



CALLN HOSTED CALL RECORDING CISCO CUCM SETUP GUIDE

Created by Paul Johansen

19 June 2020

Version 1.3.0

Table of Contents

1. Introduction	3
2. Connectivity.....	3
3. Configuration of CUCM	4
3.1. Create a SIP profile	4
3.2. Create a SIP Trunk Security Profile	5
3.3. Create a SIP trunk that points to the recording server(s).....	6
3.4. Create a recording profile	8
3.5. Create a Route Pattern.....	9
3.6. Configure Tones for Recording.....	10
3.7. Configure Codecs.....	11
4. Configure each phone for recording	13
4.1. Turn on IP Phone Built-In-Bridge to Allow Monitoring and Recording	13
4.1.1. System Wide	13
4.1.2. Per Handset.....	13
4.2. Enable Recording for a Line Appearance	14
4.3. Add a Recording Softkey or Programmable Line Key to the Device Template (Optional).....	15
4.4. Compatible handsets.....	16
5. Configuration of CallN	19
5.1. Configure call recording client software	19
5.1.1. UDP Client	19
5.1.2. TCP Client	19
5.1.3. Public IP Address.....	19
5.2. Configure machine firewall	20
5.2.1. Incoming rules	20
5.2.1. Outgoing rules.....	20
6. TAPI Setup	21
7. Create a TAPI user	21
8. Install TAPI driver	25
8.1. Download the driver	25
8.2. Install the driver	26
8.3. Configure the driver.....	26
8.4. Optional – Enable trace logging	30

1. Introduction

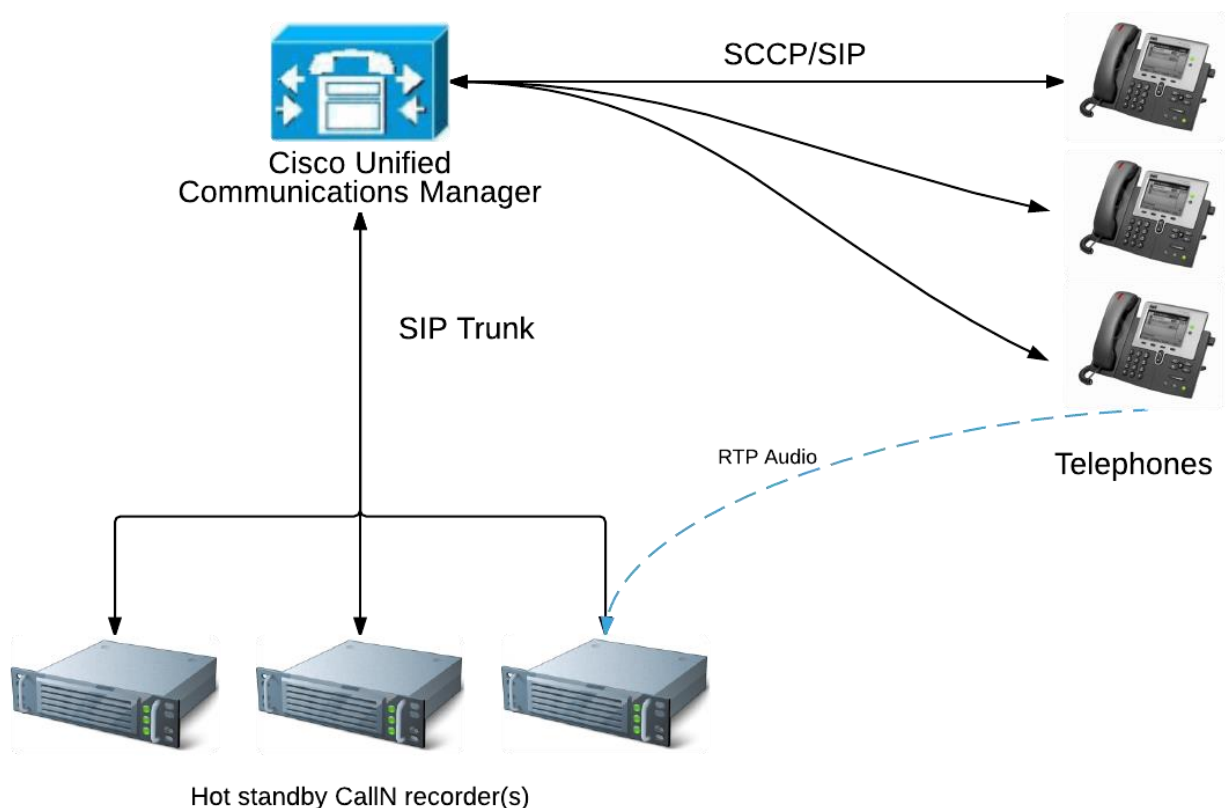
This document describes how to configure your Cisco CUCM platform as well as the CallN recording client software for the recording of telephone calls.

Note: The steps in this document are **ONLY** necessary when deploying in an Active recording integration deployment where SPAN capture is not possible nor desired.

Note: Not all CISCO handsets support the Built-in-Bridge. Supported handsets can be found at <https://developer.cisco.com/site/uc-manager-sip/documents/supported/>

2. Connectivity

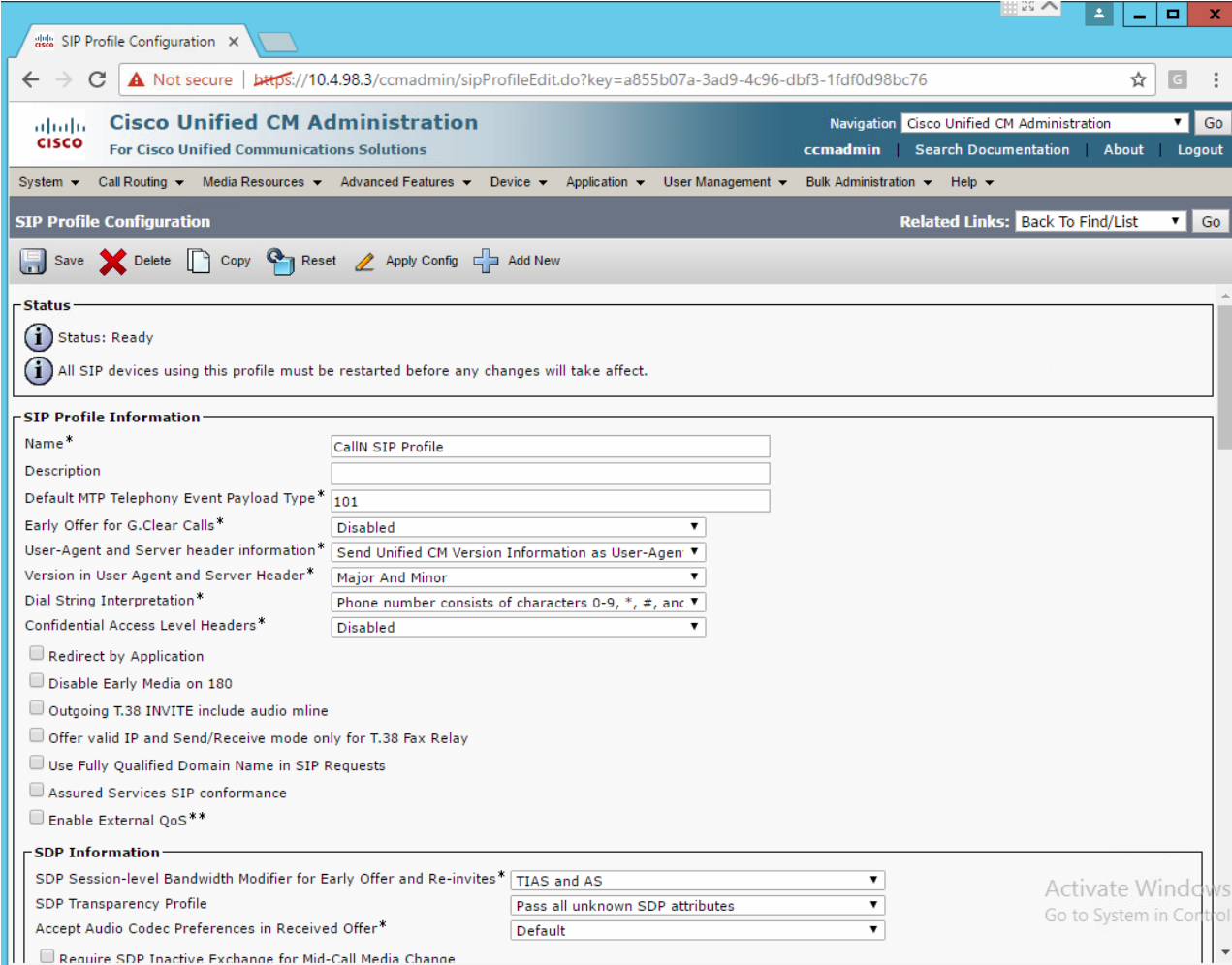
CallN utilises the Built-in-Bridge feature of the Cisco IP handset to record calls (see compatible handset list). When a call is to be recorded the CUCM initiates a SIP call to the CallN platform via a SIP trunk interface and then informs the handset to send the CallN server the RTP audio directly.



3. Configuration of CUCM

3.1. Create a SIP profile

Use the Device > Device Settings > SIP Profile menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.



SIP Profile Configuration

Navigation: Cisco Unified CM Administration | Go

ccmadmin | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

SIP Profile Configuration | Related Links: Back To Find/List | Go

Save | Delete | 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*: CallN SIP Profile

Description:

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-Agent

Version in User Agent and Server Header*: Major And Minor

Dial String Interpretation*: Phone number consists of characters 0-9, *, #, and

Confidential Access Level Headers*: Disabled

☐ Redirect by Application

☐ Disable Early Media on 180

☐ Outgoing T.38 INVITE include audio mline

☐ Offer valid IP and Send/Receive mode only for T.38 Fax Relay

☐ Use Fully Qualified Domain Name in SIP Requests

☐ Assured Services SIP conformance

☐ Enable External QoS**

SDP Information

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*: TIAS and AS

SDP Transparency Profile: Pass all unknown SDP attributes

Accept Audio Codec Preferences in Received Offer*: Default

☐ Require SDP Inactive Exchange for Mid-Call Media Change

SIP Profile Information / Name – Enter a name for this profile. Something like 'CallN SIP Profile'.

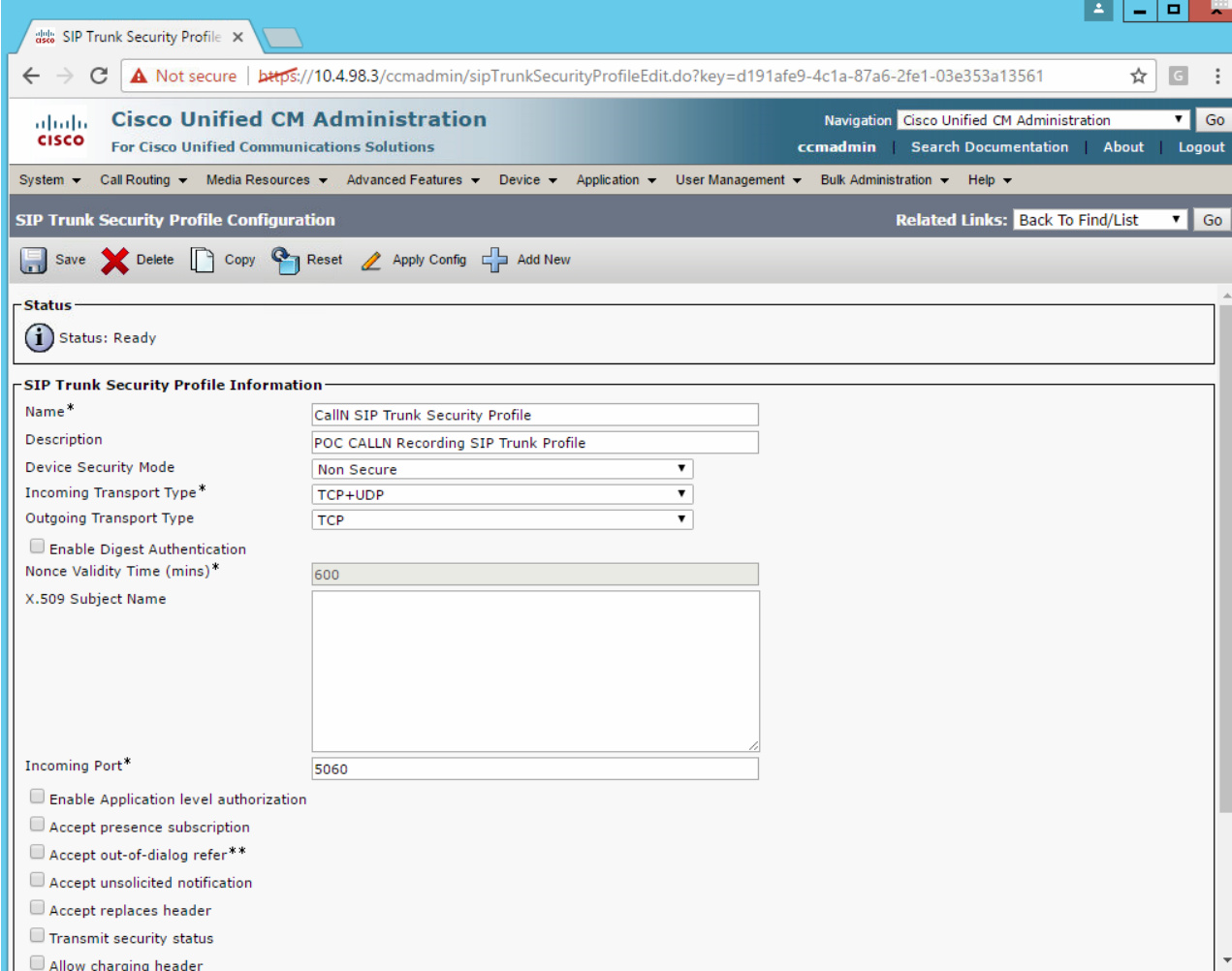
Trunk Specific Configuration / Deliver Conference Bridge Identifier – Enable this setting.

SIP Options Ping / Enable OPTIONS Ping to monitor destination status for Trunks – Enable this setting.

Click Save.

3.2. Create a SIP Trunk Security Profile

Use the System > Security > SIP Trunk Security Profile menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.



SIP Trunk Security Profile Configuration

Related Links: [Back To Find/List](#) Go

Save Delete Copy Reset Apply Config Add New

Status
 Status: Ready

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 Trunk Security Profile Information / Name – Enter a name for this security profile. Something like 'CallIN SIP Trunk Security Profile'.

SIP Trunk Security Profile Information / Incoming Transport Type – Set as 'TCP+UDP'.

SIP Trunk Security Profile Information / Outgoing Transport Type – Set as 'TCP'.

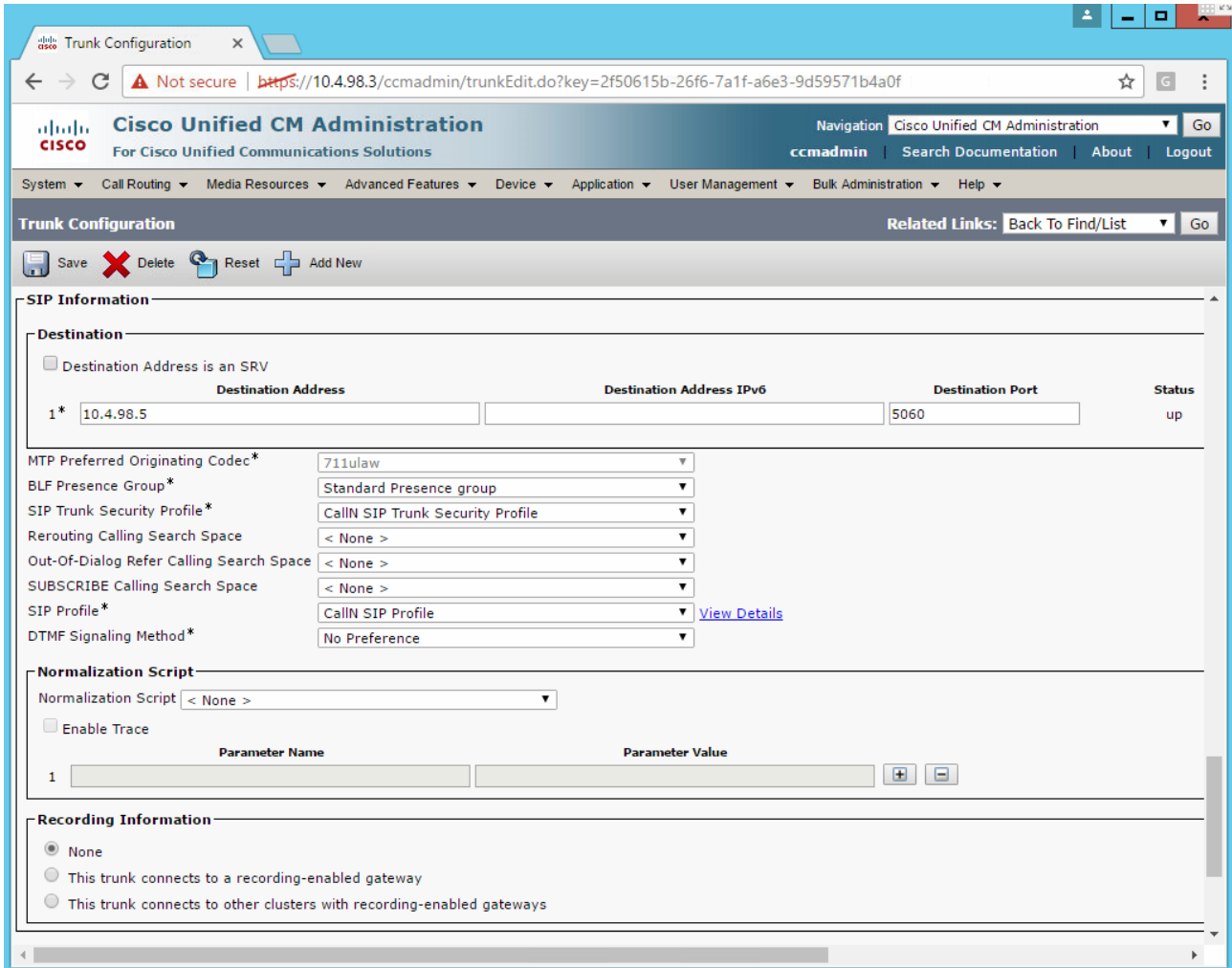
SIP Trunk Security Profile Information / Enable Digest Authentication – Uncheck.

SIP Trunk Security Profile Information / Device Security Mode – Set to 'Not Secure'.

Click Save.

3.3. Create a SIP trunk that points to the recording server(s)

Use the Device > Trunk menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration



Trunk Configuration

Navigation: Cisco Unified CM Administration | Go

ccmadmin | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Related Links: Back To Find/List | Go

Save | Delete | Reset | Add New

SIP Information

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status
1*	10.4.98.5		5060	up

MTP Preferred Originating Codec*: 711ulaw

BLF Presence Group*: Standard Presence group

SIP Trunk Security Profile*: CallIN SIP Trunk Security Profile

Rerouting Calling Search Space: < None >

Out-Of-Dialog Refer Calling Search Space: < None >

SUBSCRIBE Calling Search Space: < None >

SIP Profile*: CallIN SIP Profile [View Details](#)

DTMF Signaling Method*: No Preference

Normalization Script

Normalization Script: < None >

☐ Enable Trace

	Parameter Name	Parameter Value
1		

Recording Information

☒ None

☐ This trunk connects to a recording-enabled gateway

☐ This trunk connects to other clusters with recording-enabled gateways

Device Name – Enter a name for this trunk. Something like 'CallIN_SIP_Trunk'.

Select the Device Pool to the pool containing the phones to record.

Select the Inbound Calls / Calling Search Space to the CSS containing the phones to record.

SIP Information / SIP Trunk Security Profile - Select the CallIN SIP Trunk Security profile that you configured earlier, probably 'CallIN SIP Trunk Security Profile'.

SIP Information / SIP Profile - Select the CallIN SIP profile that you configured earlier, probably 'CallIN SIP Profile'.

SIP Information / Destination / Destination Address – set as the IP address or DNS name of the CallN recording server.

SIP Information / Destination / Destination Port – set as 5060. This should match the configuration in the CallN recording client.

Call Routing Information / SIP Privacy – Set to 'None'.

Click Save.

3.4. Create a recording profile

Use the Device > Device Settings > Recording Profile menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.

Recording Profile Information / Name – Enter a name for this recording profile. Something like 'CallN Recording Profile'.

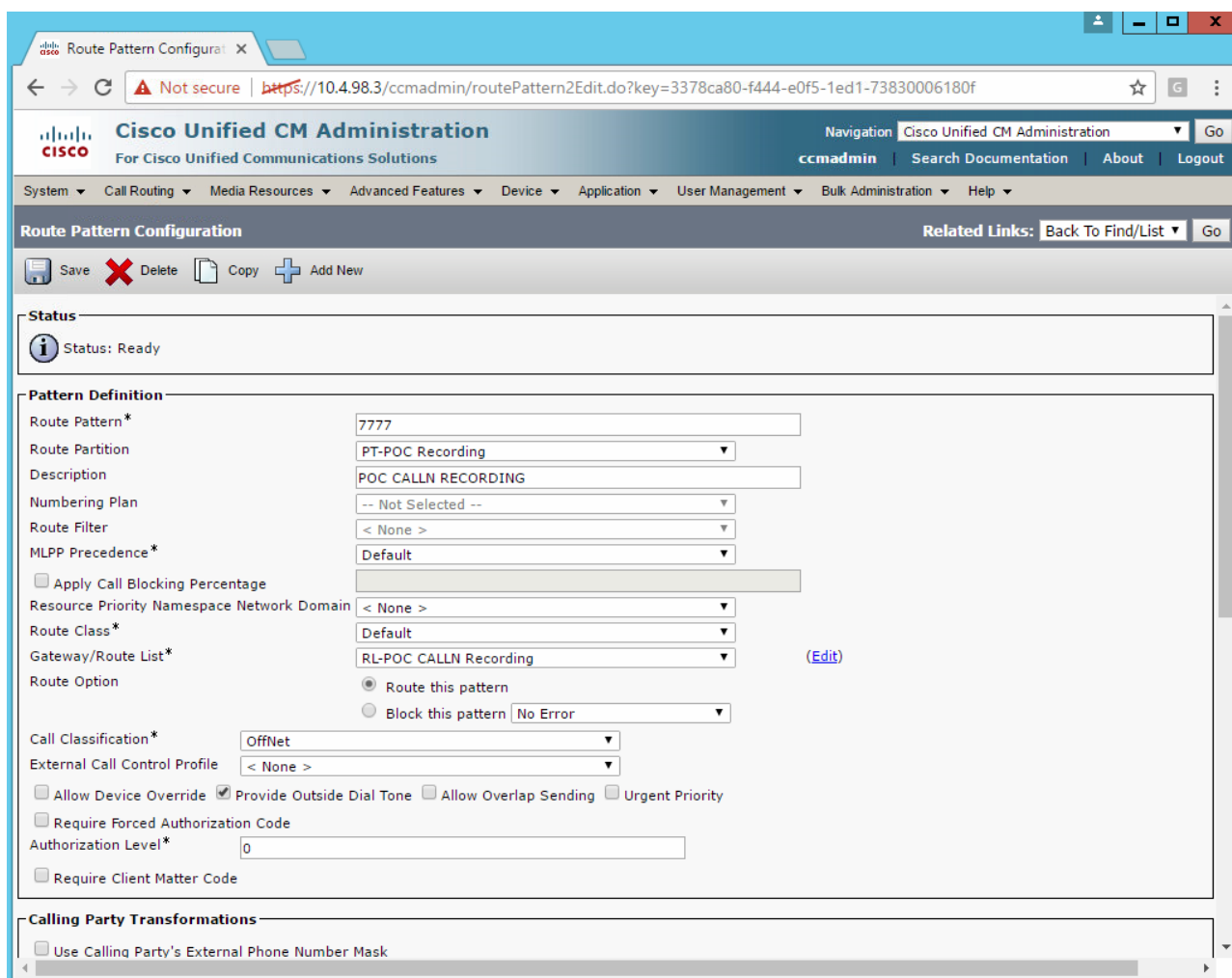
Recording Profile Information / Recording Call Search Space – Set to the CSS containing the phones to record.

Recording Profile Information / Recording Destination Address – Set as a directory number that is associated with the recorder. This number should not clash with other number plan entries. A good example is 7777.

Click Save.

3.5. Create a Route Pattern

Use the Call Routing > Route/Hunt > Route Pattern menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.



The screenshot displays the 'Route Pattern Configuration' page in the Cisco Unified CM Administration interface. The page is titled 'Route Pattern Configuration' and includes a 'Status' section indicating 'Status: Ready'. The 'Pattern Definition' section contains the following fields and values:

- Route Pattern*: 7777
- Route Partition: PT-POC Recording
- Description: POC CALLN RECORDING
- Numbering Plan: -- Not Selected --
- Route Filter: < None >
- MLPP Precedence*: Default
- Apply Call Blocking Percentage: ☐
- Resource Priority Namespace Network Domain: < None >
- Route Class*: Default
- Gateway/Route List*: RL-POC CALLN Recording (with an 'Edit' link)
- Route Option:
 - ☒ Route this pattern
 - ☐ Block this pattern No Error
- Call Classification*: OffNet
- External Call Control Profile: < None >
- Allow Device Override: ☐
- Provide Outside Dial Tone: ☒
- Allow Overlap Sending: ☐
- Urgent Priority: ☐
- Require Forced Authorization Code: ☐
- Authorization Level*: 0
- Require Client Matter Code: ☐

The 'Calling Party Transformations' section at the bottom includes the option 'Use Calling Party's External Phone Number Mask', which is currently unchecked.

Route Pattern – Set this with the same value that was configured earlier for the CallN recording profile. In the example, it was 7777.

Route Partition – Set this to the partition that contains the phones to record.

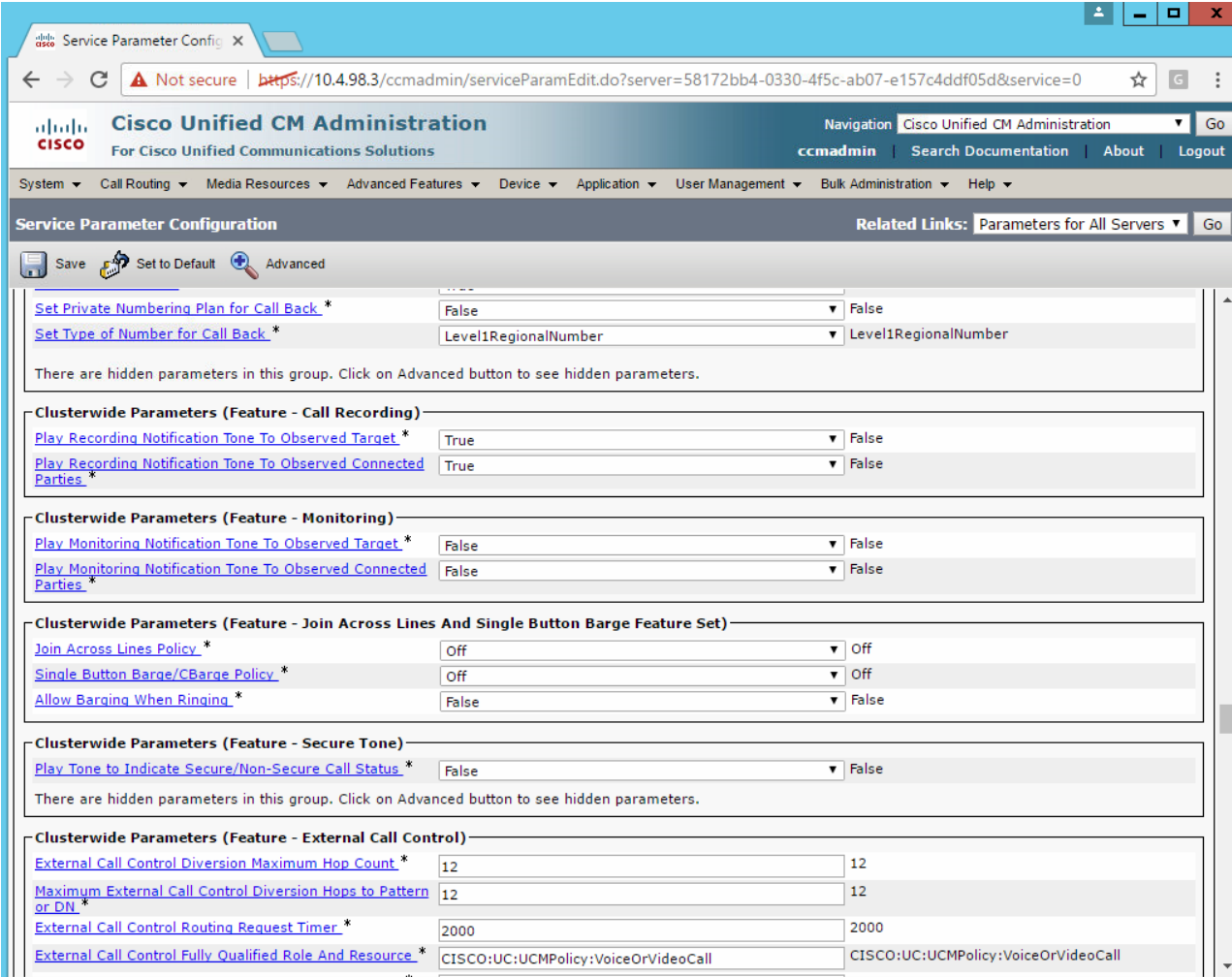
Gateway / Route List - Select the CallN SIP trunk that was created earlier.

Click Save.

3.6. Configure Tones for Recording

Use the System > Service Parameters menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.

Select your appropriate Server from the drop-down list and then select 'Cisco CallManager'.



Service Parameter Configuration

Navigation: Cisco Unified CM Administration

System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help

Related Links: Parameters for All Servers

Save Set to Default Advanced

Set Private Numbering Plan for Call Back * False False

Set Type of Number for Call Back * Level1RegionalNumber Level1RegionalNumber

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Clusterwide Parameters (Feature - Call Recording)

Play Recording Notification Tone To Observed Target * True False

Play Recording Notification Tone To Observed Connected Parties * True False

Clusterwide Parameters (Feature - Monitoring)

Play Monitoring Notification Tone To Observed Target * False False

Play Monitoring Notification Tone To Observed Connected Parties * False False

Clusterwide Parameters (Feature - Join Across Lines And Single Button Barge Feature Set)

Join Across Lines Policy * Off Off

Single Button Barge/CBarge Policy * Off Off

Allow Barging When Ringing * False False

Clusterwide Parameters (Feature - Secure Tone)

Play Tone to Indicate Secure/Non-Secure Call Status * False False

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Clusterwide Parameters (Feature - External Call Control)

External Call Control Diversion Maximum Hop Count * 12 12

Maximum External Call Control Diversion Hops to Pattern or DN * 12 12

External Call Control Routing Request Timer * 2000 2000

External Call Control Fully Qualified Role And Resource * CISCO:UC:UCMPolicy:VoiceOrVideoCall CISCO:UC:UCMPolicy:VoiceOrVideoCall

You can use the service parameters for playing tone to True to allow tone to be played either to agent only, to customer only, or to both.

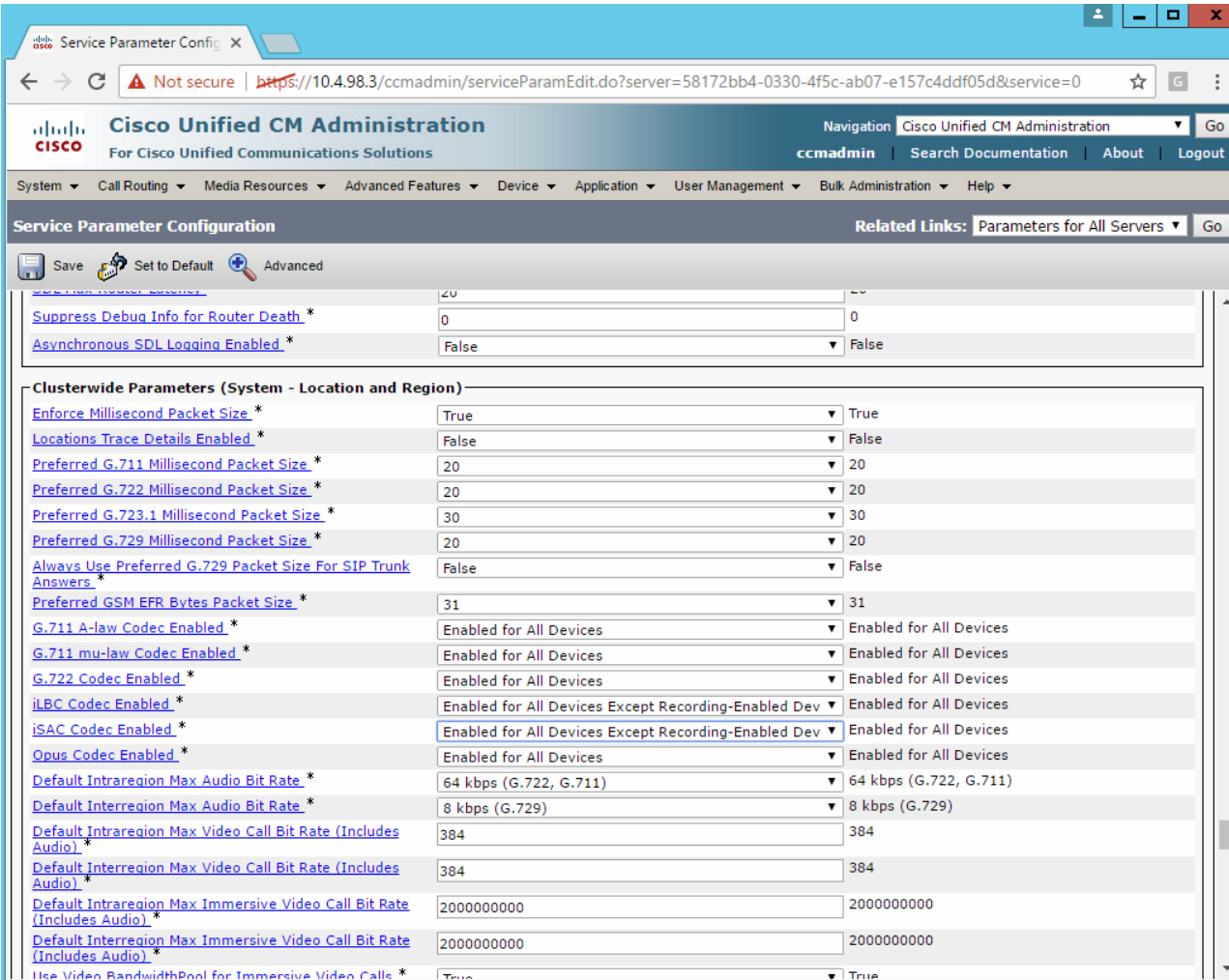
Click Save.

