NICE | UPTIVITY
Agile WFO for SMB

Customer Guide to Cisco TAPI-BiB Integrations

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Introduction

Audience

This document is written for customers and prospective customers interested in using NICE Uptivity in a Cisco TAPI-BiB telephony environment. Readers who will perform procedures in this guide should have a basic level of familiarity with IP telephony, general networking, the Windows operating system, Cisco VoIP telephony, and NICE Uptivity.

Goals

The goal of this document is to provide knowledge, reference, and procedural information necessary to understand a proposed Cisco/ NICE Uptivity integration using TAPI-BiB, and to configure the Cisco equipment to support the integration.

This document is NOT intended as a specific system or network design document. If further clarification is needed, consult with your telephony vendor(s).

Assumptions

This document assumes the reader has access to an NICE Uptivity Sales Engineer, Project Manager, or other resource to assist in applying this information to the reader's environment.

Need-to-Knows

To facilitate ease of use, this document takes advantage of PDF bookmarks. By opening the bookmark pane, readers can easily refer to the portion(s) of the guide that are relevant to their needs. For example, the Uptivity application administrator can click on the **Customer Administration Tasks** bookmark to jump directly to that section.

To expand and collapse the bookmark pane, click on the bookmark icon on the left side of the document window.

For questions related to Uptivity configuration, consult the Uptivity installation team.

Cisco TAPI-BiB can also be used, in various combinations, with Cisco MediaSense, Cisco UCCE, or Cisco UCCX. In these scenarios, refer to the *Customer Guide to Cisco MediaSense Integrations,* the *Customer Guide to Cisco UCCE Integrations,* or the *Customer Guide to Cisco UCCX Integrations*, as appropriate.

Terminology

To ensure a common frame of reference, this guide uses the following terms:

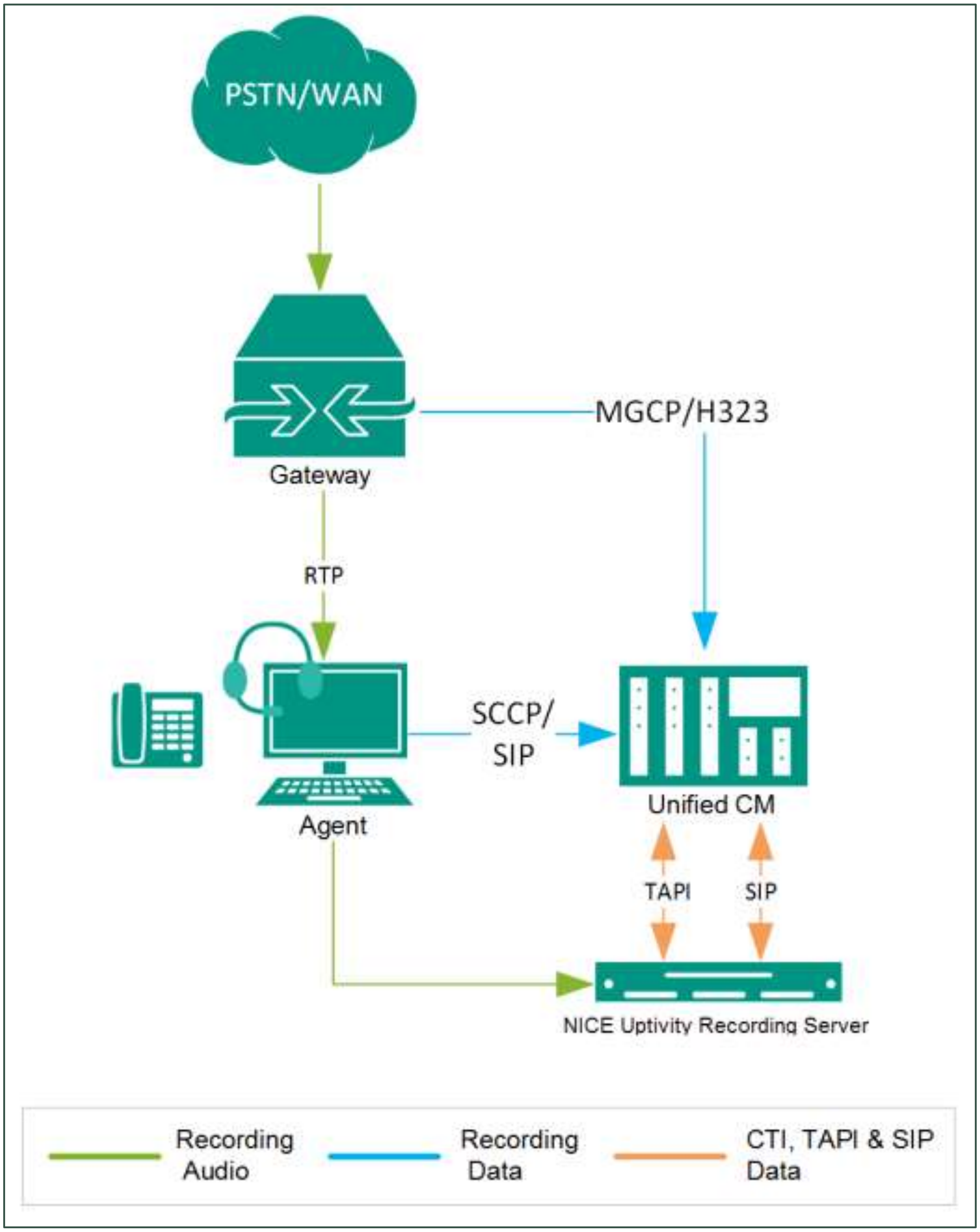
- **CUCM —** Cisco Unified Communications Manager. CUCM is a software-based call-processing system that includes gateways, routers, phones, voicemail boxes, and a variety of other VoIP components. Sometimes referred to as CallManager.
- **UCCE —** Unified Contact Center Enterprise. UCCE delivers intelligent contact routing, call treatment, network-to-desktop CTI, and multichannel contact management over an IP infrastructure. It combines multichannel ACD functionality with IP telephony in a single solution.
- **UCCX —** Unified Contact Center Express. UCCX is a single-server customer interaction management solution for up to 400 agents.
- **TAPI —** Telephony Application Programming Interface. Like JTAPI, Cisco TAPI allows custom applications to monitor and interact with the CUCM and Cisco IP phones.
- **BiB —** Built-in Bridge. Capability of some Cisco IP phone models to fork the media stream and deliver audio from both sides of a phone call to an alternate destination (for example, NICE Uptivity).
- **MediaSense —** Cisco's open-standards platform that allows for recording on the network level rather than the device level

Customer Responsibilities

You are responsible for supplying the physical connection(s), IP connection(s), or both to your telephone system and LAN, and for obtaining and loading any licensing required by Cisco. You are also responsible for configuring Cisco system components to support the recording integration. See the Customer Integration Tasks section for additional information.

