



BudgeTone-200/201 SIP Phone

User Manual



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GUI INTERFACE EXAMPLES

BT USER MANUAL

http://www.grandstream.com/support/bt_series/bt20x/documents/bt20x_gui.zip

1. SCREENSHOT OF CONFIGURATION LOGIN PAGE
2. SCREENSHOT OF STATUS PAGE
3. SCREENSHOT OF BASIC SETTINGS CONFIGURATION PAGE
4. SCREENSHOT OF ADVANCED USER CONFIGURATION PAGE
5. SCREENSHOT OF SIP ACCOUNT CONFIGURATION
6. SCREENSHOT OF SAVED CONFIGURATION CHANGES
7. SCREENSHOT OF REBOOT PAGE

Welcome

Thank you for purchasing Grandstream BudgeTone-200/201 IP Phone. You made an excellent choice and we hope you will enjoy all its capabilities.

Grandstream's BT SIP IP phone is the innovative IP telephone that offers a rich set of functionality and superb sound quality. They are fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market.

This document is subject to changes without notice. The latest electronic version of this user manual is available for download from the following location:

http://www.grandstream.com/support/bt_series/bt20x/documents/bt20x_usermanual_english.pdf

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.*

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

- This document is contains links to Grandstream GUI Interfaces. Please download these examples from http://www.grandstream.com/support/bt_series/bt20x/documents/bt20x_gui.zip 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/support/bt_series/bt20x/documents/bt20x_usermanual_english.pdf
- *Reproduction or transmittal of the entire or any part, in any form or by any means, electronic or print, for any purpose without the express written permission of Grandstream Networks, Inc. is not permitted.*

Installation

WHAT IS INCLUDED IN THE PACKAGE

Table 1: Equipment Packaging

	BT-200	BT-201
Main Case	Yes	Yes
Handset	Yes	Yes
Phone Cord	Yes	Yes
Power Adaptor	Yes	Yes
Ethernet Cable	Yes	Yes

CONNECTING YOUR PHONE

Following is a backside picture of BT, each connection port is labeled with the name in the following table:

Figure 1: BT Back Panel



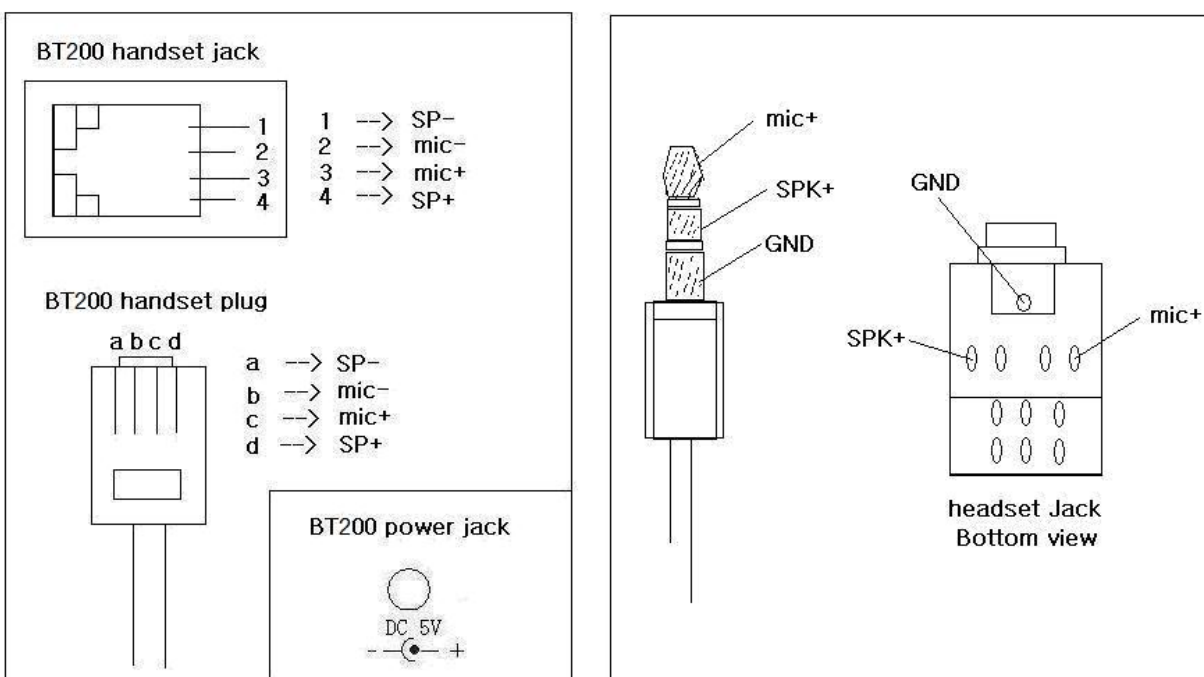
Table 2: BT Connectors

LAN	10/100 Switch LAN port for connecting to Ethernet.
PC	10/100Mbps RJ-45 ports for PC (downlink) connection.
Power Jack	5V DC power port; UL Certified
HEADSET	2.5mm Headset port

1. Connect attached telephone cord to the handset and to the phone device.
2. Put the handset on the base.
3. Connect attached LAN cable to the LAN port of the phone.
4. Connect attached power supply first to the 5V power port of the phone.
5. Connect attached power supply to the wall outlet.

