Grandstream Device Configuration

STATUS | BASIC SETTINGS | ADVANCED SETTINGS | FXS PORT1 | FXS PORT2

Account Active:	○ No ◉ Yes
Primary SIP Server:	voip4.freephoneline.ca:6060 (e.g., sip.mycompany.com, or IP address)
Failover SIP Server:	(Optional, used when primary server no response)
Prefer Primary SIP Server:	◉ No ○ Yes (yes - will register to Primary Server if Failover registration expires)
Outbound Proxy:	(e.g., proxy.myprovider.com, or IP address, if any)
Allow DHCP Option 120(override SIP server):	◉ No ○ Yes
SIP Transport:	◉ UDP ○ TCP ○ TLS (default is UDP)
NAT Traversal:	○ No ◉ Keep-Alive ○ STUN ○ UPnP
SIP User ID:	[redacted] (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:	[redacted] (optional, e.g., John Doe)
DNS Mode:	◉ A Record ○ SRV ○ NAPTR/SRV
Tel URI:	Disabled
SIP Registration:	○ No ◉ Yes
Unregister On Reboot:	◉ No ○ Yes
Outgoing Call without Registration:	◉ No ○ Yes
Register Expiration:	60 (in minutes. default 1 hour, max 45 days)
Reregister before Expiration:	0 (in seconds. Default 0 second)
SIP Registration Failure Retry Wait Time:	20 (in seconds. Between 1-3600, default is 20)
SIP Registration Failure Retry Wait Time upon 403 Forbidden:	1200 (in seconds. Between 0-3600, default is 1200. 0 means stop retry registration upon 403 response.)
Enable SIP OPTIONS Keep Alive:	◉ No ○ Yes
SIP OPTIONS Keep Alive Interval:	30 (in seconds. Between 1-64800, default is 30)
SIP OPTIONS Keep Alive Max Lost:	3 (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3)
SIP OPTIONS Keep Alive Max Lost:	3 (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3)
Layer 3 QoS:	26 SIP DSCP (Diff-Serv value in decimal, default 24) 46 RTP DSCP (Diff-Serv value in decimal, default 46)
Local SIP port:	5060 (default is 5060 for UDP and TCP; 5061 for TLS)
Local RTP port:	5004 (even number between 1024-65535, default 5004)
Use Random SIP Port:	○ No ◉ Yes
Use Random RTP Port:	○ No ◉ Yes
Hold Target Before Refer:	○ No ◉ Yes
Refer-To Use Target Contact:	◉ No ○ Yes
Transfer on Conference Hangup:	○ No ◉ Yes
Disable Bellcore Style 3-Way Conference:	◉ No ○ Yes (Using star code *23 for 3-way conference)

Remove OBP from Route Header:	(•) No ( ) Yes
Support SIP Instance ID:	( ) No (•) Yes
Validate Incoming SIP Message:	(•) No ( ) Yes
Check SIP User ID for incoming INVITE:	(•) No ( ) Yes (no direct IP calling if Yes)
Authenticate incoming INVITE:	(•) No ( ) Yes
Authenticate server certificate domain:	(•) No ( ) Yes
Authenticate server certificate chain:	(•) No ( ) Yes
Trusted CA certificates:	
Allow Incoming SIP Messages from SIP Proxy Only:	( ) No (•) Yes (no direct IP calling if Yes)
Use Privacy Header:	(•) Default ( ) No ( ) Yes
Use P-Preferred-Identity Header:	(•) Default ( ) No ( ) Yes
SIP REGISTER Contact Header Uses:	(•) LAN Address ( ) WAN Address
SIP T1 Timeout:	0.5 sec
SIP T2 Interval:	4 sec
SIP Timer D:	0 (0 - 64 seconds. Default 0)
DTMF Payload Type:	101
Preferred DTMF method: (in listed order)	Priority 1: RFC2833 Priority 2: In-audio Priority 3: SIP INFO
Disable DTMF Negotiation:	(•) No (negotiate with peer) ( ) Yes (use above DTMF order without negotiation)
Generate Continuous RFC2833 Events:	(•) No ( ) Yes (RFC2833 events are generated until key is released)
Send Hook Flash Event:	(•) No ( ) Yes (Hook Flash will be sent as a DTMF event if set to Yes)
Flash Digit Control:	(•) No ( ) Yes (Overrides the default settings for call control when both channels are in use.)
Enable Call Features:	(•) No ( ) Yes (if Yes, call features using star codes will be supported locally)
Offhook Auto-Dial:	(User ID/extension to dial automatically when offhook)
Offhook Auto-Dial Delay:	0 (0-60 seconds, default is 0)
Proxy-Require:	
Use NAT IP:	(used in SIP/SDP message if specified)
Use SIP User-Agent Header:	
Distinctive Ring Tone:	Ring Tone 1 used if incoming caller ID is Ring Tone 1 used if incoming caller ID is Ring Tone 1 used if incoming caller ID is
Disable Call-Waiting:	(•) No ( ) Yes
Disable Call-Waiting Caller ID:	(•) No ( ) Yes
Disable Call-Waiting Tone:	(•) No ( ) Yes
Disable Connected Line ID:	(•) No ( ) Yes
Disable Receiver Offhook Tone:	(•) No ( ) Yes (ROH tone will not be played after offhook for 60 seconds)

