Grandstream Networks, Inc.

HT-488
Dual FXS/FXO Port
Analog Telephone Adaptor

TABLE OF CONTENTS
HT-488 USER MANUAL

WELCOME

Congratulations on becoming an owner of HT488. You made an excellent choice and we hope you enjoy all of its capabilities.

Grandstream's HT488 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 HT488 is fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market.

Grandstream HT488 is a new addition to the popular HT product family. It is an enhanced model compared to the award-winning HT488 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 HT488 has been awarded the Best of Show product in 2005 Internet Telephony Conference and Expo.

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

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/pdf/HT488UserManual.pdf

- *Reproduction or transmittal of the entire or any part, in any form or by any means, electronic or print, for any purpose without the express written permission of Grandstream Networks, Inc. is not permitted.*

INSTALLATION

EQUIPMENT PACKAGING

The HT488 ATA package contains:

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

CONNECTING YOUR ATA

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

FIGURE 1: CONNECTING THE HT488
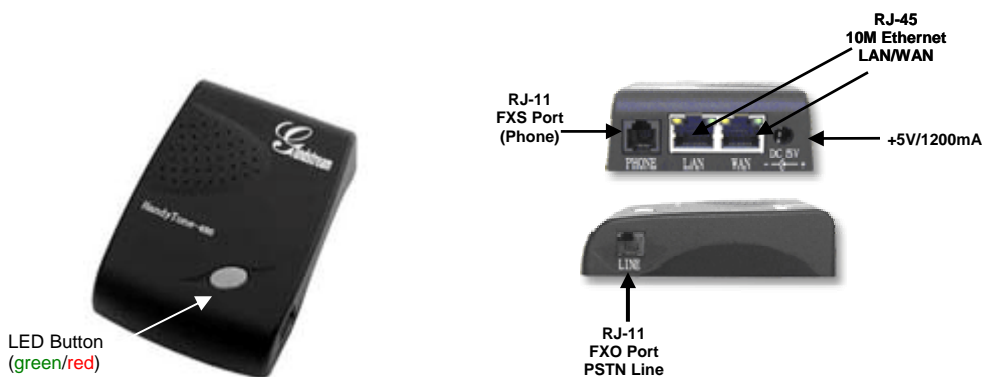
The HT488 has one FXS port and one FXO port. The PHONE port next to the LAN port is a FXS port. The LINE port on the side of the HandyTone-488 is a FXO port. Both the FXS port and the FXO port can have a separate SIP account. This is a key feature of HT488 as it supports simultaneous calls on both the FXS port and FXO port. Telephone calls can be originated from or terminated on the PSTN network remotely via the FXO port.

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. Configure the HT488 following the directions in the Configuration section of this manual.

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 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.

FIGURE 2: INTERCONNECTION DIAGRAM OF THE HT488
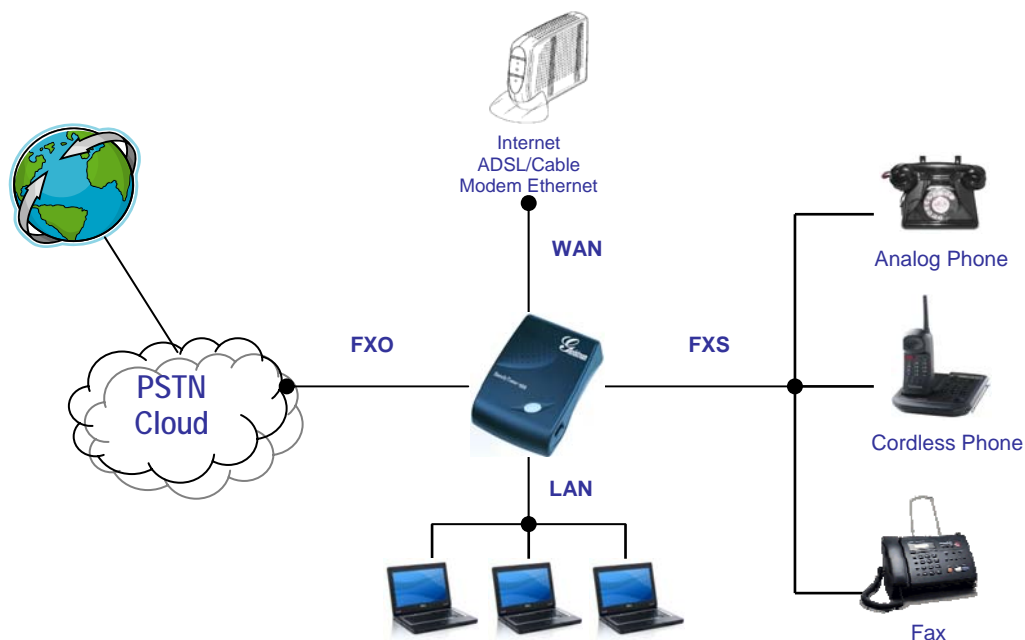
PRODUCT OVERVIEW

HT488 is a next generation dual-port SIP IAD for Internet data, voice, and fax. It has rich features and had the added functionality of a bridge that enables you to make remote calls over the internet.

