

# 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.

**End User Configuration**

Save Delete Add New

**Status**  
Status: Ready

**User Information**

User Status	Enabled Local User
User ID*	7183
Password	..... <a href="#">Edit Credential</a>
Confirm Password	.....
Self-Service User ID	7183
PIN	..... <a href="#">Edit Credential</a>
Confirm PIN	.....
Last name*	Fire1Paging
Middle name	
First name	
Title	
Directory URI	
Telephone Number	
Home Number	
Mobile Number	
Pager Number	
Mail ID	
Manager User ID	
Department	
User Locale	< None >
Associated PC	
Digest Credentials	.....
Confirm Digest Credentials	.....
User Profile	Use System Default( "Standard (Factory Default) U" <a href="#">View Details</a>
Name Dialing	Fire1Paging
Number of Digits needed for the Unique AA Name	5

## 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

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#### Status

**i** Status: Ready

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

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#### 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

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#### 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)

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#### 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

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

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**Normalization Script**

Normalization Script < None >

Enable Trace

	Parameter Name	Parameter Value	
1			<input type="button" value="+"/> <input type="button" value="-"/>

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**Incoming Requests FROM URI Settings**

Caller ID DN

Caller Name

**Trunk Specific Configuration**

Reroute Incoming Request to new Trunk based on\*

RSVP Over SIP\*

Resource Priority Namespace List

Fall back to local RSVP

SIP Rel1XX Options\*

Video Call Traffic Class\*

Calling Line Identification Presentation\*

Session Refresh Method\*

Early Offer support for voice and video calls\*

Enable ANAT

Deliver Conference Bridge Identifier

Allow Passthrough of Configured Line Device Caller Information

Reject Anonymous Incoming Calls

Reject Anonymous Outgoing Calls

Send ILS Learned Destination Route String

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**SIP OPTIONS Ping**

Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

Ping Interval for In-service and Partially In-service Trunks (seconds)\*

Ping Interval for Out-of-service Trunks (seconds)\*

Ping Retry Timer (milliseconds)\*

Ping Retry Count\*

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**SDP Information**

Send send-receive SDP in mid-call INVITE

Allow Presentation Sharing using BFCP

Allow iX Application Media

Allow multiple codecs in answer SDP

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### Add a New Device

Device > Phone > Add New.

#### Add a New Phone

 Next

#### Status


 Status: Ready


#### Create a phone using the phone type or a phone template

Phone Type\*

or

BAT Phone Template\*

 \*- indicates required item.

 \*\*- Create a phone template using the Bulk Administration Tool to enable template-based phone creation.

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**

Device is Active  
 Device is not trusted

MAC Address\*

Description

Device Pool\*  [View Details](#)

Common Device Configuration  [View Details](#)

Phone Button Template\*

Common Phone Profile\*  [View Details](#)

Calling Search Space

AAR Calling Search Space

Media Resource Group List

Location\*

AAR Group

Device Mobility Mode\*  [View Current Device Mobility Settings](#)

Owner  
 User  Anonymous (Public/Shared Space)

Owner User ID\*

Use Trusted Relay Point\*

Always Use Prime Line\*

Always Use Prime Line for Voice Message\*

Geolocation

Ignore Presentation Indicators (internal calls only)  
 Logged Into Hunt Group  
 Remote Device

**Number Presentation Transformation**

**Caller ID For Calls From This Phone**

Calling Party Transformation CSS

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

**Number Presentation Transformation**

**Caller ID For Calls From This Phone**

Calling Party Transformation CSS < None >

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

**Remote Number**

Calling Party Transformation CSS < None >

Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

**Protocol Specific Information**

BLF Presence Group\* Standard Presence\_group

MTP Preferred Originating Codec\* 711ulaw

Device Security Profile\* Third-party SIP Device Basic - Standard SIP Non-S

Rerouting Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile\* SIP Profile for Fire Station Paging [View Details](#)

Digest User < None > **Find**

Media Termination Point Required

Unattended Port

Require DTMF Reception

**MLPP and Confidential Access Level Information**

MLPP Domain < None >

Confidential Access Mode < None >

Confidential Access Level < None >

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.

**Directory Number Configuration**

Save ✗ Delete ↺ Reset ✍ Apply Config ✚ Add New

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**Status**

ℹ Update successful

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**Directory Number Information**

Directory Number\*   Urgent Priority

Route Partition

Description

Alerting Name

ASCII Alerting Name

External Call Control Profile

Associated Devices  Edit Device  
Edit Line Appearance

↕ ↕

Dissociate Devices

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**Directory Number Settings**

Voice Mail Profile  (Choose <None> to use system default)

Calling Search Space

BLF Presence Group\*

User Hold MOH Audio Source

Network Hold MOH Audio Source

Reject Anonymous Calls

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**Enterprise Alternate Number**

Add Enterprise Alternate Number

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**+E.164 Alternate Number**

Add +E.164 Alternate Number

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.

## Partition Configuration

 Save


**Status**  
 Status: Ready

**Partition Information**

To enter multiple partitions, use one line for each partition entry. You can enter up to 75 partitions; the names and descriptions can have up to a total of 1475 characters. The partition name cannot exceed 50 characters. Use a comma (',') to separate the partition name and description on each line. If a description is not entered, Cisco Unified Communications Manager uses the partition name as the description. For example:  
<< partitionName >> , << description >>  
CiscoPartition, Cisco employee partition  
DallasPartition

Name\*




 Save

 \*- indicates required item.

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




**Calling Search Space Configuration**

Save  Delete  Copy  Add New

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**Status**

 Status: Ready

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**Calling Search Space Information**

Name\*

Description

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**Route Partitions for this Calling Search Space**


Available Partitions\*\*


- COGPT\_ToD
- COGPT\_ToD\_3473
- COGRollover
- COGTollByPassToCOGPT
- COG\_9

Selected Partitions

- COG\_Local\_pt
- COG\_LongDistance\_pt
- COG911PT
- COG\_E911
- COG\_FireStation2

Save Delete Copy Add New

 \*- indicates required item.

 \*\*Selected Partitions are ordered by highest priority

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