Disable Reminder Ring for On-Hold Call:	(•) No ( ) Yes
Disable Visual MWI:	(•) No ( ) Yes
Do Not Escape '#' as %23 in SIP URI:	(•) No ( ) Yes
Disable Multiple m line in SDP:	(•) No ( ) Yes
Ring Timeout:	60 (10-300, default is 60 seconds)
Delayed Call Forward Wait Time:	20 (Allowed range 1-120, in seconds.)
No Key Entry Timeout:	4 (in seconds, default is 4 seconds)
Early Dial:	(•) No ( ) Yes (use "Yes" only if proxy supports 484 response)
Dial Plan Prefix:	(this prefix string is added to each dialed number)
Use # as Dial Key:	( ) No (•) Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
Dial Plan:	{ x+ | *x+ | *xx*x+ }
SUBSCRIBE for MWI:	(•) No, do not send SUBSCRIBE for Message Waiting Indication ( ) Yes, send periodical SUBSCRIBE for Message Waiting Indication
Send Anonymous:	(•) No ( ) Yes (caller ID will be blocked if set to Yes)
Anonymous Call Rejection:	(•) No ( ) Yes
Special Feature:	Standard
Session Expiration:	180 (in seconds. default 180 seconds)
Min-SE:	90 (in seconds. default and minimum 90 seconds)
Caller Request Timer:	(•) No ( ) Yes (Request for timer when making outbound calls)
Callee Request Timer:	(•) No ( ) Yes (When caller supports timer but did not request one)
Force Timer:	(•) No ( ) Yes (Use timer even when remote party does not support)
UAC Specify Refresher:	( ) UAC ( ) UAS (•) Omit (Recommended)
UAS Specify Refresher:	(•) UAC ( ) UAS (When UAC did not specify refresher tag)
Force INVITE:	(•) No ( ) Yes (Always refresh with INVITE instead of UPDATE)
Enable 100rel:	(•) No ( ) Yes
Add Auth Header On Initial REGISTER:	(•) No ( ) Yes
Conference URI:	
Use First Matching Vocoder in 200OK SDP:	(•) No ( ) Yes
Preferred Vocoder: *(in listed order)*	choice 1: PCMU choice 2: G729 choice 3: PCMU choice 4: PCMU choice 5: PCMU choice 6: PCMU
Voice Frames per TX:	2
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)
VAD:	(•) No ( ) Yes
Symmetric RTP:	(•) No ( ) Yes
Fax Mode:	(•) T.38 ( ) Pass-Through
Fax Tone Detection Mode:	(•) Caller ( ) Callee ( ) Caller or Callee
Re-INVITE After Fax Tone Detected:	(•) Enabled ( ) Disabled
Jitter Buffer Type:	( ) Fixed (•) Adaptive

Jitter Buffer Length:	○ Low ◉ Medium ○ High
SRTP Mode:	◉ Disabled ○ Enabled but not forced ○ Enabled and forced
Crypto Life Time:	○ Disabled ◉ Enabled
SLIC Setting:	USA 1 (BELLCORE 600 ohms)
Caller ID Scheme:	Bellcore/Telcordia
DTMF Caller ID:	*Start Tone* Default *Stop Tone* Default
Polarity Reversal:	◉ No ○ Yes (reverse polarity upon call establishment and termination)
Loop Current Disconnect:	◉ No ○ Yes (loop current disconnect upon call termination)
Loop Current Disconnect Duration:	200 (100 - 10000 milliseconds. Default 200 milliseconds)
Enable Pulse Dialing:	◉ No ○ Yes
Enable Hook Flash:	○ No ◉ Yes
Hook Flash Timing:	In 40-2000 milliseconds range, minimum: 300 maximum: 1100
On Hook Timing:	400 (In 40-2000 milliseconds range, default is 400)
Gain:	*TX* 0dB default *RX* -6dB default
Disable Line Echo Canceller (LEC):	◉ No ○ Yes
Disable Network Echo Suppressor:	◉ No ○ Yes
Outgoing Call Duration Limit:	0 (0-180 minutes, default is 0 (No Limit))
Ring Frequency:	20 (15-60 Hz, default is 20 Hz)
Enable High Ring Power:	◉ No ○ Yes

Ring Tones (Syntax: c=on1/off1-on2/off2-on3/off3;)

Ring Tone 1:	c=2000/4000;
Ring Tone 2:	c=2000/4000;
Ring Tone 3:	c=2000/4000;
Ring Tone 4:	c=2000/4000;
Ring Tone 5:	c=2000/4000;
Ring Tone 6:	c=2000/4000;
Ring Tone 7:	c=2000/4000;
Ring Tone 8:	c=2000/4000;
Ring Tone 9:	c=2000/4000;
Ring Tone 10:	c=2000/4000;

Update Apply Cancel Reboot