

Grandstream Device Configuration

STATUS **BASIC SETTINGS** **ADVANCED SETTINGS** **FXS PORT1** **FXS PORT2****Account Active:** No Yes**Primary 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)**SIP transport:** UDP TCP TLS (default is UDP)**NAT Traversal (STUN):** No No, but send keep-alive Yes**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)**DNS Mode:** A Record SRV NAPTR/SRV**Tel URI:** **SIP Registration:** No Yes**Unregister On Reboot:** No Yes**Outgoing Call without Registration:** No Yes**Register Expiration:** (in minutes. default 1 hour, max 45 days)**SIP Registration Failure Retry Wait Time:** (in seconds. Between 1-3600, default is 20)**Local SIP Port:** (default 5062)**Local RTP Port:** (1024-65535, default 5012)**Use Random Port:** No Yes**Refer-To Use Target Contact:** No Yes**Transfer on Conference Hangup:** No Yes**Enable Ring-Transfer:** No (RFC5589 Semi-Attended Transfer) 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)**Allow Incoming SIP Messages from SIP Proxy Only:** No Yes (no direct IP calling if Yes)**SIP T1 Timeout:** **SIP T2 Interval:** **DTMF Payload Type:** **Preferred DTMF method:** Priority 1: (in listed order) Priority 2: Priority 3: **Disable DTMF Negotiation:** No (negotiate with peer) Yes (use above DTMF order without negotiation)**Send Hook Flash Event:** No Yes (Hook Flash will be sent as a DTMF event if set to Yes)**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)**Proxy-Require:** **Use NAT IP:** (used in SIP/SDP message if specified)

Distinctive Ring Tone:	Ring Tone 1	used if incoming caller ID is	<input type="text"/>
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Disable Call-Waiting: No Yes
 Disable Call-Waiting Caller ID: No Yes
 Disable Call-Waiting Tone: 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
 Ring Timeout: (10-300, default is 60 seconds)
 Delayed Call Forward Wait Time: (Allowed range 1-120, in seconds.)
 No Key Entry Timeout: (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:
 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:
 Session Expiration: (in seconds. default 180 seconds)
 Min-SE: (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)
 Send Re-INVITE After Fax: No Yes
 Enable Silence Detection for Fax Disconnect: No Yes
 Enable 100rel: 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: (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: (between 96 and 127, default is 97)
 G726-32 Payload Type: (between 96 and 127, default is 112)
 AAL2-G726-16 Payload Type: (between 96 and 127, default is 100)
 AAL2-G726-24 Payload Type: (between 96 and 127, default is 99)
 AAL2-G726-32 Payload Type: (between 96 and 127, default is 104)
 AAL2-G726-40 Payload Type: (between 96 and 127, default is 103)
 G729E Payload Type: (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

SLIC Setting:

Caller ID Scheme:

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: (In 100 - 10000 milliseconds range, default is 200)

Hook Flash Timing: In 40-2000 milliseconds range, minimum: maximum:

On Hook Timing: (In 40-2000 milliseconds range, default is 400)

Gain: TX RX

Disable Line Echo Cancellor (LEC): No Yes

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

Ring Tone 1:

Ring Tone 2:

Ring Tone 3:

Ring Tone 4:

Ring Tone 5:

Ring Tone 6:

Ring Tone 7:

Ring Tone 8:

Ring Tone 9:

Ring Tone 10:

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