



Grandstream Networks, Inc.

BT-Series

BT-200 Dual Ethernet Port SIP Phone



BT-200 User Manual
Firmware Version 1.1.1.14

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GUI INTERFACES BT-200 USER'S MANUAL http://www.grandstream.com/GUI/GUI_BT200.rar

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WELCOME

Thank you for purchasing the Grandstream BT-200 SIP phone. The BT-200 is an affordable, next generation SIP phone that features excellent audio quality and rich telephony features. The BT-200 supports popular voice codecs and is designed for full SIP compatibility and interoperability with 3rd party SIP providers, thus enabling you to fully leverage the benefits of VoIP technology.

This manual will help you learn how to operate and manage your SIP phone and make the best use of its many upgraded features including 3-way conferencing, full-duplex hands-free speakerphone, voicemail with indicator, and custom ring tones. This phone is very easy to use and is designed to be a basic VoIP solution for the home or office.

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 contains links to Grandstream GUI Interfaces. Please download examples from http://www.grandstream.com/user_manuals/GUI/GUI_BT200.rar for your reference.
- This document is subject to change without notice. The latest electronic version of this user manual is available for download @: http://www.grandstream.com/user_manuals/BT-200.pdf
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CONNECTING YOUR PHONE

Connecting your BT-200 is easy. Before you begin, please verify the contents of the BT-200 package.

EQUIPMENT PACKAGING

The BT-200 phone package contains:

1. One BT-200 Main Case
2. One Handset
3. One phone cord
4. One Universal Power Adapter
5. One Ethernet cable

CONNECT YOUR PHONE

Connecting your phone is simple. First, connect the handset to the handset port. Next, connect the network port on the back of your phone to the Ethernet port on your router using the Ethernet cable included in your package. Then connect your BT-200 to your PC using the PC port. Lastly, connect your power supply to your phone and wall outlet. Please ensure your BT-200 has the latest firmware @ <http://www.grandstream.com/y-firmware.htm>.

FIGURE 1: BT-200 ILLUSTRATION



CONNECT YOUR PHONE TO THE VOIP NETWORK

Follow these steps to connect your phone to the VoIP network:

1. Ensure your phone is connected to the Internet and the power socket.
2. Wait 1 minute and press MENU.
3. Press Arrow down 1 time to see on display [2] IP Address
4. Press MENU again to see IP address on display
5. Enter this IP in Internet Explorer browser.
6. Login as administrator (password on default: admin)
 - a. Type your SIP ID numbers
 - b. Type your NAME used for Member Login
 - c. Type your PASSWORD used for Member Login
7. **REBOOT** the BT-200 phone

TABLE 1: BT-200 CONNECTOR DEFINITIONS

POWER	5V power port; Connect to your phone and the wall outlet
LAN	10/100 Mbps Switch LAN port to connect your phone to an Ethernet port.
PC	10/100 Mbps Switch port to connect your phone to the PC.
HEADSET	2.5mm Headset port

SAFETY COMPLIANCES

The BT-200 phone complies with FCC/CE and various safety standards. The BT-200 power adaptor is compliant with UL standard. Only use the universal power adapter provided with the BT- package. The manufacturer's warranty does not cover damages to the phone caused by unsupported power adaptors.

WARRANTY

If you purchased your BT-200 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. Grandstream reserves the right to remedy warranty policy without prior notification.

Warning: Please do not use a different power adaptor with the BT-200 as it may cause damage to the products and void the manufacturer warranty.

PRODUCT OVERVIEW

The BT-200 SIP phone is designed to provide all the feature functionality of a traditional phone plus advanced features like voicemail, 3-way conferencing, remote configuration, and dual 100 Mbps switched or routed network ports. The BT-200 is based on SIP Industry Standards and supports popular voice codecs and is designed for full SIP compatibility and interoperability with 3rd party SIP providers. It's simple to use, configure and manage.

KEY FEATURES

- Interoperable & compatible with SIP platforms
- Two (2) 100 Mbps auto sensing Ethernet RJ45 ports (switched or routed)
- Full duplex speakerphone
- Standard voice features and functionality
- Advanced functionality
- Voicemail indicator with light

Lines	Ethernet Ports	DHCP	Security	Voice Mail Indicator	Voice Codec	Remote Configuration
1 Line / 1 SIP account	2 RJ45 (LAN) (switched/routed)	Client/ Server	SRTP, TLS (pending)	Yes	G.722, G.723, G.711, G.729, G.726, GSM, iLBC	TFTP/HTTP

TABLE 2: TECHNICAL SOFTWARE FEATURES

Feature	Software
Lines	1 direct number
Protocol Support(1)	IETF SIP (RFC 3261 and companion RFCs)
Display	12-digit caller ID LCD
Feature Keys	8 dedicated keys: Message Button, Hold, Transfer, Conference, Flash, Speakerphone, Send, Mute/Del, 5 display/menu navigation keys
Ethernet Ports	Dual 100 Mbps Ethernet LAN and PC ports (switched or routed)
Device Management	<p>Web interface or via secure (AES encrypted) central configuration file for mass deployment</p> <p>NAT-friendly remote software upgrade (via TFTP/HTTP) for deployed devices including behind firewall/NAT</p> <p>Support Layer 2 (802.1Q, VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)</p> <p>Auto/manual provisioning system</p> <p>Keypad & LCD</p>
Audio Features	<p>Full-duplex hands-free speakerphone</p> <p>Advanced Digital Signal Processing (DSP)</p> <p>Dynamic negotiation of codec and voice payload length</p> <p>Support for G.723,1 (5.3K/6.3K), G.729A/B, G.711 μ/A, G.726, G.722 (wide-band), GSM and iLBC codecs</p> <p>In-band and out-of-band DTMF (in audio, RFC2833, SIP INFO)</p> <p>Silence Suppression, VAD (voice activity detection), CNG (comfort noise generation), ANG (automatic gain control)</p> <p>Adaptive jitter buffer control</p> <p>Packet delay & loss concealment</p>
Call Handling Features	<p>Voice mail indicator with indicator, 3-way conferencing, Caller ID display or block, Call waiting, transfer, hold, forward, flash, mute</p> <p>Redial, call log, and volume control</p> <p>Dial plans, off-hook auto dial, early dial, click-to-dial</p>
Network and Provisioning	<p>TCP/UDP/IP, RTP/RTCP, HTTP, ARP/RARP, ICMP, DNS, DHCP, NTP, TFTP, PPPoE protocols</p> <p>Manual or dynamic host configuration protocol (DHCP) network setup</p> <p>Support for IEEE 802.1p/Q tagging (VLAN), Layer 3 TOS</p> <p>RTP and NAT support traversal via STUN and external media proxy</p>
Advanced Server Features	<p>Message waiting indication</p> <p>Server redundancy – support for fail-over SIP server and DNS server</p>
Security	<p>DIGEST authentication and encryption using MD5 and MD5-sess</p> <p>SRTP and TLS (pending)</p>

