# User Manual

## HandyTone-486

### Analog Telephone Adaptor

**For Firmware Version 1.0.8.16**

# Grandstream Networks, Inc.

*www.grandstream.com*

# Table of Contents

# 1    Welcome

Congratulations on becoming an owner of HandyTone-486.  You made an excellent choice and we hope you will enjoy all its capabilities.

Grandstream's award-wining HandyTone-486 is an all-in-one VoIP integrated access device that features superb audio quality, rich functionalities, high level of integration, compactness and ultra-affordability.  The HandyTone-486 is fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market.

Grandstream HandyTone-486 has been awarded the Best of Show product in 2004 Internet Telephony Conference and Expo.

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/user_manuals/HandyTone-486UserManual.pdf
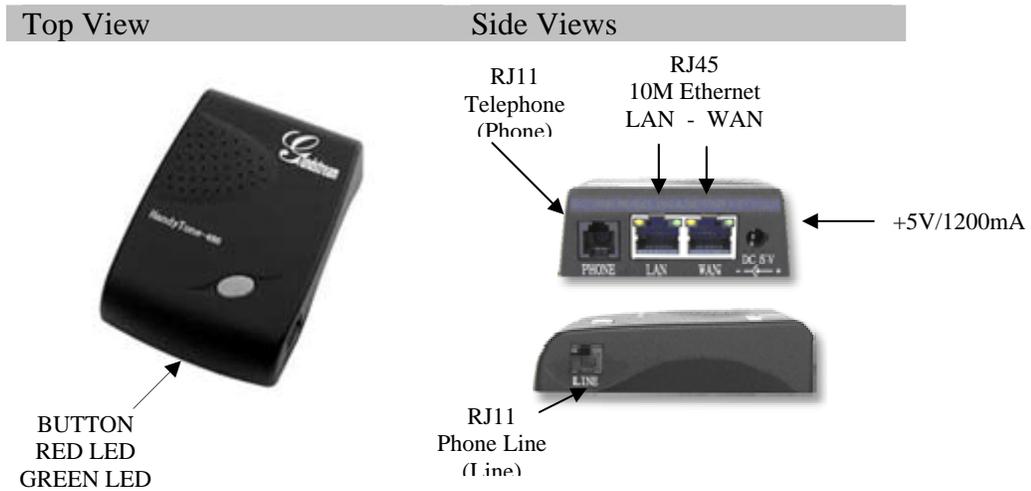
# 2    Installation

HandyTone-486 Analog Telephone Adaptor is an all-in-one VoIP integrated device designed to be a total solution for networks providing VoIP services.

The HandyTone-486 VoIP functionalities are available via a regular analog telephone.

The following photo illustrates the appearance of a HandyTone-486.



Interconnection Diagram of the HandyTone-486:
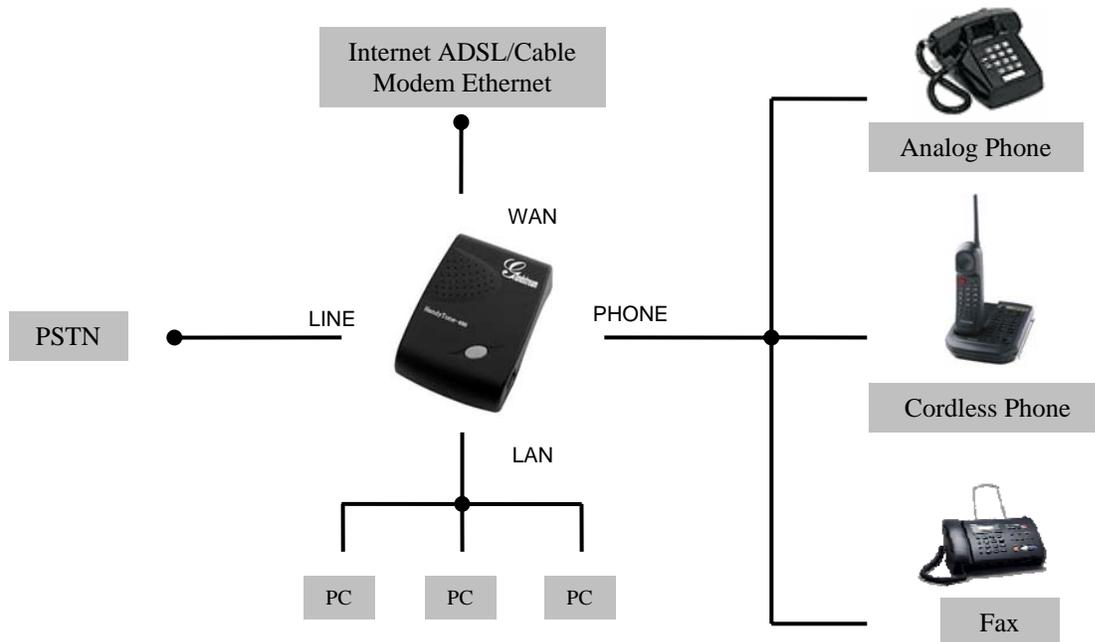
# 3    What is Included in the Package

The HandyTone-486 package contains:

  1) One HandyTone-486
  2) One universal power adaptor
  3) One Ethernet cable

## 3.1    Safety Compliances

The HandyTone-486 is compliant with various safety standards including FCC/CE and C-tick. Its power adaptor is compliant with UL standard. The HandyTone-486 should only operate with the universal power adaptor provided in the package.

## 3.2    Warranty

Grandstream has a reseller agreement with our reseller customer. End users should contact the company from whom you purchased the product for replacement, repair or refund.

If you purchased the product directly from Grandstream, contact your Grandstream Sales and Service Representative for a RMA (Return Materials Authorization) number.

Grandstream reserves the right to remedy warranty policy without prior notification.

