CALLISTO

# PLATFORM

## CONFIGURATION OF CISCO UCM

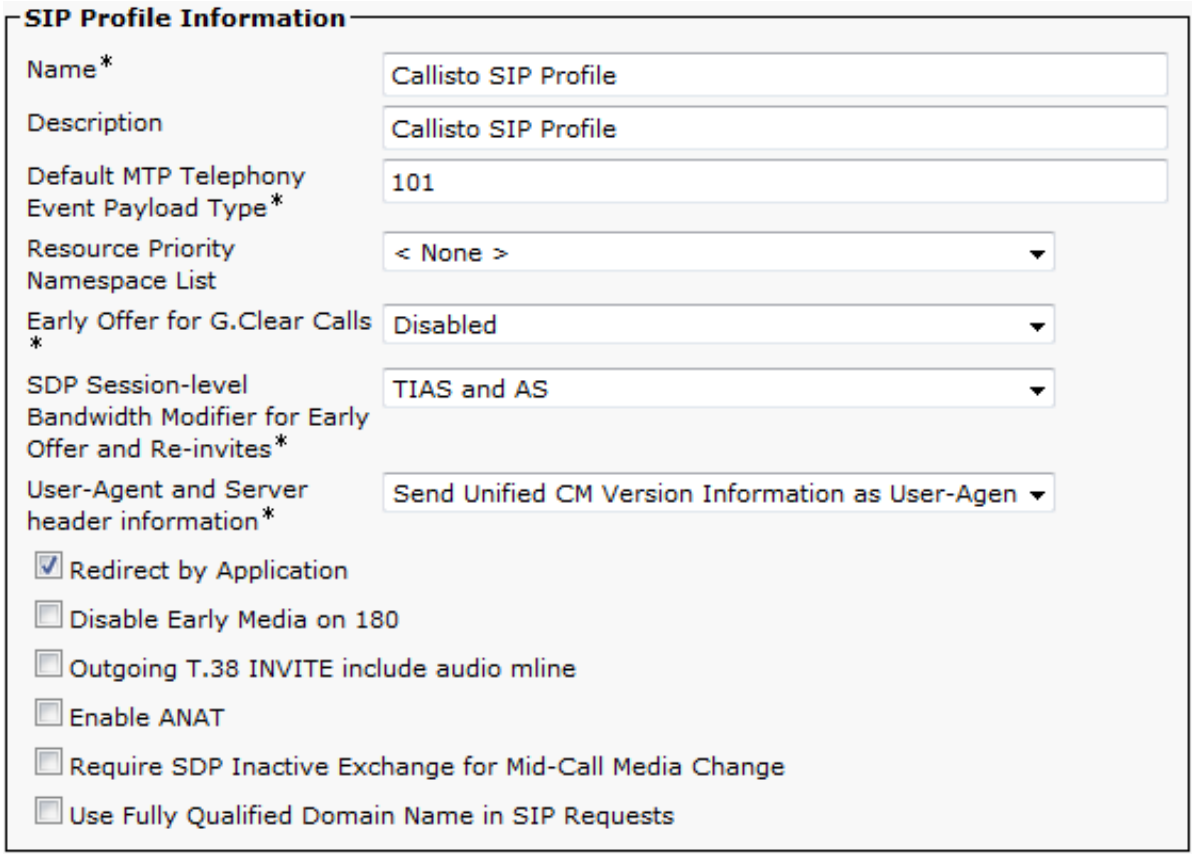
Table Of Contents

**Configuration of Cisco UCM** ..... 3
SIP Profile configuration ..... 3
SIP Trunk configuration ..... 5
Configuration of route pattern ..... 7
Configuration of Callisto services ..... 8
Configuration of Voice Mail Pilot number ..... 9
Configuration of MWI numbers ..... 10
Add application user ..... 11
Activating services ..... 14
Activating CDRs ..... 15
Configure recording ..... 16
Configure External Call Control ..... 20
Enable Transport Layer Security (TLS) ..... 24
Copyright Information, Disclaimer ..... 26

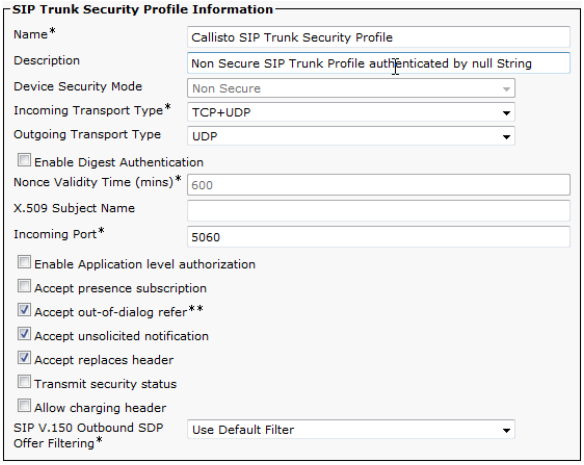
# SIP Profile configuration

SIP is the VoIP protocol of the Callisto Platform. Via a SIP Profile, general SIP settings can be configured.

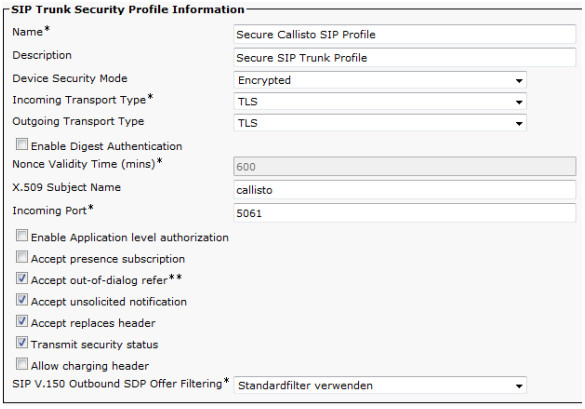
1. Add a new SIP profile or copy the Standard SIP Profile via Device > Device Settings > SIP Profile.




   Check Redirect by Application.
2. Add a new SIP Trunk Security Profile via System > Security > SIP Trunk Security Profile.
   For non-secure communication: For encrypted communication:




It is crucial to set Outgoing Transport Type to UDP.

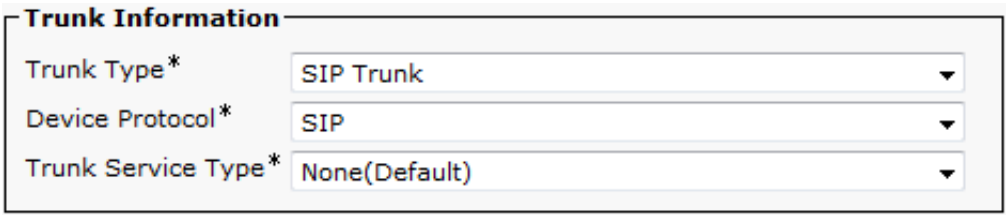



You can use the settings as seen as on the screenshot.

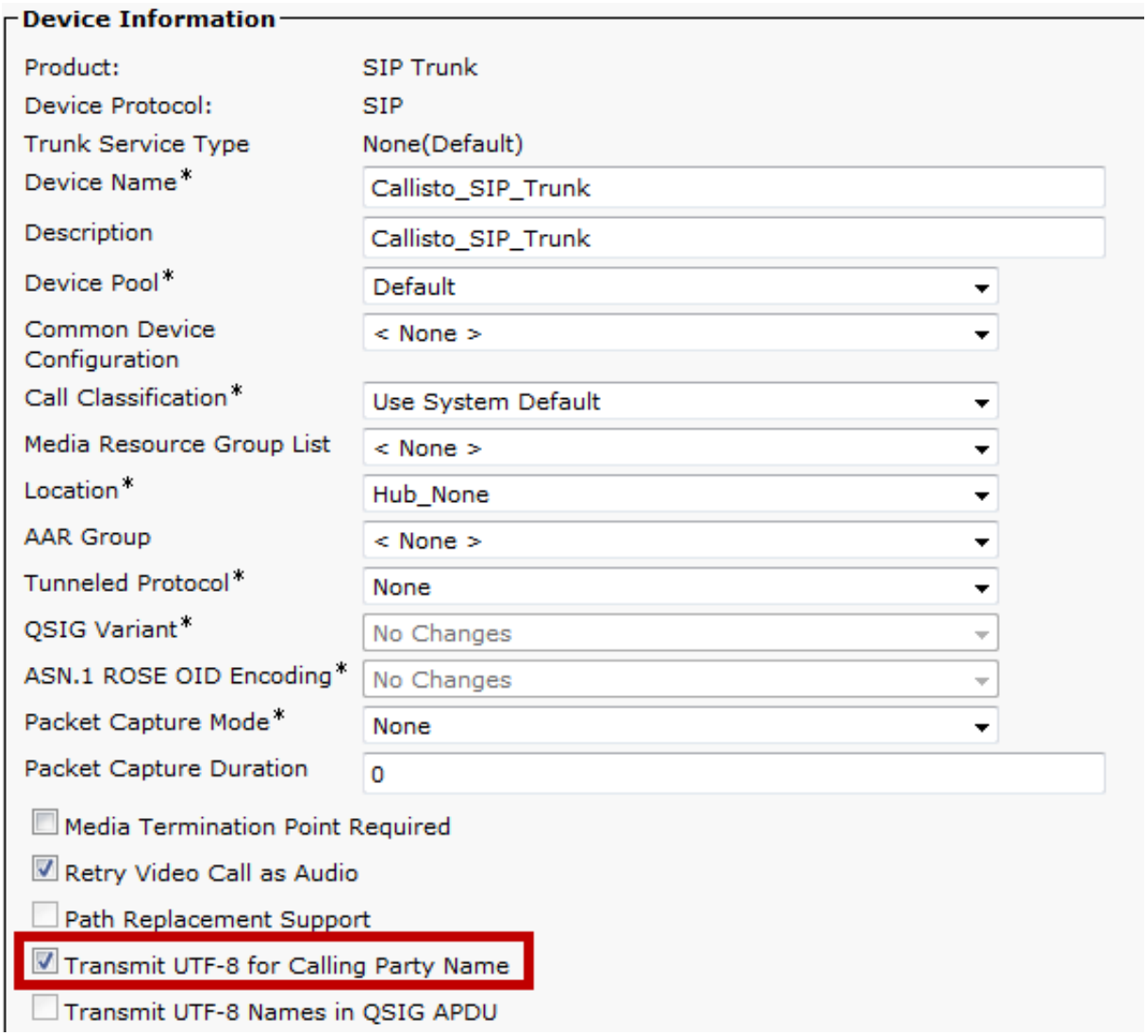
# SIP Trunk configuration

Via a Cisco Unified Communications Manager SIP Trunk, incoming calls are forwarded to Callisto, where outgoing calls will be set up.

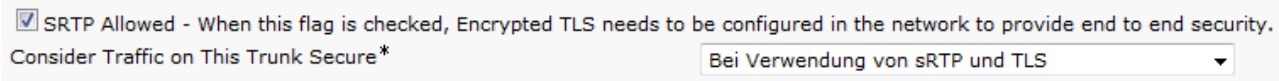
Add a new SIP Trunk via Device > Trunk




Check Transmit UTF-8 for Calling Party Name



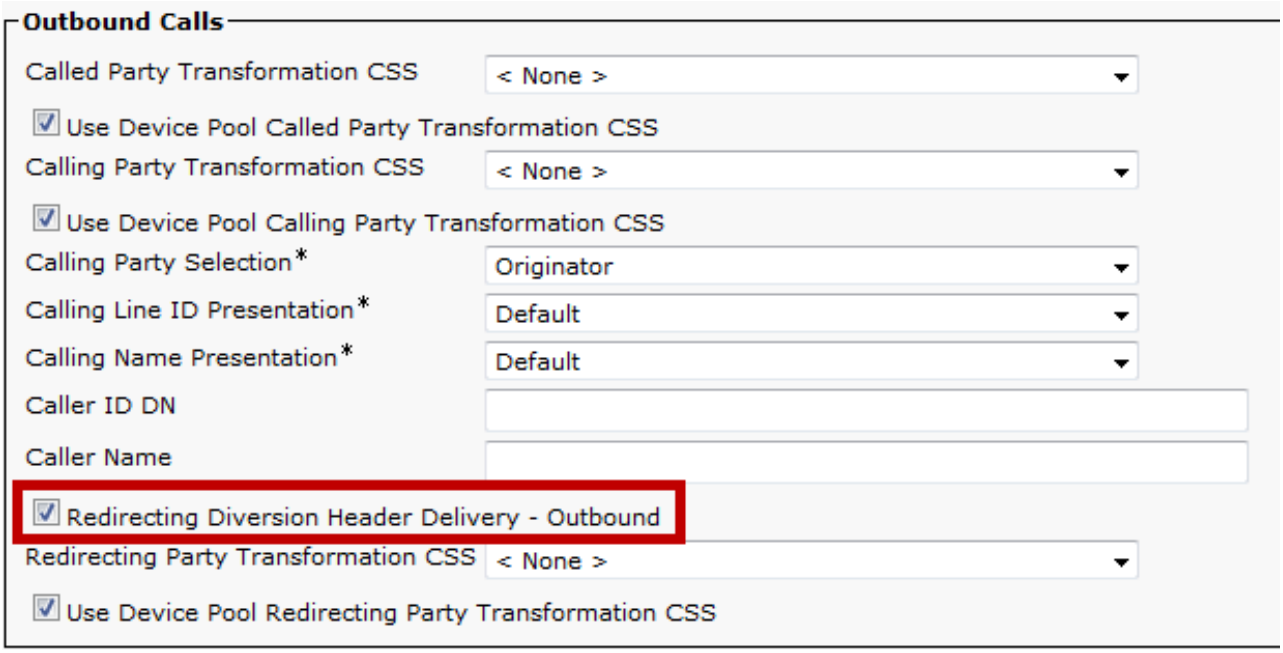

For Encrypted Communication: Check the SRTP option.




At Call Routing Information, check Remote-Party-ID and Asserted-Identity.

At Inbound Calls, check Redirecting Diversion Header Delivery – Inbound.

At Outbound Calls, check Redirecting Diversion Header Delivery – Outbound.




Via SIP Information, enter the Callisto IP address at Destination Address.

For non-secure communication, set Destination Port to 5060.

To encrypt the communication, set this port to 5061.

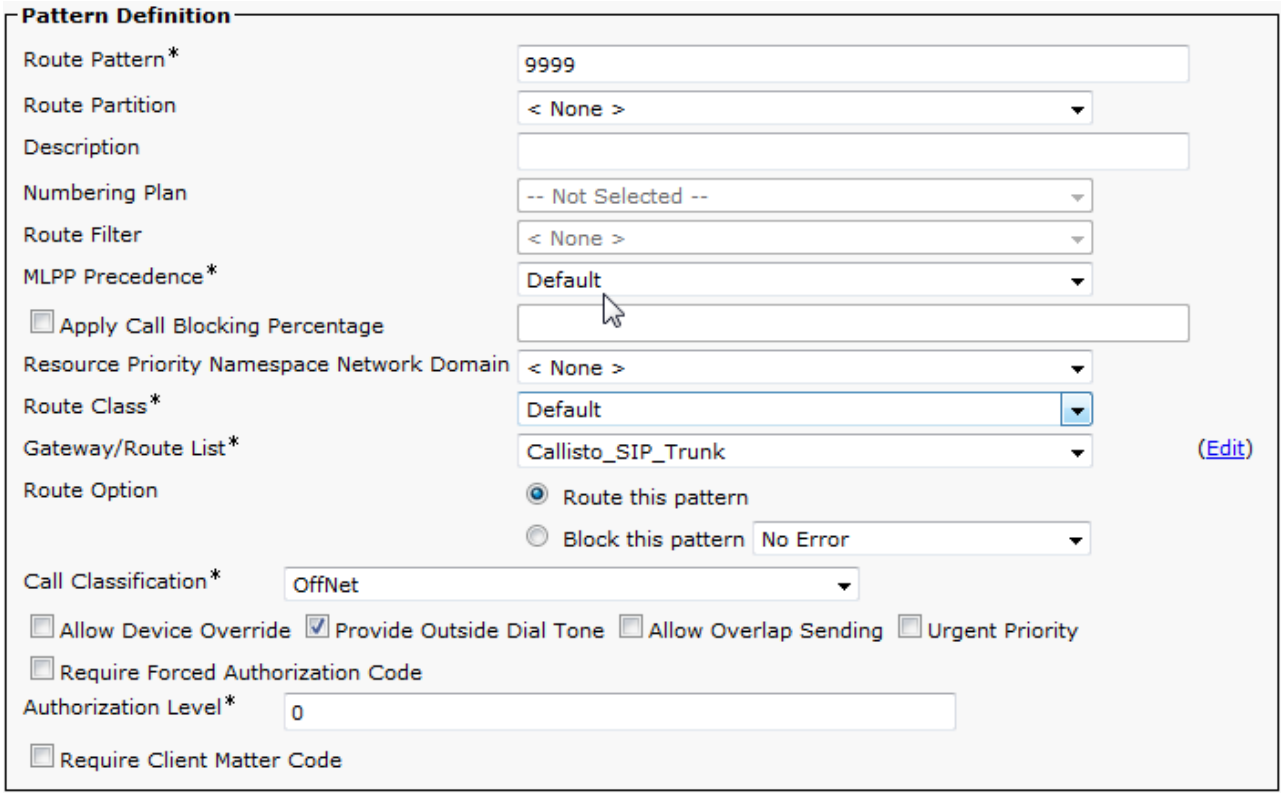
At SIP Profile, select the profile you have setup in point 1.1.

At SIP Trunk Security Profile, select the profile you have setup in point 1.2.

All Calling Search Space related settings have to be configured according the overall CUCM configuration.

# Configuration of route pattern

Callisto terminated internal numbers can be defined via Route Patterns. This applies to numbers of Voice Mail, Conference and OIM applications, and others.




Select Call Routing > Route > Hunt > Route Pattern and add a new route pattern

Enter the internal voice mail number in the field Route Pattern.
This number must be identical to the number entered in the Callisto system at System > System Parameter.

Select the newly configured SIP Trunk via Gateway > Route List.

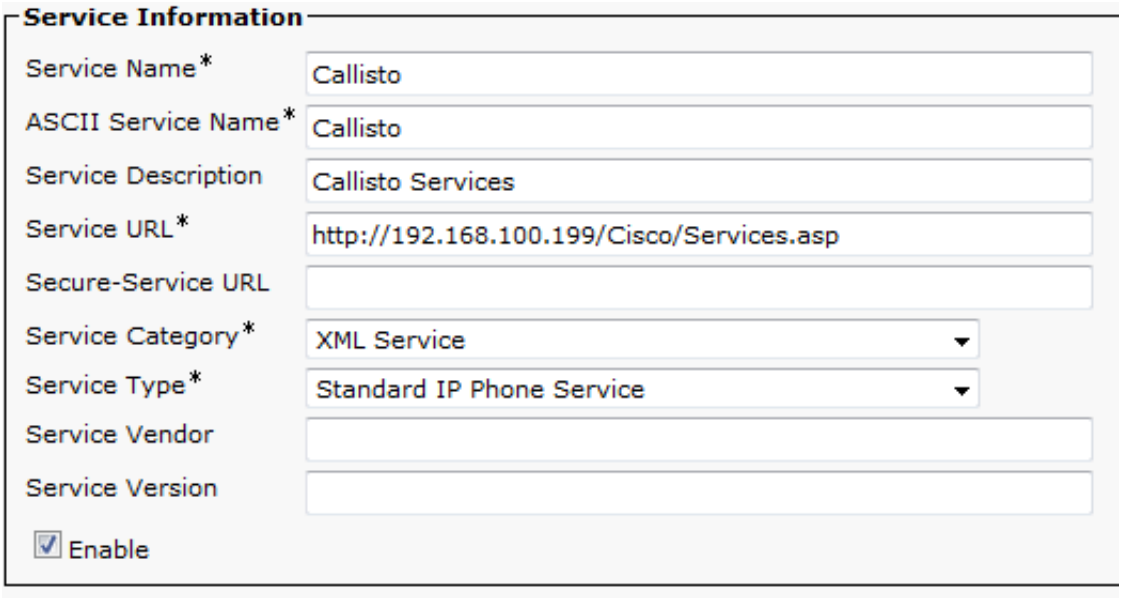
Use wildcards for the route pattern number if you want to define a range of numbers.

- X: single digit (e.g., 99XX)
- !: any digit (e.g., 99!)

# Configuration of Callisto services

The settings in this step activates the Callisto Phone Services at the Cisco IP phones.

1. Go to Device > Device Settings > Phone Services
2. Add a new service
3. for the service URL use: http://*<Callisto IP address>*/cisco/services.asp



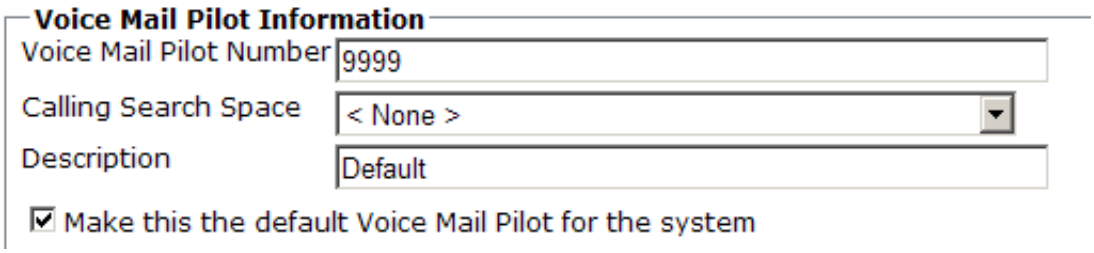

Subscribe this service to the phone.
You can either subscribe it directly in Device > Phones > Subscribe/Unsubscribe Services or trough a profile at Device > Device Settings > Device Profile > Subscribe Services

Set the Service Provisioning to internal or both, depending on your needs. This can be done in the phone settings directly or as above in the Device Profile.

# Configuration of Voice Mail Pilot number

The settings in this step define the Callisto Voice Mail number in the Cisco Unified Communications Manager. Also, the Voice Mail keys on Cisco's IP phones are configured for Callisto.

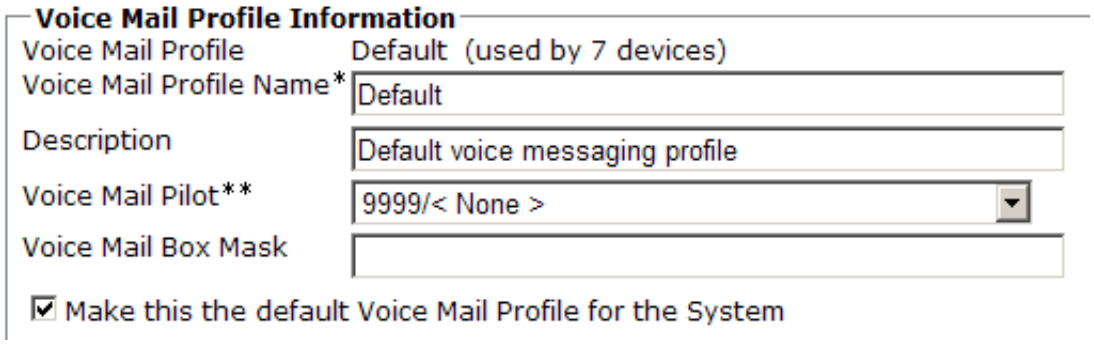
1. Open Default via Voice Mail > Voice Mail Pilot.
2. Enter the Callisto Voice Mail number.
3. Check Make this the default Voice Mail Pilot for the system




This number must be identical to the Voice Mail number entered in the Callisto web menu via System > System Parameter.

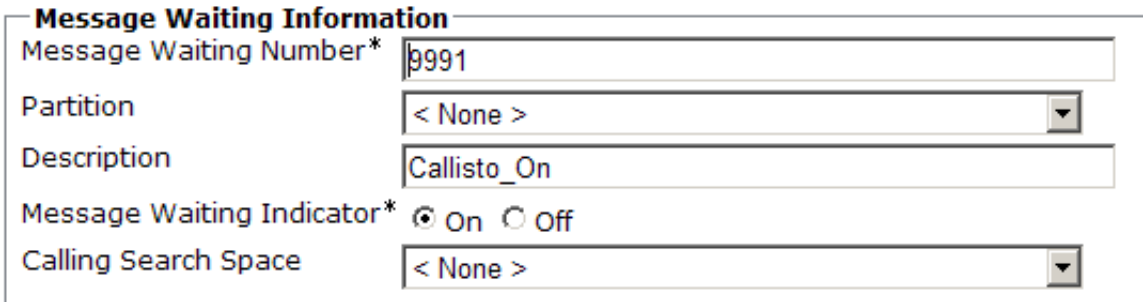
Open Default via Voice Mail > Voice Mail Profile.

Select the the new configured Voice Mail Pilot at Voice Mail Pilot.



# Configuration of MWI numbers

Add Message Waiting numbers via Voice Mail > Message Waiting. These are used by Callisto to switch the MWI lamp on and off.




1. Select Add New.
2. Enter a free internal number and set Message Waiting Indicator to On.
3. Enter another number and set Message Waiting Indicator to Off.

These two numbers must be entered in the Callisto system at System > System Parameter.

# Add application user

Callisto needs an application user to access the Cisco Unified Communications Manager.

Callisto uses three main access modes.

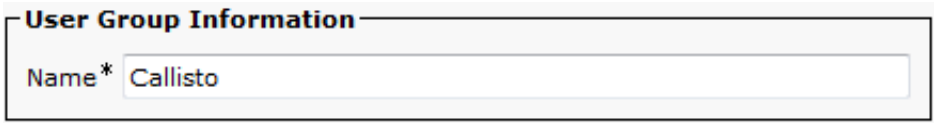
- AXL access
- Phone Web Access
- CTI with JTAPI (COC proxy option)

Within the Callisto appliance system, there is no need to distinguish between those connection methods. Therefore, the following description is simplified by exemplifying the configuration of one application user with one user group and the corresponding roles.

This user is used in Callisto for both accessing CUCM and for phone authentication. The COC proxy user is applied automatically. Please refer to the Callisto and COC administration manuals.

## Add user group

Go to User Management > User Group and add a new group.




## Assign roles

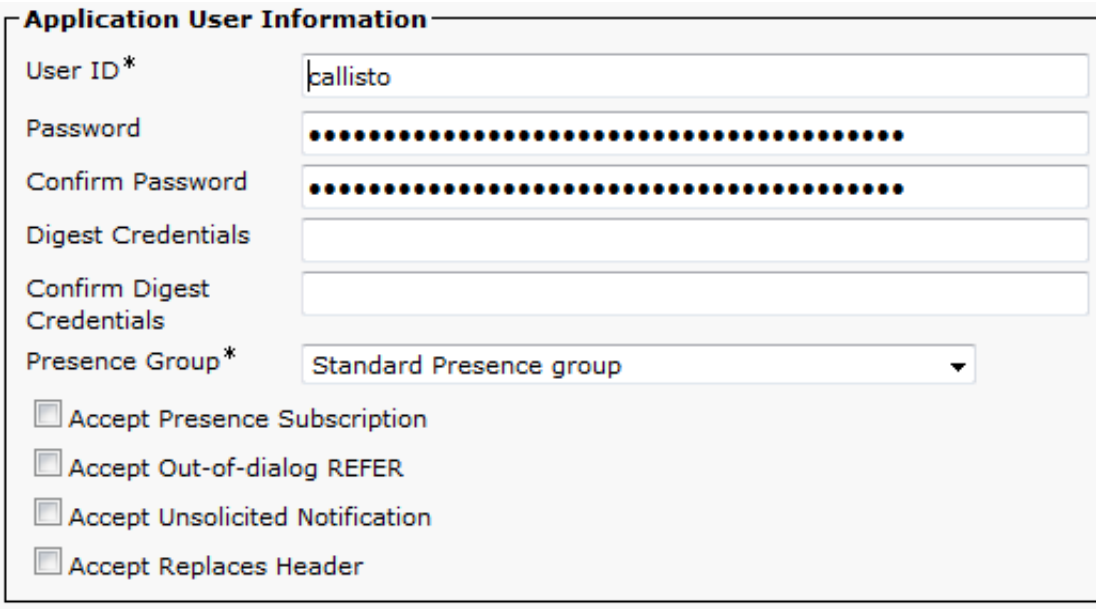
The following roles can be assigned to the user group:

| Role | Description |
|---|---|
| Standard AXL API Access | Grants access to the AXL database API. |
| Standard CCM Admin Users | Needed to get extended information from phones. |
| Standard EM Authentication Proxy Rights | Grant access to extension mobility logon information. |
| Standard RealtimeAndTraceCollection | Grants access to the phone status. |
| Standard CTI Enabled* | Enables CTI application control. |
| Standard CTI Allow Control of Phones supporting Connected Xfer and conf* | Allows control of all CTI devices that support connected transfer and conferencing. |
| Standard CTI Allow Call Park Monitoring* | This role is needed for the parking feature of COC. It should be added even if the parking feature is not used. |

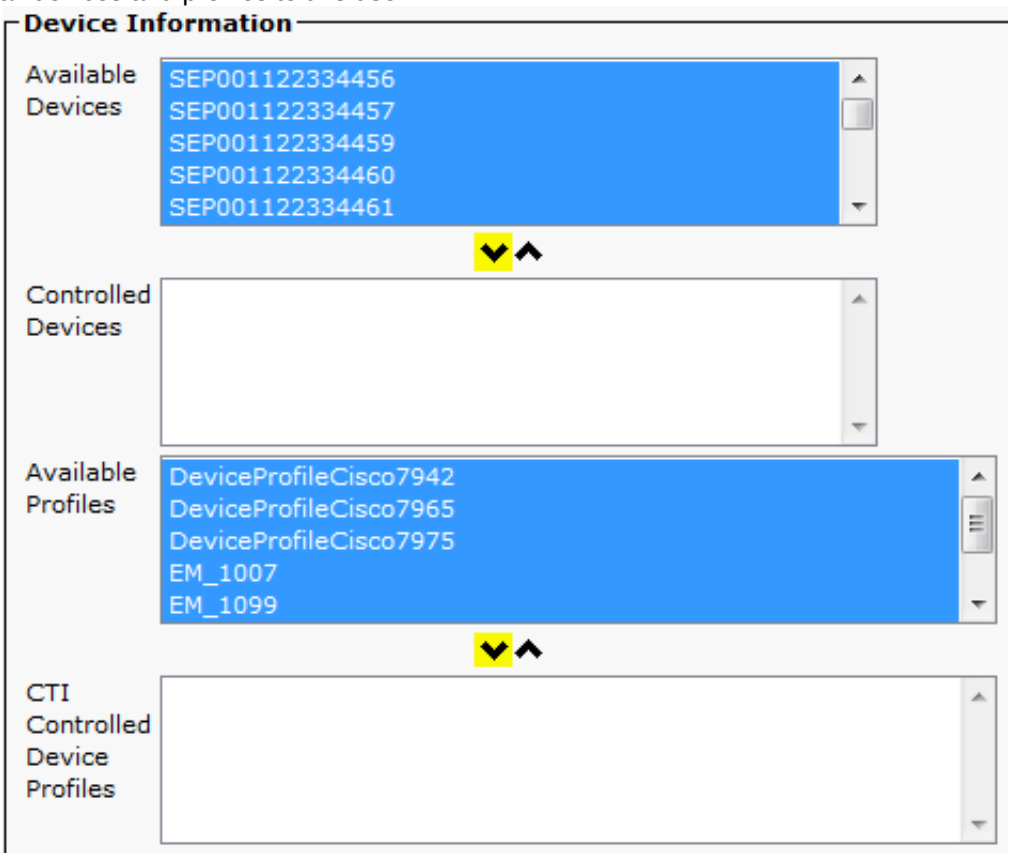
*Those roles are available if the COC Proxy option is active on Callisto.

## Add application user

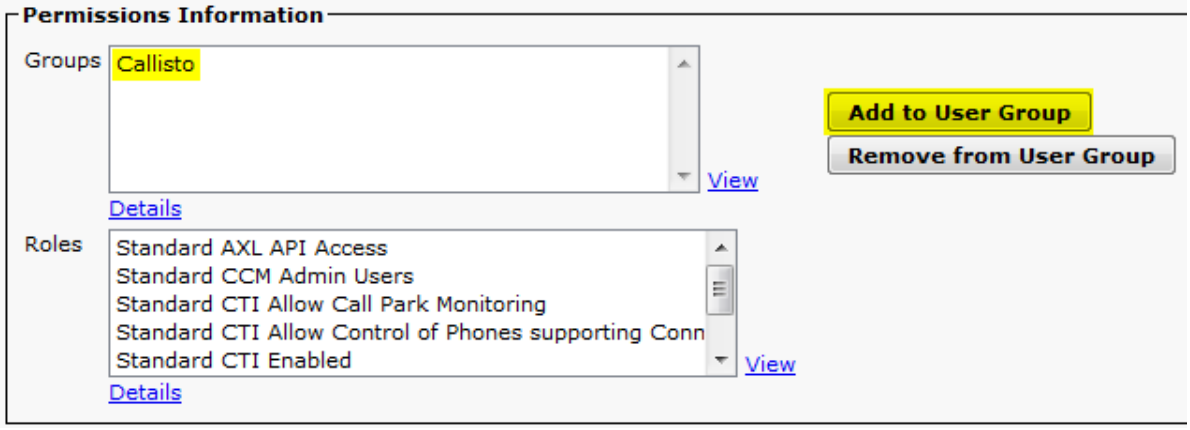
1. Go to User Management > Application User to add a new user.




2. Assign all devices and profiles to this user.




   To select all entries, click on the first entry and then press *shift-end* on your keyboard to extend the selection to the last entry.
3. Add the group you configured above.

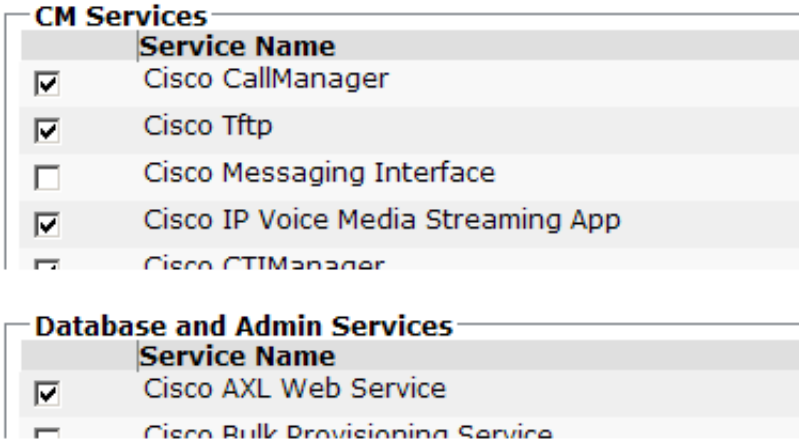



The roles will be visible after adding the group.

# Activating services

Change to Cisco Unified CallManager Serviceability.

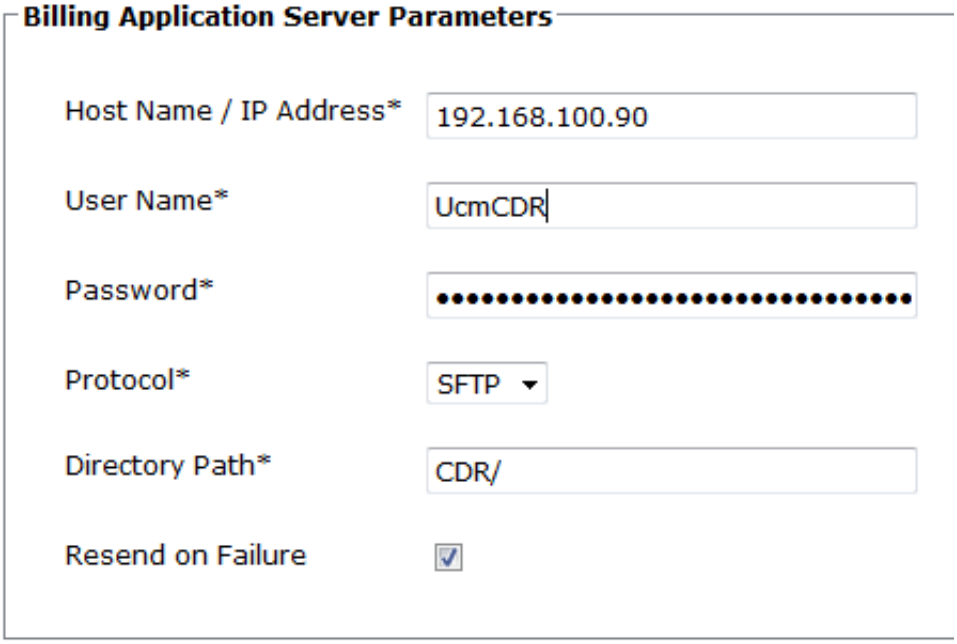
1. Open Tools > Service Activation.
2. Activate both Services Cisco IP Voice Media Streaming App and Cisco AXL Web Service.



# Activating CDRs

Stay at Cisco Unified CallManager Serviceability.

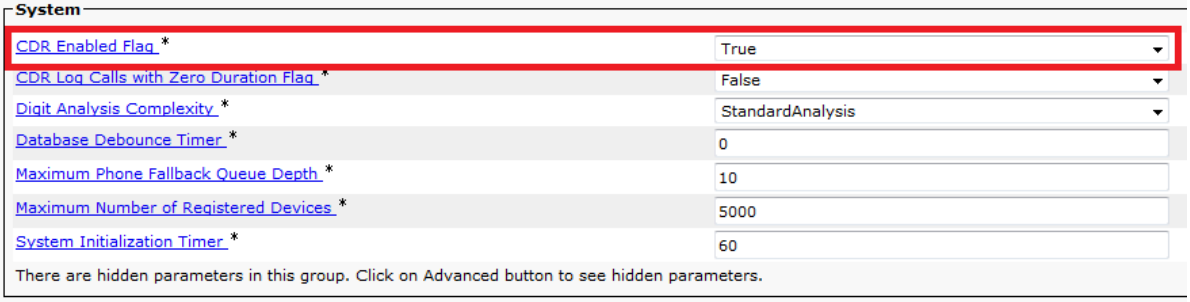
1. Go to Tools and choose CDR Management.
2. Create a new entry as following:




   The reports should be enabled and configured in Callisto before you make these settings.
3. Enter the Callisto IP address and configure the credentials, which you have already entered in Callisto Reports > Settings.
4. The protocol must be set to SFTP and Resend on Failure can be left set.

Go to Cisco Unified CM Administration.

1. In System / Service Parameters, choose your CUCM Server and select Cisco CallManager.
2. Enable the flag CDR Enabled Flag.



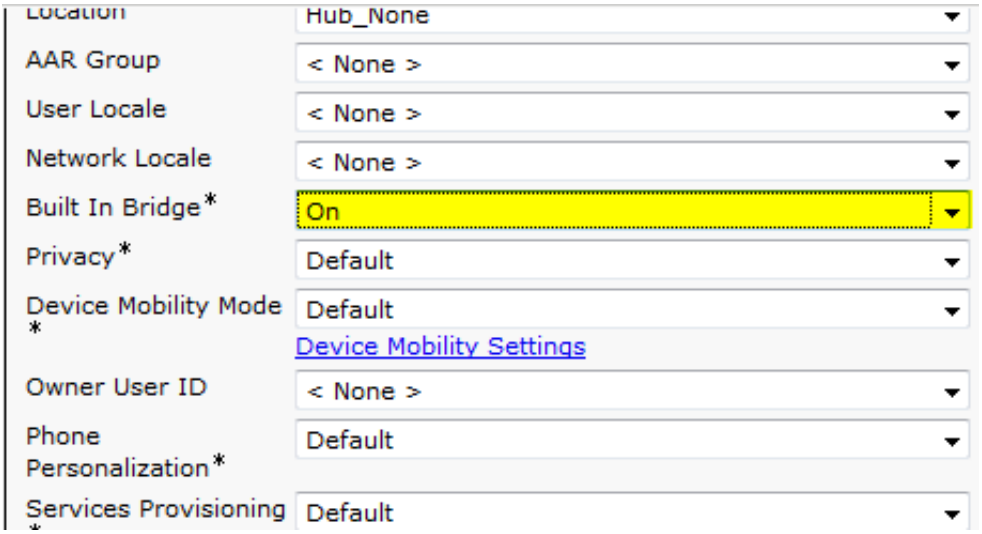
# Configure recording

Following steps are necessary if you want to use the Callisto Recording feature.

On Cisco Unified Communictions Manager V10.x or later mobility and off-network conversations can be captured using Network-recording. Please refer to the Cisco Recording documentation.

## Turn on IP phone BIB to allow recording

The Built In Bridge of the agent phone must be set to On to allow its calls to be recorded.

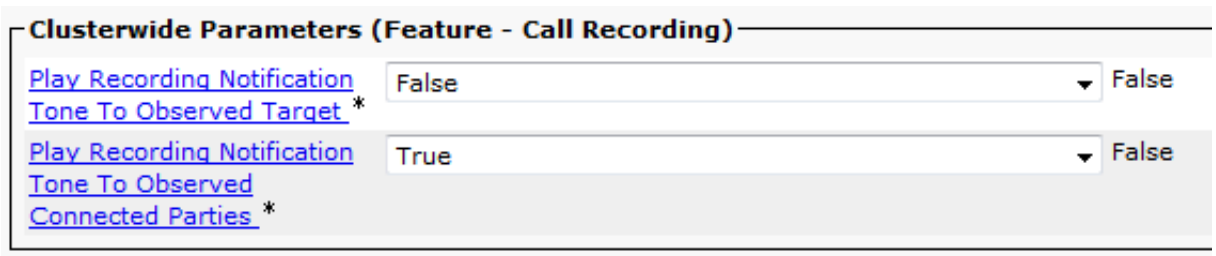



Alternatively, you can set the Built-in Bridge Enable service parameter to On and leave the Built In Bridge in the Phone Configuration window set to Default. Use the Device > Phone menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.

## Configure tones for Recording

Set the service parameters for playing a notification tone to True to allow playing it either to agent only, to customer only, or to both.

Go to System > Services (Cisco Callmanager) and set the parameters to your needs.



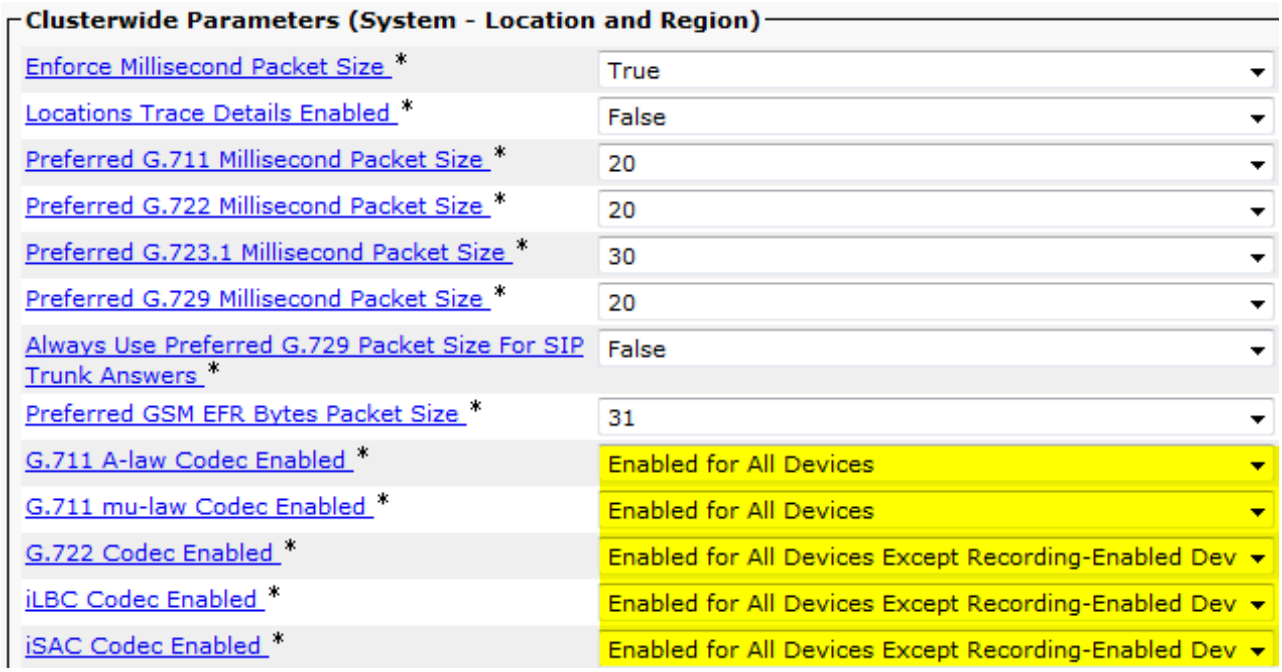

## Configure codec for recording

Set the service parameters for the used codecs.

Recording is only supporting the G.711 codec.

Set G.711 A-law and µ-law to Enabled for All Devices.

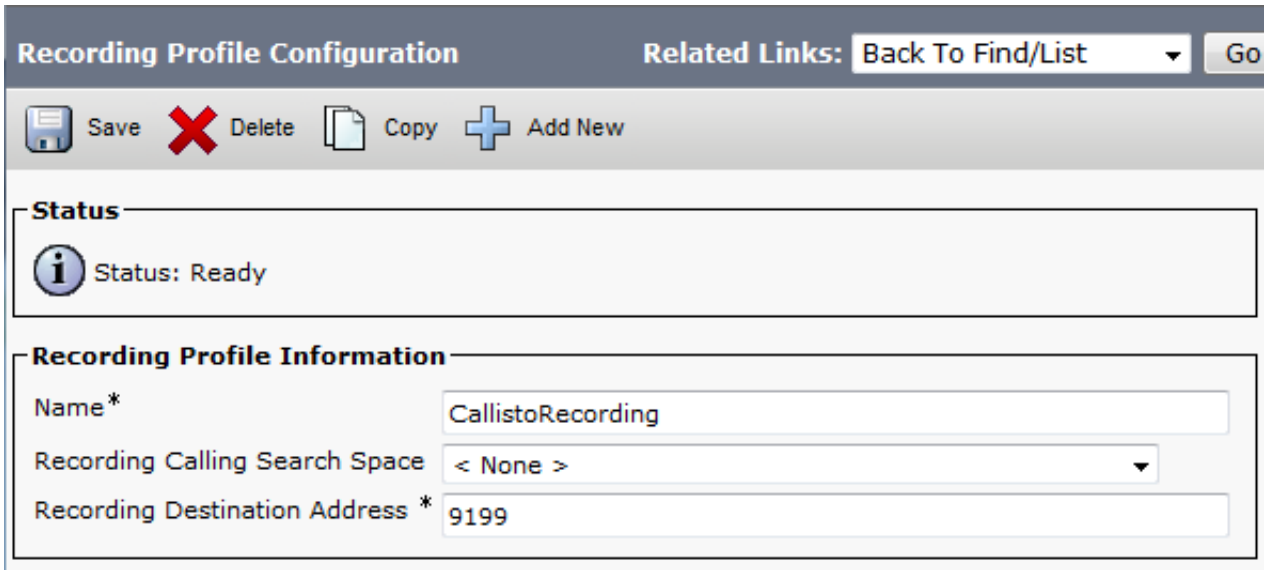
All other codecs must be set to Enabled for All Devices Except Recording-Enabled Dev.




## Create recording profile

1. Go to Device > Device Settings > Recording Profile menu in Cisco Unified Communications Manager Administration to conduct the necessary configuration.
2. Enter the recording profile name, recording calling search space, and recording destination address.
3. Add a route pattern to Callisto for this recording destination number.

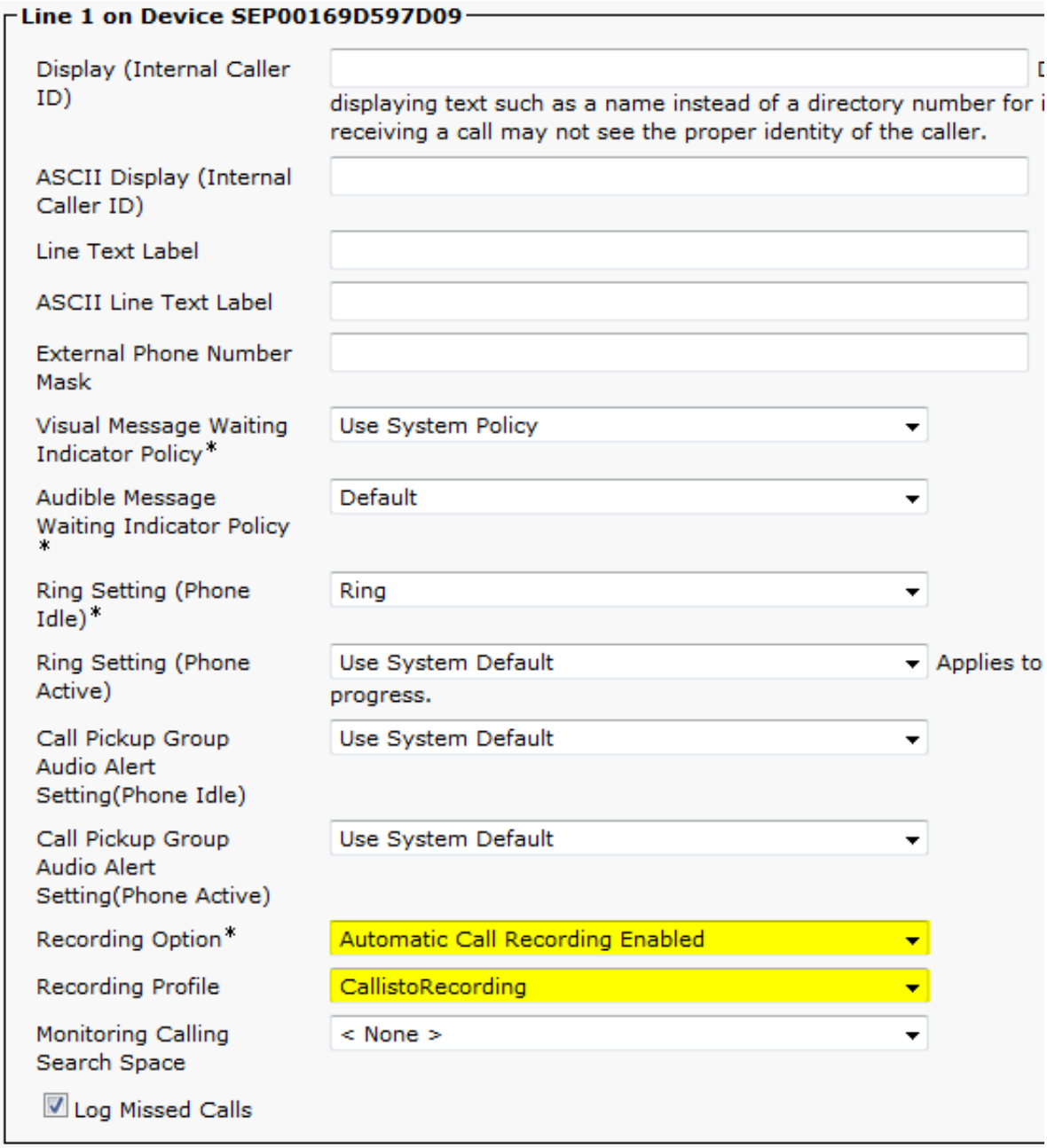
This destination number must be configured in Callisto Recording Option. Refer to the Callisto Administration Manual.




**Enable recording for a line appearance**

To enable recording of an agent, set the Recording Option in the line appearance of the agent.

Go to Call Routing > Directory Number in Cisco Unified Communications Manager Administration to conduct the necessary configuration.




To enable the automatic and manual recording during and at the end of a call, set the *Recording Option* parameter to Automatic Call Recording Enabled.

This setting can cause massive traffic to Callisto and might be very demanding on computing capacity. Also, each recording will use a line license.

To save a record during or at the end of the call, Callisto Services are used for this. Refer to the Callisto Administration Manual.

Set the recording Profile to the profile you created before.

**Only for Cisco Unified Communications Manager v9.x and higher**

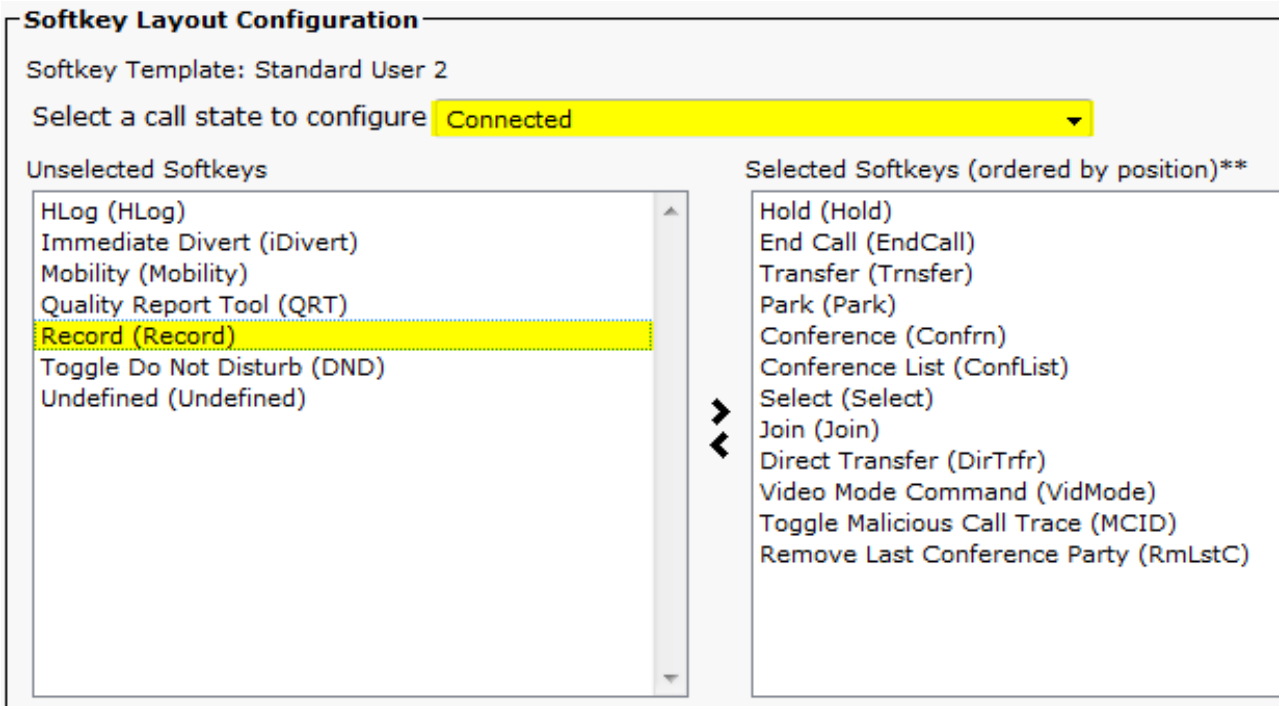
If you want to start a record manually only, set the parameter to Selective Call Recording Enabled.

With this setting, the recording is starting at the actual point of the call. This is invoked by a softkey configuration on the Cisco Unified Communications Manager as described in the next chapter.

**Add the record softkey or programmable line key to the device template**

This configuration is optional and only available with Cisco UCM v9.x and higher.

To allow a user to start and stop recording from a Cisco IP device, add a record softkey or programmable line key to the device template. This function can only be used if the phone line Recording Option in the chapter above is set to Selective Call Recording.




To add a softkey, go to Device > Device Settings > Softkey Template in Cisco Unified Communications Manager Administration to create or modify a non-standard softkey template. Configure the softkey layout for the call state connected to have the Record softkey in the selected softkeys list.

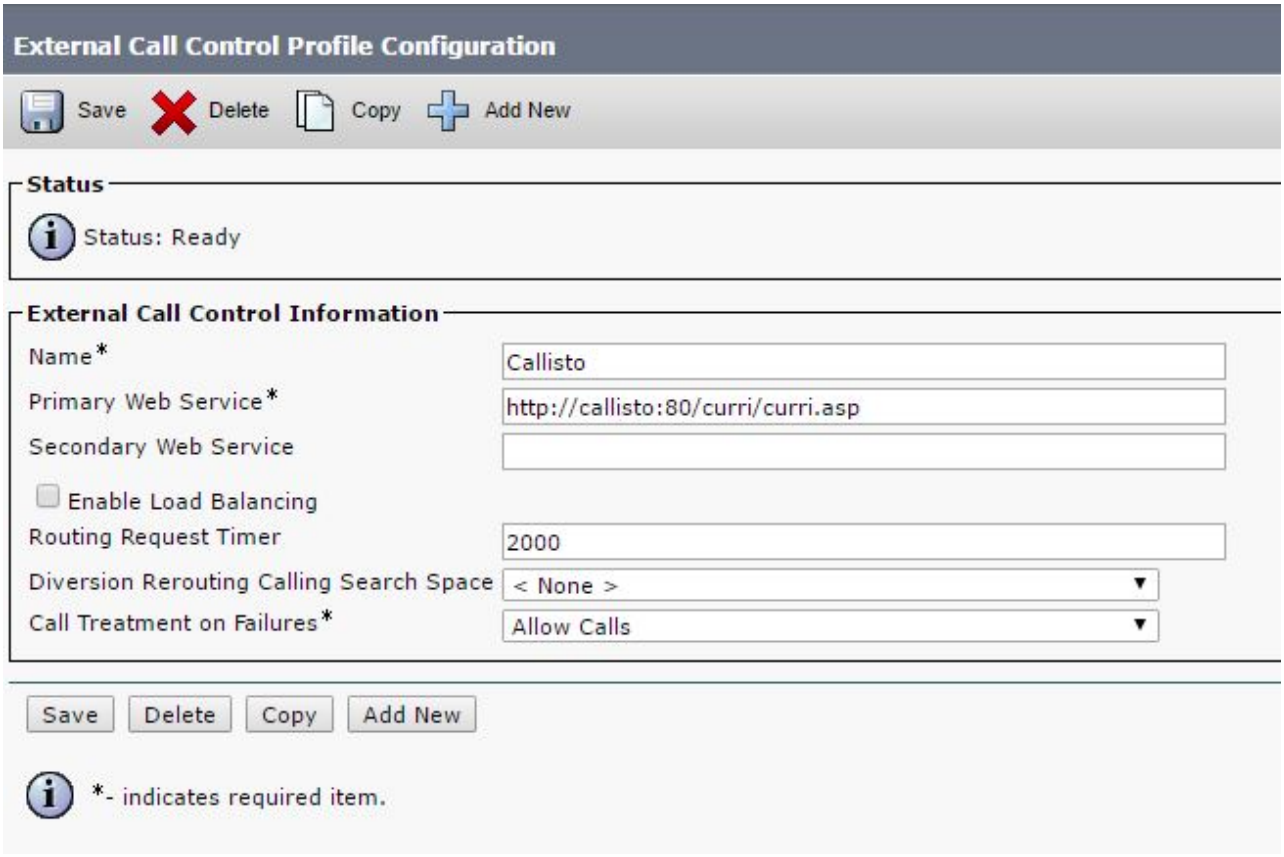
To add the Record programmable line key, go to Device > Device Settings > Phone Button Template in the Cisco Unified Communications Manager administration. Enter the button Template Name, Feature, and Label.

# Configure External Call Control

To use External Call Control in Callisto, External Call Control must first be configured in CUCM.

## Configure External Call Control Profile

Go to Call Routing > External Call Control Profile and add a new External Call Control Profile.




In the field Primary Web Service put the URL of the Callisto CURRI web service: http://*callisto*:80/curri/curri.asp .

Replace *callisto* with the respective IP address or domain name.

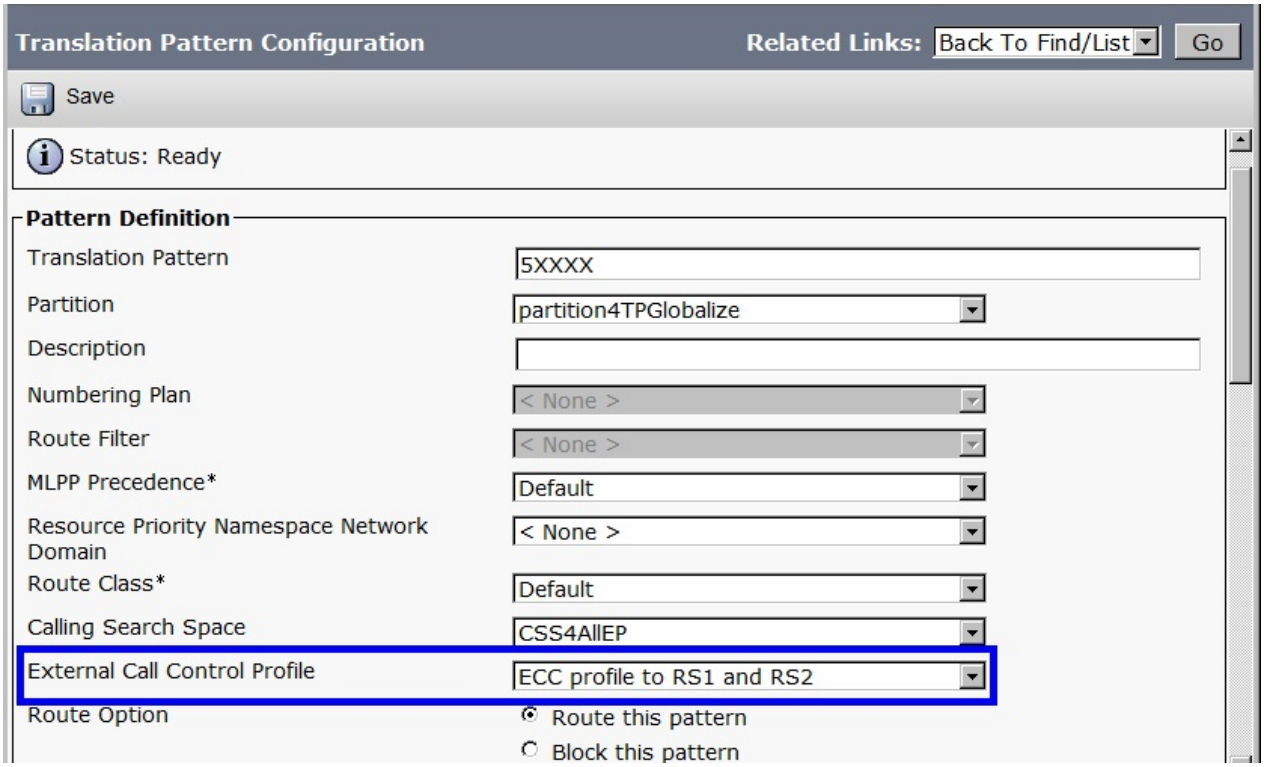
After setting up the External Call Control Profile, the trigger point needs to be set. At the trigger point, the UCM's routing logic decides which route request is chosen.

In UCM versions 8.x and 9.x, the trigger point can only be set to the *translation pattern*. In version 10.0x and higher, two new trigger points are added: *Route pattern* and *directory number*.

## Translation pattern as trigger point

Go to Call Routing > Translation Pattern and add a new translation pattern or configure an existing one.

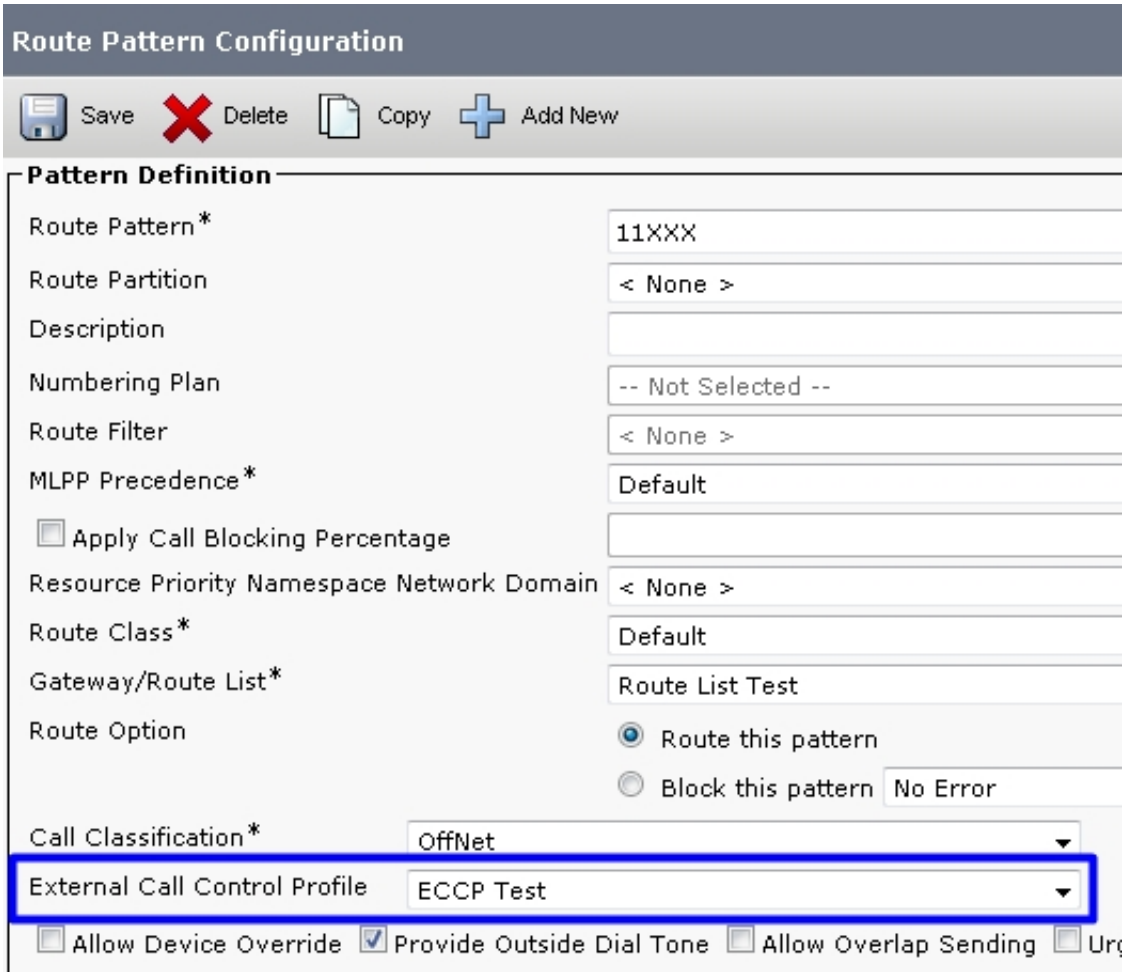
In the External Call Control Profile field set the External Call Control Profile that is configured as described above.




## Route pattern as trigger point

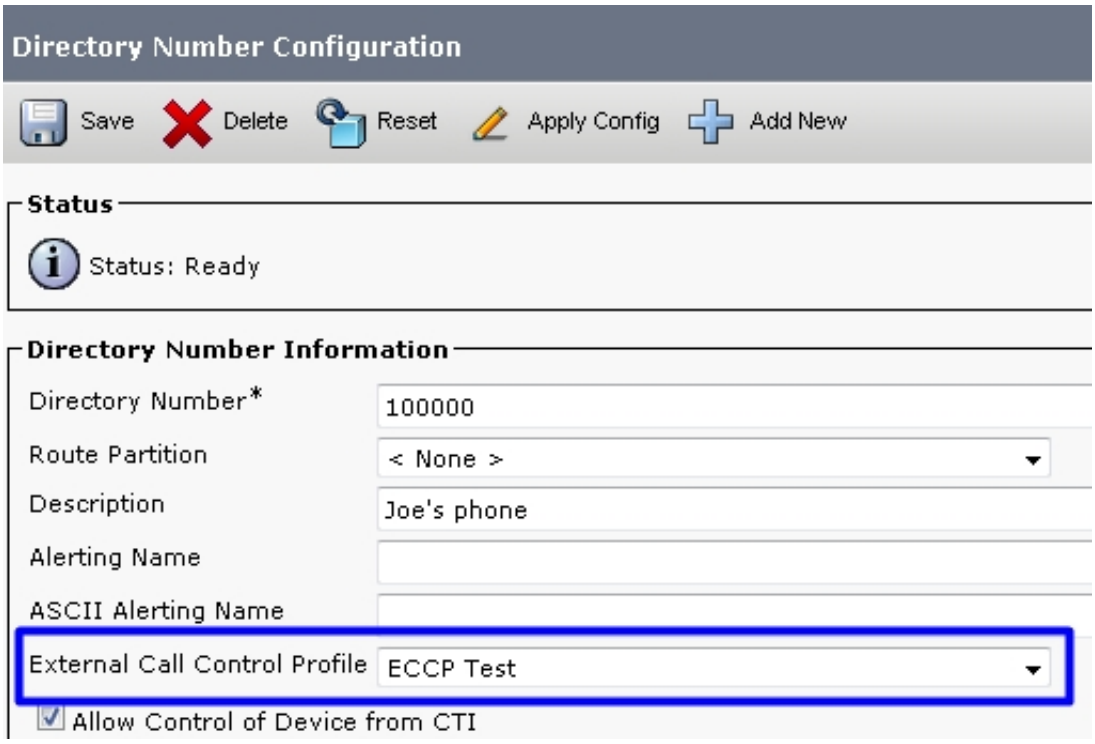
Go to Call Routing > Route/Hunt > Route Pattern and add a new route pattern or configure an existing one.

Set External Call Control Profile to the one that you created, as described above.




## Directory number as trigger point

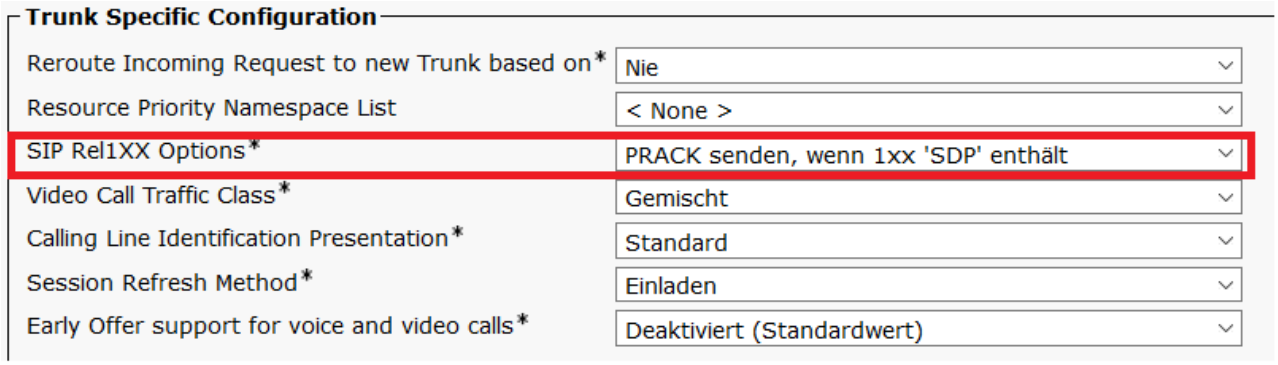
To use a directory number as a trigger point, go to Call Routing > Directory Number and set External Call Control Profile field to the one that you created, as described above.



It’s enough to set External Call Control Profile to the called number for External Call Control to work.

## Announcement over External Call Control

To be able to play an announcement over the External Call Control, before a call gets put trough, you have to modify the SIP Trunk profile from the SIP Trunk, which is connected to the PSTN.
Go to your Cisco Administration web interface, then Device > Device Settings… > SIP Profile and choose the profile which is used by the Trunk connected to the PSTN. Head down to the paragraph Trunk Specific Configuration and set the parameter SIP Rel1XX Options to Send PRACK, if 1xx contains ‘SDP’.



# Enable Transport Layer Security (TLS)

Enable TLS between Callisto UCM and Cisco UCM to encrypt their connection and achieve a high security level.

## Download certificate from Callisto

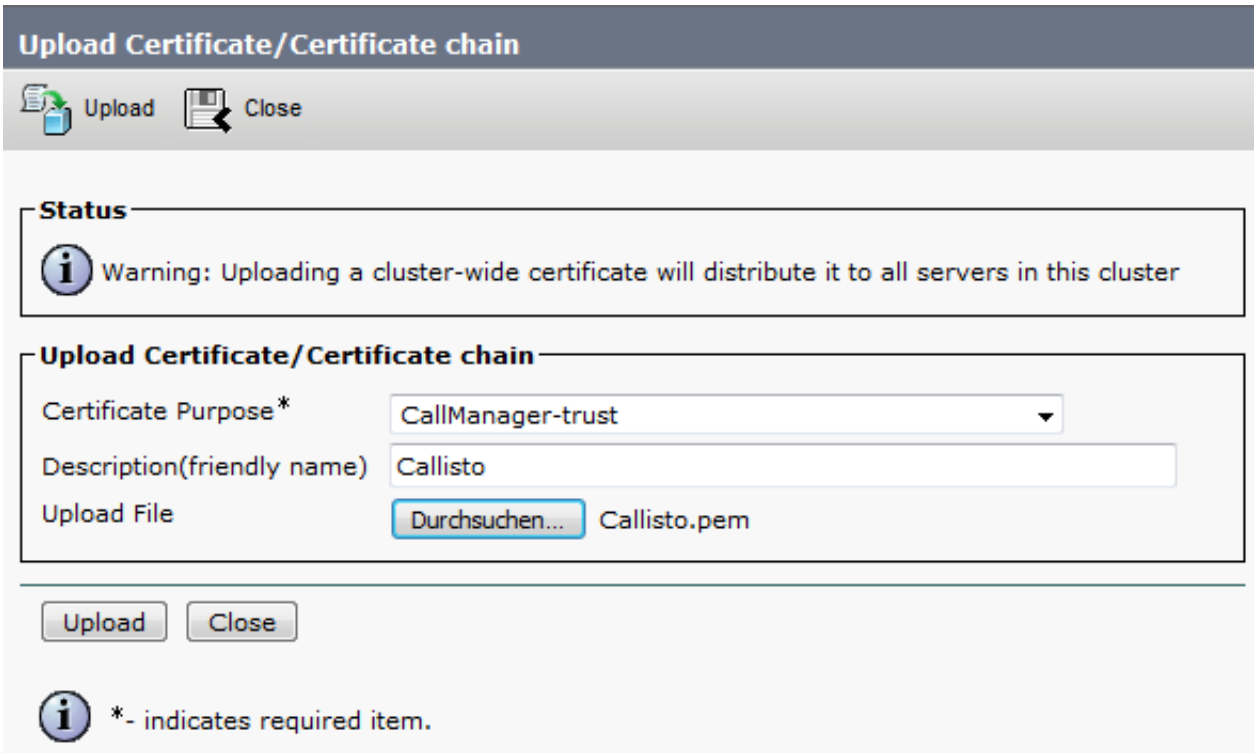
In the Callisto web interface, go to System > System Parameters > SSL Configuration (Top right corner) > Downlad PEM.
If there are no certificates yet, create a new self signed certificate… and download it.

Head back to System Parameters and make sure that the box under Miscellaneous > Secure SIP (TLS, SRTP) is checked.

## Upload certificate to Cisco UCM

Open the Cisco UCM interface in your web browser and select Cisco Unified OS Administration in the Navigation in the top right corner.

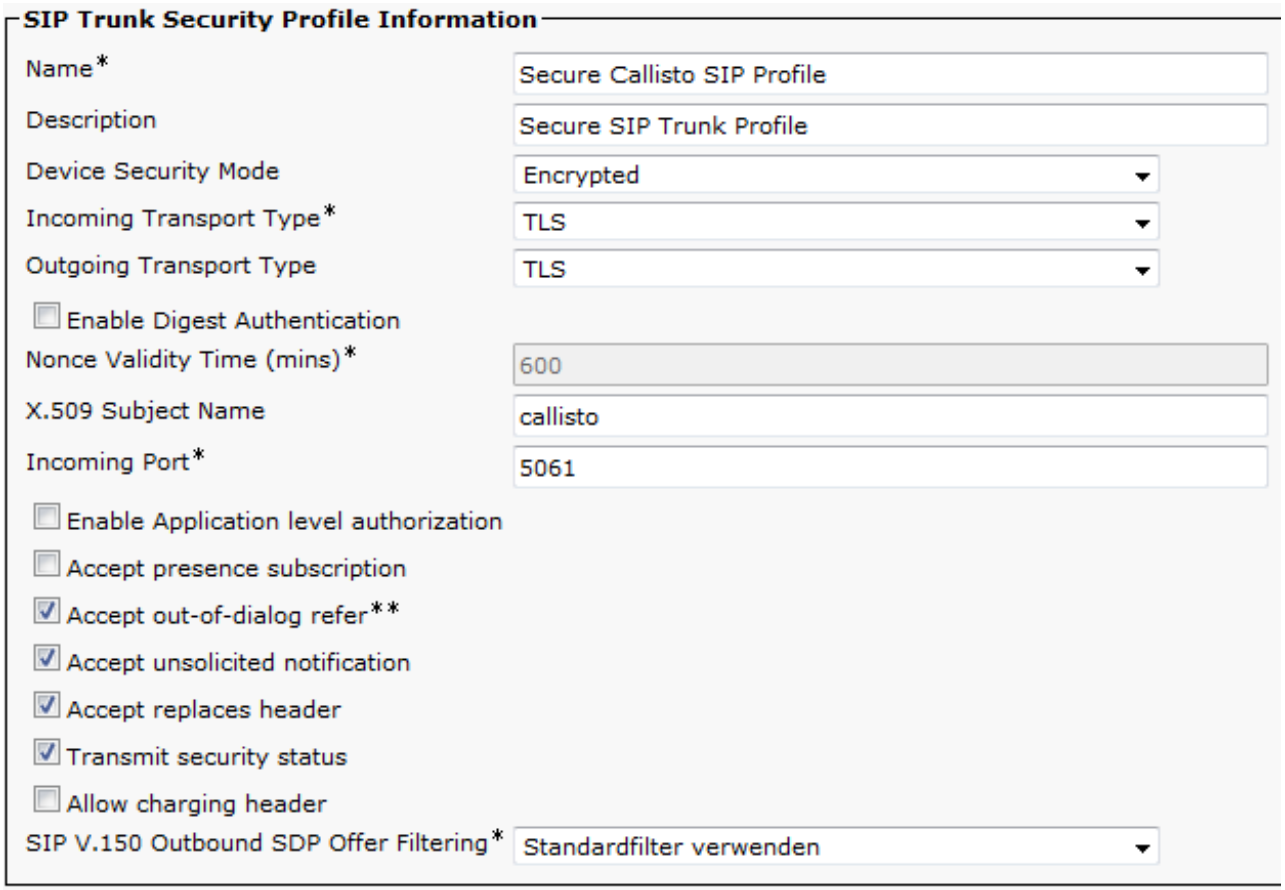
Go to Security > Certificate Management > Upload Certificate and upload the certificate that you previously downloaded from Callisto.
Choose CallManager-trust as the certificate purpose and provide a user friendly description like Callisto.




## Device security profile

Select now the Cisco Unified CM Administration in the Navigation in the top right corner and head to System > Security > SIP Trunk Security Profle and then Add New.

Fill in the profile information as following:




(Name and description are examples)

If you are not already using secure connections it might be that the LBM Security Mode is set to insecure. To change this, head to System > Enterprise Parameters > Security Parameters > LBM Security Mode and change it to at least Mixed.

## Create trunk

Create a new Trunk under Device > Trunk as described above under SIP tunk configuration.

Use the previously created Device Security Profile, set the Destinationport to 5061 and use the Callisto IP as Destination Address.

Now the Trunk is ready to be used as a secure connection, to do so define a new route pattern and use the created trunk as the Gateway / Route List to route the desired calls trough this trunk.

# Copyright

**Copyright © 2026 CTModule AG; All Rights Reserved**

This document contains proprietary information of CTModule AG. No part of the work described herein may be reproduced. Reverse engineering of the hardware or software is prohibited and is protected by patent law.

This material or any portion of it may not be copied in any form or by any means, stored in a retrieval system, adopted or transmitted in any form or by any means (electronic, mechanical, photographic, graphic, optic or otherwise), or translated to any language or computer language without the prior written permission of CTModule AG.

The information in this document is subject to change without notice. CTModule AG makes no representation or warranties with respect to the contents herein and shall not be responsible for any loss or damage caused to users either by direct or indirect use of this information. This document may contain information about third party products or processes. This third party information is out of the sphere of influence of CTModule AG. Therefore CTModule AG shall not be responsible for the correctness or legitimacy of this information.

While due care has been taken to deliver accurate documentation, CTModule AG does not warrant that this document is error-free. If you find any errors, inconsistencies, omissions or other problems related to this document, please report this in writing by email to box@ctmodule.com at CTModule AG.

CTMaker, VAS, Callisto, Callisto Express, Callisto ISDN, Callisto One, Callisto UCM, Callisto Cruise, Callisto Hospitality, Callisto X Mobile, COC Express , COC UCM and CTModule AG are trademarks and the CTModule logo is a service mark of CTModule AG.

All other products or company names mentioned herein are used for identification purposes only, and may be trademarks or registered trademarks of their respective owners.

The following description of software, hardware or process of CTModule AG or other third party provider may be included with your product and will be subject to the software, hardware or other license agreement.

# Disclaimer

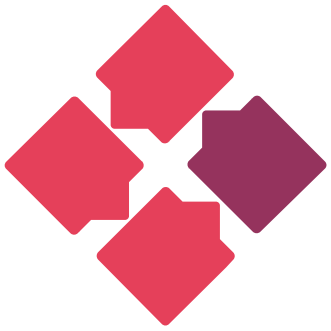
Changes are possible without notice.

The General Terms and Conditions (GTC) of CTModule AG, Switzerland apply.

All rights reserved.

**For more information about CTModule AG, visit the CTModule web site at**

www.ctmodule.com



CTMODULE

COMMUNICATION TECHNOLOGY MODULES


**CTMODULE AG**

Lehnweg 1

CH-3123 Belp/Berne

Switzerland

T: +41 (0)31 531 11 11

F: +41 (0)31 531 11 12

sales@ctmodule.com

**OFFICE GERMANY**

Frankfurter Straße 92

D-65760 Eschborn/Frankfurt

Germany

T: *+49 6196 2049173-0*

F: *+49 6196 2049173-9*

sales-d@ctmodule.com

**OFFICE SERBIA**

Gospodara Vučića 145

RS-11000 Belgrade

Serbia

T: +381 18 308076

sales@ctmodule.com