Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS FXS PORT FXO PORT

Account Active:	No Yes
Primary SIP Server:	192.168.1.40 (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)
SIP Transport:	UDP TCP TLS (default is UDP)
NAT Traversal:	No Keep-Alive STUN UPnP
SIP User ID:	7879469757 (the user part of an SIP address)
Authenticate ID:	7879469757 (can be identical to or different from **SIP User ID**)
Authenticate Password:	(purposely not displayed for security protection)
Name:	7879469757 (optional, e.g., John Doe)
DNS Mode:	A Record SRV NAPTR/SRV Use Configured IP
DNS SRV use Registered IP:	No Yes
Primary IP:	
Backup IP1:	
Backup IP2:	
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.)
Layer 3 QoS:	26 SIP DSCP (Diff-Serv value in decimal 0-63, default 26) 46 RTP DSCP (Diff-Serv value in decimal 0-63, default 46)
Local SIP port:	5062 (default 5062)
Local RTP port:	5012 (1024-65535, default 5012)
Use Random Port:	No Yes
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
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: SIP INFO Priority 3: In-audio
DTMF-RELAY Tag Respect SIP INFO:	(•) No ( ) Yes
Disable DTMF Negotiation:	(•) No (default, negotiate with peer) ( ) Yes (use above DTMF order without negotiation)
Proxy-Require:	
Use NAT IP:	(used in SIP/SDP message if specified)
Use SIP User-Agent Header:	
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)
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 "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
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)
INVITE Ring-No-Answer Timeout (sec):	40 (5-300 seconds. Default 40 seconds)
Enable 100rel:	(•) No ( ) Yes
Add Auth Header On Initial REGISTER:	(•) No ( ) Yes
Use First Matching Vocoder in 200OK SDP:	(•) No ( ) Yes
Preferred Vocoder: (in listed order)	choice 1: PCMU choice 2: PCMA choice 3: G723 choice 4: G729 choice 5: G726-32 choice 6: iLBC choice 7: G729E choice 8: AAL2-G726-16
Voice Frames per TX:	2 (default 2, from 1 to 4 for G711/G726/G729)
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)
AAL2-G726-16 Payload Type:	100 (between 96 and 127, default is 100)
AAL2-G726-24 Payload Type:	99 (between 96 and 127, default is 99)
AAL2-G726-32 payload type:	104 (between 96 and 127, default is 104)
AAL2-G726-40 Payload Type:	103 (between 96 and 127, default is 103)
G729E Payload Type:	102 (between 96 and 127, default is 102)
VAD:	(•) No ( ) Yes
Symmetric RTP:	(•) No ( ) Yes
Fax Mode:	(•) T.38 (Auto Detect) ( ) Pass-Through
Fax Tone Detection Mode:	( ) Caller (•) Callee ( ) Caller or Callee
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

Caller ID Scheme: Bellcore/Telcordia

FSK Caller ID Minimum RX Level (dB): -40 (-96 - 0dB. Default -40dB)

FSK Caller ID Seizure Bits: 70 (0 - 800 bits. Default 70)

FSK Caller ID Mark Bits: 40 (1 - 800 bits. Default 40)

Caller ID Transport Type: Relay via SIP From

Send Hook Flash To PSTN: ◉ No ○ Yes (If Yes, hook flash will be sent to PSTN upon receiving flash event from RFC2833 or SIP INFO)

Hook Flash Duration (ms): 600 (200 - 1500 milliseconds. Default 600)

Gain: *TX* 0dB default *RX* 0dB default

Disable Line Echo Canceller (LEC): ◉ No ○ Yes

FXO Termination

Enable Current Disconnect: ○ No ◉ Yes (Default Yes. If set to yes, enter threshold below)

Current Disconnect Threshold (ms): 100 (50-800 milliseconds. Default 100 milliseconds)

Enable PSTN Disconnect Tone Detection: ○ No ◉ Yes (Default No)

(If set to yes, the following tone is used as the disconnect signal)

PSTN Disconnect Tone: f1=480@-32,f2=620@-32,c=500/500;

(Syntax: f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3;)

(Allowed Range: freq = 0 to 4000Hz; vol = -40 to -24dBm)

(Default: Busy Tone: f1=480@-32,f2=620@-32,c=500/500;)

AC Termination Model ◉ Country-based ○ Impedance-based (Default Country-based)

Country-based USA

Impedance-based 600R -- 600 ohms

Number of Rings: 2 (1-50. Default 4)

(Number of rings for a PSTN incoming call before FXO port answers to accept VoIP number)

PSTN Ring Thru FXS: ◉ No ○ Yes (Default Yes)

(If set to yes, all incoming PSTN calls will ring the FXS port after the Ring Thru Delay)

PSTN Ring Thru Delay (sec): 4 (1-10 seconds. Default 4 seconds)

PSTN Ring Timeout (sec): 6 (2-10 seconds. Default 6 seconds)

(Used to detect PSTN hangup when FXO port is not answered)

PSTN Idle Wait Timeout between Outgoing Calls: 4 (0-10 seconds. Default 4 seconds)

Channel Dialing

DTMF Digit Length (ms): 100 (40-127 milliseconds, Default 100 milliseconds)

DTMF Dial Pause (ms): 100 (40-127 milliseconds, Default 100 milliseconds)

First Digit Timeout (sec): 10 (1-20 seconds. Default 10 seconds)

Inter-Digit Timeout (sec): 4 (1-15 seconds. Default 4 seconds)

Wait for Dial-Tone: ○ No ◉ Yes (Default Yes - dial upon dial-tone)

Stage Method (1/2): 1 (Default 2 - 2 stage dialing)

Min Delay Before Dial PSTN Number: 500 (default 500ms, range 50 ~ 65000ms)

Update Apply Cancel Reboot