

Grandstream Device Configuration

STATUS	BASIC SETTINGS	ADVANCED SETTINGS
Admin Password:	<input type="text"/>	(purposely not displayed for security protection)
SIP Server:	<input type="text" value="85.119.188.3"/>	(e.g., sip.mycompany.com, or IP address)
Outbound Proxy:	<input type="text" value="85.119.188.3"/>	(e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID:	<input type="text" value="02xxxxxxx"/>	(the user part of an SIP address)
Authenticate ID:	<input type="text" value="02xxxxxxx"/>	(can be identical to or different from SIP User ID)
Authenticate Password:	<input type="text"/>	(purposely not displayed for security protection)
Name:	<input type="text"/>	(optional, e.g., John Doe)

Advanced Options:

Preferred Vocoder: (in listed order)	choice 1: <input type="checkbox"/> current setting is "PCMU"	<input type="checkbox"/>
	choice 2: <input type="checkbox"/> current setting is "PCMA"	<input type="checkbox"/>
	choice 3: <input type="checkbox"/> current setting is "G723"	<input type="checkbox"/>
	choice 4: <input type="checkbox"/> current setting is "G729"	<input type="checkbox"/>
	choice 5: <input type="checkbox"/> current setting is "G726-32"	<input type="checkbox"/>
	choice 6: <input type="checkbox"/> current setting is "iLBC"	<input type="checkbox"/>
	choice 7: <input type="checkbox"/> current setting is "G722"	<input type="checkbox"/>
	choice 8: <input type="checkbox"/> current setting is "PCMU"	<input type="checkbox"/>
G723 rate:	<input checked="" type="radio"/> 6.3kbps encoding rate	<input type="radio"/> 5.3kbps encoding rate
iLBC frame size:	<input checked="" type="radio"/> 20ms	<input type="radio"/> 30ms
iLBC payload type:	<input type="text" value="97"/>	(between 96 and 127, default is 97)
Silence Suppression:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Voice Frames per TX:	<input type="text" value="4"/>	(up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
Layer 3 QoS:	<input type="text" value="48"/>	(Diff-Serv or Precedence value)
Layer 2 QoS:	802.1Q/VLAN Tag <input type="text" value="0"/>	802.1p priority value <input type="text" value="0"/> (0-7)
Allow incoming SIP messages from SIP proxy only:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Use DNS SRV:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
User ID is phone number:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
SIP Registration:	<input checked="" type="radio"/> Yes	<input type="radio"/> No
Unregister On Reboot:	<input type="radio"/> Yes	<input checked="" type="radio"/> No
Register Expiration:	<input type="text" value="300"/>	(in seconds, default 1 hour, max 45 days)
Early Dial:	<input checked="" type="radio"/> No	<input type="radio"/> Yes (use "Yes" only if proxy supports 484 response)
Allow outgoing call without Registration:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Dial Plan Prefix:	<input type="text"/>	(this prefix string is added to each dialed number)
No Key Entry Timeout:	<input type="text" value="4"/>	(in seconds, default is 4 seconds)
Use # as Dial Key:	<input type="radio"/> No	<input checked="" type="radio"/> Yes (if set to Yes, "#" will function as the Dial key)
local SIP port:	<input type="text" value="5060"/>	(default 5060)
local RTP port:	<input type="text" value="5004"/>	(1024-65535, default 5004)
Use random port:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
NAT Traversal:	<input type="radio"/> No	<input checked="" type="radio"/> Yes, STUN server is: <input type="text" value="stun01.sipphone.com:3478"/> (URI or IP:port)
keep-alive interval:	<input type="text" value="20"/>	(in seconds, default 20 seconds)
Use NAT IP	<input type="text"/>	(if specified, this IP address is used in SIP/SDP message)

Proxy-Require: (if specified, the content will appear in Proxy-Require header)

Voice Mail UserID: (User ID/extension for 3rd party voice mail system)

SUBSCRIBE for MWI: No, do not send SUBSCRIBE for Message Waiting Indication
 Yes, send periodical SUBSCRIBE for Message Waiting Indication

Auto Answer: No Yes

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

Enable Call Features: No Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally)

Disable Call-Waiting: No Yes

Send DTMF: in-audio via RTP (RFC2833) via SIP INFO

DTMF Payload Type:

Send Flash Event: No Yes (Flash will be sent as a DTMF event if set to Yes)

Onhook Threshold:

NTP Server: (URI or IP address)

Default Ring Tone: system ring tone
 custom ring tone 1, used if incoming caller ID is
 custom ring tone 2, used if incoming caller ID is
 custom ring tone 3, used if incoming caller ID is

Send Anonymous: No Yes (caller ID will be blocked if set to Yes)

Anonymous Method: Use From Header Use Privacy Header

Special Feature:

Syslog Server:

Syslog Level:

Firmware Upgrade and Provisioning: Upgrade Via TFTP HTTP
 Firmware Server Path:
 Config Server Path:
 Firmware File Prefix: Firmware File Postfix:
 Config File Prefix: Config File Postfix:

Automatic Upgrade:
 No Yes, check for upgrade every minutes (default 7 days)
 Always Check for New Firmware
 Check New Firmware only when F/W pre/suffix changes
 Always Skip the Firmware Check

Firmware Key: (in Hexadecimal Representation)

Authenticate Conf File: No Yes (cfg file would be authenticated before acceptance if set to Yes)

Lock keypad update: No Yes (configuration update via keypad is disabled if set to Yes)

Allow conf SIP Account in Basic Settings: No Yes

Override MTU Size: