

**NICE**



**UPTIVITY**  
Agile WFO for SMB

## **Customer Guide to SIP Trunk Integrations**

# Customer Guide to SIP Trunk Integrations

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

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Contact: Send suggestions or corrections regarding this guide to [documentationrequests@incontact.com](mailto: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.

## Introduction

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.

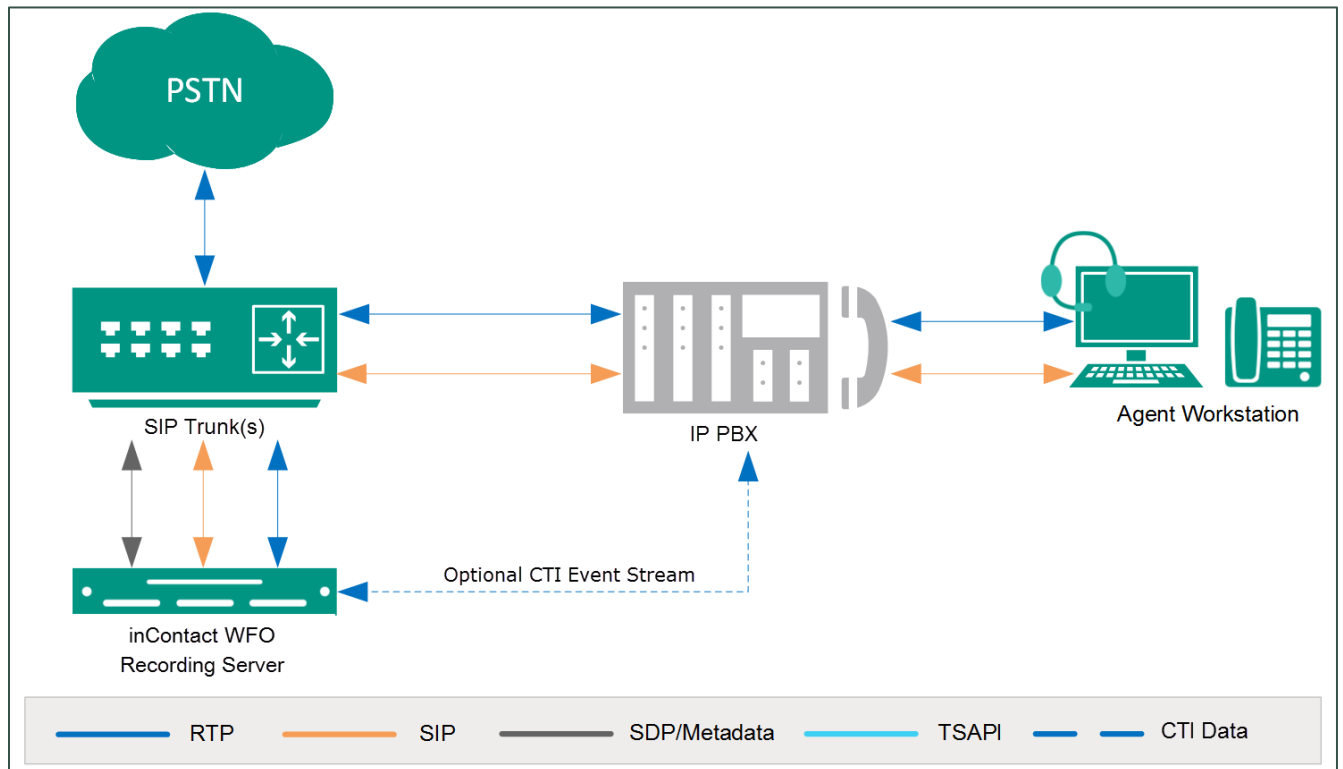
This integration supports live monitoring capability, can be used with Uptivity Screen Recording, and should work with any standard SIP trunk. It has been tested with Cisco Unified Communications Manager and the inContact Cloud Contact Center.

