Tiedosto Muokkaa Näytä Sivuhistoria Kirjanmerkit Työkalut Ohje

PortaBilling -- New Installa... | Grandstream Device Configura...

192.168.0.2/cgi-bin/config

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System Ring Cadence:	c=2000/4000;	
Call Progress Tones:	*Dial Tone:*	f1=350@-13,f2=440@-13,c=0/0;
	Ringback Tone:	f1=440@-19,f2=480@-19,c=2000/4000;
	Busy Tone:	f1=480@-24,f2=620@-24,c=500/500;
	Reorder Tone:	f1=480@-24,f2=620@-24,c=250/250;
	Confirmation Tone:	f1=350@-11,f2=440@-11,c=100/100-100/100-100/100;
	Call Waiting Tone:	f1=440@-13,c=300/10000-300/10000-0/0;
	Prompt Tone:	f1=350@-13,f2=440@-13,c=0/0;
	Syntax: `f1=val[,f2=val[,c=on1/off1[-on2/off2[-on3/off3]]]];` (Frequencies are in Hz and cadence on and off are in ms)	
Prompt Tone Access Code:	(Key pattern to get Prompt Tone. Maximum 20 digits. No default.)	
Phone LED Pattern:	(•) Pattern A (Onhook: OFF; Offhook: ON; VM: Blink) ( ) Pattern B (Onhook: ON; Offhook: OFF; VM: Blink if phone onhook)	
Lock Keypad Update:	(•) No ( ) Yes (configuration update via keypad is disabled if set to Yes)	
Disable Voice Prompt:	(•) No ( ) Yes (voice prompt is disabled if set to Yes)	
Disable Direct IP Call:	( ) No (•) Yes (direct IP call is disabled if set to Yes)	
NTP Server:	us.pool.ntp.org (URI or IP address)	
NTP Update Interval:	60 minutes(5-1440)	
Syslog Server:		
Syslog Level:	NONE	
Send SIP Log:	(•) No ( ) Yes	
Download Device Configuration:	Download	
Upload firmware:	Upload from local directory	
Upload configuration:	Upload from local directory	

Update Apply Cancel Reboot

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PortaBilling -- New Installa... | Grandstream Device Configura...

192.168.0.2/cgi-bin/config_a1

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SIP Registration Failure Retry Wait Time:	20 (in seconds. Between 1-3600, default is 20)
Local SIP Port:	5060 (default is 5060 for UDP and TCP; 5061 for TLS)
Local RTP Port:	5004 (1024-65535, default 5004)
Use Random Port:	◉ No ○ Yes
Refer-To Use Target Contact:	◉ No ○ Yes
Transfer on Conference Hangup:	◉ No ○ 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)
Authenticate incoming INVITE:	◉ 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
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: In-audio Priority 2: In-audio Priority 3: In-audio
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)
Offhook Auto-Dial Delay:	0 (0-60 seconds, default is 0)
Proxy-Require:	

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