HARDWARE FEATURES

- Headset with auto switch when in use
- 10/100 Mbps Full/Half Duplex Ethernet Switch with LAN and PC port,
- Auto detect Ethernet polarity - use either straight through or twist cable
- Message Waiting Indicator – RED LED

TABLE 2: TECHNICAL HARDWARE SPECIFICATIONS

LAN interface	2 x RJ45 10/100Mbps Ethernet Ports
Headset Jack	2.5mm Headset port
LED	1 LED in RED color
Phone Case	25-button keypad 12-digit caller ID LCD
Universal Switching Power Adaptor	Input: 100-240VAC 50-60 Hz Output: +5VDC, 1200mA, UL certified
Dimension	18cm (W) x 22cm (D) x 6.5cm (H)
Weight	0.9kg (2lbs)
Temperature	40 - 130°F 5 – 45°C
Humidity	10% - 90% (non-condensing)
Compliance	FCC / CE / C-Tick

USING THE BT-200 IP PHONE

GETTING FAMILIAR WITH THE LCD / LED

BT-200 has a numeric backlit LCD measuring 64mm x 24mm which displays icons to illustrate the mode(s) of the phone. The LCD turns on with an incoming call. If voicemail is configured, there is also a RED LED message indicator.

FIGURE 2: BT-200 LCD ILLUSTRATION



TABLE 4: LCD ICON DEFINITIONS

Icon	LCD Icon Definitions
	Connectivity Status / SIP Proxy/Server Icon: Solid – connected to SIP Server/IP address received Blinking – physical connection failed Blank – SIP Proxy/Server not registered
	Phone Status Icon: OFF when the handset is on-hook ON when the handset is off-hook
	Speaker Phone Status Icon: FLASH when phone rings or a call is pending OFF when the speakerphone is off ON when the speakerphone is on
	Calls Forwarded Icon: INDICATES calls are forwarded Currently not implemented
	Alarm Clock Icon: Currently not implemented
	RTP Encryption Icon: Ensures data security Currently not implemented
	Keypad Lock Icon: Locks the keypad Currently not implemented
	Handset, Speakerphone and Ring Volume Icon: Each icon appears next to the volume icon To adjust volume, use the up/down button
	Real-time Clock: Synchronized to Internet time server Time zone configurable via web browser AM/PM indicator
	Call Logs and IP Address Separator Icons: 01-10 for <i>CALLED</i> history (dialed number) 01-10 for <i>CALLERS</i> history (Incoming caller ID) Numbers and Characters: 0 – 9, “* key” = *, “# key” = #

GETTING FAMILIAR WITH KEYPAD

The BT-200 keypad is the same as a traditional phone, with additional feature functionality including speaker, menu keys, LCD navigation keys, and advanced feature keys which include hold, transfer, conference and flash.

FIGURE 3: BT-200 KEYPAD ILLUSTRATION



TABLE 5: KEYPAD BUTTON DEFINITIONS

Key Button	Key Button Definitions
UP ↑	Use button to select next “Menu Item” when phone is in keypad configuration mode; Or increase handset/speakerphone volume when phone is ACTIVE; Or increase ring volume when phone is in IDLE mode
DOWN ↓	Use button to select previous “Menu Item” when phone is in keypad configure mode; Or decrease handset/speakerphone volume when phone is ACTIVE; Or decrease ring volume when is in IDLE mode
MENU	Enter keypad MENU mode when phone is in IDLE mode. It is also the ENTER key once entering MENU
CALLED	After off hook, press to display the dialed numbers. When number displayed, press the SEND key can make call using that displayed number
CALLERS	After off hook, press to display the incoming Caller IDs. When number displayed, press the SEND key can make call using that displayed number
MSG	Enter to retrieve (video) voice mails or other messages
HOLD	Place ACTIVE call on hold
TRNF	TRANSFER key: Transfer an ACTIVE call to another number

CONF	Press CONF button to connect Calling/Called party into conference
FLASH	Flash to switch between two lines
SPEAKER	Enable/Disable hands-free speaker mode
SEND	Press SEND to dial a new number or redial last number dialed. Press send button to send a call immediately before “no key entry timeout” value expires
MUTE/DEL	Mute an active call; or Delete a key entry Also used to ‘REJECT’ incoming call.
0 - 9, *, #	Standard phone keypad; press # key to send call; * key for special needs

MAKING AND ANSWERING PHONE CALLS

HANDSET, SPEAKERPHONE AND HEADSET MODE

The BT-200 handset can be toggled between speaker mode (hands-free) or headset mode. To toggle between the handset and speaker/headset, press the *Hook Flash* in the handset cradle or press the SPEAKER button.

PLACING A TRADITIONAL PHONE CALL

There are four ways to place a call:

1. Use handset, headset or press SPEAKERPHONE button and dial.
2. Press the SEND button directly to redial last number. Last dialed number will be displayed on the LCD as the number is dialed.
3. Browse the CALLED/CALLER history, select the number using the navigation keys and press SEND button to place the call.
4. Use the menu key to browse the “CALLED/CALLERS” menu to scroll through the last 10 numbers dialed or received. Select number using the navigation keys and press SEND button to place the call.

Examples:

- To dial another extension on the same proxy, such as 1008, pick up handset or press speaker, dial 1008 and then press SEND.
- To dial an outside number such as 626-666-7890, enter prefix if necessary and dial number and press SEND. Check with your VoIP provider for more information about prefixes.

PLACING CALLS USING AN IP ADDRESS

Direct IP calling allows two parties using a BT-200 and another VoIP device to talk with each other in an ad hoc fashion without a SIP proxy. VoIP calls can be made between two phones if:

- both phone/VoIP device have public IP addresses, or
- both phone/VoIP device are on a same LAN/VPN using private or public IP addresses, or
- both phone/VoIP device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ)

PLACING A DIRECT IP-IP CALL

To place a direct IP to IP call, first off-hook, then press the “MENU” key, then enter a 12-digit target IP address to make the call. If port is not default 5060, 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 MENU 192 168 000 010 followed by pressing the “SEND” key or wait for seconds in the No Key Entry Timeout.
- If the target IP address/port is 192.168.1.20:5062, then the dialing convention is MENU_key 192168001020*45062 followed by the SEND key or wait for 4 seconds.

