE. The level is either one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: • product model/version on boot up (INFO level) • NAT related info (INFO level) • sent or received SIP message (DEBUG level) • SIP message summary (INFO level) • inbound and outbound calls (INFO level) • registration status change (INFO level) • negotiated codec (INFO level) • Ethernet link up (INFO level) • SLIC chip exception (WARNING and ERROR levels) • memory exception (ERROR level) The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: GS_LOG: [device MAC address][error code] error message Here is an example: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up

TABLE 10: HT488 FXS PORT SETTINGS PAGES DEFINITIONS

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	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 work.
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. This field contains the user part of the SIP address for this phone. e.g., if the SIP address is **sip:my_user_id@my_provider.com**, then the SIP User ID is: **my_user_id.** Do NOT include the preceding “sip:” scheme or the host portion of the SIP address in this field.
Authenticate ID	ID used for authentication, usually same as SIP user ID, but could be different and decided by ITSP.
Authentication Password	Password for ATA to register to (SIP) servers of ITSP. Purposely blank out once saved for security. Maximum length is 25.
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.
SIP Registration Failure Retry Wait Time	In case sip server fail from some reason, the HT488 will try to re-register not according to standard SIP timers, but according to preconfigured interval. Recommended to leave default value 20sec. Designed by special request of big service providers.
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. Multiple DTMF transmission schema can be selected.
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.
Use Bell-style 3-way Conference	Default setting is No. When it is set to yes, the user will use Bell-style to initiate conference.
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. If set to YES, the feature "Call Waiting Indication" will not be activated.
Disable Call Waiting Caller ID (CWCID)	Default is No. If set to YES, user will not be able to see CWCID information. If set to No, also requires support of this feature by analog phone connected to FXS port.
NAT Traversal (STUN)	This setting decides whether the NAT traversal mechanism is activated. It should be set to "Yes" if the device is behind a NAT router. If no outbound proxy is configured, a 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 Keep-alive interval) send a dummy UDP packet to the SIP server to pinhole the NAT.
No Key Entry Timeout	Default is 4 seconds.
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. 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.
Refer-To Uses Target Contact.	Used for Attended transfer Feature. Default is NO. If set to YES, the “Refer-To” header uses the transferred target’s “Contact” header information.
Special Features	Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Nortel, Broadsoft, etc.
Onhook Threshold	Default setting is 800ms. If the flash event is longer than the settings, it is processed as on-hook event.
FXS Impedance	Selects the impedance of the analog telephone connected to the Phone port. The following information may be useful for end user configuration: 600Ohm – North America 270+750Ohm || 150nF – Most of Europe 220+820Ohm || 120nF – Australia, New Zealand 220+820Ohm || 115nF – Austria, Bulgaria, Germany, Slovakia, South Africa 370+620Ohm || 310nF – UK, India
Caller ID Scheme	Select the Caller ID Scheme to suit the standard of different area. • Bellcore (North America) • CID - Canada • DTMF (Brazil) • DTMF (Sweden) • 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.
Volume Amplification	Handset volume adjustment. RX is for receiving volume, TX is for transmission volume. Default values are 0dB for both parameters. +6dB generates the highest volume and -6dB generates the lowest volume.

TABLE 11: HT488 FXO PORT SETTINGS PAGES DEFINITIONS

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.
Enable PSTN Disconnect Tone Detection	If set to Yes, a special tone is used as the disconnect signal. This must be pre-configured for the device to recognize the signal. In case call has been established through the FXO port and remote side has disconnected active call first, the FXO port will wait for this pre-configured signal to disconnect the VoIP call clear the line.
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
Enable Current Disconnect	The Default value is Yes. This value should be used in case the PSTN provider uses line power drop to indicate call completion to the end point. In this case the HT488 will search for a power drop for a preconfigured time frame to disconnect such calls from a VoIP extension.
If Current Disconnect Enabled Use Threshold	Given value in milliseconds. This is a preconfigured value of duration for a line power drop used by specific service providers. For example, for a configured value of 500ms the device will ignore any random voltage drops on the line less than 500ms and the call will be considered as terminated when measured voltage drop period will be equal or more then 500ms. This is useful to prevent random call drops in some low quality PSTN lines.

NOTE: General settings for the FXS port are the same as those described for the FXO port.

TABLE 12: HT488 CALL PROGRESS TONES SETTINGS PAGE DEFINITIONS

Call Progress Tones	Using these settings, user can configure tone frequencies and cadences according to their preference. By default they are set to North American frequencies. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. ON is the period of ringing ("On time" in 'ms') while OFF is the period of silence. In order to set a continuous tone, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern.

SAVING THE CONFIGURATION CHANGES

Once a change is made, users should click on the "Update" button in the Configuration page. The HT488 will then display the following screen to confirm that the changes have been saved.