*Warning: Please do not attempt to use a different power adaptor. Using other power adaptor may damage the HandyTone-486 and will void the manufacturer warranty.*

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

# 4    Product Overview

## 4.1    Key Features

- Supports SIP 2.0(RFC 3261), TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP/RARP, DNS, DHCP (both client and server), NTP, PPPoE, STUN, TFTP, etc.
- Built-in router, NAT, Gateway and DMZ port forwarding. Can also be configured to function as a two Ethernet ports bridge (NAT function is disabled)
- Powerful digital signal processing (DSP) to ensure superb audio quality; advanced adaptive jitter control and packet loss concealment technology
- Support various codecs including G.711 (PCM a-law and u-law), G.723.1 (5.3K/6.3K), G.726 (32K), as well as G.729A and iLBC
- Support Caller ID/name display or block, Call waiting caller ID, Hold, Call Waiting/Flash, Call Transfer, 3-way conference (on Rev. 2.0), Call Forward, in-band and out-of-band DTMF, etc.
- Support fax pass through (for PCMU and PCMA) and T.38 FoIP (Fax over IP)
- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168), and AGC (Automatic Gain Control)
- Support standard encryption and authentication (DIGEST using MD5 and MD5-sess)
- Support for Layer 2 (802.1Q VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
- Support automated NAT traversal without manual manipulation of firewall/NAT
- Support device configuration via built-in IVR, Web browser or encrypted configuration files through TFTP or HTTP server
- Support firmware upgrade via TFTP or HTTP
- Support PSTN pass through (on Rev.2.0)
- Support SIP Session Timer
- Support Syslog (on Rev.2.0)
- Ultra compact (wallet size) and lightweight design, great companion for travelers
- Compact, lightweight Universal Power adapter

## 4.2    Hardware Specification

The table below lists the hardware specification of HandyTone-486.

| Model | HandyTone-486 |
|---|---|
| LAN interface | 1xRJ45 10Base-T |
| WAN interface | 1xRJ45 10Base-T |
| FXS telephone port | 1xFXS |
| PSTN port | 1x PSTN pass through or life line port |
| Button | 1 |
| LED | Green and red color |
| Universal Switching Power Adaptor | Input: 100-240VAC 50-60 Hz<br>Output: +5VDC, 1200mA,<br>**UL** certified |
| Dimension | 70mm (W)<br>130mm (D)<br>27mm (H) |
| Weight | 0.6lbs (0.3kg) |
| Temperature | 40 - 130$^o$F<br>5 – 45$^o$C |
| Humidity | 10% - 90%<br>(non-condensing) |
| Compliance | FC CE C |

**NOTE:**

- HandyTone-486 has two hardware revisions. This information can be found on the label at the bottom of the device. The difference between HandyTone-486 Rev.1.0 and HandyTone-486 Rev.2.0 is that a HandyTone-486 Rev.2.0 line port can function as PSTN pass through while a HandyTone-486 (Rev.1.0, old model, no longer shipped) line port is just a life line port and will bridge to PSTN only when the device is out of power.

# 5    Basic Operations

## 5.1    Get Familiar with Key Pad and Voice Prompt

HandyTone-486 stores a voice prompt menu (Interactive Voice Response or IVR) for quick browsing and simple configuration. To enter this voice prompt menu, simply press the button on the HandyTone-486 or pick up the phone and dial "***". The following table shows how to use the voice prompt menu to configure the device.

| Menu | Voice Prompt | User's Options |
|------|-------------|----------------|
| Main Menu | "Enter a Menu Option" | Enter "*" for the next menu option<br>Enter "#" to return to the main menu<br>Enter 01 – 06, 47, 86 or 99 Menu option |
| 01 | "DHCP Mode", or<br>"Static IP Mode" | Enter '9' to toggle the selection<br>*If user selects "Static IP Mode", user need configure all the IP address information through menu 02 to 05. If user selects "Dynamic IP Mode", the device will retrieve all IP address information from DHCP server automatically when user reboots the device.* |
| 02 | "IP Address " + IP address | The current WAN IP address is announced<br>Enter 12-digit new IP address if in Static IP Mode. |
| 03 | "Subnet " + IP address | Same as Menu option 02 |
| 04 | "Gateway " + IP address | Same as Menu option 02 |
| 05 | "DNS Server " + IP address | Same as Menu option 02 |
| 06 | "TFTP Server " + IP address | Same as Menu option 02 |
| 47 | "Direct IP Calling" | When entered, user will be prompted a dial tone, dial a 12-digit IP address to make a direct IP call.<br> (For details, see "4.2.2 Make a Direct IP Call".) |
| 86 | "Voice Messages Pending"<br>"No Voice Messages" | If there are voice messages, user can dial "9" and dial pre-configured phone number to retrieve voice message. |
| 99 | "RESET" | Enter "9" to reboot the device; or<br>Enter MAC address to restore factory default setting (For details, see section 8.) |
|  | "Invalid Entry" | Automatically returns to Main Menu |

**NOTES:**

- Once the LED button is pressed, it enters voice prompt main menu. If the button is pressed again while it is already in the voice prompt menu state, it jumps to "Direct IP Calling" option and dial tone plays in this state
- "*" shifts down to the next menu option
- "#" returns to the main menu
- "9" functions as the ENTER key in many cases to confirm an option
- All entered digit sequences have known lengths - 2 digits for menu option and 12 digits for IP address. Once all digits are accumulated, the device will  automatically process them
- For IP address input, omit the dot and enter the digits directly, add 0 for those octets with less than three digits. e.g.: IP: 192.168.1.10, key in: 192168001010
- Key entry cannot be deleted but the phone may prompt error once it is detected

## 5.2     Make Phone Calls

### 5.2.1    Calling phone or extension numbers

There are currently two methods to make an extension number call:
    a) Dial the numbers directly and wait for 4 seconds (Default "No Key Entry Timeout"). Or
    b) Dial the numbers directly, and press # (assuming that "use #" as dial key is selected in web
       configuration).
Examples:
        To dial another extension on the same proxy, such as 1008, simply pick up attached phone, dial 1008 and then press the # or wait for 4 seconds.
        To dial a PSTN number such as 6266667890, you might need to enter in some prefix number followed by the phone number. Please check with your VoIP service provider to get the information. If you phone is assigned with a PSTN-like number such as 6265556789, most likely you just follow the rule to dial 16266667890 as if you were calling from a regular analog phone, followed by pressing the # or wait for 4 seconds.

