



Monitoring and Recording

This chapter provides information about Silent Call Monitoring and Call Recording.

Call centers need to be able to guarantee the quality of customer service that an agent in a call center provides. To protect themselves from legal liability, call centers need to be able to archive agent-customer conversations.

The Silent Call Monitoring feature allows a supervisor to eavesdrop on a conversation between an agent and a customer; neither the agent nor the customer can hear the supervisor voice.

The Call Recording feature allows system administrators or authorized personnel to archive conversations between the agent and the customer.

Cisco Unified Communications Manager supports Silent Call Monitoring only within a single cluster. Call Recording is supported within multi-cluster environments.

Silent call monitoring specifies one of the three call monitoring modes. (The other modes specify whisper call monitoring and active call monitoring.) In whisper monitoring mode, the supervisor can listen to the conversation between an agent and a customer and can talk to the agent without making the customer aware of the presence of the supervisor. In active monitoring mode, the supervisor can participate fully in the conversation between the agent and the customer.

The Silent Monitoring and Call Recording features specify generic features in Cisco Unified Communications Manager. Cisco makes these features available to any deployment or installation where monitoring- and recording-enabled applications are available. Descriptions in this document use terms such as supervisor, agent, and customer to see the parties that participate in call monitoring and recording sessions.

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- [Recording, on page 20](#)
- [Simultaneous Monitoring and Recording, on page 55](#)

Silent Monitoring

Customers need to guarantee the quality of customer service provided by their employees who service their customers, such as call center agents. Customers also need to meet compliance, guidelines and protect themselves from legal liability. These requirements are satisfied by monitoring and recording agent-customer conversations.

The silent call monitoring feature allows a supervisor to eavesdrop on any conversation; neither of the call participants can hear the supervisor voice.

Terminology

This document uses the following terms to discuss call monitoring:

Agent

The call party that is monitored.

Customer

Any call participant other than the agent or the supervisor.

Local stream

The media stream from agent to customer.

Remote stream

The media stream from customer to agent.

Supervisor

The person who is silently monitoring the call.

Supervisor desktop application

A silent monitoring-enabled application that is used to invoke a monitoring session.

Silent monitoring

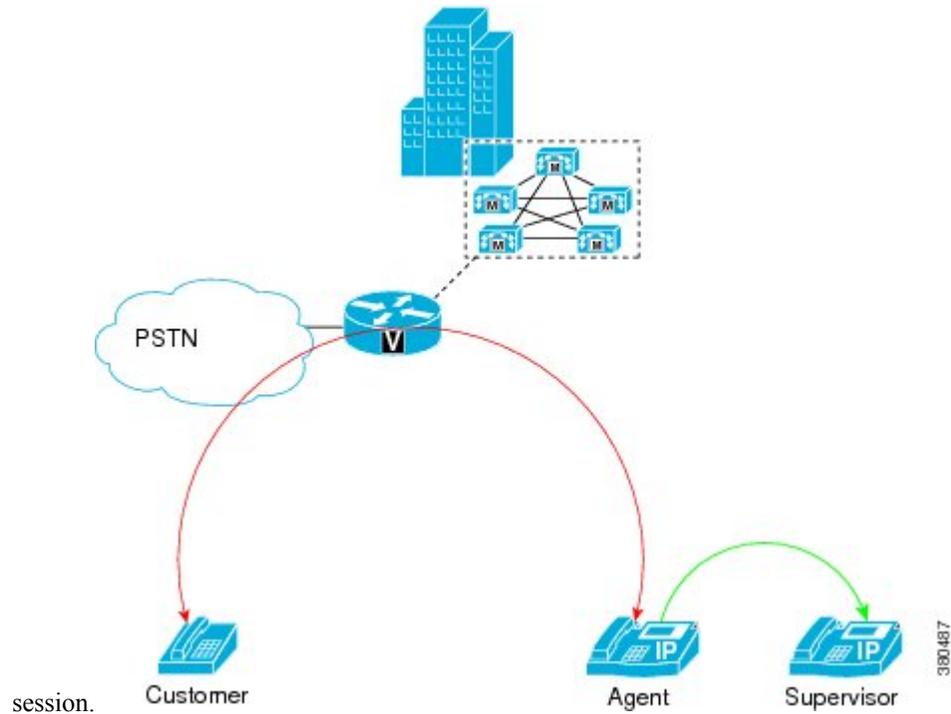
The feature which allows a monitoring party (a supervisor) to listen to a conversation between a near-end party (an agent) and a far-end party (a customer); neither the agent nor the customer hears the monitoring party voice.

Introduction

With silent call monitoring, the supervisor can listen in on any call for quality control and performance evaluation. By default, the agent is not aware of the monitoring session. In IP phone-based silent call monitoring, the monitoring stream comprises a mix of the customer voice and the agent voice. Only a CTI application can trigger a silent monitoring session.