3.7. Configure Codecs

There are a few codecs that must be disabled when recording because either CallN doesn't support them, or the Cisco handset built-in-bridge doesn't support them.

Use the System > Service Parameters menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.

Select your appropriate Server from the drop-down list and then select 'Cisco CallManager'.



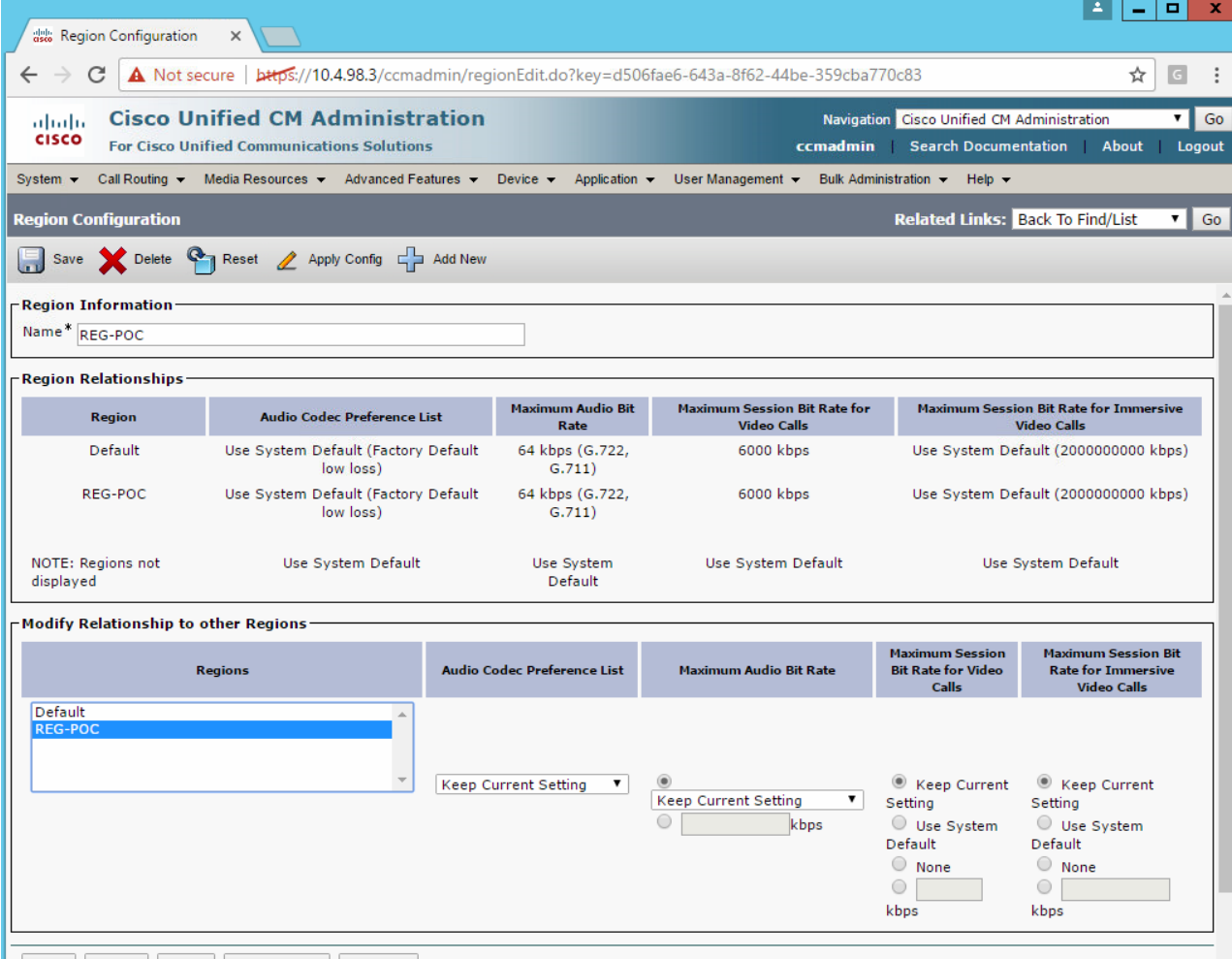
The screenshot shows the Cisco Unified CM Administration interface. The 'Service Parameter Configuration' page is displayed for the 'Cisco CallManager' server. The 'Clusterwide Parameters (System - Location and Region)' section is expanded, showing a list of parameters and their values. The 'iLBC Codec Enabled' parameter is highlighted, showing its value as 'Enabled for All Devices Except Recording-Enabled Dev'. Other parameters like 'G.711 A-law Codec Enabled', 'G.711 mu-law Codec Enabled', 'G.722 Codec Enabled', 'iSAC Codec Enabled', and 'Opus Codec Enabled' are also shown with their respective values.

Clusterwide Parameters (System – Location and Region) / iLBC Codec Enabled – Set to 'Enabled for All Devices Except Recording-Enabled Devices'

Clusterwide Parameters / iSAC Codec Enabled – Set to 'Enabled for All Devices Except Recording-Enabled Devices'

Clusterwide Parameters / Default Intra-region Max Audio Bit Rate – Set to '64 kbps (G.722, G.711)'

Use the System > Region Information > Region menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.



The screenshot shows the Cisco Unified CM Administration interface for Region Configuration. The browser address bar shows a URL starting with https://10.4.98.3/ccmadmin/regionEdit.do. The page title is "Cisco Unified CM Administration" and the navigation bar includes "ccmadmin", "Search Documentation", "About", and "Logout". The main menu includes "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The "Region Configuration" section is active, showing a "Related Links" dropdown with "Back To Find/List". Below this are buttons for "Save", "Delete", "Reset", "Apply Config", and "Add New".

Region Information

Name * REG-POC

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Default	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	6000 kbps	Use System Default (2000000000 kbps)
REG-POC	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	6000 kbps	Use System Default (2000000000 kbps)
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

Modify Relationship to other Regions

Regions	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
<div> <div>Default</div> <div>REG-POC</div> </div>	<div>Keep Current Setting</div>	<div> <input checked="" type="radio"/> Keep Current Setting <input type="radio"/> <input type="text"/> kbps </div>	<div> <input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text"/> kbps </div>	<div> <input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text"/> kbps </div>

Buttons: Save, Delete, Reset, Apply Config, Add New

Max Audio Bit Rate – Set to 'Use System Default' or '64 kbps (G.722, G.711)'.

Click Save.

4. Configure each phone for recording

4.1. Turn on IP Phone Built-In-Bridge to Allow Monitoring and Recording

The built-in bridge feature of the agent phone must be set to On to allow its calls to be recorded

This feature can be enabled as a system wide setting or on a more granular per-handset setting.

4.1.1. System Wide

Use the System > Service Parameters menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.

Select your appropriate Server from the drop-down list and then select 'Cisco CallManager'.

Clusterwide Parameters (Device – Phone) / Builtin Bridge Enable – Set to On.

4.1.2. Per Handset

Use the Device > Phone menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.

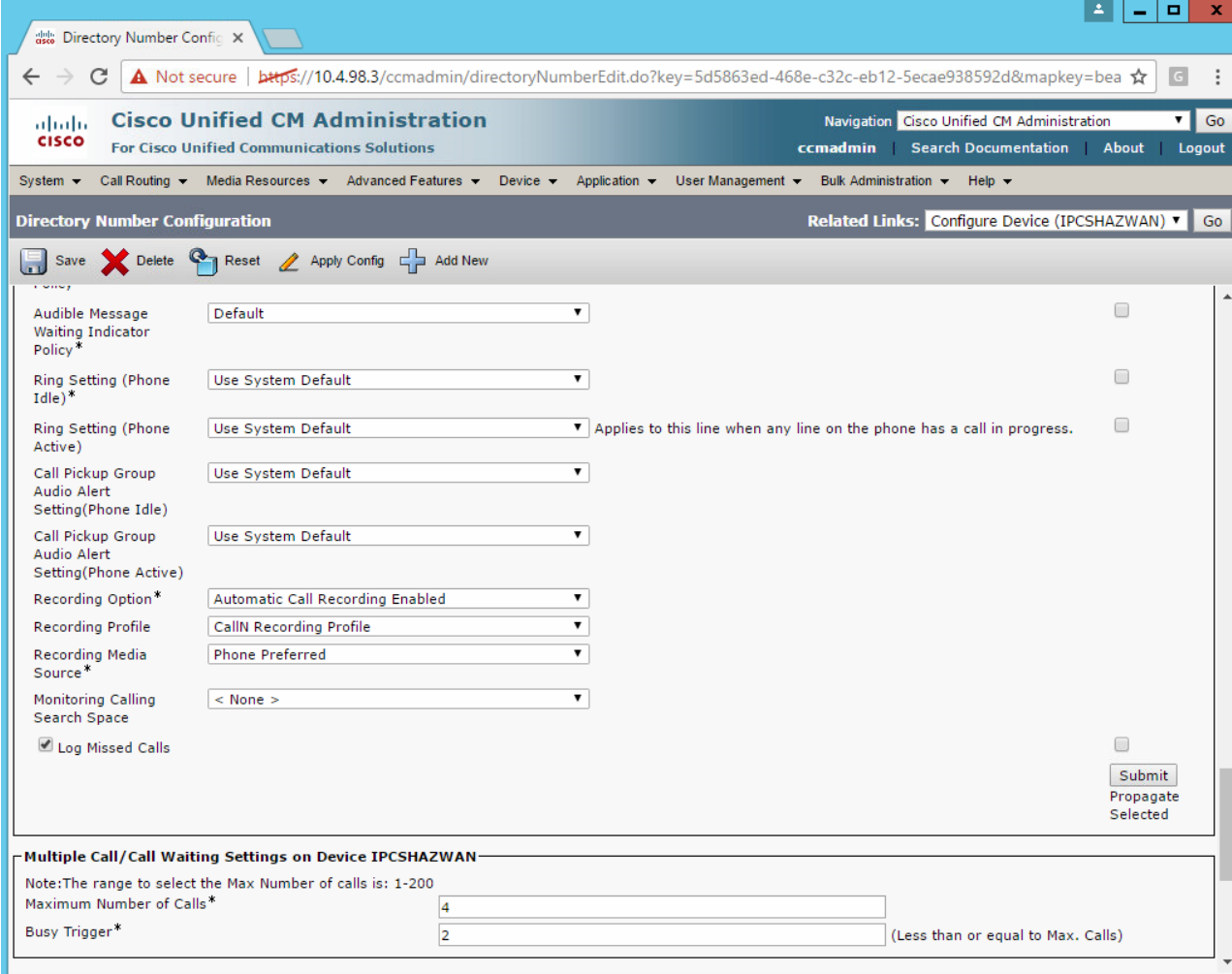
Built In Bridge – Set to On.



