User Manual

BudgeTone-100 Series
IP Phone

Version 1.0.5.6

Grandstream Networks, Inc.

www.grandstream.com

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1 Welcome

Congratulations on becoming an owner of BudgeTone-100 IP telephone! You made an excellent choice and we hope you will enjoy all its capabilities.

Grandstream's award-wining BudgeTone-100 series of SIP phones are innovative IP telephones that offer a rich set of functionality and superb sound quality at ultra-affordable price. 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 can be downloaded from the following location:
    http://www.grandstream.com/user_manuals/budgetone100.pdf

Grandstream Networks, Inc.

2 Installation

BudgeTone-100 series IP phones are designed to look and feel like standard telephones. The following photo illustrates the appearance of a BudgeTone IP phone and the use of its key buttons.

Volume & Menu Browser Key

Incoming Callers Log

Speakerphone

SEND/(Re)Dial

LCD

Menu

Out-Call Log

Message

Hold

Transfer

Conference

Flash

MUTE/Delete

2.1 What is Included in the Package

The BudgeTone-100 phone package contains:

1) One BudgeTone-100 phone
2) One universal power adaptor
3) One Ethernet cable

2.2 Safety Compliances

The BudgeTone-100 phone is compliant with various safety standards including FCC/CE. Its power adaptor is compliant with UL standard. The phone should only operate with the universal power adaptor provided with the package. Damages to the phone caused by using other unsupported power adaptors would not be covered by the manufacturer's warranty.

3 Product Overview

Grandstream IP Phone is a next generation IP network telephone based on industry open standard SIP (Session Initiation Protocol). Built on innovative technology, Grandstream IP Phone features market leading superb sound quality and rich functionalities at mass-affordable price.

3.1 Key Features

- Support SIP 2.0, TCP/UDP/IP, PPPoE, RTP/RTCP, HTTP, ARP/RARP, ICMP, DNS, DHCP, NTP/SNTP, TFTP protocols
- Support NAT traversal using IETF STUN and Symmetric RTP
- Advanced Digital Signal Processing (DSP) technology to ensure superior hi-fidelity audio quality, interoperable with various 3rd party SIP end user device, Proxy/Registrar/Server and Gateway products
- Advanced and patent pending adaptive jitter buffer control, packet delay and loss concealment technology
- Support popular codecs including G711 (a-law and u-law), G722, G.723.1 (5.3K/6.3K), G.726 (40K/32K/24K/16K), G.728, G.729A/B and iLBC. Dynamic negotiation of codec and voice payload length
- Support standard voice features such as Caller ID Display or Block, Call Waiting, Call Hold, Call Transfer, Do-Not-Disturb, Flash, Call Forwarding, in-band and out-of-band DTMF(RFC2833), SIP INFO, Dial Plans, Off-Hook Auto Dial, Auto Answer, Early Dial and Speed Dial, etc.
- Full duplex hands-free speakerphone, redial, call log, volume control, voice mail with indicator, downloadable ring tone, etc.
- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168) and AGC (Automatic Gain Control)
- Support DIGEST authentication and encryption using MD5 and MD5-sess)
- Provide easy configuration through manual operation (phone keypad), Web interface or automated provisioning via central configuration through secure TFTP file for mass deployment
- Support for Layer 2 (802.1Q VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
- NAT friendly remote software upgrade capability (via TFTP) even from behind firewall/NATs
- Support DNS SRV Look up and SIP Server Fail Over (pending)

3.2 Hardware Specification

There are three models in the BudgeTone-100 family, namely:
     BudgeTone-101
     BudgeTone-102

The table below describes the difference among these models.

Model	BudgeTone-101	BudgeTone-102
LAN interface	1xRJ45 10Base-T	2xRJ45 10Base-T
Phone Case	25-button keypad 12-digit caller ID LCD	25-button keypad 12-digit caller ID LCD
Universal Switching Power Adaptor	Input: 100-240VAC Output: +5VDC, 1.2A, UL certified	Same as left
Dimension	18cm (W) 22cm (D) 6.5cm (H)	Same as left
Weight	2 lbs (0.9 kg)	Same as left
Operating Temperature	32 - 104°F 0 - 40°C	Same as left
Humidity	10% - 95% (non-condensing)	Same as left
Compliance	FCC/CE/C-Tick	Same as left

4 Basic Operations

4.1 Get Familiar with LCD

BudgeTone-100 phone has a numeric LCD of 64mmx24mm size and a backlight. Here is the display when all segments illuminate:

The LCD is equipped with a backlight. When the phone is configured properly and in the normal idle state, the backlight is off. Whenever an event occurs, the backlight turns on automatically and brings the user's attention.

Icon	LCD Icon Definitions

	Network Status Icon:
	  FLASH in the case of Ethernet link failure
	  OFF if IP address or SIP server is not found
	  ON if IP address and SIP server are located

	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

(clock icon)	Alarm Clock Status Icon: 　　OFF when the alarm clock is not set 　　ON when the alarm clock is set
(lock icon)	Lock Status Icon: 　　OFF when the lock is set 　　ON when the lock is not set
(010 icon)	Encryption Status Icon: 　　OFF when the voice encryption is off 　　ON when the voice encryption is on
(volume icon)	Handset and Speakerphone Volume Icons: 　　0-7 scales to adjust handset / speakerphone volume
18:88	Real-time Clock: 　　Synchronized to Internet time server 　　Time zone configurable via web browser
88	Call Logs: 　　01-99 for *CALLED* history (dialed number) 　　01-99 for *CALLERS* history (caller ID)
AM PM	Time Icon: 　　AM for the morning 　　PM for the afternoon
●	IP Address Separator Icons: 　　Three icons combine to indicate valid IP address
8	Numerical Numbers and Characters: 　　0 - 9 　　* = ∟ 　　# = ⌐ 　　A, b, C, c, d, E, F, G, g, H, h, I, J, (k), L, (m), n, O, o, P, q, r, S, t, U, u, (v, w, x), y, (z)

4.2 Get Familiar with Keypad

BudgeTone-100 phone has a 25-button keypad. Underneath the keypad, there are 4 LEDs in red color.

