Set up Overhead Paging for Fire Station

Tuesday, February 21, 2017 8:33 AM

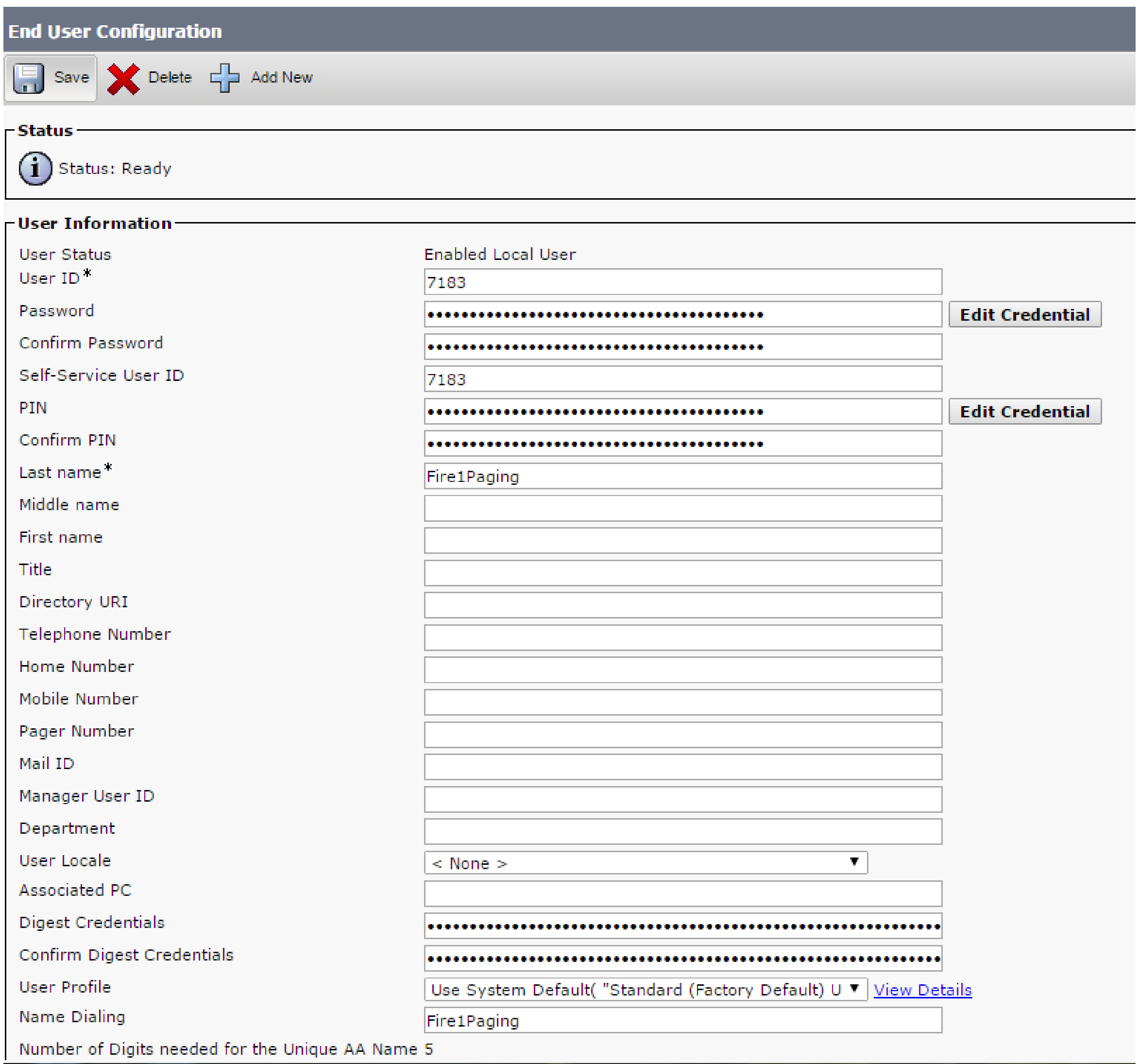
Big 'thank you!' to Brian Andersen from the City of Henderson who patiently taught me how to set this up. He's the greatest.

Steps You Have to Take

1. **Make a user for this line (USDD interface requires a pw, so you need a user)**
2. **If you don't have a SIP Profile already set up, set it up**
3. **Make a SIP Device**
4. **Assign it a Line**
5. **Lock Down Who Can Call It.**
6. **If it doesn't work, have USDD check to see if there's an update for your panel.**

Need to make an End User for these Lines

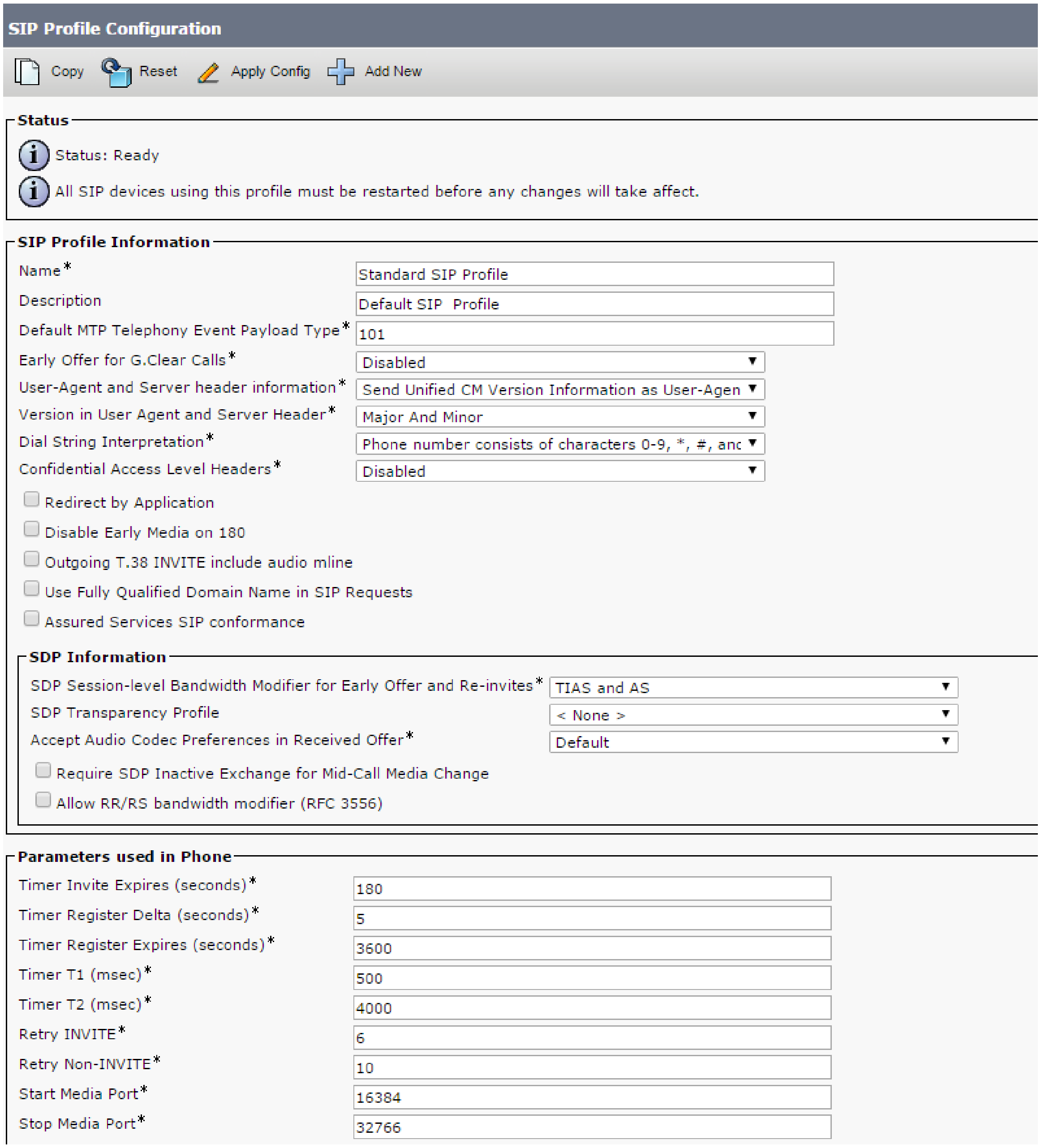
The reason these lines need an end user is because the paging system interface requires a password.
Go to User Management > End User and make a new user. Make the password and the digest credentials the same. Also set a PIN. I don't know if it matters but we didn't put special characters in the password because it was giving us a hard time, but we think that was from something else but aren't sure. Below this screenshot there are more options, but nothing else needs to be set besides pw and digest.



