Third-Party Certified Equipment

Supplemental Information

Grandstream DP715

Certification Completed On:
5/9/2016

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Grandstream DP715

Software Version 1.0.33

The Grandstream DP 715 phone is a cordless phone, communicating with its base via standard DECT technology. This is an alternative SIP/VOIP solution for customers who do not want to allocate wiring and an FXS port to support an analog DECT phone. Also, Analog DECT phones are not able to make use of functionality like transfer and conference. The Grandstream is able to signal events natively via SIP, enabling it to better express a more comprehensive feature set. A single DP715 base station can register up to 5 DECT handsets—allowing up to 4 headsets to talk concurrently. Please note, this does not require wireless engineering (802.11a/b/g/n, etc…), as the unit communicates via DECT to its base unit, and then via SIP from the base unit to CIC. That also implies that this unit will be limited to a range that is tied to the location of its base unit, as opposed to 802.11a/b/g/n units that can communicate anywhere a wireless signal is available. This should also imply a high level of voice quality, as DECT is better able to handle wireless call quality without extensive 802.11 engineering or capabilities (using its own protocols), and the base station is wired directly into the network via LAN port.

Each handset can be assigned a different extension and SIP User ID, allowing it to be bound to a different station in IA. The DP715 provisioning process allows a base unit to support up to 2 different “types” of registration—allowing some phones to be provisioned to one set of SIP Endpoints (say, the CIC servers), and other phones provision to a different SIP Endpoint.

Each handset can be provisioned with a single extension (SIP User ID). Use of Station Groups for group rings and similar items is a good idea.

Certification of this unit focused on the following items:

- Inbound and outbound calling (as a basic station, and as a station for a user logged into Interaction Client)
- Conferencing through the phone’s built in 3-party conferencing, and through Client’s Ad-hoc conferencing
- Blind Transfers through the phone (sending the call back to CIC and dropping the phone out)
- Conference transfers through the phone (using the phone’s built-in 3 party conferencing, and through Client’s Warm/Conference transfer functionality)
- G.711 and G.729 voice codecs
- Inband and out of band DTMF, including the use of dynamic audio
- Phone’s built in speed dial functionality
- Delayed media and normal media timing

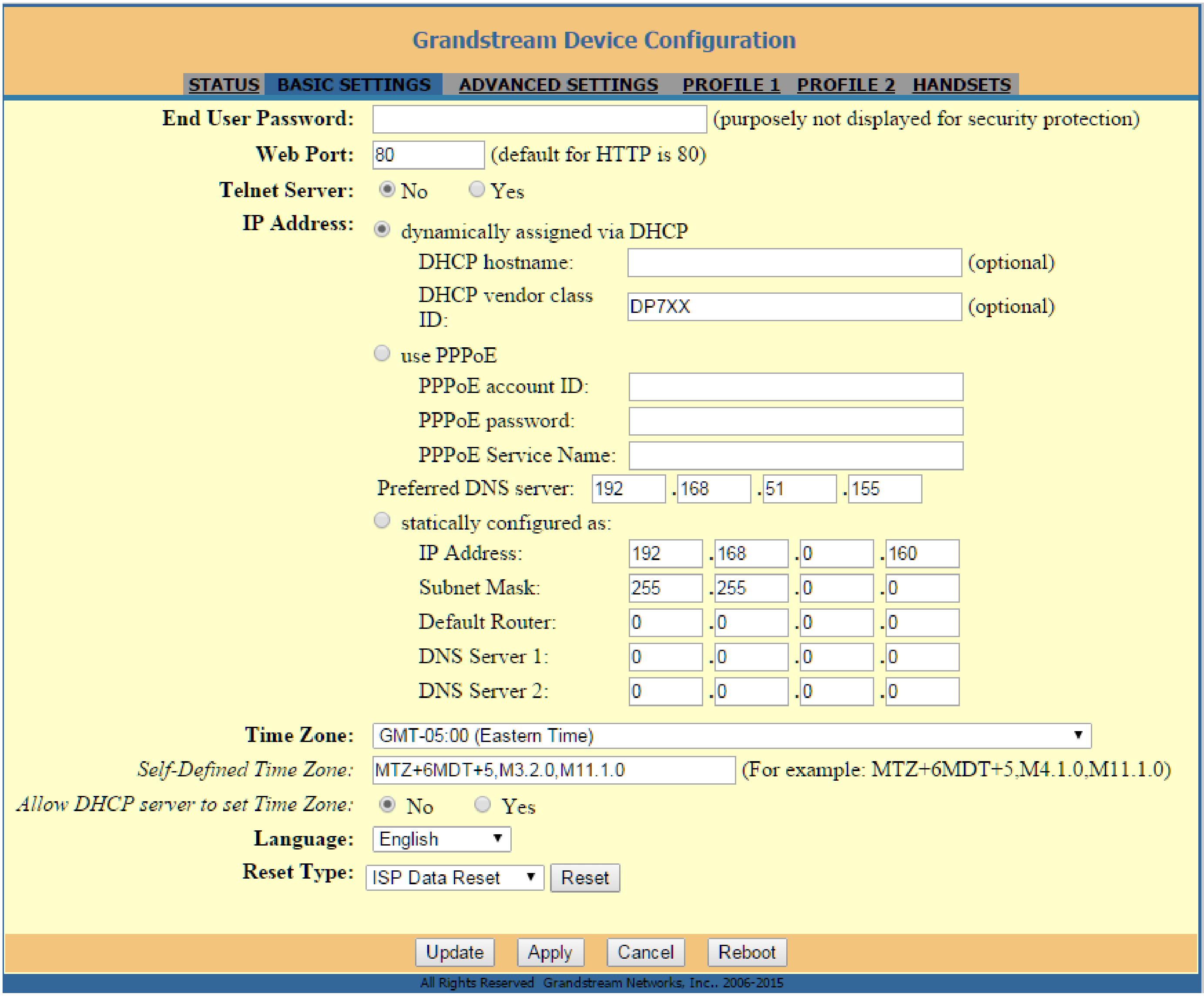
- Switchover between active and backup CIC server pairs
- UDP and TCP SIP Signaling
- Auto Answer on SIP Phones

