User Manual

BudgeTone-100 Series

IP Phone

Version 1.00

OvisLink (Canada) Inc.

http://www.ovislink.ca/news/bt100.htm

Table of Contents

1 Welcome

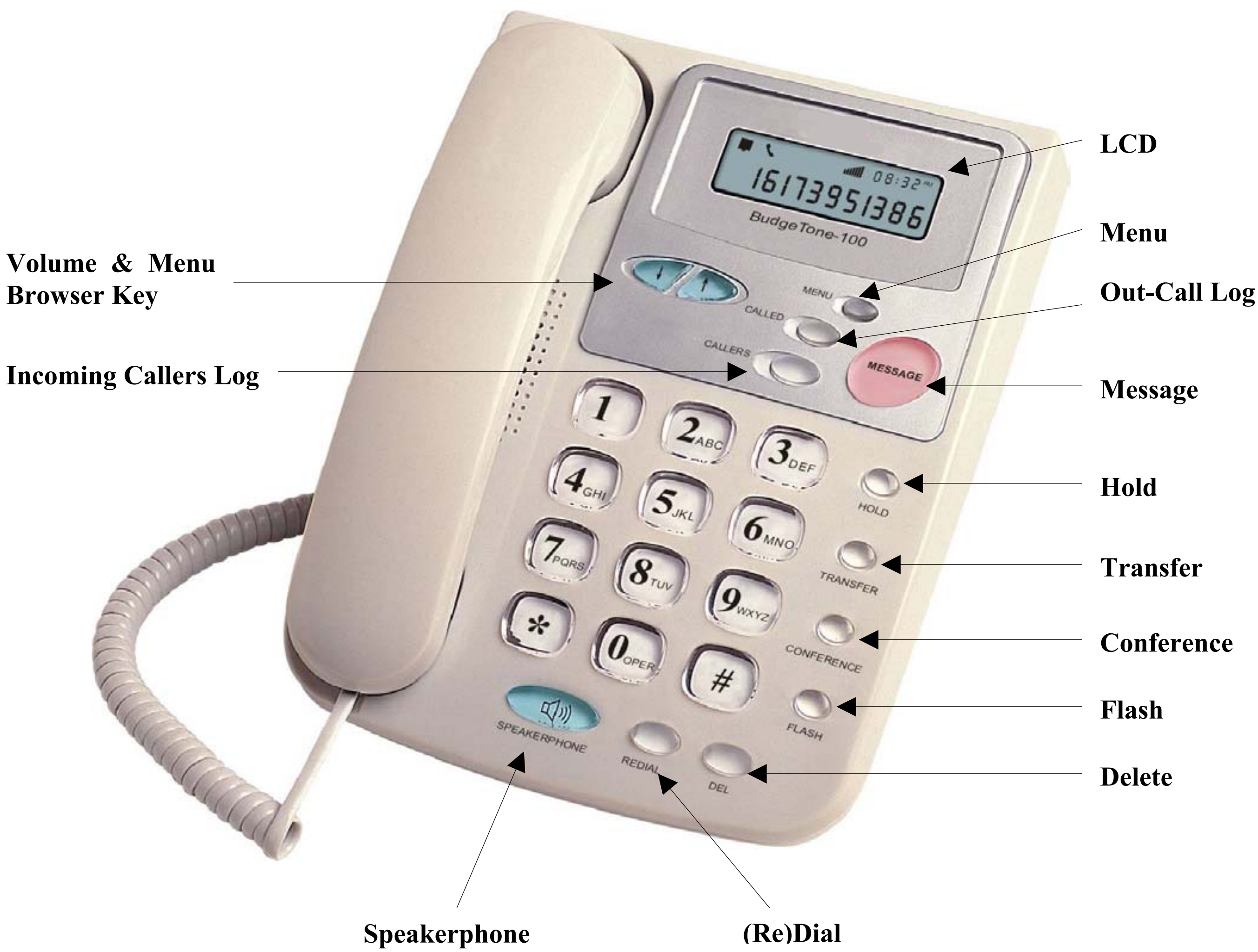
Congratulations on becoming an owner of BudgeTone-100 IP telephone! You made an excellent choice and we hope you will enjoy all its capabilities.