When power on, the phone starts the initialization procedure.

Figure 2: BT Internal Headset Wiring Schema



SAFETY COMPLIANCES

The BudgeTone-200/201 phone is compliant with various safety standards including FCC/CE. Its power adaptor is compliant with UL standard. The phone should only be operated with the universal power adaptor provided with the package. Damages to the phone caused by using other unsupported power adaptors are not covered by the manufacturer's warranty.

WARRANTY

If you purchased your BT 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.

Product Overview

The following photo illustrates the appearance of a BudgeTone IP phone.

Figure 3: BT Front View



Figure 4: BT Side View



Table 3: BT Comparison Guide

Features	BT-200	BT-201
Network Interfaces	PC Port, LAN Port	LAN Port

Table 4: BT Key Features in a Glance

Features	Benefits
Open Standards Compatible	Support SIP 2.0, TCP/UDP/IP, PPPoE, RTP/RTCP/SRTP, HTTP, ARP/RARP, ICMP, DNS, DHCP, NTP, TFTP.
Superb Audio Quality	Advanced Digital Signal Processing (DSP), Silence suppression, VAD, CNG, AGC.

Network Interfaces	Dual 10/100Mbps Ethernet ports;
Feature Rich	Traditional voice features including caller ID, call waiting, hold, transfer, forward, block, autodial, off-hook dial, and click to dial.
Advanced Features	Dedicated buttons for hold, send, speakerphone, headset, transfer, 3-way-conference, mute, message.
Advanced Functionality	Custom down-loadable ring-tones, SRTP, SIP over TLS (pending), multi-language support and adjustable positioning angles, wall mountable, AES encryption.

Table 5: Hardware Specifications

LAN Interface (Ethernet ports)	BT200	Two (2) 10/100 Mbps Full/Half Duplex Ethernet Switch with LAN and PC port with auto detection
	BT201	One (1) 10/100 Mbps Full/Half Duplex Ethernet LAN Port
Headset Jack	2.5mm Headset port	
LED	1 LED in RED color	
Phone Case	25-button keypad 12-digit caller ID LCD	
Universal Switching	Input: 100-240VAC 50-60 Hz	
Power Adaptor	Output: +5VDC, 1200mA, UL certified	
Dimension	220mm(l) x 180mm(w) x 65mm(h)	
Weight	0.9kg (21bs)	
Temperature	32 –104 °F / 5 – 45 °C	
Humidity	10% – 90% (non-condensing)	
Compliance	FCC / CE / C-Tick	

Table 6: BT Technical Specifications

Protocol Support	Support SIP 2.0, TCP/UDP/IP, PPPoE, RTP/RTCP/SRTP, HTTP, ARP/RARP, ICMP, DNS, DHCP, NTP, TFTP.
Device Management	NAT-friendly remote software upgrade (via TFTP/HTTP) for deployed devices including behind firewall/NAT Auto/manual provisioning system, GUI Interface Support Layer 2 (802.1Q, VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
Audio Features	Full-duplex hands-free speakerphone, headset enabled Advanced Digital Signal Processing (DSP) Dynamic negotiation of codec and voice payload length Support for PCMU, PCMA, GSM, G.723.1, G.729A/B, G.726-32, G.722 (wide band), iLBC codecs. In-band and out-of-band DTMF (in audio, RFC2833, SIP INFO) Silence Suppression, VAD (voice activity detection), CNG (comfort noise generation), ANG (automatic gain control) Acoustic Echo Cancellation (AEC) with Acoustic Gain Control (AGC) for speakerphone mode, Support side tone

Telephony Features	<p>Adaptive jitter buffer control (patent-pending) and packet delay & loss concealment</p> <p>Voice mail indicator, downloadable custom ring-tones, call hold, call transfer (attended/blind), call forward, call waiting, caller ID, mute, redial, call log, caller ID display or block and volume control</p> <p>3-way-conference, off-hook auto dial, auto answer, early dial</p>
Network and Provisioning	<p>Via keypad/LCD, Web browser, or secure (AES encrypted) central configuration file, manual or dynamic host configuration protocol (DHCP) network setup</p> <p>Support NAT traversal using IETF STUN and Symmetric RTP</p> <p>Support for IEEE 802.1p/Q tagging (VLAN), Layer 3 TOS</p>
Firmware Upgrades	<p>Support firmware upgrade via TFTP or HTTP,</p> <p>Support for Authenticating configuration file before accepting changes</p> <p>User specific URL for configuration file and firmware files</p>
Advanced Server Features	<p>Message waiting indication, support DNS SRV Look up and SIP Server Fail Over</p>
Security	<p>DIGEST authentication and encryption using MD5 and MD5-sess, SRTP, SIP over TLS (pending)</p>

Using the BT SIP Phone

GETTING FAMILIAR WITH THE LCD





BT phone has a numeric LCD of 64mmx24mm size with backlight. This model has a small red LED status reminder. Here is the display when all segments illuminate:






Figure 5: BT LCD



When the phone is in the normal idle state, the backlight is off. Whenever an event (call) occurs, the backlight will turn on automatically to bring the user's attention. In addition, if Voice Mail configured and there is a VM waiting, the red LED will be blinking to remind user there is a Voice Mail in the Voice Mail server.