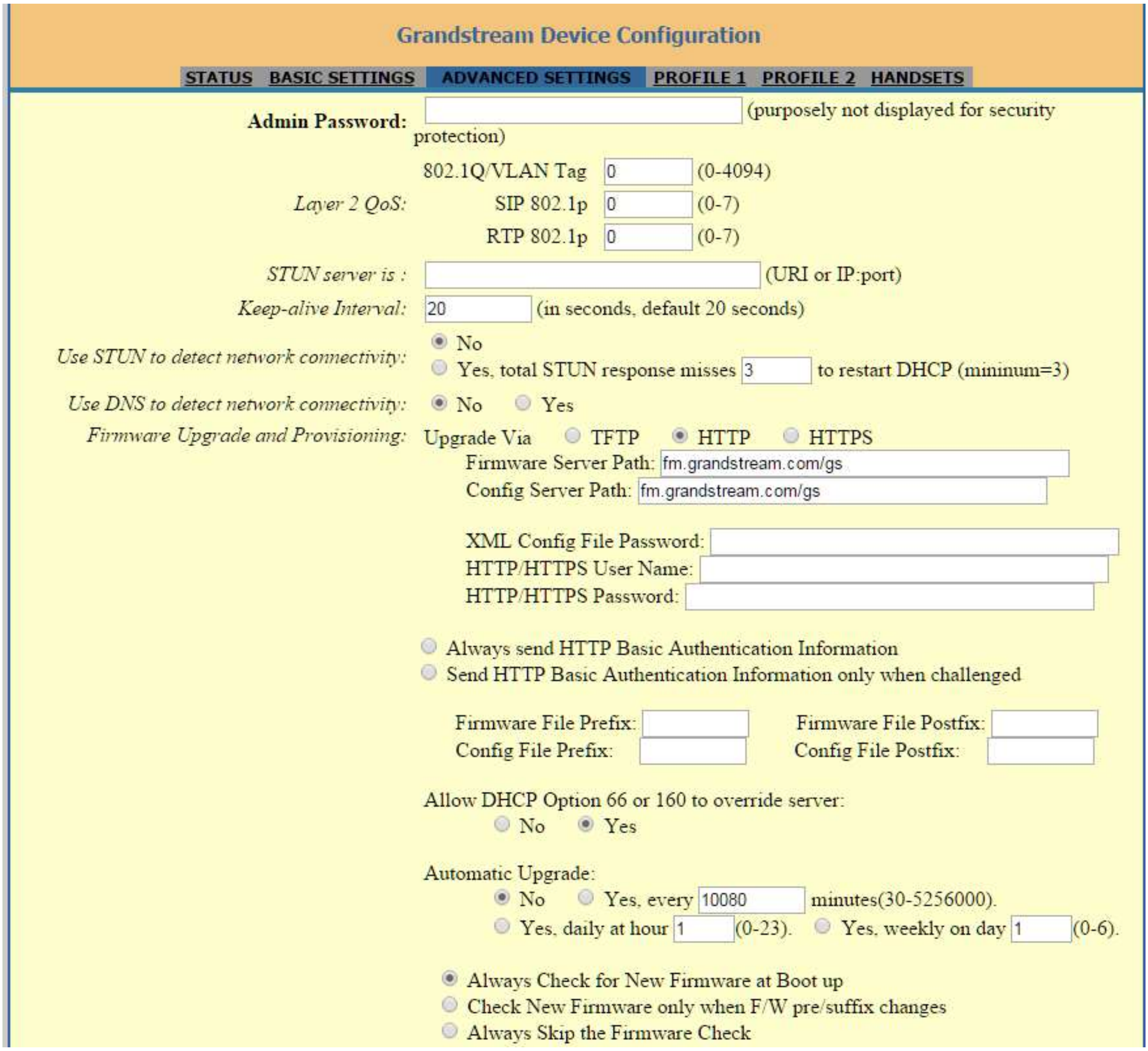
The following functionality was not validated:

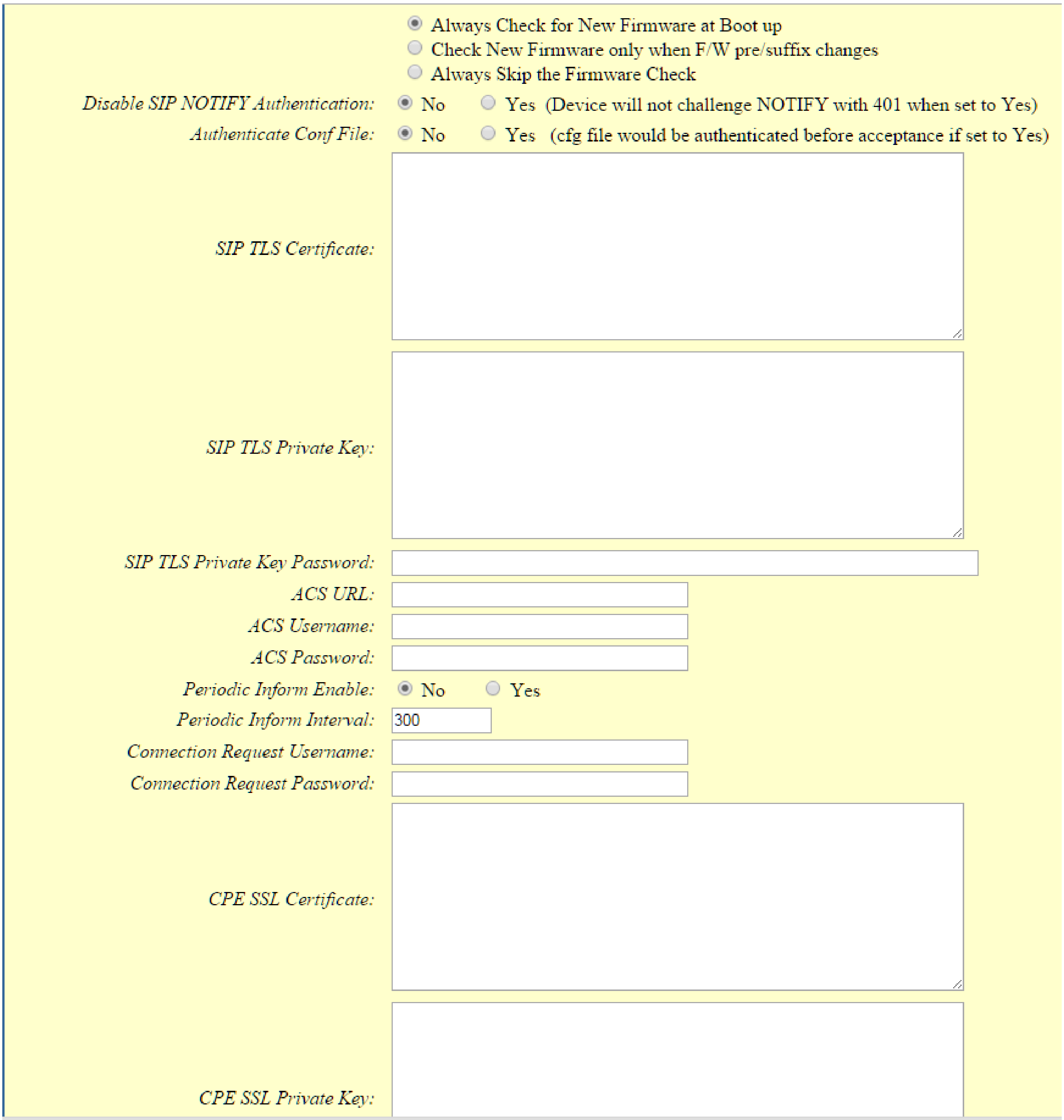
- TLS Encryption
- MWI features
- Provisioning via FTP/HTTP/TFTP separate from CIC (all provisioning was tested manually)
- Managed Phone Provisioning through CIC (it is an unmanaged phone, at present)
- Shared or Bridged line appearances (requires managed phone provisioning)

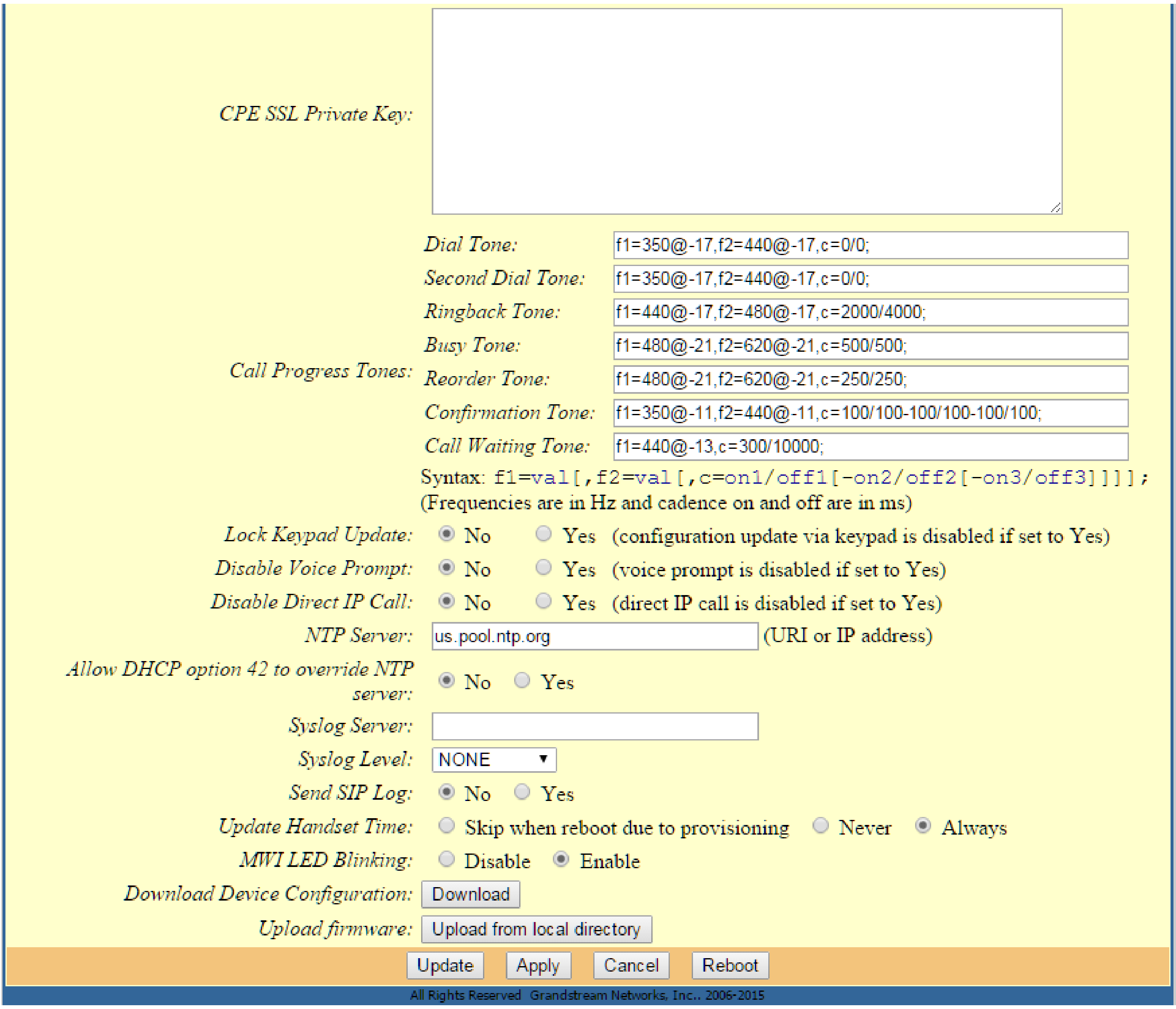
Details on Configuring MODEL

Screenshots from Grandstream web configuration:




Grandstream Device Configuration
STATUS BASIC SETTINGS ADVANCED SETTINGS PROFILE 1 PROFILE 2 HANDSETS
Admin Password: (purposely not displayed for security protection)
Layer 2 QoS:
802.1Q/VLAN Tag 0 (0-4094)
SIP 802.1p 0 (0-7)
RTP 802.1p 0 (0-7)
STUN server is : (URI or IP:port)
Keep-alive Interval: 20 (in seconds, default 20 seconds)
Use STUN to detect network connectivity: No
Yes, total STUN response misses 3 to restart DHCP (minimum=3)
Use DNS to detect network connectivity: No Yes
Firmware Upgrade and Provisioning: Upgrade Via TFTP HTTP HTTPS
Firmware Server Path: fm.grandstream.com/gs
Config Server Path: fm.grandstream.com/gs
XML Config File Password:
HTTP/HTTPS User Name:
HTTP/HTTPS Password:
Always send HTTP Basic Authentication Information
Send HTTP Basic Authentication Information only when challenged
Firmware File Prefix: Firmware File Postfix:
Config File Prefix: Config File Postfix:
Allow DHCP Option 66 or 160 to override server:
No Yes
Automatic Upgrade:
No Yes, every 10080 minutes(30-5256000).
Yes, daily at hour 1 (0-23). Yes, weekly on day 1 (0-6).
Always Check for New Firmware at Boot up
Check New Firmware only when F/W pre/suffix changes
Always Skip the Firmware Check


Always Check for New Firmware at Boot up
Check New Firmware only when F/W pre/suffix changes
Always Skip the Firmware Check
Disable SIP NOTIFY Authentication:
No
Yes (Device will not challenge NOTIFY with 401 when set to Yes)
Authenticate Conf File:
No
Yes (cfg file would be authenticated before acceptance if set to Yes)
SIP TLS Certificate:
SIP TLS Private Key:
SIP TLS Private Key Password:
ACS URL:
ACS Username:
ACS Password:
Periodic Inform Enable:
No
Yes
Periodic Inform Interval:
300
Connection Request Username:
Connection Request Password:
CPE SSL Certificate:
CPE SSL Private Key:



CPE SSL Private Key:		
Call Progress Tones:	Dial Tone:	f1=350@-17,f2=440@-17,c=0/0;
	Second Dial Tone:	f1=350@-17,f2=440@-17,c=0/0;
	Ringback Tone:	f1=440@-17,f2=480@-17,c=2000/4000;
	Busy Tone:	f1=480@-21,f2=620@-21,c=500/500;
	Reorder Tone:	f1=480@-21,f2=620@-21,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;
	Syntax: f1=val[,f2=val[,c=on1/off1[-on2/off2[-on3/off3]]]]; (Frequencies are in Hz and cadence on and off are in ms)	
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)	
Allow DHCP option 42 to override NTP server:	No Yes	
Syslog Server:		
Syslog Level:	NONE	
Send SIP Log:	No Yes	
Update Handset Time:	Skip when reboot due to provisioning Never Always	
MWI LED Blinking:	Disable Enable	
Download Device Configuration:	Download	
Upload firmware:	Upload from local directory	

Update Apply Cancel Reboot

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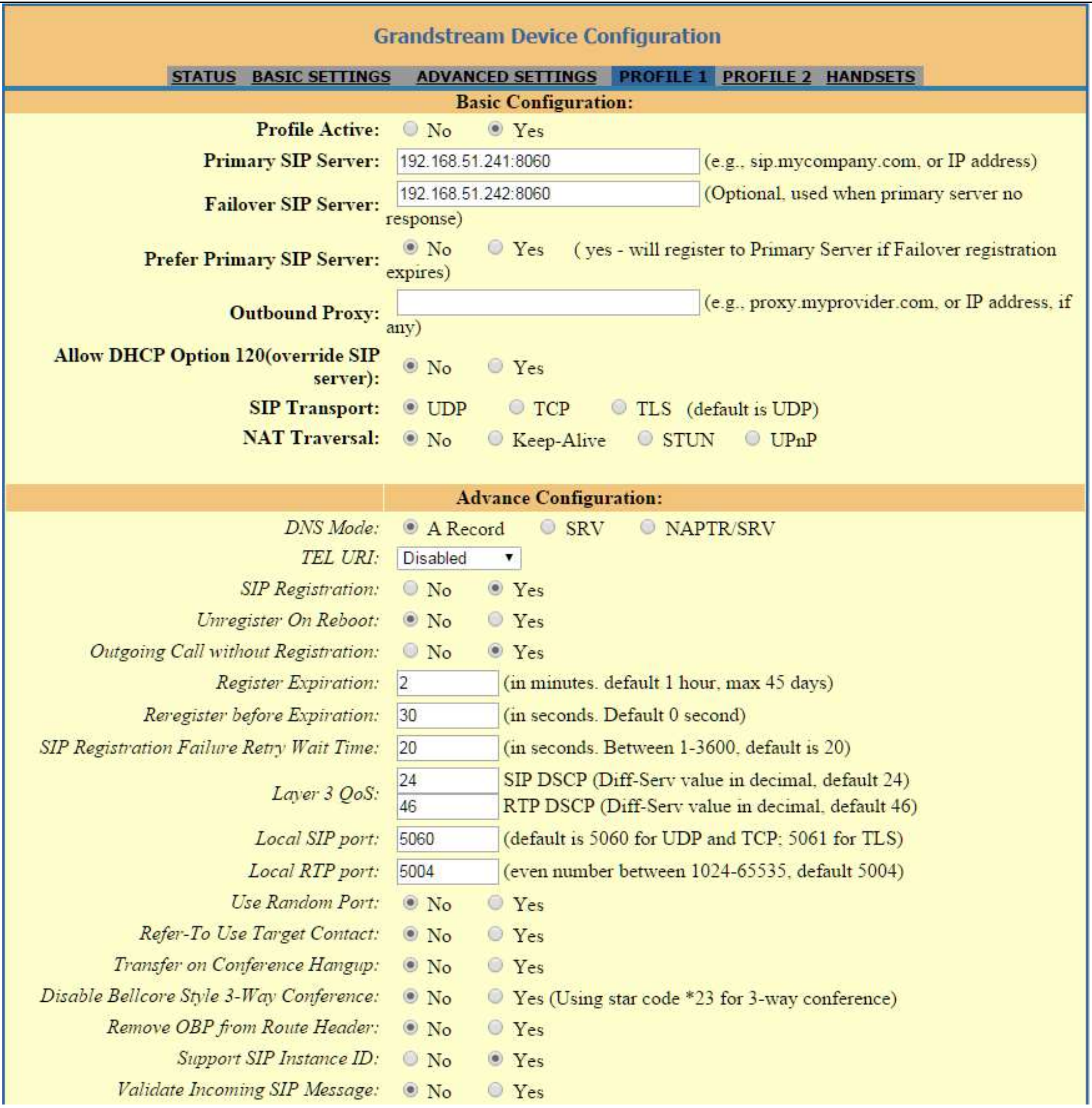

