Customer Guide to SIP Trunk Integrations

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Version: This guide should be used with NICE Uptivity (formerly Premise inContact WFO) v5.6 or later.

Contact: Send suggestions or corrections regarding this guide to documentationrequests@incontact.com.

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Introduction

Audience

This document is written for customers and prospective customers interested in using NICE Uptivity in an IP telephony environment using SIP trunks. Readers who will perform procedures in this guide should have a basic level of familiarity with IP telephony, SIP trunks, general networking, the Windows operating system, their specific IP PBX, and NICE Uptivity.

Goals

The goal of this document is to provide knowledge, reference, and procedural information necessary to understand a proposed NICE Uptivity integration using one or more SIP trunks as an audio source, and to configure the telephony 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 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 information and procedures related to Uptivity configuration, talk to your Uptivity installation team.

This integration provides a means of audio capture only; if a CTI source will be leveraged for call control and metadata, additional steps may be required. Consult the Uptivity customer guide for the applicable CTI integration.

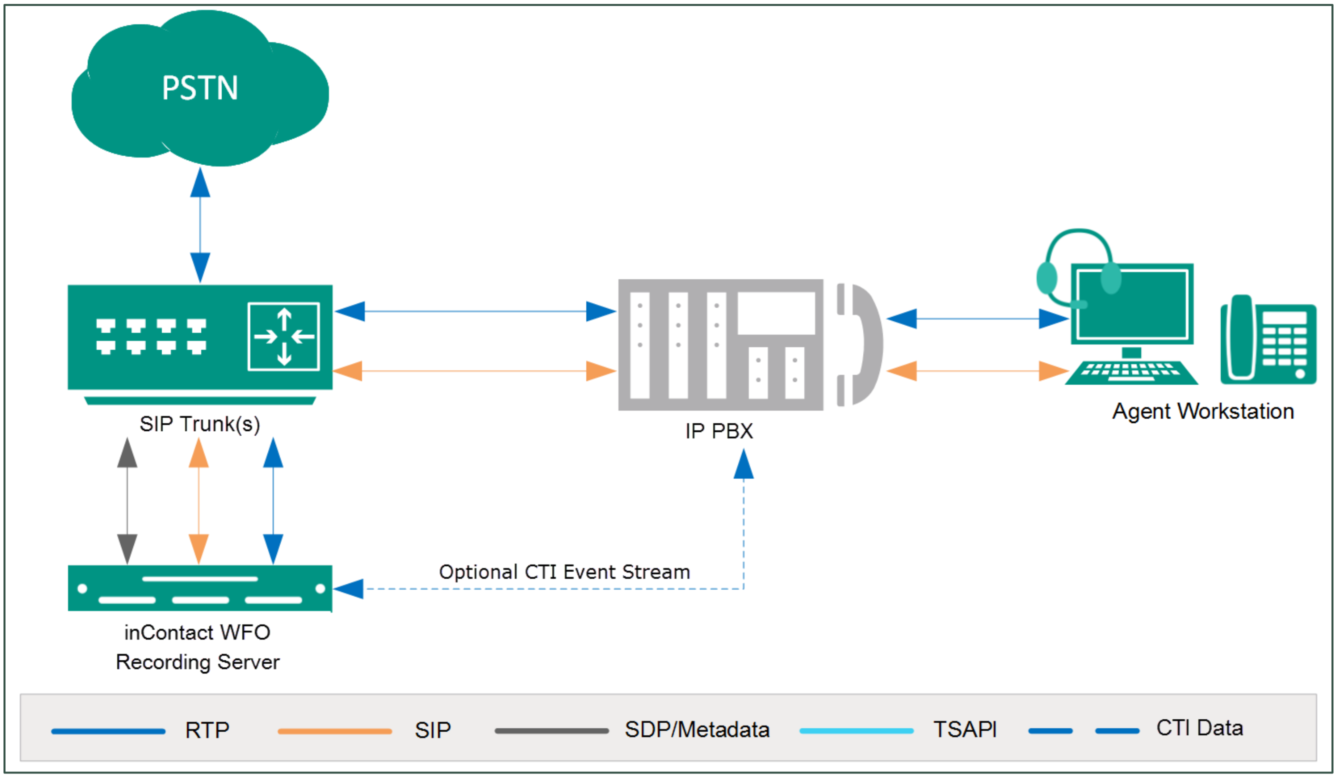
This integration supports live monitoring capability, can be used with Uptivity Screen Recording, and should work with any standard SIP trunk. It has been tested with Cisco Unified Communications Manager and the inContact Cloud Contact Center.