### 5.2.2    Direct IP calls

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

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

To make a direct IP to IP call, first pick up the analog phone or turn on the speakerphone on the analog phone, then access the voice menu prompt by dial "***" or press the button on the HandyTone-286, 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 would be:
**Voice Prompt with option 47, then 192168001020**\***45062** followed by pressing the "#" key if it is configured as a send key or wait for 4 seconds.


## 5.2.3   Call Hold

While in conversation, pressing the "FLASH" button on the attached phone will put the remote end on hold. Pressing the "FLASH" button again will release the previously Hold party and the bi-directional media will resume.


## 5.2.4   Call Waiting

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.


## 5.2.5   Call Transfer

### 5.2.5.1   Blind Transfer

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

1.  A presses "FLASH" on the analog phone ( or Hook Flash for old model phones)  to get a  dial tone.
2.  A dials **\*87** followed by C's number, then #(or wait for 4 seconds)
3.  A can hang up.

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

•  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.

- 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.

- 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.

### 5.2.5.2  Attended Transfer

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

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

**NOTE:**

- When Attended Transfer failed and A hang up, HandyTone- 486 will ring user A back again to remind A that B is still on the call. A can pick up the phone to restore conversation with B.

## 5.2.6   3-way Conferencing

HandyTone-486 rev. 2.0 supports 3-way conference.

Assume that call party A and B are in conversation. A wants to bring C in the conference:

1. A press "FLASH" button on the analog phone to get a dial tone.
2. A dials *23 followed by C's number and # (or wait for 4 seconds).
3. If C answers the call, then A press "FLASH" to bring B, C in the conference.
4. If C does not answer the call, A can press "FLASH" back to talk to B.

**NOTE:**

- "Enable Call Features" has to be set to "Yes" in web configuration page for Blind Transfer and 3-way Conferencing to work.

## 5.2.7   PSTN Pass Through/Life line

HandyTone-486 Rev. 2.0 supports PSTN pass through. User can send and receive PSTN calls with the attached analog phone.

To receive PSTN calls, simply make phone off hook when the analog phone rings.
To make a PSTN call, press the PSTN access code (*00 by default, or any number configured in web configuration page) to switch to PSTN line and get a dial tone, then dial the PSTN number.

When HandyTone-486 is out of power, it will function as a jack. User can use the same analog phone for PSTN calls.

## 5.3    Call Features

### 5.3.1   Call Features Table (star code)

Following table shows the call features (* code) of HandyTone-486.

| Key | Call Features |
|---|---|
| *23 | 3-way conference |
| *87 | Blind Transfer |
| *30 | Block CallerID (for all-config change) |
| *31 | Send CallerID (for all-config change) |
| *67 | Block CallerID (per call) |
| *82 | Send CallerID (per call) |
| *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. |
| *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 user heard the call waiting sound.<br>When in conversation and no incoming call heard, this action will switch to a new channel for a new call. |
|---|---|

# 5.4    FAX

HandyTone-486 supports FAX in two modes: T.38 (Fax over IP) and fax pass through. T.38 is the preferred method because it is more reliable and works well in most network conditions. If the service provider supports T.38, please use this method by selecting Fax mode to be T.38. If the service provider does not support T.38, pass-through mode may be used. To send or receive faxes in fax pass through mode, users will need to select all the Preferred Codecs to be PCMU/PCMA.

# 5.5    LED Light Pattern Indication

Following are the LED light pattern indications.

| RED LED always indicates not abnormal status | |
|---|---|
| DHCP Failed or WAN No Cable | Button flashes every 2 seconds (if DHCP is configured) |
| HandyTone-486 fails to register | Button flashes every 2 seconds (if SIP server is configured) |
| Firmware Upgrading | Button flashes every 2 seconds |
| Device Malfunctions | Red light steady on |

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

# 6    Configuration Guide

## 6.1    Configuring HandyTone-486 WAN IP through Voice Prompt

### 6.1.1   DHCP Mode

Follow section 5.1 with voice menu option 01 to enable HandyTone-486 to use DHCP.

### 6.1.2   STATIC IP Mode

Follow section 5.1 with voice menu option 01 to enable HandyTone-486 to use STATIC IP mode, then use option 02, 03, 04, 05 to set up IP address, Subnet Mask, Gateway, and DNS server respectively.

### 6.1.3   TFTP Server Address

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

## 6.2    Configuring HandyTone-486 with Web Browser

HandyTone-486 has an embedded Web server that will respond to HTTP GET/POST requests. It also has embedded HTML pages that allow a user to configure the HandyTone-486 through a Web browser such as Microsoft's IE and AOL's Netscape.

### 6.2.1   Access the Web Configuration Menu

HandyTone-486's web configuration page can be accessed via LAN or WAN port:

- From the LAN port:

  - Directly connect a computer to the LAN port.
  - Open a command window on the computer
  - Type in "ipconfig /release", the IP address etc. become 0.0.0.0.
  - Type in "ipconfig /renew", the computer gets an IP  address in 192.168.2.1 by default
  - Open a web browser, type in the default gateway IP address. You will see the login page of the device.

  *http://192.168.2.1*

- From the WAN port:

  The WAN port HTML configuration option is disabled by default from factory. To access the HTML configuration menu from the WAN port, first enable the "WAN side HTTP access" option

by accessing the configuration via LAN port. With the WAN side HTTP access enabled, then get the WAN IP address of the HandyTone-486 through section 5.1 with menu option 02.  The HandyTone-486's Web Configuration page can be accessed by the following URI via WAN port:

*http://HandyTone-IP-Address*

where the *HandyTone-IP-Address* is the WAN IP address of the HandyTone-486.

**NOTE:**

- To type IP address into browser to get into the configuration page, please strip out the leading "0"s as the browser will parse in octet. e.g.: if the IP address is: 192.168.001.014, please type in: 192.168.1.14.