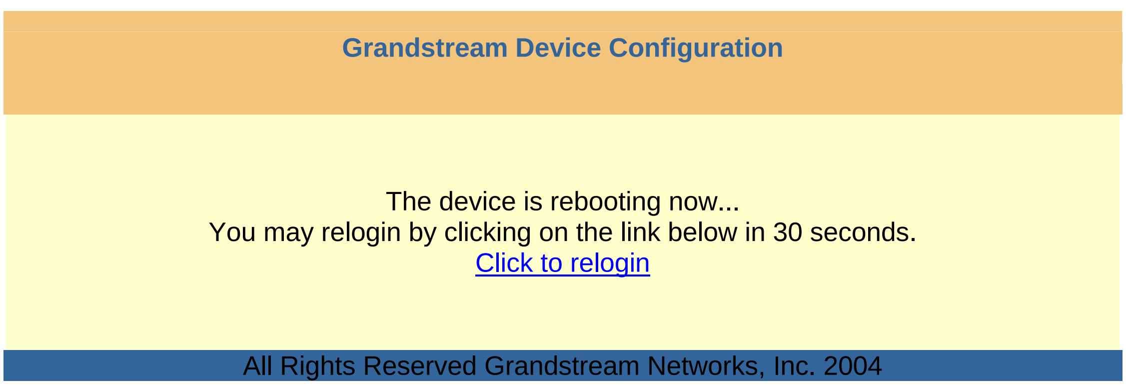
FIGURE 4: SCREENSHOT OF CONFIGURATION UPDATE MODE

Users are recommended to Reboot the HT488 after seeing the above message.

Rebooting the HT488 from Remote

Once a change is made, users should click on the "Update" button in the Configuration page. The HT488 will display a confirmation screen to confirm that the changes have been saved. Click 'Reboot' to save all changes. Please reference the GUI pages using the following link: http://www.grandstream.com/user_manuals/GUI/GUI_HT488.rar.

FIGURE 5: SCREENSHOT OF REBOOTING SCREEN



NOTE: Interrupting the 'booting up' process could permanently damage the device.

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/configurationtool.html.

SOFTWARE UPGRADE

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

FIRMWARE UPGRADE THROUGH TFTP/HTTP

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

e.g. firmware.mycompany.com:6688/Grandstream/1.0.3.86
e.g. 168.75.215.190

NOTES:

1. The TFTP server in IP address format can be configured via IVR. Please refer to the CONFIGURATION GUIDE section for instructions. If the TFTP server is in FQDN format, it must be set via the web configuration interface.
2. End users recommended using our TFTP server. Its address can be found at http://www.grandstream.com/firmware.html. Currently, the TFTP server, your HT488 can be upgraded from has an IP address 168.75.215.189. For companies, we recommend to maintain their own TFTP/ HTTP server for upgrade and provisioning procedures.
3. Once a "Firmware Server Path" is set, the 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 HT488 will attempt to retrieve the new image files by downloading them into the SRAM. During this stage, the HT503 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 HT503 will stop the TFTP/HTTP process and simply boot using the existing code image in the flash.
4. Firmware upgrades usually take around 2 minutes when 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.
5. 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.

Directions to download a free TFTP Server:

1. Unzip the file and put all of them under the root directory of the TFTP server.
2. Put the PC running the TFTP server and the HT488 device in the same LAN segment.
3. Please go to File -> Configure -> Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade.
4. Start the TFTP server, in the phone's web configuration page
5. Configure the Firmware Server Path with the IP address of the PC
6. Update the change and reboot the unit

The end-user can also choose to download the free HTTP server from http://httpd.apache.org/ or use Microsoft IIS web server.

CONFIGURATION FILE DOWNLOAD

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

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

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

FIRMWARE AND CONFIGURATION FILE PREFIX AND POSTFIX

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

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

MANAGING FIRMWARE AND CONFIGURATION FILE DOWNLOAD

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

RESTORE FACTORY DEFAULT SETTING

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

FACTORY RESET

IVR Command

Reset default factory settings using the IVR Prompt (Table 5):

1. Dial "***" for voice prompt.
2. Enter "99" and wait for "reset" voice prompt.
3. Enter the encoded MAC address (Look below on how to encode MAC address).
4. Wait 15 seconds and device will automatically reboot and restore factory settings.

Encoding the MAC Address

1. Locate the MAC address of the device. It is the 12 digit HEX number on the bottom of the unit.
2. Key in the MAC address. Use the following mapping:

   0-9: 0-9

   a. A: 22 (press the "2" key twice, "A" will show on the LCD)
   b. B: 222
   c. C: 2222
   d. D: 33 (press the "3" key twice, "D" will show on the LCD)
   e. E: 333
   f. F: 3333

For example: if the MAC address is 000***b***8200***e***395, it should be keyed in as "000***222***8200***333***395".

NOTE:

1. Factory Reset will be disabled if the "**Lock keypad update**" is set to "Yes".
2. Please be aware by default the HT488 WAN side HTTP access is disabled. After a factory reset, the device's web configuration page can be accessed only from its LAN port.
3. If the HT488 was previously locked by your local service provider, pressing the RESET button will only restart the unit. The device will not return to factory default settings.