



# CALLN HOSTED CALL RECORDING CISCO CUCM SETUP GUIDE

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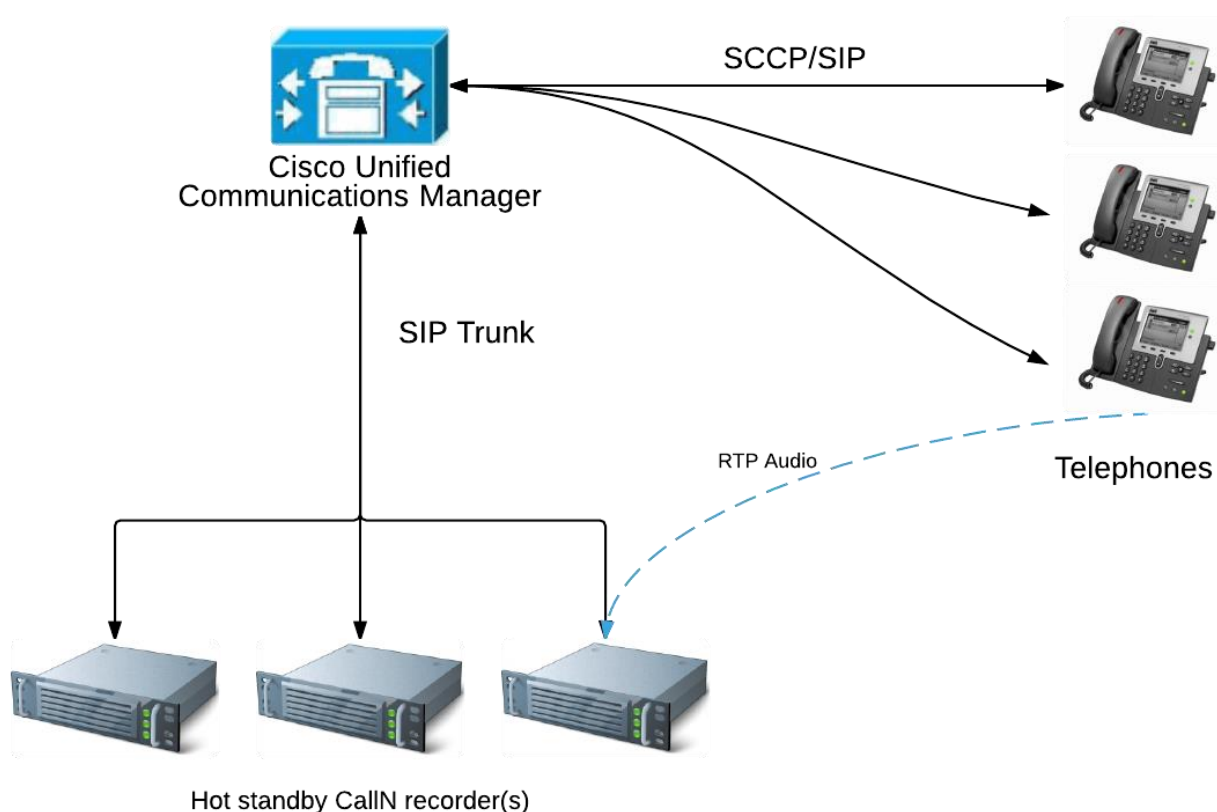
# 1. Introduction

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.

# 2. Connectivity

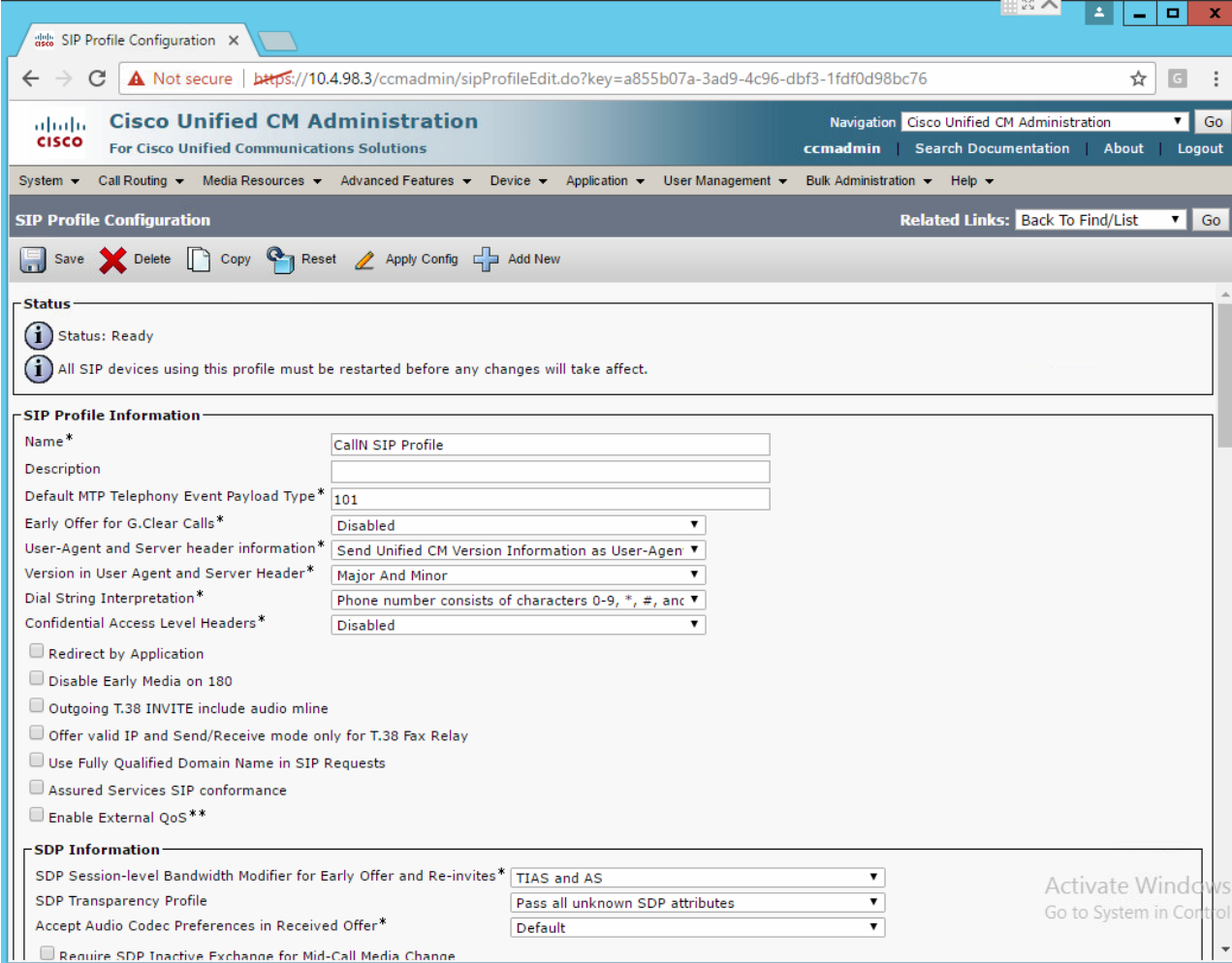
CallN utilises the Built-in-Bridge feature of the Cisco IP handset to record calls (see compatible handset list). When a call is to be recorded the CUCM initiates a SIP call to the CallN platform via a SIP trunk interface and then informs the handset to send the CallN server the RTP audio directly.