Make sure you have a SIP Profile that can be used for the new SIP lines

Device > Device Settings > SIP Profile. The only thing I changed from what it gave me as default when I hit "Add New" under Sip Profile is the SDP Transparency Profile.

Change it to "none".



SIP Profile Configuration

Copy Reset Apply Config Add New

Status

Status: Ready

All SIP devices using this profile must be restarted before any changes will take affect.

SIP Profile Information

Name*	Standard SIP Profile
Description	Default SIP Profile
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	Disabled
User-Agent and Server header information*	Send Unified CM Version Information as User-Agen
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, anc
Confidential Access Level Headers*	Disabled

- [ ] Redirect by Application
- [ ] Disable Early Media on 180
- [ ] Outgoing T.38 INVITE include audio mline
- [ ] Use Fully Qualified Domain Name in SIP Requests
- [ ] Assured Services SIP conformance

SDP Information

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	< None >
Accept Audio Codec Preferences in Received Offer*	Default

- [ ] Require SDP Inactive Exchange for Mid-Call Media Change
- [ ] Allow RR/RS bandwidth modifier (RFC 3556)

Parameters used in Phone

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766

Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

- [x] Conference Join Enabled
- [ ] RFC 2543 Hold
- [x] Semi Attended Transfer
- [ ] Enable VAD
- [ ] Stutter Message Waiting
- [ ] MLPP User Authorization

