

Configuration Example for CUCM Non-Secure SIP Integration with CUC

Contents

[Introduction](#)

[Prerequisites](#)

[Requirements](#)

[Components Used](#)

[Configure](#)

[Configuration on CUCM](#)

[Configuration on Unity Connection](#)

[Verify](#)

[Troubleshoot](#)

Introduction

This document describes the procedure to integrate Cisco Unified Communication Manager (CUCM) with Cisco Unity Connection (CUC) with the use of Session Initiation Protocol (SIP). In this example, the SIP integration is non-secure.

Prerequisites

Requirements

Cisco recommends that you have knowledge of these topics:

- CUCM
- CUC

Components Used

The information in this document is based on these software and hardware versions:

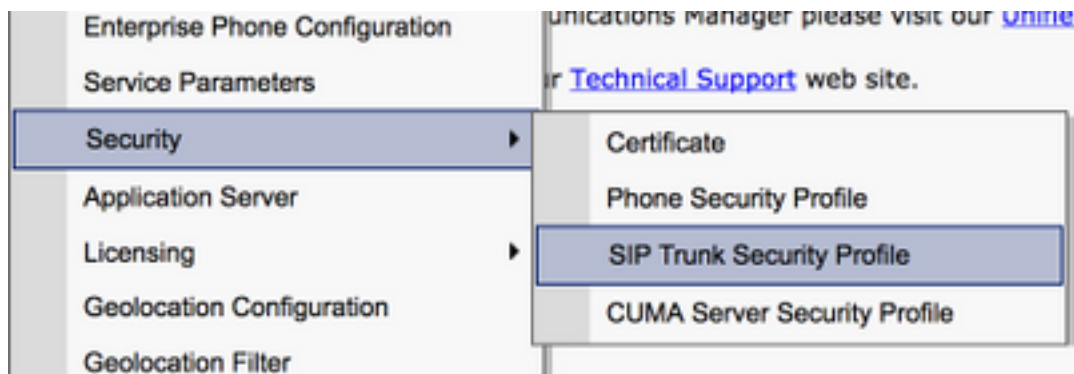
- CUCM 8.x and higher
- CUC 8.x and higher

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, ensure that you understand the potential impact of any command.