## 6.2.2    End User Configuration

Once this request is entered and sent from a Web browser, the HandyTone-486 will respond with the following login screen:

**Grandstream Device Configuration**

**Password** [                    ]

[ Login ]

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

**NOTE:**

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

After the correct password is entered into the login screen, the embedded Web server inside the device will respond with the following BASIC SETTINGS configuration page:

**Grandstream Device Configuration**

| STATUS BASIC SETTINGS ADVANCED SETTINGS | |
|---|---|
| **End User Password:** | [                    ] (purposely not displayed for security protection) |
| **IP Address:** | ⦿ dynamically assigned via DHCP (default) or PPPoE (will attempt PPPoE if DHCP fails and following is non-blank)<br><br>PPPoE account ID: [          ]<br><br>PPPoE password: [          ]<br><br>Preferred DNS server: [0].[0].[0].[0]<br><br>⦿ statically configured as:<br><br>IP Address: [192].[168].[0].[160]<br>Subnet Mask: [0].[0].[0].[0]<br>Default Router: [0].[0].[0].[0]<br>DNS Server 1: [0].[0].[0].[0]<br>DNS Server 2: [0].[0].[0].[0] |
| **Time Zone:** | GMT-5:00 (US Eastern Time, New York) ▾ |
| **Daylight Savings Time:** | ⦿ No  ⦿ Yes  (if set to Yes, display time will be 1 hour ahead of normal time) |
| | |
| *NAT/Bridge/DHCP Server Information & Configuration:* | |
| *Device Mode:* | ⦿ NAT Router  ⦿ Bridge |
| *Cloned WAN MAC Addr:* | [  ] [  ] [  ] [  ] [  ] [  ] (in hex format) |
| *LAN Subnet Mask:* | 255.255.255.0 (default is 255.255.255.0) |
| *LAN DHCP Base IP:* | 192.168.2.1 (base IP for the LAN port, default is 192.168.2.1) |
| *DHCP IP Lease Time:* | 120 (in units of hours, default is 120 hours or 5 days) |
| *DMZ IP:* | [          ] |
| *Port Forwarding:* | WAN port [0] LAN IP [          ] LAN port [0] Protocol [UDP Only ▾]<br>WAN port [0] LAN IP [          ] LAN port [0] Protocol [UDP Only ▾]<br>WAN port [0] LAN IP [          ] LAN port [0] Protocol [UDP Only ▾]<br>WAN port [0] LAN IP [          ] LAN port [0] Protocol [UDP Only ▾]<br>WAN port [0] LAN IP [          ] LAN port [0] Protocol [UDP Only ▾] |

| WAN port | 0 | LAN IP | | LAN port | 0 | Protocol | UDP Only ▼ |
| WAN port | 0 | LAN IP | | LAN port | 0 | Protocol | UDP Only ▼ |
| WAN port | 0 | LAN IP | | LAN port | 0 | Protocol | UDP Only ▼ |

Update      Cancel      Reboot

| | |
|---|---|
| **End User Password** | This field contains the password to access the Web Configuration Menu. The password is case sensitive with a maximum of 25 characters. |
| **IP Address** | There are 2 modes under which the HandyTone ATA can operate:<br><br>- If DHCP mode is enabled, then all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory) and the IP phone will acquire its IP address from the first DHCP server it discovers on the LAN it attaches to.<br><br>To use PPPoE feature please set the PPPoE account settings if the HandyTone ATA is connected directly to a DSL modem. The HandyTone ATA will attempt to establish a PPPoE session if any of the PPPoE fields are set.<br><br>- If Static IP mode is selected, then the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields will need to be configured. These fields are set to zero by default. |
| **Time Zone** | This parameters decides how the displayed date/time will be adjusted according to the specified time zone. |
| **Daylight Savings Time** | This parameter decides whether the displayed time will be daylight savings time or not. If set to "Yes", then the displayed time will be 1 hour ahead of normal time. Default is No. |
| **Device Mode** | Default is NAT router mode. HandyTone-486 Rev.2.0 can be configured in Bridge mode so the device functions as a bridge. |
| **Cloned WAN MAC Address:** | Allow 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. If bridge mode is selected, the LAN settings have no effect. |
| **LAN DHCP Base IP:** | Base IP for the LAN port, which functions as a gateway for its LAN. Default value is 192.168.2.1 |
| **DHCP IP Lease Time** | The amount of time that a given IP address will be valid for a LAN client. Value is set in units of hours. Default value is 120hr (5 Days). |

| | |
|---|---|
| **DMZ IP:** | Forward all WAN IP traffic to a specific IP address if no matching port is used by HandyTone-486 itself or in the defined port forwarding. |
| **Port Forwarding:** | Allow user to forward a matching (TCP/UDP) port to a specific LAN IP address with a specific (TCP/UDP) port. |

In addition to the Basic Settings configuration page, end users also have access to the device Status page. The following is a snap shot of the device Status page. Details will be explained next.

## Grandstream Device Configuration

**STATUS  BASIC SETTINGS  ADVANCED SETTINGS**

| | |
|---|---|
| **MAC Address:** | 00.0B.82.03.31.D2 |
| **WAN IP Address:** | 192.168.1.180 |
| **Product Model:** | HT486 REV 2.0 |
| **Software Version:** | Program-- 1.0.8.16    Bootloader-- 1.0.8.9    HTML-- 1.0.8.16    VOC-- 1.0.0.10 |
| **Configuration Version:** | |
| **System Up Time:** | 0 day(s) 1 hour(s) 2 minute(s) |
| **Registered:** | Yes |
| **PPPoE Link Up:** | disabled |
| **NAT:** | detected NAT type is port restricted cone |
| **NAT Mapped IP:** | 67.153.142.88 |
| **NAT Mapped Port:** | 5060 |
| **Total Inbound Calls:** | 0 |
| **Total Outbound Calls:** | 0 |
| **Total Missed Calls:** | 0 |
| **Total Call Time (in minutes):** | 0 |
| **Total SIP Message Sent:** | 4 |
| **Total SIP Message Received:** | 4 |
| **Total RTP Packet Sent:** | 0 |
| **Total RTP Packet Received:** | 0 |
| **Total RTP Packet Loss:** | 0 |
| | |