Customer Responsibilities

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

SIP Trunk Integration Overview

SIP trunks are a standard means of delivering IP telephone services and unified communications to customers with a SIP-based IP-PBX. This integration can be used to record any traffic routed over a SIP trunk. It can also be used to provide ad hoc recording of calls that are not otherwise captured by Uptivity. Agents create a conference between themselves, the customer, and the SIP trunk. Uptivity records the trunk and thus the conference call.



General architectural example of the SIP trunk integration

Component	Function
SIP Trunk(s)	Provides the audio connection to Uptivity; may also provide call control events in the form of SIP signaling.
IP PBX	The IP PBX negotiates audio stream network ports and codec between the phone and Uptivity. Audio is redirected to the Uptivity server through the SIP Trunk(s). May also provide call control events in the form of CTI data.
NICE Uptivity Recording 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 • Providing a SIP Trunk endpoint for recording • Copying the finished recordings to the Uptivity storage location

Known Limitations

The following limitations apply when the integration involves Cisco Unified Communications Manager (CUCM)

- 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*)
- 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
- CUCM 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 PBX/ACD. If you have any difficulties enabling a specific codec, please contact your telephony vendor for assistance.

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

Telephony Requirements

SIP trunk integration is dependent on the PBX and network topologies employed in the phone system. Due to the varying configurations and complexities possible, an Uptivity Sales Engineer must determine whether SIP trunk integration is viable, and if so, how to deploy it properly.

NICE Uptivity Requirements

Network

Sufficient network bandwidth is required to support audio traffic between each agent phone(s), the SIP trunk(s), and Uptivity.

Hardware

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

Software

- NICE Uptivity v 5.6 or later

Additional third-party software is required for this integration:

- CACE WinPcap version 4.1.x, available from Uptivity or 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 required if the system includes optional features (for example, inContact Screen Recording)

Customer Integration Tasks

SIP Trunk Integration in Cisco Environments

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 tasks assume that 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*.

inContact **strongly** recommends that administrators performing the configuration tasks for any Cisco integration print the Customer Configuration Overview for Cisco SIP Trunk Integrations table and check each customer step as it is completed. You may also wish to print each configuration task 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.