Table 7: LCD Icons

Icon	LCD Icon Definitions
	Connectivity Status / SIP Proxy/Server Icon: OFF – IP address of Sip server is not found ON – IP address of Sip server are located Blinking –Ethernet link failure or the phone is not registered properly
	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 OFF when the speakerphone is off ON when the speakerphone is on
	Handset and Speakerphone/Headset Volume Icons: 0-7 scales to adjust handset / speakerphone volume

	<p>Real-time Clock: Synchronized to Internet time server Time zone configurable via web browser AM/PM indicator</p>
	<p>Call Logs: 01-10 for CALLED history (dialed number) 01-10 for CALLERS history (Incoming caller ID)</p>
	<p>Time Icon: AM for the morning PM for the afternoon</p>
	<p>IP Address Separator Icons</p>
	<p>Numerical Numbers and Characters: 0 - 9 * = L # = J A, b, C, c, d, E, F, G, g, H, h, I, L, n, O, o, P, q, r, S, t, U, u, Y</p>

GETTING FAMILIAR WITH KEYPAD



Table 8: BT Keypad Buttons

Key Button	Key Button Definitions
0 - 9, *, #	Digit, star and pound keys are usually used to make phone calls
↓	1) Reduce handset, speakerphone/headset volume after off hook the phone via handset or speaker 2) Reduce ring tone volume when phone in IDLE and off hook to confirm the changed ring tone volume 3) Next menu item browsing when phone is in IDLE mode after MENU key pressed, off hook to interrupt and exit
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
MESSAGE	Enter to retrieve voice mails from Voice Mail Portal or Server
HOLD	Temporarily hold the active call
TRANSFER	Transfer the active call to another party
CONFERENCE	Establish 3-way conferencing call
FLASH	Flash event to switch between two lines
MUTE/DEL	Mute an active call; or Delete a key entry Also used to 'REJECT' incoming call.
SEND/(RE)DIAL	Dial a new number inputted or Redial the number last dialed. After entering the phone number, pressing this key would force a call to go out immediately before timeout
SPEAKERPHONE	Enter hands-free mode
↑	1) Increase handset, speakerphone/headset volume after off hook the phone via handset or speaker 2) Increase ring tone volume when phone in IDLE and off hook to confirm the changed ring tone volume 3) Next menu item browsing when phone is in IDLE mode after MENU key pressed, off hook to interrupt and exit

MAKING PHONE CALLS

Handset, Speakerphone and Headset Mode

The regular Handset mode can be switched with either the Speaker mode (Hand free) or the Headset mode, however, whenever the Headset is plugged in, Speaker mode will be switched to the Headset mode automatically.

To Switch between Handset and Speaker/Headset, simply press the Hook Flash in the Handset cradle or the Speaker button.

Make Calls using Numbers

There are FIVE ways to make phone calls:

- Pick up handset or press SPEAKERPHONE button, and then enter the phone numbers
- Press the SEND button directly to redial the number last called.
- Once pressed, the last dialed number will be displayed on the LCD as the corresponding DTMF tones are played out and an outgoing call is sent.
- Browse the CALLED/CALLER history and press the SEND/REDIAL button.
- Pick up the handset or press the speakerphone button, then press the “CALLED/CALLERS” button to browse thru the last 10 numbers dialed out. Once the desired number is identified and displayed on the LCD screen, press the SEND button and a new call to that displayed number will be sent out immediately.

Examples:

- To dial another extension on the same proxy, such as 1008, simply pick up handset or press speakerphone, dial 1008 and then press the “SEND” button.
- 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 “SEND” button.

Make Calls using IP Address

Direct IP calling allows two parties, that is, a BT 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 BT phone and other VoIP Device(i.e., another IP Phone or BT SIP phone or other VoIP unit) have public IP addresses, or
- Both BT phone and other VoIP Device are on the same LAN using private or public IP addresses, or
- Both BT phone and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

This model has the ability to dial an IP address under the same LAN segment by simply pressing the last octet in the IP address.

In the Advanced Settings page there is an option "Use Quick IP-call mode", by default it is set to No. 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.

#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 #

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, Direct IP-IP call will still work. However, if you are using STUN, Direct IP-IP call will also use STUN.

OR

To make a direct IP to IP call, first off hook, then press "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_key 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 would be: **MENU_key 192168001020*45062** followed by pressing the "SEND" key wait for seconds in the No Key Entry Timeout.

Quick IP Call Mode

The BT also supports Quick IP call mode. This enables the phone to make direct IP-calls, using only the last few digits (last octet) of the target phone's IP-number.

This is possible only if both phones are in under the same LAN/VPN. This simulates a PBX function using the CMSA/CD without a SIP server. Controlled static IP usage is recommended.

Setting up the phone to make Quick IP calls

To enable Quick IP calls, the phone has to be setup first. This is done through the web-setup function. 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 OK).

For example:

192.168.0.2 calling 192.168.0.3 -- dial #3 follow by SEND or #

192.168.0.2 calling 192.168.0.23 -- dial #23 follow by SEND or #

192.168.0.2 calling 192.168.0.123 -- dial #123 follow by SEND or #

192.168.0.2: dial #3 and #03 and #003 results in the same call -- call 192.168.0.3

NOTE: If you have a SIP Server configured, a Direct IP-IP 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

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 SPEAKERPHONE button to answer in speakerphone or headset mode

Call Hold

While in conversation, pressing the “Hold” button will put the remote end on hold. Pressing the “Hold” button again will release the previously Hold state and resume the bi-directional media.

Call Waiting and Call Flashing

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 then can press FLASH button to put the current call party on hold automatically and switch to the other call. Pressing flash button toggles between two active calls.

Call Transfer

Two transfer operations are supported.

BLIND TRANSFER

User can transfer an active call to a third party without announcement.

User presses the “TRANSFER” button and if the other voice channel is available (i.e., there is no other active conversation besides the current one), user will hear a dial tone. User can then dial the third party’s phone number followed by pressing SEND button.

NOTE:

- “Enable Call Feature” has to be configured to “Yes” in web configuration page in order to make the features to work.

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) follows by a dial tone. This indicates the transfer has been successful. At this point, the user can either hang up or make another call.
- A quick busy tone followed by a restored call (On supported platforms only). This means the transfer has failed due to the failed response sent from server and the phone 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 the phone has failed to receive the final response and decide to time out. Be advised that this does not indicate the transfer has been successful, nor does it indicate the transfer has failed.

ATTENDED TRANSFER

User can transfer an active call to a third party with announcement.

User presses the “FLASH” button and hears a dial tone, then dial the third party’s phone number followed by pressing “SEND” button. If the call is answered, press “TRANSFER” to complete the

transfer operation and hand up, if the call is not answered, pressing “FLASH” button to resume the original call.

NOTE:

- *When Attended Transfer failed, if A hangs up, the BudgeTone phone 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.*

Conference Call

BT phone supports 3-way conference.

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

1. A presses the “CONFERENCE” button to get a dial tone and put B on hold
2. A dials C’s number then “SEND” key to make the call
3. If C answers the call, then A presses “CONFERENCE” button 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:

- During the conference, if B or C drops the call, the remaining two parties can still talk. However, if A the conference initiator hangs up, all calls will be terminated.

Checking Message and Message Waiting Indication

When BudgeTone-200 is on-hook, pressing the MESSAGE button will trigger the phone to call the VM Server (VMS) configured for the Account.

The MWI (Message Waiting Indicator) LED will flash in red color in three quarters of a second when voicemail server sends message waiting information to BudgeTone-200/201.

Mute and Delete

When in conversation with an ACTIVE LINE, pressing “MUTE/DEL” will mute the conversation, that is, you can hear the other party but the other party cannot hear you. Pressing the button again will resume the conversation.

When dialing a number, press “MUTE/DEL” will delete the last entered digit.

When receiving incoming call, press “MUTE/DEL” will Reject the call and forward to voice mail.

CALL FEATURES

BT series phone supports a list of call features: Caller ID Block (or Anonymous Call), Disable/Enable Call Waiting, Call Forward on Busy, Delay, or Unconditional, etc.

Following table shows the call features of BudgeTone-200 series phone.

Table 9: BT Call Features

Key	Call Features
*30	Block Caller ID (for all subsequent calls)
*31	Send Caller ID (for all subsequent calls)
*67	Block Caller ID (per call)
*82	Send Caller ID (per call)
*50	Disable Call Waiting (for all subsequent calls)
*51	Enable Call Waiting (for all subsequent calls)
*70	Disable Call Waiting (per Call)
*71	Enable Call Waiting (per Call)
*72	Unconditional Call Forward To use this feature, dial “*72” and get the dial tone. Dial the forward number and “#” for a dial tone, then hang up.
*73	Cancel Unconditional Call Forward To cancel “Unconditional Call Forward”, dial “*73” and get the dial tone, then hang up.
*90	Busy Call Forward To use this feature, dial “*90” and get the dial tone. Dial the forward number and “#” for a dial tone, then hang up.
*91	Cancel Busy Call Forward To cancel “Busy Call Forward”, dial “*91” and get the dial tone, then hang up.
*92	Delayed Call Forward To use this feature, dial “*92” and get the dial tone. Dial the forward number and “#” for a dial tone, then hang up.
*93	Cancel Delayed Call Forward To cancel this Forward, dial “*93” and get the dial tone, then hang up.

Configuration Guide

The BT can be configured in two ways. Firstly, using the Key Pad Configuration Menu on the phone; secondly, through embedded web-configuration menu.

CONFIGURATION VIA KEYPAD

The BT 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.

1. Be sure the phone is idle.
2. Press the MENU button to enter the keypad MENU to configure the phone.

Using the Keypad

1. To enter the MENU, press MENU button.
2. Navigate the menu by using the UP/DOWN
3. Press the MENU button to confirm a menu selection.
4. To delete an entry by pressing the MUTE/DEL button.

Press the MENU button to enter the key the Key Pad Menu. The menu options available are listed in table 8.

Table 10: Key Pad Configuration Menu

Manual Option	Call Features
1	Display “[1] dhcP On ” or “[1] dhcP oFF” for the current selection Press <i>MENU</i> key to enter edit mode
2	Display “[2] IP Addr ” Press <i>MENU</i> to display the current IP address Enter new IP address if DHCP is OFF
3	Display “[3] SubNet ” Press <i>MENU</i> to display the Subnet mask Enter new Subnet mask if DHCP is OFF
4	Display “[4] routEr ” Press <i>MENU</i> to display the Router/Gateway address Enter new Router/Gateway address if DHCP is OFF
5	Display “[5] dnS ” Press <i>MENU</i> to display the DNS address Enter new DNS address if DHCP is OFF
6	Display “[6] tFtP ” Press <i>MENU</i> to display the TFTP address Enter new TFTP server IP address
7	Display “[7] G-711u 2” Press <i>MENU</i> to select new codec Press ‘↓’ or ‘↑’ to browse a list of available codecs
8	Display “[8] codE rEL” Press <i>Menu</i> to display the code releases Press ‘↓’ or ‘↑’ to browse
9	Display “[9] Phy Addr” Press <i>MENU</i> to display the physical / MAC address.
10	Display “[10] ring 0”

	Press MENU to hear the selected ring tone, press '↓' or '↑' to select the stored ring tones.
11	Display "[11]trANSPort" Press MENU to select SIP Transport.
Factory Reset	Display "-- rESEt --". Please be very <i>careful</i> when using this function. Two options: factory reset or manual reboot. Factory Reset: 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 VIA WEB BROWSER

The BT embedded Web server responds to HTTP/HTTPS GET/POST requests. Embedded HTML pages allow a user to configure the IP phone through a Web browser such as Microsoft's IE or Mozilla Firefox.

Access the Web Configuration Menu

To access the phone's Web Configuration Menu

- Connect the computer to the same network as the phone¹
- Make sure the phone is turned on and shows its IP-address
- Start a Web-browser on your computer
- Enter the phone's IP-address in the address bar of the browser²
- Enter the administrator's password to access the Web Configuration Menu³

¹ The Web-enabled computer has to be connected to the same sub-network as the phone. This can easily be done by connecting the computer to the same hub or switch as the phone is connected to. In absence of a hub/switch (or free ports on the hub/switch), please connect the computer directly to the phone using the PC-port on the phone.

² If the phone is properly connected to a working Internet connection, the phone will display its IP address. This address has the format: xxx.xxx.xxx.xxx, where xxx stands for a number from 0-255. You will need this number to access the Web Configuration Menu. **e.g.** if the phone shows 192.168.0.60, please use "*http://192.168.0.60*" in the address bar your browser.

³ The default administrator password is "**admin**"; the default end-user password is "**123**".

NOTE: When changing any settings, always SUBMIT them by pressing the button on the bottom of the page. Reboot the phone to have the changes take effect. If, after having submitted some changes, more settings have to be changed, press the menu option needed.

Definitions

This section will describe the options in the Web configuration user interface. As mentioned, a user can log in as an administrator or end-user.

Functions available for the end-user are:

- **Status:** Displays the network status, account statuses, software version and MAC-address of the phone
- **Basic:** Basic preferences such as date and time settings, multi-purpose keys and LCD settings can be set here.

Additional functions available to administrators are:

- **Advanced Settings:** To set advanced network settings, codec settings.
- **Account:** To configure each of the SIP accounts.

Table 11: Device Configuration - Status

MAC Address	The device ID, in HEXADECIMAL format.
IP Address	This field shows IP address of BT
Product Model	This field contains the product model information.
Part Number	This field contains the product part number.
Software Version	<ul style="list-style-type: none"> • Program: This is the main software (firmware) release number, always used to identify the software (firmware) system of the phone. • Boot: Booting code version number
System Up Time	This field shows system up time since the last reboot.
System Time	This field shows the current time in the phone.
Registered	Indicates whether accounts are registered to the related SIP server(s). BT can support four unique SIP profiles.
PPPoE Link Up	Indicates whether the PPPoE connection is enabled (connected to a modem).

Table 12: Device Configuration – Basic Settings

End User Password	This contains the password to access the Web Configuration Menu. This field is <u>case sensitive</u> with a maximum length of 25 characters.
IP Address	<p>There BT operates in two modes:</p> <ol style="list-style-type: none"> 1. DHCP mode: all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The BT acquires its IP address from the first DHCP server it discovers on its LAN. The DHCP option is reserved for NAT router mode. To use the PPPoE feature, set the PPPoE account settings. The BT establishes a PPPoE session if any of the PPPoE fields are set. 2. Static IP mode: configure all of the following fields: IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary). These fields are set to zero by default.

Time Zone	This parameter controls the date/time display according to the specified time zone.
Daylight Savings Time	<p>This parameter controls time displayed in daylight savings time. If set to “Yes”, then the displayed time will be 1 hour ahead of normal time.</p> <p>The “Optional Rule” is configured to automatically adjust the Daylight Savings Time (DST) based on the rule set in this field.</p> <p>Rule Syntax:</p> <ul style="list-style-type: none"> • start-time; end-time; saving • Both start-time and end-time have the same syntax: month,day,weekday,hour,minute <ul style="list-style-type: none"> ○ 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>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” the iteration of the weekday (e.g.: 1st Sunday, 3rd Tuesday etc). If “weekday” is not zero and “day” is negative, then the daylight saving starts on the last “day” the iteration of the weekday (e.g.: last Sunday, 3rd last Tuesday etc).</p> <p>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.</p> <p>The default value is set for US, the “Automatic Daylight Saving Time Rule” shall be set to “3,2,7,2,0;11,1,7,2,0;60”</p> <p><u>Examples</u> US/Canada where daylight saving time is applicable: 03,02,7,02,00;11,1,7,02,00;60 This means the daylight saving time starts from the second Sunday of March at 2AM and ends the first Sunday of November at 2AM. The saving is 60 minutes.</p>
Date Display Format	<p>Choose one of the following formats:</p> <ul style="list-style-type: none"> • Year-Month-Day • Month-Day-Year • Day-Month-Year
Display Account Name instead of Date	Choose to display Account Name or date on LCD.
Mute Speaker Ringer	Default is No. This field lets user to choose whether to ring the phone Speaker when headset is connected.

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.

Table 13: Advanced Settings

Admin Password	Administrator password. Only the administrator can access the “Advanced Settings” and “Account Settings” page. Password field is purposely blank for security reasons after clicking update and saved. The maximum password length is 25 characters.
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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 controls the silence suppression/VAD feature of the audio codec G.723 and G.729. If set to “Yes”, when silence is detected, a 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 IP phone 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>Please be careful when editing these parameters. Adjusting these parameters will also change the dynamic jitter buffer. The BT has a patent dynamic jitter buffer handling algorithm. The jitter buffer range is 20 ~ 200 ms.</p> <p>Grandstream recommends using the default settings provided. 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/faqscodec.html for more technical detail.</p>
Layer 3 QoS	This field defines the layer 3 QoS parameter. It is 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.
Data VLAN Tag	Default is 0 for [0-2]. Enabling the Data VLAN filtering will help reduce the load on the phone, but it isn’t necessary in most environments. This is primarily for VLAN filtering where tagged traffic will be forwarded to the DSP.
No Key Entry Timeout	Default is 4 seconds.
Use # as Dial 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 “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 used to 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 BTs are behind the same NAT. Default is No.
Keep-alive interval	This parameter specifies how often the BT 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. STUN resolution result will display in the STATUS page of the Web UI.
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.
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 will attempt to retrieve a new configuration file or new code image from the specified TFTP server at boot time. It will make up to 3 attempts before timeout and then it will 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: Grandstream strongly recommends that the user upgrade firmware locally in a LAN environment if using TFTP to upgrade. Please do NOT interrupt the TFTP upgrade process (especially the power supply) as this will damage the device.</p>
Via HTTP Server	<p>The HTTP server URL used for firmware upgrade and configuration via HTTP. For example: http://provisioning.mycompany.com:6688/Grandstream/1.2.2.19 Here “:6688” is the specific TCP port that the HTTP server is using; omit if using default port 80.</p> <p>Note: If Auto Upgrade is set to No, BT will only perform HTTP download once at boot up.</p>
Config Server Path	IP address or domain name of firmware server.
Firmware File Prefix/Postfix	Default is blank. If configured, BT will request the firmware file with the prefix/postfix. This setting is useful for ITSPs. End user should keep it blank.
Config File Prefix/Postfix	Default is blank. End user should keep it blank.
Allow DHCP Option 43 and Option 66 to override server	Default is Yes. This allows device gets provisioned automatically.
Authenticate Conf File	Default is “No”. If set to “Yes”, configuration file would be authenticated before acceptance. End user should use default setting.

Automatic Upgrade	<p>This function is used by ITSP. End user should NOT touch these parameters.</p> <p>Default is No. Choose “Yes” to enable automatic HTTP upgrade and provisioning. In “Check for upgrade every” field, enter the number of minutes to check the HTTP server for firmware upgrade or configuration changes. When set to “No”, the phone will only perform HTTP upgrade and configuration check once at boot up.</p>
Offhook Auto Dial	To configure a User ID/extension to dial automatically when the phone is taken offhook.
DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833. Default is 101 .
Onhook Threshold	Time period when the cradle is pressed (Hook Flash) to simulate FLASH. To prevent unwanted activation of the Flash/Hold and automatic phone ring-back, adjust this time value.
Syslog Server	The IP address or URL of System log server. This feature is especially useful for ITSPs.
Syslog Level	<p>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:</p> <ul style="list-style-type: none"> • product model/version on boot up (INFO level) • NAT related info (INFO level) • sent or received SIP message (DEBUG level) • SIP message summary (INFO level) • inbound and outbound calls (INFO level) • registration status change (INFO level) • negotiated codec (INFO level) • Ethernet link up (INFO level) • SLIC chip exception (WARNING and ERROR levels) • memory exception (ERROR level) <p>The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: <i>GS_LOG: [device MAC address][error code] error message</i> <i>For 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	This parameter defines the URI or IP address of the NTP (Network Time Protocol) server. It is used to display the current date/time.
Distinctive Ring Tone	Caller ID must be configured. Select a Distinctive Ring Tone 1 through 3 for a particular Caller ID. The BT will ONLY use selected ring tones for particular Caller IDs. For all other calls, the BT will use System Ring Tone. When selected and no Caller ID is configured, the selected ring tone will be used for all incoming calls.
System Ring Tone	System ring tone. Default is North American standard. Adjust system ring tone frequencies and cadences based on local telecom standard.