## **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**

## SIP Trunk Integration Overview

Component	Function
<b>SIP Trunk(s)</b>	Provides the audio connection to Uptivity; may also provide call control events in the form of SIP signaling.
<b>IP PBX</b>	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.
<b>NICE Uptivity Recording Server</b>	Receives call control events and business data and provides a CTI interface for recording. The Uptivity server has these responsibilities: <ul style="list-style-type: none"><li>• Sending call start/call stop messages using the Uptivity API</li><li>• Starting and stopping recordings</li><li>• Providing a SIP Trunk endpoint for recording</li><li>• Copying the finished recordings to the Uptivity storage location</li></ul>

### 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.729a
- G.722
- 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

## SIP Trunk Integration Overview

### 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	<a href="#">Configure a SIP Trunk Security Profile for the Cisco Recording Trunk.</a>
2	<a href="#">Configure a SIP Profile for the Cisco Recording Trunk.</a>
3	<a href="#">Create a Cisco SIP Recording Trunk.</a>
4	<a href="#">Configure Cisco Phones.</a>

**Configure a SIP Trunk Security Profile for the Cisco Recording Trunk**

**SIP Trunk Security Profile Information**

Name\*

Description

Device Security Mode

Incoming Transport Type\*

Outgoing Transport Type

Enable Digest Authentication

Nonce Validity Time (mins)\*

X.509 Subject Name

Incoming Port\*

Enable Application level authorization

Accept presence subscription

Accept out-of-dialog refer\*\*

Accept unsolicited notification

Accept replaces header

Transmit security status

Allow charging header

SIP V.150 Outbound SDP Offer Filtering\*

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](#).

## Customer Integration Tasks

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, *, #, and
<input type="checkbox"/> Redirect by Application	
<input type="checkbox"/> Disable Early Media on 180	
<input type="checkbox"/> Outgoing T.38 INVITE include audio mline	
<input type="checkbox"/> Enable ANAT	
<input type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change	
<input type="checkbox"/> Use Fully Qualified Domain Name in SIP Requests	
<input type="checkbox"/> Assured Services SIP conformance	

Parameters used in Phone	
Timer Invite Expires (seconds)*	<input type="text" value="180"/>
Timer Register Delta (seconds)*	<input type="text" value="5"/>
Timer Register Expires (seconds)*	<input type="text" value="3600"/>
Timer T1 (msec)*	<input type="text" value="500"/>
Timer T2 (msec)*	<input type="text" value="4000"/>
Retry INVITE*	<input type="text" value="6"/>
Retry Non-INVITE*	<input type="text" value="10"/>
Start Media Port*	<input type="text" value="16384"/>
Stop Media Port*	<input type="text" value="32766"/>
Call Pickup URI*	<input type="text" value="x-cisco-serviceuri-pickup"/>
Call Pickup Group Other URI*	<input type="text" value="x-cisco-serviceuri-opickup"/>
Call Pickup Group URI*	<input type="text" value="x-cisco-serviceuri-gpickup"/>
Meet Me Service URI*	<input type="text" value="x-cisco-serviceuri-meetme"/>
User Info*	<input type="text" value="None"/>
DTMF DB Level*	<input type="text" value="Nominal"/>
Call Hold Ring Back*	<input type="text" value="Off"/>
Anonymous Call Block*	<input type="text" value="Off"/>
Caller ID Blocking*	<input type="text" value="Off"/>
Do Not Disturb Control*	<input type="text" value="User"/>
Telnet Level for 7940 and 7960*	<input type="text" value="Disabled"/>
Resource Priority Namespace	<input type="text" value="&lt; None &gt;"/>
Timer Keep Alive Expires (seconds)*	<input type="text" value="120"/>
Timer Subscribe Expires (seconds)*	<input type="text" value="120"/>
Timer Subscribe Delta (seconds)*	<input type="text" value="5"/>
Maximum Redirections*	<input type="text" value="70"/>
Off Hook To First Digit Timer (milliseconds)*	<input type="text" value="15000"/>
Call Forward URI*	<input type="text" value="x-cisco-serviceuri-cfwdall"/>
Speed Dial (Abbreviated Dial) URI*	<input type="text" value="x-cisco-serviceuri-abbrdial"/>
<input checked="" type="checkbox"/> Conference Join Enabled <input type="checkbox"/> RFC 2543 Hold <input checked="" type="checkbox"/> Semi Attended Transfer <input type="checkbox"/> Enable VAD <input type="checkbox"/> Stutter Message Waiting <input type="checkbox"/> MLPP User Authorization	

