

Grandstream Device Configuration

STATUS **BASIC SETTINGS** **ADVANCED SETTINGS** **FXS PORT1** **FXS PORT2**Account Active: ☐ No ☒ YesPrimary SIP Server: (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)Backup Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)Prefer Primary Outbound Proxy: ☒ No ☐ Yes (yes - will reregister via Primary Outbound Proxy if registration expires)Allow DHCP Option 120 (override SIP server): ☒ No ☐ YesSIP Transport: ☒ UDP ☐ TCP ☐ TLS (default is UDP)SIP URI Scheme When Using TLS: ☐ sip ☒ sipsUse Actual Ephemeral Port in Contact with TCP/TLS: ☒ No ☐ YesNAT Traversal: ☐ No ☒ Keep-Alive ☐ STUN ☐ UPnP ☐ VPNSIP 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)DNS Mode: ☒ A Record ☐ SRV ☐ NAPTR/SRV ☐ Use Configured IPDNS SRV use Registered IP: ☒ No ☐ YesDNS SRV Failover Mode: Failback Timer: (in minutes. default 60 minutes, max 45 days)Register Before DNS SRV Failover: ☒ No ☐ YesPrimary IP: Backup IP1: Backup IP2: Tel URI: SIP Registration: ☐ No ☒ YesUnregister On Reboot: ☒ No ☐ YesOutgoing Call without Registration: ☒ No ☐ YesRegister Expiration: (in minutes. default 1 hour, max 45 days)Reregister before Expiration: (0-64800. Default 0 second)SIP Registration Failure Retry Wait Time: (in seconds. Between 1-3600, default is 20)SIP Registration Failure: (in seconds. Between 0-3600, default is 1200. 0 means stop retry registration upon 403)

Retry Wait Time upon 403 response.)

Forbidden:

Enable SIP

OPTIONS/NOTIFY Keep ☐ No ☐ OPTIONS ☒ NOTIFY

Alive:

SIP OPTIONS/NOTIFY

Keep Alive Interval: (in seconds. Between 1-64800, default is 30)

SIP OPTIONS/NOTIFY

Keep Alive Max Lost: (Number of max lost packets for SIP OPTIONS/NOTIFY Keep Alive before re-registration. Between 3-10, default is 3)

Layer 3 QoS:

SIP DSCP (Diff-Serv value in decimal, 0-63, default 26)

RTP DSCP (Diff-Serv value in decimal, 0-63, default 46)

Local SIP Port:

(default is 5060 for UDP; 5061 for TLS)

Local RTP Port:

(even number between 1024-65535, default 5004)

Use Random SIP Port: ☐ No ☒ Yes

Use Random RTP Port: ☐ No ☒ Yes

Enable RTCP: ☐ No ☒ Yes

Hold Target Before Refer: ☐ No ☒ Yes

Refer-To Use Target

Contact: ☒ No ☐ Yes

Transfer on Conference

Hangup: ☐ 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

Allow Incoming SIP

Messages ☐ No ☒ Yes (no direct IP calling if Yes)

from SIP Proxy Only:

Use Privacy Header: ☒ Default ☐ No ☐ Yes

Use P-Preferred-Identity

Header: ☒ Default ☐ No ☐ Yes

Use P-Access-Network-Info

Header: ☐ No ☒ Yes

Use P-Emergency-Info

Header: ☐ No ☒ Yes

Use P-Asserted-Identity

Header: ☒ No ☐ Yes

SIP REGISTER Contact

Header Uses: ☐ LAN Address ☒ WAN Address

Caller ID Fetch Order:

☒ Auto ☐ Disabled ☐ From Header

Allow SIP Factory Reset:

☒ No ☐ Yes

Maximum Number of SIP Request Retries: (between 1 and 10, default is 4.)

SIP T1 Timeout:

SIP T2 Interval:

SIP Timer D: (0 - 64 seconds. Default 0)

DTMF Payload Type:

Preferred DTMF method (in listed order):
 Priority 1:
 Priority 2:
 Priority 3:

Inband DTMF Duration: In 40-2000 milliseconds range, duration: inter-duration:

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.)

Callee Flash to 3WC: ☒ No ☐ Yes

Offhook Auto-Dial: (User ID/extension to dial automatically when offhook)

Offhook Auto-Dial Delay: (0-60 seconds, default is 0)

Offhook Auto-Dial DTMF: (DTMF digits to send after offhook auto-dial call gets connected)

Proxy-Require:

Use NAT IP: (used in SIP/SDP message if specified)

SIP User-Agent:

SIP User-Agent Postfix:

RFC2543 Hold: ☐ No ☒ Yes

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 Reminder Ring for DND: ☒ No ☐ Yes

Disable Visual MWI: ☒ No ☐ Yes

Visual MWI Type: ☒ FSK ☐ NEON

Do Not Escape '#' as %23 in SIP URI: ☒ No ☐ Yes

Disable Multiple m line in SDP: ☒ No ☐ Yes

Ring Timeout: (0-300, default is 60 seconds, 0 means no timeout)

Delayed Call Forward Wait Time: (Allowed range 1-120, in seconds.)

No Key Entry Timeout: (1-15, 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)

Disable # as Redial Key: ☒ No ☐ Yes (if set to Yes, "#" will not function as ReDial key)

Dial Plan:

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:

Enable Session Timer: ☐ No ☒ Yes

Session Expiration: (90-64800. default 180 seconds)

Min-SE: (90-64800. default 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: | <input type="text" value="PCMU"/> |
| choice 2: | <input type="text" value="G729"/> |
| choice 3: | <input type="text" value="PCMU"/> |
| choice 4: | <input type="text" value="PCMU"/> |
| choice 5: | <input type="text" value="PCMU"/> |
| choice 6: | <input type="text" value="PCMU"/> |
| choice 7: | <input type="text" value="PCMU"/> |
| choice 8: | <input type="text" value="PCMU"/> |

Voice Frames per TX:

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

iLBC Frame Size: ☒ 20ms ☐ 30ms

Disable OPUS Stereo in SDP: ☒ No ☐ Yes (removes "/2" from offer)

iLBC Payload Type: (between 96 and 127, default is 97)

OPUS Payload Type: (between 96 and 127, default is 123)

VAD: ☒ No ☐ Yes

Symmetric RTP: ☒ No ☐ Yes

Fax Mode: ☒ T.38 ☐ Pass-Through

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 ▼

Disable Unknown Caller ID: ☒ No ☐ Yes

Replace Beginning '+' with 00 in Caller ID: ☒ No ☐ Yes

Number of Beginning Digits to Strip from Caller ID: 0 (between 0 and 10, default is 0.)

Polarity Reversal: ☒ No ☐ Yes (reverse polarity upon call establishment and termination)

Loop Current Disconnect: ☒ No ☐ Yes (loop current disconnect upon call termination)

Play busy/reorder tone before Loop Current Disconnect: ☒ No ☐ Yes (play busy/reorder tone before loop current disconnect upon call fail)

Loop Current Disconnect Duration: 200 (100 - 10000 milliseconds. Default 200 milliseconds)

Enable Pulse Dialing: ☒ No ☐ Yes

Pulse Dialing Standard: General Standard ▼

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: 20Hz default ▼

Enable High Ring Power: ☒ No ☐ Yes

OnHook DC Feed Current: 30mA default ▼

RFC2833 Events Count: 8 (between 2 and 10, default is 8)

RFC2833 End Events Count: 3 (between 2 and 10, default is 3)

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

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:

Ring Tone 8:

Ring Tone 9:

Ring Tone 10:

Distinctive Call Waiting Tone: used if incoming caller ID is

used if incoming caller ID is

used if incoming caller ID is

Call Waiting Tones

Syntax: f1=**val**[, f2=**val**[, c=**on1/off1**[-**on2/off2**[-**on3/off3**]]]];
 (Frequencies are in (10, 4000) Hz and cadence on and off are in (0, 64000) ms)

Call Waiting Tone 1:

Call Waiting Tone 2:

Call Waiting Tone 3:

Call Waiting Tone 4:

Call Waiting Tone 5:

Call Waiting Tone 6:

Call Waiting Tone 7:

Call Waiting Tone 8:

Call Waiting Tone 9:

Call Waiting Tone 10:

Call Features Settings

Enable Call Features: ☒ No ☐ Yes ☐ Enable All

(When enabled, Do No Disturb, Call Forward and other call features can be used via the local feature codes on the phone. Otherwise, the ITSP feature codes will be used. Enable All will override all individual features enable setting.)

Reset Call Features: ☒ No ☐ Yes

SRTP Feature: ☐ No ☒ Yes

Enable SRTP:

Disable SRTP:

SRTP per call Feature: ☐ No ☒ Yes

Enable SRTP per call:

Disable SRTP per call:

CID Feature: ☐ No ☒ Yes

Enable CID:

Disable CID:

CID per call Feature: ☐ No ☒ Yes

Enable CID per call:

Disable CID per call:

Direct IP Calling Feature: ☐ No ☒ Yes

Direct IP Calling:

CW Feature: ☐ No ☒ Yes

Enable CW:

Disable CW:

CW per call Feature: ☐ No ☒ Yes

Enable CW per call:

Disable CW per call:

Call Return Feature: ☐ No ☒ Yes

Call Return:

Unconditional Forward
Feature: ☐ No ☒ Yes

Enable Unconditional
Forward:

Disable Unconditional
Forward:

Busy Forward Feature: ☐ No ☒ Yes

Enable Busy Forward:

Disable Busy Forward:

Delayed Forward Feature: ☐ No ☒ Yes

Enable Delayed Forward:

Disable Delayed Forward:

Paging Feature: ☐ No ☒ Yes

Paging:

DND Feature: ☐ No ☒ Yes

Enable DND:

Disable DND:

Blind Transfer Feature: ☐ No ☒ Yes

Enable Blind Transfer:

Disable LEC per call
Feature: ☐ No ☒ Yes

Disable LEC per call:

Disable Bellcore Style 3-
Way Conference: ☒ No ☐ Yes

Star Code 3WC Feature: ☐ No ☒ Yes

Star Code 3WC:

Forced Codec Feature: ☐ No ☒ Yes

Forced Codec:

PCMU Codec Feature: ☐ No ☒ Yes

PCMU Codec:

PCMA Codec Feature: ☐ No ☒ Yes

PCMA Codec:

G723 Codec Feature: ☐ No ☒ Yes

G723 Codec:

G729 Codec Feature: ☐ No ☒ Yes

G729 Codec:

iLBC Codec Feature: ☐ No ☒ Yes

iLBC Codec:

G722 Codec Feature: ☐ No ☒ Yes

G722 Codec:

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