## 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.



The screenshot displays the 'SIP Profile Configuration' page in the Cisco Unified CM Administration interface. The browser address bar shows a URL starting with 'https://10.4.98.3/ccmadmin/sipProfileEdit.do?'. The page title is 'Cisco Unified CM Administration'. The navigation menu includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The 'SIP Profile Configuration' section is active, showing a 'Status' of 'Ready' and a message: 'All SIP devices using this profile must be restarted before any changes will take effect.' The 'SIP Profile Information' section contains the following fields and values:

- Name\*: CallN SIP Profile
- Description: (empty)
- 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, \*, #, and
- Confidential Access Level Headers\*: Disabled
- Redirect by Application: (unchecked)
- Disable Early Media on 180: (unchecked)
- Outgoing T.38 INVITE include audio mline: (unchecked)
- Offer valid IP and Send/Receive mode only for T.38 Fax Relay: (unchecked)
- Use Fully Qualified Domain Name in SIP Requests: (unchecked)
- Assured Services SIP conformance: (unchecked)
- Enable External QoS\*\*: (unchecked)

The 'SDP Information' section contains the following fields and values:

- 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: (unchecked)

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

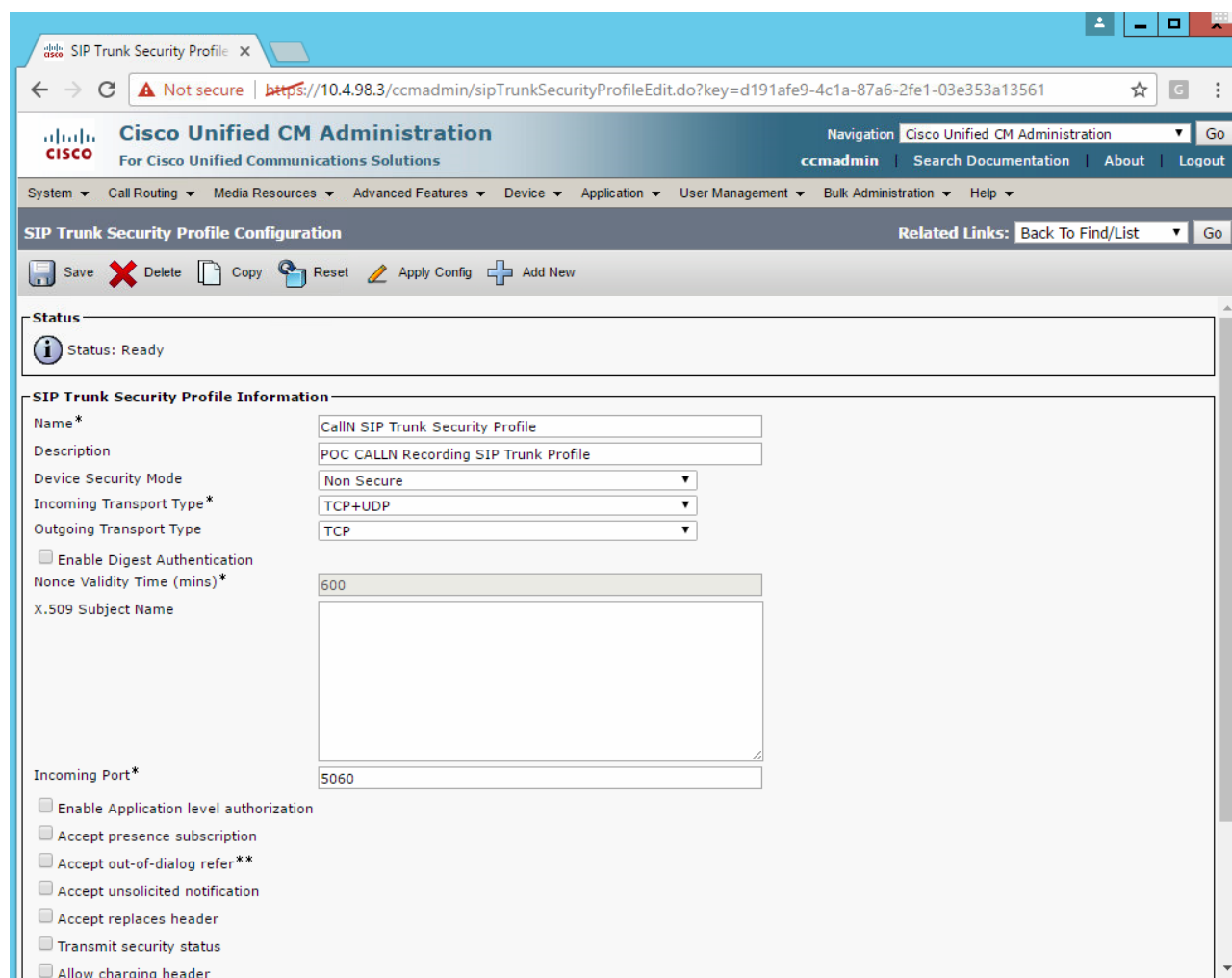
**Trunk Specific Configuration / Deliver Conference Bridge Identifier** – Enable this setting.