| | |
|---|---|
| **MAC Address** | The device ID, in HEX format.   This is very important ID for ISP troubleshooting. |
| **WAN IP Address** | This field shows WAN port IP address. |
| **Product Model** | This field contains the product model info,  such as HT486 Rev:2.0 |
| **Software Version** | **Program:** This is the main software release. This number is always used for firmware upgrade.  Current release is 1.0.8.12. <br> **Bootloader:** current version is 1.0.8.9. <br> **HTML:**   current version 1.0.8.12. <br> **VOC:**   current version is 1.0.0.10 |
| **System Uptime** | This shows system up time since last reboot. |
| **Registered** | Whether the unit is registered to 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 HandyTone ATA is connected to via its WAN port.  It is based on STUN protocol. |
| **NAT Mapped IP** | WAN side public IP if connected to LAN of a SOHO router. |
| **NAT Mapped Port** | WAN side SIP port if connected to LAN of a SOHO router. |
| **Other Statistical Status of ATA** | Self explainable, please see the page displayed. |

### 6.2.3   Advanced User Configuration

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

Advanced User configuration includes not only the end user configuration, but also advanced configuration such as SIP configuration, Codec selection, NAT Traversal Setting and other miscellaneous configuration.   Following is a snap shot of the advanced configuration page.

**Grandstream Device Configuration**

| STATUS BASIC SETTINGS ADVANCED SETTINGS | |
|---|---|
| **Admin Password:** | (purposely not displayed for security protection) |
| **SIP Server:** | sip.mycompany.com (e.g., sip.mycompany.com, or IP address) |
| **Outbound Proxy:** | (e.g., proxy.myprovider.com, or IP address, if any) |
| **SIP User ID:** | 3123763006 (the user part of an SIP address) |
| **Authenticate ID:** | 3123763006 (can be identical to or different from **SIP User ID**) |
| **Authenticate Password:** | (purposely not displayed for security protection) |
| **Name:** | John Doe (optional, e.g., John Doe) |
| **Home NPA:** | |

*Advanced Options:*

| | |
|---|---|
| *Preferred Vocoder:* *(in listed order)* | choice 1: current setting is " PCMU" |
| | choice 2: current setting is " PCMA" |
| | choice 3: current setting is " G723" |
| | choice 4: current setting is " G729" |
| | choice 5: current setting is " G726-32" |
| | choice 6: current setting is " iLBC" |
| | choice 7: current setting is " PCMU" |
| *G723 rate:* | ⊙ 6.3kbps encoding rate     ○ 5.3kbps encoding rate |
| *iLBC frame size:* | ⊙ 20ms     ○ 30ms |
| *iLBC payload type:* | 97 (between 96 and 127, default is 97) |
| *Silence Suppression:* | ⊙ No     ○ Yes |
| *Voice Frames per TX:* | 2 (up to 10/20/32/64 for G711/G726/G723/other codecs respectively) |
| *Fax Mode:* | ⊙ T.38 (Auto Detect)     ○ Pass-Through |
| *Layer 3 QoS:* | 48 (Diff-Serv or Precedence value) |
| *Layer 2 QoS (VoIP):* | 802.1Q/VLAN Tag 0     802.1p priority value 0 (0-7) |
| *Layer 2 QoS (PC):* | 802.1Q/VLAN Tag 0     802.1p priority value 0 (0-7) |
| *Allow incoming SIP messages from SIP proxy only:* | ⊙ No     ○ Yes |

| | |
|---|---|
| *Use DNS SRV:* | ⦿ No  ☐ Yes |
| *User ID is phone number:* | ⦿ No  ☐ Yes |
| *SIP Registration:* | ⦿ Yes  ☐ No |
| *Unregister On Reboot:* | ⦿ Yes  ☐ No |
| *Register Expiration:* | 3600 (in seconds. default 1 hour, max 45 days) |
| *Early Dial:* | ⦿ No  ☐ Yes (use "Yes" only if proxy supports 484 response) |
| *Dial Plan Prefix:* | [ ] (this prefix string is added to each dialed number) |
| *No Key Entry Timeout:* | 4 (in seconds, default is 4 seconds) |
| *Use # as Dial Key:* | ☐ No  ⦿ Yes (if set to Yes, "#" will function as the Dial key) |
| *local SIP port:* | 5060 (default 5060) |
| *local RTP port:* | 5004 (1024-65535, default 5004) |
| *Use random port:* | ⦿ No  ☐ Yes |
| *NAT Traversal:* | ☐ No <br> ⦿ Yes, STUN server is: [stun.mycompany.com] (URI or IP:port) |
| *keep-alive interval:* | 20 (in seconds, default 20 seconds) |
| *Use NAT IP* | [ ] (used in SIP/SDP message if specified) |
| *Proxy-Require:* | [ ] |
| *SUBSCRIBE for MWI:* | ⦿ No, do not send SUBSCRIBE for Message Waiting Indication <br> ☐ Yes, send periodical SUBSCRIBE for Message Waiting Indication |
| *Offhook Auto-Dial:* | [ ] (User ID/extension to dial automatically when offhook) |
| *Enable Call Features:* | ☐ No  ⦿ Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally) |
| *Disable Call-Waiting:* | ⦿ No  ☐ Yes |
| *Send DTMF:* | ⦿ in-audio  ☐ via RTP (RFC2833)  ☐ via SIP INFO |
| *DTMF Payload Type:* | 101 |
| *Send Flash Event:* | ⦿ No  ☐ Yes  (Flash will be sent as a DTMF event if set to Yes) |
| *Onhook Threshold:* | 800 ms ▾ |