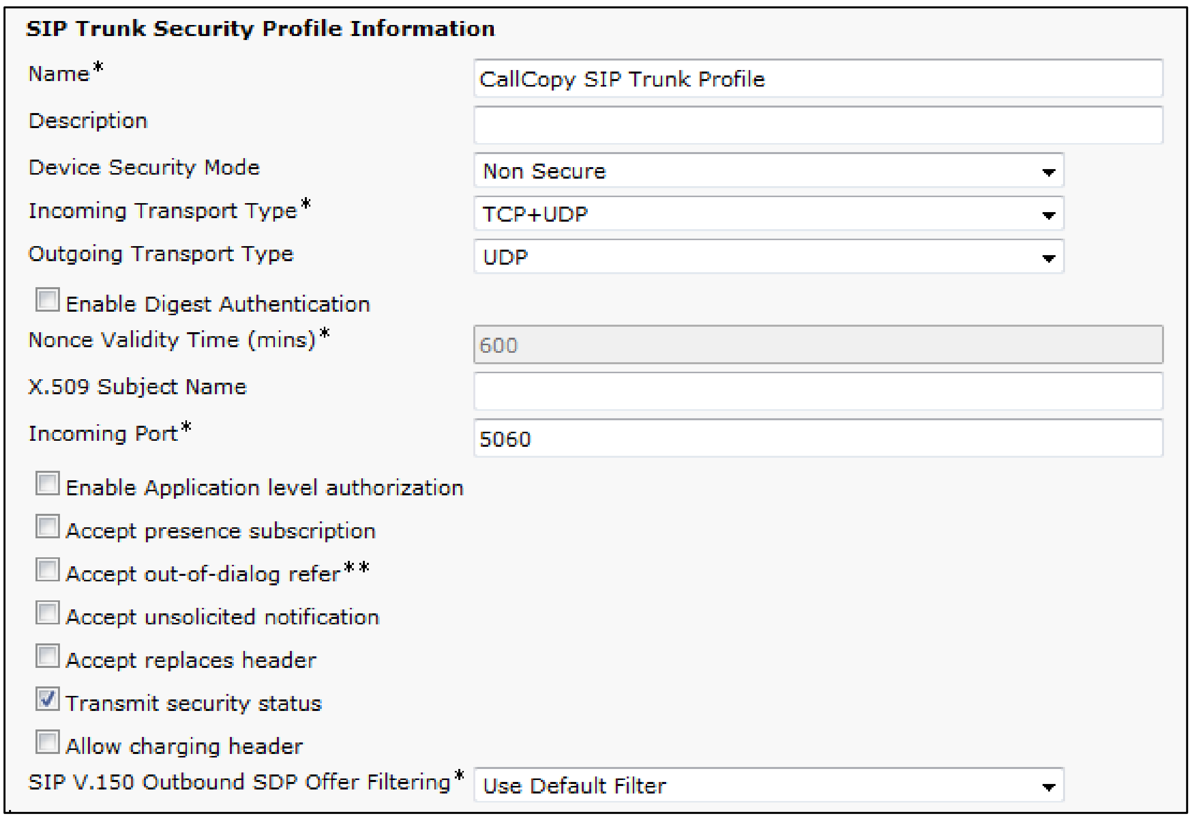
If you are combining the SIP trunk integration with an additional CTI integration, complete the customer procedures for this integration first. Then complete the tasks in the additional appropriate guide(s).

Customer Configuration Overview for Cisco SIP Trunk Integrations

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

Customer Configuration Steps for Cisco SIP Trunk Integrations	
1	Configure a SIP Trunk Security Profile for the Cisco Recording Trunk.
2	Configure a SIP Profile for the Cisco Recording Trunk.
3	Create a Cisco SIP Recording Trunk.
4	Configure Cisco Phones.

Configure a SIP Trunk Security Profile for the Cisco Recording Trunk


SIP Trunk Security Profile Information

Name*	CallCopy SIP Trunk Profile
Description	
Device Security Mode	Non Secure
Incoming Transport Type*	TCP+UDP
Outgoing Transport Type	UDP
☐ Enable Digest Authentication	
Nonce Validity Time (mins)*	600
X.509 Subject Name	
Incoming Port*	5060
☐ Enable Application level authorization	
☐ Accept presence subscription	
☐ Accept out-of-dialog refer**	
☐ Accept unsolicited notification	
☐ Accept replaces header	
☑ Transmit security status	
☐ Allow charging header	
SIP V.150 Outbound SDP Offer Filtering*	Use Default Filter

inContact recommends creating a separate SIP trunk security profile for the trunk between the CUCM and the Uptivity 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. Click **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 check box for **Enable Digest Authentication**.

Leave all other settings at their default values. After you complete this procedure, return to the Customer Configuration Overview for Cisco SIP Trunk Integrations.

Configure a SIP Profile for the Cisco 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 included 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.

