



# Customer Guide to Cisco JTAPI-BiB Integrations

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**Recording**

# Customer Guide to Cisco JTAPI-BiB Integrations

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## Introduction

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### Audience

This document is written for customers and prospective customers interested in using inContact Call Recording in a Cisco JTAPI-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 inContact WFO.

### Goals

The goal of this document is to provide knowledge, reference, and procedural information necessary to understand a proposed Cisco/inContact WFO integration using JTAPI-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 inContact WFO 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 inContact WFO 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 inContact WFO configuration, consult the inContact WFO installation team.

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Cisco JTAPI-BiB can also be used with Cisco UCCE or UCCX. In these scenarios, refer to the *inContact WFO Customer Guide to Cisco UCCE Integrations* or the *inContact WFO Customer Guide to Cisco UCCX Integrations*, as appropriate.

### Terminology

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

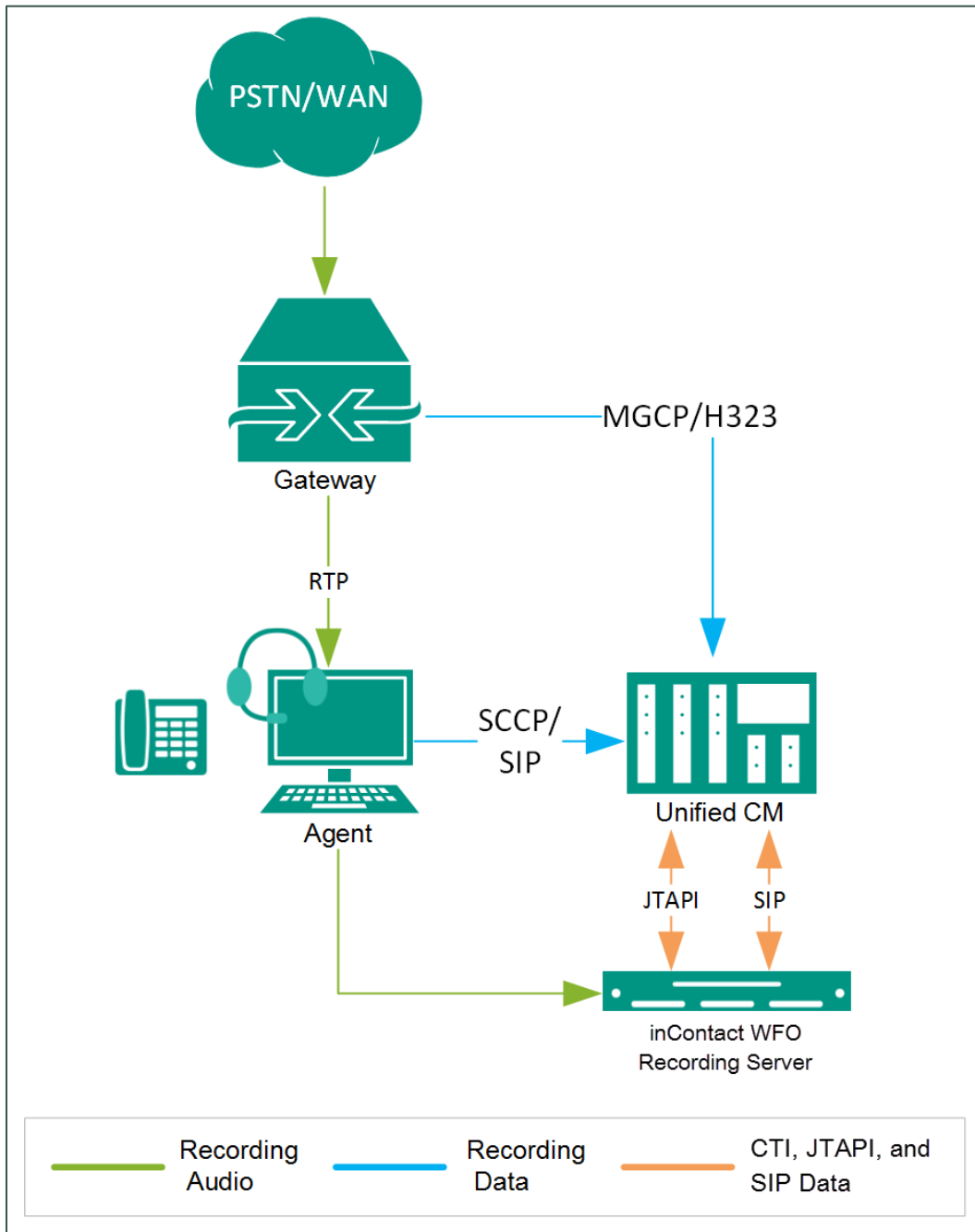
- **CUCM:** Cisco Unified Communications Manager. CUCM is a software-based call-processing system that includes gateways, routers, phones, voicemail boxes, and a variety of other VoIP components. Sometimes referred to as CallManager.
- **UCCE:** Unified Contact Center Enterprise. UCCE delivers intelligent 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.
- **JTAPI:** Java Telephony Application Programming Interface. Cisco JTAPI 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, inContact WFO).

### Customer Responsibilities

You are responsible for supplying the physical and/or IP connection(s) 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 JTAPI-BiB Integration Overview

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



General architectural example of the Cisco JTAPI-BiB integration

## Customer Guide to Cisco JTAPI -BiB Integrations

Component	Function
<b>Voice Gateway</b>	Connects the customer network to the public network.
<b>Cisco JTAPI</b>	When a call is placed to or received by a monitored device, inContact WFO receives the event via JTAPI and issues a record start message to the CUCM, also using JTAPI.
<b>Cisco Unified Communications Manager (UCM)</b>	The CUCM negotiates audio stream network ports and codecs between the phone and inContact WFO using SIP on the inContact WFO side and SCCP or SIP to the phone. Audio is redirected to the inContact WFO recording server through a SIP Trunk.
<b>inContact WFO Application Server</b>	Receives call control events and business data and provides a CTI interface to the inContact WFO recording node.  The inContact WFO server has these responsibilities: <ul style="list-style-type: none"><li>• Sending call start/call stop messages using the inContact WFO API.</li><li>• Starting and stopping recordings using JTAPI.</li><li>• Providing a SIP Trunk endpoint the CUCM uses for recording.</li><li>• Copying finished recordings to the inContact WFO storage location.</li></ul>
<b>Third Generation Phones</b>	Each third generation phone being recorded uses built-in bridge to forward audio streams for each side of the call to inContact WFO.

## Known Limitations

- This integration requires Cisco device firmware version SCCP42.9-2-1S or higher. Firmware can typically be obtained from Cisco's software download site.
- Monitoring and recording the calls of secure-capable agents are not allowed. This limitation is imposed by the CUCM. (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 inContact WFO 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

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The features utilized in this method of recording require third generation phones that have Built-in Bridge capability (BiB). 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 any questions, consult your Cisco account management team to determine whether your telephone sets have this capability.

### Software

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- Cisco Unified Communications Manager v9.0 – v10.5(2)
- Cisco JTAPI Client installed on the inContact WFO system

### Licensing

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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 Cisco Unified Communications Manager to determine capacity. The integration utilizes the JTAPI and BiB capabilities for each recorded phone.

## inContact WFO Requirements

### Network

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

### Hardware

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inContact WFO hardware requirements vary depending on system configurations. Appropriate hardware is identified during the system implementation process. For additional information, see *Customer Site Requirements for inContact WFO*.

### Software

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- inContact WFO v5.6 or later

Additional third-party software is required for this integration:

- CACE WinPcap version 4.1.x (available from the WinPcap website)
- Java Development Kit for Windows x86, version 6u45

### Licensing

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- 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 JTAPI-BiB integrations. Links are provided for tasks that are covered in this guide.

Customer Configuration Steps for Cisco JTAPI-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	<a href="#">Identify Phones that Support Recording</a> . This step is to verify that all desired recording locations have an appropriate phone.
4	<a href="#">Configure a JTAPI User Account for inContact WFO</a> .
5	<a href="#">Configure a SIP Trunk Security Profile for the Recording Trunk</a> .
6	<a href="#">Configure a SIP Profile for the Recording Trunk</a> .
7	<a href="#">Create the SIP Recording Trunk</a> .
8	<a href="#">Create a Route Pattern for the SIP Recording Trunk</a> .
9	<a href="#">Create a Recording Profile</a> .
10	For all phones to be recorded, <a href="#">Enable Built-in-Bridge, Disable Privacy</a> .
11	For all phones to be recorded, <a href="#">Add Recording Option and Recording Profile to Line Appearances</a> . Tell your inContact WFO Installation team which Recording Option you

## Customer Integration Tasks

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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 manuals and/or guides 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 inContact WFO installation team.

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

inContact WFO 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 JTAPI-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 inContact WFO deployments which experience initial errors do so because of a Cisco setting being missed.

If you are combining the JTAPI-BiB integration with an additional integration like Cisco UCCE or Cisco 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*.

The screenshot displays the Cisco Unified Reporting application interface. The main content area shows a success message: "OK: Report generated successfully." Below this is the "Unified CM Phone Feature List" report, which provides a complete list of features available to products supported by Unified CM. The report is created on Wed Nov 06 13:56:15 EST 2013. The interface includes a navigation menu on the left, a main content area with a success message, and a table of device features.

Product	Protocol	Feature	Parameters
Cisco 6911	SCCP	Record	
Cisco 6921	SCCP	Record	
Cisco 6941	SCCP	Record	
Cisco 6949	SCCP	Record	
Cisco 6961	SCCP	Record	
Cisco 7906	SCCP	Record	
Cisco 7910	SCCP	Record	
Cisco 7911	SCCP	Record	
Cisco 7921	SCCP	Record	
Cisco 7925	SCCP	Record	
Cisco 7926	SCCP	Record	
Cisco 7931	SCCP	Record	
Cisco 7937	SCCP	Record	

The Cisco Unified Reporting application can be used to generate a complete list of devices that support monitoring and recording for a particular release and device pack. To generate this list:

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.

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5. For **Feature**, select Record from the drop-down list.
6. Click the **Submit** button.

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 JTAPI User Account for inContact WFO

A user account must be created on the CUCM for inContact WFO to use to connect and receive JTAPI events for phones. This user account must be configured to monitor all devices that you wish 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).

inContact WFO's 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 JTAPI 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.

To configure a JTAPI user:

1. Log in to the **Cisco Unified CM Administration** portal with an administrative account.
2. From the menu bar, select the **User Management** menu and click **Application User**.
3. On the Find and List Application Users page, click the **Add New** button.
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 inContact WFO Installation team.
5. Under **Device Information**, move any devices that inContact WFO will record to the **Controlled Devices** panel.

**Device Information**

Available Devices

- SEP001380C29B23
- SEP001E4A924C13
- SEP001EF7C3F62B
- SEP001F6C810D39
- SEP44ADD9BC39EE

Controlled Devices

- SEP001280E52C21
- SEP00137F0031C8
- SEP00137F0031C9
- SEP001646CB51B2
- SEP001E7AC340E0

Find more Phones

Find more Route Points

- Under **Permissions Information**, click **Add to Access Control Group**.
- Select the appropriate groups and click **Add Selected**.

<input type="checkbox"/>	Name ^	Roles	Copy
<input checked="" type="checkbox"/>	Standard CTI Allow Call Monitoring		
<input checked="" type="checkbox"/>	Standard CTI Allow Call Park Monitoring		
<input checked="" type="checkbox"/>	Standard CTI Allow Call Recording		
<input checked="" type="checkbox"/>	Standard CTI Allow Calling Number Modification		
<input checked="" type="checkbox"/>	Standard CTI Allow Control of All Devices		
<input checked="" type="checkbox"/>	Standard CTI Allow Control of Phones supporting Connected Xfer and conf		
<input checked="" type="checkbox"/>	Standard CTI Allow Control of Phones supporting Rollover Mode		
<input type="checkbox"/>	<u>Standard CTI Allow Reception of SRTP Key Material</u>		
<input checked="" type="checkbox"/>	Standard CTI Enabled		
<input type="checkbox"/>	<u>Standard CTI Secure Connection</u>		

Select All Clear All Add Selected Close

## Configure a SIP Trunk Security Profile for the Recording Trunk

The screenshot shows a configuration form titled "SIP Trunk Security Profile Information". The form contains the following fields and options:

- Name\*: CallCopy SIP Trunk Security Profile
- Description: CallCopy SIP Trunk Security Profile
- Device Security Mode: Non Secure (dropdown)
- Incoming Transport Type\*: TCP+UDP (dropdown)
- Outgoing Transport Type: UDP (dropdown)
- Enable Digest Authentication
- Nonce Validity Time (mins)\*: 600
- X.509 Subject Name: (empty)
- Incoming Port\*: 5060
- Enable Application Level Authorization
- Accept Presence Subscription
- Accept Out-of-Dialog REFER\*\*
- Accept Unsolicited Notification
- Accept Replaces Header
- Transmit Security Status

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.

To configure a SIP trunk security profile:

1. Log in to the **Cisco Unified CM Administration** portal with an administrative account.
2. From the menu bar, select the **System** menu, then **Security**, and click **SIP Trunk Security Profile**.
3. Click **Add New** and enter a **Name** and a **Description** for this trunk.
4. For **Device Security Mode**, select **Non Secure** from the drop-down list.
5. For **Incoming Transport Type**, select **TCP+UDP** from the drop-down list.
6. For **Outgoing Transport Type**, select **UDP** from the drop-down list.
7. Do not select (or clear if selected) the check box 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 included in this section. If any settings in your SIP profile do not match, discuss this with your inContact WFO Installation team.

To configure a SIP profile for the recording trunk:

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

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	

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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
<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	

<b>Normalization Script</b>							
Normalization Script	< None >						
<input type="checkbox"/> Enable Trace							
	<table border="1"> <thead> <tr> <th></th> <th>Parameter Name</th> <th>Parameter Value</th> </tr> </thead> <tbody> <tr> <td>1</td> <td><input type="text"/></td> <td><input type="text"/></td> </tr> </tbody> </table>		Parameter Name	Parameter Value	1	<input type="text"/>	<input type="text"/>
	Parameter Name	Parameter Value					
1	<input type="text"/>	<input type="text"/>					
<b>Incoming Requests FROM URI Settings</b>							
Caller ID DN	<input type="text"/>						
Caller Name	<input type="text"/>						
<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 the SIP Recording Trunk

The audio streams to be recorded will be routed to inContact WFO over a SIP trunk configured on the CUCM. Except where specifically noted, all settings should match 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 inContact WFO Installation team.

To create the SIP recording trunk:

1. Log in to the **Cisco Unified CM Administration** portal with an administrative account.
2. Select the **Device** menu and click **Trunk**.
3. Click **Add New**.

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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 inContact WFO server on which the CTI Core for this integration is running.
10. In **Destination Port** in the **SIP Information**, enter 5060.
11. In **SIP Trunk Security Profile** in the **SIP Information**, enter the name of the profile this trunk will use (see [Configure a SIP Trunk Security Profile for the Recording Trunk](#)).
12. In **SIP Profile** in the **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**.

<b>Device Information</b>	
Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	<input type="text" value="CallCopyRecorder"/>
Description	<input type="text"/>
Device Pool*	<input type="text" value="Default"/>
Common Device Configuration	<input type="text" value="&lt; None &gt;"/>
Call Classification*	<input type="text" value="Use System Default"/>
Media Resource Group List	<input type="text" value="&lt; None &gt;"/>
Location*	<input type="text" value="Hub_None"/>
AAR Group	<input type="text" value="&lt; None &gt;"/>
Tunneled Protocol*	<input type="text" value="None"/>
QSIG Variant*	<input type="text" value="No Changes"/>
ASN.1 ROSE OID Encoding*	<input type="text" value="No Changes"/>
Packet Capture Mode*	<input type="text" value="None"/>
Packet Capture Duration	<input type="text" value="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*	<input type="text" value="When using both sRTP and TLS"/>
Route Class Signaling Enabled*	<input type="text" value="Default"/>
Use Trusted Relay Point*	<input type="text" value="Default"/>
<input checked="" type="checkbox"/> PSTN Access	
<input type="checkbox"/> Run On All Active Unified CM Nodes	

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**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 >	<input checked="" type="checkbox"/>

**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

1\* Destination Address Destination Address IPv6 Destination Port 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

1 Parameter Name Parameter Value

## 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. This pattern cannot require a forced authorization code or inContact WFO will not receive call audio.

To configure this feature:

1. Log in to the **Cisco Unified CM Administration** portal with an administrative account.
2. Select the **Call Routing** menu, then select **Route/Hunt**, and click on **Route Pattern**.
3. Click **Add New**.
4. Enter a **Route Pattern** number.
5. Set **Gateway/Route List** to the SIP trunk.
6. Do not select (or clear if selected) the check box for **Require Forced Authorization Code**.
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

Put your section name here	
Name*	CallCopy Recorder
Recording Calling Search Space	< None >
Recording Destination Address*	7778
Save	

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 the **Device** menu, then select **Device Settings** and click **Recording Profile**.
3. Click **Add New** and enter a **Name**.
4. For **Recording Calling Search Space**, select None, Default, or the appropriate search space from the drop-down list.

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5. For **Recording Destination Address**, enter the Route Pattern assigned to the SIP recording trunk.
6. Click **Save**.

### Enable Built-in-Bridge, Disable Privacy

Built-in-Bridge and Privacy mode can be configured at both the device and server level. The instructions below are for devices. If these settings are configured at the server level, the device-level settings must be set to Default. 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.

On 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.

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 inContact WFO system for recording. If desired, you can 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.

Device Information	
Registration	Registered with Cisco Unified Communications Manager cucm85
IP Address	<a href="#">10.100.10.63</a>
Active Load ID	SCCP42.9-1-1SR15
<input checked="" type="checkbox"/> Device is Active	
<input checked="" type="checkbox"/> Device is trusted	
MAC Address*	001E4A924C13
Description	SEP001E4A924C13
Device Pool*	Default <a href="#">View Details</a>
Common Device Configuration	< None > <a href="#">View Details</a>
Network Locale	< None >
<b>Built In Bridge*</b>	<b>On</b>
<b>Privacy*</b>	<b>Off</b>
Device Mobility Mode*	Default <a href="#">View Current Device Mobility</a>

To configure this feature for individual phones:

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



3. 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.
4. From the resulting list of phones, click the **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**. Your inContact WFO Installation team needs to know which Recording Option you are using.

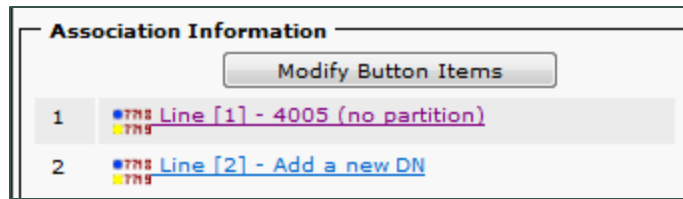
An alternate way of locating Directory numbers is to select the **Call Routing** menu, then click the **Directory Number** option. If desired, you can also use Bulk Administration to schedule a job that will update multiple line appearances at once.

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

To configure recording options and set the recording profile for a line appearance:

1. Log in to the **Cisco Unified CM Administration** portal with an administrative account.
2. Select the **Device** menu and click the **Phone** link.
3. 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.
4. From the resulting list of phones, click the **Device Name** to edit the **Phone Configuration**.
5. Click the **Line Appearance** to be recorded under the **Association Information** section.

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6. Enable the desired **Recording Option** in the **Line Settings** section.
7. Set **Recording Profile** to the desired profile.
8. Set **Recording Media Source** to **Phone Preferred** and click **Save**.

The screenshot shows the configuration page for "Line 1 on Device SEP001E4A924C13". The page contains various settings for the line, including display text, policies, and recording options. The recording options are highlighted:

- Recording Option\*: Selective Call Recording Enabled
- Recording Profile: QA
- Recording Media Source\*: Phone Preferred

The "Log Missed Calls" checkbox is checked.

Repeat this task for any lines that will be recorded.

## Customer Administration Tasks

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During ongoing use of the system, your inContact WFO administrator may need to configure new channels or reconfigure existing channels. This integration requires changes to the Voice Boards page in the inContact WFO Web Portal only when channels are added.

### Voice Boards Overview

Voice Boards control how inContact WFO acquires audio. This component provides *what* inContact WFO is to record. At least one Voice Board is required for most integrations. While Voice Boards can correspond to physical audio capture boards in some integrations, they are not those boards.

inContact WFO uses per-channel licensing, and each Voice Board software component maintains the count of licensed, used and available channels associated with it. The system will not use any Voice Boards or channels for which it is not licensed.

### Voice Board Configuration

The Cisco JTAPI-BiB integration streams audio from the phone to inContact WFO, 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 inContact WFO Voice Board.

If channels are added to your system, you must increase the channel count on the associated Voice Board. You must restart the Recorder service (cc\_cticore.exe) after any changes to Voice Boards or channels.

Any other Voice Board changes should only be done under direct supervision from inContact WFO 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.

## Document Revision History

Revision	Change Description	Effective Date
0	Initial version for this release	2016-04-05