

Configuration pour GrandStream HT-286 sur freephoneline.ca

Pour accéder la page de web pour configure l'ATA: poussez sur l'étoile trois fois; '***' en suite '02' du telephone attacher a l'ATA. Après ca, ajouter l'adresse IP qui est donne sur la page web. Le mot de passe « Admin Password » est **admin**

Cliquez sur 'Advanced Settings 1', vous allez trouvez tout les options pour faire le HT-286 fonctionner avec freephoneline.ca

Grandstream Device Configuration

STATUS

BASIC SETTINGS

ADVANCED SETTINGS 1

ADVANCED SETTINGS 2

Admin Password:

(purposely not displayed for security protection)

SIP Server:

voip.freephoneline.ca

(e.g., sip.mycompany.com, or IP address)

Outbound Proxy:

(e.g., proxy.myprovider.com, or IP address, if any)

SIP User ID:

15198045747

(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:

FPL Display

(optional, e.g., John Doe)

Home NPA:

Advanced Options:

Preferred Vocoder:

(in listed order)

choice 1:

current setting is "PCMU"

choice 2:

current setting is "G729"

choice 3:

current setting is "PCMU"

choice 4:

current setting is "PCMU"

choice 5:

current setting is "PCMU"

choice 6:

current setting is "PCMU"

choice 7:

current setting is "PCMU"

G723 rate:

☒ 6.3kbps encoding rate

☐ 5.3kbps encoding rate

iLBC frame size:

☒ 20ms

☐ 30ms

iLBC payload type:

97

(between 96 and 127, default is 97)

Silence Suppression:

☒ No

☐ Yes

Voice Frames per TX:

2

(up to 10/20/32/64 for G711/G726/G723/other codecs respectively)

Fax Mode:

☒ T.38 (Auto Detect)

☐ Pass-Through

Layer 3 QoS:

48

(Diff-Serv or Precedence value)

Layer 2 QoS:

802.1Q/VLAN Tag

0

802.1p priority value

0

(0-7)

SIP Server: voip.freephoneline.ca

SIP User ID: Votre numéro de FreePhoneLine

Authenticate Password: Votre mot de passé SIP

Name: Le nom que vous voulez afficher sur le téléphone que vous appelez.

Preferred Vocoder: Suivez l'image sur dessus.

Continué...

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: (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: (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: (default 5060)
 local RTP port: (1024-65535, default 5004)
 Use random port: ☐ No ☒ Yes
 SIP Registration Failure Retry Wait Time: (in seconds. Between 1-3600, default is 20)
 NAT Traversal: ☐ No
☒ Yes, STUN server is: (URI or IP:port)
 keep-alive interval: (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) 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:
 Send Flash Event: ☒ No ☐ Yes (Flash will be sent as a DTMF event if set to Yes)

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Register Expiration: 3600 (Normalement choisis)

No Key Entry Timeout: 2 (montant de secondes avant que le numéro est appeler après l'avoir inscrit)

Use random port: YES

Disable Call-Waiting Caller-ID: NO

Send DTMF: Choisis 'in-audio' et 'via RTP (RFC2833)'

CLIQUEZ SUR 'UPDATE'! SI NON RIEN NE VA SAUVER!