KEY FEATURES

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

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
Physical Design	Stylish and compact design; small universal power supply, ideal for travel

HARDWARE SPECIFICATION

The table below lists the hardware specification of HT488.

TABLE 3: HT488 HARDWARE SPECIFICATION

LAN interface	1xRJ45 10Base-T
WAN interface	1xRJ45 10Base-T
FXS telephone port	1 x FXS
FXO telephone port (PSTN Port)	1x PSTN pass-through and life line port
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	FC CE C

BASIC OPERATIONS

GET FAMILIAR WITH VOICE PROMPT

HT488 has a stored voice prompt menu for quick browsing and simple configuration. Currently, the voice prompt menu and the LED button is designed for the FXS port <u>only</u>. To enter the voice prompt menu, press the **LED button** or "***" from the analog phone.

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:

- 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. 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 the 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 may need a prefix number followed by the phone number. Please check with your VoIP service provider for this information. If your phone is assigned a PSTN-like number such as 6265556789, you will most likely follow the rule 1 + (the number) – 16266667890. Press # 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 HT-488 and other VoIP Device (i.e. another Handytone ATA or Budgetone SIP phone or other VoIP unit) have public IP addresses, or

- Both HT-488 and other VoIP Device are on the same LAN using private IP addresses, or

- Both HT-488 and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

TO PLACE A DIRECT IP CALL:

1. Pick up the analog phone (or use the speakerphone),
2. Access the voice menu prompt by dial "***" or press the button on the HT-488
3. Dial "47" to access the direct IP call menu
4. At voice prompt "Direct IP Calling" and 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:

1. 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.

2. 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.

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

CALL HOLD

This function is applicable on the **FXS port** for VoIP calls only. While in conversation, pressing the "flash" button on the connected phone (if the phone has that button) places the remote end on hold. Pressing the "flash" button again releases the previously held party and the conversation can resume. If no "flash" button is available, then on-off hook quickly (hook flash) will do the same thing. You may lose the call if 'hook flash' is not quick 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.

CALL TRANSFER

The HT488 supports both blind transfer and attended transfer.

Blind Transfer
This function is applicable using the FXS port for VoIP calls only. Assume that parties **A** and **B** are in conversation. Party A wants to *Blind Transfer* Party B to C:

1. **A** presses FLASH on the analog phone to hear the dial tone.
2. Then **A** dials ***87**, then dials C's number, and then presses #
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 was successful (transferee has received a 200 OK from transfer target). 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 indicates the transfer has failed.
3. Busy tone keeps playing. This means we have failed to receive the second NOTIFY from the transferee and the call has timed 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 situation occurs. In bad network scenarios, this could also happen, although the transfer may have been completed successfully.

Attended Transfer
This function is applicable on the **FXS port** for VoIP calls only. Assume that parties **A** and **B** are in conversation. Party A wants to *Attend Transfer* Party 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 HT488 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

The HT488 supports both Star Code Style and Bellcore Style 3-way conferencing.

Star Code Style 3-way Conference
This function is applicable on the **FXS port** for VoIP calls only. Assume that parties **A** and **B** are in conversation. Party **A** wants to bring C into a 3-way 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 <u>does not answer</u> 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
To use the Bellcore Style conference, the "Use Bell-style 3-way Conference" field in FXS PORT web configuration must be enabled.

Assume that parties **A** and **B** are in conversation. Party **A** (using the HT-488) wants to bring C into a 3-way 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 <u>does not answer</u> the call, **A** can press FLASH back to talk to **B.**
5.   If **A** presses FLASH during the conference, C will be dropped out.

Note: Party A is the call initiator for both calls with party B and party C.

PSTN PASS THROUGH

HT488 supports PSTN pass through using the **FXS port**. The user can place and receive PSTN calls when the FXS port is in use.

- To receive PSTN calls, pick up the phone when it rings;
- To complete a PSTN call, press the PSTN access code (*00 is default, or any number configured in web configuration page) to switch to the PSTN line, hear a dial tone, then dial the number.

It the HT488 looses power, it will function as a jack, enabling a direct connection to the PSTN Line.

VOIP-TO-PSTN CALLS

This function is available using the **FXO port**. The FXO port functions as a bridge between the Internet and PSTN. The user can remotely use a 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, otherwise, the caller will hear a dial tone.
2. Enter the pin code (configured on the configuration page). The caller will hear a dial tone and be connected to the PSTN line if the pin code is valid. If the pin code is invalid, the continuous tone is played to prompt caller to enter the pin code again. The user may try up to 3 times to enter a correct pin code. After three (3) tries, the HT488 hangs up.
3. After the caller hears a dial tone from PSTN line, the caller can place the next call.

Note:
- Users can choose whether or not to apply password protection for VoIP-to-PSTN calls. A PIN (Pin for PSTN calls) consists of up to 8 numeric digits and can be configured using the 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.
- The special continuous tone is the prompt to enter a valid PIN code. If a caller doesn't enter a valid PIN, the HT488 times out after 10 seconds. Users may press the "#" key to indicate the end of an input or wait 4 seconds.
- On the web configuration page, if the "*Forward to PSTN*" is configured, the second stage dialing format is eliminated, so after dialing into the FXO SIP account number, the PSTN number will be called automatically

PSTN-TO-VOIP CALLS

This function is available using the **FXO port**. The FXO port functions as a bridge between the Internet and PSTN and enables calls to be passed from the PSTN network to VoIP. The user can make VoIP calls remotely by dialing into the 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 phone will ring for 4 times by default (this setting is configurable on the configuration page).

2. If no one answers the call after 4 rings (default configuration), then the caller hears either a special continuous tone (prompting a PIN number) or a dial tone.

3. Enter a valid PIN. The caller will hear dial tone and be bridged to VoIP. If an incorrect PIN is inputted, the continuous tone prompts caller to enter a valid PIN. The caller may try 3 times to enter a valid PIN, then the 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 or not to apply password protection for VoIP-to-PSTN calls. A PIN (Pin for PSTN calls) consists of up to 8 numeric digits and can be configured using the 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.

- The special continuous tone is the prompt to enter a valid PIN code. If a caller doesn't enter a valid PIN, the HT488 times out after 10 seconds. Users may press the "#" key to indicate the end of an input or wait 4 seconds.

- On the web configuration page, if the "*Forward to VoIP*" is configured, the second stage dialing format is eliminated, so after dialing into the FXO SIP account number, the PSTN number will be called automatically

ROUTE CALLS TO PSTN

The FXO port enables access to the PSTN network. By default, the HT488 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.

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.

FAX SUPPORT

HT488 supports FAX in two modes: 1) T.38 (Fax over IP) and 2) fax pass through. T.38 is the preferred method because it is more reliable and works well in most network conditions. If the service provider supports T.38, please use this method by selecting 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

Key	Call Features
*23	**3-way Conferencing** Please refer to 3 way Calling section.
*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** Please refer to Blind Transfer section.
*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, flash/hook switches to incoming call (call waiting is enabled). When in conversation and no incoming call heard, flash/hook switches provides a new dial tone.

LED Light Pattern Indication

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

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 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. There are three options:
1) always check, 2) check when pre/suffix changes, and 3) never upgrade

CONFIGURING HT- 488 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 is 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**
        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://HT-488-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 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. Please reference the GUI pages using the following link: http://www.grandstream.com/user_manuals/GUI/GUI_HT488.rar.

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

NOTE: If you **can not** log into the configuration page by using the default password, please check with the VoIP service provider. It is 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 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 HT-488.
Product Model	This field contains the product model info, such as HT-488.
Software Version	*Program:* This is the main software release. *This number is always used for firmware upgrade.* Current release is 1.0.3.64 *Bootloader:* current version is 1.1.0.1. *HTML:* current version 1.0.3.64. *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 time displayed in daylight savings time. If set to "Yes", then the displayed time will be 1 hour ahead of normal time.
	
	The "Optional Rule" is configured to automatically adjust the Daylight Savings Time (DST) based on the rule set in this field. Rule 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 (e.g.: 1st Sunday, 3rd Tuesday etc). If "weekday" is 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** is set for US, the "Automatic Daylight Saving Time Rule" shall be set to **"3,2,7,2,0;11,1,7,2,0;60"** <u>Examples</u> US/Canada where daylight saving time is applicable: 03,02,7,02,00;11,1,7,02,00;60 This means the daylight saving time starts from the second Sunday of March at 2AM and ends the first Sunday of November at 2AM. The saving is 60 minutes.
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
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.

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, HT- 488 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.
Caller ID Scheme	• 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 different area or PBX.
Polarity Reversal	Default is No. If set to Yes, polarity will be reversed upon call establishment and termination.
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.
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.
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 1. G.711 A/µ law, 2. G.723.1, 3. G.726-32, 4. G.729A, 5. 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 <u>first</u> vocoder in the vocoder preference list or the actual used payload type negotiated between the 2 conversation parties at run time. **FOR EXAMPLE:** 1. If the first vocoder is configured as G723 and the "Voice Frames per TX" is set to 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. 2. Similarly, if this field is set to 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.
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.
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

Note: General settings have same meaning as explained in above for FXS port page. Here are described only a parameters related to FXO port.