## Customer Integration Tasks

<b>Normalization Script</b>	
Normalization Script < None >	
<input type="checkbox"/> Enable Trace	
Parameter Name	Parameter Value
1	
<b>Incoming Requests FROM URI Settings</b>	
Caller ID DN	
Caller Name	
<b>Trunk Specific Configuration</b>	
Reroute Incoming Request to new Trunk based on*	Never
RSVP Over SIP*	Local RSVP
Resource Priority Namespace List	< None >
<input checked="" type="checkbox"/> Fall back to local RSVP	
SIP Rel1XX Options*	Disabled
Video Call Traffic Class*	Mixed
Calling Line Identification Presentation*	Default
<input type="checkbox"/> Deliver Conference Bridge Identifier	
<input type="checkbox"/> Early Offer support for voice and video calls (insert MTP if needed)	
<input type="checkbox"/> Send send-receive SDP in mid-call INVITE	
<input type="checkbox"/> Allow Presentation Sharing using BFCP	
<input type="checkbox"/> Allow iX Application Media	
<input type="checkbox"/> Allow Passthrough of Configured Line Device Caller Information	
<input type="checkbox"/> Reject Anonymous Incoming Calls	
<input type="checkbox"/> Reject Anonymous Outgoing Calls	
<b>SIP OPTIONS Ping</b>	
<input type="checkbox"/> Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"	
Ping Interval for In-service and Partially In-service Trunks (seconds)*	60
Ping Interval for Out-of-service Trunks (seconds)*	120
Ping Retry Timer (milliseconds)*	500
Ping Retry Count*	6

### 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.



Trunk Information	
Trunk Type*	SIP Trunk
Device Protocol*	SIP
Trunk Service Type*	None(Default)
<input type="button" value="Next"/>	

- Click **Next** to continue.
- Enter a **Device Name**.
- 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.

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
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> 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
<input checked="" type="checkbox"/> PSTN Access	
<input type="checkbox"/> Run On All Active Unified CM Nodes	

## Customer Integration Tasks

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*.

The screenshot shows the 'Outbound Calls' configuration section. It includes several dropdown menus and checkboxes. The 'Calling Line ID Presentation\*' and 'Calling Name Presentation\*' dropdowns are highlighted with a red box and are both set to 'Allowed'. Other visible options include 'Called Party Transformation CSS' (set to '< None >'), 'Use Device Pool Called Party Transformation CSS' (checked), 'Calling Party Transformation CSS' (set to '< None >'), 'Use Device Pool Calling Party Transformation CSS' (checked), 'Calling Party Selection\*' (set to 'Originator'), 'Caller ID DN', 'Caller Name', and 'Redirecting Diversion Header Delivery - Outbound' (unchecked).

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

The screenshot shows the 'SIP Information' configuration section. It includes a 'Destination' section with a table for 'Destination Address', 'Destination Address IPv6', and 'Destination Port'. The 'Destination Address' is set to '1' and 'Destination Port' is set to '5060'. Below this are several dropdown menus: 'MTP Preferred Originating Codec\*' (set to '711ulaw'), 'Presence Group\*' (set to 'Standard Presence group'), 'SIP Trunk Security Profile\*' (set to 'CallCopy SIP Trunk Security Profile'), 'Rerouting Calling Search Space' (set to '< None >'), 'Out-Of-Dialog Refer Calling Search Space' (set to '< None >'), 'SUBSCRIBE Calling Search Space' (set to '< None >'), 'SIP Profile\*' (set to 'CallCopy SIP Profile'), and 'DTMF Signaling Method\*' (set to 'No Preference'). There is also a 'Normalization Script' section with a dropdown set to '< None >' and an 'Enable Trace' checkbox (unchecked).

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**.

The screenshot shows the 'Find and List Phones' page. At the top, there is a '+ Add New' button. Below that, the 'Phone' section is active. The search criteria are set to 'Device Name' and 'begins with'. There is a 'Find' button and a 'Clear Filter' button. Below the search bar, there is a message: 'No active query. Please enter your search criteria using the options above.'

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

The screenshot shows the 'Association Information' dialog box. It has a 'Modify Button Items' button at the top. Below that, there is a list of two lines:

1	Line [1] - 3003 (no partition)
2	Line [2] - Add a new DN

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

## Customer Integration Tasks

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

Line 1 on Device SEP44ADD9BC39C5	
Display (Caller ID)	<input type="text"/>
ASCII Display (Caller ID)	<input type="text"/>
Line Text Label	<input type="text"/>
ASCII Line Text Label	<input type="text"/>
External Phone Number Mask	<input type="text"/>
Visual Message Waiting Indicator Policy*	Use System Policy ▾
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 ▾
<b>Recording Option*</b>	<b>Automatic Call Recording Enabled ▾</b>
Recording Profile	dthomasRecorder ▾
Monitoring Calling Search Space	< None > ▾
<input checked="" type="checkbox"/> Log Missed Calls	

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.