Placing a Quick IP Call

The BT-200 can dial an IP address under the same LAN segment by dialing the last octet in the IP address. Controlled static IP usage is recommended.

In the “Advanced Settings” page, set the “Use Quick IP-call Mode” to YES. When #xxx is dialed, where x is 0 – 9 and xxx <= 255, a direct IP call to aaa.bbb.ccc.XXX is completed. “aaa.bbb.ccc” is from the local IP address *regardless* of subnet mask. The numbers #xx or #x are also valid. The leading 0 is not required but can be used.

Examples:

- 192.168.0.2 calling 192.168.0.3 just dial #3 follow by SEND or #
- 192.168.0.2 calling 192.168.0.23 just dial #23 follow by SEND or #
- 192.168.0.2 calling 192.168.0.123 just dial #123 follow by SEND or #
- 192.168.0.2 dial # 3 and # 03 and # 003 has same effect --> call 192.168.0.3

NOTE: If you have a SIP Server configured, a Direct IP-IP call still works. If you are using STUN, the Direct IP-IP call will also use STUN. Configure the “Use Random Port” to “NO” when completing Direct IP calls.

ANSWER AN INCOMING CALL

The BT-200 works like any other traditional phone. There are two ways to answer an incoming call:

1. Pick up the handset to answer the call normally using handset, or
2. Press the SPEAKER button to answer in speakerphone or headset mode

Call Hold

Press the HOLD button to place a call on hold. Press the HOLD button again to release the previously held call.

Call Waiting and Call Flashing

If call waiting feature is enabled, a call waiting tone (stutter tone) will alert the caller to an incoming call. Press the FLASH button to place the current party on hold and switch to the incoming call. Press the flash button to toggle between two active calls.

Blind and Attended Call Transfer

BT-200 supports both *blind* and *attended* call transfer. Each is easy to use. Use *blind* transfer if you want to transfer a call without speaking with someone first; use *attended* transfer if you want to speak with the xxxxxxxx prior to transferring call.

Blind Transfer

Transfer an active call to a third party without announcement.

Press the TRANSFER button and wait for a dial tone. Dial the third party's phone number followed by the SEND button.

Expected outcomes:

- a) A quick confirmation tone (call waiting tone) followed by a dial tone. This indicates the transfer is successful. Hang up or place another call.
- b) A quick busy tone followed by a restored call (on supported platforms only). The busy tone indicates the transfer failed.
- c) Continuous busy tone. The phone call has timed out. **NOTE:** continuous busy tone does not indicate the transfer has been successful, nor does it indicate the transfer has failed. It often means there was a failure to receive second NOTIFY – check firmware for most recent release.

NOTE: The “*Enable Call Feature*” must be configured to “Yes” in the web configuration page to enable this feature.

Attended Transfer

Transfer an active call to a third party with an accompanying message. User has the option to speak to the intended receiver of call prior to transferring the call.

Press the FLASH button for a second dial tone. Dial the third party's phone number and press the SEND button. If the call is answered, press TRANSFER to complete the transfer and hang up. If the call is not answered, press FLASH to resume the original call.

NOTE: If Attended Transfer fails, and party A hangs up, the HandTone-496 will ring party A to remind A that party B is still on the line. Party A can pick up the phone to resume conversation with party B.

3-way Conference

The BT-200 supports the advanced 3-way conferencing feature. To use 3-way conferencing, assume Caller A and Caller B wish to conference Caller C.

1. Press CONFERENCE for dial-tone, placing Caller B on hold. Dial third number and press SEND.
2. Press CONFERENCE again to establish 3-way conference call.
3. If Caller C does not answer, press FLASH to resume conversation with Caller B.

NOTE: If either Caller B or Caller C drops the call, the two remaining parties can still talk. However, if Caller A (conference initiator) ends the call, the conference is terminated and all lines hang up.

Retrieving Messages

The blinking red MWI (Message Waiting Indicator) indicates a message is waiting. Press the MSG button to retrieve the message. An IVR will prompt the user through the message retrieval menu.

Mute/Delete

1. To enable/disable mute during a call, press MUTE/DEL button.

2. To delete entries when dialing, press MUTE/DEL to delete the digit.
3. To send a call directly to voicemail (without picking up the call), press MUTE/DEL

CALL FEATURES

The BT-200 supports an advanced call feature list including but not limited to Caller ID Block (or Anonymous Call), Disable/Enable Call Waiting, Call Forward on Busy, Delay, or Unconditional. The following table outlines each call feature available on the BT-200.

TABLE 6: CALL FEATURE DEFINITIONS

Key	Call Features
*30	Block Caller ID (for all subsequent calls)
*31	Send Caller ID (for all subsequent calls)
*67	Block Caller ID (per call)
*82	Send Caller ID (per call)
*70	Disable Call Waiting (per Call)
*71	Enable Call Waiting (per Call)
*72	Unconditional Call Forward Dial “*72” for a dial tone. Dial the forwarding number followed by “#”. Wait for dial tone. LCD will display “Call FWD Activated”.
*73	Cancel Unconditional Call Forward: dial “*73” and get the dial tone, then hang up. LCD will display “Call FWD Activated”.
*90	Busy Call Forward Dial “*90” for a dial tone. Dial the forwarding number followed by “#”. Wait for a dial tone. Hang up.
*91	Cancel Busy Call Forward: dial “*91”. Wait for dial tone. Hang up.
*92	Delayed Call Forward Dial “*92” for a dial tone. Dial the forwarding number followed by “#”. Wait for a dial tone. Hang up. LCD will display “Call FWD Activated”.
*93	Cancel Delayed Call Forward Dial “*93” for a dial tone, then hang up.

CONFIGURATION GUIDE

This section of the BT-200 manual will help guide you to configure your phone, either using the keypad or through the web interface. Examples of the web interfaces can be downloaded @ http://www.grandstream.com/user_manuals/GUI/GUI_BT200.rar.

CONFIGURATION WITH KEYPAD

Configuring the BT-200 using the key pad is simple. If at anytime the phone rings or there is a 20+ second idle time, while in the configuration state, the phone will exit the configuration menu. You can answer an incoming call as usual during the configuration stage.

First, be sure your phone is not in use. Press the MENU button to enter key pad menu. There are eleven main menu functions ranging from 1 – 11. Navigate the menu options using the navigation keys on the keypad. Definitions are found in Table 7.