**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.



The screenshot shows the Cisco Unified CM Administration web interface. The browser address bar displays a URL starting with https://10.4.98.3/ccmadmin/sipTrunkSecurityProfileEdit.do. The page title is 'SIP Trunk Security Profile Configuration'. The navigation bar includes tabs for System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area shows the configuration form for a SIP Trunk Security Profile. The form includes fields for Name, Description, Device Security Mode, Incoming Transport Type, Outgoing Transport Type, and checkboxes for Enable Digest Authentication, Accept presence subscription, Accept out-of-dialog refer, Accept unsolicited notification, Accept replaces header, Transmit security status, and Allow charging header. The Incoming Port is set to 5060.

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'.

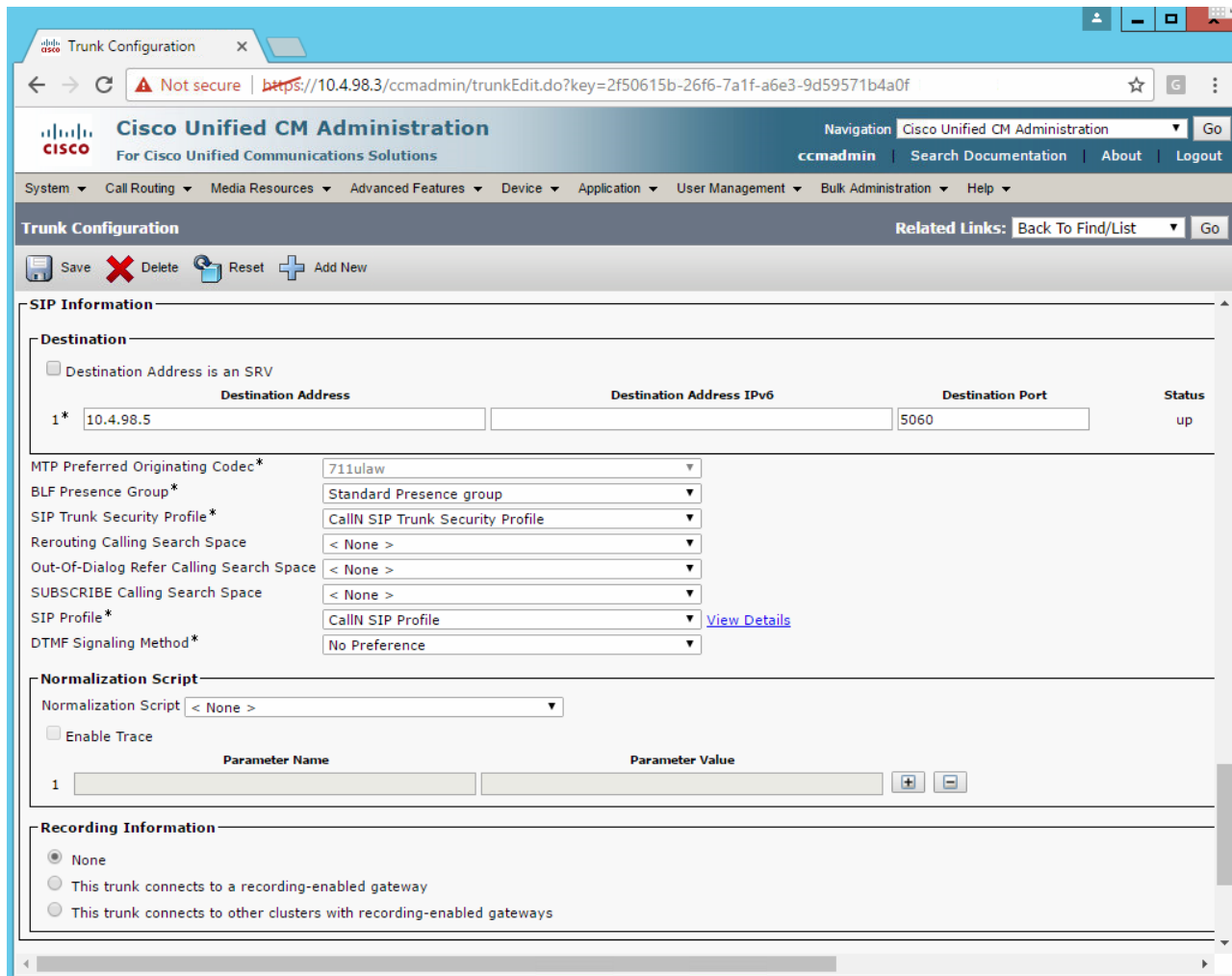
SIP Trunk Security Profile Information / Enable Digest Authentication – Uncheck.

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



The screenshot shows the 'Trunk Configuration' page in Cisco Unified CM Administration. The 'SIP Information' section is expanded, showing the following configuration:

- Destination:**
  - ☐ Destination Address is an SRV
  - Destination Address:** 10.4.98.5
  - Destination Address IPv6:**
  - Destination Port:** 5060
  - Status:** up
- MTP Preferred Originating Codec\*:** 711ulaw
- BLF Presence Group\*:** Standard Presence group
- SIP Trunk Security Profile\*:** CallN SIP Trunk Security Profile
- Rerouting Calling Search Space:** < None >
- Out-Of-Dialog Refer Calling Search Space:** < None >
- SUBSCRIBE Calling Search Space:** < None >
- SIP Profile\*:** CallN SIP Profile (with a 'View Details' link)
- DTMF Signaling Method\*:** No Preference

The **Normalization Script** section shows:

- Normalization Script:** < None >
- ☐ Enable Trace
- Parameter Name / Parameter Value table:**

	Parameter Name	Parameter Value
1		

The **Recording Information** section at the bottom has three radio button options:

- ☒ 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 'CallN\_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 CallN SIP Trunk Security profile that you configured earlier, probably 'CallN SIP Trunk Security Profile'.

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

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

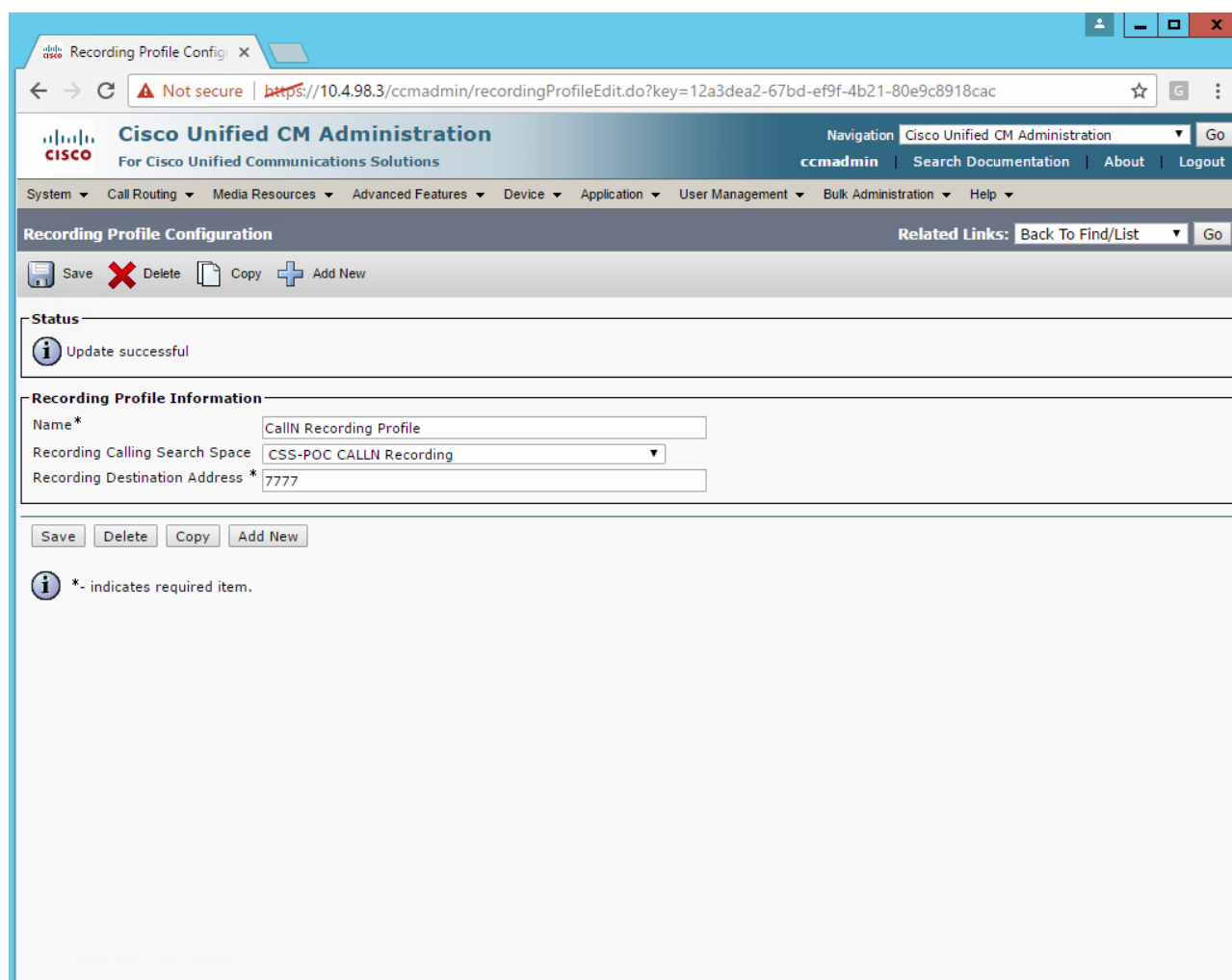
SIP Information / Destination / Destination Port – set as 5060. This should match the configuration in the CallN recording client.

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 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

Recording Profile Configuration Related Links: Back To Find/List Go

Save Delete Copy Add New

**Status**

Update successful

**Recording Profile Information**

Name\* CallN Recording Profile

Recording Calling Search Space CSS-POC CALLN Recording

Recording Destination Address\* 7777

Save Delete Copy Add New

\*- indicates required item.

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.

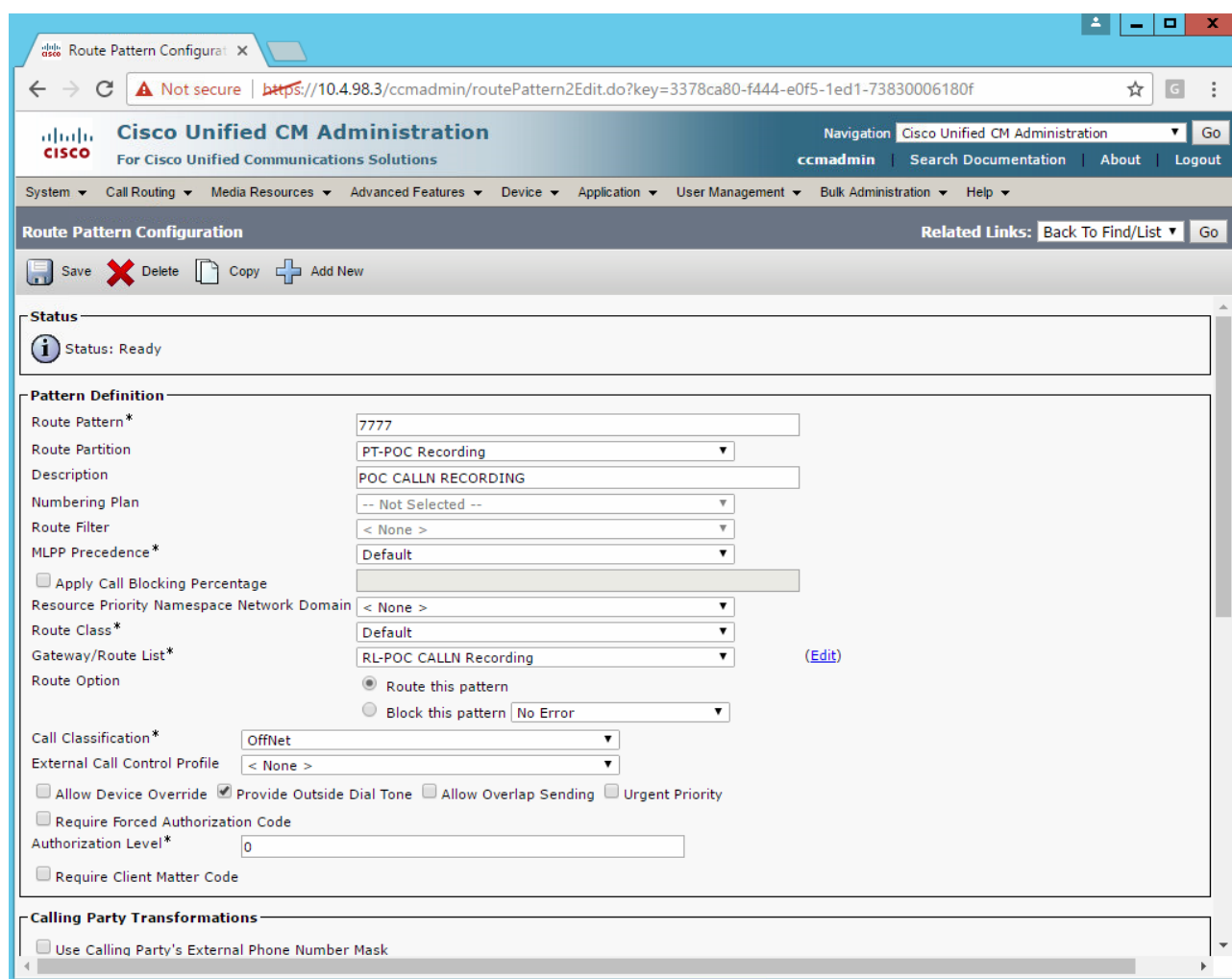
Recording Profile Information / Recording Destination Address – Set as a directory number that is associated with the recorder. This number should not clash with other number plan entries. A good example is 7777.

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 shows the Cisco Unified CM Administration web interface. The browser address bar displays a URL starting with 'https://10.4.98.3/ccmadmin/routePattern2Edit.do?'. The page title is 'Route Pattern Configuration'. The navigation bar includes 'Cisco Unified CM Administration' and 'ccmadmin'. The main content area is titled 'Route Pattern Configuration' and includes a 'Related Links' section with 'Back To Find/List'. The configuration form is divided into three main sections: 'Status' (showing 'Status: Ready'), 'Pattern Definition', and 'Calling Party Transformations'. 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' (unchecked), 'Resource Priority Namespace Network Domain' (< None >), 'Route Class\*' (Default), 'Gateway/Route List\*' (RL-POC CALLN Recording), 'Route Option' (Route this pattern), 'Call Classification\*' (OffNet), 'External Call Control Profile' (< None >), 'Allow Device Override' (unchecked), 'Provide Outside Dial Tone' (checked), 'Allow Overlap Sending' (unchecked), 'Urgent Priority' (unchecked), 'Require Forced Authorization Code' (unchecked), 'Authorization Level\*' (0), and 'Require Client Matter Code' (unchecked). The 'Calling Party Transformations' section has a checkbox for 'Use Calling Party's External Phone Number Mask' which is 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.

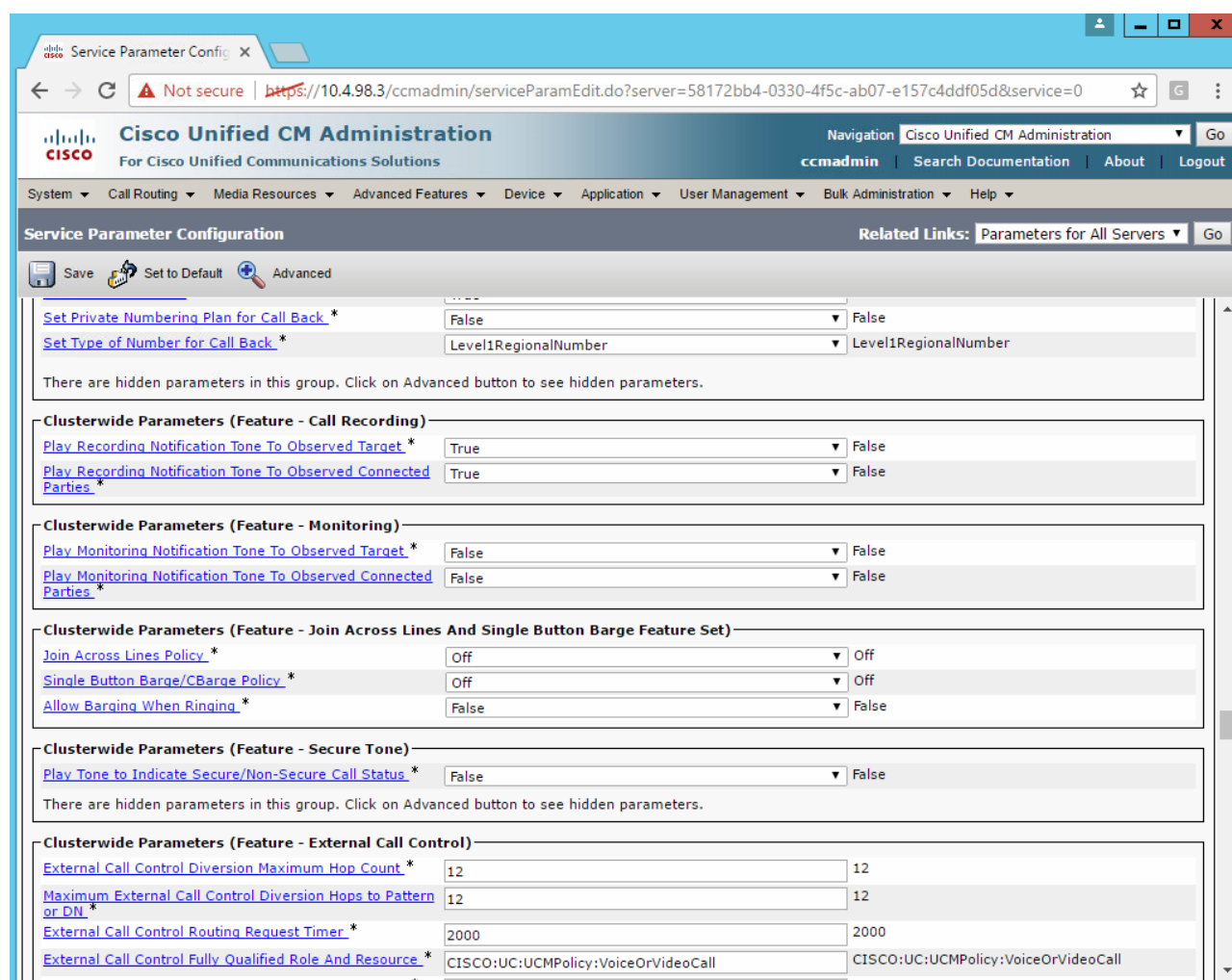
Gateway / Route List - Select the CallN SIP trunk that was created earlier.

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.

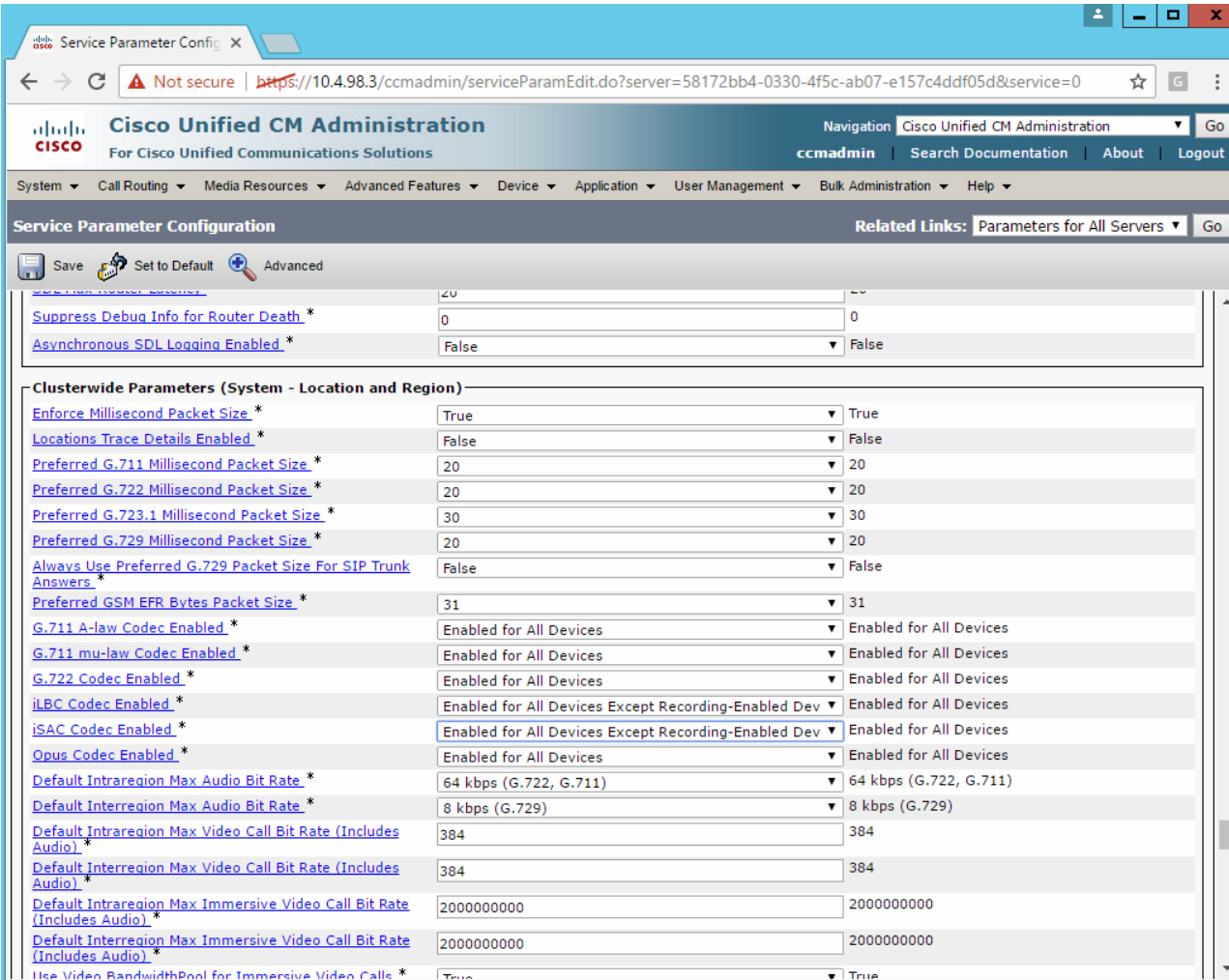
Click Save.