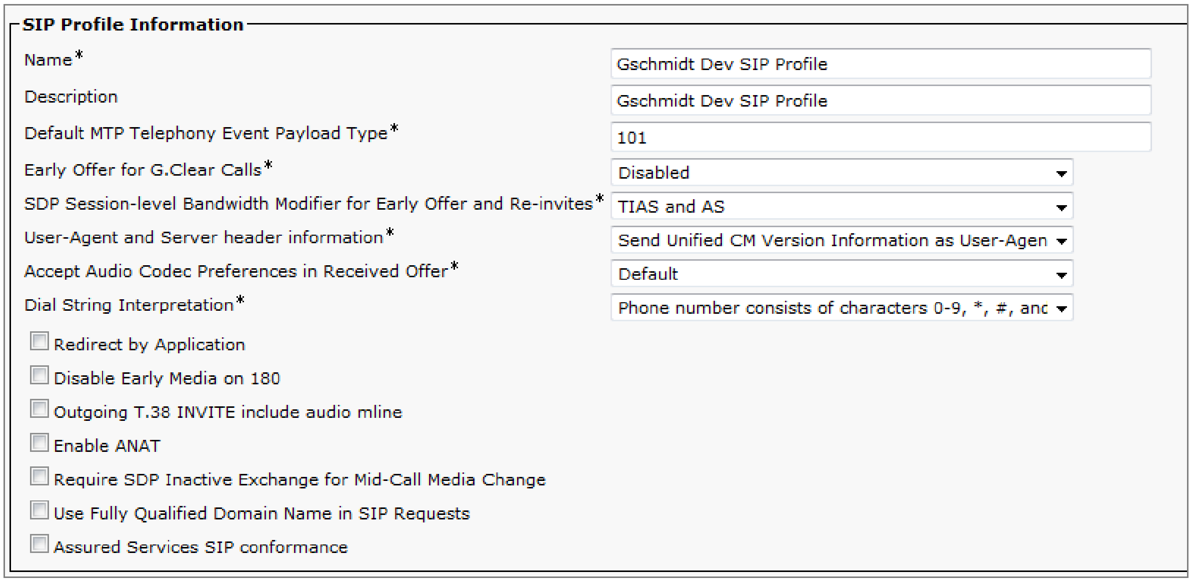
2. Select **Device** > **Device Settings** > **SIP Profile**.

3. Click **Add New.**

4. Enter a **Name** and **Description** for the SIP profile.

5. Verify the settings as shown in the following three images and then click **Save**.

After you complete this procedure, return to the Customer Configuration Overview for Cisco SIP Trunk Integrations.