| | |
|---|---|
| *FXS Impedance:* | 600 Ohm (North America) ▼ |
| *Caller ID Scheme:* | Bellcore (North America) ▼ |
| *Onhook Voltage:* | 36V ▼ |
| *Polarity Reversal:* | ⦿ No  ◯ Yes  (reverse polarity upon call establishment and termination) |
| *NTP Server:* | time.nist.gov  (URI or IP address) |
| *Send Anonymous:* | ⦿ No  ◯ Yes  (caller ID will be blocked if set to Yes) |
| *Anonymous Method:* | ⦿ Use From Header  ◯ Use Privacy Header |
| *Special Feature:* | Standard ▼ |
| *Reply to ICMP on WAN port:* | ⦿ No  ◯ Yes  (Unit will not respond to PING from WAN side if set to No) |
| *WAN Side HTTP Access:* | ◯ No  ⦿ Yes  (WAN side access to http server will be rejected if set to No) |
| *PSTN Access Code:* | _____  (key pattern to use the PSTN line, default is "*00") |
| *Syslog Server:* | _____ |
| *Syslog Level:* | NONE ▼ |
| *Session Expiration:* | 180  (in seconds. default 180 seconds) |
| *Min-SE:* | 90  (in seconds. default and minimum 90 seconds) |
| *Caller Request Timer:* | ◯ Yes  ⦿ No (Request for timer when making outbound calls) |
| *Callee Request Timer:* | ◯ Yes  ⦿ No (When caller supports timer but did not request one) |
| *Force Timer:* | ◯ Yes  ⦿ No (Use timer even when remote party does not support) |
| *UAC Specify Refresher:* | ◯ UAC  ◯ UAS  ⦿ Omit (Recommended) |
| *UAS Specify Refresher:* | ⦿ UAC  ◯ UAS (When UAC did not specify refresher tag) |
| *Force INVITE:* | ◯ Yes  ⦿ No (Always refresh with INVITE instead of UPDATE) |
| *Firmware Upgrade and Provisioning:* | Upgrade Via  ◯ TFTP  ⦿ HTTP<br><br>Firmware Server Path: fm.grandstream.com/gs<br><br>Config Server Path: fm.grandstream.com/gs<br><br>Firmware File Prefix: _____   Firmware File Postfix: _____<br><br>Config File Prefix: _____   Config File Postfix: _____ |

Automatic Upgrade:

⦿ No        ◯ Yes, check for upgrade every [10080] minutes (default 7 days)

⦿ Always Check for New Firmware

◯ Check New Firmware only when F/W pre/suffix changes

| | |
|---|---|
| *Firmware Key:* | [_____] (in Hexadecimal Representation) |
| *Authenticate Conf File:* | ⦿ No   ◯ Yes   (cfg file would be authenticated before acceptance if set to Yes) |
| *Lock keypad update:* | ⦿ No   ◯ Yes   (configuration update via keypad is disabled if set to Yes) |
| *Allow conf SIP Account in Basic Settings:* | ⦿ No   ◯ Yes |
| *Override MTU Size:* | [0] |

[ Update ]   [ Cancel ]   [ Reboot ]

| | |
|---|---|
| **Admin Password** | Administrator password. Only administrator can configure the "Advanced Settings" page. Password field is purposely left blank for security reason after user clicks UPDATE button.  This field is case sensitive and the maximum password length is 25 characters. |
| **SIP Server** | IP address or Domain name provided by VoIP service provider |
| **Outbound Proxy** | IP address or Domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by ATA for firewall or NAT penetration in different network environment. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy will provide solution for it. |
| **SIP User ID** | User account information, provided by VoIP service provider (ITSP), usually has the form of digit similar to phone number or actually a phone number. |
| **Authenticate ID** | ID used for authentication, usually same as SIP user ID, but could be different and decided by ITSP. |
| **Authentication Password** | Account information, password for ATA to register to (SIP) servers of ITSP. |
| **Name** | SIP service subscriber's name which will be used for Caller ID display. |

| | |
|---|---|
| **Preferred Vocoder** | The HandyTone ATA supports 6 different codec types including G.711 A/U law , G.723.1, G.726, G.729A/B, iLBC. A user can configure Codecs in a preference list that will be included with the same preference order in SDP message. |
| **G723 Rate:** | Encoding rate for G723 codec. By default, 6.3kbps rate is set. |
| **iLBC frame size:** | iLBC packet frame size. Default is 20ms. For Asterisk PBX, 30ms might be required. |
| **iLBC payload type:** | Payload type for iLBC. Default value is 97. The valid range is between 96 and 127. |
| **Silence Suppression** | This enables/disables the silence suppression/VAD feature of G723 and G729. If set to "Yes", when a silence is detected, small quantity of VAD packets (instead of audio packets) will be sent during this period. If set to "No", this feature is disabled. |
| **Voice Frames per TX** | This field contains the number of voice frames 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 codec in the above codec Preference List or the actual used payload type negotiated between the 2 conversation parties at run time. e.g., if the first codec 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 codec 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 codec choice. The maximum value for PCM is 10(x10ms) frames. For G726, it is 20 (x10ms) frames. And 32 (x30ms) and 64 (x10ms) frames for G723 and G729 respectively. |
| **Fax Mode** | T.38 (Auto Detect) FoIP by default, or fax Pass-Through. |
| **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 ( VoIP)** | Layer 2 QoS settings for VoIP traffic. Default setting is blank. VLAN supported equipment is required if user needs to change these settings. |
| **Layer 2 QoS ( PC)** | Layer 2 QoS settings for LAN port device traffic. Default setting is blank. VLAN supported equipment is required if user needs to change these settings. |

| | |
|---|---|
| **Allow incoming SIP messages from SIP proxy only** | If set to "Yes", the device will ignore any SIP message that does not come from the IP address (Source IP in the IP header) that it is registered to. Default setting is "No". |
| **Use DNS SRV:** | Default is "No". If set to "Yes" the client will use DNS SRV to lookup for the server. |
| **User ID is Phone Number** | If set to yes, a "user=phone" parameter will be attached to the "From" header in SIP request. |
| **SIP Registration** | This parameter decides 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 previous bindings. |
| **Register Expiration** | This parameter allows the user to specify the time frequency (in minutes) the HandyTone ATA will refresh its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days). |
| **Early Dial** | Default is "No". Use only if proxy supports 484 response. |
| **Dial Plan Prefix** | Sets the prefix added to each dialed number. |
| **No Key Entry timeout** | Default is 4 seconds. |
| **Use # as Send Key** | This parameter allows user to configure the "#" key to be used as the "Send" (or "Dial") key. Once set to "Yes", pressing this key will immediately trigger the sending of dialed string collected so far. If set to "No", the "#" key will then be included as part of the dialed string to be sent out. |
| **Local SIP port** | This parameter defines the local SIP port the HandyTone ATA listens and transmits. The default value is 5060. |
| **Local RTP port** | This parameter defines the local RTP-RTCP port pair the HandyTone ATA listens and transmits. It is the base RTP port for channel 0. When configured, channel 0 will use this port_value for RTP and the port_value+1 for its RTCP; channel 1 will use port_value+2 for RTP and port_value+3 for its RTCP. The default value is 5004. |
| **Use Random Port** | 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. |

| NAT Traversal | 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.<br>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. |
|---|---|
| Keep-alive interval | Default is 20 seconds. This is the interval of sending dummy UDP packet to keep the NAT "pin hole" open. |
| Use NAT IP: | If configured, the NAT IP address will be used in SIP/SDP message. Default is blank. |
| Proxy-Require | SIP Extension to notify SIP server that the unit is behind the NAT/Firewall. |
| Subscribe for MWI: | Default is "No". When set to "Yes" a SUBSCRIBE for Message Waiting Indication will be sent periodically. |
| Offhook Auto-Dial | This parameter allows a user to configure a User ID or extension number to be automatically dialed upon offhook. Please note that only the user part of a SIP address needs to be entered here. The HandyTone ATA will automatically append the "@" and the host portion of the corresponding SIP address.<br>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. |
| Enable Call Features | Default is "Yes".  Advanced call features and feature codes functions are supported locally. |
| Disable Call Waiting | Default is "No". User can use star codes to enable/disable call waiting. |
| Send DTMF | This parameter specifies the mechanism to transmit DTMF digit. There are 3 modes supported: in audio which means DTMF is combined in audio signal (not very reliable with low-bit-rate codec), via RTP (RFC2833), or via SIP INFO. |
| DTMF Payload Type | This parameter sets the payload type for DTMF using RFC2833 |
| Send Flash Event | Default is "No". If set to "Yes", flash will be sent as a DTMF event. |
| Onhook Threshold | The amount of time the hookflash is pressed that will cause the device to onhook. Default is 800ms. |
| FXS Impedance | Selects the impedance of the analog telephone connected to the Phone port. |
| Caller ID Scheme | Select the Caller ID Scheme to suit the standard of different area. |
| Onhook Voltage | Select the onhook voltage to suit different area or PBX. |

| | |
|---|---|
| **Polarity Reversal** | Select Polarity Reversal to adapt some call charge/billing system. Default is "No". |
| **NTP server** | This is the URI or IP address of the NTP (Network Time Protocol) server, which the HandyTone ATA will use to synchronize the date/time. |
| **Send Anonymous** | If this parameter is set to "Yes", user ID will be sent as anonymous, essentially blocks the Caller ID from displaying. |
| **Anonymous Method** | Allows displayed Caller ID to be taken from Privacy Header or From Header. |
| **Special Features** | Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Nortel, Broadsoft, etc. |
| **Syslog Server** | The IP address or URL of System log server. This feature is especially useful for ITSP (Internet Telephone Service Provider). |
| **Syslog Level** | Select the ATA to report the log level. Default is NONE. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events:<br><br>• product model/version on boot up (INFO level)<br><br>• NAT related info (INFO level)<br><br>• sent or received SIP message (DEBUG level)<br><br>• SIP message summary (INFO level)<br><br>• inbound and outbound calls (INFO level)<br><br>• registration status change (INFO level)<br><br>• negotiated codec (INFO level)<br><br>• Ethernet link up (INFO level)<br><br>• SLIC chip exception (WARNING and ERROR levels)<br><br>• memory exception (ERROR level)<br><br>The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components:<br> GS_LOG: [device MAC address][error code] error message<br><br>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 |
| **Reply to ICMP on WAN port** | Unit will not respond to PING from WAN side if set to "No". |

| | |
|---|---|
| **WAN Side HTTP Access** | Default is "No". The access to configuration page via WAN port is disabled. This setting has no effect if the device is in Bridge mode. |
| **PSTN Access Code** | Default is "*00". User can switch the phone to PSTN line connected to the Line port of ATA and make outgoing calls. |
| **Session Expiration** | Grandstream implemented SIP Session Timer. The session timer extension enables SIP sessions to be periodically "refreshed" via a re-INVITE request. Once the session interval expires, if there is no refresh via a re-INVITE message, the session will be terminated.<br><br>Session Expiration is the time (in seconds) at which the session is considered timed out, if no successful session refresh transaction occurs beforehand. The default value is 180 seconds. |
| **Min-SE** | The minimum session expiration (in seconds).  The default value is 90 seconds. |
| **Caller Request Timer** | If selecting "Yes" the phone will use session timer when it makes outbound calls if remote party supports session timer. |
| **Callee Request Timer** | If selecting "Yes" the phone will use session timer when it receives inbound calls with session timer request. |
| **Force Timer** | If selecting "Yes" the phone will use session timer even if the remote party does not support this feature. Selecting "No" will allow the phone to enable session timer only when the remote party support this feature.<br>To turn off Session Timer, select "No" for Caller Request Timer, Callee Request Timer, and Force Timer. |
| **UAC Specify Refresher** | As a Caller, select UAC to use the phone as the refresher, or UAS to use the Callee or proxy server as the refresher. |
| **UAS Specify Refresher** | As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the phone as the refresher. |
| **Force INVITE** | Session Timer can be refreshed using INVITE method or UPDATE method. Select "Yes" to use INVITE method to refresh the session timer. |
| **Firmware Upgrade and Provisioning** | Default 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, HT486 rev. 2.0 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** | Default is "No". If set to "Yes", configuration file would be authenticated before acceptance. End user should use default setting. |
| **Lock keypad update** | If this parameter is set to "Yes", except for IVR MENU items 1 to 5, the configuration update via keypad is disabled. |
| **Allow conf SIP Account in Basic Settings** | Default "No". If set to "Yes", user ID, authentication IP, authentication password and display name can be configured in BASIC SETTINGS page. |

### 6.2.4   Saving the Configuration Changes

Once a change is made, users should click on the "Update" button in the Configuration page.  The HandyTone-486 will then display the following screen to confirm that the changes have been saved.