## 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 'Service Parameter Configuration' page in the Cisco Unified CM Administration interface. The 'Clusterwide Parameters (System - Location and Region)' section is expanded, displaying 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 include 'Enforce Millisecond Packet Size' (True), 'Locations Trace Details Enabled' (False), 'Preferred G.711 Millisecond Packet Size' (20), 'Preferred G.722 Millisecond Packet Size' (20), 'Preferred G.723.1 Millisecond Packet Size' (30), 'Preferred G.729 Millisecond Packet Size' (20), 'Always Use Preferred G.729 Packet Size For SIP Trunk Answers' (False), 'Preferred GSM EFR Bytes Packet Size' (31), 'G.711 A-law Codec Enabled' (Enabled for All Devices), 'G.711 mu-law Codec Enabled' (Enabled for All Devices), 'G.722 Codec Enabled' (Enabled for All Devices), 'iSAC Codec Enabled' (Enabled for All Devices Except Recording-Enabled Dev), 'Opus Codec Enabled' (Enabled for All Devices), 'Default Intra-region Max Audio Bit Rate' (64 kbps (G.722, G.711)), 'Default Inter-region Max Audio Bit Rate' (8 kbps (G.729)), 'Default Intra-region Max Video Call Bit Rate (Includes Audio)' (384), 'Default Inter-region Max Video Call Bit Rate (Includes Audio)' (384), 'Default Intra-region Max Immersive Video Call Bit Rate (Includes Audio)' (2000000000), 'Default Inter-region Max Immersive Video Call Bit Rate (Includes Audio)' (2000000000), and 'Use Video Bandwidth Pool for Immersive Video Calls' (True).

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)'

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## 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.

System • Call Routing • Media Resources • Voice Mail • Device • Application • User Management • Bulk Administration • Help •

### Phone Configuration

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

Save X Delete Copy Reset Add New

18 Call Pickup	Audio Source	
19 Conference List	Location*	Hub_None
20 Conference	AAR Group	< None >
21 Do Not Disturb	User Locale	< None >
22 End Call	Network Locale	< None >
23 Forward All	Built In Bridge*	On
24 Group Call Pickup	Privacy*	Off
25 Hold	Device Mobility	Default
26 Hunt Group Logout	Mode*	<a href="#">Mobility Settings</a>
27 <a href="#">Intercom [1] - Add a new Intercom</a>	Owner User ID	< None >
28 Malicious Call Identification	Phone Load Name	SCCP70.MU-1-0-13DEV
29 Meet Me Conference		
30 Mobility		
31 New Call		
32 Other Pickup		

Retrv Video Call as Audio

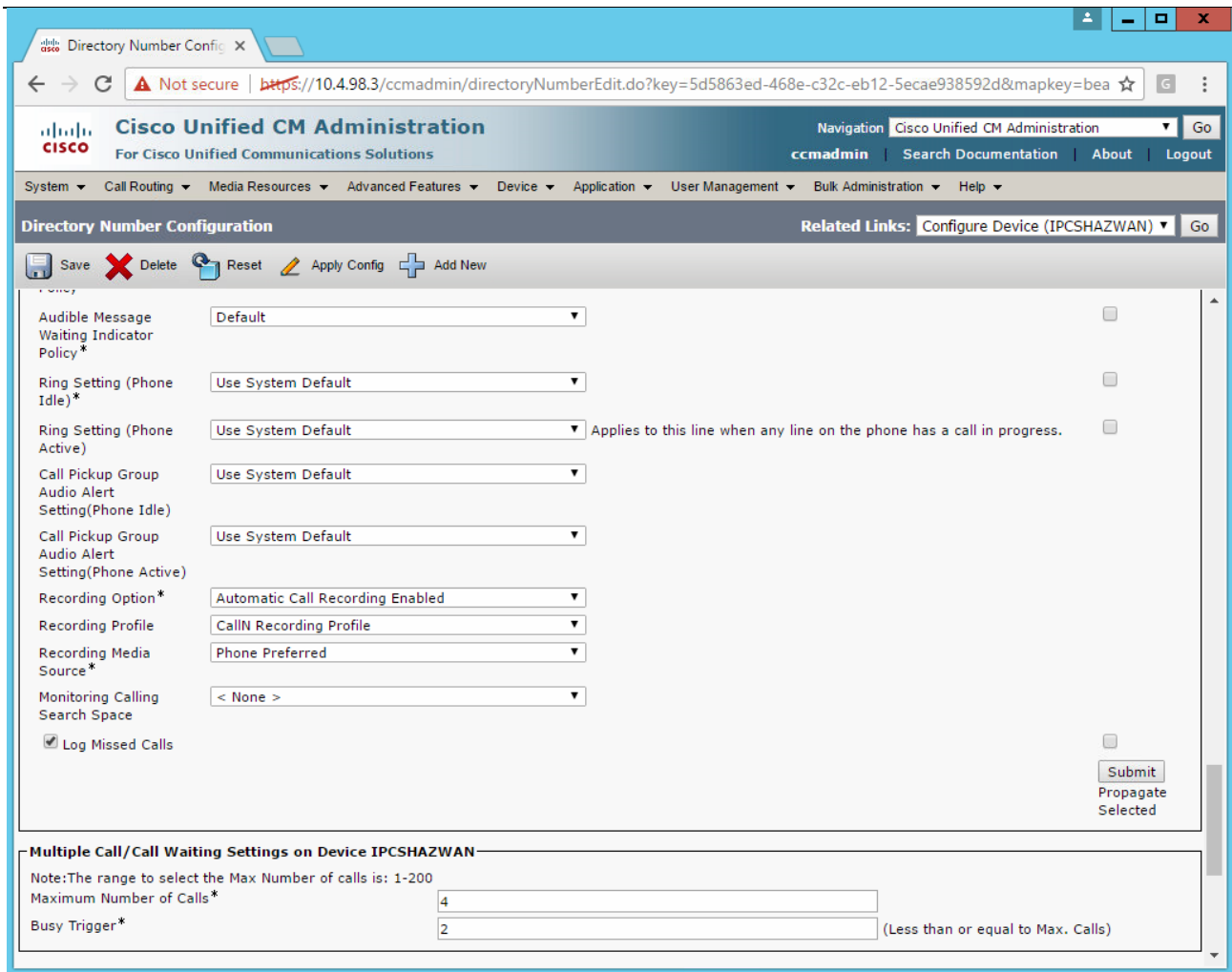
8202338

*Set BIB 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 web interface. The browser address bar displays a URL starting with https://10.4.98.3/ccmadmin/directoryNumberEdit.do. The page title is 'Directory Number Configuration'. The navigation bar includes links for System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Directory Number Configuration' and contains a list of settings for a directory number. The settings include: 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 (CallIN Recording Profile), Recording Media Source\* (Phone Preferred), and Monitoring Calling Search Space (< None >). There are also checkboxes for 'Log Missed Calls' and 'Propagate Selected'. At the bottom, there is a section for 'Multiple Call/Call Waiting Settings on Device IPCSHAZWAN' with input fields for 'Maximum Number of Calls\*' (4) and 'Busy Trigger\*' (2).

Set the Recording Option to

- Automatic Call Recording Enabled
- Selective Call Recording Enabled

Set Recording Profile to 'CallIN 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

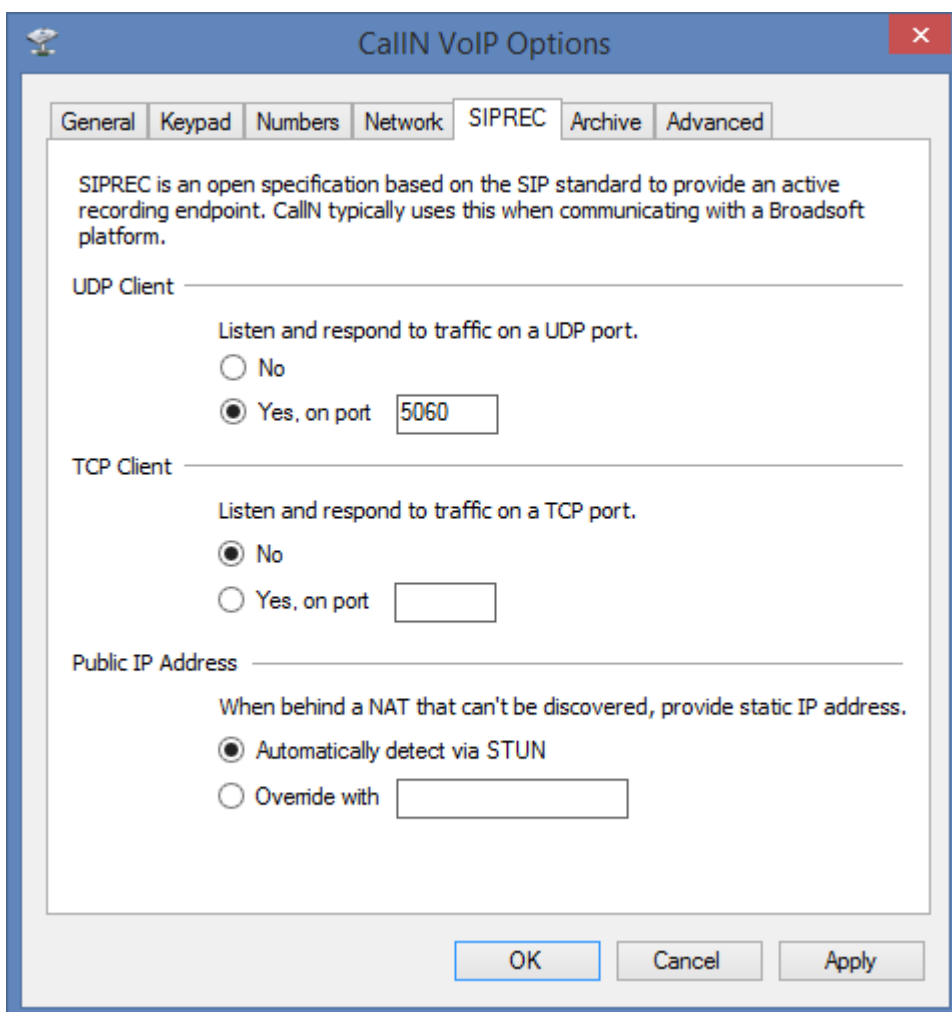
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CTI Port	not supported
CTI Remote Device	not supported
CTI Route Point	not supported

## 5. Configuration of CallN

### 5.1. Configure call recording client software

The recording client contains various settings 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).



The screenshot shows the 'CallN VoIP Options' dialog box with the 'SIPREC' tab selected. The dialog has tabs for General, Keypad, Numbers, Network, SIPREC, Archive, and Advanced. The SIPREC tab contains the following settings:

- SIPREC Description:** 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

At the bottom of the dialog are buttons for OK, Cancel, and 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.