Cisco TAPI-BiB Integration Overview

The Cisco TAPI-BiB integration uses the built-in bridge functionality of specific Cisco IP phones to fork the audio stream and deliver duplicate audio to NICE Uptivity, while simultaneously receiving call control events and metadata from the CUCM.



General architectural example of the Cisco TAPI-BiB integration

Component	Function
Voice Gateway	Connects the customer network to the public network.
Cisco TAPI	When a call is placed to or received by a monitored device, Uptivity receives the event via TAPI and issues a record start message to the CUCM, also using TAPI.
Cisco Unified Communications Manager (UCM)	The CUCM negotiates audio stream network ports and codecs between the phone and Uptivity using SIP on the Uptivity side and SCCP or SIP to the phone. Audio is redirected to the Uptivity recording server through a SIP Trunk.
NICE Uptivity Application Server	Receives call control events and business data and provides a CTI interface for recording. The Uptivity server has these responsibilities: • Sending call start/call stop messages using the Uptivity API • Starting and stopping recordings using TAPI • Providing a SIP Trunk endpoint the CUCM uses for recording • Copying finished recordings to the Uptivity storage location
Third Generation Phones	Each third generation phone being recorded uses built-in bridge to forward audio streams for each side of the call to Uptivity.

Known Limitations

- The CUCM does not allow monitoring and recording the calls of secure-capable agents (see "Security Handling in Monitoring and Recording" in the *Cisco Unified CM Features and Services Guide*).
- The CUCM does not allow monitoring or recording of whisper intercom and talkback intercom calls (see "Intercom" in the *Cisco Unified CM Features and Services Guide*).
- DSP limitations in some phone models require both inbound and outbound audio streams on a phone to utilize the same codec.
- Cisco does not support BiB recording for phones that route through a phone proxy.
- This integration does not support Digest Authentication on the SIP trunk or SRTP/Media Encryption.

Audio Codec Support

The following codecs are supported by Uptivity for recording. Depending on the phone model used and DSP resources available, not all codecs may be supported by your CUCM. If you have any difficulties enabling a specific codec, please contact your Cisco support resource for assistance.

- G.711
- G.722
- G.729a
- iLBC

Cisco Requirements

Hardware

The features used in this method of recording require third generation phones that have BiB capability. Agent devices (phones) must be able to mix media for monitoring and to fork media for recording. The list of devices that support the monitoring and recording features varies per version and device pack. If you have questions, consult your Cisco account management team to determine whether your telephone sets have this capability.

Software

- Cisco Unified Communications Manager v9.0 – v10.5(2)
- Cisco TAPI Service Provider (TSP) installed on the Uptivity system

Licensing

Each device (Cisco Unified IP Phones, soft phones, third-party devices, and video devices) provisioned in the system corresponds to a number of device license units (DLUs), depending on its capabilities. The total number of units is managed in CUCM to determine capacity. The integration uses the TAPI and BiB capabilities for each recorded phone.

NICE Uptivity Requirements

Network

Sufficient network bandwidth is required to support audio traffic between each agent phone being recorded and Uptivity.

Hardware

Uptivity hardware requirements vary depending on system configurations. Appropriate hardware is identified during the system implementation process. For additional information, search online help for keyword *site requirements.*

Software

- NICE Uptivity, v5.6 or later

Additional third-party software is required for this integration:

- CACE WinPcap version 4.1.x (available from the WinPcap website)

Licensing

- One (1) Voice seat license per named agent **or**
- One (1) Voice concurrent session license for each simultaneous call that will be recorded
- Additional licensing may be needed if optional features (such as inContact Screen Recording) are included in the system.

Customer Configuration Overview

The following table provides a high-level overview of the customer configuration steps in Cisco TAPI-BiB integrations. Links are provided for tasks that are covered in this guide.

Customer Configuration Steps for Cisco TAPI-BiB Integrations	
1	Complete all necessary physical and IP connections between the recording server(s) and the LAN.
2	Obtain any necessary Cisco software and licensing.
3	Identify Phones that Support Recording. This step is to verify that all desired recording locations have an appropriate phone.
4	Configure a TAPI User Account for Uptivity.
5	Configure a SIP Trunk Security Profile for the Recording Trunk.
6	Configure a SIP Profile for the Recording Trunk.
7	Create the SIP Recording Trunk.
8	Create a Route Pattern for the SIP Recording Trunk.
9	Create a Recording Profile.
10	For all phones to be recorded, Enable Built-in-Bridge, Disable Privacy.
11	For all phones to be recorded, Add Recording Option and Recording Profile to Line Appearances. Tell your Uptivity installation team which **Recording Option** you select.

Customer Integration Tasks

The information in this section is provided for your reference only. Detailed steps for Cisco configuration can be found in Cisco's documentation, which is available on the Cisco website. You should always use the appropriate documentation from Cisco to install and configure Cisco components.

Instructions in these procedures assume that your Cisco environment uses the default settings. You should note any non-default settings and discuss them with your Uptivity installation team.

Most of the instructions in this section are based on CUCM Administrator v9.1. Other versions may have different settings.

Uptivity supports buddy core failover/resiliency, in which multiple trunks are configured on the CUCM and added to a route group. A route group can be added to a route list, and the route list would be selected inside the route pattern instead of pointing it directly at a trunk. For detailed information on configuring these specific items, refer to the *Cisco Unified Communications Manager Administration Guide*.

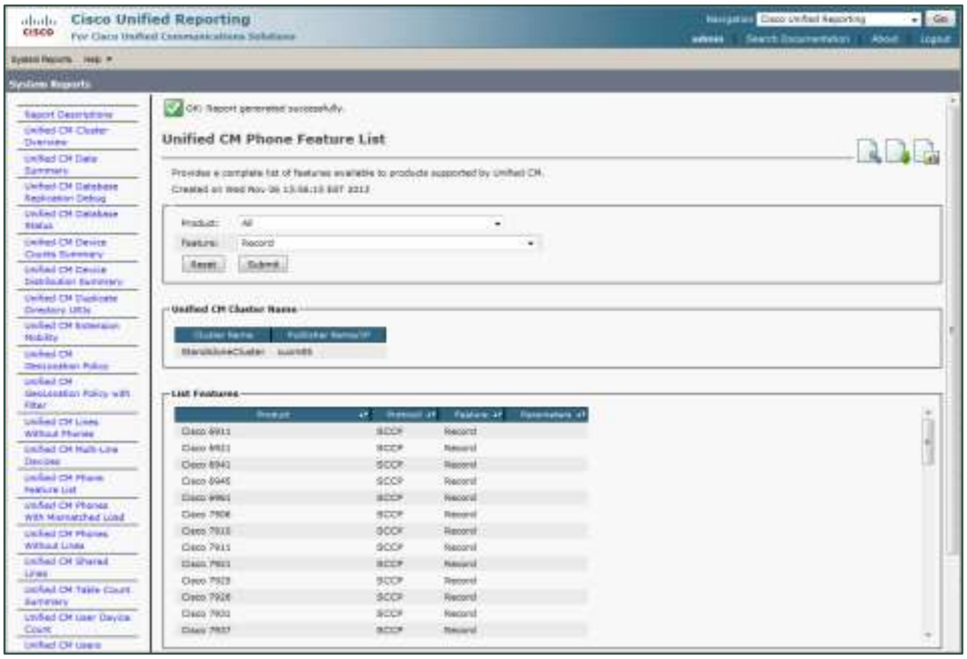
> ℹ While the TAPI-BiB integration supports buddy core failover configurations, it does ***not*** support multiple CUCM failover scenarios.

inContact **strongly** recommends that administrators performing the configuration tasks for any Cisco integration print the Customer Configuration Overview table and check each customer step as it is completed. You may also print each configuration procedure and check each step in the procedure as you complete it. The majority of Uptivity deployments which experience initial errors do so because of a Cisco setting being missed.