**Grandstream Device Configuration**

STATUS BASIC SETTINGS ADVANCED SETTINGS

Your configuration changes have been saved.
They will take effect on next reboot.

Reboot

All Rights Reserved Grandstream Networks, Inc. 2005
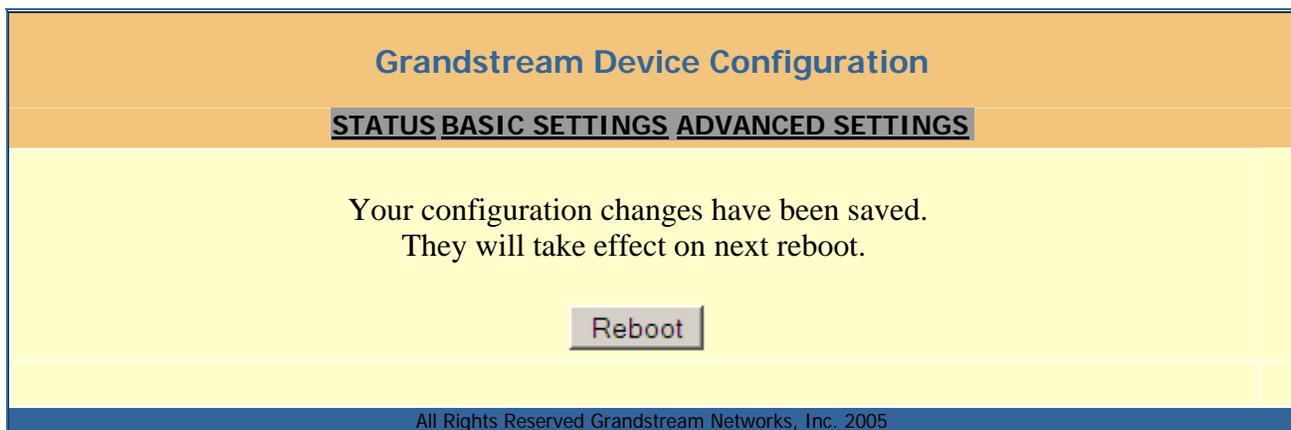
### 6.2.5   Rebooting the HandyTone-486 from remote

The user/administrator of the HandyTone-486 can remotely reboot the HandyTone-486 by pressing the "Reboot" button at the bottom of the configuration page. Once done, the following screen will be displayed to indicate that rebooting is underway.

**Grandstream Device Configuration**

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

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# 6.3     Configuration through a Central Server

Grandstream HandyTone ATAs can be automatically configured from a central provisioning system.

When HandyTone ATA boot up, it will send TFTP or HTTP request to download configuration file, "cfg000b82xxxxxx", where "000b82xxxxxx" is the MAC address of the HandyTone ATA.

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

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

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

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

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

# 7    Software Upgrade

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

## 7.1    Firmware Upgrade through TFTP/HTTP

To upgrade via TFTP or HTTP, the "Firmware Upgrade and Provisioning upgrade via" field needs to be set to TFTP or HTTP, respectively.  "Firmware Server Path" needs to be set to a valid URL of a TFTP or HTTP server, server name can be in either FQDN or IP address format. Here are examples of some valid URL.
     e.g.  firmware.mycompany.com:6688/Grandstream/1.0.8.16
     e.g. 168.75.215.189

**NOTES:**

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

## 7.2      Configuration File Download

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

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

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

## 7.3      Firmware and Configuration File Prefix and Postfix

Starting from firmware version 1.0.7.11 for HandyTone-486 Rev 2.0, adding prefix and postfix for both firmware and configuration file is supported.

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

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

## 7.4      Managing Firmware and Configuration File Download

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

# 8    Restore Factory Default Setting

**Warning !!!**

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

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

**Step 1:**
Find the MAC Address of the device. It is a 12 digits HEX number located on the bottom of the unit.

**Step 2:**
Encode the MAC address. Please use the following mapping:

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

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

**Step 3:**
To perform factory reset:

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

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

**NOTE:**

- Please be aware by default the HandyTone-486 WAN side HTTP access is disabled. After a factory reset, the device's web configuration page can be accessed only from its LAN port, please refer to instructions in section 6.2.1 for details.

# 9    Glossary of Terms

**ADSL**

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

**AGC**

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

**ARP**

Address Resolution Protocol is a protocol used by the Internet Protocol (IP) [RFC826], pecifically 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, geneate artificial background noise used in radio and wireless communications to fill the silent time in a transmission resulting from voice activity detection.

**DATAGRAM**

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

**DECIMATE**

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

**DECT**

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

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

**DNS**

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

**DID**

Direct Inward Dialing

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

**DSP**

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

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

Grandstream products all have DSP chips built inside.

**DTMF**

Dual Tone Multi Frequency

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

**FQDN**

Fully Qualified Domain Name

A FQDN consists of a host and domain name, including top-level domain. For example, [www.grandstream.com](http://www.grandstream.com) is a fully qualified domain name. www is the host, grandstream is the second-level domain, and.com is the top level domain.

**FXO**

Foreign eXchange Office

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

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

FXO is complimentary to FXS (and the PSTN).

**FXS**

Foreign eXchange Station

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

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

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

**DHCP**

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

**ECHO CANCELLATION**

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

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

**H.323**

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

**HTTP**

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

**IP**

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

**IP-PBX**

IP-based Private Branch Exchange

**IP Telephony**

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

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

## IVR

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

## MTU

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

## NAT

Network Address Translation

## NTP

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

## OBP/SBC

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

## PPPoE

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

## PSTN

Public Switched Telephone Network

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

**RTCP**

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

**RTP**

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

**SDP**

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

**SIP**

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

**STUN**

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

**TCP**

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

**TFTP**

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

**UDP**

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

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

**VAD**

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

**VLAN**

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

**VoIP**

Voice over IP

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