Call Progress Tones	Using these settings, user can configure tone frequencies 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 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.

Rebooting the HT488 from Remote

The administrator can remotely reboot the HT488 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. You can login again after about 30 seconds.

FIGURE 4: SCREENSHOT OF REBOOTING SCREEN

Grandstream Device Configuration

The device is rebooting now...
You may relogin by clicking on the link below in 30 seconds.
Click to relogin

All Rights Reserved Grandstream Networks, Inc. 2004
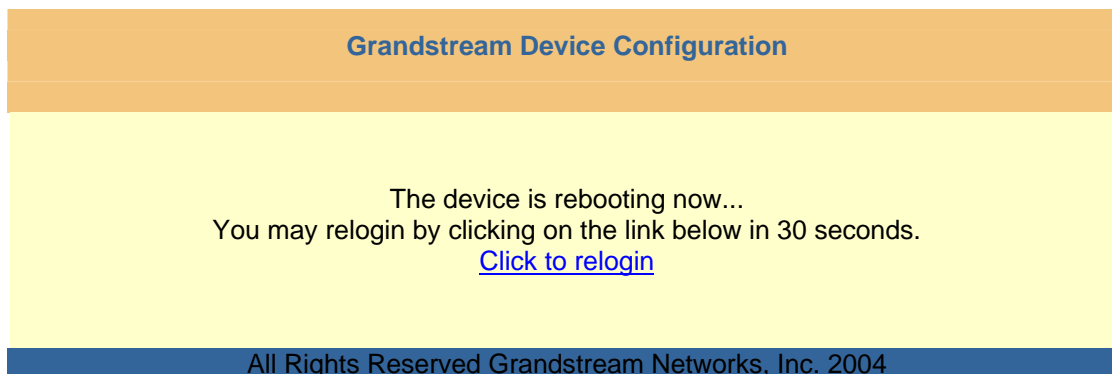
NOTE: Interrupting the 'booting up' process could permanently damage the device.

CONFIGURATION THROUGH A CENTRAL SERVER

The Grandstream HT488 can be automatically configured from a central provisioning system.

When the HT488 boots up, it will send TFTP or HTTP request to download configuration file, "cfg000b82xxxxxx", where "000b82xxxxxx" is the LAN side MAC address of the HT488

The configuration files can be downloaded via TFTP or HTTP from the central server. A service provider or an enterprise with large deployment of HT488 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 HT488. 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 HT488 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/1.0.3.64
> e.g. 168.75.215.190

NOTES:

1. TFTP server in IP address format can be configured via IVR. Please refer to section CONFIGURATION GUIDE for instructions. If TFTP server is in FQDN format, it must be set via web configuration interface.
2. End users recommended using our TFTP server. Its address can be found at http://grandstream.com/y-firmware.htm. Currently, the TFTP server, your HT-488 can be upgraded from has an IP address 168.75.215.190. For companies, we recommend to maintain their own TFTP/ HTTP server for upgrade and provisioning procedures.
3. Once a "Firmware Server Path" is set, user needs to update the settings and reboot the device. If the configured firmware server is found and a new code image is available, the HT ATA will attempt to retrieve the new image files by downloading them into the HT ATA's SRAM. During this stage, the HT ATA's LEDs will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will then be saved into the Flash. If TFTP/HTTP fails for any reason (e.g., TFTP/HTTP server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the HT ATA will stop the TFTP/HTTP process and simply boot using the existing code image in the flash.
4. 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.
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.

<u>Directions to download a free TFTP Server</u>:

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 GXW400X 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", the 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, however they are defined. This allows the device to 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, the 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 GXW400X WAN side HTTP access is disabled. After a factory reset, the device's web configuration page can be accessed only from its LAN port.

GLOSSARY OF VoIP 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 sub-sampling.

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 multi-carrier 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 multi-frame (16 frames). Voice signals are digitally encoded into a 32 Kbit/s signal using Adaptive Differential Pulse Code Modulation.

DNS Short for *D*omain *N*ame *S*ystem (or *S*ervice or *S*erver), an Internet service that translates *domain names* into IP addresses

DID 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 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 and.com is the top level domain.

FXS Foreign eXchange Office. An FXS 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 FXS interface will accept calls from FXS or PSTN interfaces. All countries and regions have their own standards.
- FXS is complimentary to FXS (and the PSTN).

FXS Foreign eXchange Station. An FXS device has hardware to generate the ring signal to the FXS extension (usually an analog phone).
- An FXS device will allow any FXS 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 FXS devices (by an FXS 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 Caller. The OBP/SBC acts as if it was the called VoIP phone and places a second call to the called Caller. The effect of this behavior 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. 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 well 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 the Internet. VoIP encompasses many protocols. All the protocols do some form of signaling of call capabilities and transport of voice data from one point to another. e.g.: SIP, H.323, etc.