If you are combining the TAPI-BiB integration with an additional integration like MediaSense, UCCE, or UCCX, complete the customer procedures for this integration first. Then complete the tasks in the additional appropriate guide(s).

Identify Phones that Support Recording

> ℹ For additional information about the Cisco Unified Reporting application, refer to the *Cisco Unified Reporting Administration Guide.*



You can use the Cisco Unified Reporting application to generate a complete list of devices that support monitoring and recording for a particular release and device pack.

1. Start Cisco Unified Reporting in one of these ways:
   - Choose **Cisco Unified Reporting** in the Navigation menu in Cisco Unified Communications Manager Administration and click **Go**.
   - Choose **File** > **Cisco Unified Reporting** at the Cisco Unified Real Time Monitoring Tool (RTMT) menu.
   - Enter https://<server name or IP address>:8443/cucreports/ and then enter your authorized username and password.
2. Click **System Reports** in the navigation bar and in the left column list of reports, click **Unified CM Phone Feature List**.
3. Click **Generate a new report** to generate a new report or click **Unified CM Phone Feature List** if the report already exists.
4. For **Product**, select ***All*** from the drop-down list.

5. For **Feature**, select ***Record*** from the drop-down list.

6. Click **Submit**.

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

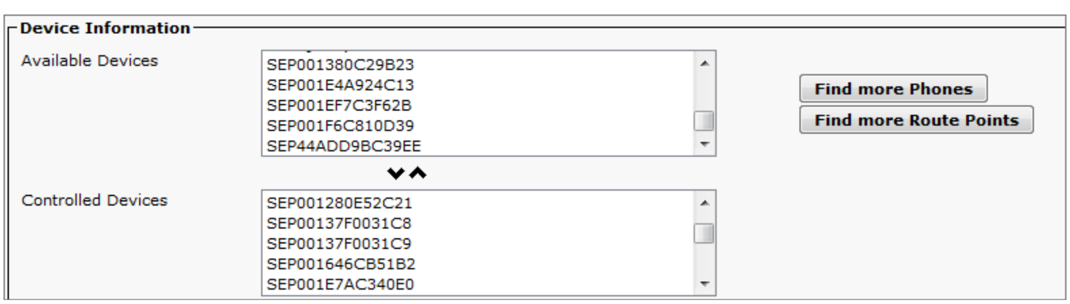
Configure a TAPI User Account for Uptivity

A user account must be created on the CUCM for Uptivity to use to connect and receive TAPI events for phones. This user account must be configured to monitor all devices that you want to record. If a device is not listed as a **Controlled Device** in the **Device Information** section of the user account, it will **not** be monitored or recorded. Device IDs shown in this section are the Selsius identifiers of the phones ("SEP" followed by the MAC address of the device).

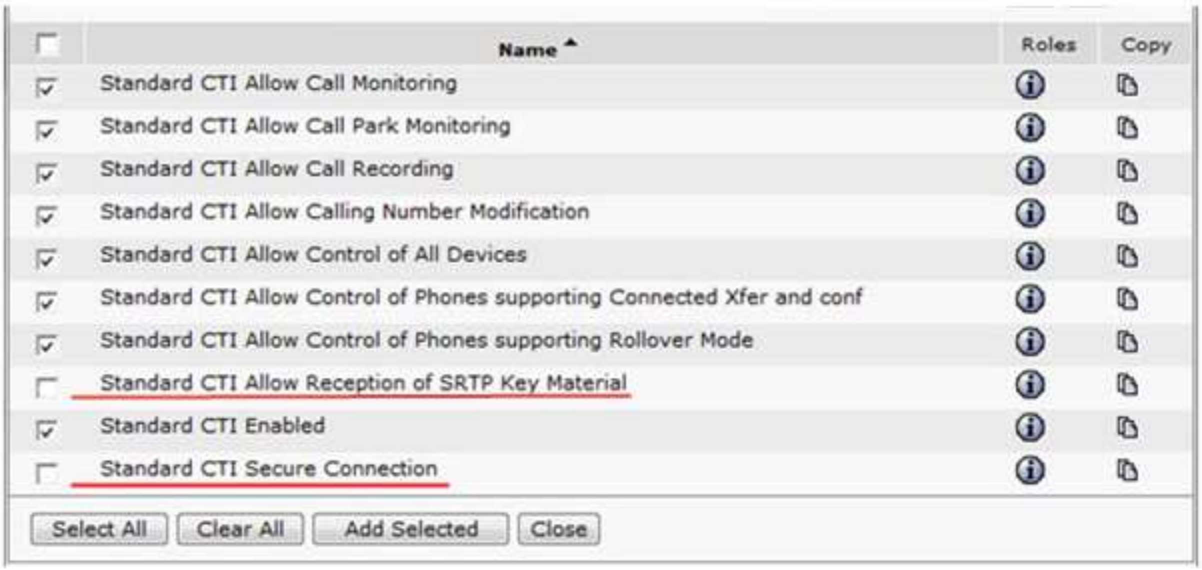
The Uptivity user account must also be added to all Access Control Groups whose names begin with "Standard CTI" **except** for the "Standard CTI Allow Reception of SRTP Key Material" and "Standard CTI Secure Connection" groups. Only select the Secure Connection group if TAPI encryption is configured. If you select this group and encryption is not configured, the CUCM can refuse "insecure" or non-encrypted connections, and call recording will not occur.

1. Log in to the **Cisco Unified CM Administration** portal with an administrative account.
2. Select **User Management** > **Application User**.
3. On the **Find and List Application Users** page, click **Add New**.
4. On the **Application User Configuration** page, set the **User ID** field and the **Password** in the **Application User Information** section. Note these values and provide them to your Uptivity installation team.

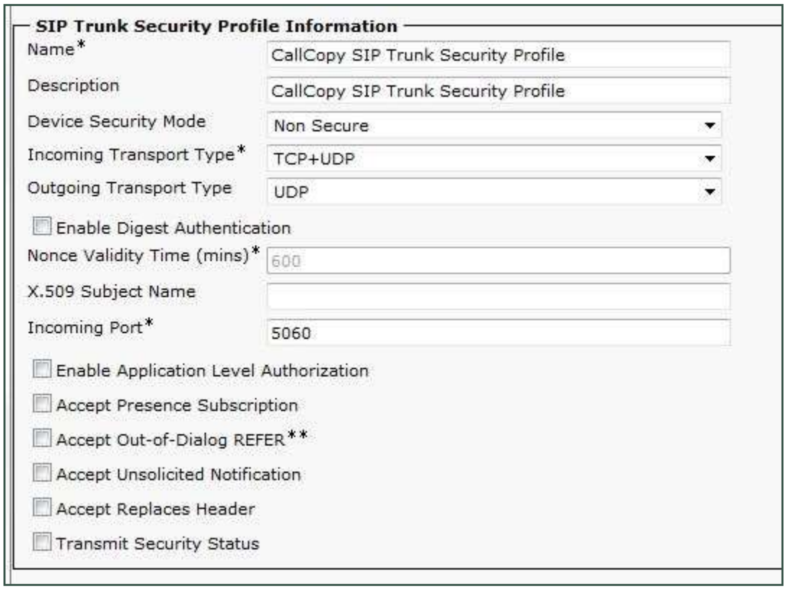
5. Under **Device Information**, move any devices that Uptivity will record to the **Controlled Devices** panel.



6. Under **Permissions Information**, click **Add to Access Control Group**.
7. Select the appropriate groups (note the excluded groups in the following image) and click **Add Selected**.



Configure a SIP Trunk Security Profile for the Recording Trunk



inContact recommends creating a separate SIP trunk security profile for the trunk between the CUCM and the recording server. This profile prevents changes made to other security profiles from interfering with call recording. Since changing security settings requires a restart of any trunks using these settings, creating a separate security profile also minimizes the need to reset existing SIP trunks.

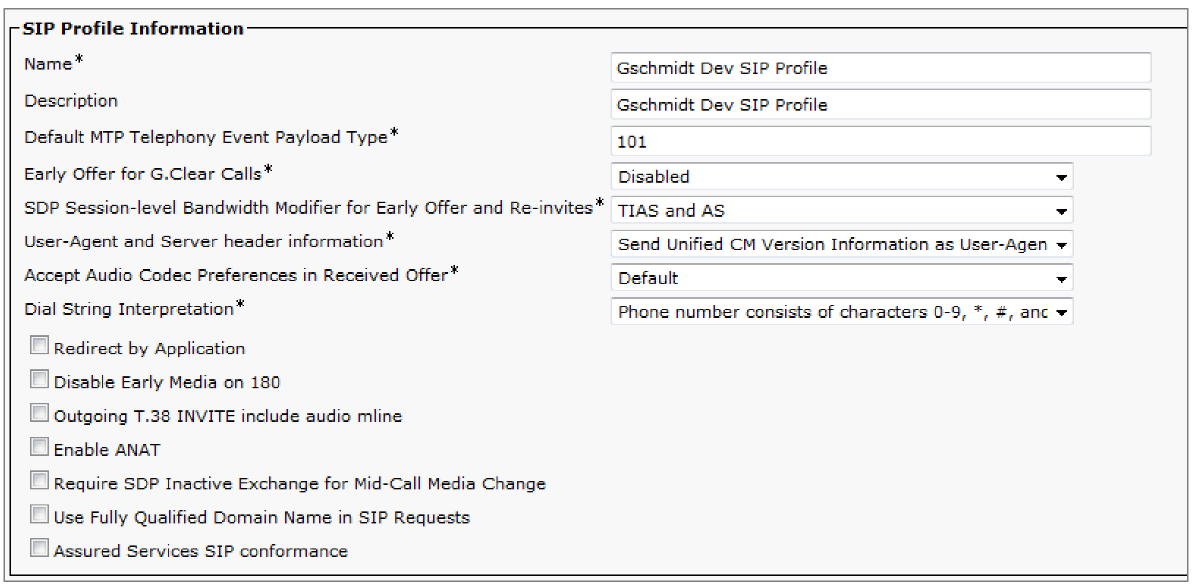
1. Log in to the **Cisco Unified CM Administration** portal with an administrative account.
2. Select **System** > **Security** > **SIP Trunk Security Profile**.
3. Click **Add New**.
4. Enter a **Name** and a **Description** for this trunk.
5. For **Device Security Mode**, select ***Non Secure*** from the drop-down list.
6. For **Incoming Transport Type**, select ***TCP+UDP*** from the drop-down list.
7. For **Outgoing Transport Type**, select ***UDP*** from the drop-down list.
8. Do not select (or clear if selected) the checkbox for **Enable Digest Authentication**.

Leave all other settings at their default values.

Configure a SIP Profile for the Recording Trunk

inContact recommends creating a separate SIP profile for the recording trunk, which protects it from changes made to SIP profiles for other trunks. All SIP devices using this profile must be restarted before any changes will take effect. This SIP profile should use the default settings shown in the images in this section. If any settings in your SIP profile do not match, discuss this with your Uptivity installation team.