Normalization Script

Normalization Script: < None >

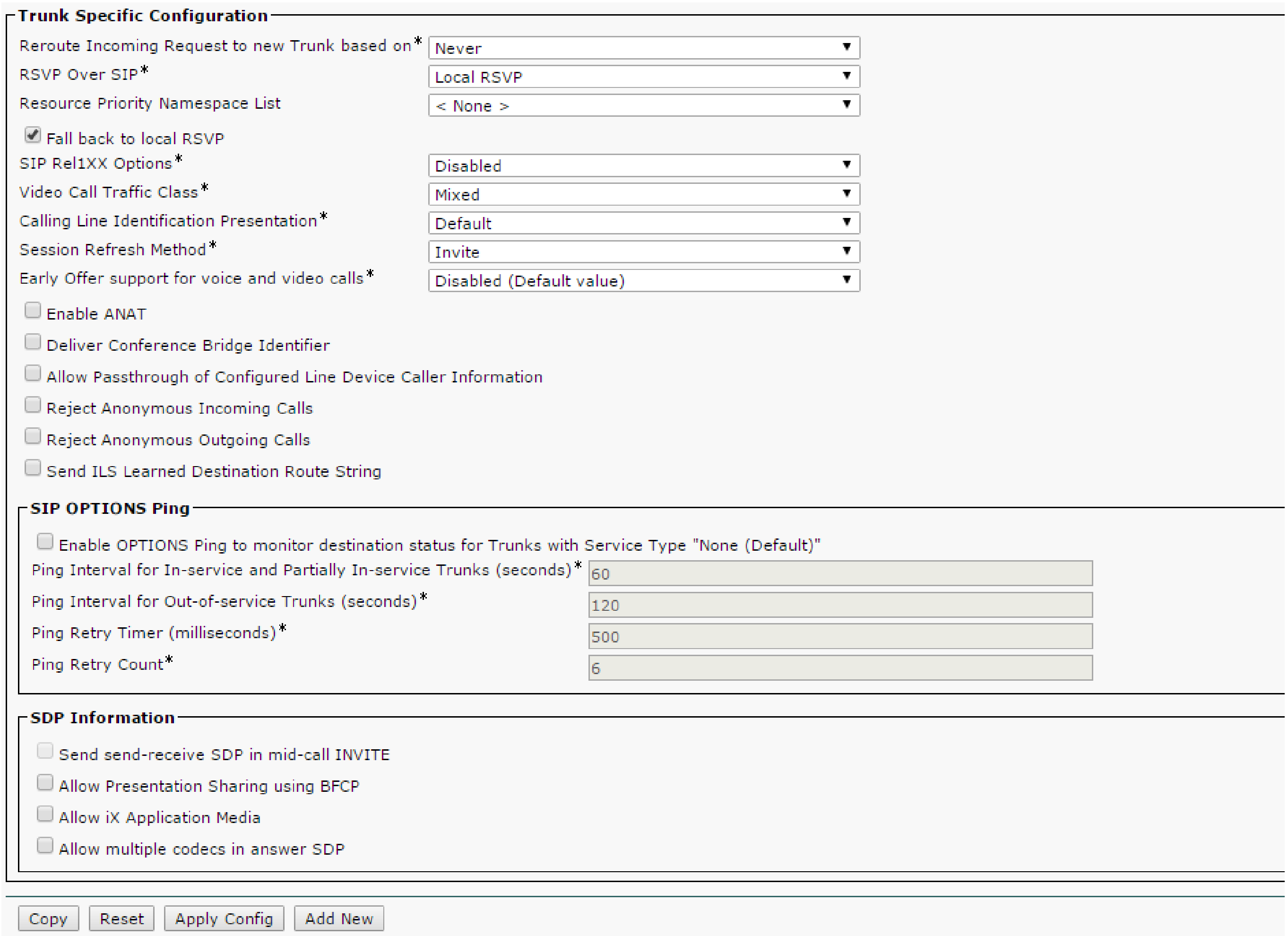
- [ ] Enable Trace

	Parameter Name	Parameter Value
1		

Incoming Requests FROM URI Settings

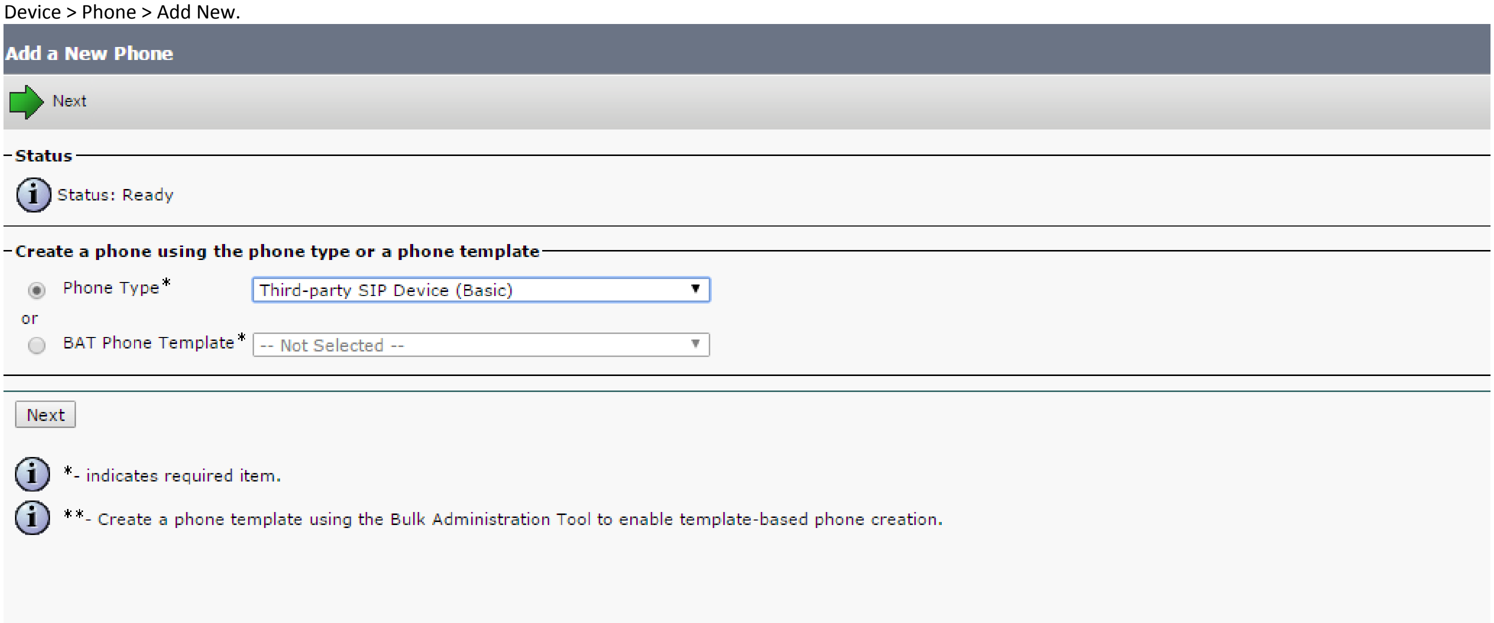
Caller ID DN

Caller Name



Add a New Device

Device > Phone > Add New.



You're making up a MAC address, but we did 0's and then the extension. Notice the "Owner ID"

Phone Type

Product Type: **Third-party SIP Device (Basic)**
Device Protocol: **SIP**