Key Button	Key Button Definitions
0 - 9, *, #	Digit, star and pound keys are usually used to make phone calls
↓	Next menu item when phone is in IDLE mode Or reduce handset/speakerphone volume
↑	Previous menu item when phone is in IDLE mode Or increase handset/speakerphone volume
MENU	Enter MENU mode when phone is in IDLE mode. It is also the ENTER key once entering MENU
CALLED	Display the phone numbers called
CALLERS	Display the caller IDs
MESSAGE	Enter to retrieve voice mails or other messages
HOLD	Temporarily hold the active call
TRANSFER	Transfer the active call to another number
CONFERENCE	Enter 3-way conferencing call
FLASH	Flash event to switch between two lines
MUTE/DEL	Mute an active call; or Delete a key entry, call log, voice mail and etc
SEND/(RE)DIAL	Dial a new number 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

4.3 Make Phone Calls

4.3.1 Make Calls Using Regular Phone or Extension Numbers

There are four ways to make phone calls:

1. Pick up handset or press SPEAKERPHONE button, and then enter the phone numbers
2. Press the SEND/REDIAL button directly to redial the number just 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.
3. Browse the CALLED history and press the SEND/REDIAL button.
   Pick up the handset or press the speakerphone button, and then press the "Called" button to browse through the last 10 numbers dialed out. Once the desired number is identified and displayed on the LCD screen, press the SEND/(Re)Dial button and a new call to that displayed number will be sent out immediately.
4. Browse the CALLERS history and press the SEND/REDIAL button
   Pick up the handset or press the speakerphone button, then press the "Callers" button to browse thru the last 10 caller IDs received. Once the desired number is identified and displayed on the LCD screen, press the SEND/(Re)Dial button and a new call to that displayed number will be sent out immediately.

Examples:

If the phone is configured with a user part of "1000" with a SIP proxy, then to dial the user extension "1008", simply just dial 1008 and then press the "SEND/(Re)Dial" button.

If the phone is configured with a regular PSTN number 16172223333 with a service provider's server, then to dial any other PSTN number (say, 16266667890) will be simply to dial that number (16266667890) as if you were calling from a regular analog phone, followed by pressing the "SEND/(Re)Dial" button.

If the "SEND/(Re)Dial" button is not pressed, the phone will wait for about 5 seconds before initiating the call.

4.3.2 Make Calls using IP Address

Direct IP calling allows 2 phones to talk to each other in an ad hoc fashion without a SIP proxy. VoIP calls can be made between two phones if

- both phones have public IP addresses, or
- both phones are on a same LAN using private or public IP addresses, or

- both phones can be connected through a router using public or private IP addresses.

To make a direct IP calling, first pick up the phone or turn on the speakerphone, then press "Menu" button followed by the 12-digit target IP address. If there is a user part, press "Menu" button and then the encoded user part, followed by *3 (encoding for "@") and then followed by the 12-digit target IP address. Destination ports can also be specified using *4 (encoding for ":") followed by the encoded port number.

The follow is a table of the encoding scheme for the most commonly used characters:

00	0
01	1
02	2
03	3
04	4
05	5
06	6
07	7
08	8
09	9
*0	. (dot character)
*1	_ (underscore character)
*2	- (hyphen character)
*3	@
*4	: (column character)
21	a
22	b
23	c
31	d
32	e
33	f
41	g
42	h
43	i
51	j
52	k
53	l

61	m
62	n
63	o
71	p
72	q
73	r
74	s
81	t
82	u
83	v
91	w
92	x
93	y
94	z

The rule of thumb to remember these encoding is: "a" is the first letter on button "1" so its encoding is "11". "b" is the 2nd letter on button "1" and its encoding is "12". "c" is the 3rd letter on button "1" and its encoding is "13". Likewise, "d" is the first letter on button "2" and its encoding is "21". This pattern and rule applies to all other alphabetic encoding.

Examples:
If the target IP address is 192.168.0.160, the dialing convention is
     Menu_key 192168000160
followed by pressing the "(Re)Dial" button or the "#" key is it is configured as a send key. In this case, the default destination port 5060 is used if no port is specified.

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 "(Re)Dial" button or the "#" key is it is configured as a send key.

If the target address is john@192.168.1.100:5062, then the dialing convention would be:
     Menu_key 51634262*3192168001100*45062
followed by pressing the "SEND/(Re)Dial" button or the "#" key is it is configured as a send key.

4.3.3   Answer an Incoming Call

There are two ways to answer an incoming call:
1. Pick up the handset to answer the call normally
2. Press the SPEAKERPHONE button to answer in speakerphone mode

4.3.4   Handset Mode and Speakerphone Mode

Handset mode and Speakerphone mode cannot be enabled at the same time. Pressing the hook-switch or Speakerphone button would toggle the phone between these two modes.

4.3.5   Call Hold

While in conversation, pressing the "Hold" button will put the remote end on hold. This is achieved by sending a Re-INVITE with "a=sendonly" attribute and a zero IP address for media in the SDP message. Pressing the "Hold" button again will release the previously Hold state and the bi-directional media will resume. This is triggered by sending another Re-INVITE with "a=sendrecv" attribute and a non-zero IP address for media in the SDP message.

