

Grandstream IP Phone Configuration

MAC Address: 00.0B.82.02.BD.3D

Product Model: BT100

Software Version: Program--1.0.5.11 Bootloader--1.0.0.18 HTML--1.0.0.37 VOC--1.0.0.6

Custom Ring Tone: ring1--1.0.0.0 ring2--1.0.0.0 ring3--1.0.0.0
(all zeroes means unavailable or unsupported)
detected NAT type is open Internet

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): . . .

statically configured as:

IP Address: . . .

Subnet Mask: . . .

Default Router: . . .

DNS Server 1: . . .

DNS Server 2: . . .

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: choice 1: current setting is "PCMA" ▼

(in listed order) choice 2: current setting is "PCMU" ▼

choice 3: current setting is "iLBC" ▼

choice 4: current setting is "iLBC" ▼

choice 5: current setting is "iLBC" ▼

choice 6: current setting is "iLBC" ▼

choice 7: current setting is "iLBC" ▼

choice 8: current setting is "iLBC" ▼

G723 rate: 6.3kbps encoding rate 5.3kbps encoding rate

iLBC frame size: 20ms 30ms

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

Silence Suppression: No Yes

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

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

Layer 2 QoS: 802.1Q/VLAN Tag 802.1p priority value (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: (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: (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: (default 5060)
 local RTP port: (1024-65535, default 5004)
 Use random port: No Yes
 NAT Traversal: No
 Yes, STUN server is: (URI or IP:port)
 keep-alive interval: (in seconds, default 20 seconds)
 Use NAT IP (if specified, this IP address is used in SIP/SDP message)
 TFTP Server: . . . (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:
 Send Flash Event: No Yes (Flash will be sent as a DTMF event if set to Yes)
 NTP Server: (URI or IP address)
 Time Zone:
 Year-Month-Day
 Date Display Format: 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)
 system ring tone
 Default 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)