Real-time Device Status

Registration: Unknown
IPv4 Address: None

Device Information

- [x] Device is Active
- Device is not trusted

MAC Address*	000000007184	
Description	Fire 2 - Paging System	
Device Pool*	dp_FIRE_Admin_phones	View Details
Common Device Configuration	MigratedCommonProfile2	View Details
Phone Button Template*	Third-party SIP Device (Basic)	
Common Phone Profile*	Standard Common Phone Profile	View Details
Calling Search Space	COGLocalCS	
AAR Calling Search Space	< None >	
Media Resource Group List	< None >	
Location*	Hub_None	
AAR Group	< None >	
Device Mobility Mode*	Default	View Current Device Mobility Settings
Owner	(•) User ( ) Anonymous (Public/Shared Space)	
Owner User ID*	7184	Find
Use Trusted Relay Point*	Default	
Always Use Prime Line*	Default	
Always Use Prime Line for Voice Message*	Default	
Geolocation	< None >	

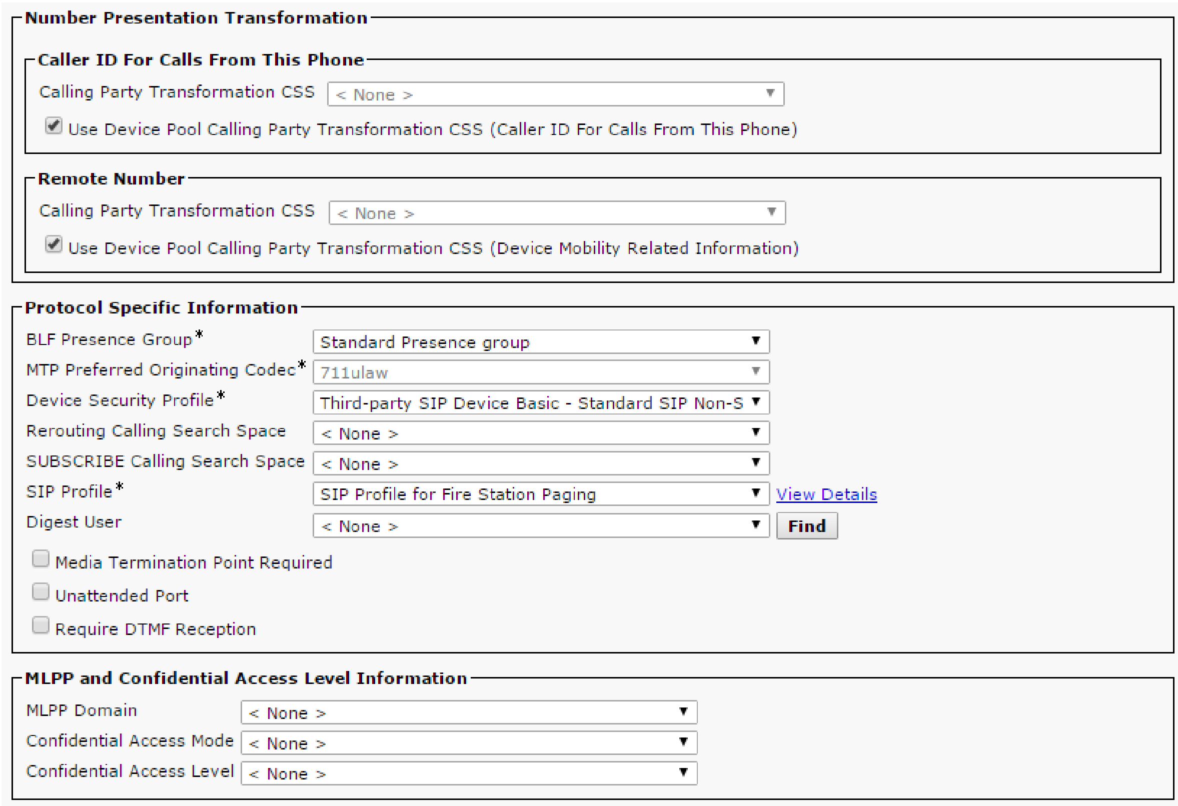
- [ ] Ignore Presentation Indicators (internal calls only)
- [x] Logged Into Hunt Group
- [ ] Remote Device