Call Progress Tones	<p>Using these settings, users can configure ring or tone frequencies based on parameters from local telecom. By default, they are set to North American standard. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds.</p> <p>Syntax: f1=val,f2=val[,c=on1/off1[-on2/off2[-on3/off3]]]; (Frequencies are in Hz and cadence on and off are in 10ms) ON is the period of ringing (“On time” in ‘ms’) while OFF is the period of silence. In order to set a continuous ring, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern. Up to three cadences are supported.</p>
Disable Call Waiting	Default is No. If set to Yes, the call waiting will be disabled.
Disable Direct IP calls	Default is No. If set to Yes, direct IP calls will be disabled.
Use Quick IP Call Mode	<p>Dial an IP address under the same LAN/VPN segment by entering the last octet in the IP address.</p> <p>In the Advanced Settings page there is an option “Use Quick IP-call mode”. Default setting is No. When 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). See Quick IP Call Mode for details.</p>
Disable Conference	Default is No. If set to Yes, conference calls will be disabled
Lock keypad update	If set to “Yes”, the configuration changes via keypad are disabled.
Disable DND	Default is No. If set to ‘Yes’, DND key on keypad will be disabled
Disable Transfer	Default is No. If set to Yes, transfer calls will be disabled
Disable Multicast Filter	Default is No. If set to Yes, the phone will not filter out (discard) multicast packets.
Send Flash Event	Default is No. If set to yes, flash will be sent as DTMF event.
Headset TX gain(dB)	Transmission gain. Its a headset setting to control the voice intensity
Headset RX gain (dB)	Receive gain. It’s a headset setting to control the voice intensity.

Table 14: SIP Account Settings

Account Name	The name associated with each account - displayed on 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, Media Gateway, or Session Border Controller. Used for firewall or NAT penetration in different network environment. If the system detects symmetric NAT, STUN will not work. ONLY outbound proxy can provide solution for symmetric NAT.
SIP User ID	User account information provided by VoIP service provider (ITSP); either an actual phone number or formatted like one.
Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. It can be identical to or different from SIP User ID.
Authenticate Password	SIP service subscriber's account password for BT to register to (SIP) servers of ITSP.
Name	SIP service subscriber's name that is 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 phone 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 sending REGISTER messages to the proxy server. The default setting is "Yes".
Un-register 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 refreshes its registration with the specified registrar. The default interval is 60 minutes. The maximum interval is 65,535 minutes (about 45 days).
Local SIP Port	This parameter defines the local SIP port used to 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 Registration Failure Retry Wait Time	Retry registration if the process failed. Default is 20 seconds.
SIP T1 Timeout	RFC 3261 SIP T1 timer. Default is 1 second.
SIP T2 Interval	RFC 3261 SIP T2 timer. Default is 0.5 seconds.
SIP Transport	Choose SIP Transport between UDP and TCP. Default is UDP.
Use RFC3581 Symmetric Routing	This option allows a SIP client to request that the server send the response back to the source IP address and port from which the request originated.

NAT Traversal (STUN)	This parameter activates the NAT traversal mechanism. If activated (by choosing “Yes”) and a STUN server is also specified, the phone performs according to the STUN client specification. Using this mode, the embedded STUN client detects if and what type of NAT/Firewall configuration is used. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the phone will 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 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.
SUBSCRIBE for Registration Event	Default is No. This is mainly used for IMS purposes. When enabled, the terminals should store the Service-Route header values after successfully registered, and thereafter add a route header with the values stored in the Service-Route when initiating a request excluding REGISTER.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Voice Mail UserID	When configured, user can access messages by pressing “MSG” button. This ID is usually the VM portal access number.
Send DTMF	This parameter specifies the mechanism to transmit DTMF digit. There are 3 supported modes: 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.
Delayed Call Forward Wait Time	Time waited before the call is forward to a number or VM. Default is 20 seconds.
Enable Call Features	Default is Yes. If set to “Yes”, all star code call features will be supported locally.
Disable Call Log	User can choose to disable Call Log.
Session Expiration	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. 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.
Min-SE	Defines the minimum session expiration (in seconds). Default is 90 seconds .
Caller Request Timer	If set to “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 set to “Yes”, the phone will use session timer even if the remote party does not support this feature. If set to “No”, the session timer is enabled only when the remote party supports this feature. 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.
Enable 100rel	PRACK (Provisional Acknowledgment) method enables reliability to SIP provisional responses (1xx series). This is required to support PSTN inter-networking..
Account Ring Tone	There are 4 uniquely defined ring tones: <ul style="list-style-type: none"> • One (1) System Ring Tone: when selected, all calls will ring with system ring tone. • Three (3) Customer Ring Tones: when selected, incoming calls from designated account will play selected ring tone.
Ring Timeout	Defines how long ring will ring when receiving a call. Default is 60 seconds.
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.
Anonymous Method	Whether to use “< sip:anonymous@anonymous.invalid>” in the From Header or P-Asserted-Identity header.
Anonymous Call Rejection	Default is NO. If set to YES, anonymous call will be rejected
Auto Answer	Default is No. If set to “Yes”, BT will automatically switch on speaker to answer the incoming call. Set to Intercom/Paging mode, it will answer the call based on the SIP info header from the server.
Allow Auto Answer by Call-Info	If the Call-Info header contains answer-after=0, the call be answered automatically (so called paging mode).
Turn off speaker on remote disconnect	When BYE is received, the phone will turn off its speaker automatically.
Check SIP User ID for incoming INVITE	Check the SIP User ID in Request URI. If they don't match, the call will be rejected.
Refer-To Use Target Contact	Default is NO. If set to YES, then for Attended Transfer, the “Refer-To” header uses the transferred target's Contact header information.
Disable Multiple Media Attribute in SDP	Default is No.

Preferred Vocoder	BT supports up to 5 different Vocoder types including G.711(a/μ) (also known as PCMU/PCMA), GSM, G.723.1, G.729A/B. Configure Vocoders 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”.
SRTP Mode	Enable SRTP mode based on selection. Default is No.
Special Feature	Default is Standard. Choose the selection to meet special requirements from Soft Switch vendors.

SAVING THE CONFIGURATION CHANGES

After the user makes a change to the configuration, press the “Update” button in the Configuration Menu. The web browser will then display a message window to confirm saved changes.

Grandstream recommends reboot or power cycle the IP phone after saving changes.

REBOOTING THE PHONE REMOTELY

Press the “Reboot” button at the bottom of the configuration menu to reboot the phone remotely. The web browser will then display a message window to confirm that reboot is underway. Wait 30 seconds to log in again.

Software Upgrade & Customization

Software (or firmware) upgrades are completed via either TFTP or HTTP. The corresponding configuration settings are in the ADVANCED SETTINGS configuration page.

FIRMWARE UPGRADE THROUGH TFTP/HTTP

To upgrade via TFTP or HTTP, select TFTP or HTTP upgrade method. “Upgrade Server” needs to be set to a valid URL of a HTTP server. Server name can be in either FQDN or IP address format. Here are examples of some valid URLs.

- firmware.mycompany.com:6688/Grandstream/1.2.2.19
- 72.172.83.110

There are two ways to set up the Upgrade Server to upgrade firmware: via Key Pad Menu and Web Configuration Interface.

Key Pad Menu

Only TFTP would be used when configure the phone using key pad. To configure the Upgrade Server via Key Pad Menu options, select “[6]tFtP” from the Main Menu, then press MENU button to enter this option. Enter the 12-digit IP address. Then, press MENU button to save. Must reboot the phone to take effect.

Web Configuration Interface

To configure the Upgrade Server via the Web configuration interface, open the web browser. Enter the BT IP address. Enter the admin password to access the web configuration interface. In the ADVANCED SETTINGS page, enter the Upgrade Server’s IP address or FQDN in the “Firmware Server Path” field. Select TFTP or HTTP upgrade method. Update the change by clicking the “Update” button. “Reboot” or power cycle the phone to update the new firmware.

During this stage, the LCD will display the firmware file downloading process. If a firmware upgrade 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 phone will stop the upgrading process and re-boot using the existing firmware/software.

Firmware upgrades take around 60 seconds in a controlled LAN or 5-10 minutes over the Internet. Grandstream recommends completing firmware upgrades in a controlled LAN environment whenever possible.

No Local TFTP/HTTP Server

For users who do not have a local TFTP/HTTP server, Grandstream provides a HTTP server on the public Internet for users to download the latest firmware upgrade automatically. Please check the Support/Download section of our website to obtain this HTTP server IP address: <http://www.grandstream.com/firmware.html> .

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>.

Instructions for local TFTP Upgrade:

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

User can also choose to download the free HTTP server from <http://httpd.apache.org/> or use Microsoft IIS web server.

NOTE:

- When BT phone boots up, it will send TFTP or HTTP request to download configuration file "cfg000b82xxxxx", where "000b82xxxxx" is the MAC address of the BT phone. This file is for provisioning purpose. For normal TFTP or HTTP firmware upgrades, 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. Configuration File Download"*

CONFIGURATION FILE DOWNLOAD

The BT 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 the configuration file. It needs to be set to a valid URL, either in FQDN or IP address format.

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 4 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 configuration template of the firmware.

Once the BT boots up (or re-booted), it will request a configuration file named "cfgxxxxxxxxxxx", where "xxxxxxxxxxx" is the MAC address of the device, i.e., "cfg000b820102ab". The configuration file name should be in lower cases.

Managing Firmware and Configuration File Download

When "Automatic Upgrade" is set to "Yes", a Service Provider can use P193 (Auto Check Interval, in minutes, default and minimum is 60 minutes) to have the devices periodically check for upgrades at pre-scheduled time intervals. By defining different intervals in P193 for different devices, a Server Provider can manage and reduce the Firmware or Provisioning Server load at any given time.

Restore Factory Default Setting

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

INSTRUCTIONS FOR RESTORATION:

Step 1: Press the MENU button for Key Pad Menu options and press the Up button to see “reset”.

Step 2: Enter the MAC address printed on the bottom of the sticker. Please use the following mapping:

0-9: 0-9

A: 22 (press the “2” key twice, “A” will show on the LCD)

B: 222

C: 2222

D: 33 (press the “3” key twice, “D” will show on the LCD)

E: 333

F: 3333

Example: if the MAC address is 000**b**8200**e**395, it should be key in as “000**222**8200**333**395”.

NOTE: If there are digits like “22” in the MAC, you need to type “2” then press “->” right arrow key to move the cursor or wait for 4 seconds to continue to key in another “2”.

Step 3: Press the **MENU** button again. If the MAC address is correct, the phone will reboot. Wait for phone reboot and the LCD backlight finish flashing.