Grandstream's Device Configuration to use SiP Micro's SIP VoIP Account

Grandstream Device Configuration

STATUS | BASIC SETTINGS | ADVANCED SETTINGS

Admin Password:		(purposely not displayed for security protection)
FXS Port :		
SIP Server:	SIP3.SIPTELGK.NET	(e.g., sip.mycompany.com, or IP address)
Outbound Proxy:	SIP3.SIPTELGK.NET	(e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID:	Enter given Account number	(the user part of an SIP address)
Authenticate ID:	Enter given Account number	(can be identical to or different from SIP User ID)
Authenticate Password:	Enter given Account number	(purposely not displayed for security protection)
Name:		(optional, e.g., John Doe)
Register Expiration:	60	(in minutes. default 1 hour, max 45 days)
local SIP port:	5060	(default 5060)
local RTP port:	5004	(1024-65535, default 5004)
Enable Call Features:	○ No ⦿ Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally)	
Send DTMF:	⦿ in-audio ○ via RTP (RFC2833) ○ via SIP INFO	
DTMF Payload Type:	101	
FXO Port :		
SIP Server:		(e.g., sip.mycompany.com, or IP address)
Outbound Proxy:		(e.g., proxy.myprovider.com, or IP address, if any)
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)
Register Expiration:	60	(in minutes. default 1 hour, max 45 days)
local SIP port:	5062	(default 5062)
local RTP port:	5008	(1024-65535, default 5008)
Send DTMF:	⦿ in-audio ○ via RTP (RFC2833) ○ via SIP INFO	
DTMF Payload Type:	101	
General Misc. Settings:		
Preferred Vocoder: (in listed order)	choice 1: current setting is " G723" choice 2: current setting is " PCMA" choice 3: current setting is " G723" choice 4: current setting is " G729" choice 5: current setting is " G726-32" choice 6: current setting is " iLBC"	
Voice Frames per TX:	2	(up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
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)
Silence Suppression:	○ No ⦿ Yes	
SIP Registration:	⦿ Yes ○ No	
Unregister On Reboot:	○ Yes ⦿ No	
Fax Mode:	⦿ T.38 (Auto Detect) ○ Pass-Through	
Layer 3 QoS:	48	(Diff-Serv or Precedence value)
Layer 2 QoS:	802.1Q/VLAN Tag 0 802.1p priority value 0 (0-7)	
Use DNS SRV:	⦿ No ○ Yes	
User ID is phone number:	⦿ No ○ Yes	
Early Dial:	⦿ No ○ Yes (use "Yes" only if proxy supports 484 response)	
Dial Plan Prefix:		(this prefix string is added to each dialed number)
No Key Entry Timeout:	4	(in seconds, default is 4 seconds)
Use # as Dial Key:	○ No ⦿ Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)	
Use random port:	⦿ No ○ Yes	
NAT Traversal:	○ No ⦿ Yes, STUN server is:	(URI or IP:port)
keep-alive interval:	20	(in seconds, default 20 seconds)
Use NAT IP		(if specified, this IP address is used in SIP/SDP message)
Proxy-Require:		(if specified, the content will appear in Proxy-Require header)
TFTP Upgrade Server:	168 . 75 . 215 . 188	(for remote software upgrade and configuration)
SUBSCRIBE for MWI:	⦿ No, do not send SUBSCRIBE for Message Waiting Indication ○ Yes, send periodical SUBSCRIBE for Message Waiting Indication	
Offhook Auto-Dial:		(User ID/extension to dial automatically when offhook)
Send Flash Event:	⦿ No ○ Yes (Flash will be sent as a DTMF event if set to Yes)	
FXS Impedance:	current setting is " 600 Ohm (North America)"	
Caller ID Scheme:	current setting is " Bellcore"	
Onhook Voltage:	current setting is " 48V"	
Disable Call-Waiting:	⦿ No ○ Yes	
Polarity Reversal:	⦿ No ○ Yes (reverse polarity upon call establishment and termination)	
NTP Server:	time.nist.gov	(URI or IP address)
Send Anonymous:	⦿ No ○ Yes (caller ID will be blocked if set to Yes)	
Lock keypad update:	⦿ No ○ Yes (configuration update via keypad is disabled if set to Yes)	
WAN side http access:	○ No ⦿ Yes (WAN side access to http server will be rejected if set to No)	
FXS phone line failover:	⦿ No ○ Yes (Phone automatically connects to PSTN when Ethernet down or lose SIP registration if set to Yes. No IVR available if set to Yes)	

Update | Cancel | Reboot