4.3.6   Flash

This button is equivalent to putting an active call on Hold and then switching to the other voice channel. If the other channel has an active conversation going on, this is essentially a switching of the "talking" channel and the other channel will be activated. If the other channel is idle with no active conversation going on, then the user will hear a dial tone.
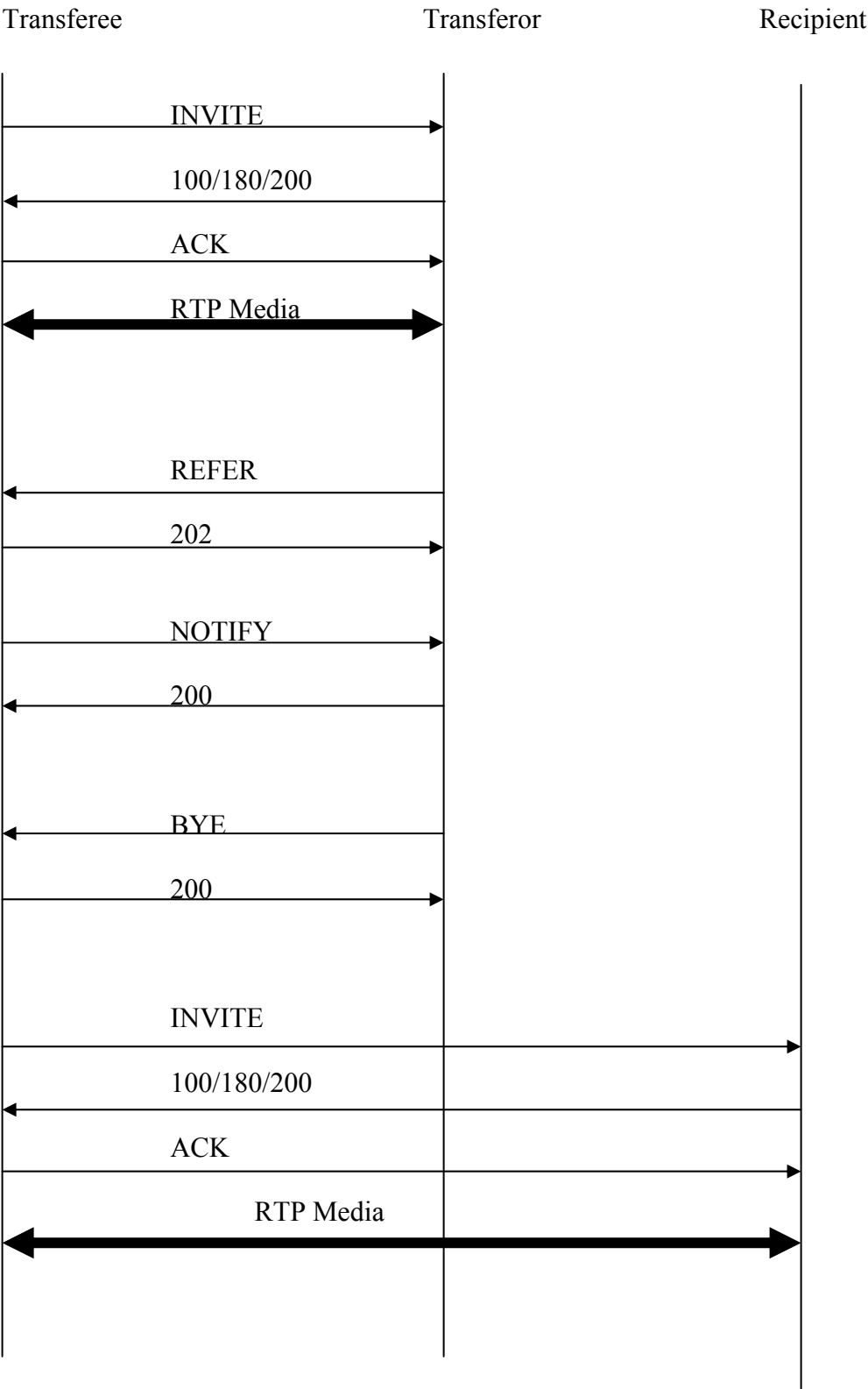
4.3.7   Call Transfer

The user can transfer an active call to a third phone by using the "Transfer" button. The sequence is like this:

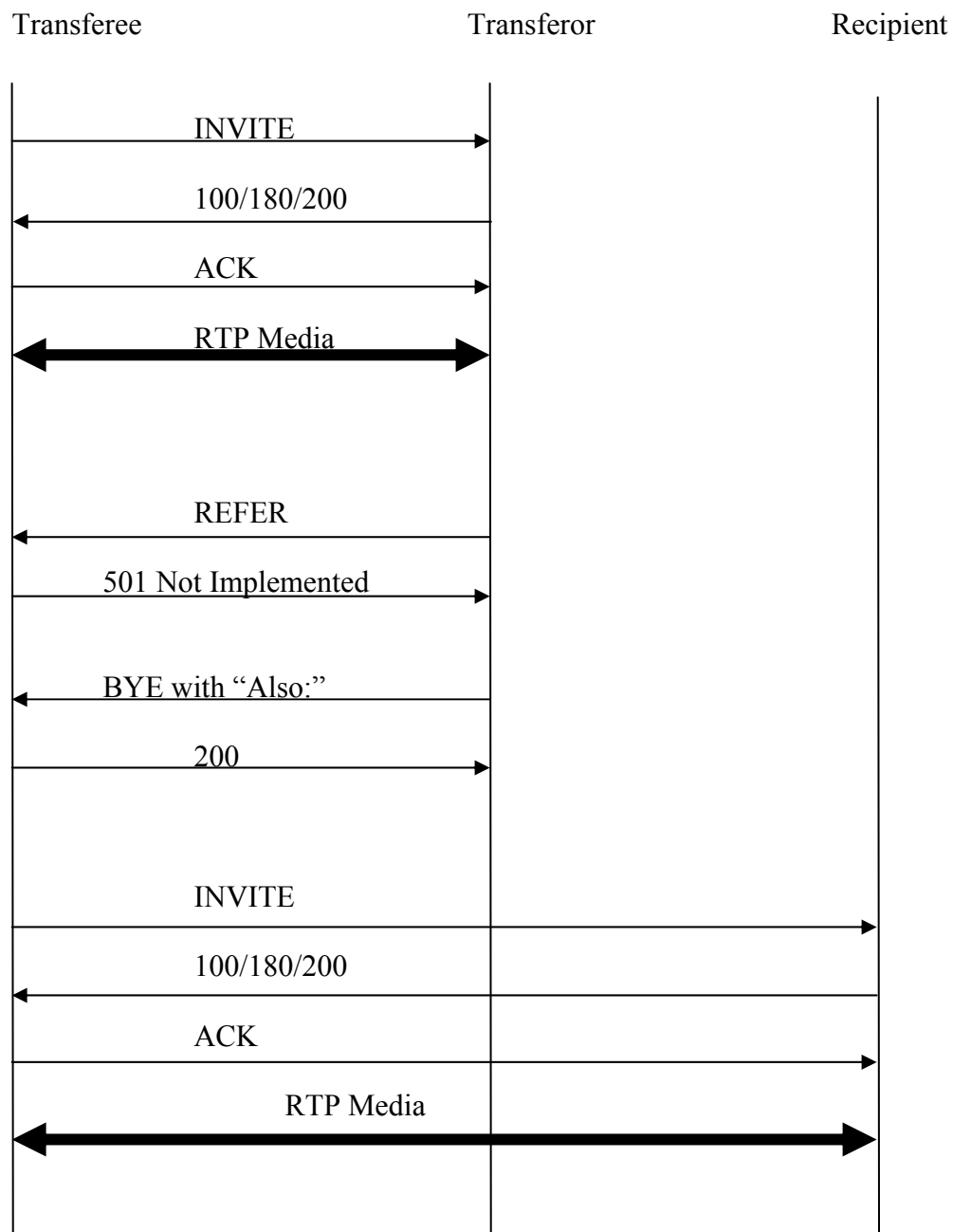> *The user presses the "Transfer" button and if the other voice channel is available (i.e., there is no other active conversation besides the current one), he/she will hear a dial tone. He/She can then dial the 3rd phone and then hangs up his own phone.*

Two kinds of blind call transfers are supported: using REFER and using BYE/Also. The SIP message flow based on SIP REGER method looks something like this:

Call Flow Diagram For Blind Call Transfer

Transferee   Transferor   Recipient

INVITE

100/180/200

ACK

RTP Media

REFER

202

NOTIFY

200

BYE

200

INVITE

100/180/200

ACK

RTP Media

The SIP message flow based on BYE/Also: method looks something like this:

Transferee   Transferor   Recipient

INVITE

100/180/200

ACK

RTP Media

REFER

501 Not Implemented

BYE with "Also:"

200

INVITE

100/180/200

ACK

RTP Media

4.4   Call Features

Following table shows the call features of BudgeTone series phone.

Dial Key	Call Features
*70	Do not disturb, it will give caller busy tone when called. This setting will be released when user hangs up the phone.
*72	Unconditional Call Forward. To use this feature, dial "*72" and get the dial tone. Then dial the forward number and 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. Then dial the forward number and hang up.
*91	Cancel Busy Call Forward To cancel "Unconditional 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. Then dial the forward number and hang up.
*93	Cancel Delayed Call Forward To cancel this Forward, dial "*93" and get the dial tone, then hang up
Flash/Hook	When in conversation, this action will switch to the new incoming call if user heard the call waiting sound. When in conversation and no incoming call heard, this action will switch to a new channel for a new call.

5 Configuration Guide

5.1 Configuration with Keypad

When the phone is on-hook, press the *MENU* button to enter MENU state. When the phone goes off-hook or a call comes in, the phone automatically exits the MENU state and prepare for the call. It also exits the MENU state if left idle for 20 seconds.

Here are the Menu options supported:

Menu Item	Menu Functions
1	Display "**[1] dhcP On** " or "**[1] dhcP oFF**" for the current selection Press *Menu* to enter edit mode Press '↓' or '↑' to toggle the selection Press *Menu* to save and exit Must recycle power 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 to take effect
3	Display "**[3] SubNet** " Press *Menu* to display the Subnet address Enter new Subnet address if DHCP is OFF Press '↓' or '↑' to exit Press *Menu* to (save and) exit Must recycle power 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 to take effect

Menu Item	Menu Functions
5	Display "`[5] dnS        `" Press *Menu* to display the DNS address Enter new DNS address if DHCP is OFF Press '↓' or '↑' to exit Press *Menu* to (save and) exit Must recycle power to take effect
6	Display "`[6] tFtP      `" Press *Menu* to display the TFTP address Enter new TFTP server address Press '↓' or '↑' to exit Press *Menu* to save and exit

7	Display "`[7] G-711u 2`"
	Press *Menu* to select new codec
	Press '↓' or '↑' to browse a list of available codecs
	  line 1 "　**- G-711A 2**"
	       2 "　**- G-722 2**"
	       3 "　**- G-723 1**"
	       4 "　**- G-726 2**"
	       5 "　**- G-728 8**"
	       6 "　**- G-729 2**"
	       7 "　**- iLBC 1**"
	Press 1 to 9 to indicate number of frames per TX packet
	Press *Menu* to save and exit. The codec selected takes effect immediately. No power cycle needed

8	Display "`[8] SIP SP-1`"
	Press *Menu* to display the SIP Server/Service Provider
	Press '↓' or '↑' to browse the valid SIP Server (1-9)
	Press *Menu* to save and exit
	      SIP Server(s) must be configured via Web browser
	      Only configured SIP server(s) are displayed
	Take effect immediately

Menu Item	Menu Functions

9	Display "`[9] codE rEL`"
	 Press *Menu* to display the code releases
	 Press '↓' or '↑' to browse
	   line 1 "`b 2004-06-12`" – date: *boot* code
	      2 " `1. 0. 0.18`" – version: *boot* code
	      3 "`P 2004-06-21`" – date: *phone* code
	      4 " `1. 0. 5. 3`" – version: *phone* code
	      5 "`c 2004-05-06`" – date: *vocoder*
	      6 " `1. 0. 0. 6`" – version: *vocoder*
	      7 "`h 2004-06-17`" – date: *web* server
	      8 " `1. 0. 0. 36`" – version: *web* server
	      9 "`1r 2004-05-12`" – date: 1st *ring* tone
	      10 " `1. 0. 0. 0`" – version: *ring* tone
	      11 "`2r 2004-05-12`" – date: 2nd *ring* tone
	      12 " `1. 0. 0. 0`" – version: *ring* tone
	      13 "`3r 0000-00-00`" – date: 3rd *ring* tone
	      14 " `0. 0. 0. 0`" – version: *ring* tone
	      (all zeroes means unavailable or unsupported)
	 Press *Menu* to exit, take effect immediately

10	Display "`[10] Phy Addr`" Press *Menu* to display the physical / MAC address Press *Menu*, '↓' or '↑' to exit
11	Display "**[11] ring 0**" Press *Menu* to hear the selected ring tone, press '↓' or '↑' to select the stored ring tones. Now only 3 are available, ring 0 (default), ring 1 and ring 2. ring 3 is unavailable or unsupported. Press *Menu* to select and exit, take effect immediately

	Display " **-- rESEt --**", please be very **CAREFUL** here
	            A> Key in the physical / MAC address on back of the phone, Press *Menu*, phone will be reset back to **FACTORY DEFAULT** setting, all your setting will be erased and gone
	            B> Press *Menu* without key in anything, phone will function the same as power cycle / reboot

Menu Item	Menu Functions
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)*, to designed one, then off and on the handset to set the ring tone volume • Press "**SPEAKERPHONE**" button, or off hook and pick up handset, press '↓' or '↑' to adjust the speakerphone or phone volume

5.2 Configuration with Web Browser

BudgeTone-100 series IP phone 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 a Web browser such as Microsoft's IE.