TABLE 7: CONFIGURATION MENU DEFINITIONS

Menu Item	Menu Functions
1	Display “[1] dhcP On ” or “[1] dhcP oFF” for the current selection Press MENU key to enter edit mode Press ‘↓’ or ‘↑’ to toggle the selection Press MENU to save and exit Must recycle power for changes to take effect.
2	Display “[2] IP Addr ” Press MENU to display the current IP address Enter new IP address if DHCP is OFF Press ‘↓’ or ‘↑’ to exit Press MENU to (save and) exit Must recycle power for changes to take effect.
3	Display “[3] SubNet ” Press MENU to display the Subnet mask Enter new Subnet mask if DHCP is OFF Press ‘↓’ or ‘↑’ to exit Press MENU to (save and) exit Must recycle power for changes to take effect.
4	Display “[4] routEr ” Press MENU to display the Router/Gateway address Enter new Router/Gateway address if DHCP is OFF Press ‘↓’ or ‘↑’ to exit Press MENU to (save and) exit Must recycle power for changes to take effect.

Menu Item	Menu Functions
5	<p>Display “[5] dnS ”</p> <p>Press MENU to display the DNS address</p> <p>Enter new DNS address if DHCP is OFF</p> <p>Press ‘↓’ or ‘↑’ to exit</p> <p>Press MENU to (save and) exit</p> <p>Must recycle power for changes to take effect.</p>
6	<p>Display “[6] tFTP ”</p> <p>Press MENU to display the TFTP address</p> <p>Enter new TFTP server IP address</p> <p>Press MENU to save</p> <p>Press ‘↓’ or ‘↑’ to exit</p>
7	<p>Display “[7] G-711u 2”</p> <p>Press MENU to select new codec</p> <p>Press ‘↓’ or ‘↑’ to browse a list of available codecs</p> <p>line 2 “ - G-711A 2”</p> <p>3 “ - G-723 1”</p> <p>4 “ - G-729 2”</p> <p>5 “ - GS 1”</p> <p>Press 1 to 9 to indicate number of frames per TX packet</p> <p>Press MENU to save and exit</p> <p>Must recycle power for changes to take effect.</p>
8	<p>Display “[8] SIP SP-1”</p> <p>Reserve for future products.</p>
9	<p>Display “[9] codE rEL”</p> <p>Press Menu to display the code releases</p> <p>Press ‘↓’ or ‘↑’ to browse</p> <p>line 1 “b 2006-09-28” – date: boot code</p> <p>2 “ 1. 1. 1. 3” – version: boot code</p> <p>3 “P 2006-10-02” – date: phone code</p> <p>4 “ 1. 1. 1. 14 – version: phone code</p> <p>5 “1r 2004-05-12” – date: 1st ring tone</p> <p>6 “ 1. 0. 0. 0” – version: ring tone</p> <p>7 “2r 2004-05-12” – date: 2nd ring tone</p> <p>8 “ 1. 0. 0. 0” – version: ring tone</p> <p>9 “3r 0000-00-00” – date: 3rd ring tone</p> <p>10 “ 1. 0. 0. 0” – version: ring tone</p> <p>(all zeroes means unavailable or unsupported)</p> <p>Press MENU to exit.</p>

Menu Item	Menu Functions
10	Display “[10] Phy Addr” Press MENU to display the physical / MAC address. Press ‘↓’ or ‘↑’ to exit.
11	Display “[11] ring 0” Press MENU to hear the selected ring tone, press ‘↓’ or ‘↑’ to select the stored ring tones. There are 3 available ring-tones: ring 0 (default), ring 1 and ring 2. ring 3 is unavailable or unsupported. Press MENU to select and exit.
Factory Reset	Display “ -- rESet --”. Please be very <u>careful</u> when using this function. Two options: factory reset or manual reboot. Factory Reset: <ol style="list-style-type: none"> 1. Key in the physical / MAC address on back of the phone. 2. Press MENU to reset phone to FACTORY DEFAULT settings. All your setting will be erased. Manual Reboot: Press MENU without keying anything.
Others	When phone is powered on and time is displayed Press ‘↓’ or ‘↑’, Display “ ring [4] ”, press ‘↓’ or ‘↑’ again to hear and adjust the ring tone volume, from 0 (off) to 7 (maximum), off and on hook to set Press SPEAKERPHONE button, or off hook and pick up handset, press ‘↓’ or ‘↑’ to adjust the speakerphone/headset or handset volume.

CONFIGURATION WITH WEB BROWSER

The BT-200 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 IP phone through any web browser.

Access the Web Configuration Menu

The IP Phone Web Configuration Menu can be accessed by the following URI: <http://Phone-IP-Address> where the [Phone-IP-Address](#) is the IP address of the phone. To find your IP address, press MENU and select the Status item to see “IP: IP Address”.

NOTE: When typing the IP address into a browser, strip out the leading “0” 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.

END USER CONFIGURATION

Once this HTTP request is sent from a Web browser, the web server will display a login screen. The password is case sensitive with maximum length of 25 characters. The factory default password for the End User is “123”. After login, the embedded Web server will display the Configuration page, explained in detail in Table 8.

TABLE 8: KEYPAD BUTTON DEFINITIONS

End User Password	This contains the password to access the Web Configuration Menu. This field is case sensitive with a maximum length of 25 characters.
IP Address	<p>There are two modes under which the BT-200 can operate:</p> <ol style="list-style-type: none"> 1. <u>DHCP Mode</u>: all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The BT-200 will acquire its IP address from the first DHCP server it discovers from the LAN it is connected. To use the PPPoE feature the PPPoE account settings need to be set. The BT-200 will attempt to establish a PPPoE session if any of the PPPoE fields is set. 2. <u>STATIC IP Mode</u>: 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 parameter controls how the date/time is displayed according to the specified time zone.
Daylight Savings Time	<p>This parameter controls whether the displayed time will be daylight savings time or not. If set to "Yes" and the Optional Rule is empty, then the displayed time will be 1 hour ahead of normal time. The "Automatic Daylight Saving Time Rule" shall have the following syntax: start-time;end-time;saving Both start-time and end-time have the same syntax: month,day,weekday,hour,minute month: 1,2,3,...,12 (for Jan, Feb, ..., Dec) day: [+]-1,2,3,...,31 weekday: 1, 2, 3, ..., 7 (for Mon, Tue, ..., Sun), or 0 which means the daylight saving rule is not based on week days but based on the day of the month. hour: hour (0-23), minute: minute (0-59)</p> <p>If "weekday" is 0, it means the date to start or end daylight saving is at exactly the given date. In that case, the "day" value must not be negative. If "weekday" is not zero and "day" is positive, then the daylight saving starts on the first "day"th iteration of the weekday (1st Sunday, 3rd Tuesday etc). If "weekday" 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 for "Automatic Daylight Saving Time Rule" shall be set to "04,01,7,02,00;10,-1,7,02,00;60" which is the rule for US.</p> <p>Examples US/Canada where daylight saving time is applicable: 04,01,7,02,00;10,-1,7,02,00;60 This means the daylight saving time starts from the first Sunday of April at 2AM and ends the last Sunday of October at 2AM. The saving is 60 minutes (1hour).</p>

