

GrandStream HT-286 Configuration for freephoneline.ca

To access the web interface: dial '***' then '02' from the phone attached to the ATA.

Next, enter the IP address provided in your web browser. Login password is **admin**

Under the 'Advanced Settings 1' page, you will find all the settings necessary to make the HT-286 work with the freephoneline.ca services.

Grandstream Device Configuration

STATUS BASIC SETTINGS **ADVANCED SETTINGS 1** ADVANCED SETTINGS 2

Admin Password: (purposely not displayed for security protection)

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: (purposely not displayed for security protection)

Name: (optional, e.g., John Doe)

Home NPA:

Advanced Options:

Preferred Vocoder: (in listed order)

- choice 1:
- choice 2:
- choice 3:
- choice 4:
- choice 5:
- choice 6:
- choice 7:

G723 rate: ☒ 6.3kbps encoding rate ☐ 5.3kbps encoding rate

iLBC frame size: ☒ 20ms ☐ 30ms

iLBC payload type: (between 96 and 127, default is 97)

Silence Suppression: ☒ No ☐ Yes

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

Fax Mode: ☒ T.38 (Auto Detect) ☐ Pass-Through

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

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

SIP Server: voip.freephoneline.ca

SIP User ID: freephoneline number

Authenticate Password: SIP password

Name: Caller-ID Name (currently only supported on SIP to SIP calls)

Preferred Vocoder: Follow diagram above

Continued...

Allow incoming SIP messages from SIP proxy only: ☒ No ☐ Yes
 Use DNS SRV: ☒ No ☐ Yes
 User ID is phone number: ☒ No ☐ Yes
 SIP Registration: ☒ Yes ☐ No
 Unregister On Reboot: ☐ Yes ☒ No
 Register Expiration: 3600 (in seconds, default 1 hour, max 45 days)
 Early Dial: ☒ No ☐ Yes (use "Yes" only if proxy supports 484 response)
 Allow outgoing call without Registration: ☒ No ☐ Yes
 Dial Plan Prefix: (this prefix string is added to each dialed number)
 No Key Entry Timeout: 2 (in seconds, default is 4 seconds)
 Use # as Dial Key: ☐ No ☒ Yes (if set to Yes, "#" will function as the Dial key)
 local SIP port: 5060 (default 5060)
 local RTP port: 5004 (1024-65535, default 5004)
 Use random port: ☐ No ☒ Yes
 SIP Registration Failure Retry Wait Time: 20 (in seconds, Between 1-3600, default is 20)
 NAT Traversal: ☐ No
☒ Yes, STUN server is: (URI or IP:port)
 keep-alive interval: 20 (in seconds, default 20 seconds)
 Use NAT IP (used in SIP/SDP message if specified)
 Use STUN keep-alive to detect networks connectivity: ☒ No
☐ Yes, total STUN response misses (minimum=3) 5 before restart
 Proxy-Require:
 SUBSCRIBE for MWI: ☒ No, do not send SUBSCRIBE for Message Waiting Indication
☐ Yes, send periodical SUBSCRIBE for Message Waiting Indication
 Offhook Auto-Dial: (User ID/extension to dial automatically when offhook)
 Enable Call Features: ☐ No ☒ Yes
 (if yes, call features using star codes will be supported locally)
 Use Bell-style 3-way Conference: ☒ No ☐ Yes (if Yes, *23 will be disabled)
 Disable Call-Waiting: ☒ No ☐ Yes
 Disable Call-Waiting Caller-ID: ☒ 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)

Update Cancel Reboot

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Register Expiration: 3600 (usually default)

No Key Entry Timeout: 2 (number of seconds to wait before dialing after a number is entered)

Use random port: YES

Disable Call-Waiting Caller-ID: NO

Send DTMF: Check 'in-audio' and 'via RTP (RFC2833)'

PRESS UPDATE OR SETTINGS WILL NOT APPLY!