Number Presentation Transformation

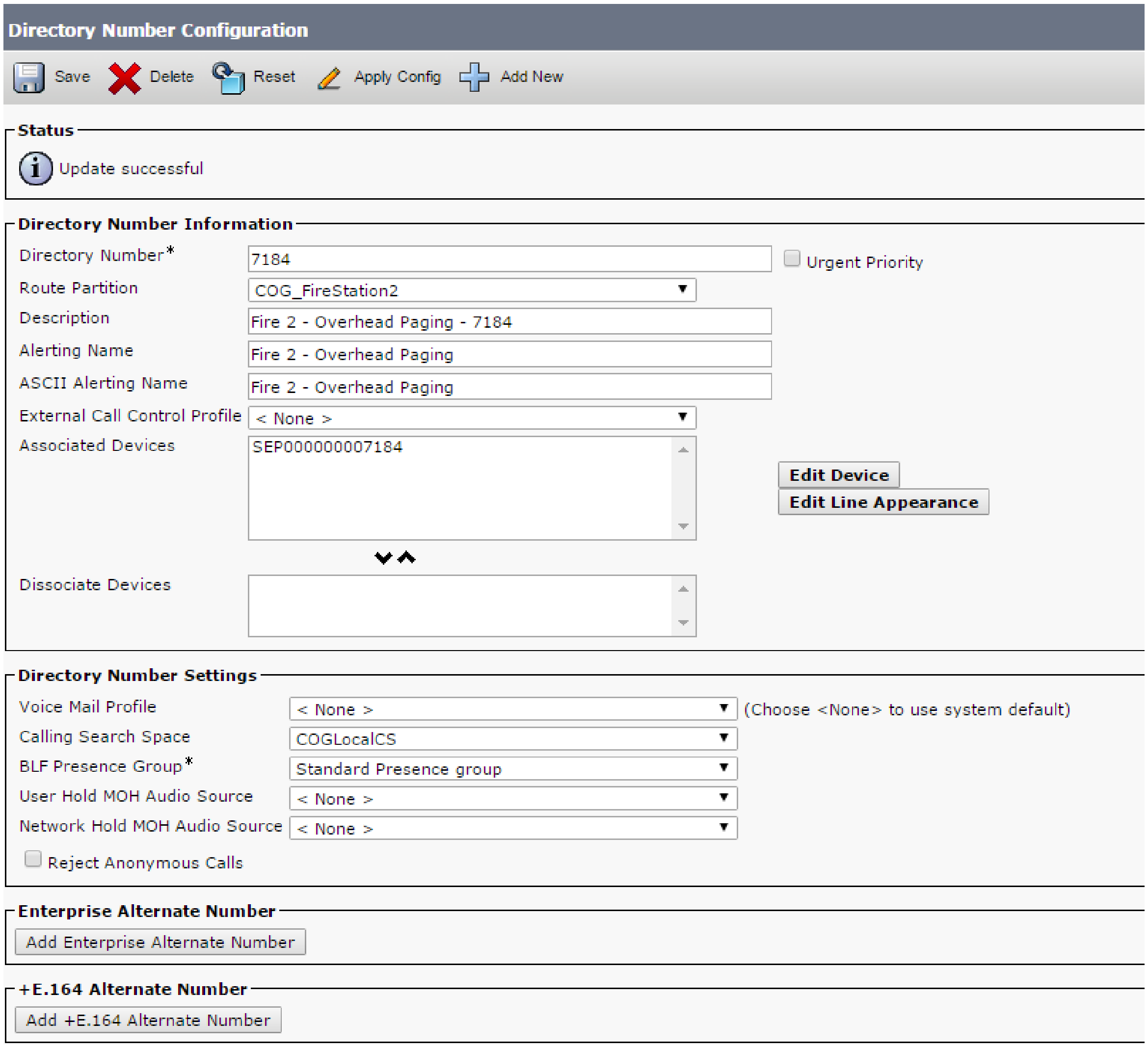
Caller ID For Calls From This Phone

Calling Party Transformation CSS: < None >

- [x] Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)



Notice how the Device Security Profile is set and the SIP profile is set to the new one you just made. Once you hit 'save' you'll see that you can add a line to it.

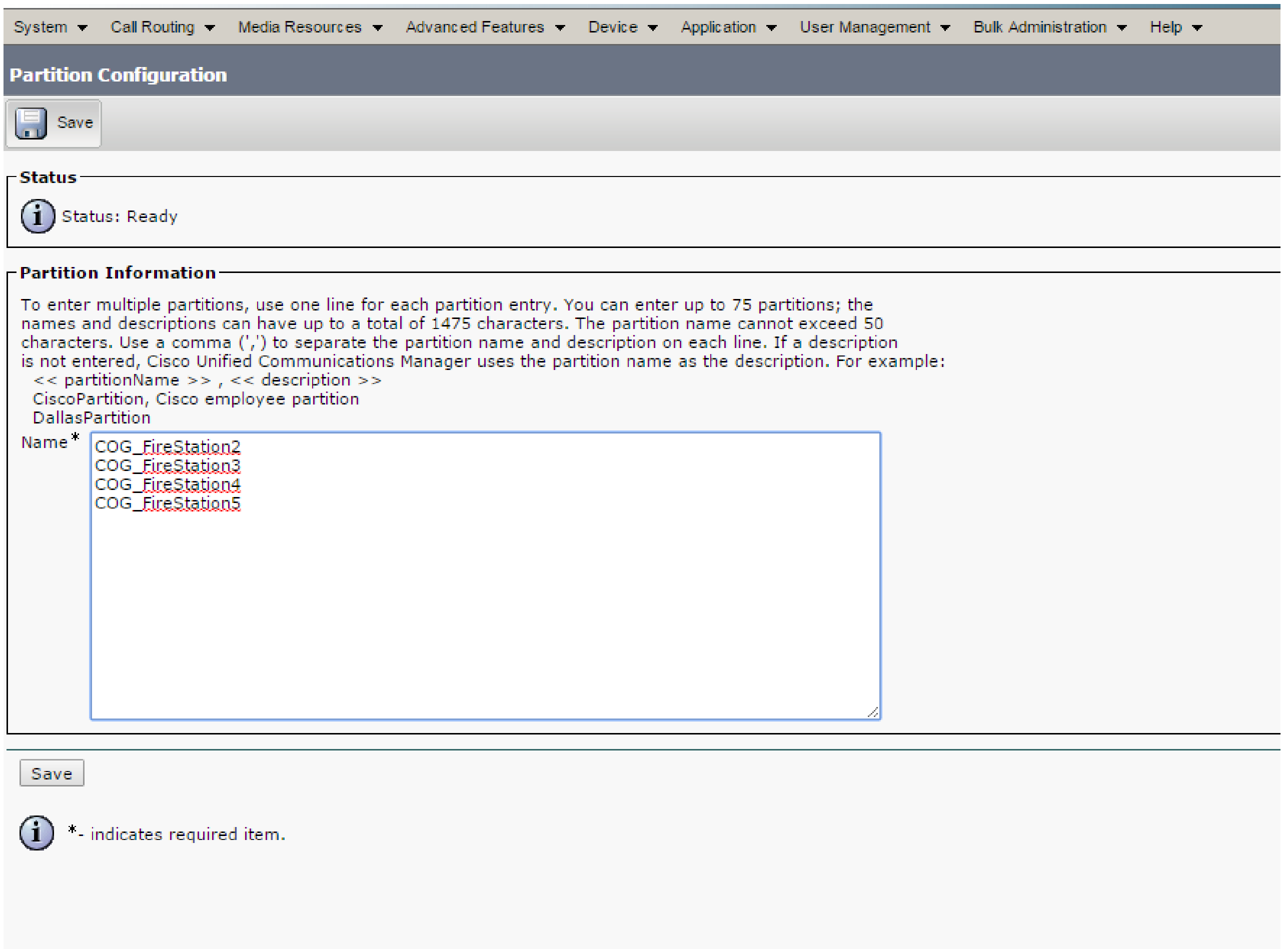


The choices below this, I didn't change also I added the end user to the 'associate end users' button at the bottom, but it was working before I did that so it's not necessary. Please note how this line is in the partition for that FireStation. This means you have to have the right CSS on your phone to call this line.

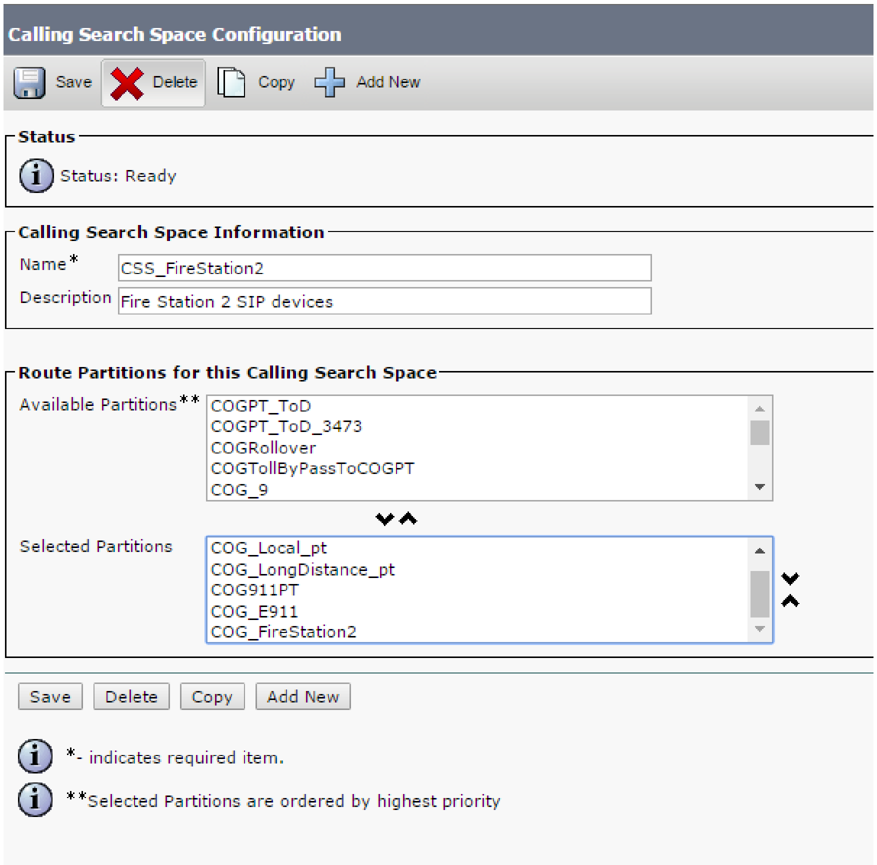
Locking Down the New Lines

Remember, partition means who can call you, and calling search space means who can you call.

So you don't want firemen from other stations (or police men) prank calling the fire station's overhead paging. You really only want the phones at that fire station calling that fire station's paging system. So put the paging system's phone numbers into their own partitions (Call Routing > Class of Control > Partition) Make the partitions for the new lines.



Then make the Call Search Spaces that will be able to call those lines. These CSSs will be applied to all the phones at that fire station, so they'll need all the regular partitions (long distance, 911, etc) but then add the FireStationX partition that the new extension will live in



This means any phone with this CSS can call the paging system and any other partitions it has access to.