Figure 1: Silent Call Monitoring Session

The following figure shows a typical monitoring



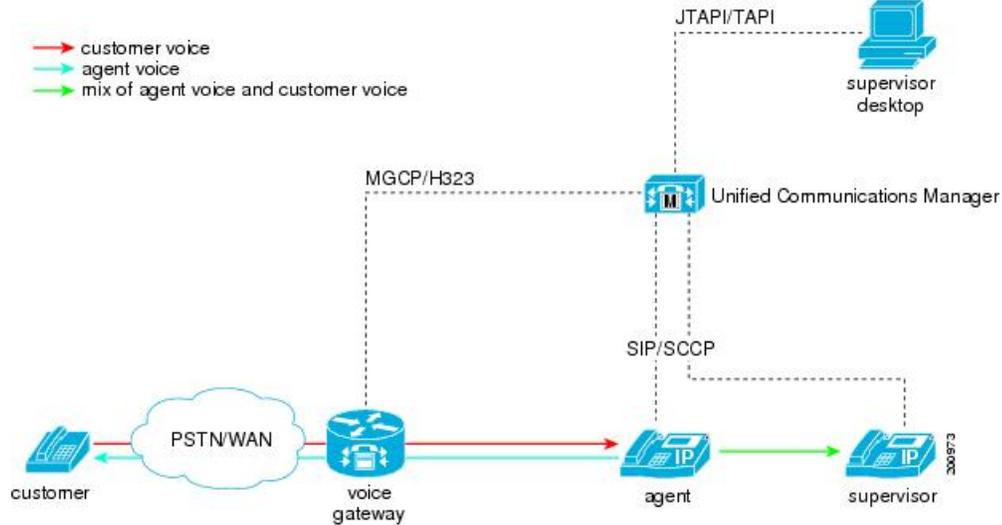
Architecture

Cisco Unified Communications Manager uses an IP phone-based architecture to provide call monitoring whereby the agent phone mixes the agent voice with the customer voice and sends a single, combined, or mixed stream of both voices to the supervisor phone.

Figure 2: IP Phone-Based Architecture for Monitoring

The following figure illustrates the IP phone-based architecture for monitoring. In the figure, the blue lines indicate the agent voice stream, the red lines indicate the customer voice stream, and the green line indicates

the mix of customer and agent voice streams that gets sent to the supervisor.



Applications can initiate monitoring through the JTAPI or TAPI interfaces. Many Cisco applications, such as Cisco Unified Contact Center Enterprise and Cisco Unified Contact Center Express have the ability to use the silent monitoring feature.

Monitoring exhibits the following characteristics:

- Silent monitoring is call based; the supervisor selects a specific call on a line appearance of an agent phone to be monitored.
- The start-monitoring request from the application triggers the supervisor phone to go off hook automatically to make a special monitoring call to the agent.
- The agent phone automatically answers the monitoring call. The monitoring call is not presented to the agent.

The following requirements apply:

- The CTI application user needs to be a member of the Standard CTI Allow Call Monitoring user group.
- The agent device needs to be in the CTI application user's controlled device list.

For a monitoring call, the supervisor phone displays “From Monitoring [agent username/DN]”.

One-Way Media

A monitoring call comprises one-way media from an agent phone to a supervisor phone.

Monitoring calls go through normal call admission control.

Network Address Translation (NAT) separating agent and supervisor or customer remains transparent within the limitations of Cisco Unified Communications Manager.

Firewall Considerations

Firewall software needs to know the destination IP address and destination port as well as the source IP address to allow the RTP streams.

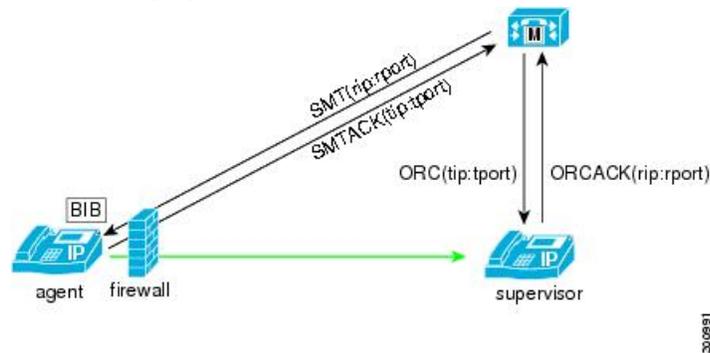
Be aware that SCCP messages for media are not symmetric; SIP is acceptable.

The SCCP version 12 enhancement for one-way media provides the following additions:

- New StartMediaTransmissionAck (SMTACK) message with transmission IP and port
- OpenReceiveChannel (ORC) with additional transmission IP and port

Figure 3: One-Way Media and Firewall

The following figure illustrates one-way media and a firewall.



Codec Selection

The agent phone and the supervisor phone negotiate the codec for the monitoring call using Cisco Unified Communications Manager region settings.

Call Preservation

If the agent call that is being monitored goes to call preservation, Cisco Unified Communications Manager puts the monitoring call into call preservation mode.

The agent call is not affected if the monitoring call goes to call preservation mode.

Notification Tones

In certain jurisdictions, a notification tone must be played to either the agent, customer, or both indicating the call is being monitored.