SIP Profile Information

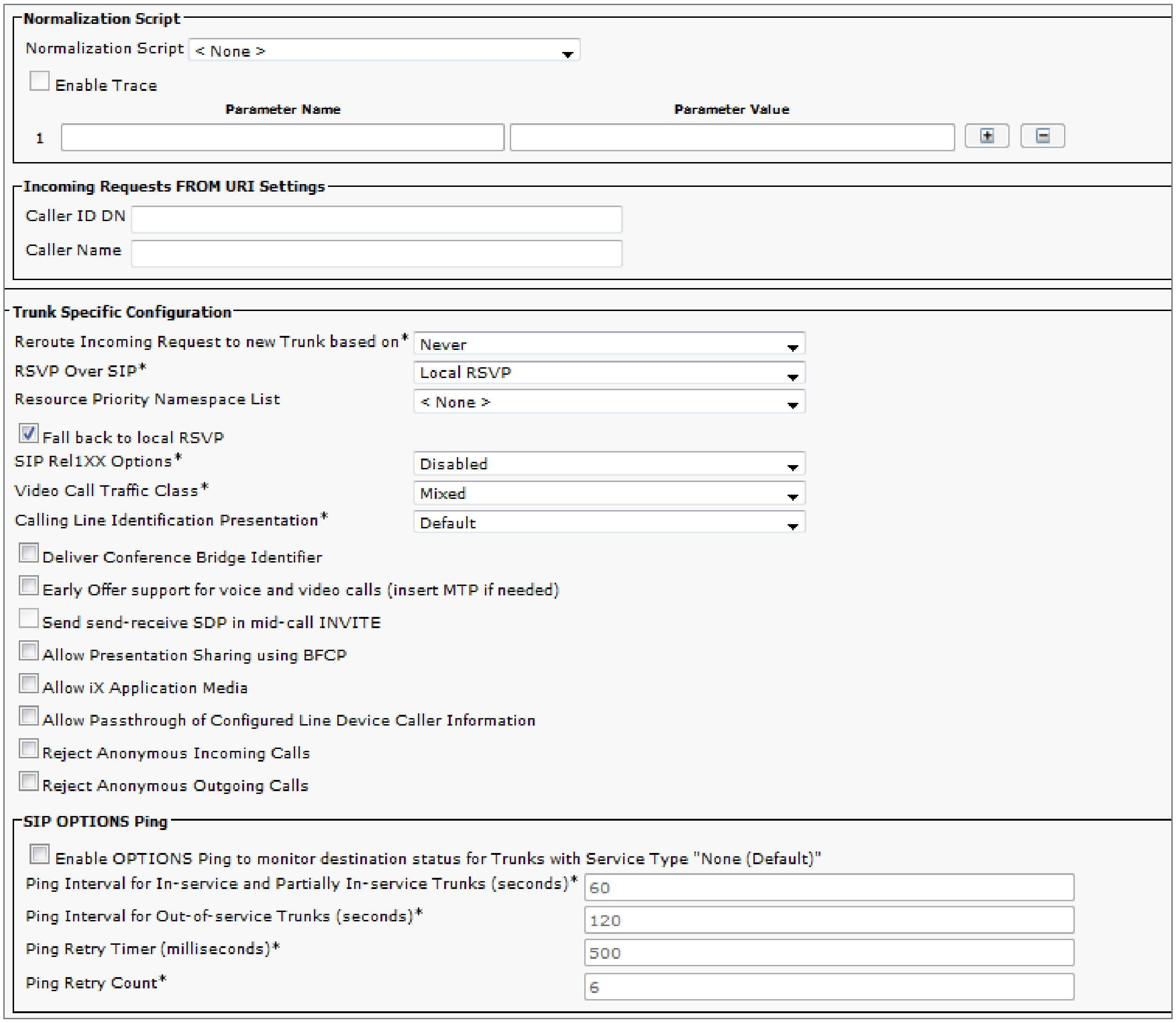
Name*	Gschmidt Dev SIP Profile
Description	Gschmidt Dev SIP Profile
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	Disabled
SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
User-Agent and Server header information*	Send Unified CM Version Information as User-Agen
Accept Audio Codec Preferences in Received Offer*	Default
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, anc

- [ ] Redirect by Application
- [ ] Disable Early Media on 180
- [ ] Outgoing T.38 INVITE include audio mline
- [ ] Enable ANAT
- [ ] Require SDP Inactive Exchange for Mid-Call Media Change
- [ ] Use Fully Qualified Domain Name in SIP Requests
- [ ] Assured Services SIP conformance

Parameters used in Phone

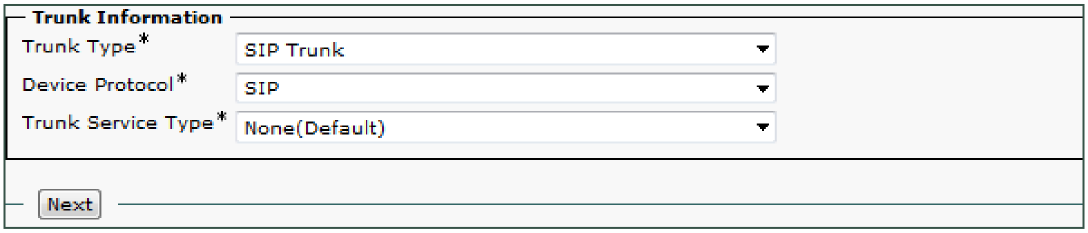
Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other 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

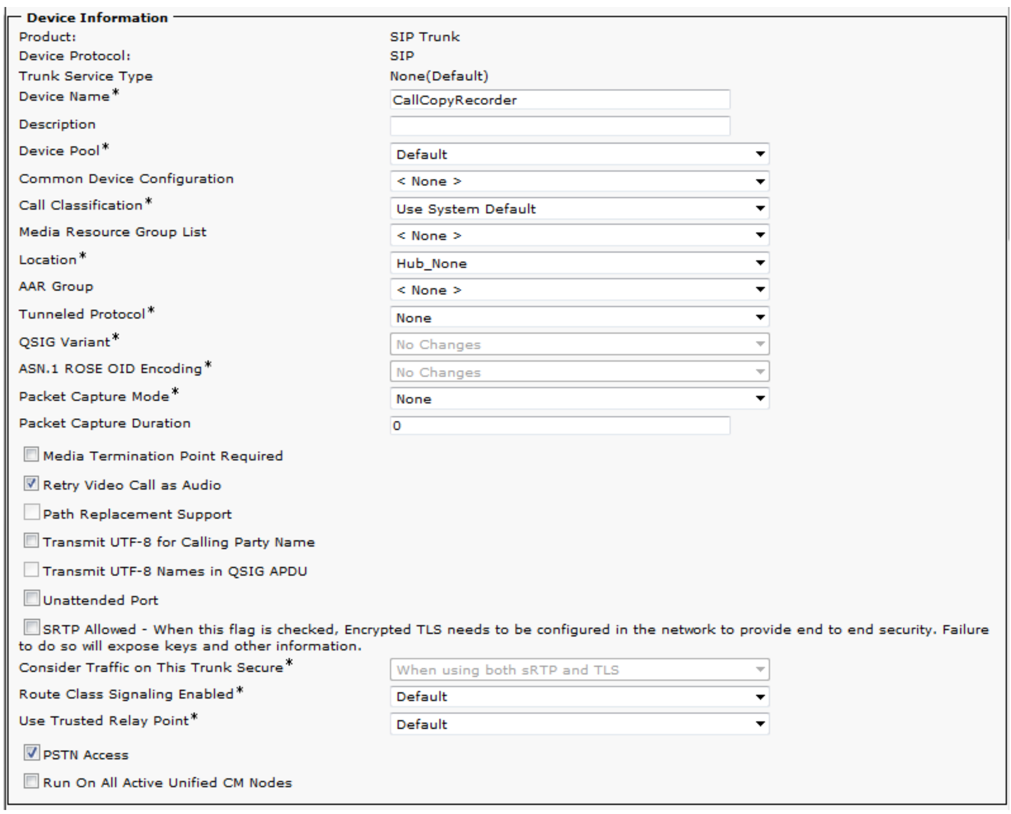


Create a Cisco SIP Recording Trunk

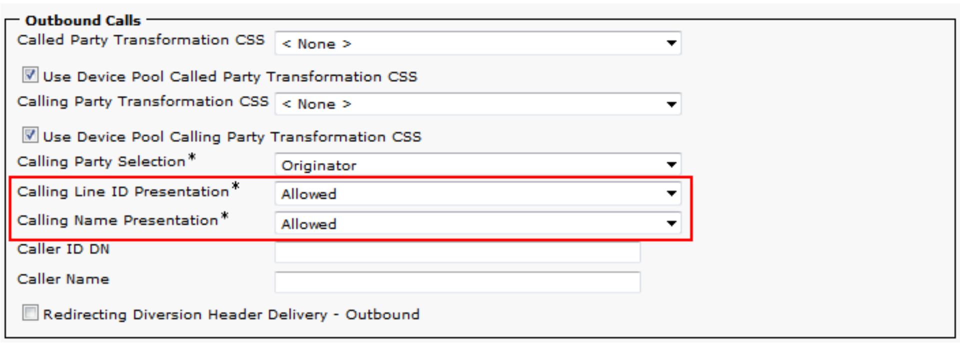
1. Log in to the **Cisco Unified CM Administration** portal with an administrative account.
2. Select **Device** > **Trunk**.
3. On the **Find and List Trunks** page, click **Add New**.
4. On the Trunk Information section of the form, configure the settings as shown in the following image.