Date Display Format	Allow user to choose among the following three formats: 1. Year-Month-Day 2. Month-Day-Year 3. Day-Month-Year
Display Account Name instead of Date	If set to Yes, it will display the information entered on Account Name field from Account page.
Device Mode	This parameter controls whether the device is working in NAT router mode or Bridge mode. Need save the setting and reboot the device before the setting start to work.
WAN side HTTP access	If set to "Yes", user can access the configuration page through the WAN port, instead of connecting PC and BT200 through the "PC" port to do the configuration. On the other hand, it exposes the BT200 to others, and may cause some security issues for users. Default is No.
Reply to ICMP on WAN port	If set to "Yes", The BT200 will respond to the PING command from other computers for testing, but it also is vulnerable to the DOS attack. Default is No.
Cloned WAN MAC Addr	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 function as a Gateway for the subnet. Default value is 192.168.2.1.
DHCP IP Lease Time	Value is set in units of hours. Default value is 120hr (5 Days.) The time IP address is assigned to the LAN clients.
DMZ IP	Forward all WAN IP traffic to a specific IP address if no matching port is used by HandyTone-486 itself or in the defined port forwarding.
Port Forwarding	Allow the 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, the end user can also access to the device Status page. The Device Status page allows the user to find device specific data. After login, the embedded Web server will display the Device Status page, explained in detail in **Table 9: Device Status Page Definitions**.

TABLE 9: DEVICE STATUS DEFINITIONS

MAC Address	The device ID, in HEX format. This is a very important ID for ISP troubleshooting.
IP Address	This field shows LAN IP address of BT-200
Product Model	This field contains the product model info.
Software Version	Program: This is the main software release. This number is always used for firmware upgrade. Boot and loader are not normally changed.
System Up Time	This field shows system up time since the last reboot.
Registered	This field indicates whether the device is registered to the SIP server(s).
PPPoE Link Up	This field shows whether the PPPoE connection is up if connected to DSL modem.
Detected NAT Type	Displays type of NAT the BT-200 is connected to via its LAN port. It is based on STUN protocol.

ADVANCED USER CONFIGURATION

To login to the Advanced User Configuration page, please follow the instructions in [End User Configuration](#) to access the login page. The password is case sensitive with a maximum length of 25 characters 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. After login, the embedded Web server will display the Advanced Configuration page, explained in detail in **Table 10: Advanced Configuration Page Definitions**.

FIGURE 4: ADVANCED USER CONFIGURATION SCREENSHOT

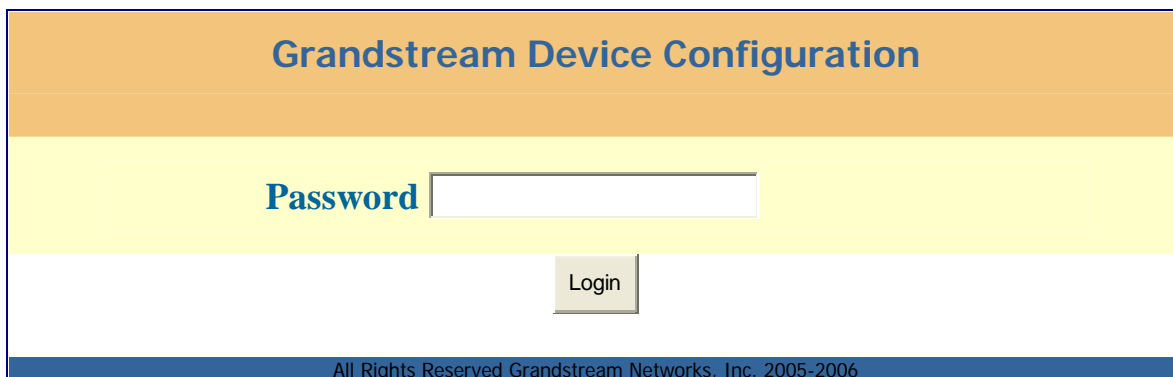


TABLE 10: ADVANCED CONFIGURATION PAGE DEFINITIONS

Admin Password	Administrator password. Only administrator can configure the “Advanced Settings” page. Password field is purposely left blank for security reason after clicking update and saved. The maximum password length is 25 characters.
Silence Suppression	This controls the silence suppression/VAD feature of G723 and G729. If set to “Yes”, when a silence is detected, small quantity of VAD packets (instead of audio packets) will be sent during the period of no talking. If set to “No”, this feature is disabled.
Voice Frames per TX	<p>This field contains the number of voice frames to be transmitted in a single Ethernet packet (be advised the IS limit is based on the maximum size of Ethernet packet is 1500 byte (or 120kbps)).</p> <p>When setting this value, be aware of the requested packet time (ptime, used in SDP message) is a result of configuring this parameter. This parameter is associated with the <u>first</u> 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 G.723 and the “Voice Frames per TX” is set to 2, then the “ptime” value in the SDP message of an INVITE request will be <u>60ms</u> because each G.723 voice frame contains 30ms of audio. Similarly, if this field is set to 2 and the first codec is G.729 or G.711 or G.726, then the “ptime” value in the SDP message of an INVITE request will be <u>20ms</u>.</p> <p>If the configured voice frames per TX exceeds the maximum allowed value, the BT-200 will use and save the maximum allowed value for the corresponding first codec choice. The maximum value for PCM is 10 (x10ms) frames; for G.726, it is 20 (x10ms) frames; for G.723, it is 32 (x30ms) frames; for G.729/G.728, 64 (x10ms) and 64 (x2.5ms) frames respectively.</p> <p>Grandstream recommends using the default settings. Grandstream does not recommend adjusting these parameters if you are an average user. Incorrect settings will affect the voice quality. Please refer to the Codec FAQ at http://www.grandstream.com/FAQ/FAQ-Codec.pdf for more technical detail.</p>
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	This contains the value used for layer 2 VLAN tag. Default setting is blank.
No Key Entry Timeout	Default is 4 seconds.
Use # as Send Key	This parameter allows users to configure the “#” key as the “Send” (or “Dial”) key. If set to “Yes”, the “#” key will immediately send the call. In this case, this key is essentially equivalent to the “(Re)Dial” key. If set to “No”, the “#” key is included as part of the dial string.
Local RTP port	This parameter defines the local RTP-RTCP port pair the BT-200 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 is 5004.