Grandstream Device Configuration
STATUS BASIC SETTINGS ADVANCED SETTINGS PROFILE 1 PROFILE 2 HANDSETS
Basic Configuration:
Profile Active: No Yes
Primary SIP Server: 192.168.51.241:8060 (e.g., sip.mycompany.com, or IP address)
Failover SIP Server: 192.168.51.242:8060 (Optional, used when primary server 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)
Allow DHCP Option 120(override SIP server): No Yes
SIP Transport: UDP TCP TLS (default is UDP)
NAT Traversal: No Keep-Alive STUN UPnP
Advance Configuration:
DNS Mode: A Record SRV NAPTR/SRV
TEL URI: Disabled
SIP Registration: No Yes
Unregister On Reboot: No Yes
Outgoing Call without Registration: No Yes
Register Expiration: 2 (in minutes. default 1 hour, max 45 days)
Reregister before Expiration: 30 (in seconds. Default 0 second)
SIP Registration Failure Retry Wait Time: 20 (in seconds. Between 1-3600, default is 20)
Layer 3 QoS: 24 SIP DSCP (Diff-Serv value in decimal, default 24)
46 RTP DSCP (Diff-Serv value in decimal, default 46)
Local SIP port: 5060 (default is 5060 for UDP and TCP; 5061 for TLS)
Local RTP port: 5004 (even number between 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

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)
Caller ID Display:	Auto Disabled From Header
Use Privacy Header:	Default No Yes
Use P-Preferred-Identity Header:	Default No Yes
SIP REGISTER Contact Header Uses:	LAN Address WAN Address
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
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)
Proxy-Require:	
Use NAT IP:	(used in SIP/SDP message if specified)
Use SIP User-Agent Header:	
Do Not Escape '#' as %23 in SIP URI:	No Yes
Disable Multiple m line in SDP:	No Yes
Ring Timeout:	60 (10-300, default is 60 seconds)
Hunting Group Ring Timeout:	20 (5-300, default is 20 seconds)
Hunting Group Type:	Linear Parallel Shared Line
Delayed Call Forward Wait Time:	20 (Allowed range 1-120, in seconds.)
No Key Entry Timeout:	4 (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:	{ 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)

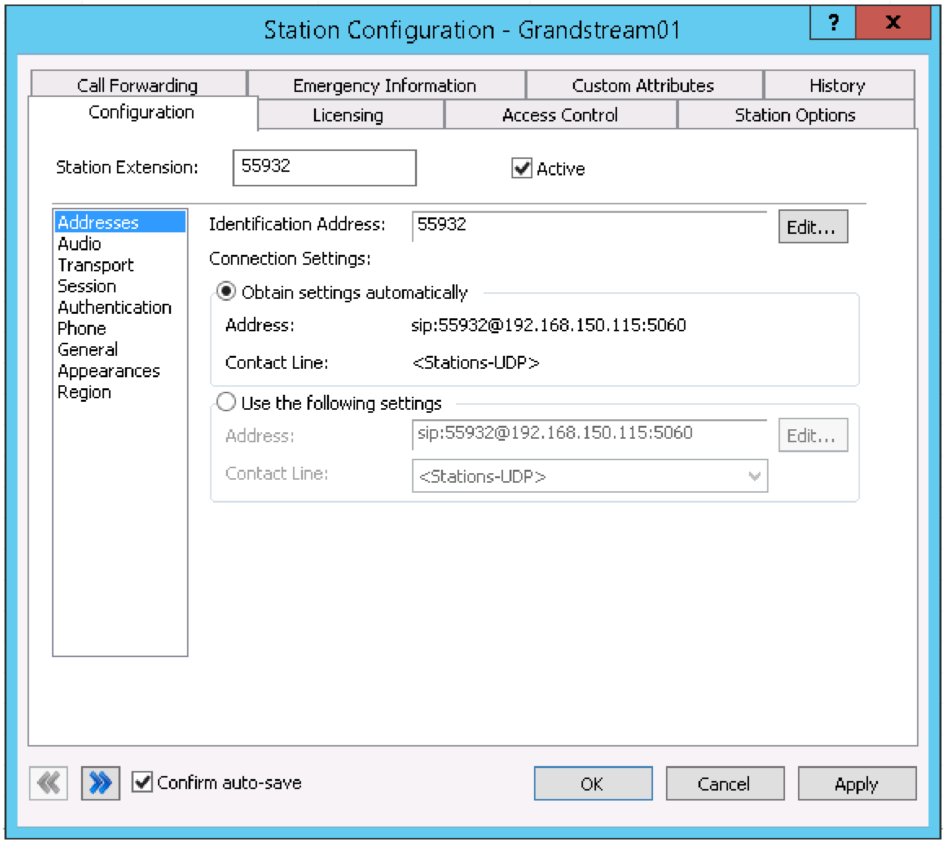
SUBSCRIBE for MWI:	No, do not send SUBSCRIBE for Message Waiting Indication Yes, send periodical SUBSCRIBE for Message Waiting Indication
Send Anonymous:	No Yes (caller ID will be blocked if set to Yes)
Disable Call-Waiting:	No Yes
Disable Call-Waiting Caller ID:	No Yes
Disable Call-Waiting Tone:	No Yes
Disable Reminder Ring for On-Hold Call:	No Yes
Anonymous Call Rejection:	No Yes
Special Feature:	Standard
Session Expiration:	180 (in seconds. default 180 seconds)
Min-SE:	90 (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)
Enable 100rel:	No Yes
Add Auth Header On Initial REGISTER:	No Yes

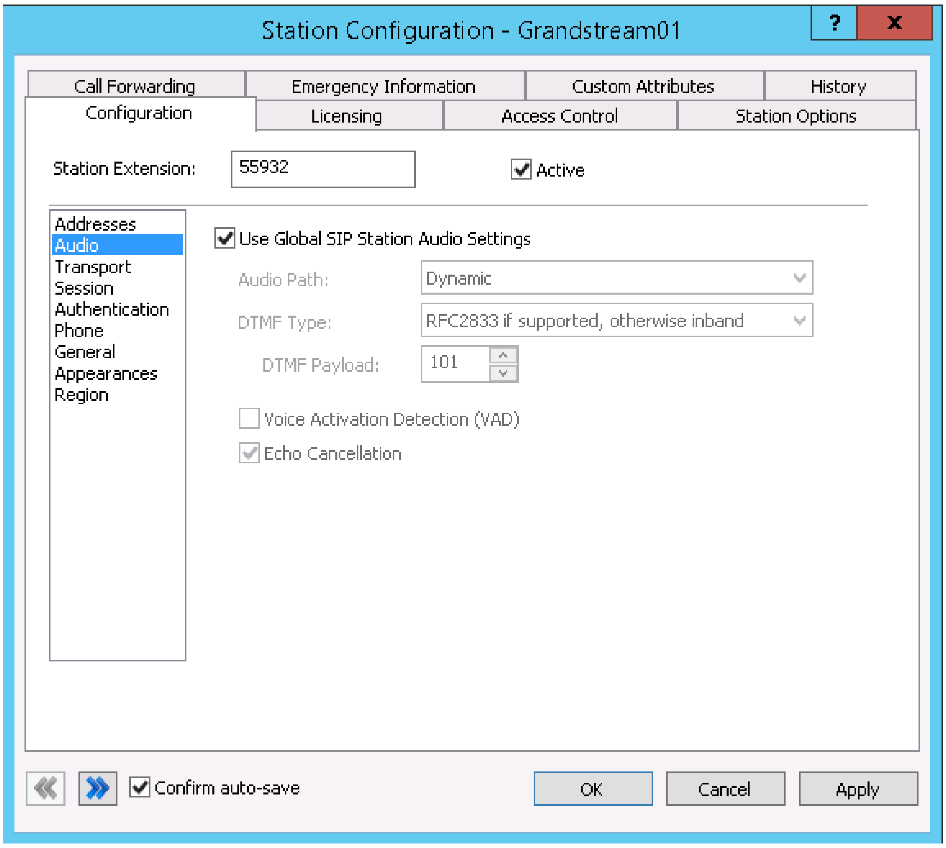
Codec Configuration:

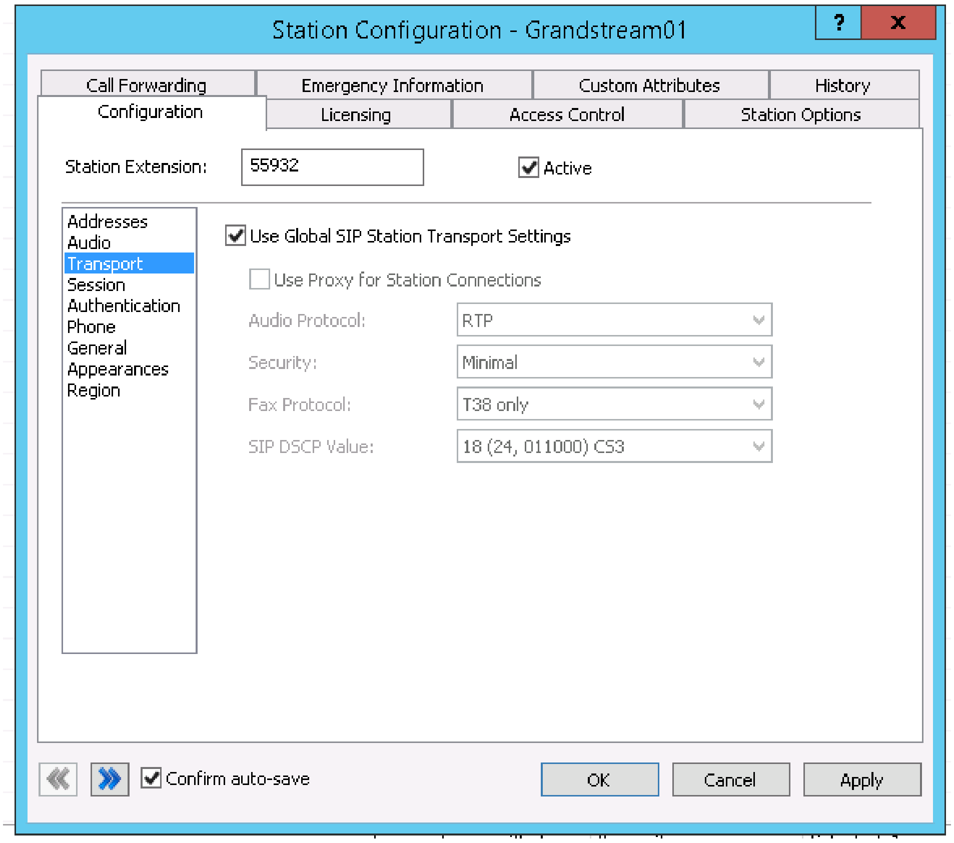
Preferred Vocoder: (in listed order)	choice 1: PCMU choice 2: PCMA choice 3: G729 choice 4: G723 choice 5: G726-32 choice 6: iLBC
VAD:	No Yes
Jitter Buffer Type:	Fixed Adaptive
Jitter Buffer Length:	Low Medium High
SRTP Mode:	Disabled Enabled but not forced Enabled and forced
G723 Rate:	6.3kbps encoding rate 5.3kbps encoding rate
Use First Matching Vocoder in 200OK SDP:	No Yes
iLBC Frame Size:	20ms 30ms
iLBC Payload Type:	97 (between 96 and 127, default is 97)
Voice Frames per TX:	2
Symmetric RTP:	No Yes

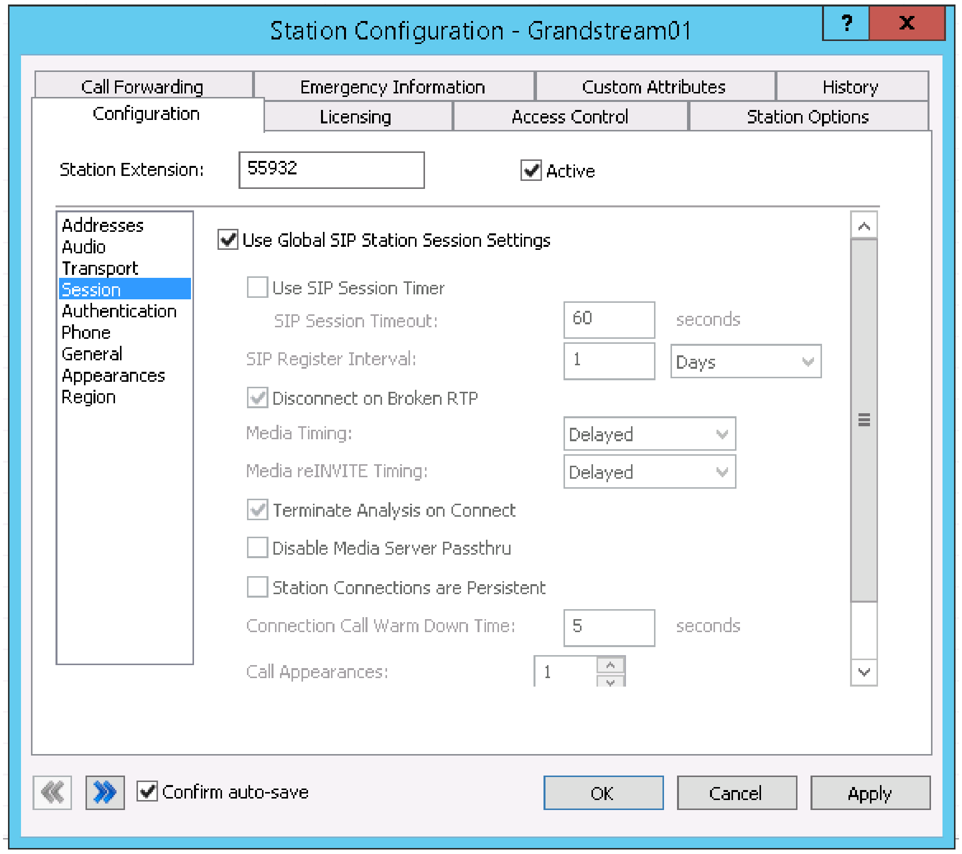
Update Apply Cancel Reboot

Administrator:






Station Configuration - Grandstream01
Call Forwarding
Emergency Information
Custom Attributes
History
Configuration
Licensing
Access Control
Station Options
Station Extension:
55932
Active
Addresses
Audio
Transport
Session
Authentication
Phone
General
Appearances
Region
Use Global SIP Station Transport Settings
Use Proxy for Station Connections
Audio Protocol:
RTP
Security:
Minimal
Fax Protocol:
T38 only
SIP DSCP Value:
18 (24, 011000) CS3
Confirm auto-save
OK
Cancel
Apply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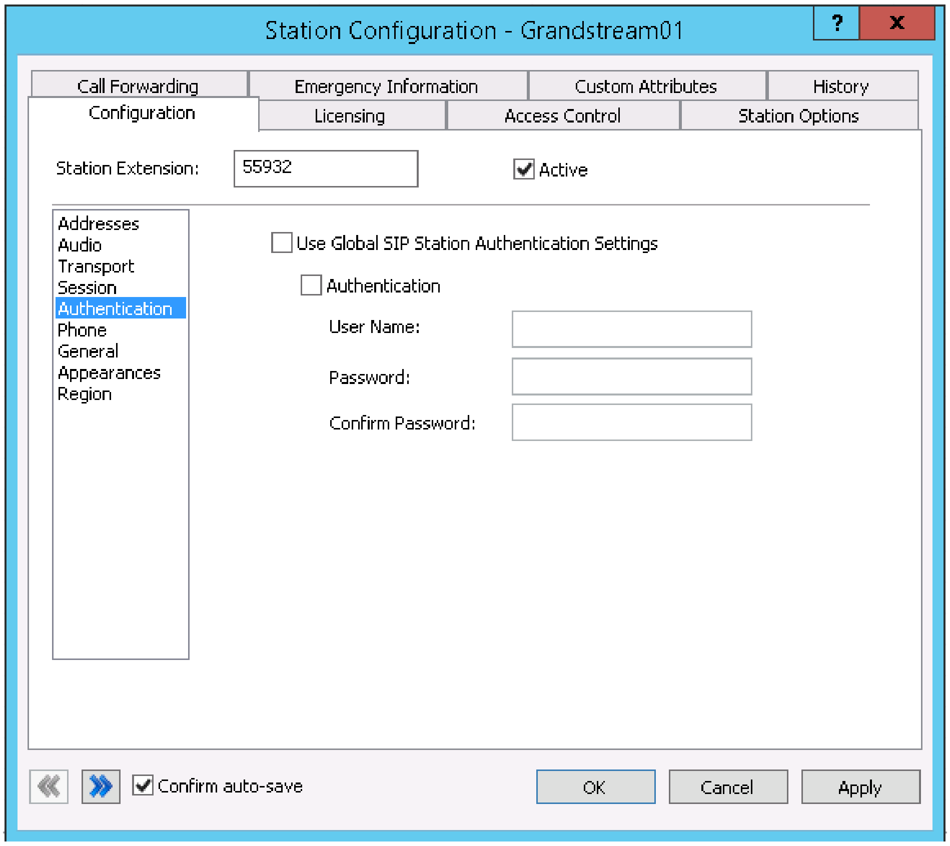
Session
Authentication
Phone
General
Appearances
Region
Use Global SIP Station Session Settings
Use SIP Session Timer
SIP Session Timeout:
60
seconds
SIP Register Interval:
1
Days
Disconnect on Broken RTP
Media Timing:
Delayed
Media reINVITE Timing:
Delayed
Terminate Analysis on Connect
Disable Media Server Passthru
Station Connections are Persistent
Connection Call Warm Down Time:
5
seconds
Call Appearances:
1
Confirm auto-save
OK
Cancel
Apply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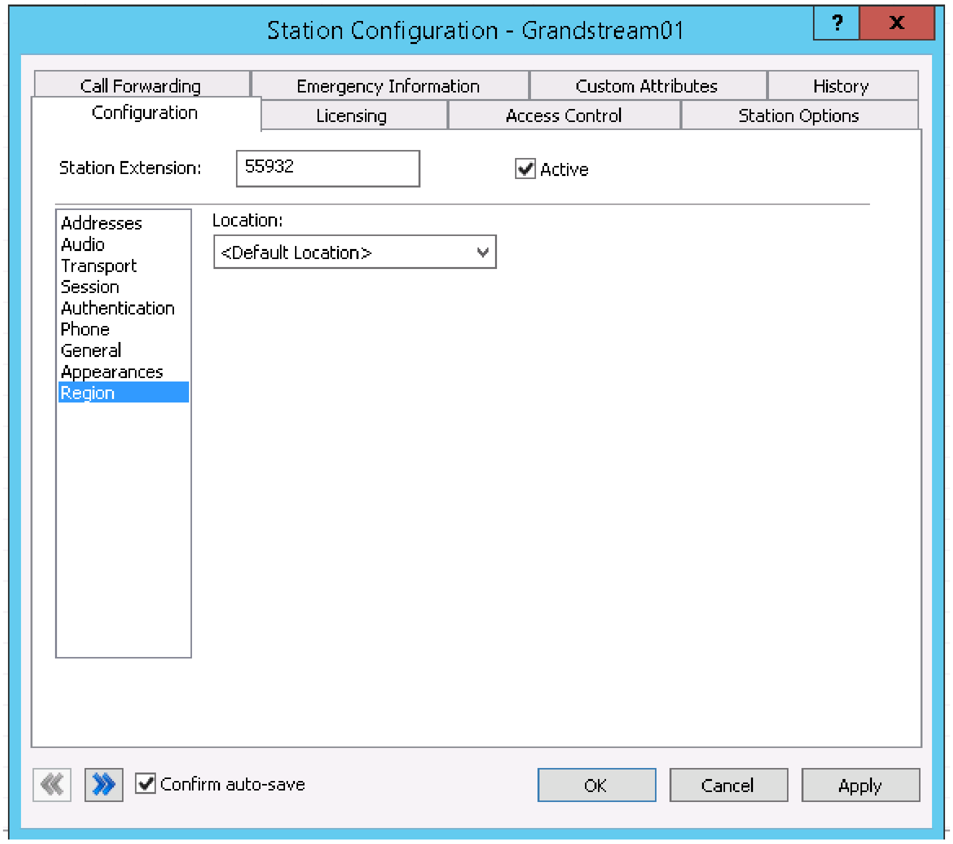
Station Options
Station Extension:
55932
Active
Addresses
Audio
Transport
Session
Authentication
Phone
General
Appearances
Region
Use Global SIP Station Authentication Settings
Authentication
User Name:
Password:
Confirm Password:
Confirm auto-save
OK
Cancel
Apply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Emergency Information
Custom Attributes
History
Configuration
Licensing
Access Control
Station Options
Station Extension:
55932
Active
Addresses
Audio
Transport
Session
Authentication
Phone
General
Appearances
Region
Location:
<Default Location>
Confirm auto-save
OK
Cancel
Apply

References

- User Manual, English:
  - http://www.grandstream.com/products/dp_series/dp71x/documents/dp71x_usermanual_english.pdf
- Provisioning guide (XML configuration via FTP/HTTP)
  - http://www.grandstream.com/general/gs_provisioning_guide_public.pdf
  - Please note, this was not validated in the interoperability test.