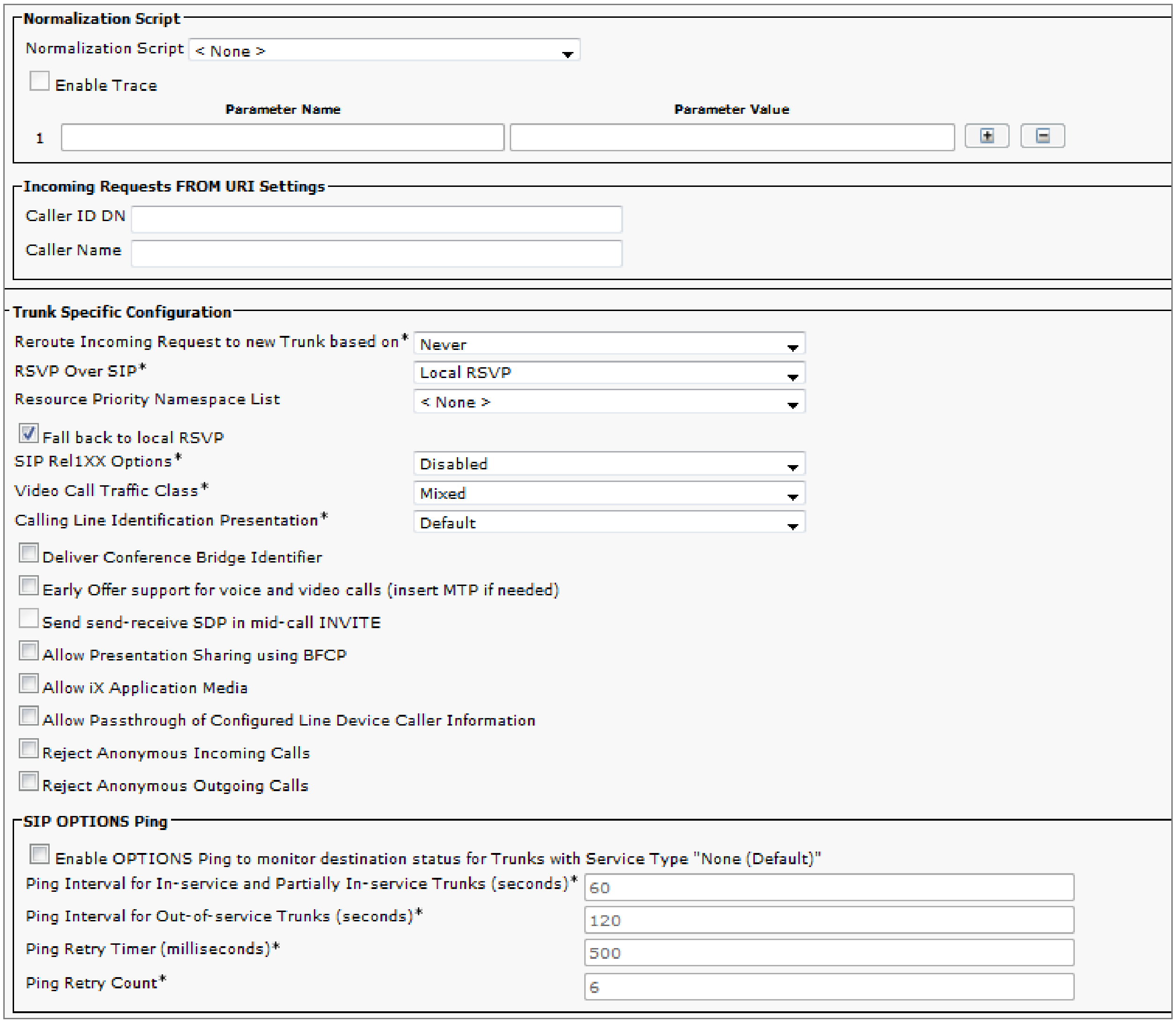
1. Log in to the **Cisco Unified CM Administration** portal with an administrative account.
2. Select **Device** > **Device Settings** > **SIP Profile**.
3. Click **Add New**.
4. Enter a **Name** and **Description** for the SIP profile.
5. Verify the settings as shown and then click **Save**.



Parameters used in Phone

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

- [x] Conference Join Enabled
- [ ] RFC 2543 Hold
- [x] Semi Attended Transfer
- [ ] Enable VAD
- [ ] Stutter Message Waiting
- [ ] MLPP User Authorization



Create the SIP Recording Trunk

The audio streams to be recorded will be routed to Uptivity over a SIP trunk configured on the CUCM. Except where specifically noted, all settings should match the default settings shown in the images in this section. If any settings in your SIP profile do not match, discuss this with your Uptivity installation team.

1. Log in to the **Cisco Unified CM Administration** portal with an administrative account.

2. Select **Device** > **Trunk**.

3. Click **Add New**.

4. For **Trunk Type**, select ***SIP Trunk*** from the drop-down list.

5. For **Device Protocol**, select ***SIP*** from the drop-down list.

6. For **Trunk Service Type**, leave the setting at ***None (Default)***, and click **Next**.

7. Enter a **Device Name**.

8. For **Device Pool**, select ***Default*** from the drop-down list.

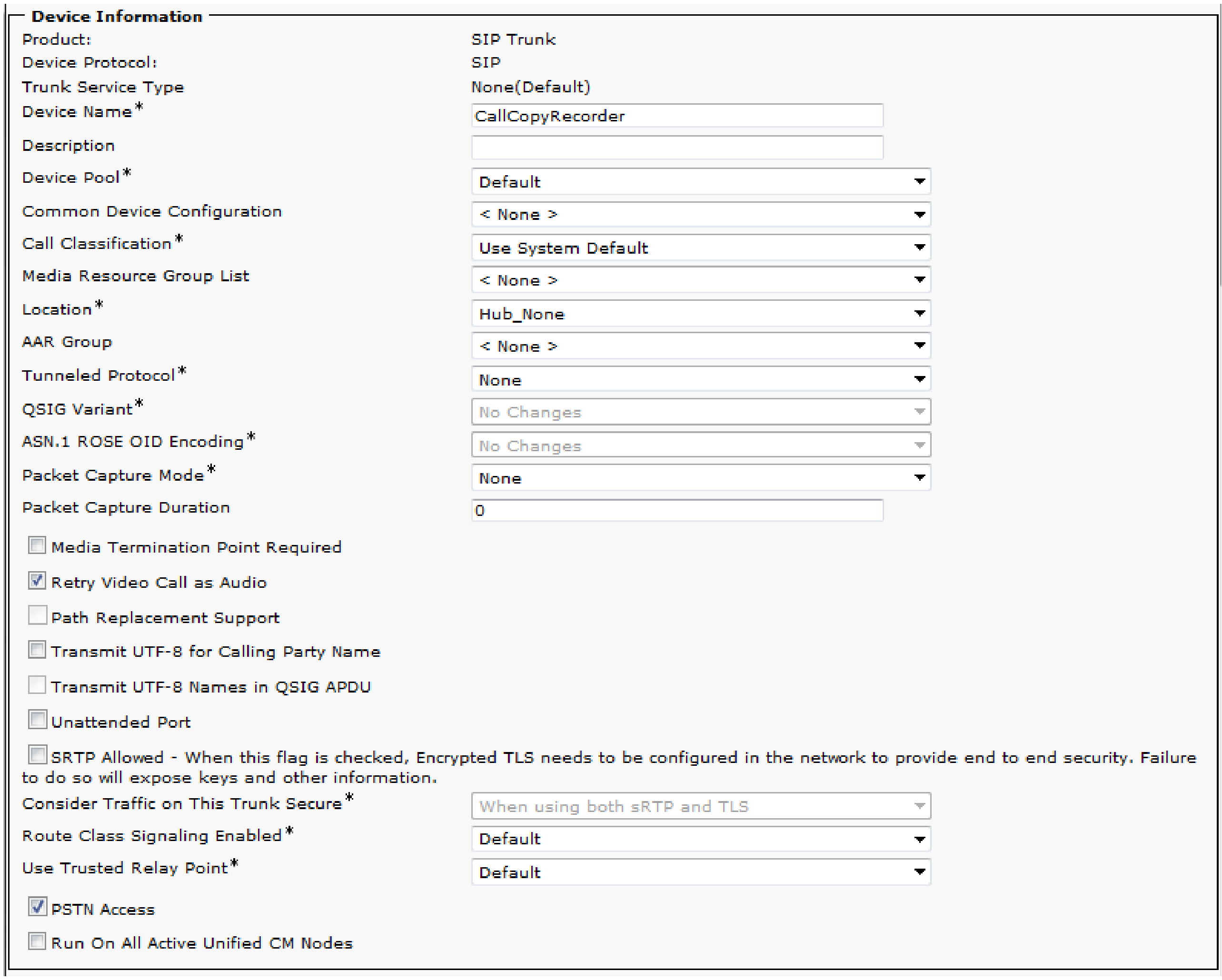
9. In **Destination Address** in the **SIP Information**, enter the IP address assigned to the Uptivity recording server (that is, the server on which the **CTI Core** service for this integration is running).

10. In **Destination Port** in the **SIP Information**, enter ***5060***.

11. For **SIP Trunk Security Profile** under **SIP Information**, enter the name of the profile this trunk will use (see Configure a SIP Trunk Security Profile for the Recording Trunk).

12. For **SIP Profile** under **SIP Information**, enter the name of the profile this trunk will use (see Configure a SIP Profile for the Recording Trunk).

13. Verify that all remaining settings on the **Trunk Configuration** page match the default settings in the images shown here, and click **Save**.



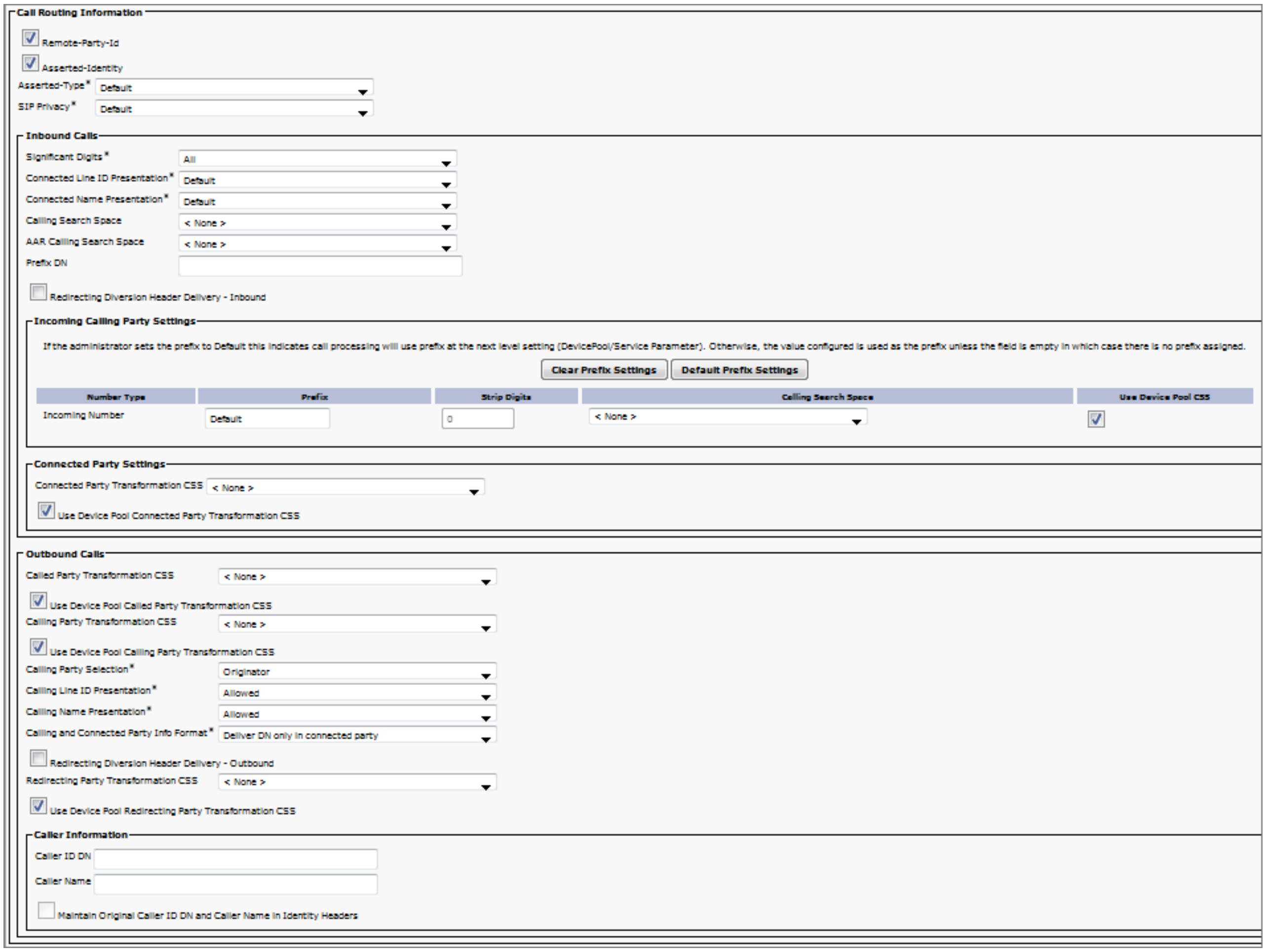
Device Information

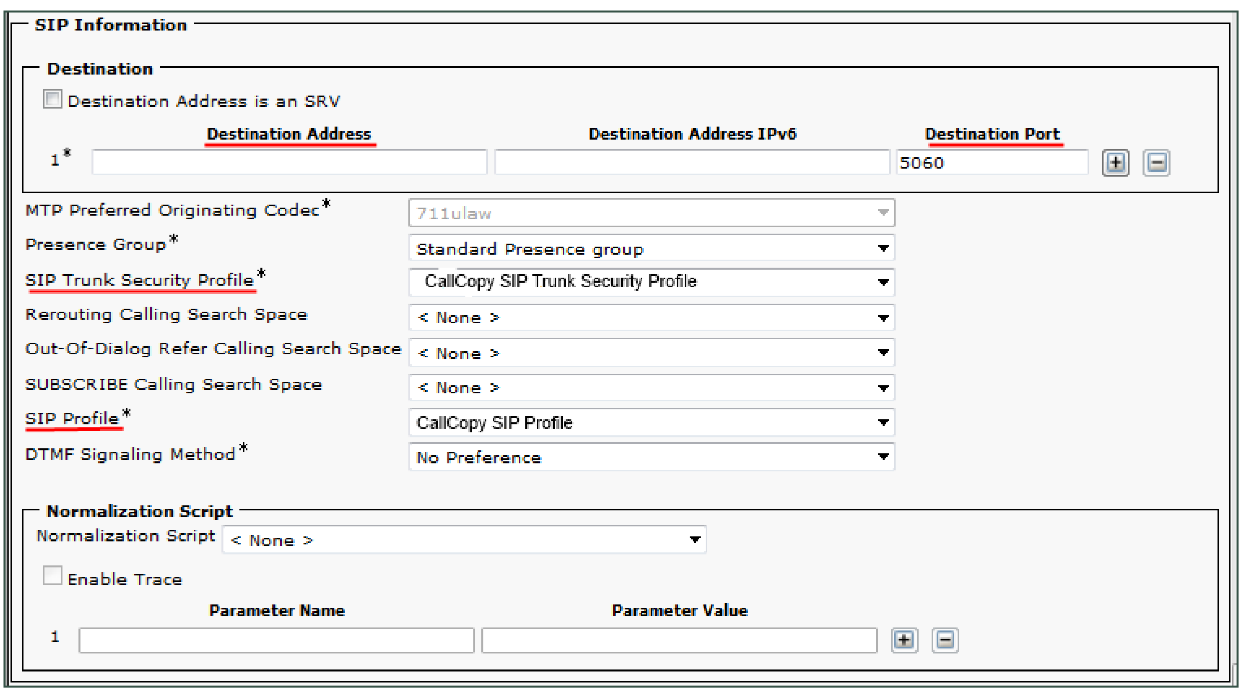
Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	CallCopyRecorder
Description	
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*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

- [ ] Media Termination Point Required
- [x] Retry Video Call as Audio
- [ ] Path Replacement Support
- [ ] Transmit UTF-8 for Calling Party Name
- [ ] Transmit UTF-8 Names in QSIG APDU
- [ ] Unattended Port
- [ ] SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default

- [x] PSTN Access
- [ ] Run On All Active Unified CM Nodes


Call Routing Information
Remote-Party-Id
Asserted-Identity
Asserted-Type* Default
SIP Privacy* Default
Inbound Calls
Significant Digits* All
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Calling Search Space < None >
AAR Calling Search Space < None >
Prefix DN
Redirecting Diversion Header Delivery - Inbound
Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.
Clear Prefix Settings
Default Prefix Settings
Number Type
Prefix
Strip Digits
Calling Search Space
Use Device Pool CSS
Incoming Number
Default
0
< None >
Connected Party Settings
Connected Party Transformation CSS < None >
Use Device Pool Connected Party Transformation CSS
Outbound Calls
Called Party Transformation CSS < None >
Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS < None >
Use Device Pool Calling Party Transformation CSS
Calling Party Selection* Originator
Calling Line ID Presentation* Allowed
Calling Name Presentation* Allowed
Calling and Connected Party Info Format* Deliver DN only in connected party
Redirecting Diversion Header Delivery - Outbound
Redirecting Party Transformation CSS < None >
Use Device Pool Redirecting Party Transformation CSS
Caller Information
Caller ID DN
Caller Name
Maintain Original Caller ID DN and Caller Name in Identity Headers


SIP Information
Destination
Destination Address is an SRV
Destination Address
Destination Address IPv6
Destination Port
1*
5060
MTP Preferred Originating Codec* 711ulaw
Presence Group* Standard Presence group
SIP Trunk Security Profile* CallCopy SIP Trunk Security Profile
Rerouting Calling Search Space < None >
Out-Of-Dialog Refer Calling Search Space < None >
SUBSCRIBE Calling Search Space < None >
SIP Profile* CallCopy SIP Profile
DTMF Signaling Method* No Preference
Normalization Script
Normalization Script < None >
Enable Trace
Parameter Name
Parameter Value
1

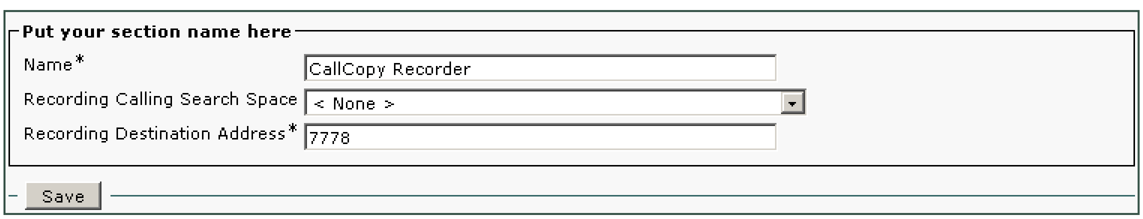
Create a Route Pattern for the SIP Recording Trunk

You must configure a route pattern extension to route audio streams from the BiB to the newly-created SIP trunk.

1. Log in to the **Cisco Unified CM Administration** portal with an administrative account.
2. Select **Call Routing** > **Route/Hunt** > **Route Pattern.**
3. Click **Add New**.
4. Enter a **Route Pattern** number.
5. Set the **Gateway/Route List** value to the SIP trunk.
6. Do not select (or clear if selected) the check box for **Require Forced Authorization Code**. This pattern cannot require a forced authorization code or Uptivity will not receive call audio.
7. Click **Save**.

If you receive messages about activating an Authorization Code and resetting the Gateway, follow the direction specified in those messages.

Create a Recording Profile



You must create a recording profile for the route pattern assigned to the SIP recording trunk. This recording profile must reference the correct recording calling search space for the phones you wish to record. A misconfigured search space can result in recordings with no audio and line errors in logging.

1. Log in to the **Cisco Unified CM Administration** portal with an administrative account.
2. Select **Device** > **Device Settings** > **Recording Profile**.
3. Click **Add New**.
4. Enter a **Name**.

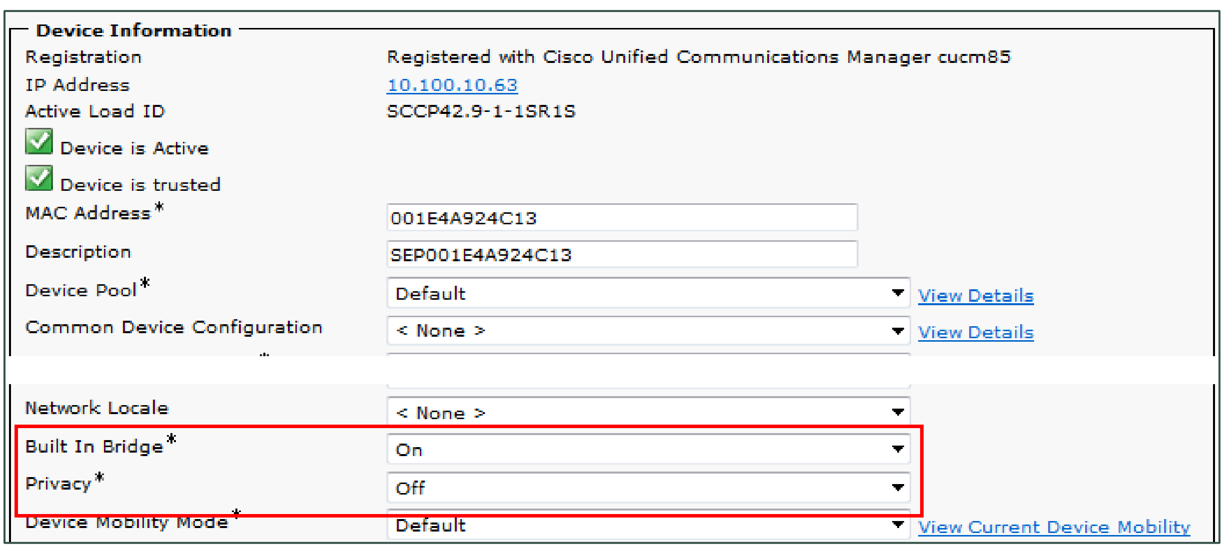
5. For **Recording Calling Search Space**, select ***None***, ***Default***, or the appropriate search space from the drop-down list.
6. For **Recording Destination Address**, enter the route pattern assigned to the SIP recording trunk.
7. Click **Save**.

Enable Built-in-Bridge, Disable Privacy

Built-in-Bridge (BiB) and Privacy mode can be configured at both the device and server level. The instructions below are for devices.

BiB and Privacy have these setting options: ***On***, ***Off***, and ***Default***. The ***Default*** option causes the device to use the server-level value for this setting and you must select this value if these settings are configured at the server level. If ***Default*** is not used, the device-level setting overrides the server-level setting. Conflicts between the server and device-level settings prevent call recording.

You must enable the BiB feature on every phone to be recorded. This feature is what creates a separate audio stream of any in-progress call and routes it to the Uptivity system for recording. You can optionally use the Bulk Administration Tool to create and schedule a job that will update multiple phones at once. See your Cisco documentation for specific instructions.



1. Log in to the **Cisco Unified CM Administration** portal with an administrative account.
2. Select **Device** > **Phone**.

3. Run a query to locate devices you want to record. Queries can be run against many parameters, such as **Device Name** or a particular **Directory Number** associated with a device.

4. From the resulting list of phones, click **Device Name** to edit the **Phone Configuration**.

5. Under the Device Information section, set **Built-in-Bridge** to ***On***.

6. Set **Privacy** to ***Off***. Phones with the privacy feature enabled cannot be recorded.

Repeat this task for all phones to be recorded.

Add Recording Option and Recording Profile to Line Appearances

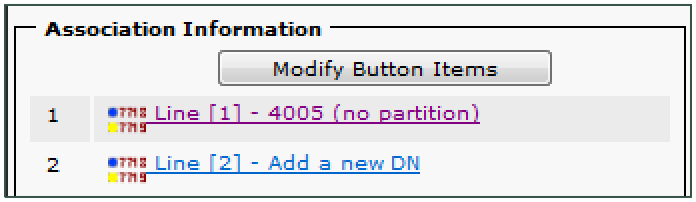
The recording profile (see Create a Recording Profile) must be added to each individual line appearance on a phone so that recording is allowed for that appearance. You also need to configure recording options for the appearance. This integration supports the **Selective Call Recording Enabled** and **Automatic Call Recording Enabled** options. Your Uptivity installation team needs to know which recording option you are using.

An alternate way of locating directory numbers is to select **Call Routing** > **Directory Number**. You can optionally use the Bulk Administration Tool to schedule a job that will update multiple line appearances at once.

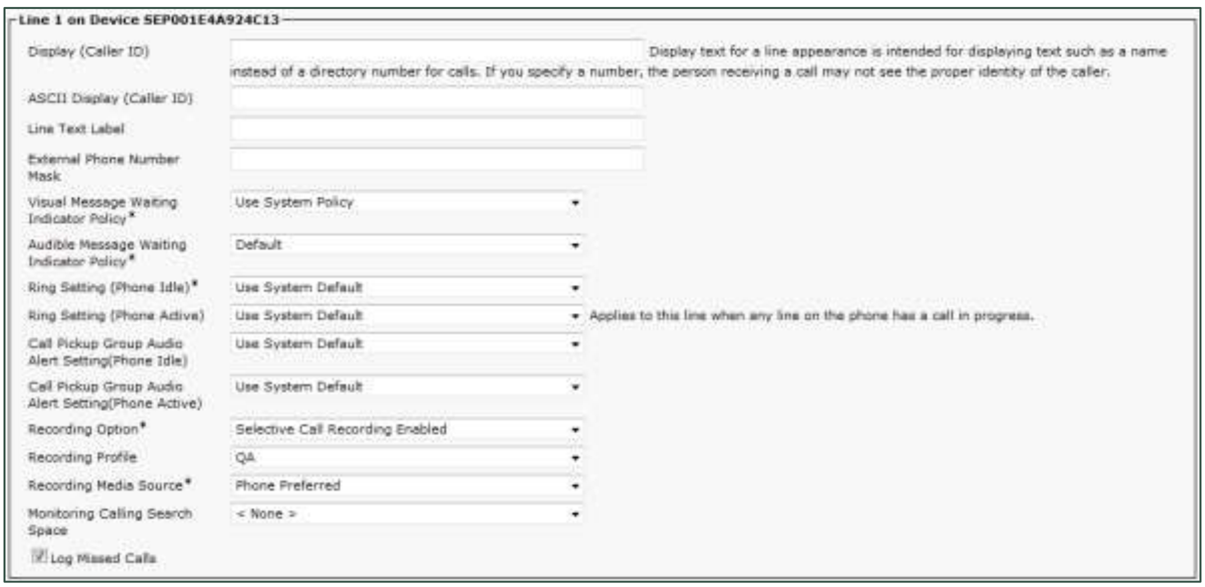
> When you create new phones, be aware that using the **Copy** or **Super Copy** option does not copy the line/directory number information. The Add Recording Option and Recording Profile to Line Appearances task must be performed manually.

1. Log in to the **Cisco Unified CM Administration** portal with an administrative account.
2. Select **Device** > **Phone**.
3. Run a query to locate devices you want to record. Queries can be run against many parameters, such as **Device Name** or a particular **Directory Number** associated with a device.
4. From the resulting list of phones, click **Device Name** to edit the **Phone Configuration**.

5. Click the **Line Appearance** to be recorded under the **Association Information** section.



6. Enable the correct **Recording Option** in the **Line Settings** section.
7. Set **Recording Profile** to the profile created earlier.
8. Set **Recording Media Source** to ***Phone Preferred***.
9. Click **Save**.



Repeat this task for any lines that will be recorded.

Customer Administration Tasks

There are no regular, ongoing administrative tasks related to this integration. If you add channels to your system, your Uptivity administrator will need to increase the channel count on the voice board in the **Web Portal**. For more information on voice board tasks, search online help for keyword *voice boards*.

> You must restart the **CTI Core** service after any changes to voice boards, channels, or both.

Any other voice board changes should only be done under direct supervision from Uptivity Support. Done incorrectly, voice board modifications can have serious negative impact to your system. In addition, altering the hardware configuration of your system may void your warranty.

The Cisco JTAPI-BiB integration streams audio from the phone to Uptivity, and supports multiple calls on the same line or multiple lines on the same phone simultaneously. Therefore, it is important ***not*** to specify any devices on the voice board.

Document Revision History

Revision	Change Description	Effective Date
0	Initial release for this version	2016-04-05
1	Rebranded for NICE Uptivity.	2017-03-31