Q: Are BudgeTone SIP phones certified for use in North America and Europe?
A: Yes. The BudgeTone-101 and BudgeTone-102 IP phones are FCC (part 15) and CE certified and thus can be used in the US and European market. Its universal power adaptors are UL certified.

Q: What are the differences among BudgeTone-101, 102 and 102D models?
A: Within the BudgeTone-100 series family, model 101 and 102 have the same software functions and the only difference is that model 101 has 1 (one) Ethernet interface and model 102 has 2 (two) Ethernet interfaces.
Model 102D will have a better character based LCD, 2 (two) Ethernet interfaces, and more software functions such as 3-way conferencing, support for SIMPLE, support for more vocoders, future support for power-over-Ethernet, ear phone interface, etc.

Q: Can I call another IP phone using its IP address directly without a proxy?
Yes. Direct IP-to-IP calling is supported. Please refer to Users Manual for details.

Q: ***Where to get the latest software release for BugeTone 100 series SIP phones?***
A: It is available on the our website.

Q: How do I upgrade my Grandstream BT100 series phone's firmware?
A: First configure your tftp server on the phone and then power cycle or reboot the phone. Please consult user manual for detail.

Q: How do I setup my tftp server on the phone?
A: There are 2 ways setting up your tftp server.
1) From the phone's web page.
2) From phone keypad, press menu button and down arrow to item number 6, press menu button one more time to get into the "Editing" mode. If the tftp server IP displayed is not the one you want, enter the entire 12 digits of the IP address of your tftp server. e.g., if your tftp server is 192.168.1.100, enter 192168001100.

Q: How do I know which firmware version my IP phone is running?
A: There are two ways that you can see your phone's firmware version:
1) use "menu" button.

Press “menu” and then up arrow button 3 times to get to item number 9, “codE rEL”, and then press "menu" one more time to get into the “codE rEL” checking mode, use up or down arrow button to see “B (bootloader)” and “P (program)” code date and version.
2) From Grandstream phone’s web page.
The software version should be displayed on top of the web configuration page in a format similar to the following:
Software Version: Program--1.0.X.X Bootloader--1.0.X.X HTML--1.0.X.X

Q: Why is my phone’s LCD keep lighting up without showing any date?
A: The most possible problem will be that the phone is not getting responses from the NTP server. Check your network connection, DNS server or try to use another NTP server.

Q: Why is my phone showing date “1900-01-02”?
A: This is probably because either DNS is not resolving the NTP server correctly, or the NTP server is not responding. This 1900-01-02 is the default date shown under this circumstance.

Q: Which NTP server can I use?
A: We use "time.nist.gov" as default NTP server. If that or your own NTP server does not work, try to select an NTP server from following link:
http://www.eecis.udel.edu/~mills/ntp/clock1a.html
Or you can get more info from www.ntp.org.

Q: What is Outbound proxy? Should I put an Outbound proxy in the field?
A: An Outbound proxy is mostly used in presence of a firewall/NAT to handle the signaling and media traffic across the firewall. Generally, if you have an outbound proxy and you are not using STUN or other firewall/NAT traversal mechanisms, you can use it. However, if you are using STUN or other firewall/MAT traversal tools, do not use an outbound proxy at the same time.

Q: What is the difference between “User ID” and “Authentication ID”?
A: User ID is the user part of the SIP address of the phone and this is usually the information displayed as Caller ID on the LCD. e.g., typically it is a phone number or extension number or a user’s name. Authentication ID is an ID used strictly for authentication purpose when the phone attempts to contact the SIP server. This may or may not be the same as User ID.

Q: What if my SIP URI domain is different from the SIP proxy server FQDN (Fully Qualified Domain Name)?
A: With firmware 1.0.3.60 and later, you can put the your SIP URI domain name into the **SIP Server** field, and put the actual sip server FQDN into **Outbound Proxy** field. The phone will use the domain name in **SIP Server** as part of SIP URI but send

and receive SIP messages through the SIP proxy server defined in the **Outbound Proxy** field.