Budge Tone 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:

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.



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
4) User Manual

2.2 Safety Compliances

The BudgeTone-100 phone is compliant with various safety standards including FCC/CE/UL. 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

Budge Tone IP Phone is a next generation IP network telephone based on industry open standard SIP (Session Initiation Protocol). Built on innovative technology, Budge Tone 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, RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, TFTP protocols
- Support IETF STUN and SIMPLE standard extension (Model 102D)
- Interoperable with various 3rd party SIP end user device, Proxy / Registrar / Server, and gateway products.
- Advanced Digital Signal Processing (DSP) technology to ensure superior audio quality
- Advanced and patent pending adaptive jitter buffer control, packet delay and loss concealment technology
- Support popular vocoders including G.723.1 (5.3K/6.3K), G.729A/B, G.711 (a-law and u-law), G.726 (40K/32K/24K/16K), as well as G.728 (Model 102D)
- Support standard voice features such as Caller ID Display or Block, Call Waiting, Hold, Transfer, Forward, in-band and out-of-band DTMF (RFC2833), Dial Plans
- Support 3-way conferencing (Model 102D), full duplex hands-free speakerphone, redial, call log, volume control, voice mail with indicator, downloadable ring tone (Model 102D)
- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168), and AGC (Automatic Gain Control)
- Support BASIC and DIGEST authentication (MD5, MD5-sess)
- Provide easy configuration thru manual operation (phone keypad and Web interface) or automated centralized configuration file.
- Support for Layer 2 (802.1Q VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
- Remote software upgrade capability via TFTP
- Built-in alarm-clock with downloadable music ringing tone (Model 102D)
- Optional voice encryption (secure RTP, Model 102D)

3.2 Hardware Specification

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

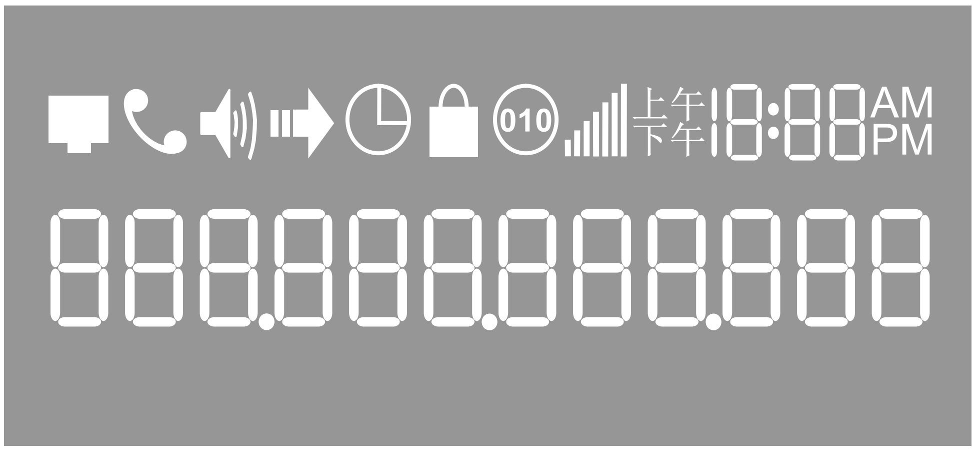
The table below describes the difference among these models.

Model	**BudgeTone-101**	**BudgeTone-102**	**BudgeTone-102D**
LAN interface	1xRJ45 10Base-T	2xRJ45 10Base-T	2xRJ45 10Base-T
Phone Case	25-button keypad 12-digit caller ID LCD	25-button keypad 12-digit caller ID LCD	25-button keypad 16x2 character LCD
Universal Switching Power Adaptor	Input: 100-240VAC Output: +5VDC, 400mA	Same as left	Same as left
Dimension	18cm (W) 22cm (D) 6.5cm (H)	Same as left	Same as left
Weight	2 lbs (0.9 kg)	Same as left	Same as left
Operating Temperature	32 - 104^{o}F 0 - 40^{o}C	Same as left	Same as left
Humidity	10% - 95% (non-condensing)	Same as left	Same as left
Compliance	UL/FCC/CE	Same as left	Same as left

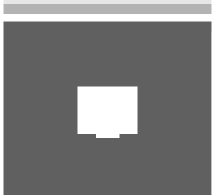
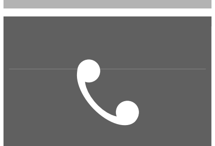
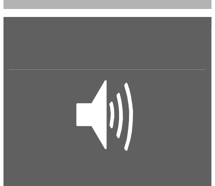
4 Basic Operations

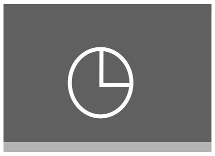
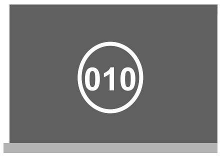
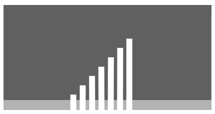
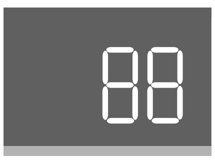
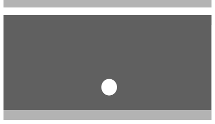
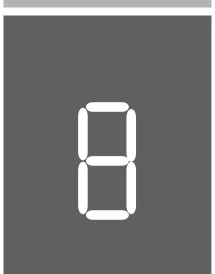
4.1 Get Familiar with LCD

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

	Alarm Clock Status Icon: OFF when the alarm clock is not set ON when the alarm clock is set
	Lock Status Icon: OFF when the lock is set ON when the lock is not set
	Encryption Status Icon: OFF when the voice encryption is off ON when the voice encryption is on
	Handset and Speakerphone Volume Icons: 0-7 scales to adjust handset / speakerphone volume
	Real-time Clock: Synchronized to Internet time server Time zone configurable via web browser
	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
	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
DEL	Delete a key entry, call log, voice mail and etc
(RE)DIAL	Redial the number dialed last time. 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 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 REDIAL button.
   Pick up the handset or press the speakerphone button, then press the "Called" button to browse thru the last 10 numbers dialed out. Once the desired number is identified and displayed on the LCD screen, press the (Re)Dial button and a new call to that displayed number will be sent out immediately.
4. Browse the CALLERS history and press the 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 (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 "(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 "(Re)Dial" button.

If the "(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 “(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 again. 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 basically 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