Configure

Configuration on CUCM

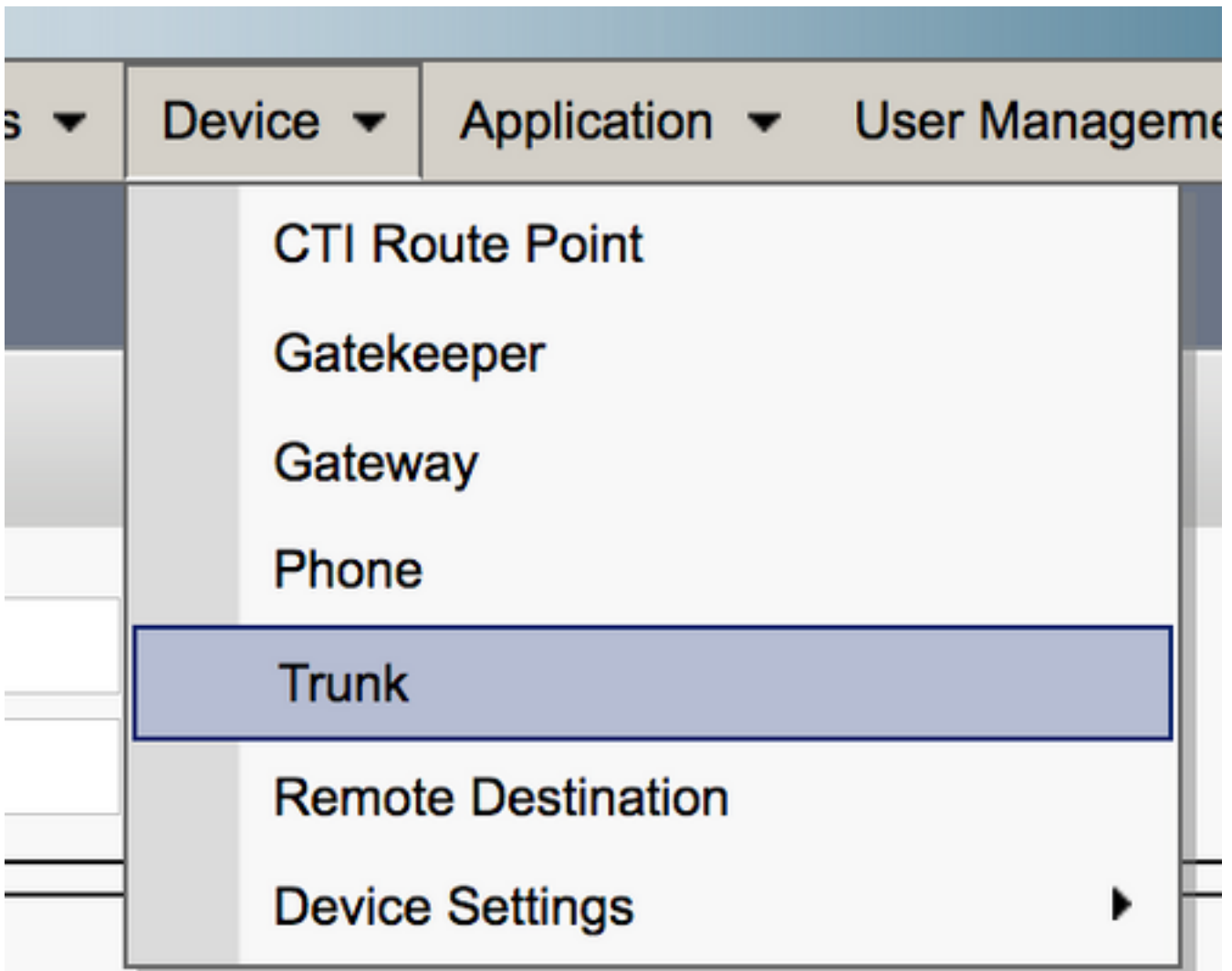
Step 1. On CUCM Admin page, navigate to **System > Security > SIP Trunk Security Profile**. Make a copy of the available profile. The default profile is **Non-Secure SIP Trunk Profile**. On the new profile, check these options; **Accept out-of-dialog refer**, **Accept unsolicited notification** and **Accept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



SIP Trunk Security Profile Information

Name*	Non Secure SIP Trunk Profile--Unity
Description	Non Secure SIP Trunk Profile authenticated by null S
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

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)

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

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Unity-trunk
Description	
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0

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*	All
Connected Line ID Presentation*	Default
Connected Name Presentation*	Default
Calling Search Space	< None >
AAR Calling Search Space	< None >
Prefix DN	
<input checked="" type="checkbox"/> 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	< None >
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	
Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling and Connected Party Info Format*	Deliver DN only in connected party
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Outbound	
Redirecting Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Redirecting Party Transformation CSS	

Step 7. In the **Destination Address** field, enter the IP address of the Unity Connection server to which the CUCM connects.