Q What Codec should I use for my Granstream phone?
A: By default, PCMU(G711u) will be used. Both PCMU and PCMA will give you toll quality but their bandwidth consumption is also the highest (64kbps). If your network bandwidth is low, you can choose other lower-bit-rate codec such as G723 or G729 which will give you near toll quality at much smaller bandwidth consumption (G723 consumes 5.3/6.3kbps and G729 consumes 8kbps). If bandwidth is not a concern and you want good voice quality, try using PCMU or PCMA, or even the new wide band codec G722 (64kbps) which will provide hi-fidelity voice that is better than toll quality.

Q: What number should I use for "Voice Frames per TX"?
It depends on what codec you choose and balance between bandwidth utilization and impact of packet loss. The bigger this value, the higher bandwidth utilization because more voice frames are packed into the payload field of a UDP/RTP packet and thus the network header overhead would be lower. However, the impact of a packet loss on perceived voice quality will be bigger.
For PCMU/PCMA, the default is 2 and max is 10
For G723, the default is 1 and max is 32
For G726-32, the default is 2 and max is 20
For G729, the default is 2 and max is 64
For G728, the default is 4 and max is 64

Q: What is "Early Dial"? Should I use it?
A: When you dial a number, if you do not press the "Dial" ("Redial") or "#" key if it is configured to function as the "Send" key at the end of your dialed string, the phone will wait for about 4 seconds before timeout and then sends the actual INVITE message. If you set "Early Dial" to be YES, then the phone will attempt to send out INVITE at each key input using the entered dial string collected so far. If the SIP server supports 484 Incomplete Address response, the phone will keep trying with each new key entry until the complete dialed string is entered. This will essentially eliminate the 4-second wait time mentioned above.
Please note that this option can be used ONLY when the SIP server supports 484 Incomplete Address response. Otherwise, any other negative responses from the SIP server (such as 404 Not Found) will cause immediate termination of the call.

Q: What is STUN? Should I use STUN?
A: STUN stands for Simple Traversal of UDP over NAT. It is a protocol which enables an IP phone to detect the presence and type of NAT behind which the phone is placed. An IP phone that supports STUN can intelligently modify the private IP address and port in its SIP/SDP message by using the NAT mapped public IP address and port through a series of STUN queries against a STUN server located on the public Internet. This will allow SIP signaling and RTP media to successfully traverse a NAT without requiring any configuration changes on the NAT.

STUN presents a working solution for most NATs that are not symmetric NAT, e.g., most of the SOHO routers have non-symmatric NAT and in this case, it is OK to use STUN. However, STUN does NOT work with symmetric NAT and if your routers have built-in symmetric NAT, do not use STUN.

Note: NOT ALL SIP PROXY SERVER WILL WORK with A STUN TRANSLATED SIP MESSAGES, PLEASE CONSULT YOUR SERVICE PROVIDER FOR DETAIL.

Q: Do I still need to put in "Outbound" proxy if my phone is working under STUN?
A: NO.

Q: Which other 3rd party SIP applications and products are BudgeTone SIP phones compatible with?
A: We have been active participants in the SIPit events and have done extensive tests successfully with a number of 3rd party SIP products directly or indirectly through our customers or partners. A partial list of other products with which we have successfully tested basic and in some cases advanced features include:

Cisco (7960/7905 IP phones, ATA186, SIP proxy, 5300/3640, etc)
Microsoft (Messenger, RTC Server)
DynamicSoft (SIP proxy)
Broadsoft (softswitch)
Santera/Tekelec (softswitch)
Siemens (IP phone)
Nortel (softswitch, softphone)
Intel (gateway)
Lucent (media server)
Alcatel
Jasomi (border controller)
iptel/SER (open source SIP proxy)
Digium/Asterisk (open source IP PBX)
vovida.org/VOCAL (open source IP PBX)
Mitel (IP phone)
Pingtel (IP phone, softphone)
Teledex (IP phone)
Dlink (IP phone)
Ingate/Intertex (SIP aware firewall)
Hotsip (SIP proxy, STUN server)
Radvision
Sylantro
Hughes Software Systems
Avaya
Ericsson
Nokia
Sharp
TI/Telogy
eDial (conferencing server)
octave (conferencing server)