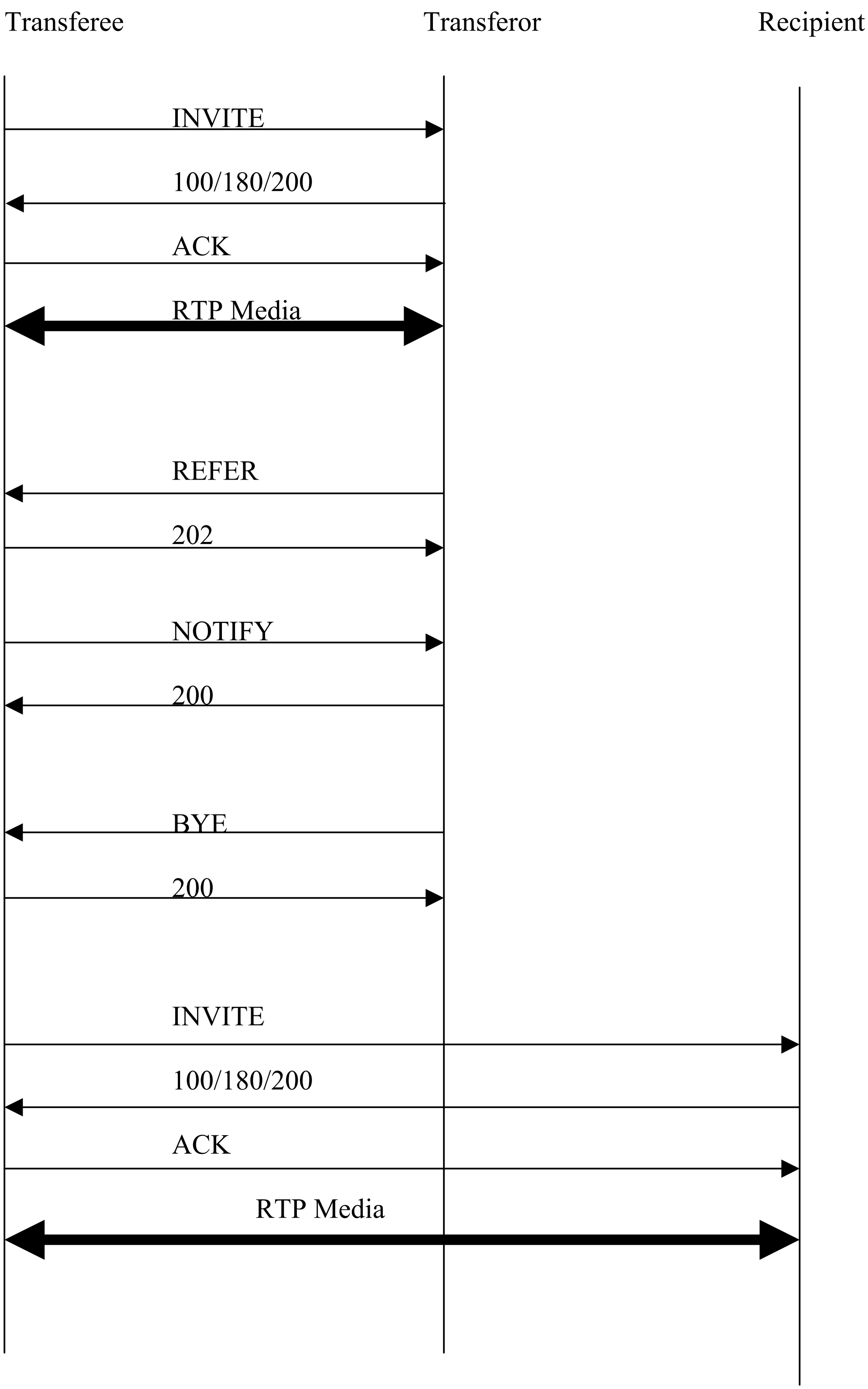
The user can transfer an active call to a third phone by using the "Transfer" button. The sequence is like this:

> *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.*

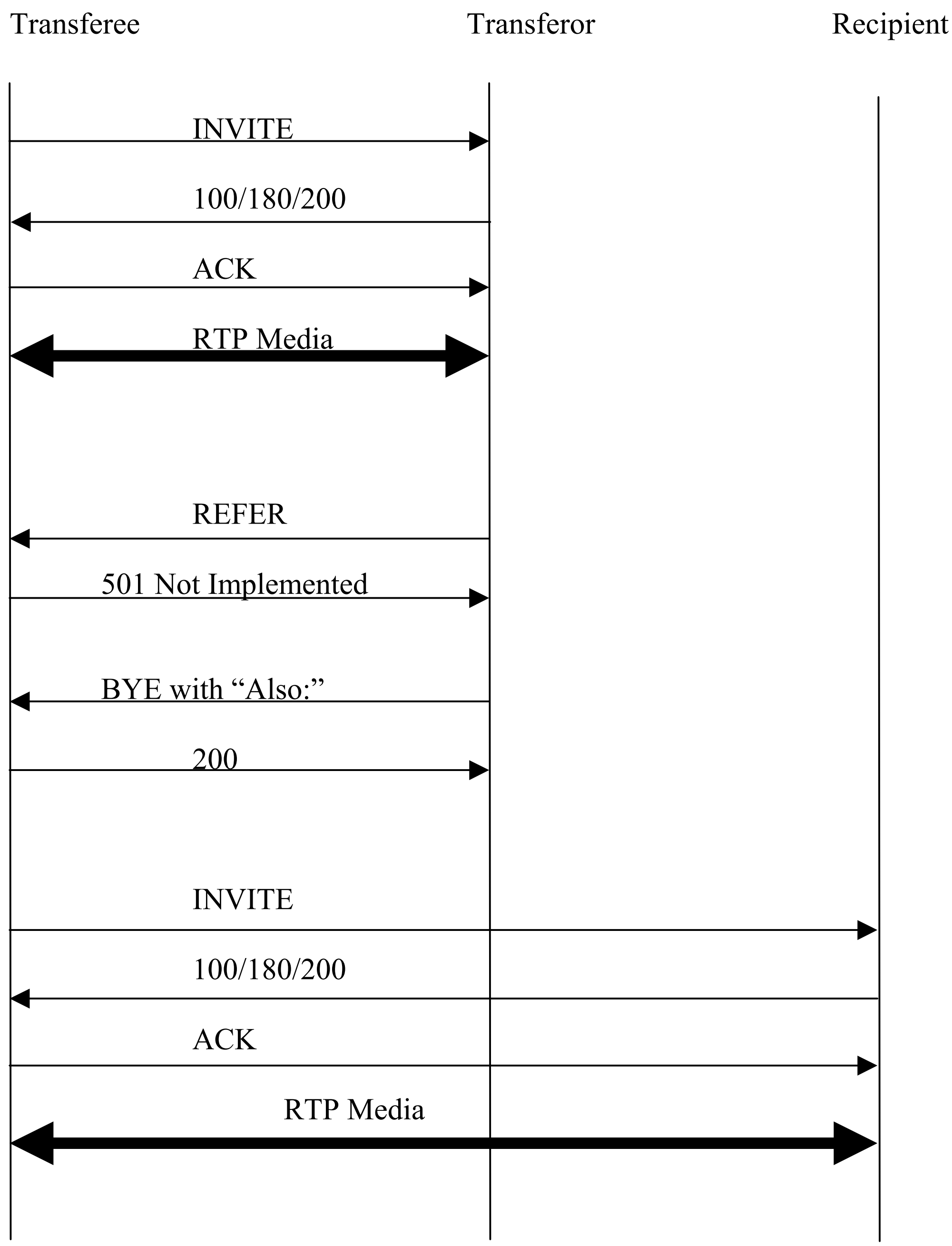
2 kinds of blind call transfers are supported: using REFER and using BYE/Also:.

The SIP message flow based on SIP REGER method looks something like this:

Call Flow Diagram For Blind Call Transfer



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



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 effective!!!
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 effective!!!
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 effective!!!
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 effective!!!

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 effective!!!
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-723 1**" Press *Menu* to select new vocoder Press '↓' or '↑' to browse a list of available vocoders line 1 " **- G-711u 2**" 2 " **- G-711A 2**" 3 " **- G-723 1**" 4 " **- G-726 1**" 5 " **- G-728 4**" 6 " **- G-729 1**" Press 1 to 9 to indicate number of frames per TX packet Press *Menu* to save and exit Must recycle power to take effective!!!
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 effective immediately!

Menu Item	Menu Functions
9	Display "**[9] codE reL**" Press *Menu* to display the code releases Press '↓' or '↑' to browse line 1 "**b 2003-02-15**" – date: *boot* code 2 " **1. 0. 0. 2**" – version: *boot* code 3 "**P 2003-02-16**" – date: *phone* code 4 " **1. 0. 0. 5**" – version: *phone* code 5 "**c 2003-02-16**" – date: *vocoder* 6 " **1. 0. 0. 5**" – version: *vocoder* 7 "**h 2003-02-16**" – date: *web* server 8 " **1. 0. 0. 5**" – version: *web* server 9 "**r 2003-02-16**" – date: *ring* tone 10 " **1. 0. 0. 5**" – version: *ring* tone Press *Menu* to exit
10	Display "**10] Phy Addr**" Press *Menu* to display the physical / MAC address Press *Menu*, '↓' or '↑' to exit

5.2 Configuration with Web Browser

BudgeTone-100 series IP phone has an embedded Web server that will respond to HTTP GET/POST requests from a Web browser. 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:

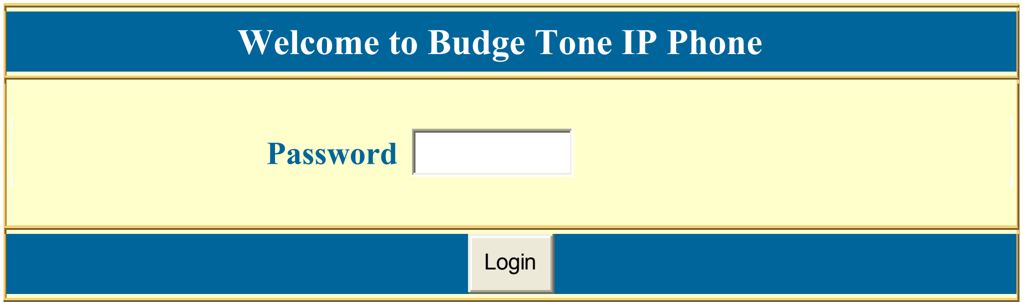
http://Phone-IP-Address,

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:



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

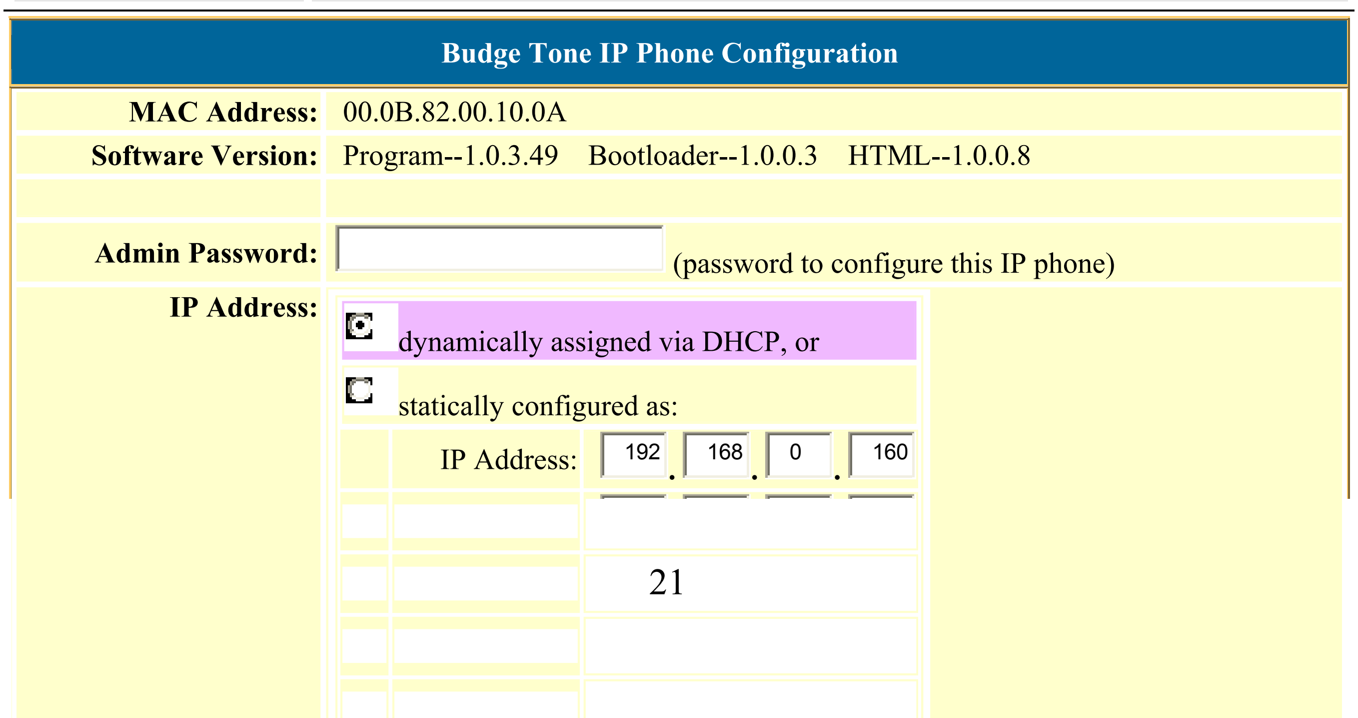
5.2.2 Configuration Menu

After the correct password is entered in the login screen, the embedded Web server inside the IP phone will respond with the Configuration Menu screen which is explained in details below.

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

Password	This contains the password to access the Web Configuration Menu. This field is case sensitive.
IP Address	There are 2 modes under which the IP phone can operate: - If DHCP mode is enabled, then all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory) and the IP phone will acquire its IP address from the first DHCP server it discovers on the LAN it attaches to. - 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.

	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.
SIP User ID is Phone Number	If the IP 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 Login 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.
SIP Password	This field contains the password used for SIP authentication. It is used together with the above SIP Login ID



	Subnet Mask: 255 . 255 . 255 . 0 Default Router: 0 . 0 . 0 . 0 DNS Server 1: 0 . 0 . 0 . 0 DNS Server 2: 0 . 0 . 0 . 0
SIP Server:	sipprovider.com (e.g., sip.mycompany.com, or IP address)
Outbound Proxy:	(e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID:	8001
Authenticate ID:	8001 (can be identical to or different from **SIP User ID**)
Authenticate Password:	
Name:	John Doe (optional, e.g., John Doe)
Advanced Options:	
Preferred Vocoder: *(in listed order)*	Choice1: current setting is "PCMU" choice2: current setting is "G729" choice3: G.723.1 choice4: PCMA choice5: current setting is "G726-32" choice6: current setting is "G728"
G723 rate:	6.3kbps encoding rate 5.3kbps encoding rate
Silence Suppression:	No Yes
Voice Frames per TX:	2 (up to 10/20/32/64 frames for G711/G726/G723/othercodecs respectively)
IP QoS:	48 (IP Diff-Serv or Precedence value for RTP)
VLAN Tag:	0 (VLAN classification for RTP)
SIP User ID is phone number:	No Yes
Dial Plan:	(dial plan prefix, used only when **SIP User ID** is phone number)
SIP Registration:	Yes No
Register Expiration:	1 (in minutes. default 1 hour, max 45 days)

	◉ No ○ Yes (use "Yes" only if proxy supports 484 response)
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)
NAT Traversal:	◉ No ○ Yes, STUN server is: 0 . 0 . 0 . 0
TFTP Server:	. . . (for remote software upgrade and configuration)
Voice Mail UserID:	(User ID/extension for 3rd party voice mail system)
Offhook Auto-Dial:	(User ID/extension to dial automatically when offhook)
NTP Server:	time.nist.gov (URI or IP address)
Time Zone:	current setting is "GMT-5:00 (US Eastern Time, New York)"
	◉ No ○ Yes (caller ID will be blocked if set to Yes)
	Update

Preferred Vocoder	The BudgeTone IP phone supports up to 6 different vocoder types including G711-ulaw, G711-alaw, G723, G729A/B, G726-32 (ADPCM), and G728. Depending on the product model, some of these vocoders may not be provided in standard release. A user can configure vocoders 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 6”.
G723 Rate:	This defines the encoding rate for G723 vocoder. By default, 6.3kbps rate is chosen.
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 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.
IP 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.
VLAN Tag	This contains the value used for layer 2 VLAN tag. Default setting is blank.
Dial Plan	This value contains the dial plan prefix string (typically an ASCII numeric string). If it is not blank, then this string will be used as a prefix to the target URI string in the "To" header field of an INVITE message.

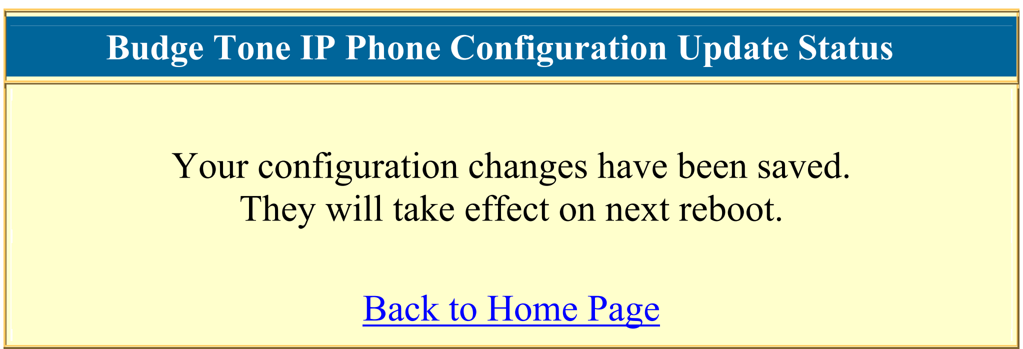
Early Dial	This parameter controls whether the phone will attempt to send an early INVITE each time a key is pressed when a user dials a number. If set to "Yes", an INVITE is sent using the dial-number collected thus far; Otherwise, no INVITE is sent until the "(Re-)Dial" button is pressed or after about 5 seconds have elapsed if the user forgets to press the "(Re-)Dial" button. The "Yes" option should be used ONLY if there is a SIP proxy configured and the proxy server supports 484 Incomplete Address response. Otherwise, the call will most likely be rejected by the proxy (with a 404 Not Found error). Please note that this feature is NOT designed to work with and should NOT be enabled for direct IP-to-IP calling.
Use # as Send 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.
SIP Registration	This parameter controls whether the IP phone needs to send REGISTER messages to the proxy server. The default setting is "Yes".
Registration Interval	This parameter allows the user to specify the time frequency (in minutes) the 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).
Local SIP port	This parameter defines the local SIP port the IP phone will listen and transmit on. The default value is 5060.
Local RTP port	This parameter defines the local RTP-RTCP port pair the IP phone will listen 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.

NAT Traversal	This parameter defines whether the phone NAT traversal mechanism will be activated or not. If activated (by choosing "Yes") and a STUN server is also specified, then the phone will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the 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 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 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.
TFTP Server	This is the IP address of the configured tftp server. If it is non-zero or not blank, the IP 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 User ID	This parameter defines the User ID (or extension number) of a 3rd party voice mail system where the user may have an account. By defining this Voice Mail extension, when the user presses the "Message" button on the phone, an INVITE message will be sent to that extension to allow the user to retrieve messages.
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 phone will automatically append the "@" and the host portion of the corresponding SIP address.
NTP server	This parameter defines the URI or IP address of the NTP server which the IP phone will use to display the current date/time.
Time Zone	This parameter controls how the displayed date/time will be adjusted according to the specified time zone.

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.

5.2.3 Saving the Configuration Changes

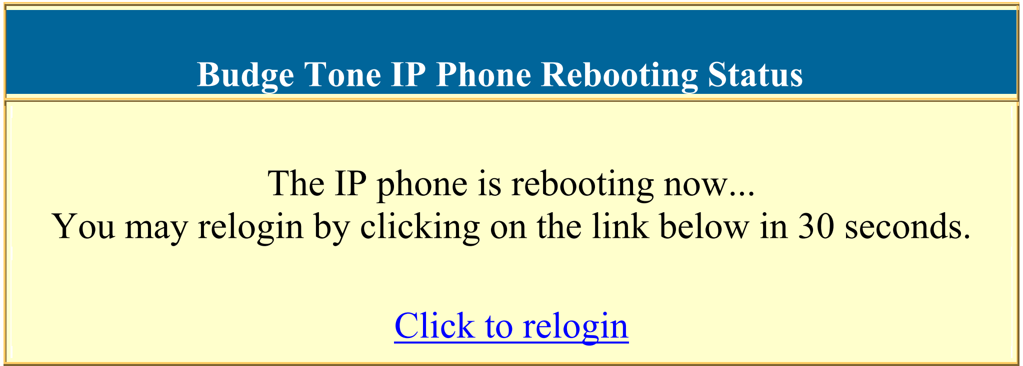
Once a change is made, the user should press the "Update" button in the Configuration Menu. The IP phone will then display the following screen to confirm that the changes have been saved.



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.



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

5.3 Configuration through a Central Server

The content of this section will be provided when this feature is implemented in the near future.

6 Software Upgrade

6.1 Upgrade with TFTP

To upgrade software, BudgeTone-100 phone can be configured with a tftp server where the new code image is stored. The phone can be configured in either static IP or DHCP mode using private or public IP address. TFTP server must have either public IP address or be on the same LAN with the phone. Once this is done, 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 4 minutes over Internet, or just a few seconds if it is performed on a LAN. It is generally recommended to use conduct tftp upgrade in a controlled LAN environment.