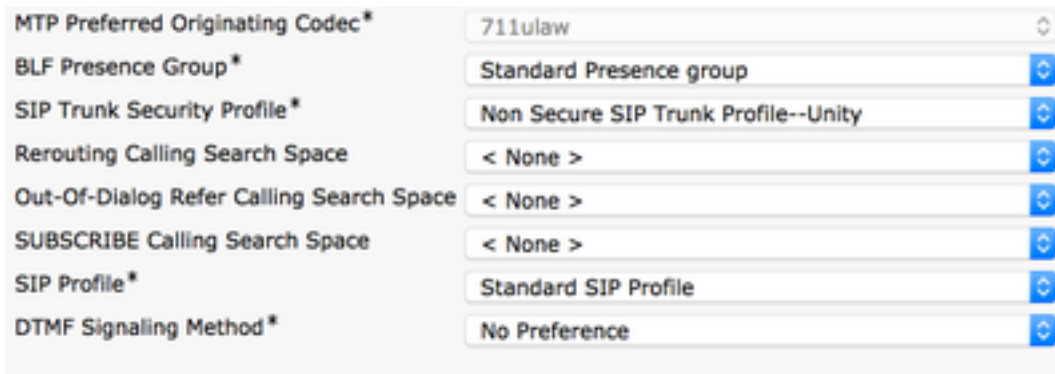
Destination

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	10.127.226.5		5060

Note: For a Unity Connection cluster (Publisher and Subscriber), create 2 SIP trunks. Each SIP trunk points to one Unity Connection server.

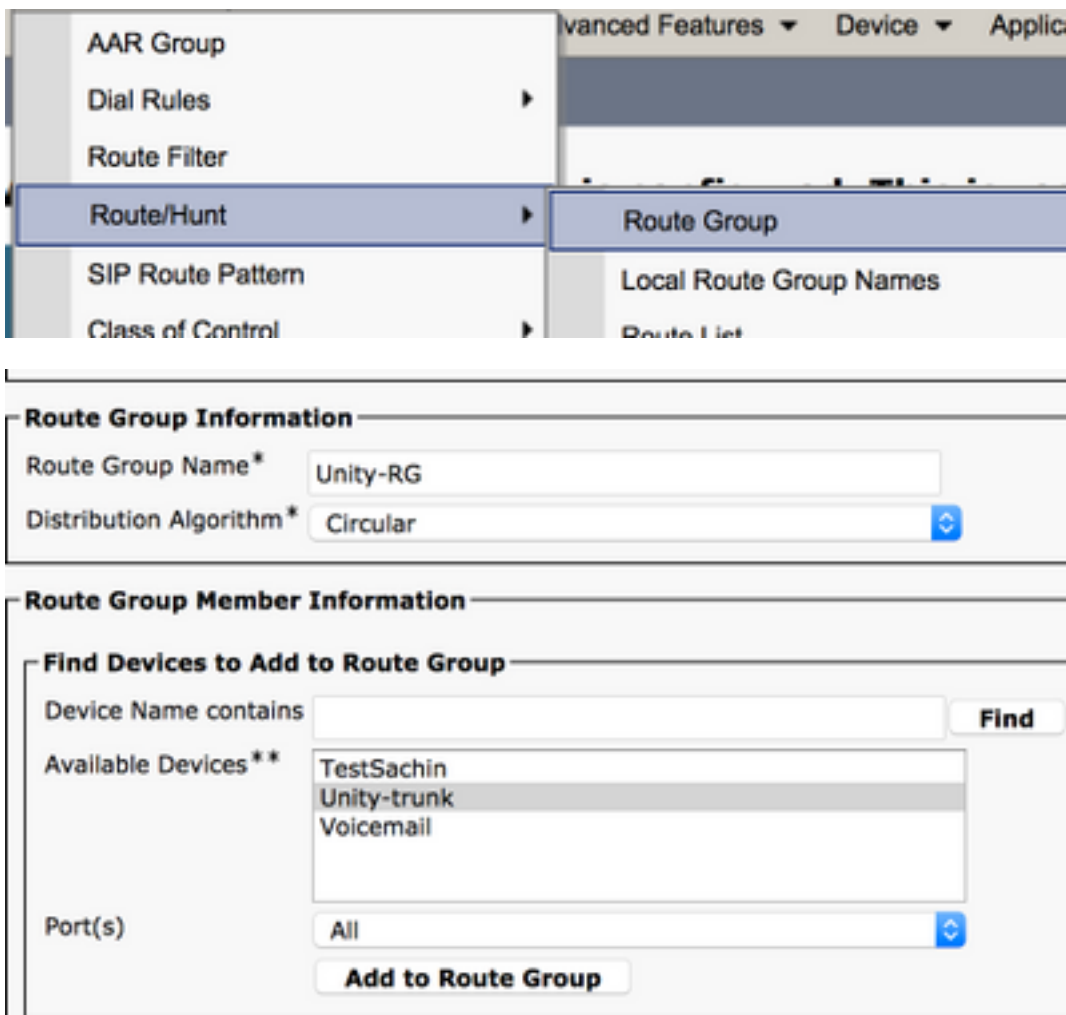
Step 8. Select the **SIP trunk security profile** from the drop down menu. Choose the new Security Profile created in Step 1. Select the **Rerouting CSS**. This CSS comes into picture for calls transferred back to the CUCM from the Unity Connection and must have access to the user phones. For **SIP Profile**, select the **Standard SIP Profile** from the drop down.



The screenshot shows a configuration page for a SIP Trunk with the following settings:

MTP Preferred Originating Codec*	711ulaw
BLF Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile--Unity
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile
DTMF Signaling Method*	No Preference

Step 9. Create a Route Group. Navigate to **Call Routing > Route/Hunt > Route Group**. Add a new Route Group and give it an appropriate name. Select the SIP Trunk created in Step 2 and click on **Add to Route Group**. Hit **Save**.



The screenshot shows the configuration page for a Route Group, divided into two sections:

Route Group Information

- Route Group Name*: Unity-RG
- Distribution Algorithm*: Circular

Route Group Member Information

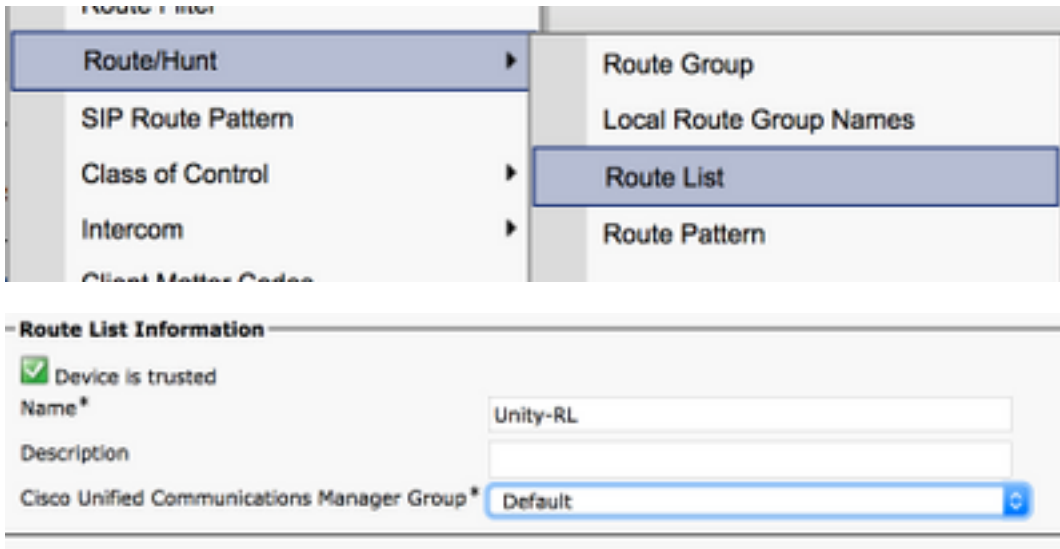
Find Devices to Add to Route Group

- Device Name contains: [Empty field] **Find**
- Available Devices**: TestSachin, Unity-trunk, Voicemail
- Port(s): All
- Add to Route Group**

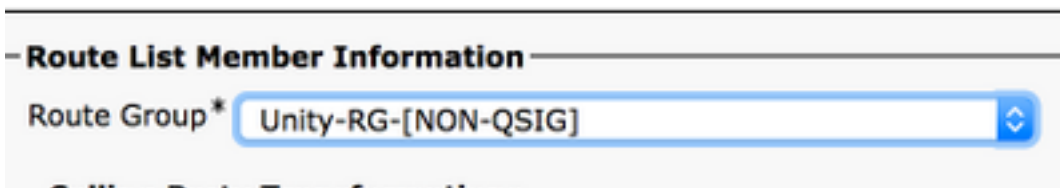
Note: In case of Unity Connection cluster, you can add a separate Route Group for the second SIP Trunk created. Alternatively, you can choose to add the second SIP trunk to the

same Route Group. The order is selected from the **Distribution Algorithm** drop down menu: Circular, Round Robin, etc.

Step 10. Create a Route List. Navigate to **Call Routing > Route/Hunt > Route List**. Click on **Add new** and give an appropriate name to the Route List. Select the **CUCM Group** from the drop down menu which contains the CUCM servers to which the CUC server establishes a SIP trunk connection.

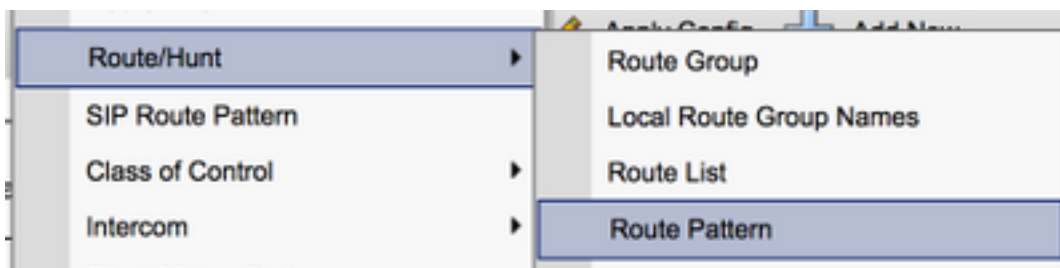


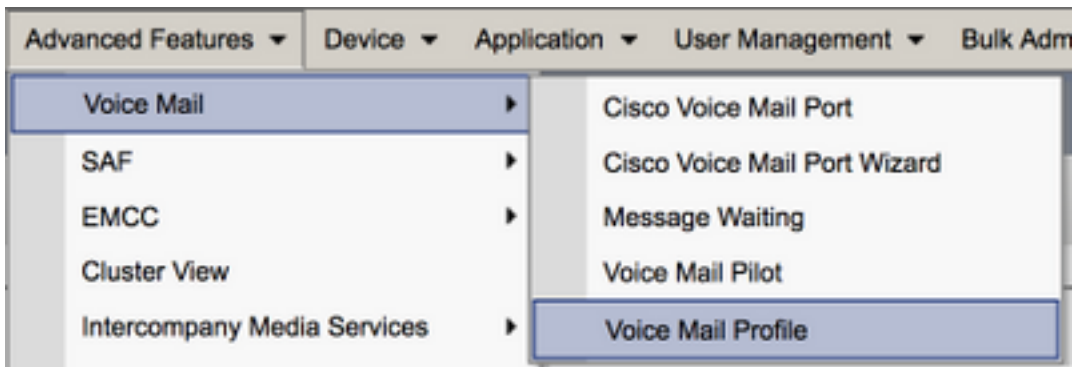
Step 11. Click **Save**. Post this there is an option to select a **Route group** for this Route List. Click on **Add Route Group** and select the Route group you created in Step 9.



Note: If you create multiple Route Group, each for one trunk, select all Route Groups and arrange them in order of preference. CUCM selects the route group at the top first to route the call.

Step 12. Add a **Route Pattern**. Navigate to **Call routing > Route/Hunt > Route Pattern**. Click on **add new** and provide the voicemail pilot number for unity connection. This is the number users use to call into the Unity connection server. Select the Route List created in Step 10 from the drop down option **Gateway/Route List**.





Step 16. Click on **add new** and provide an appropriate name. Choose the voice mail pilot created in Step 13. 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.**

A screenshot of a web form titled 'Voice Mail Profile Information'. The form contains the following fields and options:

- 'Voice Mail Profile Name*': A text input field containing 'Unity-Profile'.
- 'Description': An empty text input field.
- 'Voice Mail Pilot**': A dropdown menu showing '3000/< None >'.
- 'Voice Mail Box Mask': An empty text input field.
- A checkbox labeled 'Make this the default Voice Mail Profile for the System' which is currently unchecked.

Configuration on Unity Connection

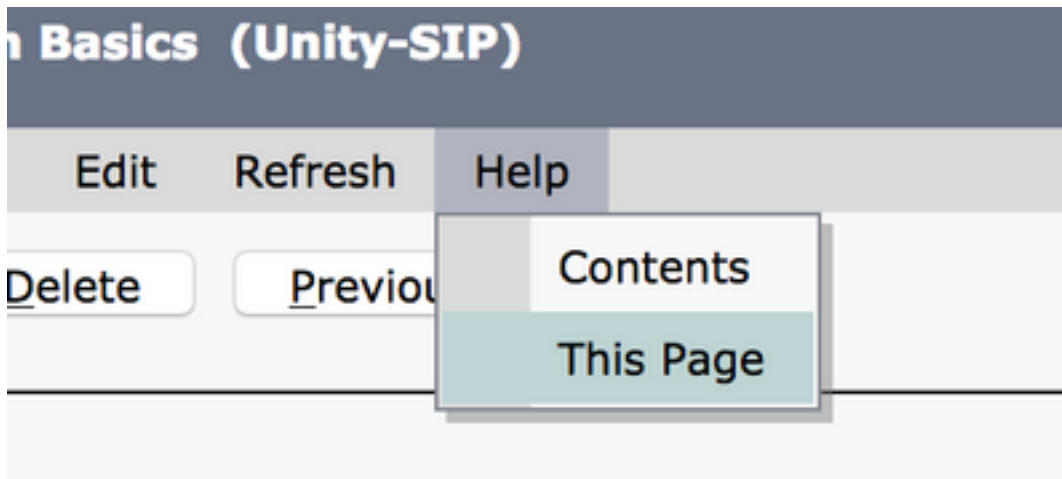
Step 1. Navigate to CUC Admin page and expand **Telephony Integration**. Select the first option, **Phone System**.

Step 2. Click on **Add New** and give the Phone System a name.

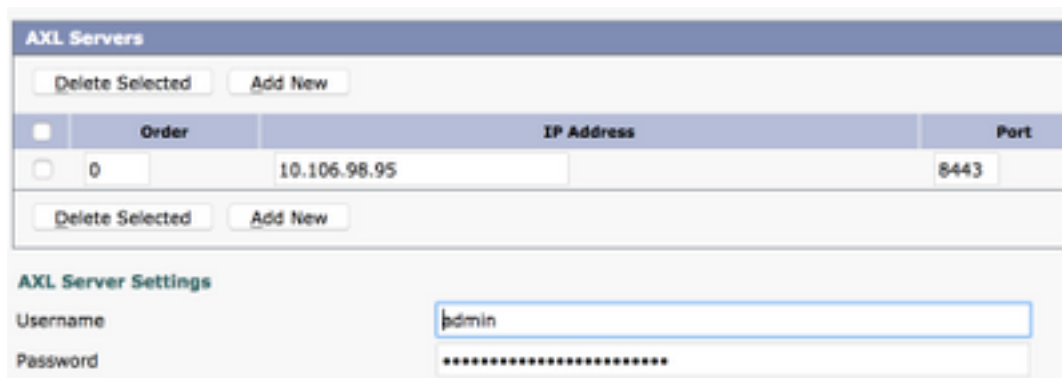
A screenshot of a web form titled 'Phone System'. The form contains the following elements:

- The title 'Phone System' in a large, bold, teal font.
- 'Phone System Name*': A text input field containing 'Unity-SIP'.
- A 'Save' button with a blue border and shadow.

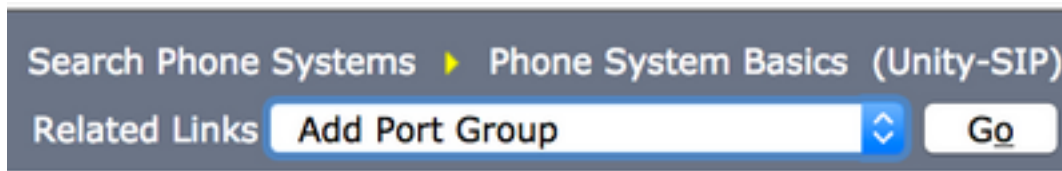
Step 3. The defaults are used on the Phone System Basics page. In order to view information about the additional configuration for the Phone System, navigate to **Help > This page**.



Step 4. [Optional] In order to import CUCM users to CUC, configure AXL servers on the Phone System. Navigate to **Edit > Cisco Unified Communications Manager AXL server**.



Step 5. Navigate back to the Phone System basic page. On the top right corner, select **Add a Port Group** from the related links menu.



Step 6. Create a Port Group. Provide a Display Name for the Port Group. Change the **Port Group type** to SIP. Enter the FQDN/IP address of the CUCM server to which this SIP trunk registers to.

New Port Group

Phone System

Create From Port Group Type Port Group

Port Group Description

Display Name*

Authenticate with SIP Server

Authentication Username

Authentication Password

Contact Line Name

SIP Security Profile

SIP Transport Protocol

Primary Server Settings

IPv4 Address or Host Name

IPv6 Address or Host Name

Port

Step 7. Go to Related Links on the top right corner and select **Add Ports**.

Search Port Groups ▶ Port Group Basics (Unity-SIP-1)

Related Links

Step 8. Enter the number of ports desired. Select the appropriate **Phone System** and **Port Group** name and hit **save**.

New Phone System Port

Enabled

Number of Ports

Phone System

Port Group

Server

Port Behavior

Answer Calls

Perform Message Notification

Send MWI Requests (may also be disabled by the port group)

Allow TRAP Connections

Note: From the **Server** drop down menu, select the Publisher CUC server and create ports. To add ports for the Subscriber CUC server, navigate to the same Port Group **Unity-SIP-1** and choose **Add Ports** from the **Related Links** menu on the top right corner. On the **New Phone System Port** page, choose the Subscriber server from the **Server** drop down menu. Alternatively, create a new port group in the same Phone System with a different device name prefix for the Subscriber ports.

Step 9. Navigate back to **Telephony Integration > Port Group** and select the SIP Port group. Navigate to **Edit > Server** and add the additional CUCM servers in the same cluster for failover. Assign a preference with the help of **Order** number. Order 0 has highest preference followed by 1, 2 and so on. The ports register to the CUCM server with Order 0. If this server is not available, the ports register to the subsequent servers in the list.

Check the **Reconnect to a Higher-Order Cisco Unified Communications Manager When Available** for the ports to fall back to the higher order CUCM server once it becomes available. Otherwise, the ports remain registered to the lower preference server.

Verify

Use this section in order to confirm that your configuration works properly.

If the ports are unregistered,

Step 1. Check if the ports are successfully created on the Unity Connection. Navigate to **Telephony Integration > Ports**.

<input type="checkbox"/>	Unity-SIP-1-001	Unity-SIP	cuc1052	X	X	X	X	X
<input type="checkbox"/>	Unity-SIP-1-002	Unity-SIP	cuc1052	X	X	X	X	X
<input type="checkbox"/>	Unity-SIP-1-003	Unity-SIP	cuc1052	X	X	X	X	X
<input type="checkbox"/>	Unity-SIP-1-004	Unity-SIP	cuc1052	X	X	X	X	X
<input type="checkbox"/>	Unity-SIP-1-005	Unity-SIP	cuc1052	X	X	X	X	X

Step 2. Navigate to **Telephony Integration > Port Group**. Select the SIP Port Group. In the Related Links drop-down list, select **Check Telephony Configuration** and select **Go** to confirm the phone system integration settings. If the test is not successful, the Task Execution Results displays one or more messages with troubleshoot steps. Correct the problem and test the connection again.

Troubleshoot

There is currently no specific troubleshooting information available for this configuration.