5.2.1 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. There are two ways to retrieve this IP address from the phone:

1) When the phone is in *on-hook* state, press *Menu* button and then the browsing arrow keys to check "**[2] IP Addr** "
2) When the phone is in *off-hook or speakerphone* state, simply press *Menu* button

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

Welcome to Grandstream IP Phone
Password [   ]
Login

The password is case sensitive and the factory default password is lower case '*admin*'.

5.2.2 Configuration Menu

After key in the correct password in the login screen, the embedded web server inside the IP phone will respond with the Configuration Menu screen. It will be displayed like following. You will know the MAC address of the phone, product model, software/firmware version and the available custom ring tones from the showed page.

Grandstream IP Phone Configuration	
MAC Address:	00.0B.82.00.A1.25
Product Model:	BT100
Software Version:	Program--1.0.5.3 Bootloader--1.0.0.18 HTML--1.0.0.36 VOC--1.0.0.6
Custom Ring Tone:	ring1--1.0.0.0 ring2--1.0.0.0 ring3--0.0.0.0 (all zeroes means unavailable or unsupported)
Admin Password:	[ ] (password to configure this IP phone)
IP Address:	⊙ dynamically assigned via DHCP (default) or PPPoE (will attempt PPPoE if DHCP fails and following is non-blank)
	PPPoE account ID: [ ]
	PPPoE password: [ ]
	Use this DNS server (if specified): [0].[0].[0].[0]
	⊙ statically configured as:
	IP Address: [192].[168].[0].[160]
	Subnet Mask: [0].[0].[0].[0]
	Default Router: [0].[0].[0].[0]
	DNS Server 1: [0].[0].[0].[0]
	DNS Server 2: [0].[0].[0].[0]
SIP Server:	[ ] (e.g., sip.mycompany.com, or IP address)
Outbound Proxy:	[ ] (e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID:	[ ] (the user part of an SIP address)
Authenticate ID:	[ ] (can be identical to or different from **SIP User ID**)
Authenticate Password:	[ ]

Name: [_____] (optional, e.g., John Doe)

Advanced Options:

Preferred Vocoder:
(in listed order)

choice 1: current setting is "PCMU"
choice 2: current setting is "PCMA"
choice 3: current setting is "G723"
choice 4: current setting is "G729"
choice 5: current setting is "G726-32"
choice 6: current setting is "G728"
choice 7: current setting is "iLBC"
choice 8: current setting is "G722"

G723 rate: ● 6.3kbps encoding rate ○ 5.3kbps encoding rate

iLBC frame size: ● 20ms ○ 30ms

iLBC payload type: [99] (between 96 and 127, default is 98)

Silence Suppression: ● No ○ Yes

Voice Frames per TX: [2] (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)

Layer 3 QoS: [48] (Diff-Serv or Precedence value)

Layer 2 QoS: 802.1Q/VLAN Tag [0] 802.1p priority value [0] (0-7)

Use DNS SRV: ● No ○ Yes

User ID is phone number: ● No ○ Yes

SIP Registration: ● Yes ○ No

Unregister On Reboot: ○ Yes ● No

Register Expiration: [60] (in minutes. default 1 hour, max 45 days)

Early Dial: ● No ○ Yes (use "Yes" only if proxy supports 484 response)

Dial Plan Prefix: [____] (this prefix string is added to each dialed number)

No Key Entry Timeout: [4] (in seconds, default is 4 seconds)

Use # as Dial Key: ○ No ● Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)

local SIP port: [5060] (default 5060)

local RTP port:	`5004` (1024-65535, default 5004)
Use random port:	⊙ No ○ Yes
NAT Traversal:	⊙ No
	○ Yes, STUN server is: `_____` (URI or IP:port)
keep-alive interval:	`20` (in seconds, default 20 seconds)
Use NAT IP	`_____` (if specified, this IP address is used in SIP/SDP message)
TFTP Server:	`67`.`153`.`142`.`69` (for remote software upgrade and configuration)
Voice Mail UserID:	`_____` (User ID/extension for 3rd party voice mail system)
SUBSCRIBE for MWI:	⊙ No, do not send SUBSCRIBE for Message Waiting Indication
	○ Yes, send periodical SUBSCRIBE for Message Waiting Indication
Auto Answer:	⊙ No ○ Yes
Offhook Auto-Dial:	`_____` (User ID/extension to dial automatically when offhook)
Enable Call Features:	⊙ No ○ Yes (if Yes, Call Forwarding & Do-Not-Disturb are supported locally)
Disable Call-Waiting:	⊙ No ○ Yes
Send DTMF:	⊙ in-audio ○ via RTP (RFC2833) ○ via SIP INFO
DTMF Payload Type:	`101`
Send Flash Event:	⊙ No ○ Yes (Flash will be sent as a DTMF event if set to Yes)
NTP Server:	`time.nist.gov` (URI or IP address)
Time Zone:	current setting is "GMT-5:00 (US Eastern Time, New York)" ▼
Date Display Format:	⊙ Year-Month-Day
	○ Month-Day-Year
	○ Day-Month-Year
Daylight Savings Time:	⊙ No ○ Yes (if set to Yes, display time will be 1 hour ahead of normal time)
Default Ring Tone:	○ system ring tone
	○ custom ring tone 1, used if incoming caller ID is `_____`

⊙	custom ring tone 2, used if incoming caller ID is	
○	custom ring tone 3, used if incoming caller ID is	

Send Anonymous: ⊙ No   ○ Yes   (caller ID will be blocked if set to Yes)

Lock keypad update: ⊙ No   ○ Yes   (configuration update via keypad is disabled if set to Yes)

[Update]   [Cancel]   [Reboot]

The definitions for all the configuration parameters in the Configuration Menu are:

Admin Password	This contains the password to access the Web Configuration Page. This field is case sensitive and maximum length is 16 characters.
IP Address	There are 2 modes under which the IP phone can operate: 1. DHCP (default) or PPPoE Mode If DHCP mode is enabled (default), then all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory) and the IP phone will acquire its IP address from the first DHCP server it discovers on the LAN it attaches to. In addition, the phone will attempt PPPoE if DHCP fails. Be careful if the PPPoE account ID or password field is not empty, the phone will try PPPoE from then on and never use DHCP. So leave these two fields BLANK if using DHCP. With this feature, the phone can be directly connected to ISP or broadband service providers' Cable or DSL modem. Use this DNS server (if specified), any IP put here will become the primary DNS server, if also the other field "Use DNS SVR" is set to "Yes". This helps in some cases if the DNS server get from DHCP or PPPoE not working properly, client can put own DNS server here. Default is blank and "Use DNS SVR" is "No". 2. Static IP Mode If Static IP mode is selected, then the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields will need to be configured. These fields are reset to zero by default.
SIP Server	This field contains the URI string or the IP address (and port, if different from 5060) of the SIP proxy server. e.g., the following are some valid examples: sip.my-voip-provider.com, or sip:my-company-sip-server.com, or 192.168.1.200:5066

Outbound Proxy	This field contains the URI string or the IP address (and port, if different from 5060) of the outbound proxy. If there is no outbound proxy, this field **SHOULD** be left blank. If not blank, all outgoing requests will be sent to this outbound proxy.
SIP User ID	This field contains the user part of the SIP address for this phone. e.g., if the SIP address is: sip:my_user_id@my_provider.com, then the SIP User ID is: my_user_id. Please do NOT include the preceding "sip:" scheme or the host portion of the SIP address in this field.
Authenticate ID	This field contains the login ID used for SIP authentication. Typically, this is the account number on an SIP server for this IP phone. It can be the same as or different from the above SIP User ID, depending on whether a separate account ID is used for authentication.
Authenticate Password	This field contains the password used for SIP authentication. It is used together with the above SIP User ID.
Name	This field contains user's real name, it's optional
Preferred Vocoder	The BudgeTone IP phone supports up to 8 different vocoder types including G711-ulaw, G711-alaw, G722, G723, G729A/B, G726-32 (ADPCM), G728 and iLBC. Depending on the product model, some of these codecs may not be provided in standard release. A user can configure codecs in a preference list that will be included with the same preference order in SDP message. The first vocoder in this list can be entered by choosing the appropriate option in "Choice 1". Similarly, the last vocoder in this list can be entered by choosing the appropriate option in "Choice 8".
G723 Rate	This defines the encoding rate for G723 codec. By default, 6.3kbps rate is chosen.
iLBC frame size	This defines the size of the iLBC codec frame. The default setting is 20ms.
iLBC payload type	This defines payload time for iLBC. Default value is 98. The valid range is between 96 and 127.

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	This field contains the number of voice frames to be transmitted in a single packet. When setting this value, the user should be aware of the requested packet time (used in SDP message) as a result of configuring this parameter. This parameter is associated with the first vocoder in the above vocoder Preference List or the actual used payload type negotiated between the 2 conversation parties at run time. e.g., if the first vocoder is configured as G723 and the "Voice Frames per TX" is set to be 2, then the "ptime" value in the SDP message of an INVITE request will be 60ms because each G723 voice frame contains 30ms of audio. Similarly, if this field is set to be 2 and if the first vocoder chosen is G729 or G711 or G726, then the "ptime" value in the SDP message of an INVITE request will be 20ms. If the configured voice frames per TX exceeds the maximum allowed value, the BudgeTone phone will use and save the maximum allowed value for the corresponding first vocoder choice. The maximum value for PCM is 10(x10ms) frames; for G726, it is 20 (x10ms) frames; for G723, it is 32 (x30ms) frames; for G729/G728, 64 (x10ms) and 64 (x2.5ms) frames respectively.
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.
Use DNS SRV:	Default is No. If set to Yes, the phone will use the previous specified DNS server and put DHCP/PPPoE DNS server as 2nd/backup choice
User ID is Phone Number	If the BudgeTone phone has an assigned PSTN telephone number, then this field will 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 BudgeTone phone 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 will be unregistered on reboot.
Register Expiration	This parameter allows the user to specify the time frequency (in minutes) the BudgeTone phone will refresh its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
Early Dial	Default is No. Use only if proxy supports 484 response
Dial Plan Prefix	Sets the prefix added to each dialed number
No Key Entry Timeout	Default is 4 seconds.
Use # as Dial Key	This parameter allows the user to configure the "#" key to be used as the "Send"(or "Dial") key. Once set to "Yes", pressing this key will immediately trigger the sending of dialed string collected so far. In this case, this key is essentially equivalent to the "(Re)Dial" key. If set to "No", this # key will then be included as part of the dial string to be sent out.
Local SIP port	This parameter defines the local SIP port the BudgeTone phone will listen to and transmit on. The default value is 5060.
Local RTP port	This parameter defines the local RTP-RTCP port pair the BudgeTone phone will listen to and transmit on. 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 BudgeTone phones are behind the same NAT.

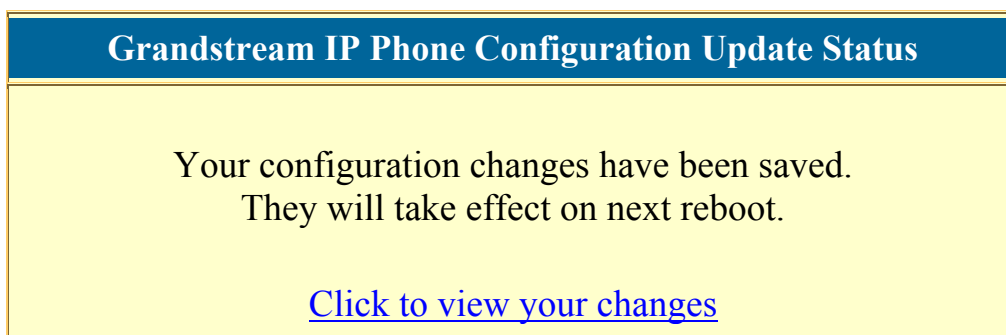|	
NAT Traversal	This parameter defines whether the BudgeTone phone NAT traversal mechanism will be activated or not. If activated (by choosing "Yes") and a STUN server is also specified, then the BudgeTone phone will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the BudgeTone phone will attempt to detect if and what type of firewall/NAT it is behind through communication with the specified STUN server. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the BudgeTone phone will attempt to use its mapped public IP address and port in all the SIP and SDP messages it sends out. If this field is set to "Yes" with no specified STUN server, then the BudgeTone phone will periodically (every 10 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open.
keep-alive interval	Default is 20 seconds. This field specify the time period to send out packets to keep router's NAT mapping port open
Use NAT IP	NAT IP address used in SIP/SDP message. Default is blank.
TFTP Server	This is the IP address of the configured tftp server. If it is non-zero or not blank, the BudgeTone phone will attempt to retrieve 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.
Voice Mail UserID	If the ID number put here, when "MESSAGE" button pressed, the phone will automatically send out a SIP request. For example, if 8005 put here, when "MESSAGE" button pressed, a SIP message "sip:8005@my_provider.com" will be sent to SIP server.
Subscribe for MWI	Default is NO. When set to Yes a SUBSCRIBE for Message Waiting Indication will be sent periodically.
Auto Answer	Default is NO. When set to Yes, any incoming call will be automatically answered via speakerphone after a Beep. This is somewhat similar to Intercom but still different.

Offhook Auto-Dial	This parameter allows the user to configure a User ID or extension number to be automatically dialed upon offhook. Please note that only the user part of a SIP address needs to be entered here. The BudgeTone phone will automatically append the "@" and the host portion of the corresponding SIP address.
Enable Call Features	Default is No. If set to Yes, Call Forwarding & Do-Not-Disturb are supported locally
Disable Call Waiting	Default is No.
Send DTMF	This parameter controls the way DTMF events are transmitted. There are 3 ways: in audio which means DTMF is combined in audio signal (not very reliable with low-bit-rate codec), via RTP (RFC2833), or via SIP INFO.
DTMF Payload Type	Default is 101. This is the inband DTMF payload
Send Flash Event	This parameter allows the user to control whether to send an SIP NOTIFY message indicating the Flash event, or just to switch to the voice channel when the user presses the Flash key.
NTP server	This parameter defines the URI or IP address of the NTP server that the BudgeTone phone will use to display the current date/time. Default is "time.nist.gov"
Time Zone	This parameter controls how the displayed date/time will be adjusted according to the specified time zone.
Date Display Format	Date display format for different countries
Daylight Savings Time	This parameter controls whether the displayed time will be daylight savings time or not. If set to Yes, then the displayed time will be 1 hour ahead of normal time.
Default Ring Tone	Different system ring tones (now have 3) can be selected. Also can use different ring tone to identify different incoming call by caller ID
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.

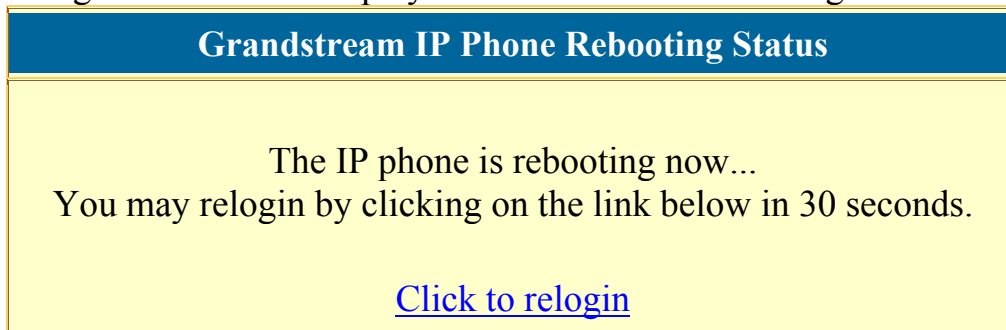|	
Lock keypad update	If this parameter is set to "Yes", the configuration update via keypad is disabled.

5.2.3 Saving the Configuration Changes

Once a change is made, the user should press the "Update" button in the Configuration Menu. The IP phone will then display the following screen to confirm that the changes have been saved.

Grandstream IP Phone Configuration Update Status
Your configuration changes have been saved. They will take effect on next reboot. Click to view your changes

The user is recommended to power cycle the IP phone after seeing the above message.

5.2.4 Rebooting the phone from remotely

The administrator of the phone can remotely reboot the phone by pressing the "Reboot" button at the bottom of the configuration menu. Once done, the following screen will be displayed to indicate that rebooting is underway.

Grandstream IP Phone Rebooting Status
The IP phone is rebooting now... You may relogin by clicking on the link below in 30 seconds. Click to relogin

At this point, the user can relogin to the phone after waiting for about 30 seconds.

5.3 Configuration through a Central Server

Grandstream IP phones can be automatically configured via a central provisioning system called Grandstream Automated Provisioning System (GAPS).

With GAPS, a service provider or an enterprise with large deployment of IP phones can easily manage the configuration and service provisioning of individual devices remotely and automatically via a central server. GAPS uses enhanced (NAT friendly) tftp and other communication protocols to communicate with each individual IP phone even if the phone is behind a NAT.

GAPS must be used to support automated configuration of an IP phone. To enable this feature on the phone, the user just needs to enter the IP address of GAPS server in the tftp server field of the configuration screen. Then power cycle the phone.

For details on how GAPS works, please refer to the documentation of GAPS product.

6 Software Upgrade

6.1 Upgrade with TFTP

Warning !!!

*The device **WILL** get damaged if there is a power outage during firmware upgrade. Grandstream **STRONGLY** recommend customer maintain **UNINTERRUPTED POWER SUPPLY** during firmware upgrade. This damage is **NOT** covered by the manufacture warranty. Grandstream will **NOT** take any responsibility for this kind of damage. Please be very **CAREFUL** when doing firmware upgrade.*

To upgrade software, BudgeTone-100 phone can be configured with a TFTP server where the new code image is located. The TFTP upgrade can work in either static IP or DHCP mode using private or public IP address. It is recommended that the TFTP server must have either public IP address or be on the same LAN segment with the phone.