4.2. Enable Recording for a Line Appearance

Select a pre-created recording profile from the drop-down list box. (Use Device > Device Settings > Recording Profile to configure a recording profile.)

Use the Call Routing > Directory Number menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.



The screenshot shows the Cisco Unified CM Administration interface for configuring a directory number on device IPCSHAZWAN. The 'Recording Option' is set to 'Automatic Call Recording Enabled', the 'Recording Profile' is 'CallN Recording Profile', and the 'Recording Media Source' is 'Phone Preferred'. The 'Log Missed Calls' checkbox is checked. Below the main configuration area, there is a section for 'Multiple Call/Call Waiting Settings on Device IPCSHAZWAN' with a note about the maximum number of calls (1-200). The 'Maximum Number of Calls' is set to 4 and the 'Busy Trigger' is set to 2 (Less than or equal to Max. Calls).

Configuration Item	Value
Audible Message Waiting Indicator Policy*	Default
Ring Setting (Phone Idle)*	Use System Default
Ring Setting (Phone Active)	Use System Default
Call Pickup Group Audio Alert Setting(Phone Idle)	Use System Default
Call Pickup Group Audio Alert Setting(Phone Active)	Use System Default
Recording Option*	Automatic Call Recording Enabled
Recording Profile	CallN Recording Profile
Recording Media Source*	Phone Preferred
Monitoring Calling Search Space	< None >
Log Missed Calls	<input checked="" type="checkbox"/>

Multiple Call/Call Waiting Settings on Device IPCSHAZWAN

Note: The range to select the Max Number of calls is: 1-200

Maximum Number of Calls*	4
Busy Trigger*	2 (Less than or equal to Max. Calls)

Set the Recording Option to

- Automatic Call Recording Enabled
- Selective Call Recording Enabled

Set Recording Profile to 'CallN Record Profile'.

Set Recording Media Source to 'Phone Preferred'.

Click Save.

4.3. Add a Recording Softkey or Programmable Line Key to the Device Template (Optional)

To enable a user to start and stop recording from a Cisco IP device, add the Record softkey or programmable line key to the device template.

To add the Record softkey, use the Device > Device Settings > Softkey Template menu option in Cisco Unified Communications Manager Administration to create or modify a nonstandard softkey template. Configure the softkey layout for call state *connected* to have the Record softkey in the selected softkeys list.

To add the Record programmable line key, use the Device > Device Settings > Phone Button Template menu option in Cisco Unified Communications Manager Administration. Enter the button template name, feature, and label.

4.4. Compatible handsets

Below is a list of compatible handsets which support the Built-in-Bridge feature.

Phone model	Status
Cisco 6901	not supported
Cisco 12 S	not supported
Cisco 12 SP	not supported
Cisco 30 SP+	not supported
Cisco 3905	not supported
Cisco 3911	not supported
Cisco 6901	not supported
Cisco 6911	supported
Cisco 6921	supported
Cisco 6941	supported
Cisco 6945	supported
Cisco 6961	supported
Cisco 7811	supported
Cisco 7821	supported
Cisco 7841	supported
Cisco 7861	supported
Cisco 7902	not supported
Cisco 7905	not supported
Cisco 7906	supported
Cisco 7910	not supported
Cisco 7911	supported
Cisco 7912	not supported
Cisco 7914 Sidecar	supported
Cisco 7915 Sidecar	supported
Cisco CKEM Sidecar	supported
Cisco 7920	not supported
Cisco 7921	supported
Cisco 7925	supported
Cisco 7926	supported
Cisco 7931	supported
Cisco 7935	not supported
Cisco 7936	not supported
Cisco 7937	supported
Cisco 7940	not supported
Cisco 7941	supported

Cisco 7941G-GE	supported
Cisco 7942	supported
Cisco 7945	supported
Cisco 7960	not supported
Cisco 7961	supported
Cisco 7961G-GE	supported
Cisco 7962	supported
Cisco 7965	supported
Cisco 7970	supported
Cisco 7971	supported
Cisco 7975	supported
Cisco 7985	supported
Cisco 8811	supported
Cisco 8831	supported
Cisco 8841	supported
Cisco 8845	supported
Cisco 8851	supported
Cisco 8861	supported
Cisco 8865	supported
Cisco 8941	supported
Cisco 8945	supported
Cisco 8961	supported
Cisco 9951	supported
Cisco 9971	supported
Cisco DX650	supported
Cisco E20	not supported
Cisco EX60	not supported
Cisco EX90	not supported
Cisco CTS 500	not supported
Cisco CTS 500-32	not supported
Cisco ATA 186	not supported
Cisco ATA 187	not supported
Cisco ATA 188	not supported
Cisco IP Communicator	supported
Cisco Jabber for Windows	supported
Cisco Jabber for Mac	supported
Cisco Jabber for iPad	not supported
Cisco Jabber for Android	not supported
Cisco Unified Personal Communicator	not supported
Cisco VGC Phone	not supported
VG224	not supported
VG248	not supported

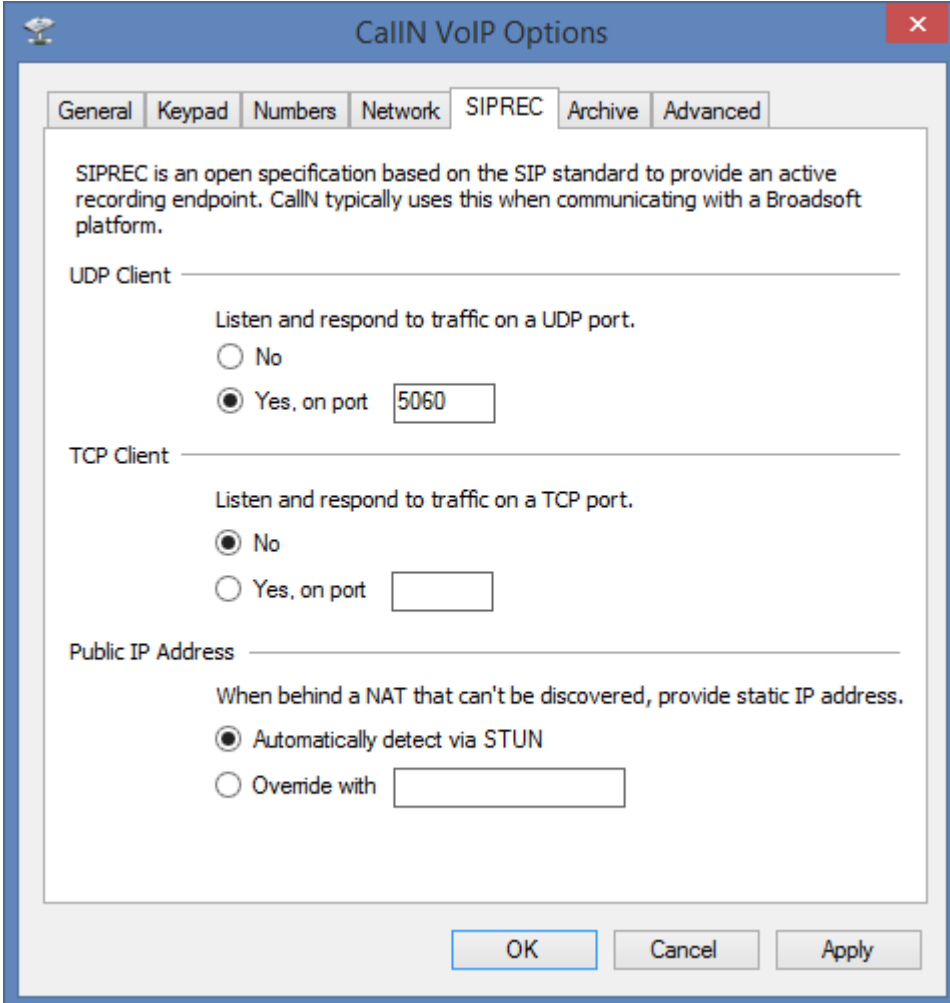
CTI Port	not supported
CTI Remote Device	not supported
CTI Route Point	not supported

Note: List of supported handsets can also be found at
<https://developer.cisco.com/site/uc-manager-sip/documents/supported/>

5. Configuration of CallN

5.1. Configure call recording client software

The recording client contains various setting to enable communication with the CUCM Platform. Please make sure they match the settings configured in section 3.3 – Create a SIP trunk that points to the recording server(s).



CallN VoIP Options

General | Keypad | Numbers | Network | **SIPREC** | Archive | Advanced

SIPREC is an open specification based on the SIP standard to provide an active recording endpoint. CallN typically uses this when communicating with a Broadsoft platform.

UDP Client

Listen and respond to traffic on a UDP port.

☐ No

☒ Yes, on port

TCP Client

Listen and respond to traffic on a TCP port.

☒ No

☐ Yes, on port

Public IP Address

When behind a NAT that can't be discovered, provide static IP address.

☒ Automatically detect via STUN

☐ Override with

OK Cancel Apply

5.1.1. UDP Client

The UDP port number to listen on for incoming SIP messages.

5.1.2. TCP Client

The TCP port number to listen on for incoming SIP messages.

5.1.3. Public IP Address

When the machine is behind a firewall and the Public IP address cannot be discovered via STUN then enter an override public IP address in this field.

5.2. Configure machine firewall

5.2.1. Incoming rules

Make sure the machine allows the following inbound traffic.

Please note: When receiving traffic from the internet, it is also best practice to limit traffic to the incoming source IP address as well.

Protocol	Port	Description
UDP	5060	When listening as a SIP UDP client, the port that was selected. By default, usually 5060.
TCP	5060	When listening as a SIP TCP client, the port that was selected. By default, usually 5060.
UDP	16384 - 32767	Port range for RTP media.

5.2.1. Outgoing rules

Generally, by default traffic is not limited outbound, but make sure the machine allows the following outbound traffic.

Protocol	Port	Description
UDP	5060	When listening as a SIP UDP client, the port that was selected. By default, usually 5060.
TCP	5060	When listening as a SIP TCP client, the port that was selected. By default, usually 5060.
UDP	16384 - 32767	Port range for RTP media. Verify with CUCM.