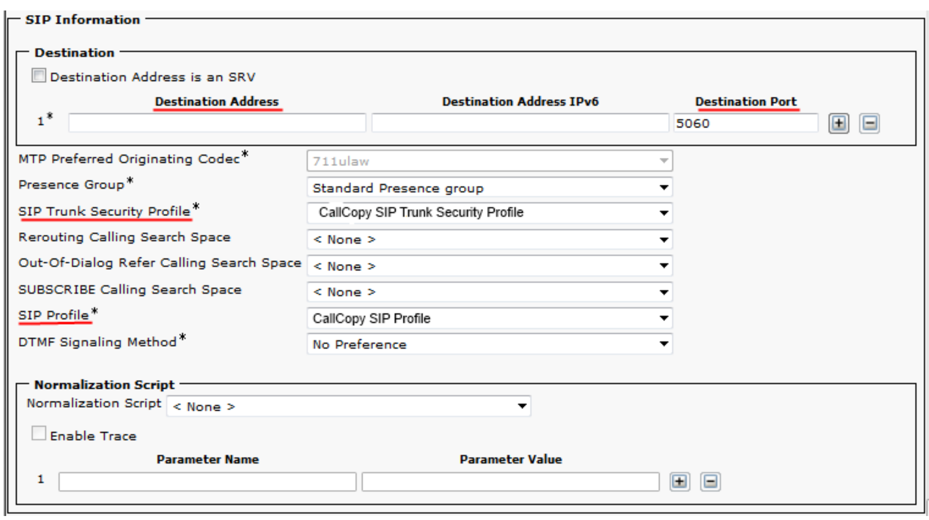
5. Click **Next** to continue.
6. Enter a **Device Name**.
7. For **Device Pool**, select ***Default***. Unless otherwise specified, the default settings on the **Trunk Configuration** page can be used. Make a note of any differences between the defaults shown in this image and the settings in your environment.



8. In the **Outbound Calls** section of the **Trunk Configuration** page, the following options should be configured:
   - **Calling Line ID Presentation**: Set to ***Allowed***.
   - **Calling Line Name Presentation**: Set to ***Allowed***.



9. In the **SIP Information** section of the **Trunk Configuration** page, the following options should be configured:
   - **Destination Address —** IP address of the Uptivity server where the Uptivity Cisco Active Recording module is installed
   - **Destination Port —** should be set to ***5060***
   - **SIP Trunk Security Profile** — should be set to the profile configured earlier
   - **SIP Profile** — should be set to the profile set earlier



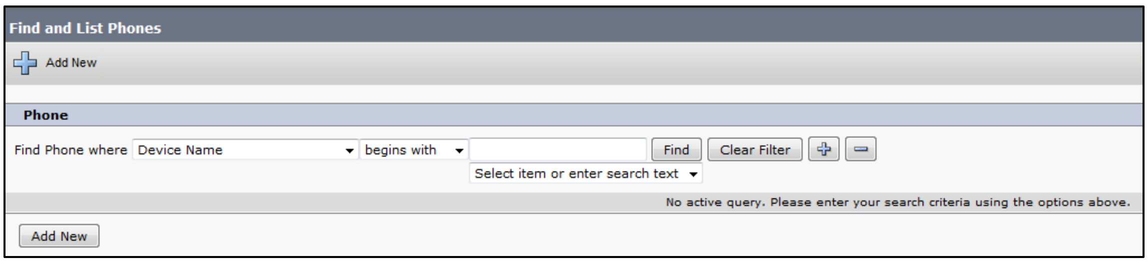
10. Click **Save**.

After you complete this procedure, reset the SIP trunk and return to the Customer Configuration Overview for Cisco SIP Trunk Integrations.

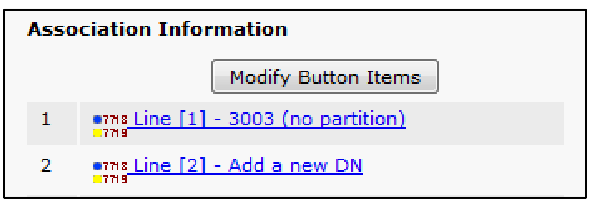
Configure Cisco Phones

If there are Cisco phones to be recorded with this integration, they must support and be configured for automatic call recording. You will typically need to run a query to locate devices you wish to record. Queries can be run against many parameters, such as the Device Name or a particular Directory Number associated with a device.

1. Log into Cisco Unified CM Administration with an appropriately-permissioned account.
2. Click **Device** > **Phone**.
3. Enter the desired query parameters and click **Find**.

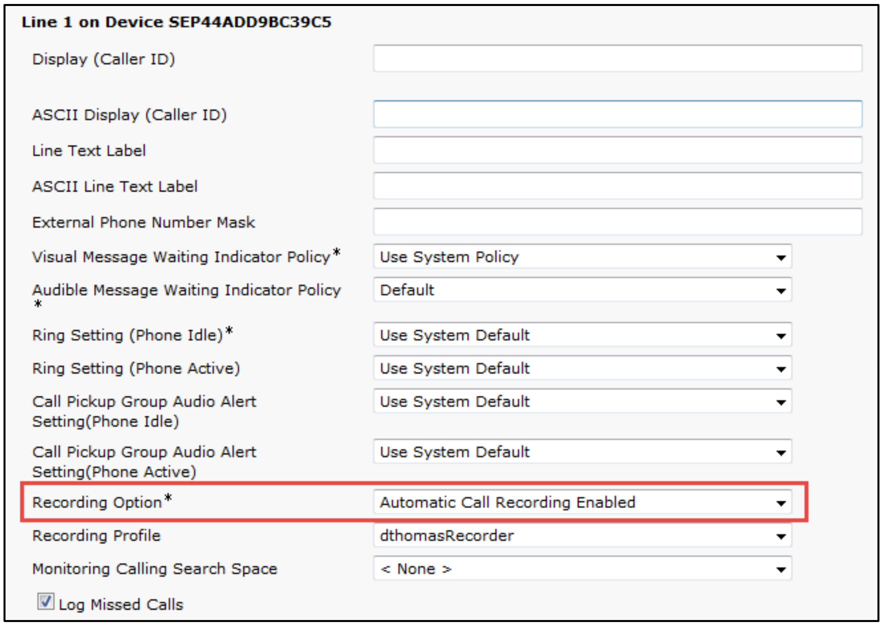


4. From the resulting list of phones, click the desired entry in the **Device Name** column.



5. Under **Association Information**, click the desired **Line** (extension).

6. Scroll to the section labeled **Line # on Device #**.



7. For **Recording Option**, select ***Automatic Call Recording Enabled*** from the drop-down list.

Repeat this procedure for any lines that will be recorded.

Customer Administration Tasks

During ongoing use of the system, your Uptivity administrator may need to configure new channels or reconfigure existing channels. At those times, this integration requires changes to the **Voice Boards** page in the **Web Portal**. If the integration uses an alternate CTI source, additional tasks may be required; refer to the appropriate customer guide for that integration.

The SIP trunk integration records conferences established between the agent phone, the SIP trunk, and Uptivity. The integration 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.

If channels are added to your system, you must increase the channel count on the associated voice board. 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.