Grandstream Device Configuration

STATUS **BASIC SETTINGS** **ADVANCED SETTINGS** **PROFILE 1** **PROFILE 2** **FXS PORTS**

Profile Active:	○ No ◉ Yes
Primary SIP Server:	tel.t-online.de (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:	tel.t-online.de (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 ○ Yes
SIP Transport:	◉ UDP ○ TCP ○ TLS (default is UDP)
SIP URI Scheme When Using TLS:	○ sip ◉ sips
Use Actual Ephemeral Port in Contact with TCP/TLS:	◉ No ○ Yes
NAT Traversal:	◉ No ○ Keep-Alive ○ STUN ○ UPnP ○ VPN
DNS Mode:	○ A Record ◉ SRV ○ NAPTR/SRV ○ Use Configured IP
DNS SRV use Registered IP:	◉ No ○ Yes
DNS SRV Failover Mode:	Default
Failback Timer:	60 (in minutes. default 60 minutes, max 45 days)
Register Before DNS SRV Failover:	◉ No ○ Yes
Primary IP:	
Backup IP1:	
Backup IP2:	
Tel URI:	Disabled
Use Request Routing ID in SIP INVITE Header:	◉ No ○ Yes
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 (0-64800. 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/NOTIFY Keep Alive:	◉ No ○ OPTIONS ○ NOTIFY
SIP OPTIONS/NOTIFY Keep Alive Interval:	30 (in seconds. Between 1-64800, default is 30)
SIP OPTIONS/NOTIFY Keep Alive Max Lost:	3 (Number of max lost packets for SIP OPTIONS/NOTIFY Keep Alive before re-registration. Between 3-10, default is 3)
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:	5060 (default is 5060 for UDP; 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
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 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
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:	4 (between 1 and 10, default is 4.)
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
Inband DTMF Duration:	In 40-2000 milliseconds range, duration: 100 inter-duration: 50
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 Delay:	0 (0-60 seconds, default is 0)
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:	60 (0-300, default is 60 seconds, 0 means no timeout)
Hunting Group Ring Timeout:	20 (5-300, default is 20 seconds)
Hunting Group Type:	● Circular ○ Linear
Delayed Call Forward Wait Time:	20 (Allowed range 1-120, in seconds.)
No Key Entry Timeout:	4 (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:	{ x+ | \+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
Enable Session Timer:	○ No ● Yes
Session Expiration:	180 (90-64800. default 180 seconds)
Min-SE:	90 (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: PCMU choice 2: PCMA

	choice 3: G723
	choice 4: G729
	choice 5: G726-32
	choice 6: iLBC
	choice 7: OPUS
	choice 8: G722
Voice Frames per TX:	2
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:	97 (between 96 and 127, default is 97)
OPUS Payload Type:	123 (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:	GERMANY
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
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:	c=2000/4000;
Ring Tone 8:	c=2000/4000;
Ring Tone 9:	c=2000/4000;
Ring Tone 10:	c=2000/4000;

Distinctive Call Waiting Tone:	Call Waiting Tone 1	used if incoming caller ID is
	Call Waiting Tone 1	used if incoming caller ID is
	Call Waiting Tone 1	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:	f1=440@-13,c=300/10000;
Call Waiting Tone 2:	f1=440@-13,c=300/10000;
Call Waiting Tone 3:	f1=440@-13,c=300/10000;
Call Waiting Tone 4:	f1=440@-13,c=300/10000;
Call Waiting Tone 5:	f1=440@-13,c=300/10000;
Call Waiting Tone 6:	f1=440@-13,c=300/10000;
Call Waiting Tone 7:	f1=440@-13,c=300/10000;
Call Waiting Tone 8:	f1=440@-13,c=300/10000;
Call Waiting Tone 9:	f1=440@-13,c=300/10000;
Call Waiting Tone 10:	f1=440@-13,c=300/10000;

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:	16
Disable SRTP:	17

SRTP per call Feature:	○ No ◉ Yes
Enable SRTP per call:	18
Disable SRTP per call:	19

CID Feature:	○ No ◉ Yes
Enable CID:	31
Disable CID:	30

CID per call Feature:	○ No ◉ Yes
Enable CID per call:	82

Disable CID per call:	67

Direct IP Calling Feature:	○ No ● Yes
Direct IP Calling:	47

CW Feature:	○ No ● Yes
Enable CW:	51
Disable CW:	50

CW per call Feature:	○ No ● Yes
Enable CW per call:	71
Disable CW per call:	70

Call Return Feature:	○ No ● Yes
Call Return:	69

Unconditional Forward Feature:	○ No ● Yes
Enable Unconditional Forward:	72
Disable Unconditional Forward:	73

Busy Forward Feature:	○ No ● Yes
Enable Busy Forward:	90
Disable Busy Forward:	91

Delayed Forward Feature:	○ No ● Yes
Enable Delayed Forward:	92
Disable Delayed Forward:	93

Paging Feature:	○ No ● Yes
Paging:	74

DND Feature:	○ No ● Yes
Enable DND:	78
Disable DND:	79

Blind Transfer Feature:	○ No ● Yes
Enable Blind Transfer:	87

Disable LEC per call Feature:	○ No ● Yes
Disable LEC per call:	03

Disable Bellcore Style 3-Way Conference:	● No ○ Yes
Star Code 3WC Feature:	○ No ● Yes
Star Code 3WC:	23

Forced Codec Feature:	○ No ● Yes
Forced Codec:	02
PCMU Codec Feature:	○ No ● Yes

PCMU Codec:	7110	
PCMA Codec Feature:	○ No	◉ Yes
PCMA Codec:	7111	
G723 Codec Feature:	○ No	◉ Yes
G723 Codec:	723	
G729 Codec Feature:	○ No	◉ Yes
G729 Codec:	729	
iLBC Codec Feature:	○ No	◉ Yes
iLBC Codec:	7201	
G722 Codec Feature:	○ No	◉ Yes
G722 Codec:	722	

Update Apply Cancel Reboot