

Mutare Voice™ SIP Integration with Cisco Unified Communication Manager

Overview

This document outlines the configuration steps to integrate the Mutare Voice (SAM) using Session Initiation Protocol (SIP) with the Cisco Unified Communication Manager (CUCM).

Site Configuration

Cisco Unified Communication Manager must be at release 8.X or higher.

For this document, the configuration was as follows:

- Cisco CUCM v 8.2
- Mutare Voice v 1.8

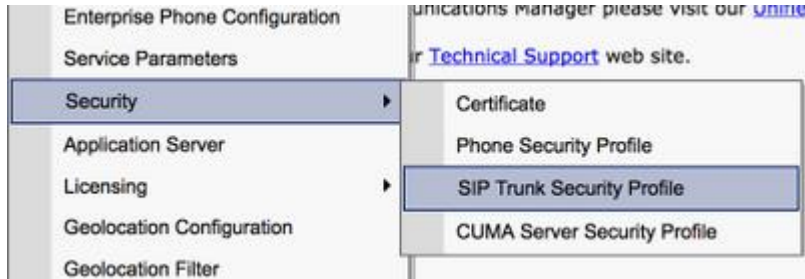
For the purposes of the configuration examples below, the following IP configuration was used:

- Mutare Voice (SAM) - 192.168.1.79
- CUCM – 192.168.1.30

Customer Initials:

Configure Cisco Unified Communication Manager

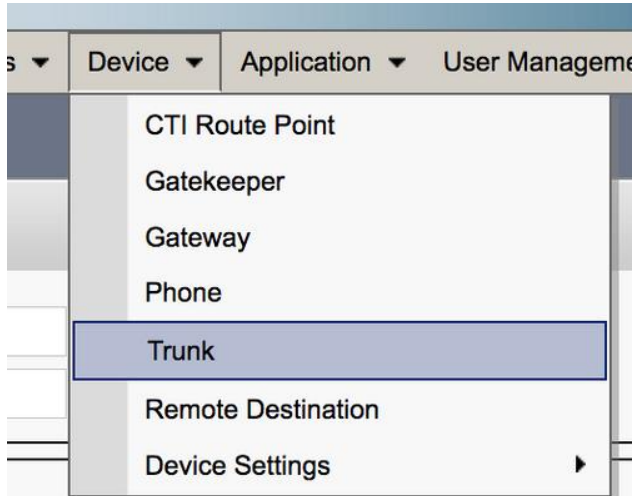
Step 1. On CUCM Admin page, navigate to **System > Security > SIP Trunk Security Profile**. Make a copy of the available profile. The default profile is **Non-Secure SIP Trunk Profile**. On the new profile, check these options; **Accept out-of-dialog refer**, **Accept unsolicited notification** and **Accept replaces header**.



SIP Trunk Security Profile Information	
Name *	Non Secure SIP trunk to Mutare SAM
Description	SIP trunk for SAM call completion
Device Security Mode	Non Secure ▼
Incoming Transport Type *	TCP+UDP ▼
Outgoing Transport Type	TCP ▼
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins) *	600
X.509 Subject Name	
Incoming Port *	5060
<input type="checkbox"/> Enable Application level authorization	
<input type="checkbox"/> Accept presence subscription	
<input checked="" type="checkbox"/> Accept out-of-dialog refer**	
<input checked="" type="checkbox"/> Accept unsolicited notification	
<input checked="" type="checkbox"/> Accept replaces header	
<input type="checkbox"/> Transmit security status	
<input type="checkbox"/> Allow charging header	
SIP V.150 Outbound SDP Offer Filtering *	Use Default Filter ▼

Customer Initials:

Step 2. In order to create a SIP trunk, navigate to Device > Trunk and select Add New.




Step 3. Select the Type as **SIP trunk**. Rest of the fields auto-populate.

Trunk Information	
Trunk Type*	SIP Trunk
Device Protocol*	SIP
Trunk Service Type*	None(Default)

Customer Initials:

Step 4. Provide a name for the Trunk and assign an appropriate Device Pool.

Status

 Add successful

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	<input type="text" value="SAM"/>
Description	<input type="text" value="SAM call completion"/>
Device Pool*	<input type="text" value="Default"/>
Common Device Configuration	<input type="text" value="< None >"/>
Call Classification*	<input type="text" value="Use System Default"/>
Media Resource Group List	<input type="text" value="< None >"/>
Location*	<input type="text" value="Hub_None"/>
AAR Group	<input type="text" value="< None >"/>
Tunneled Protocol*	<input type="text" value="None"/>
QSIG Variant*	<input type="text" value="No Changes"/>
ASN.1 ROSE OID Encoding*	<input type="text" value="No Changes"/>
Packet Capture Mode*	<input type="text" value="None"/>
Packet Capture Duration	<input type="text" value="0"/>

Media Termination Point Required

Step 5. For the Inbound Calls settings, select the appropriate CSS which has access to the phones. Also, check the box Redirecting Diversion Header Delivery-Inbound.