6. TAPI Setup

The following describes how to configure your Cisco CUCM platform for TAPI integration which will allow additional metadata to be captured with each call. The primary one is call direction.

Note: This is an additional set after setting up your CUCM for Active recording.

For CISCO TSP Windows compatibility and TSP CUCM compatibility please check the following links:

<https://developer.cisco.com/site/tapi/documents/supported-windows-os/>

<https://developer.cisco.com/site/tapi/documents/tapi-operations-by-release/>

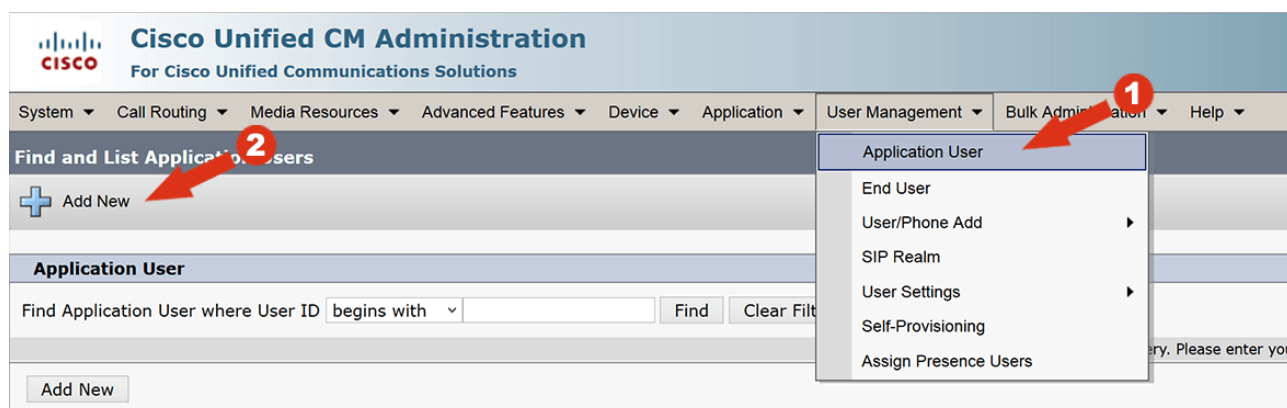
7. Create a TAPI user

An application user is required for CallN to monitor and control handsets.

Open the Cisco Unified CM Administration web portal

Navigate to User Management / Application User


Click the Add New button



In the Application User Information set the User ID as CallN and the password as with high strength as per your password policies.


In the list **Available Devices** select all of the devices which should be monitored and click arrow **V** to move these devices to the list **Controlled Devices**.


In case of Extension Mobility, you can use **CTI Controlled Device Profiles** instead of **Controlled Devices** list.


Cisco Unified CM Administration
 For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾

Application User Configuration

 Save

Status
 Status: Ready

Application User Information

User ID*
 Password
 Confirm Password
 Digest Credentials
 Confirm Digest Credentials
 BLF Presence Group*
☐ Accept Presence Subscription
☐ Accept Out-of-dialog REFER
☐ Accept Unsolicited Notification
☐ Accept Replaces Header

Device Information

Available Devices

Auto-registration Template
 Sample Device Template with TAG usage examples
 SEP001EBE90DACA

☒

Controlled Devices

SEP108CCF7416B6

Available Profiles

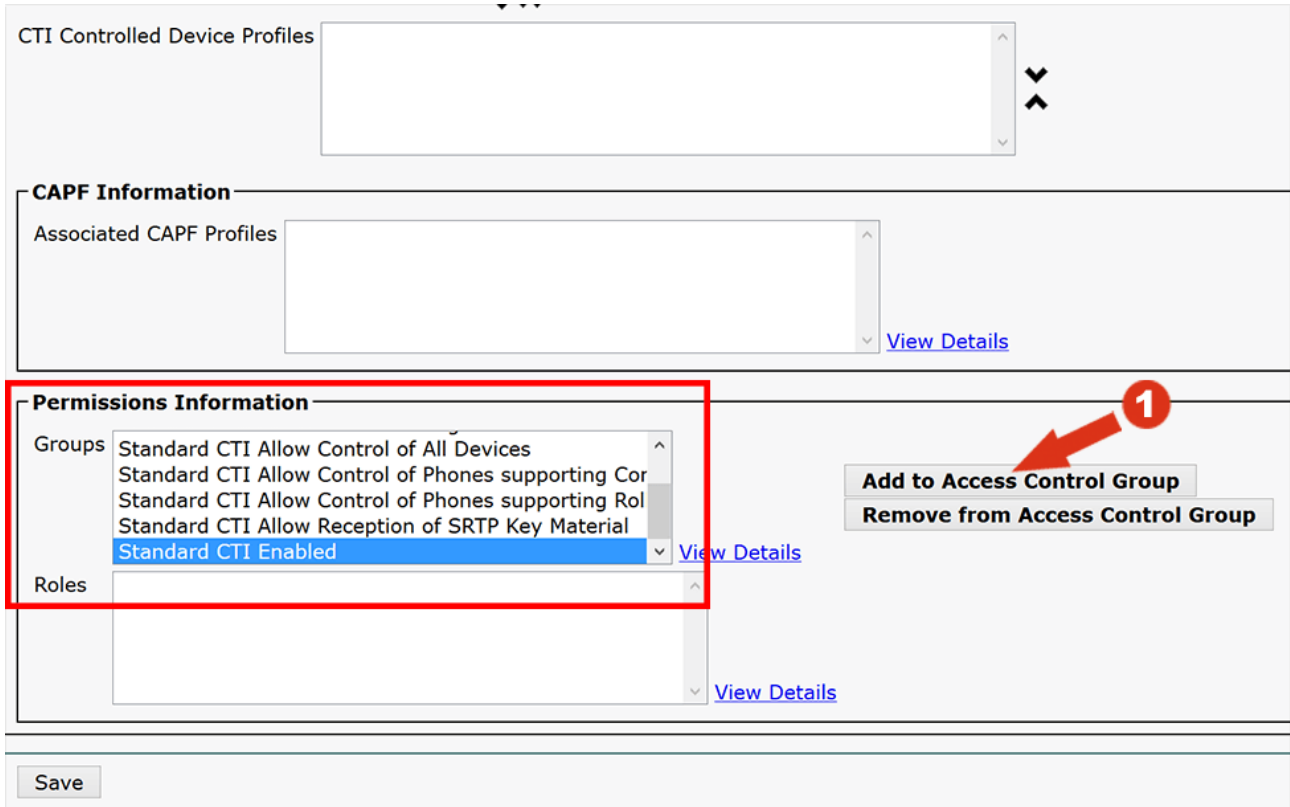
☒

CTI Controlled Device Profiles

☒

Device Association
[Find more Route Points](#)

In the section **Permissions Information** click the **Add to Access Control Group** button to select permissions for application user.



CTI Controlled Device Profiles

CAPF Information

Associated CAPF Profiles

Permissions Information

Groups

- Standard CTI Allow Control of All Devices
- Standard CTI Allow Control of Phones supporting Cor
- Standard CTI Allow Control of Phones supporting Rol
- Standard CTI Allow Reception of SRTP Key Material
- Standard CTI Enabled

Roles

Add to Access Control Group

Remove from Access Control Group

1

Save

In the new pop-up window select the following required options:

- Standard CTI Allow Control of All Devices
- Standard CTI Allow Control of Phones supporting Connected Xfer and conf
- Standard CTI Allow Control of Phones supporting Rollover Mode
- Standard CTI Enabled

Other options are not required.

Find and List Access Control Groups

☐ Select All
 ☐ Clear All
 ☐ Add Selected

<input type="checkbox"/>	Standard CCM End Users
<input type="checkbox"/>	Standard CCM Gateway Administration
<input type="checkbox"/>	Standard CCM Phone Administration
<input type="checkbox"/>	Standard CCM Read Only
<input type="checkbox"/>	Standard CCM Server Maintenance
<input type="checkbox"/>	Standard CCM Server Monitoring
<input type="checkbox"/>	Standard CCM Super Users
<input checked="" type="checkbox"/>	Standard CTI Allow Call Monitoring
<input type="checkbox"/>	Standard CTI Allow Call Park Monitoring
<input checked="" type="checkbox"/>	Standard CTI Allow Call Recording
<input type="checkbox"/>	Standard CTI Allow Calling Number Modification
<input checked="" type="checkbox"/>	Standard CTI Allow Control of All Devices
<input checked="" type="checkbox"/>	Standard CTI Allow Control of Phones supporting Connected Xfer and conf
<input checked="" type="checkbox"/>	Standard CTI Allow Control of Phones supporting Rollover Mode
<input checked="" type="checkbox"/>	Standard CTI Allow Reception of SRTP Key Material
<input checked="" type="checkbox"/>	Standard CTI Enabled
<input type="checkbox"/>	Standard CTI Secure Connection
<input type="checkbox"/>	Standard Confidential Access Level Users
<input type="checkbox"/>	Standard EM Authentication Proxy Rights
<input type="checkbox"/>	Standard Packet Sniffer Users
<input type="checkbox"/>	Standard RealtimeAndTraceCollection
<input type="checkbox"/>	Standard TabSync User
<input type="checkbox"/>	Third Party Application Users

Save the settings of new application user.

8. Install TAPI driver

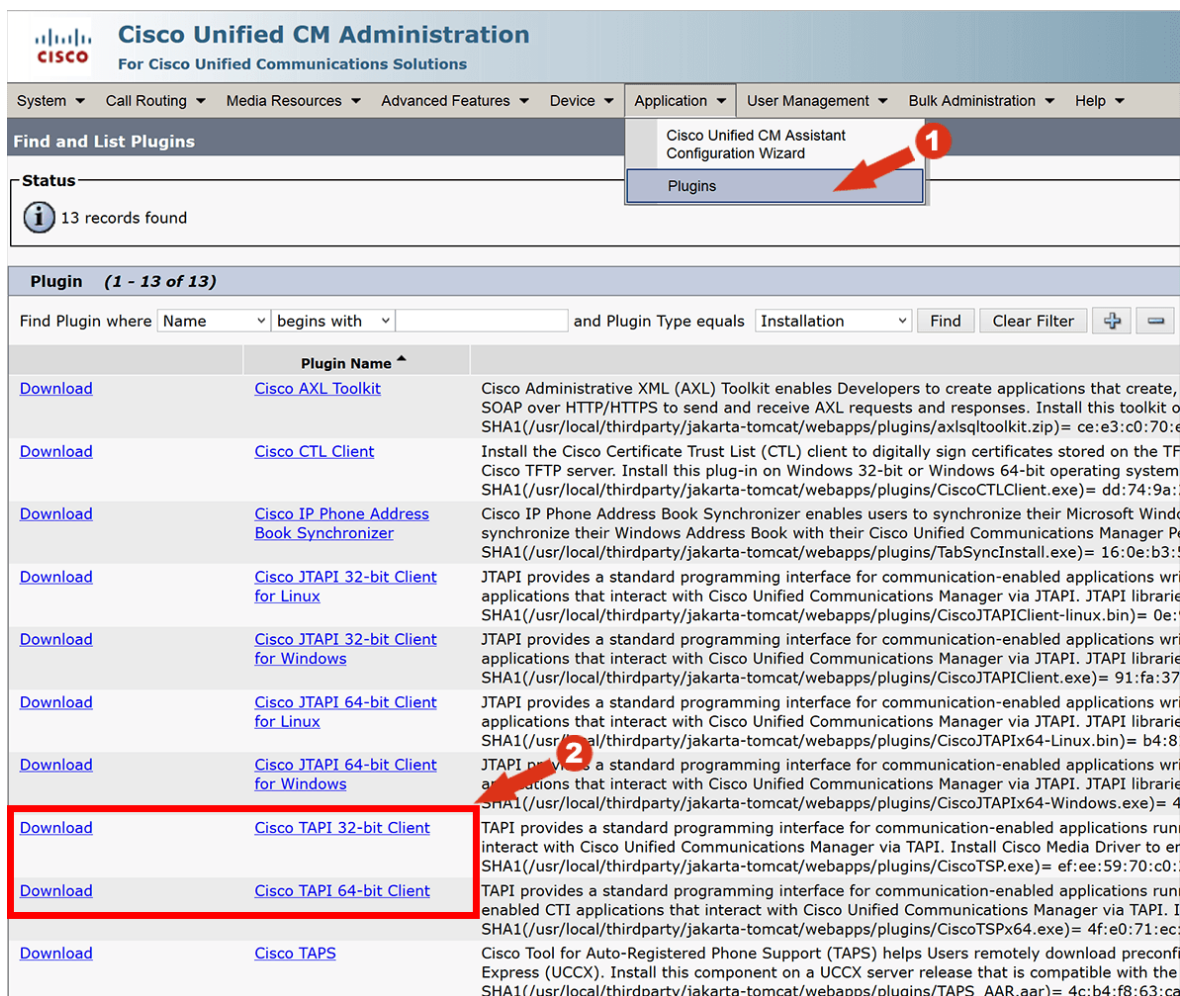
The Cisco TAPI Service Provider (TSP) is a TAPI driver that is installed on the Windows server running the CallN recording software that allows communication between CallN and the Cisco UCM. For TSP Windows compatibility and TSP CUCM compatibility please see the following links:

<https://developer.cisco.com/site/tapi/documents/supported-windows-os/>
<https://developer.cisco.com/site/tapi/documents/tapi-operations-by-release/>

8.1. Download the driver

The installer for the TAPI driver can be obtained from the Cisco Unified CM Administration portal using the following steps:

1. Open **Cisco Unified CM Administration** portal in a web browser and log in with an administrator account.
2. Navigate to **Application** menu across the top of the site and click the **Plugins** link.
3. On the Find and List Plugins page, enter "**Cisco TAPI**" into the search field and click **Find**.
4. The plugin list will load. Click the **Download** link on either 32-bit or 64-bit client depending on your operating system.



Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Find and List Plugins

Status
13 records found

Plugin (1 - 13 of 13)

Find Plugin where Name ▾ begins with ▾ and Plugin Type equals Installation ▾ Find Clear Filter + -

	Plugin Name ^	
Download	Cisco AXL Toolkit	Cisco Administrative XML (AXL) Toolkit enables Developers to create applications that create, SOAP over HTTP/HTTPS to send and receive AXL requests and responses. Install this toolkit o SHA1(/usr/local/thirdparty/jakarta-tomcat/webapps/plugins/axlsqltoolkit.zip)= ce:e3:c0:70:ε
Download	Cisco CTL Client	Install the Cisco Certificate Trust List (CTL) client to digitally sign certificates stored on the TF Cisco TFTP server. Install this plug-in on Windows 32-bit or Windows 64-bit operating system SHA1(/usr/local/thirdparty/jakarta-tomcat/webapps/plugins/CiscoCTLClient.exe)= dd:74:9a::
Download	Cisco IP Phone Address Book Synchronizer	Cisco IP Phone Address Book Synchronizer enables users to synchronize their Microsoft Wind synchronize their Windows Address Book with their Cisco Unified Communications Manager P SHA1(/usr/local/thirdparty/jakarta-tomcat/webapps/plugins/TabSyncInstall.exe)= 16:0e:b3:!
Download	Cisco JTAPI 32-bit Client for Linux	JTAPI provides a standard programming interface for communication-enabled applications wri applications that interact with Cisco Unified Communications Manager via JTAPI. JTAPI librarie SHA1(/usr/local/thirdparty/jakarta-tomcat/webapps/plugins/CiscoJTAPIClient-linux.bin)= 0e:!
Download	Cisco JTAPI 32-bit Client for Windows	JTAPI provides a standard programming interface for communication-enabled applications wri applications that interact with Cisco Unified Communications Manager via JTAPI. JTAPI librarie SHA1(/usr/local/thirdparty/jakarta-tomcat/webapps/plugins/CiscoJTAPIClient.exe)= 91:fa:37
Download	Cisco JTAPI 64-bit Client for Linux	JTAPI provides a standard programming interface for communication-enabled applications wri applications that interact with Cisco Unified Communications Manager via JTAPI. JTAPI librarie SHA1(/usr/local/thirdparty/jakarta-tomcat/webapps/plugins/CiscoJTAPIx64-Linux.bin)= b4:8
Download	Cisco JTAPI 64-bit Client for Windows	JTAPI provides a standard programming interface for communication-enabled applications wri applications that interact with Cisco Unified Communications Manager via JTAPI. JTAPI librarie SHA1(/usr/local/thirdparty/jakarta-tomcat/webapps/plugins/CiscoJTAPIx64-Windows.exe)= 4
Download	Cisco TAPI 32-bit Client	TAPI provides a standard programming interface for communication-enabled applications runi interact with Cisco Unified Communications Manager via TAPI. Install Cisco Media Driver to er SHA1(/usr/local/thirdparty/jakarta-tomcat/webapps/plugins/CiscoTSP.exe)= ef:ee:59:70:c0:.
Download	Cisco TAPI 64-bit Client	TAPI provides a standard programming interface for communication-enabled applications runi enabled CTI applications that interact with Cisco Unified Communications Manager via TAPI. I SHA1(/usr/local/thirdparty/jakarta-tomcat/webapps/plugins/CiscoTSPx64.exe)= 4f:e0:71:ec:
Download	Cisco TAPS	Cisco Tool for Auto-Registered Phone Support (TAPS) helps Users remotely download preconfi Express (UCCX). Install this component on a UCCX server release that is compatible with the SHA1(/usr/local/thirdparty/jakarta-tomcat/webapps/plugins/TAPS_AAR.aar)= 4c:b4:f8:63:ca

8.2. Install the driver

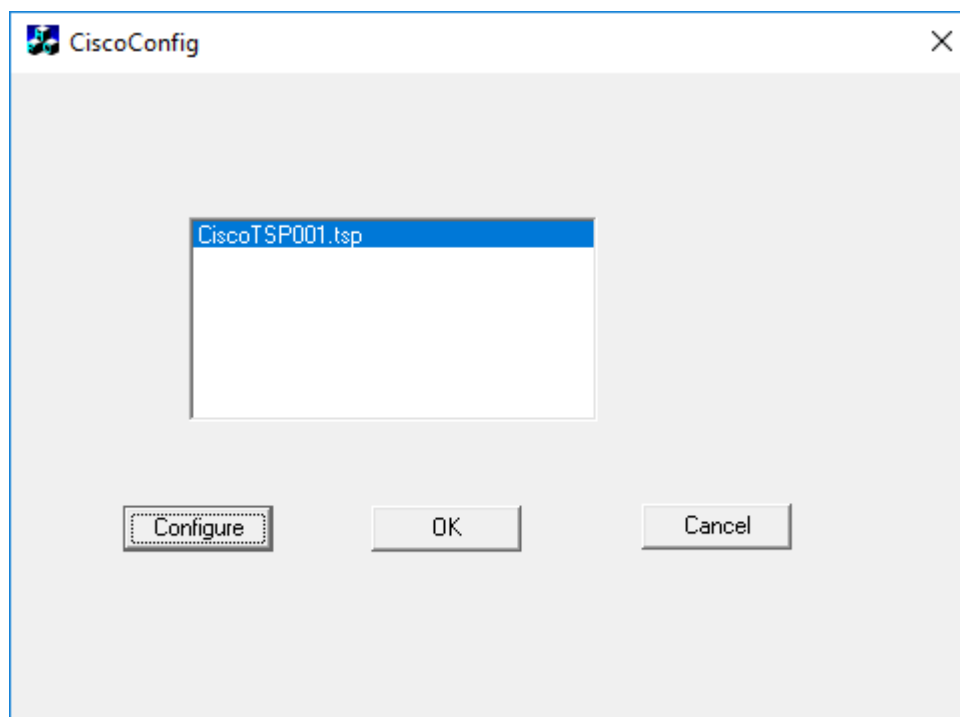
Open the **CiscoTSP.exe** installer and follow instructions on screen. You will be asked for Cisco Call Manager address and application user/password as created in previous steps.

Restart operating system is required after installation of Cisco TAPI driver.

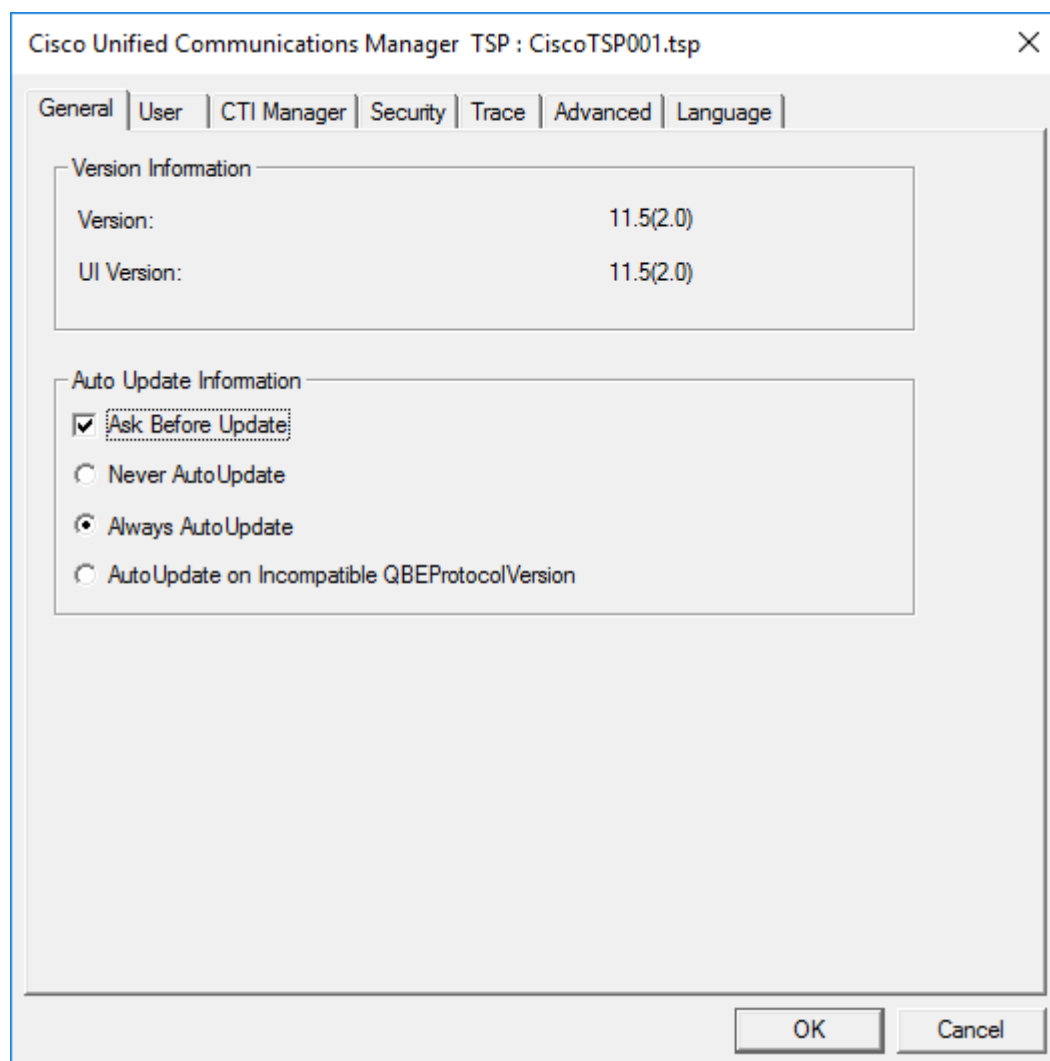
8.3. Configure the driver

Open the **CiscoConfig.exe** utility, which is installed with TAPI driver.