Cosmocom (contact center application)
IVR Technologies (software based IVR system)
SoftFront (SIP proxy, UA)
Vegastream (gateway)
UTStarcom (SIP proxy, media gateway)
Tangerine (SIP proxy/registrar)
and many others ……

Q: Do you have other colors than the white one?
A: We will have another dark gray color option in the near future.

Q: Which SIP based IP telephone service providers do you currently support?
A: We have tested with and support the following service providers' SIP network:
MCI
Nikotel
Delta3
Telic.net
Go2Call
Free World Dialup
This list will continue to expand and please check for updates from time to time.

Q: How do I setup my Grandstream Phone for Delta3/iconnect network?
A: typical configuration is:
SIP Server: natrelay.deltathree.com
outbound proxy: leave it blank
User ID: xxxxxx (your Delta3 account number)
Authentication/Login ID: xxxxx (same as above, your Delta3 account number)
Password: xxxxx (your Delta3 password)
Dial plan: 6666

Q: How do I setup my Grandstream Phone for nikotel network?
A: typical configuration is:
SIP Server: calamar0.nikotel.com
Outbound proxy: leave it blank
User ID: xxxxx (your nikotel account number)
Authentication ID: same as your User ID
Password: your nikotel password
NAT Traversal: YES (WITHOUT setting the STUN server)

Q: How do I setup my Grandstream Phone for MCI(test) network?
A: typical configuration is:
SIP Server: siptest.mci.com
Outbound proxy: (use an outbound proxy if MCI provides one for you)
User ID: xxxxx (your MCI assigned account/phone number)
Authentication ID: (Your MCI assigned id, i.e., foo)

Password: your MCI password
NAT Traversal: No (You need to set up your STUN server if you don't have outbound proxy)

Note: MCI Proxy server seems to respond our phone client SIP messages correctly.

Q: How do I setup my Grandstream Phone for telic.net network?
A: typical configuration is:
SIP Server: sip.telic.net
Outbound proxy: (Use outbound proxy, it will not work under STUN for now)
User ID: xxxxx (your Telic.net account number)
Authentication ID: same as your User ID
Password: your Telic.net password

Note: STUN is not working yet against Telic.net's SIP proxy server for now.

Q: How do I setup my Grandstream Phone for go2call network?
A: typical configuration is:
SIP Server: voip01.go2call.com
Outbound proxy: (Should leave it blank, because it's a GW)
User ID: xxxxx (your Go2Call PIN number)
Authentication ID: same as your User ID
Password: xxxxxxx (Your Go2Call password)
NAT Traversal: YES (WITHOUT setting the STUN server)

Q: How do I setup my Grandstream Phone for FWD service?
A typical configuration is:
SIP Server: fwd.pulver.com
outbound proxy: 192.246.69.247:5082 (used only when behind firewall, otherwise leave it blank)
User ID: xxxxxx (your FWD account number)
Authentication/Login ID: xxxxx (same as above, your FWD account number)
Password: xxxxx (your FWD password)
NAT Traversal: No (You need to set up your STUN server if you don't have outbound proxy)

Q: How to reset my Budgetone 100 phone to factory default setting?
A:
1. set up the phone to have NO tftp server and then boot without network
2. Press the Menu and then the upward arrow key, you will see the Reset option on the LCD. At this point, enter the full 12 digit of the MAC address or Product ID of your phone (printed at the back of your phone). e.g., for "A/B/C", press the "2" button until you see "A" or "B" or "C" appears, for "D/E/F", press the "3" button until you see the corresponding character. Then continue to enter the complete 12 digit of the MAC address. After that, press "Menu" again.