Inbound Calls

Significant Digits*

Connected Line ID Presentation*

Connected Name Presentation*

Calling Search Space

AAR Calling Search Space

Prefix DN

Redirecting Diversion Header Delivery - Inbound

Step 6. For the Outbound Call settings, check the box Redirecting Diversion Header Delivery – Outbound.

Outbound Calls

Called Party Transformation CSS

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

Use Device Pool Calling Party Transformation CSS

Calling Party Selection*

Calling Line ID Presentation*

Calling Name Presentation*

Calling and Connected Party Info Format*

Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS

Use Device Pool Redirecting Party Transformation CSS

Customer Initials:

Step 7. In the **Destination Address** field, enter the IP address of the Mutare Voice (SAM) server to which the CUCM connects.

SIP Information

Destination

Destination Address is an SRV

Destination Address	Destination Address IPv6	Destination Port
1* 192.168.1.79		5060

Step 8. Select the **SIP trunk security profile** from the drop-down menu. Choose the new Security Profile created in Step 1.

SIP Trunk Security Profile Information

Name*

Description

Device Security Mode

Incoming Transport Type*

Outgoing Transport Type

Enable Digest Authentication

Nonce Validity Time (mins)*

X.509 Subject Name

Incoming Port*

Enable Application level authorization

Accept presence subscription

Accept out-of-dialog refer**

Accept unsolicited notification

Accept replaces header

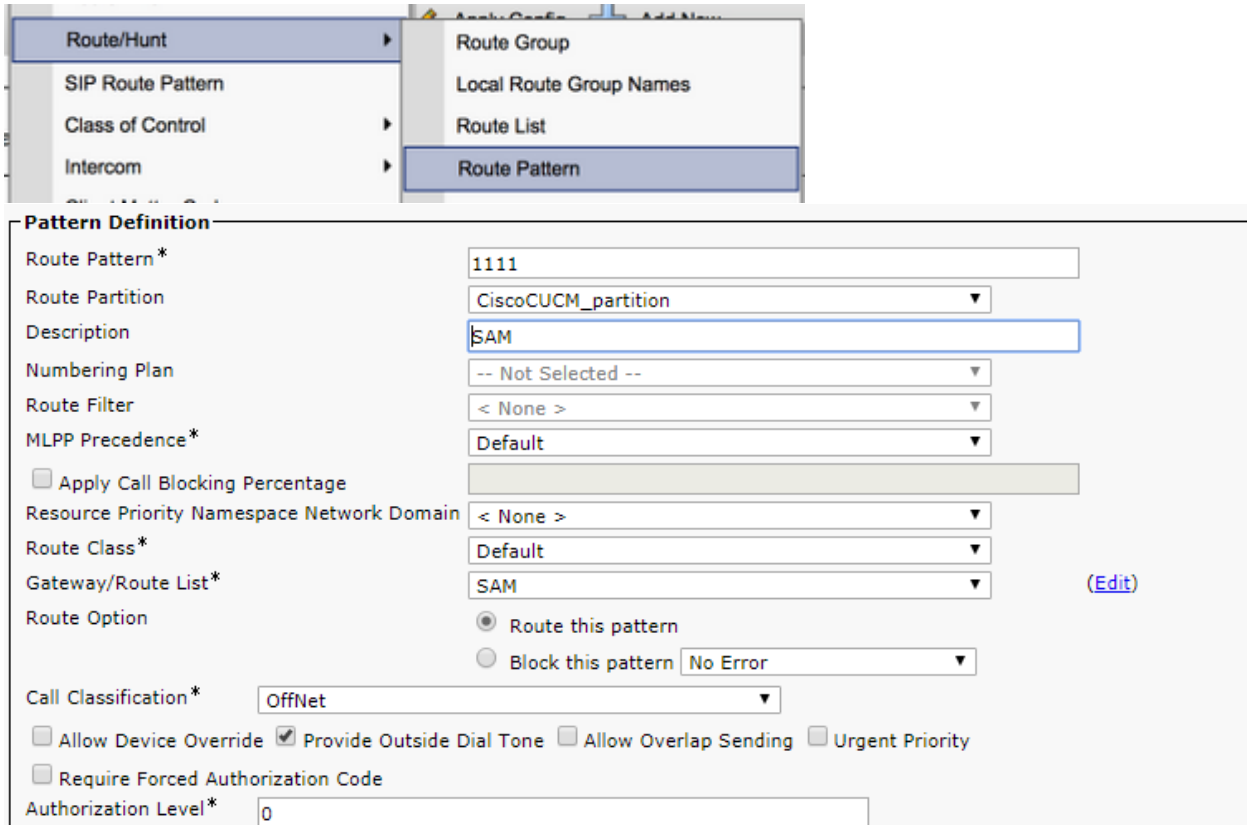
Transmit security status

Allow charging header

SIP V.150 Outbound SDP Offer Filtering*

Customer Initials:

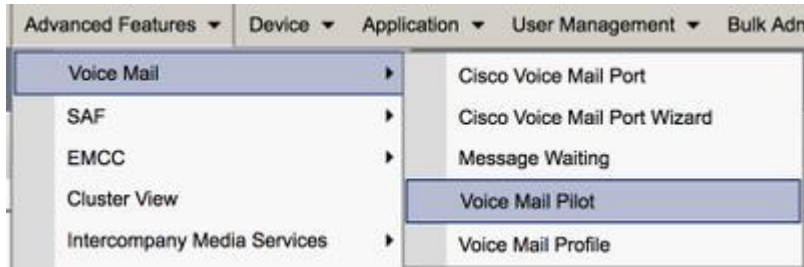
Step 9. Add a Route Pattern. Navigate to Call routing > Route/Hunt > Route Pattern. Click on add new and provide the voicemail pilot number for Mutare Voice (SAM) connection. This is the number users use to call into the Mutare Voice server. Select SAM as the Gateway/Route List mandatory field.



The screenshot shows the 'Route Pattern' configuration page in the Cisco Unified CM Administration interface. The 'Route Pattern' field is set to '1111'. The 'Route Partition' is set to 'CiscoCUCM_partition'. The 'Description' field is set to 'SAM'. The 'Gateway/Route List' is set to 'SAM'. The 'Route Option' is set to 'Route this pattern' with 'No Error' selected. The 'Call Classification' is set to 'OffNet'. The 'Authorization Level' is set to '0'. There are several checkboxes for 'Allow Device Override', 'Provide Outside Dial Tone', 'Allow Overlap Sending', 'Urgent Priority', and 'Require Forced Authorization Code'. An '(Edit)' link is visible next to the 'Gateway/Route List' dropdown.


Customer Initials:

Step 10. In order to add the Voicemail Pilot number, navigate to Advanced Features > Voicemail > Voicemail pilot.



Step 11. Click on Add new and provide the voicemail pilot number. This number must match the Route Pattern created in Step 12. You can choose to make this the Default voicemail pilot number for the entire CUCM cluster. In order to do this, check Make this the default voice mail pilot for the system.

Status

 Status: Ready

Voice Mail Pilot Information

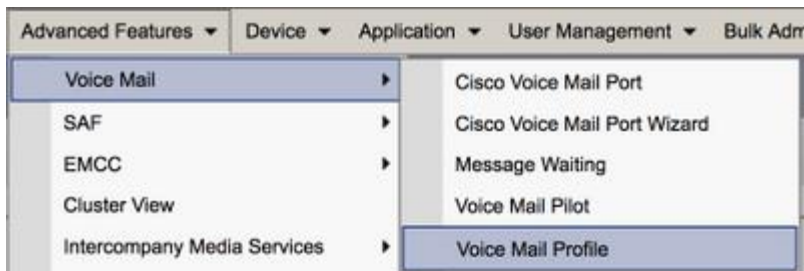
Voice Mail Pilot Number

Calling Search Space

Description

Make this the default Voice Mail Pilot for the system

Step 12. Add a voicemail profile for this voicemail system. Navigate to Advanced Features > Voicemail > Voice mail profile.



Customer Initials:

Step 13. Click on add new and provide an appropriate name. Choose the voice mail pilot created in Step 10. from the drop down. You can choose to make this the default voicemail profile for the system. In order to do this, check Make this the default voice mail profile for the system.

Voice Mail Profile Information

Voice Mail Profile: SAM (used by 1 devices)

Voice Mail Profile Name*:

Description:

Voice Mail Pilot**:

Voice Mail Box Mask:

Make this the default Voice Mail Profile for the System

Step 14. Assign Voicemail profile to device. Navigate to Devices > Phone > Selected Phone(s). Assign SAM voicemail profile to selected devices.

Directory Number*:

Route Partition:

Description:

Alerting Name:

ASCII Alerting Name:

Allow Control of Device from CTI

Associated Devices:

▼ ▲

Dissociate Devices:

Directory Number Settings

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

Calling Search Space:

Presence Group*:

User Hold MOH Audio Source:

Network Hold MOH Audio Source:

Auto Answer*:

Customer Initials: