Cisco UCM
Integration Guide

Encore Workforce Optimization Solution
Version 7.1 or later

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For Dealer and Customer Use Only

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Introduction

The Encore system integrates with the Cisco UCM via the Cisco TSP Server. This integration allows the Encore system to successfully perform the following functions:

- Audio Collection – Capture the audio that needs to be recorded. If using Cisco Built in Bridge (BIB) Recording or Station-side RTP Packet Capture, the audio recording can be generated as either a stereo or mono recording file.
- Recording Control – Receive the necessary events that signal when the Encore must start and stop recording.
- Data Capture – Receive data associated with the call.

Supported Data Capture

The following is a list of the supported data elements that can be collected with each recording. Not every element is applicable for each call. For a description of each data element, refer to "Appendix 2: Glossary" on page 25.

- ACD Name[1]
- ACD Number [1]
- ANI/Caller ID
- Call Direction
- Call Origin
- Connected Identifier
- Consultation Call
- Device Name
- Dialed Number
- DNIS/Called ID
- Extension
- Fin Call Type[2]
- Fin Team Name[2]
- Global Call Identifier
- Hold Duration
- Other Call Identifier
- Other Party Name
- Other Party Number
- Recorded Party Number
- Redirecting Identifier
- Redirection Identifier

1. Available if using the Cisco UCCX or Finesse applications

2. Available if using the Cisco Finesse application

Supported Recording Features

The following matrix should be used to determine which audio collection is best for your business needs. If you find that more than one collection method will work for you, talk to your Encore representative about which method is more cost-effective. For a description of each feature, refer to "Appendix 1: Glossary" on page 25.

RECORDING FEATURE	AUDIO COLLECTION METHOD	
	STATION-SIDE RTP PACKET CAPTURE (PASSIVE CAPTURE)	SUBSCRIPTION-BASED SIP AUDIO STREAM
Max. Recording Ports per Server	500	500
Record External Calls	YES	YES

RECORDING FEATURE	AUDIO COLLECTION METHOD	
	STATION-SIDE RTP PACKET CAPTURE (PASSIVE CAPTURE)	SUBSCRIPTION-BASED SIP AUDIO STREAM
Record Internal Calls	YES	YES
Record Encrypted Calls		
Related Call Lookup	YES	YES
Suspend/Resume on Hold	YES	YES
Extension Mobility	YES	YES
Single Number Reach	NO	YES
Shared Line Recording[1]	NO	YES
Cisco Finesse Data Capture[2]	NO	YES

1. *For shared lines and multi-line phones using a different extension for each recorded (sub) line, Encore does not support screen recording. Also, Encore is unable to capture the agent name since many agents may use a shared line. This prevents Encore from associating a recording with a specific agent which will affect the evaluation process.*
2. *Cisco Finesse Data Capture cannot not be used with Shared Line Recording.*

Software and Hardware Requirements

SYSTEM	SOFTWARE REQUIREMENTS

Cisco system	• Station-side RTP Packet Capture
	  ○ Cisco UCM v10 or later
	• Subscription-based SIP Audio Stream
	  ○ Cisco UCM v10 or later

Encore system	• All audio collection methods
	  ○ The 64-bit TSP Client is required.
	  ○ The Cisco TSP Server software must be installed on the Encore server
	  ○ Encore communicates with Cisco over TCP/IP port 5060 and UDP ports 40000 – 49000. Verify the network configuration or firewall is not blocking these ports.

SYSTEM	HARDWARE REQUIREMENTS

Cisco system	• Station-side RTP Packet Capture
	  ○ Phones must be configured to use one of these codecs: G.711a, G.711u, and G.729
	  ○ Span port on network to route all RTP traffic for recorded stations

	to Encore server ○ DHCP IP address reservation or static IP assignment for each station to be recorded • Subscription-based SIP Audio Stream ○ Phones must support the BIB (Built-In Bridge) ○ Cisco publishes a list of phones that support the BIB on their website. Reference the Device-based (built-in-bridge) RTP-Unencrypted Media column in this link: https://developer.cisco.com/site/collaboration/call-control/uc-manager-sip/faq/supported/index.gsp. Consult your Cisco expert for the latest list. ○ You may also generate a report on your CUCM system that provides a list of supported devices for your CUCM version. See "Step 1: List of Devices" on page 8 for steps to generate this report. ○ Phones must be configured to use one of these codecs: G.711a, G.711u, and G.729
Encore system	• No special hardware is required

Certification

As of February 2015, Encore v6.0.1 is certified to operate with Cisco UCM 10.5.1.10000-7.

Documentation Overview

This document provides integration information for a specific phone system. It helps a user to understand the features and benefits of the integration as well as what needs to be configured on the phone system. Conventions used in this guide include:

1. Computer commands needed to complete a task appear like this: **Sample** (in black)
2. Keyboard strokes that need to be entered appear like this: [Sample]

The screenshot examples in this guide were taken from Cisco UCM v7.1 and v8.5, and may vary from other versions of the Cisco UCM software.

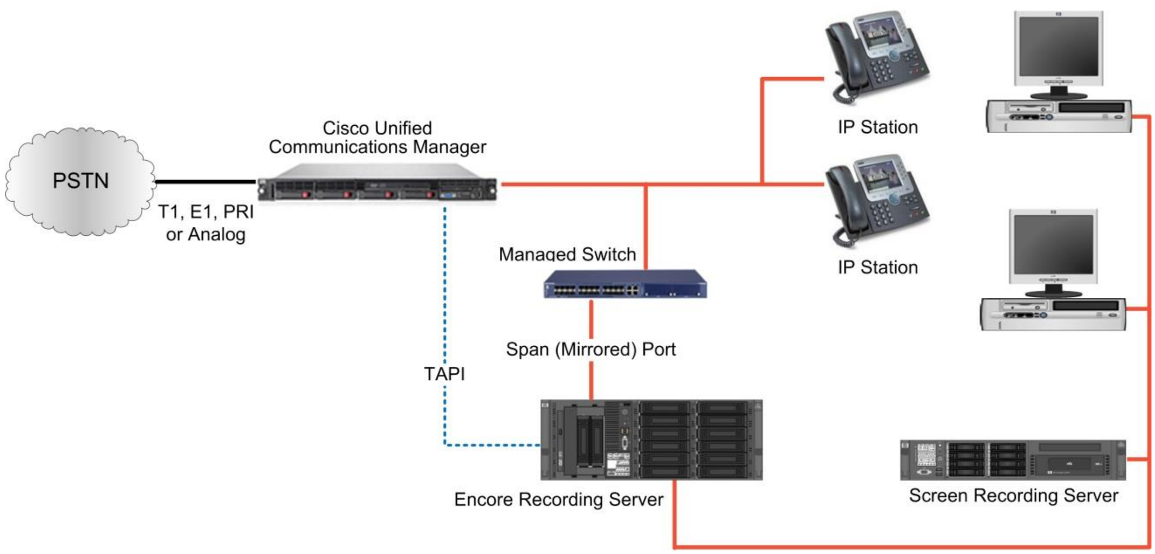
Overview

This section provides an overview of each audio collection method. For simplicity sake, the diagrams only display a single Encore server but there can be multiple Encore servers depending on the number of stations to be recorded.

Station-side RTP Packet Capture

If using Cisco UCM v3 up to v6.1, use the Station-side RTP Packet Capture method to record calls. This audio collection method also works with systems using a version of UCM that is later than v6.1.

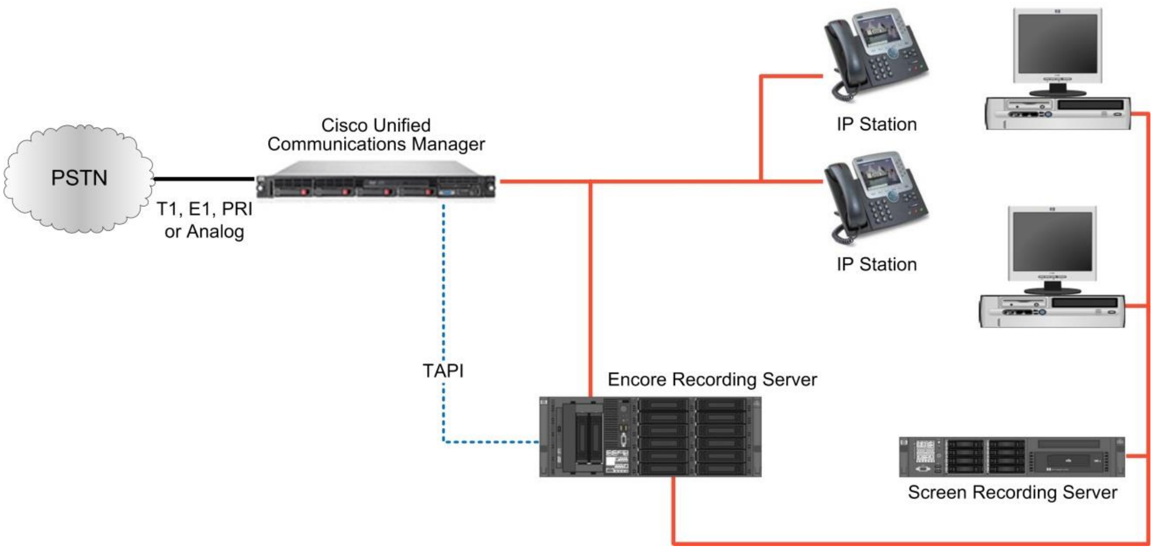
This method uses a span port to collect the RTP audio packets directly from the network segment that includes the VoIP traffic. Based on the TAPI messages received from the Cisco UCM and sent to the Cisco TSP (installed on the Encore server), the Encore server collects the RTP packets for a specific IP or MAC address. It then converts the RTP data to an audio recording file and collects data associated with the call from the TAPI messages.



Subscription-based SIP Audio Stream

This audio collection method works with systems using Cisco UCM v6.1 or later. It uses the events received from the Cisco TSP Server installed on the Encore server to start and stop recording. Encore collects data associated with the call from the TAPI messages.

NOTE Busy Hour Call Completions (BHCC) is the number of expected calls completed during the Busy Hour. For sizing CUCM and traffic analysis, each recording session adds two calls to BHCC. (*Source: "Silent Monitoring & Call Recording", SIPTrunkMonitoringRecordingOverview.ppt, Cisco Systems, Inc.*)



In order to obtain the audio from the phone stations, Encore uses the Cisco Call Recording feature configured in the Automatic Recording mode. After an agent call becomes active, two server calls are made to the Built-in Bridge (BIB) of the agent phone. The agent phone automatically answers. Two server calls are then redirected to the Encore server to provide audio using SIP protocol. This is accomplished by assigning a directory number to the Encore server and configuring a route pattern for the SIP trunk.

Configure Cisco System

The steps to configure the Cisco system are based on Cisco UCM v8.5 and the screen shots of the Cisco UCM are based on Cisco UCM v7.1 and v8.5. Your screens may be different. It is assumed that the reader has a working knowledge of Cisco UCM and only needs specific configuration assistance.

Only complete the steps in this section if using the Subscription-based SIP Audio Stream method to record calls.

Step 1: List of Devices

Agent devices must be able to fork media for recording. The list of devices that support the recording feature varies per version and device pack.

Complete the following steps to generate a complete list of devices that support recording for a particular release and device pack.

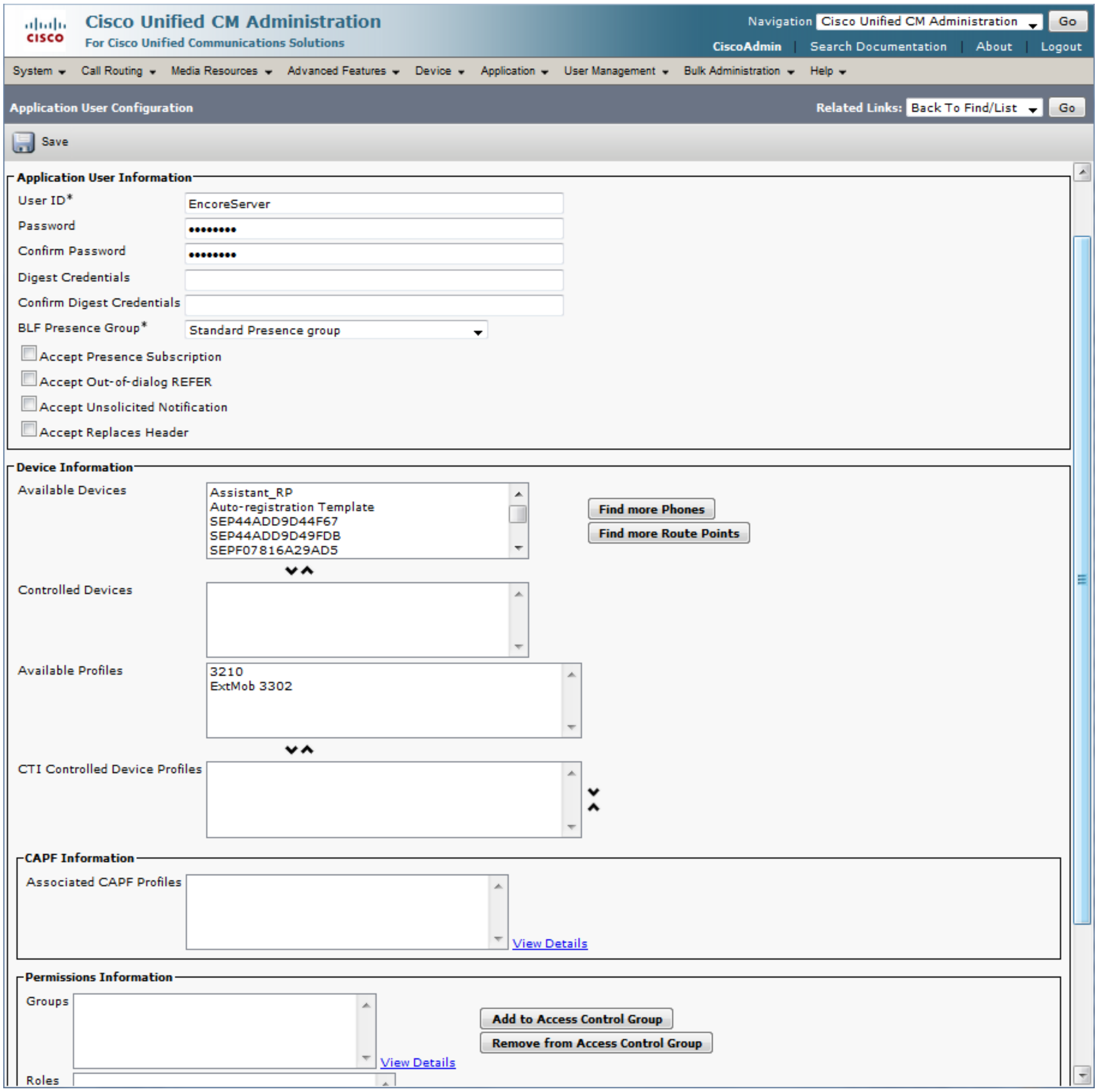
1. Start Cisco Unified Reporting by using any of the methods that follow. The system uses the Cisco Tomcat service to authenticate users before allowing access to the web application. You can access the application by:
2. Choosing Cisco Unified Reporting in the **Navigation** menu in **Cisco Unified Communications Manager Administration** and clicking **Go**.
3. Choosing **File | Cisco Unified Reporting** at the **Cisco Unified Real Time Monitoring Tool (RTMT)** menu.
4. Entering **https://<*server name or IP address*>:8443/cucreports/** and then entering your authorized user name and password.
5. Click **System Reports** in the navigation bar.
6. In the list of reports that displays in the left column, click the **Unified CM Phone Feature List** option.
7. Click the **Generate a new report** link to generate a new report, or click the **Unified CM Phone Feature List** link if a report already exists.
8. To generate a report of all devices that support recording, choose these settings from the respective drop-down list boxes and click the **Submit** button:
    - **Product: All**
    - **Feature: Record**

    The **List Features** pane displays a list of all devices that support the recording feature. Click the up and down arrows next to the column headers (**Product** or **Protocol**) to sort the list.

Step 2: Configure Application User for CT Gateway

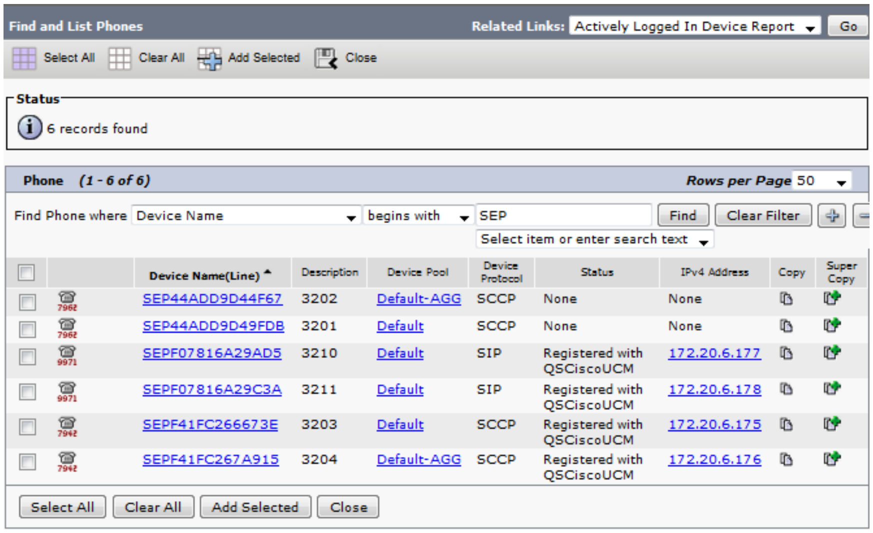
TAPI security requires an application user to be configured on the switch for the Encore server.

1. Log in as a CUCM administrator on the switch.
2. Move the mouse pointer over the **User Management** option and select **Application User** from the drop-down menu.
3. Click the **Add New** button at the bottom of the window to add a new user.
4. Enter the user identifier (**EncoreServer** is recommended) and a password.



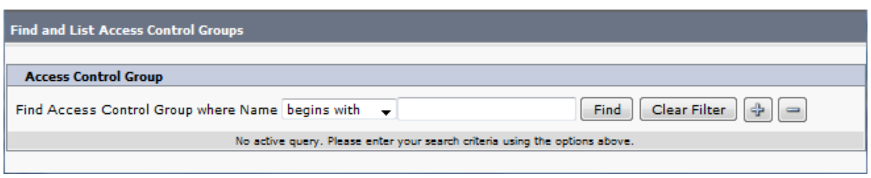
5. From the **Available Devices** list, select all the phone devices that need to be recorded. Click the down-arrow button to add the devices to the **Controlled Devices** list. If you do not know a

phone's device name, use the **Find more Phones** button to locate it. Select the devices you would like to add and click the **Add Selected** button at the top of the window to accept them.



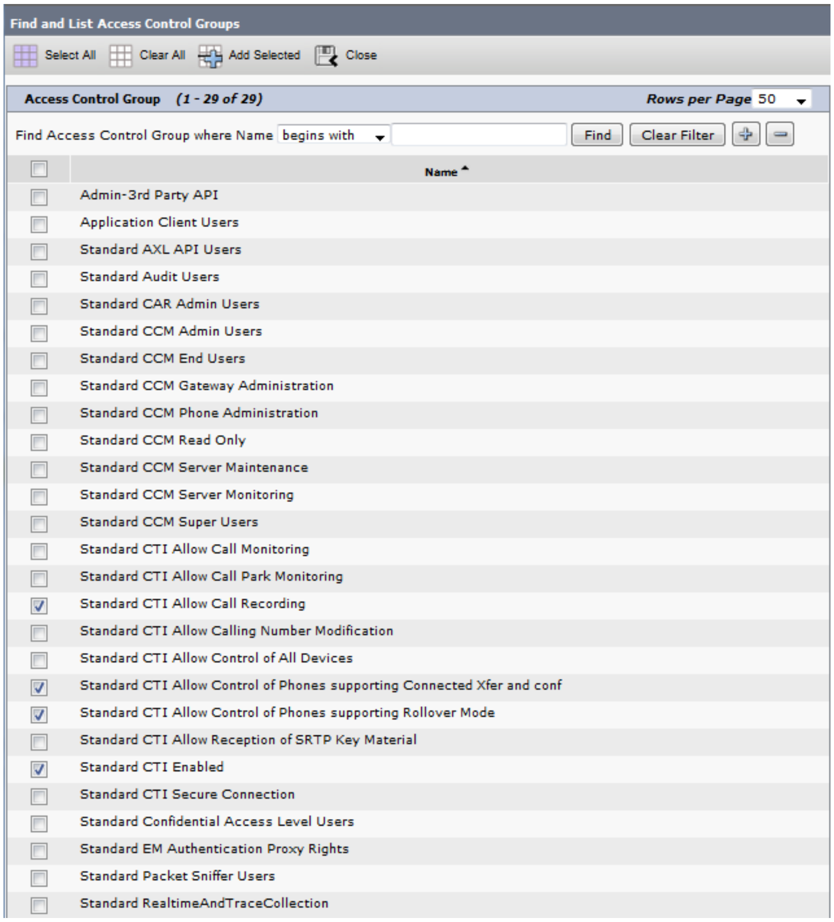
WARNING Only add phone devices that need to be recorded to the **Controlled Devices** list. CT Gateway uses this list to determine which devices to record.

6. After all phones have been added to the **Controlled Devices** list, scroll down to the **Permissions Information** section and click the **Add to Access Control Group** button on the **Application User Configuration** window shown on page9. The following window opens; click **Find**.



7. Select the following **Groups** and click the **Add Selected** button at the top of the window:

   - **Standard CTI Allow Call Recording**
   - **Standard CTI Enabled**
   - **Standard CTI Allow Control of Phones supporting Connected Xfer and Conf**
   - **Standard CTI Allow Control of Phones supporting Rollover Mode**



8. Click **Save**.

Step 3: Configure Cisco UCM to Send Audio

This section configures the CUCM to send audio to the Encore server:

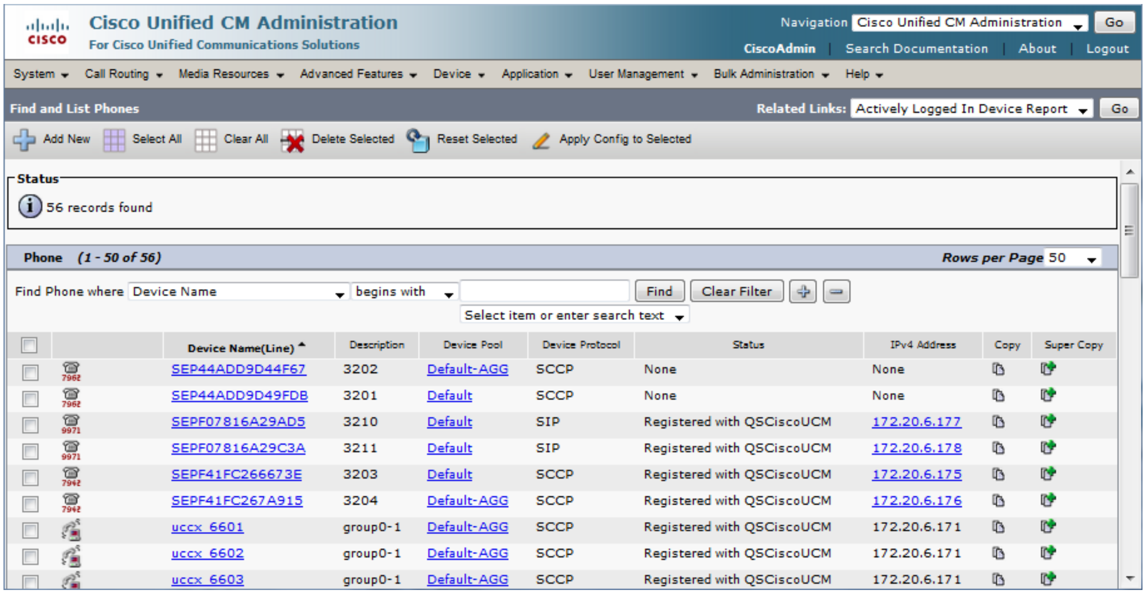
- Turn on the IP phone BIB to allow recording
- Configure tones for recording
- Create a SIP trunk that points to the Encore server
- Create a Route Pattern Definition for the Encore server
- Create a recording profile
- Enable recording for a line appearance
- Limit codec usage for recording calls

Complete instructions for configuring the Call Recording feature for CUCM are detailed in Cisco documentation. The information pertinent to Encore has been extracted for this guide.

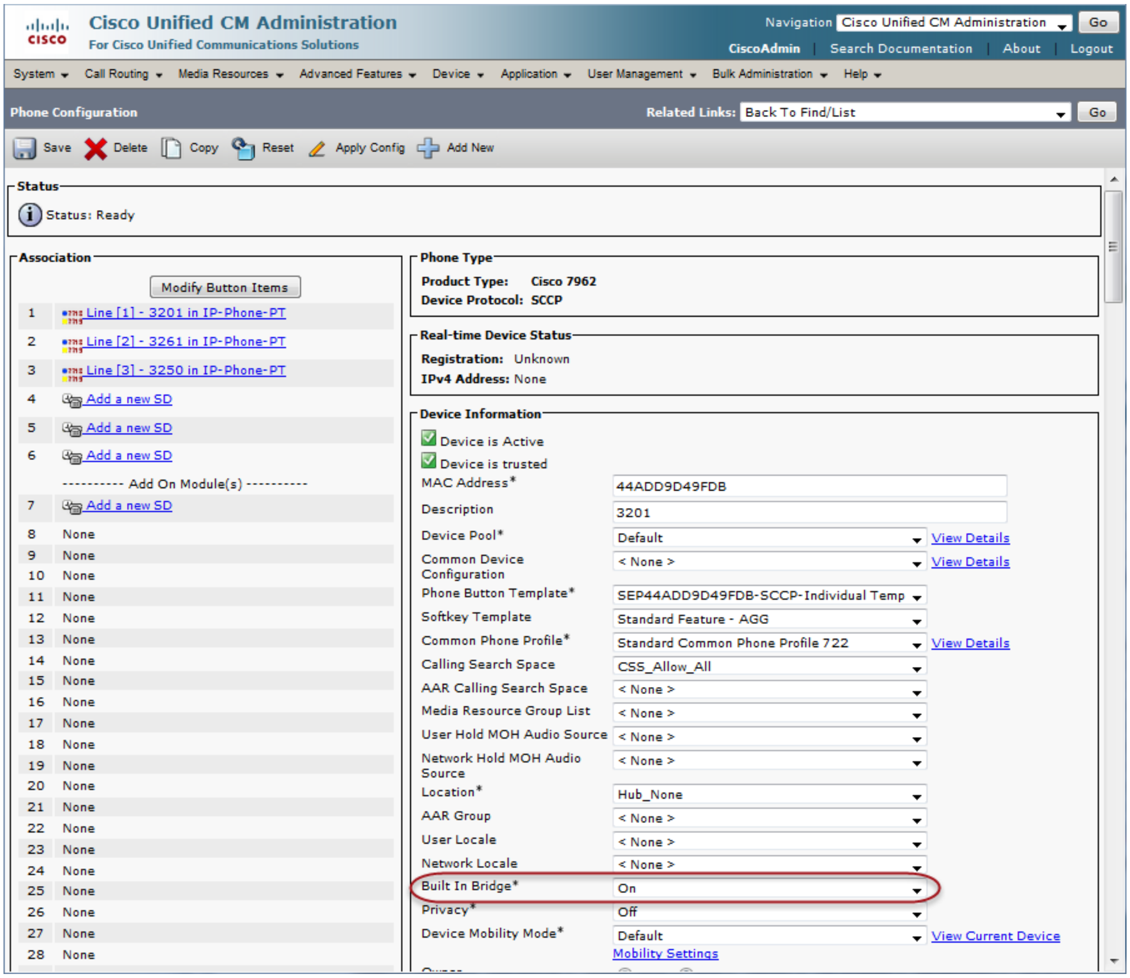
Turn on the IP phone BIB to allow recording

This step enables the Built-in Bridge (BIB) feature of the agent's phone so that its calls can be recorded.

1. Log in as a CUCM administrator on the switch.
2. Select the **Device | Phone** menu option in the Cisco Unified CM Administration.
3. Click the **Find** button.
4. In the **Device Name (Line)** column, click the link for the device you need to record.



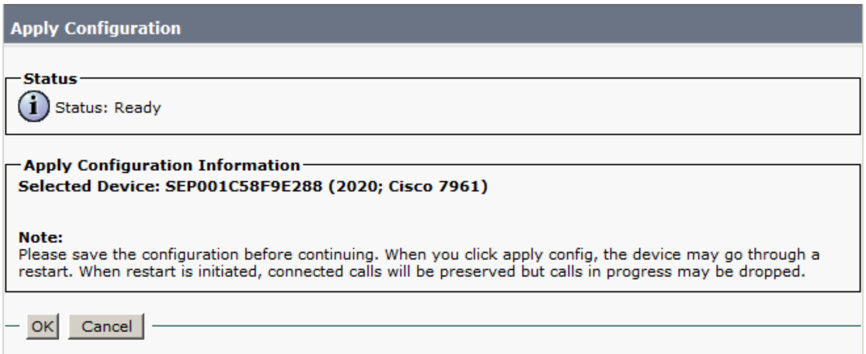
The **Phone Configuration** window (shown below) opens. In the **Device Information** section, scroll down to the **Built-in Bridge** option. Click the drop-down arrow and select **On**.



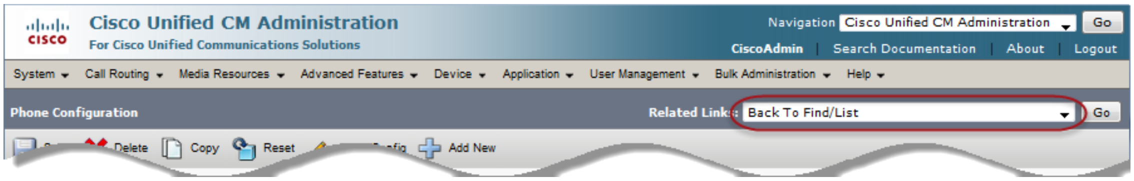
NOTE	You can also set the **Built-in Bridge Enable** service parameter to **On** and leave the **Built-in Bridge** option in the **Phone Configuration** window set to **Default**.

5. Click **Save**. A message opens reminding you to click **Apply Config** to have the changes take effect. Click **OK**.
6. Click **Apply Config** to have the changes take effect on the device now.

7. If the following window appears, click **OK** to continue. The phone reloads its configuration. If a call is connected the call will be preserved, but any calls that are in progress may be dropped.

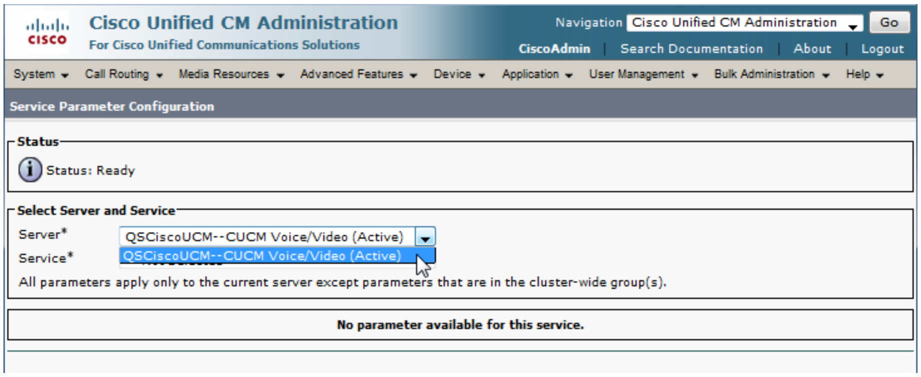


8. To select another device that needs these changes, locate the **Related Links** box in the upper right corner and click **Back to Find/List**. Click **Go**. Repeat Steps 4-7 for each individual phone device you need to record.

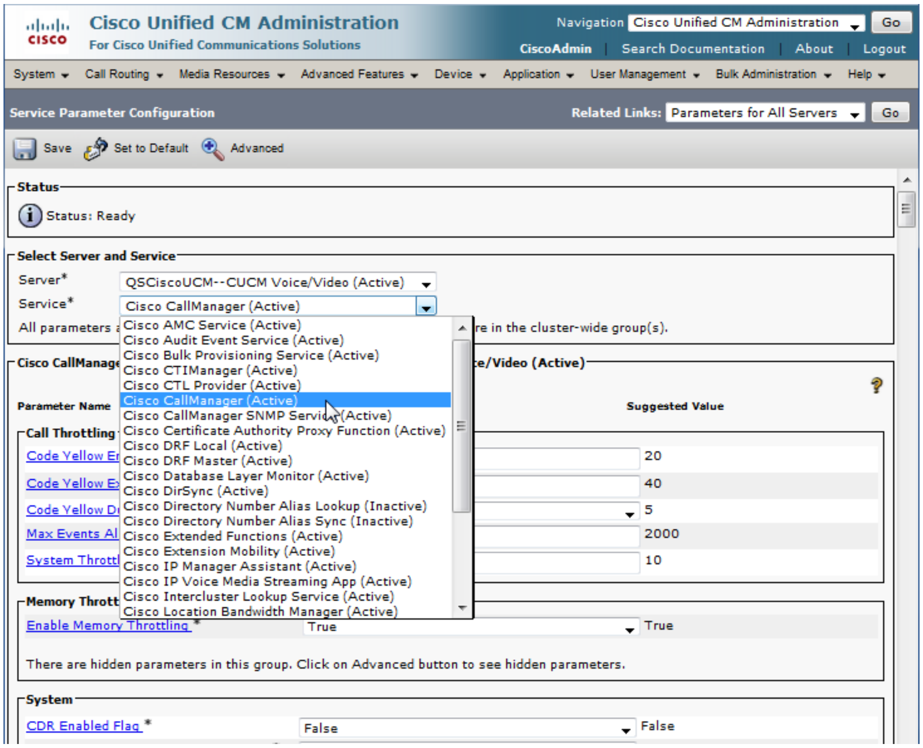


Configure tones for recording

1. If a recording warning tone is required, select the **System | Service Parameters** menu option in the Cisco Unified CM Administration.
2. Select the server name or IP address from the **Server** drop-down menu.



3. Select the **Cisco Call Manager (Active)** from the **Service** drop-down menu.



4. Scroll just over half way down to the **Clusterwide Parameters (Feature - Call Recording)** and set the following parameters to **True** to allow the Recording Notification tone to be played to the agent only, to the customer only, or to both.

   - **Play Recording Notification Tone to Observed Target** – When True, tone plays to agent
   - **Play Recording Notification Tone to Observed Connected Parties** – When True, tone plays to customer

> **NOTE** These are global settings and apply to all recorded phones.

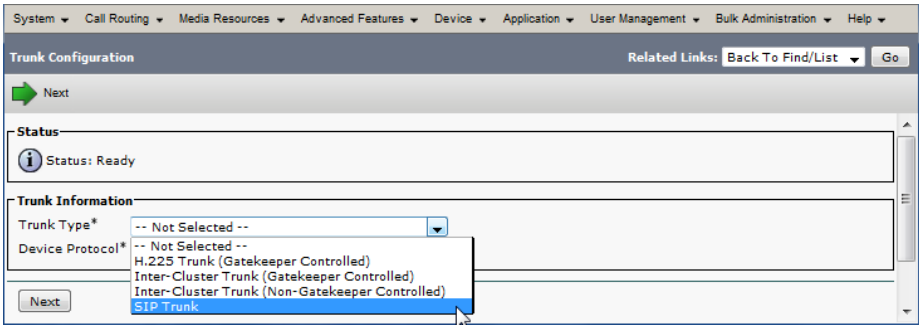
5. Click **Save.**

Create a SIP trunk that points to the Encore server

To create a SIP trunk that points to the Encore server, enter a Directory Number (DN) for the Encore server. The DN must match a route pattern for the SIP trunk or a route list that includes the Encore server.

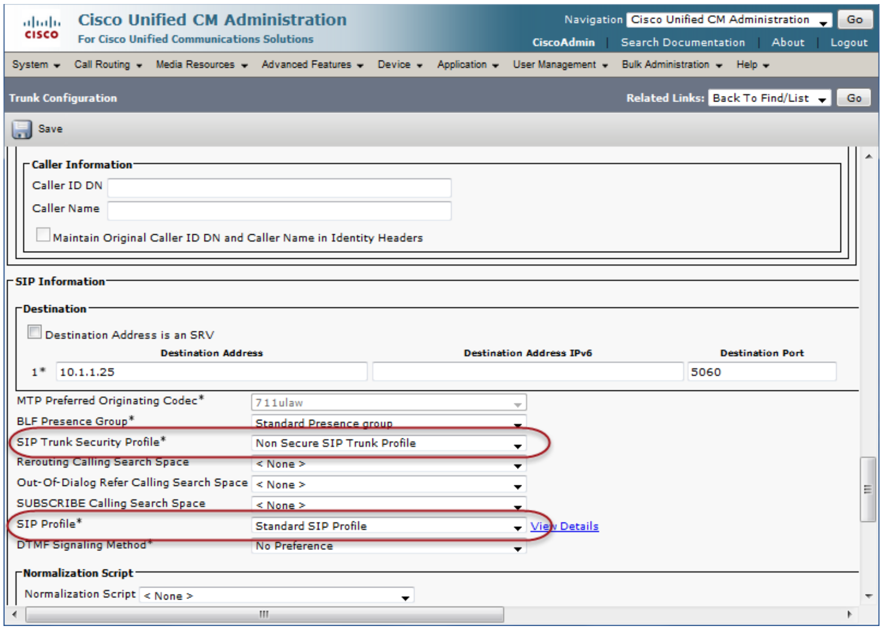
1. Select the **Device | Trunk** menu option in Cisco Unified CM Administration.
2. Click **Add New**.

3. In the **Trunk Type** drop-down menu, select **SIP Trunk**.



4. The **Device Protocol** auto-populates with **SIP**. Accept the default **Trunk Service Type**.
5. Click **Next**.
6. In the **Device Information** section, enter **Encore** in the **Device Name**.
7. In the **Device Information** section, verify the **Device Pool** has a value selected.
8. Scroll down to the **SIP Information** section and set the **Destination Address** to the IP address of the Encore server's NIC that collects audio. Accept the default **Destination Port** of **5060**.
9. Check to verify that the **SIP Trunk Security Profile** and the **SIP Profile** have values assigned in the **SIP Information** section of the window.

> **NOTE** Ensure the chosen **SIP Trunk Security Profile** is using an **Outgoing Transport Type** of **TCP**.
>
> Ensure the **SIP Profile** has the **Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"** unchecked in the **SIP OPTIONS Ping** section of the **SIP Profile Configuration**.

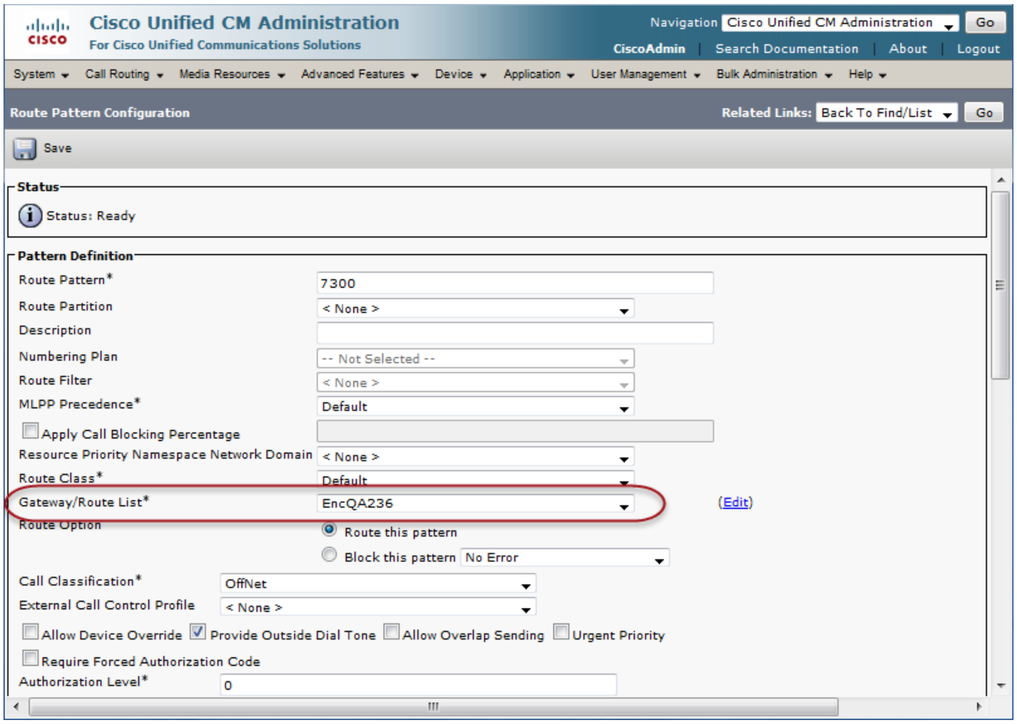


10. Click **Save**.
11. Click **OK** if the following prompt opens, "The configuration changes will not take effect on the trunk until a reset is performed. Use the Reset button or the Job Scheduler to execute the reset." Then click the **Reset** button at the top of the window if you would like to reset the trunk device at this time.
12. When prompted to **Reset**, **Restart** or **Close**, choose **Reset**. After the status changes to indicate that the reset request was sent successfully, click **Close**.

Create a Route Pattern Definition for the Encore server

Create a Route Pattern Definition for the Encore server's SIP trunk. The Route Pattern number must match the **Recording Destination Address** in the Recording Profile you will create in "Create a recording profile" on page 18.

1. Select the **Call Routing | Route/Hunt | Route Pattern** menu option in the Cisco Unified CM Administration.
2. Click **Add New**. In the **Route Pattern** field, enter the route pattern number you wish to create.
3. In the **Gateway/Route List** field, select the SIP trunk that points to the Encore server, or select a route list in which the server is a member.



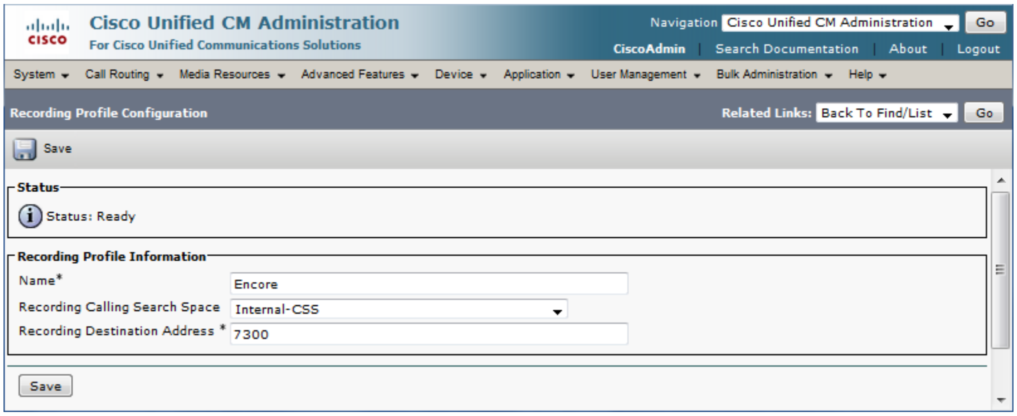
4. Click **Save**.

5. If the following message opens, click **OK** to proceed. No authorization code is needed for this route pattern.

6. Click **OK** if this message opens, “Any update to this Route Pattern automatically resets the associated gateway or Route List.”

Create a recording profile

1. Create a recording profile by selecting **Device | Device Settings | Recording Profile** from the menu bar.
2. Select the **Add New** button.
3. Enter **Encore** in the **Name** field.
4. On the **Recording Calling Search Space** drop-down menu, select the **Calling Search Space** that is associated with the devices you intend to record.
5. Enter **the previously created** Route Pattern Definition in the **Recording Destination Address**.



6. Click **Save**.

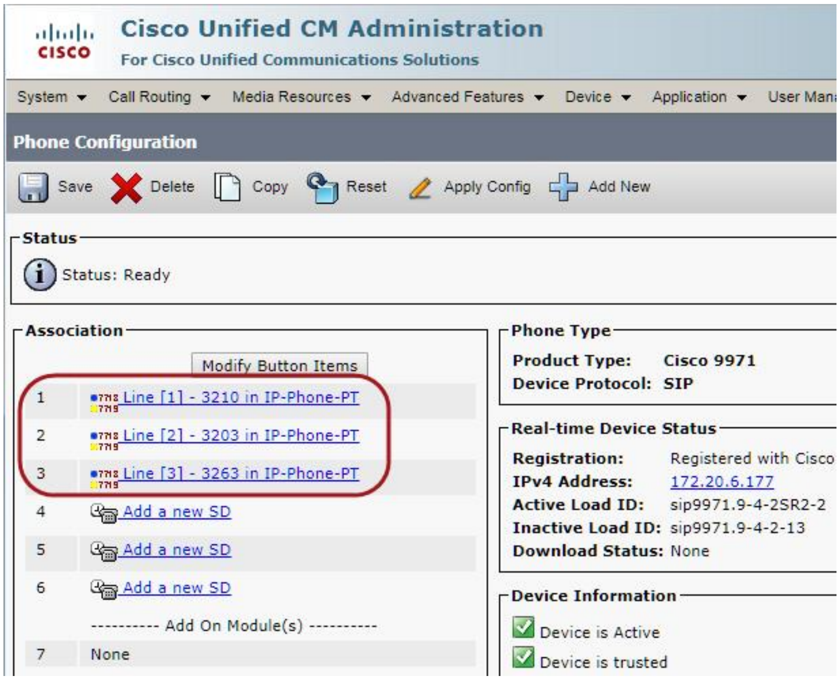
Enable recording for a line appearance

To enable recording of an agent, set the **Recording Option** in the line appearance of the agent to **Automatic Call Recording Enabled**.

For non-extension mobility line appearance:

1. Select the **Device | Phone** menu option.
2. In the **Device Name (Line)** column, click the link for the device name you need to change.

3. In the **Association** section, click each line appearance of the device that requires recording.

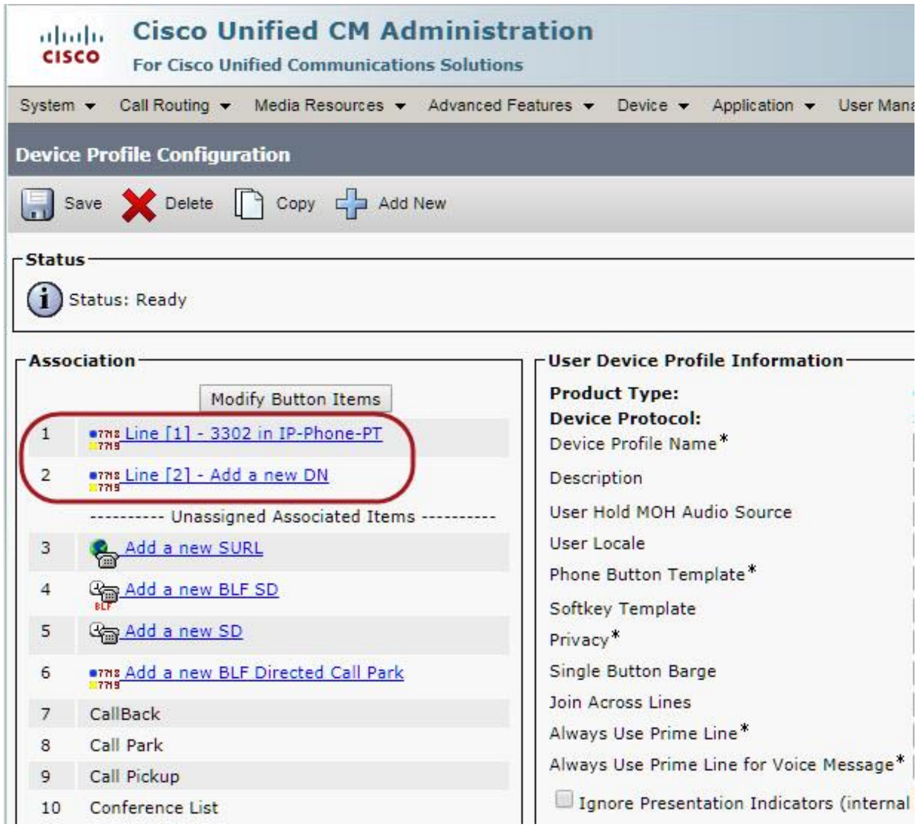


4. On the **Phone Configuration** window that opens, scroll down to the **Association** section and locate the Line/Device you wish to change. Set the **Recording Option** to **Automatic Call Recording Enabled** and set the **Recording Profile** to **Encore**.
5. Click **Save**.
6. If you need to select another line appearance to configure, click the **Go** button located to the right of the **Related Links** drop-down menu.
7. Once you have made changes to all line appearances, click **Apply Config** and then click **OK** on the **Apply Configuration** window to restart the device.

For extension mobility line appearance:

1. Select the **Device | Device Settings | Device Profile** menu option.
2. Click the **Find** button.
3. Select a **Device**.

4. Click each line appearance of the device that requires recording.



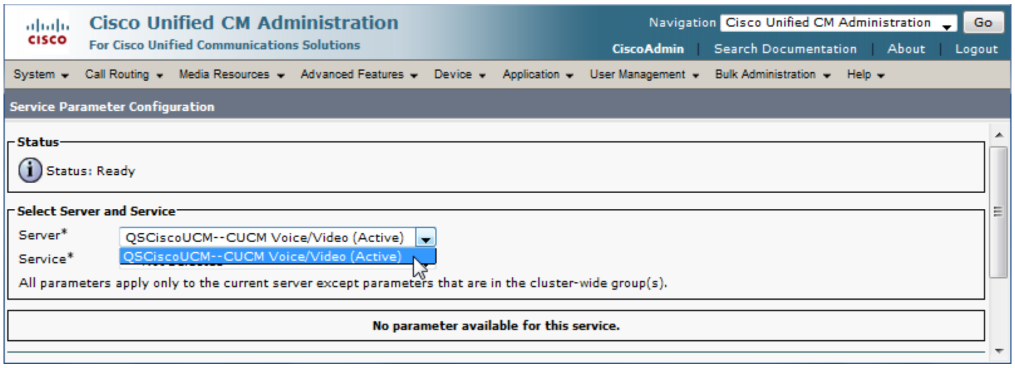
5. On the **Device Profile Configuration** window that opens, scroll down to the **Association** section and locate the Line/Device you wish to change. Set the **Recording Option** to **Automatic Call Recording Enabled** and set the **Recording Profile** to **Encore** from the drop-down menu.

6. Click **Save**.

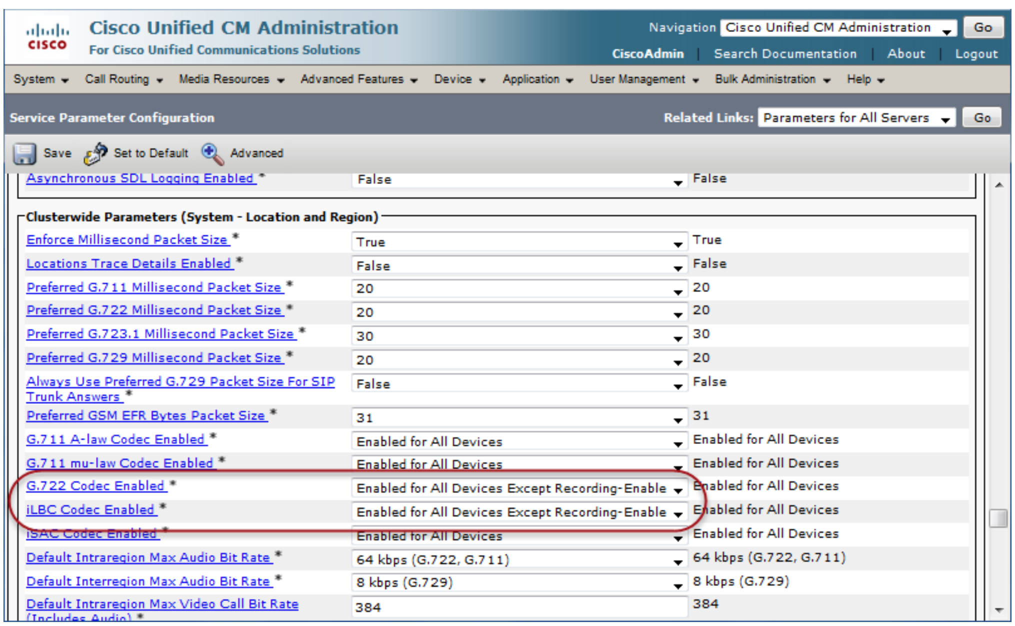
Limit codec usage for recording calls

By default, newer Cisco phones use the G.722 codec. For recording purposes, this codec is not supported. Complete the following steps to disable usage of the G.722 and iLBC codecs:

1. Select the **System | Service Parameter** menu option.
2. Select the Server Name or IP Address from the **Server** drop-down menu.



3. Select **Cisco CallManager (Active)** in the **Service** drop-down menu.
4. Scroll down to **Clusterwide Parameters (System – Location and Region)** section and change the **G722 Codec Enabled** and **iLBC Codec Enabled** values to **Enabled for All devices Except Recording-Enabled Devices**.

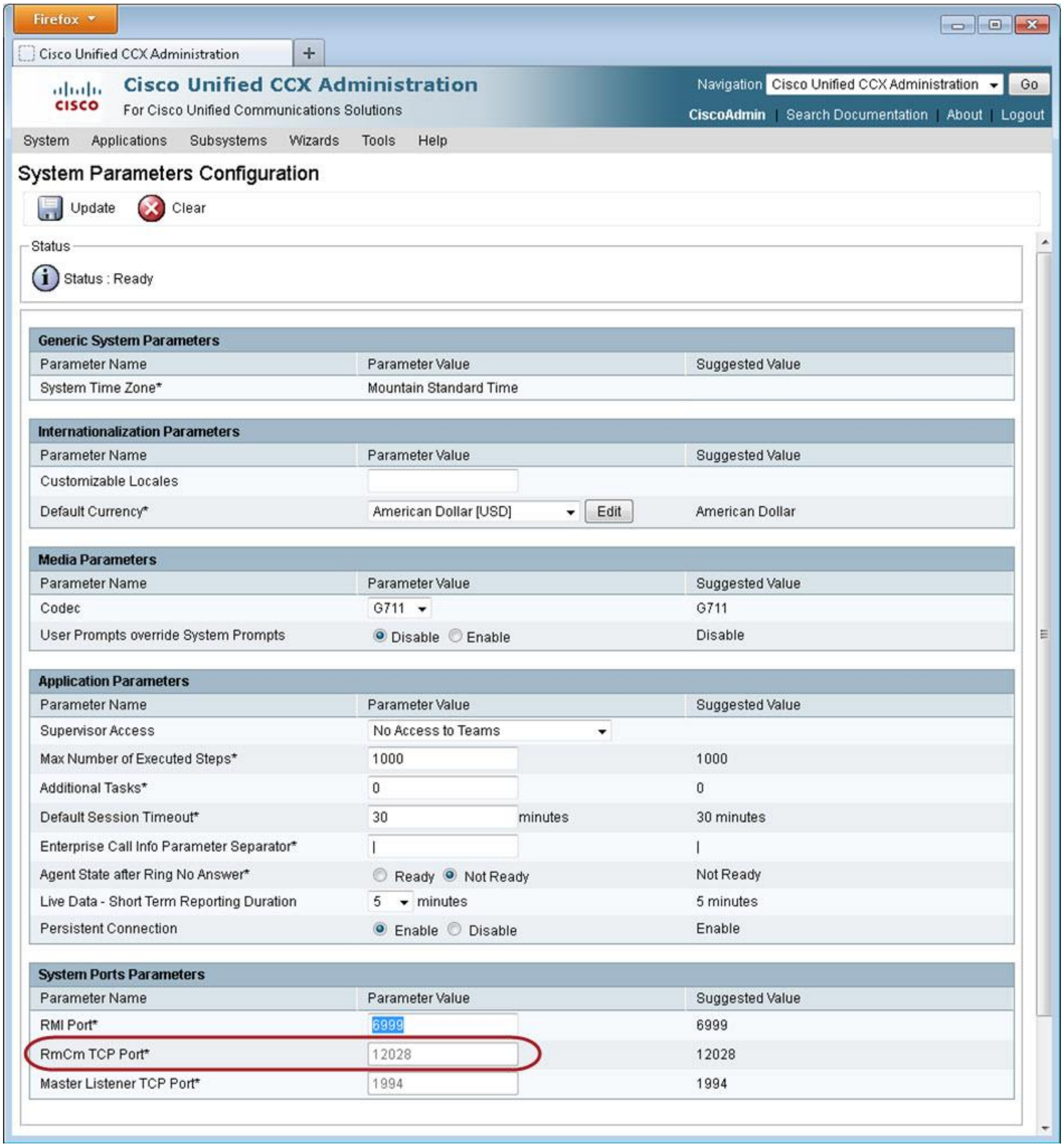


5. Click **Save**.

All phone devices that need to be recorded must be configured to use the G.711a, G.711u, or G.729 codec in the device's profile; see the Cisco documentation or the switch technician for detailed steps.

Step 4: Obtain UCCX Port Number

1. Open the Cisco Unified CCX Administration application.
2. Browse to the **System Parameters Configuration** page.
3. In the **System Ports Parameters** section, find the **RmCm TCP Port** field and write down the value.



Step 5: Configure Finesse – Optional

This section configures Finesse to send information to the Encore server:

- Create an Encore Supervisor Account
- Download Finesse Self-signed SSL Certificate

Create Finesse Supervisor Accounts

One or more Finesse supervisor accounts must be created that Encore uses to get information for the agents logged into finesse. Each Finesse supervisor account can monitor a maximum of twenty (20) teams. If you have more than twenty teams with agents that need to be recorded, you will need to create enough supervisor accounts to cover all necessary teams. Encore supports up to ten (10) supervisor accounts for a total of 200 teams. If you need to monitor more than 200 teams, contact your DVS technician.

Each supervisor account needs the following configuration:

- The account must only be used by Encore.
- The account must have Single Sign-on (SSO) disabled.
- Teams to be monitored by Encore for agent recording must be in one of the account's **Supervised Teams** list.

Provide the DVS Installation Technician with the following information:

- The fully-qualified domain name (FQDN) of the Cisco Finesse server.
- The name and password of the supervisor account created for Encore.

Download Finesse Self-signed SSL Certificate

If the Cisco Finesse Tomcat software uses a self-signed SSL certificate, download the certificate:

1. Sign in to the **Cisco Unified Operating System Administration** using the URL: https://FQDN:8443/cmplatform where FQDN is the fully-qualified domain name of the primary Finesse server.
2. Click the **Security** menu option and then click **Certificate Management**.
3. In the **Find Certificate List where** query, enter 'Certificate', 'is exactly', and 'tomcat' and click **Find**.
4. Click the FQDN link in the **Common Name** field of the tomcat certificate record.
5. Note the expiration date of the certificate and set a reminder to generate a new certificate before the expiration date and provide the new certificate to DVS support.

   CAUTION If the Finesse certificate expires, the Encore system will not receive Finesse data until a new certificate is applied.

6. In the **Certificate Details** window that appears, click the **Download .DER File** button to download the file.
7. Provide the DVS Installation Technician with the DER file.

Call Handling Scenarios

This section explains how different calls are displayed in Encore.

- **Hold** – When a call is put on hold, the recording is suspended. When the call is retrieved, the audio is appended to the recording to create one audio recording.
- **Consultation Call** – If an agent is on a call and then places a consultation call, the first call is put on hold and the recording is suspended. Assuming the called party is also using a recorded phone, the consultation call is recorded as two separate recordings – one for each extension. When the agent hangs up the consultation call and retrieves the caller, the two recordings end and the first recording resumes; the second portion of the recording is appended to the first portion. All three recordings have different Segment IDs (SIDs) and share the same Relation ID (RID).
- **Blind Transfer** – When a call is blind transferred (also called an unannounced transfer), the first recording ends after the agent presses the transfer button and hangs up the handset. The second recording begins when the second agent answers the transferred call. The second recording ends when the second agent hangs up the call. Separate SIDs are associated with each recording and they usually share the same RID. If the call is transferred to an ACD queue or Hunt Group, it may not be possible to show the relationship between the recordings and the same RID may not be associated with both recordings.
- **Supervised Transfer** – For a supervised transfer, the first recording is suspended when the agent puts the caller on hold to initiate the consultation call. The agent makes the call to the supervisor and the supervisor answers. Two new recordings start, one for the agent and one for the supervisor. The agent's call data will show this as a consultation call. When the agent completes the transfer, the agent's first and second recordings end. The recording for the supervisor continues, but will now reflect call data for the transferred party and will end when either party hangs up. All three recordings have different SIDs and share the same RID.
- **Conference Call** – When an agent decides to bring a third party into a current call, the agent usually puts the caller on hold to first consult with the third party. The first recording of the agent and the outside caller suspends during the consultation call. Assuming the third party is using a recorded phone, the consultation call creates two new recordings – one for the agent and another for the third party. After the consultation call ends, the agent's second recording stops and the three parties are joined into the conference, the first recording resumes and it ends when the agent hangs up. The recording of the third party continues until the third party hangs up. All three recordings have different SIDs and share the same RID.
- **Internal Call** – If both extensions are monitored by Encore, two recordings are created – one for each extension. Separate SIDs are associated with each recording but they share the same RID.

Appendix 1: Single Number Reach Recording

To record Cisco Single Number Reach or Mobile Jabber Client requires a second server dedicated to recording these calls. The Cisco UCM version must be 10.x or higher and there must be a compatible CUBE device or Voice Gateway.

Configuration Steps:

1. Create a new SIP trunk that points to the secondary Encore server's LAN address. Give it a name that's reflective of the secondary Encore server.
2. Create a new **Route Pattern Definition** using the newly created SIP trunk.
3. Create a new **Recording Profile** that points to the newly created **Route Pattern Definition**. Give it a name that's reflective of the secondary Encore server.
4. For **Single Number Reach** (**SNR**), open the **Remote Destination Profile** (**Device | Device Settings | Remote Destination Profile**) and modify the needed records. In each record, in the **Association** section and within the line appearance for the associated extension, scroll down to the **Line <#>** on **Device <*name*>** section and make sure the following is set:
    a. **Recording Profile** = **Recording Profile** created in Step 3 above. (This is what points to the secondary Encore server.)
    b. **Recording Option** = **Automatic Call Recording**
    c. **Recording Media Source** = **Gateway Preferred**
5. For mobile Jabber clients, the steps are like Step 4, except these are considered phone devices and would be listed as type **Cisco Dual Mode for IPhone** or **Cisco Dual Mode for Android**.

Appendix 2: Glossary

ACD Name

The CSQ Name.

ACD Number

The CSQ ID.

ANI

Automatic Number Identification. For inbound calls, this is the phone number from which the customer is calling (may not be supported by the trunk). For outbound calls, this is the dialed number.

call direction

The direction is either incoming (inbound) or outgoing (outbound).

call origin

One of the following origins is provided by the TAPI stream:

- Conference – The call originated from a conference call.
- External Incoming – The call originated as an incoming call on an external line.
- Internal Incoming – The call originated as an incoming call at a station internal to the same switching environment. If using the UCCX, external incoming UCCX calls will always show as Internal Incoming.
- Outgoing – The call originated from this station as an outgoing call.
- Unavailable – The call origin is not available and will never become known for this call.
- Unknown – The call origin is currently unknown but may become known later.

call record

An entry in a database that holds the data associated with a call.

called ID

The ID of the station that receives/answers a call.

caller ID

The ID of the station that initiates/places a call.

connected identifier

Identifier for the party to which the call was actually connected. May be different from the called party if the call was diverted.

consultation call

A call that is made while the customer (original call) is on hold. In the database, the Consultation Call field shows **Yes** when the recording is a consultation call.

device name

Phone's name usually contains the phone's Media Access Control (MAC) address. This field allows Encore to identify the physical device where the call is recorded.

dialed number

Captured for outbound calls and is stored in the ANI field.

digital recording

A method of recording that converts analog sound into a series of pulses that are translated into binary code, which is read by computers.

DNIS

Dialed Number Identification Service. For inbound calls, this is the number the customer dialed or the agent's extension number (may not be supported by the trunk).

encrypted calls

Calls that have the audio RTP packets encrypted. This prevents 3rd party applications, such as the Encore system, from using the RTP packets for recording.

extension

The number associated with a person's station. For the Cisco integration, the extension may be used as an agent identifier. Extension and station are sometimes used interchangeably.

Extension Mobility

Cisco Extension Mobility allows a user to log into and out of a phone. It loads a user Device Profile (including line, speed dial numbers, and so on defined for the user) onto the phone when the user logs in.

external calls

In these calls, the calling or called parties are outside the PBX.

Fin Call Type

The Finesse call type of the recorded call.

Fin Login Name

The Finesse login name of the recorded agent.

Fin Team Name

The Finesse team name of the recorded agent.

global call identifier

Allows a call to be tracked as it moves through the call center (hold, transfer, and conference).

hold duration capture

The sum of all hold durations that occurred during the recording.

inbound

Calls which are received/answered by a recorded party.

internal calls

Calls made between extensions on the same PBX.

media encrypted calls

Calls that have the audio RTP packets encrypted. This prevents 3rd party applications, such as the Encore system, from using the RTP packets for recording.

other call ID

A unique call ID assigned by the PBX, showing the relationship to other call ID's assigned by the PBX. For example, this identifier allows the Encore server to relate the "customer-agent" call to the "agent-supervisor" consultation call.

other party name

Name of the other party on the line with the person being recorded; may be blank if this is an external call.

other party number

Number of the other party on the line with the person being recorded.

outbound

Calls which are initialed/placed by a recorded party.

pause/resume on hold

A method that pauses the recording of audio and screen when a call is placed on hold, and resumes recording when the hold is taken off.

recorded party number

Extension of the recorded station.

Recording Notification Tone
This notification informs the customer that he is being recorded by playing a certain tone periodically during the conversation. The tone may play for the agent only or for both the agent and the customer. Recording phone calls in certain jurisdictions may require caller notification. If you are unsure of whether this requirement pertains to your situation, contact the local public utilities commission.

redirecting identifier

Identifies the address that redirected the call; for a transfer call this is the transferring party.

redirection identifier

Identifies the address to which the call was redirected; for a transfer call this is the destination of the transfer.

station

A phone connected to the PBX.

TAPI

Telephone Application Programming Interface. A telephony software interface included in Microsoft Windows operating system that supports the incorporation of telephony control by other applications.

trunk

The connection between the phone company and the PBX that carries incoming calls.

TSP Server

Cisco's TAPI Service Provider Server. Used to control recording start/stop commands in Encore for the Cisco integration.

UCCX

Cisco Unified Contact Center Express is an IP-based Automated Call Distribution (ACD) system.

UCM

Unified Communications Manager. Cisco call-processing system; previously called CallManager.