Click **Configure** button.

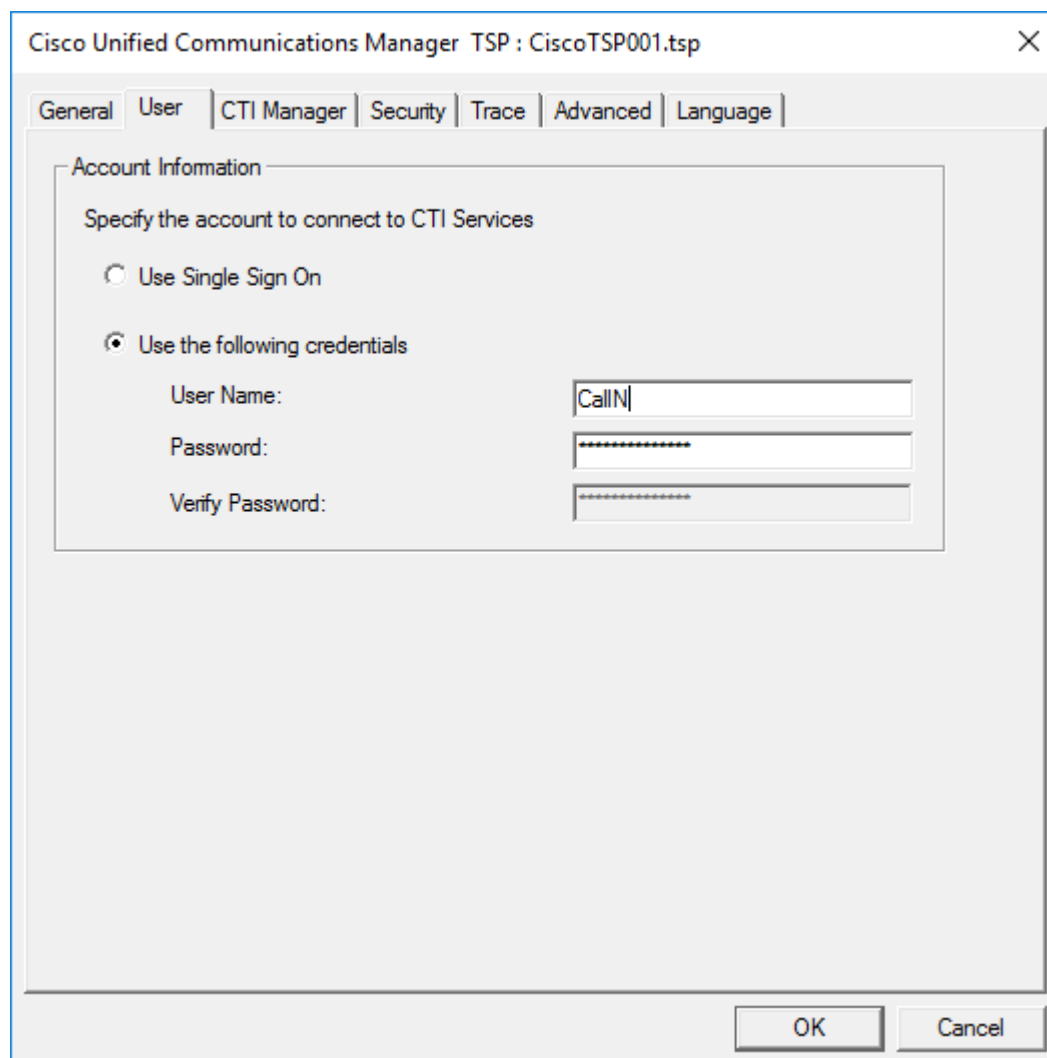


The general page will open and display what version is currently installed.



Click the **User** tab

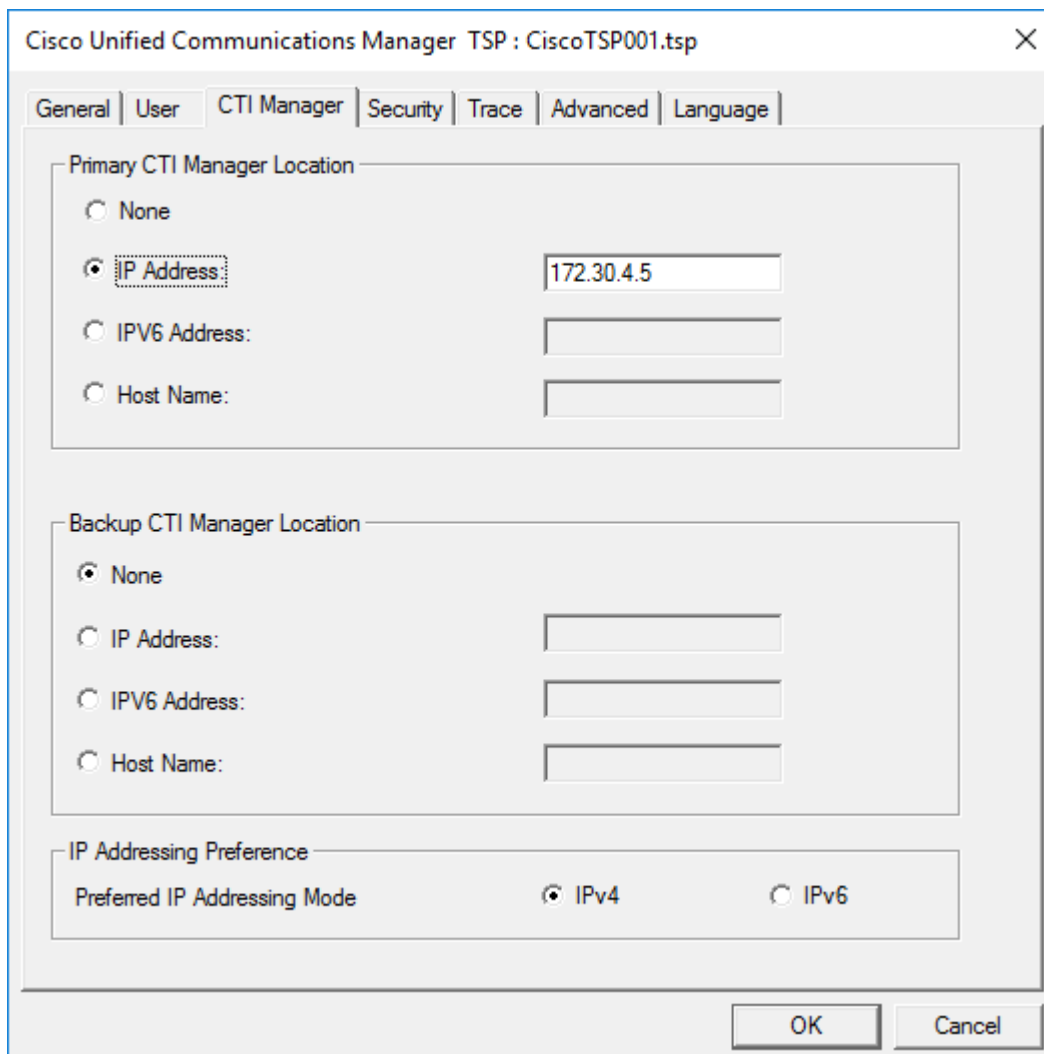
Enter the credentials that were used when created the Application User.



The screenshot shows a configuration window titled "Cisco Unified Communications Manager TSP : CiscoTSP001.tsp". It has several tabs: General, User, CTI Manager, Security, Trace, Advanced, and Language. The "User" tab is selected. Inside the "User" tab, there is a section titled "Account Information" with the instruction "Specify the account to connect to CTI Services". There are two radio buttons: "Use Single Sign On" (unselected) and "Use the following credentials" (selected). Below the selected radio button, there are three text input fields: "User Name:" with the value "CallN", "Password:" with masked characters, and "Verify Password:" with masked characters. At the bottom right of the window are "OK" and "Cancel" buttons.

Click the **CTI Manager** tab

Enter the IP address of your Call Manager installation.



The screenshot shows the 'Cisco Unified Communications Manager TSP : CiscoTSP001.tsp' window with the 'CTI Manager' tab selected. The window contains three main sections: 'Primary CTI Manager Location', 'Backup CTI Manager Location', and 'IP Addressing Preference'. In the 'Primary CTI Manager Location' section, the 'IP Address' radio button is selected, and the text '172.30.4.5' is entered in the adjacent text box. The 'Backup CTI Manager Location' section has the 'None' radio button selected. The 'IP Addressing Preference' section shows the 'Preferred IP Addressing Mode' with 'IPv4' selected. At the bottom right are 'OK' and 'Cancel' buttons.

Cisco Unified Communications Manager TSP : CiscoTSP001.tsp

General | User | **CTI Manager** | Security | Trace | Advanced | Language

Primary CTI Manager Location

☐ None

☒ IP Address: 172.30.4.5

☐ IPV6 Address:

☐ Host Name:

Backup CTI Manager Location

☒ None

☐ IP Address:

☐ IPV6 Address:

☐ Host Name:

IP Addressing Preference

Preferred IP Addressing Mode ☒ IPv4 ☐ IPv6

OK Cancel

8.4. Optional – Enable trace logging

Open the **CiscoConfig.exe** utility, which is installed with TAPI driver.

Click **Configure** button.

Click the **Trace** tab

Enable the Trace **On** check box.

Set the File Size to 10

Set the No. of files to 100

Set up a **Directory** that is the path for the trace log. For example, C:\CiscoTAPILog

Select **Detailed** to log internal messages for debugging purposes.

Enable the following events; -

- **TSP Trace** to trace the TSP internal messages.
- **CTI Trace** to trace the messages sent between CTI and TSP.
- **TSPI Trace** to trace the requests and events that are sent between TSP and TAPI.

Cisco Unified Communications Manager TSP : CiscoTSP001.tsp

General | User | CTI Manager | Security | **Trace** | Advanced | Language

Trace

☒ On

File Size

No. of files

Directory

☒ TSP Trace ☐ Error ☒ Detailed

☒ CTI Trace

☒ TSPI Trace

OK Cancel