There are 2 ways to set up the TFTP server to upgrade the firmware, namely via keypad and via the phone's Web configuration interface. To configure the TFTP server via keypad, press "Menu" key and then press the down arrow key 5 times until you see the "tftp" option. Press the "Menu" again and the current TFTP server IP address will be displayed on the LCD. Enter the new TFTP server's full 12-digit IP address and wait about 2 seconds for the 3 separating "." to show up on the LCD. E.g., if the new TFTP server IP is 192.168.1.90, enter 192168001090. After about 2 seconds, the LCD will display 192.168.001.090. At that time, press the "Menu" key to confirm and quit the TFTP IP editing mode.

To configure the TFTP server via the Web configuration interface, open up your browser to point at the IP address of the phone. Input the admin password to enter the configuration screen. From there, enter the TFTP server address in the designated field towards the bottom of the configuration screen.

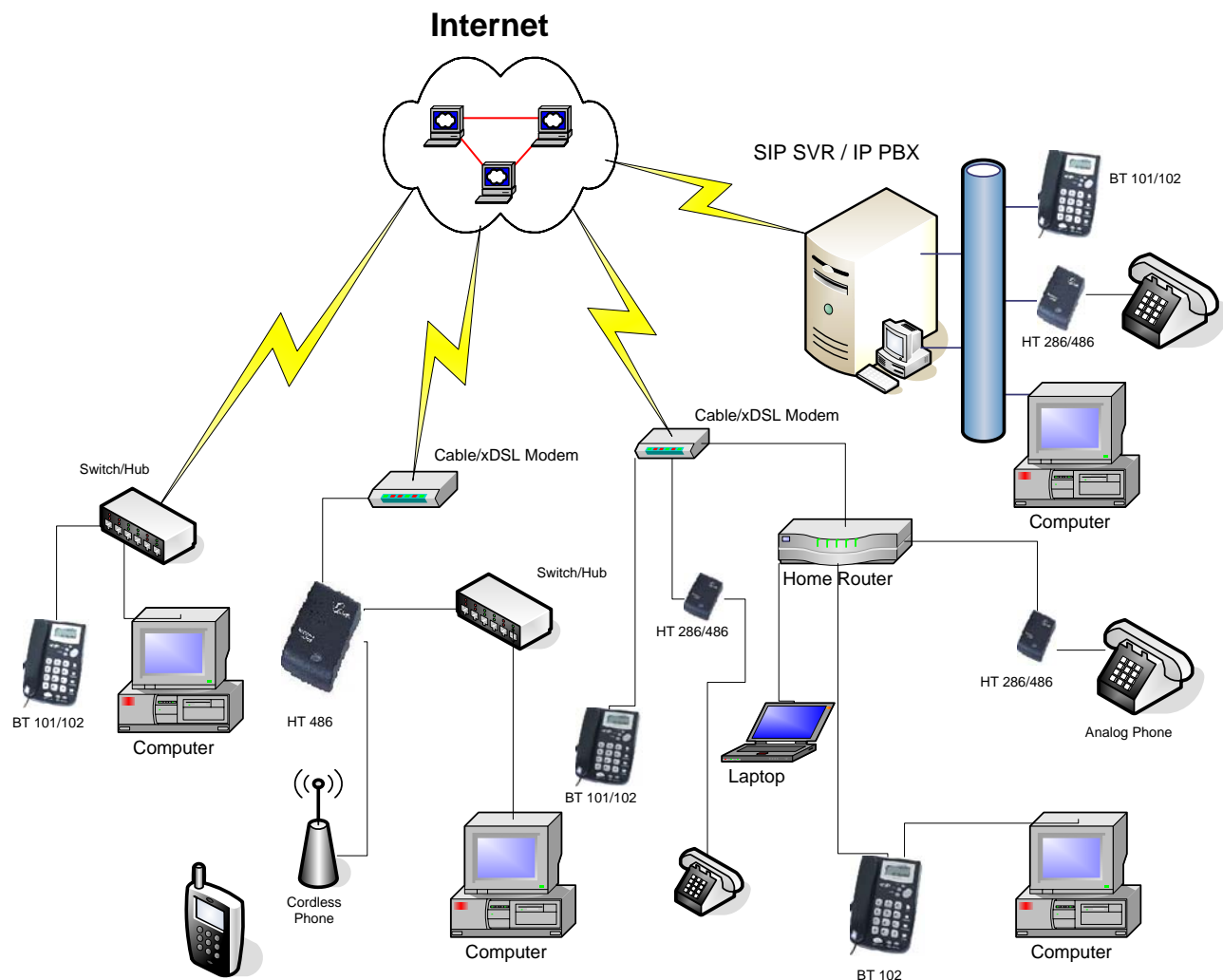Once the TFTP server is configured, power cycle the IP phone.

TFTP checking is only performed during the initial power up. If the configured tftp server is found and a new code image is available, the phone will attempt to retrieve the new image files by downloading them into the phone's SRAM. During this stage, the Keypad LEDs will blink with 0.25 second ON and 0.25 second OFF pattern 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 phone will stop the TFTP process and simply boot using the existing code image in the flash.

TFTP may take as long as 1—10 minutes over Internet, or just 20+ seconds if it is performed on a LAN. It is generally recommended to conduct TFTP upgrade in a controlled LAN environment if possible because the network environment like firewall, NAT could affect TFTP function properly. For users who do not have local TFTP server, Grandstream provides a NAT-friendly TFTP server on the public Internet for users to download the latest firmware upgrade automatically. Please check the Service or Support section of Grandstream's Web site to obtain this TFTP server IP address.

7 Connection Diagram

Network connection diagrams for Grandstream series products.

8 Restore Factory Default Setting

Warning !!!

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

Step 1:

Find the Mac Address of the device. The Mac address of the device is located on the bottom of the device. It is a 12-digit number.

Step 2:

Encode the Mac address. The encode rule is:
"2" is the first letter on the button "2" so its encoding is "2".
"A" is the second letter on button "2" so its encoding is "22".
"B" is the third letter on button "2" and its encoding is "222".
"C" is the fourth letter on button "2" and its encoding is "2222".

For example, the Mac address is 000b8200e395,
User should encode it as "0002228200333395".

Step 3:

Access the phone screen menu, then select the "-- reset --" with the up or down arrows keys.

Step 4:

Dial in the encode of the Mac address. Once the correct Mac address is displayed in the LCD screen, press "MENU" button, the device will reboot automatically and restore to the factory default setting.