Use Random Port	This parameter, when set to Yes, will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple BT-200s are behind the same NAT.
Keep-alive interval	This parameter specifies how often the BT-200 sends a blank UDP packet to the SIP server in order to keep the “hole” on the NAT open. Default is 20 seconds.
Use NAT IP	NAT IP address used in SIP/SDP message. Default is blank.
STUN Server	IP address or Domain name of the STUN server.
Firmware Upgrade and provisioning	This radio button will enable BT-200 to download firmware or configuration file through either TFTP or HTTP.
Via TFTP Server	<p>This is the IP address of the configured TFTP server. If selected and it is non-zero or not blank, the BT-200 will retrieve new configuration file or new code image from the specified TFTP server at boot time. If after 3 attempts boot-up fails, the server will timeout and then start the boot process using the existing code image in the Flash memory. If a TFTP server is configured and a new code image is retrieved, the new downloaded image will be verified and then saved into the Flash memory.</p> <p>NOTE: Please do NOT interrupt the TFTP upgrade process (especially the power supply) as this will damage the device. Depending on the network environment this process can take up to 15 or 20 minutes.</p>
Via HTTP Server	<p>The URL for the HTTP server used for firmware upgrade and configuration via HTTP. i.e. http://provisioning.mycompany.com:6688/Grandstream/1.0.5.16 Here “:6688” is the specific TCP port that the HTTP server is listening at, it can be omitted if using default port 80.</p> <p>NOTE: If Auto Upgrade is set to No, BT-200 will only do HTTP download once at boot up.</p>
Allow DHCP Option 66 to override server	DHCP Option 66 is used to identify a TFTP server when the 'sname' field in the DHCP header has been used for DHCP options. If you choose yes, BT-200 will use the TFTP server resolved from DHCP, instead of the one you specified in the "TFTP Server" option above.
Automatic Upgrade	<p>Choose Yes to enable automatic upgrade and provisioning.</p> <p>In “Check for new firmware every” field, enter the number of days to enable BT-200 to check the server for firmware upgrade or configuration in the defined period of days. When set to No, BT-200 will only upgrade once at boot up.</p> <p>You also have the following options:</p> <ul style="list-style-type: none"> • “Always check for New Firmware” • “Check New Firmware only when F/W pre/suffix changes” • “Always skip firmware check.”
Authenticate Conf File	If set to Yes, cfg file will be authenticated before acceptance. Useful to protect against unauthorized changes to the device configuration.

Off-hook Auto Dial	This parameter allows the user to configure a User ID or extension number to be automatically dialed upon off-hook. Please NOTE that only the user part of a SIP address needs to be entered here. The phone will automatically append the "@" and the host portion of the corresponding SIP address.
DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833.
Syslog Server	The IP address or URL of System log server. This feature is especially useful for ITSP (Internet Telephone Service Provider).
Syslog Level	<p>Select the BT-200 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:</p> <ol style="list-style-type: none"> 1. product model/version on boot up (INFO level) 2. NAT related info (INFO level) 3. sent or received SIP message (DEBUG level) 4. SIP message summary (INFO level) 5. inbound and outbound calls (INFO level) 6. registration status change (INFO level) 7. negotiated codec (INFO level) 8. Ethernet link up (INFO level) 9. SLIC chip exception (WARNING and ERROR levels) 10. memory exception (ERROR level) <p>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</p> <p><i>Example:</i> May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up</p>
NTP server	URI or IP address of the NTP (Network Time Protocol) server, which will be used by the phone to synchronize the date and time.
Allow DHCP Option 42 to override NTP server	DHCP Option 42 specifies a list of IP addresses for Network Time Protocol (NTP) servers available to the client. If you choose yes, BT200 will use the NTP servers resolved from DHCP, instead of the one you specified in the "NTP Server" option above.
Distinctive Ring Tone	Customer Ring Tone 1 to 3 with associate Caller ID: when selected, if Caller ID is configured, then the device will ONLY sound this ring tone when the incoming call is from the Caller ID, device will use System Ring Tone for all other calls. When selected but no Caller ID is configured, the selected ring tone will be used for all incoming calls.
Disable Call Waiting	Default is No.

Quick IP Call Mode	<p>Dial an IP address under the same LAN segment by simply pressing the last octet in the IP address.</p> <p>In the Advanced Settings page the option "Use Quick IP-call mode", is set to No by default. When this option is set to YES, and #XXX is dialed, where X is 0-9 and XXX <=255, phone will make direct IP call to aaa.bbb.ccc.XXX where aaa.bbb.ccc comes from the local IP address REGARDLESS of subnet mask.</p> <p>#XX or #X are also valid so leading 0 is not required (but OK). eg. 192.168.0.2 calling 192.168.0.3 just dial #3 follow by SEND or # 192.168.0.2 calling 192.168.0.23 just dial #23 follow by SEND or # 192.168.0.2 calling 192.168.0.123 just dial #123 follow by SEND or #</p> <p>192.168.0.2 dial #3 and #03 and #003 has same effect --> call 192.168.0.3</p> <p>NOTE: If you have a SIP Server configured, Direct IP-IP call will still work. However, if you are using STUN, Direct IP-IP call will also use STUN.</p>
Lock keypad update	<p>If this parameter is set to "Yes", the configuration updates via keypad for Menu Item 7, 9, 12 are disabled.</p>

TABLE 11: INDIVIDUAL SETTINGS DEFINITIONS

Account Active	This field indicates whether the account is active or not. The default value for the primary account Account 1 is Yes. The default values for the other three accounts are No.
Account Name	A name to identify an account which will be displayed in LCD.
SIP Server	SIP Server's 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 BT-200 for firewall or NAT penetration in different network environment. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy can 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	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
Authenticate Password	SIP service subscriber's account password for BT-200 to register to (SIP) servers of ITSP.
Name	SIP service subscriber's name which will be used for Caller ID display.
Use DNS SRV:	Default is No . If set to Yes the client will use DNS SRV to look up server.

User ID is Phone Number	If the BT-200 has an assigned PSTN telephone number, this field should be set to "Yes". Otherwise, set it to "No". If "Yes" is set, a "user=phone" parameter will be attached to the "From" header in SIP request
SIP Registration	This parameter controls whether the BT-200 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 SIP user's registration information will be cleared on reboot.
Register Expiration	This parameter allows user to specify the time frequency (in minutes) that BT-200 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 BT-200 will listen and transmit. The default value for Account 1 is 5060. It is 5062, 5064, 5066 for Account 2, Account 3 and Account 4 respectively.
SIP T1 Timeout	T1 is an estimate of the round-trip time (RTT) between the client and server transactions. If the network latency is high, select bigger value for reliable usage.
SIP T2 Interval	This element sets the value of the SIP protocol T2 timer, in seconds. Timer T2 defines the retransmit interval for INVITE responses and non-INVITE requests. The SIP protocol default value is 4 seconds.
NAT Traversal	This parameter defines whether the BT-200 NAT traversal mechanism will be activated or not. If activated (by choosing "Yes") and a STUN server is also specified, then the BT-200 will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the BT-200 will attempt to detect if and what type of firewall/NAT it is sitting behind through communication with the specified STUN server. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the BT-200 will attempt to use its mapped public IP address and port in all of its SIP and SDP messages. If the NAT Traversal field is set to "Yes" with no specified STUN server, the BT-200 will periodically (every 20 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open.
Subscribe for MWI:	Default is No . When set to "Yes" a SUBSCRIBE for Message Waiting Indication will be sent periodically.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Voice Mail User ID	When configured, user will be able to dial voice mail server by pressing "MSG" button.
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.
Early Dial	Default is No . Use only if proxy supports 484 response.
Dial Plan Prefix	Sets the prefix added to each dialed number.

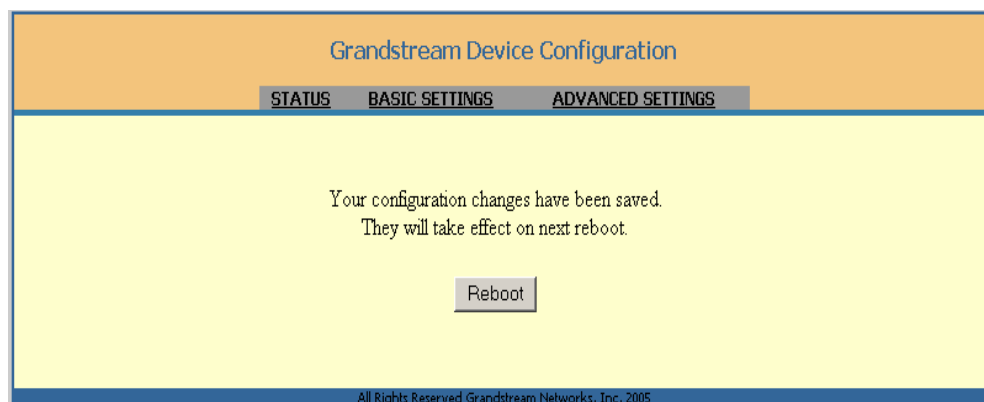
Enable Call Features	Default is No . If set to Yes, Call transfer, Call Forwarding & Do-Not-Disturb are supported locally.
Disable Missed-Call	Default is No . If set to Yes, missed calls will not be recorded for your review.
Session Expiration	<p>The SIP Session Timer extension enables SIP sessions to be periodically “refreshed” via a SIP request (UPDATE, or re-INVITE. Once the session interval expires, if there is no refresh via a UPDATE or re-INVITE message, the session is terminated.</p> <p>Session Expiration is the time (in seconds) at which the session is considered timed out, provided no successful session refresh transaction occurs beforehand. The default value is 180 seconds.</p>
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	<p>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.</p> <p>To turn off Session Timer, select “No” for Caller Request Timer, Callee Request Timer, and Force Timer.</p>
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.
Enable 100rel	The use of the PRACK (Provisional Acknowledgment) method enables reliability to be offered to SIP provisional responses (1xx series). This is very important if PSTN internetworking is to be supported. A user’s wish to use reliable provisional responses is invoked by the 100rel tag which is appended to the value of the required header of initial signalling messages.
Account Ring Tone	<p>There are 4 different ring tone that are defined:</p> <p>System Ring Tone: when selected, all calls will ring with system ring tone.</p> <p>Customer Ring Tone 1 to 3: when selected, BT-200 will ONLY play this ring tone for all the incoming calls for this account.</p>
Send Anonymous	If this parameter is set to “Yes”, the “From” header in outgoing INVITE message will be set to anonymous, essentially blocking the Caller ID from displaying.
Auto Answer	When set to “Yes”, BT-200 will automatically switch to speaker when there is an incoming call.

Allow Auto Answer by Call-Info	Default is No . If set to Yes, auto answer depends on the Call-Info in the SIP message. This feature needs the support of IP-PBX.
Turn off speaker on remote disconnect	Default is No . If set to Yes, the speaker will turn off, and the phone will go back to idle status, after the other party of the call hands up.
Check SIP User ID for Incoming Invite	When turned on, it will validate the To URI for incoming INVITE before accepting it. If it differs from locally configured URI, it should reject the call with 404 Not Found message.
Preferred Vocoder	<p>The BT-200 supports up to 5 different Vocoder types including G.711(a/μ) (also known as PCMU/PCMA), GSM, G.723.1, G.729A/B.</p> <p>Configure Vcoders in a preference list that is included with the same preference order in SDP message. Enter the first Vocoder in this list by choosing the appropriate option in "Choice 1". Similarly, enter the last Vocoder in this list by choosing the appropriate option in "Choice 8".</p>
SRTP Mode	<p>Disabled: BT-200 will use RTP for both inbound and outbound calls. (Default setting)</p> <p><u>Enabled but not forced</u>: BT-200 provides crypto suites to others SIP phones. If they support SRTP, then use SRTP, otherwise, use RTP.</p> <p><u>Enabled and forced</u>: All calls using this account must use SRTP. If the other party doesn't support SRTP, BT200 will decline the call. If we invite others, and receive a "200 Ok" message without crypto suite, we will send a "BYE" message followed by the "ACK" message to terminate the call. If someone invites us without crypto suite, we will response a "488 Not Acceptable Here" message.</p>
Special Feature	Default is Standard . Choose the selection to meet some special requirements from SoftSwitch vendors.

SAVING THE CONFIGURATION CHANGES

Once a change is made, click the "Update" button in the Configuration Menu. The IP phone will display a confirmation page. Reboot or power cycle the BT-200 after all changes to ensure the changes take effect.

FIGURE 5: SAVE CONFIGURATION SCREENSHOT



REBOOTING THE PHONE FROM REMOTE

The administrator can remotely reboot the phone by clicking the “Reboot” button at the bottom of the configuration menu. Once done, the web server will display a confirmation page to indicate that rebooting is underway. When finished, login again after waiting for about 30 seconds.

CONFIGURATION THROUGH CENTRAL PROVISIONING SERVER

The BT-200 can be automatically configured from a central provisioning system. The configuration files can be downloaded via TFTP or HTTP from the central server. A service provider or an enterprise with large deployment of BT-200 can easily manage the configuration and service provisioning of individual devices remotely from a central server.

When the BT-200 boots up, it sends a TFTP or HTTP request to download configuration files. There are two configuration files, one is “cfg.txt” and the other is “cfg000b82xxxxxx”, where “000b82xxxxxx” is the MAC address of the BT-200.

Grandstream’s GAPS provisioning server is the default server for all Grandstream devices. Based on the unique MAC address, Grandstream’s default GAPS server provisions the devices with re-direction settings to point to a customer’s TFTP or HTTP server for further provisioning. This could be simple re-direction or with special provisioning settings.

GAPSLITE - Grandstream Automated Provisioning System – supports the automated configuration of the HT-496. It is a licensed-based software. GAPSLITE uses enhanced (NAT friendly) TFTP or HTTP and other communication protocols to communicate with each individual BT-200 for firmware upgrade, remote reboot, etc. The GAPSLITE software package also has a configuration tool to generate device configuration files.

The configuration tool and configuration templates is available on the Grandstream website: <http://www.grandstream.com/y-configurationtool.htm>. Please refer to GAPSLITE product documentation or contact sales directly @ sales@grandstream.com.

FIRMWARE UPGRADE

UPGRADE THROUGH HTTP

To upgrade firmware via HTTP, the field “*Firmware Upgrade and Provisioning: Upgrade Via*” needs to be set to HTTP. The “*Firmware Server Path*” should be set to where the firmware files are located.

For example, the user can use the following URL in the Firmware Server Path: firmware.mycompany.com:6688/Grandstream/1.0.0.29 where firmware.mycompany.com is the FQDN of the HTTP server. It can also be in IP address format. “:6688” is the TCP port the HTTP server listening to, default http server listens to port 80. “/Grandstream/1.0.0.29” is the RELATIVE directory to the root dir on HTTP web server.

NOTE: If the “Auto Upgrade” field in the *Advanced Configuration page* is set to “No”, HTTP upgrade will be performed only once during boot-up. If it is set to “Yes”, the BT-200 will check the HTTP server in the interval defined in “Check for new firmware every” field.

UPGRADE THROUGH TFTP

To upgrade firmware, the BT-200 can be configured with a TFTP server where the new code image is located. It is recommended to set the TFTP server address in either a public IP address or on the same LAN with the BT-200. There are two ways to use the TFTP server to upgrade the BT-200 firmware

1. the keypad configuration menu
2. the BT-200's Web configuration interface.

Keypad Configuration Prompt Menu

To configure the TFTP server **via menu prompt**, please refer to [Table 7: Configuration Menu Definitions, Menu Option #06](#). After the TFTP IP address is set up, re-boot or power cycle the BT-200 to run the new firmware.

Web Configuration Interface

To configure the TFTP server via the Web configuration interface, open up your browser to point at the IP address of the BT-200.

1. Input the [admin password](#) to enter the configuration screen.
2. Enter the TFTP server address in the designated field located at the bottom of the configuration screen.
3. Update the change by clicking the "Update" button.
4. Reboot or power cycle the phone, the firmware files will be fetched upon booting up.

The system will check the TFTP server only during the initial power up. If the configured TFTP server is found and a new code image is available, the BT-200 will retrieve the new image files by downloading them into the BT-200's SRAM. During this stage, the BT-200's LEDs will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will then be saved into the Flash. If TFTP fails for any reason (e.g., TFTP server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the BT-200 will stop the TFTP process and boot using the existing code image in the flash.

The TFTP process may take up to 2 minutes over the Internet, or 20+ seconds if it is performed on a LAN. Grandstream recommends conducting firmware upgrades in a controlled LAN environment if possible.

NOTE for TFTP/HTTP Firmware Upgrade:

1. When the BT-200 boots up, it will send TFTP or HTTP request to download configuration files. There are two configuration files, 1) "cfg.txt" and 2) "cfg000b82xxxxxx", where "000b82xxxxxx" is the MAC address of the BT-200.
2. These two files are for initial automatic provisioning purposes only. If using a normal TFTP or HTTP firmware upgrade, the following error messages in a TFTP or HTTP server log can be ignored.
 - TFTP Error from [IP ADDRESS] requesting cfg000b82023dd4 : File does not exist
 - TFTP Error from [IP ADDRESS] requesting cfg.txt : File does not exist

No Local TFTP Server

For those who do not have a local TFTP server, Grandstream provides a NAT-friendly TFTP server on the public Internet for firmware upgrade. Please check the Support/Download section of our website to obtain this TFTP server IP address: <http://www.grandstream.com/y-firmware.htm>. Each product has a unique TFTP server address.

Alternatively, download and install a free TFTP or HTTP server to the LAN to perform firmware upgrades. A free Windows version TFTP server is available:
<http://support.solarwinds.net/updates/New-customerFree.cfm>.

RESTORE FACTORY DEFAULT SETTING

WARNING: Restoring the Factory Default Setting will **delete** all configuration information of the phone. Please **backup** or **print** all the settings before you restoring factory default settings. Grandstream is not responsible for restoring lost parameters and cannot connect to your VoIP service provider.

Directions for Restoration:

Disconnect network cable and power cycle the unit before resetting factory default settings.

1. Find the MAC address of the device. It is a 12 digit HEX number located on the bottom of the unit.
2. Press the MENU button for Key Pad Menu options.
3. Press the Up button to "reset".
4. Enter the MAC address with alphabets and numbers.
5. Press the MENU button again

The BT-200 will reboot automatically and restore to factory default setting. The reset is finished when the LCD stops flashing.

GLOSSARY OF TERMS

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

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

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

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

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

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

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

DECIMATE To discard portions of a signal in order to reduce the amount of information to be encoded or compressed. Lossy compression algorithms ordinarily decimate while 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 **Domain Name System** (or **Service** or **Server**), 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 .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.