Use the following service parameters to set the default notification-tone options:

- Play Monitoring Notification Tone To Observed Target (agent)
- Play Monitoring Notification Tone To Observed Connected Parties (customer)

CTI applications can specify the notification tone option in the monitoring request.

Play Tone Behavior

The following table specifies the behavior of tones during a monitoring session.

Table 1: Play Tone Behavior

Play To	Agent Hears	Customer Hears	Supervisor Hears
None	Nothing	Nothing	Nothing
Agent	Tone	Nothing	Nothing
Customer	Nothing	Tone	Nothing
Both	Tone	Tone	Nothing

Silent Monitoring Usage Scenarios

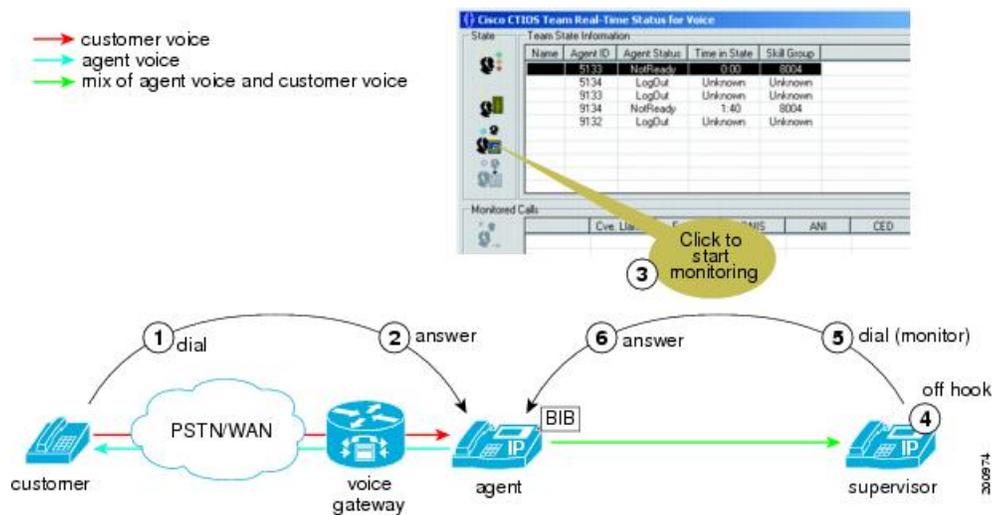
The following section provides usage scenarios for Call Monitoring.

Invocation of a Silent Monitoring Session

A supervisor can initiate a silent monitoring session by using a desktop application after the agent answers a call.

The following figure illustrates a silent monitoring session.

Figure 4: Silent Monitoring Session



When the supervisor initiates a monitoring session, the following steps take place:

1. The customer calls into the call center. The call gets routed to the agent.
2. The agent answers the call. A two-way media stream is set up between the agent IP phone and the customer.
3. The supervisor selects the agent from the desktop application, and then clicks Monitoring.
4. The supervisor phone automatically goes off hook.
5. The supervisor phone makes a monitoring call to the agent.

- The built-in bridge (BIB) of the agent phone automatically accepts the monitoring call. The agent phone starts to mix media of the agent voice and the customer voice and sends the mix to the supervisor phone.

The supervisor can transfer the monitoring call anywhere after the monitoring call is initiated.

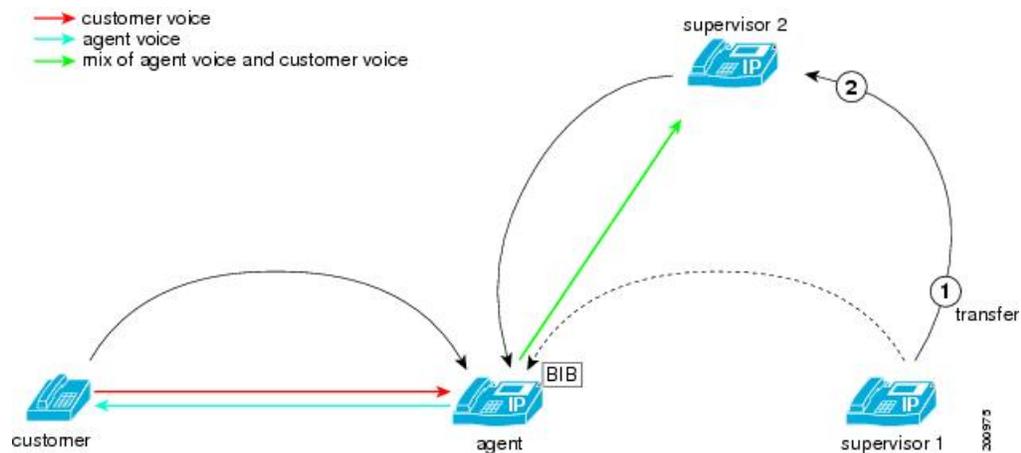
The supervisor can stop monitoring the call anytime after the call starts, either through the application or simply by hanging up.

The supervisor can put the monitoring call on hold (no MOH gets inserted) and resume the monitoring call from the same or a different device.

Supervisor Transfers the Monitoring Call

The following figure illustrates the supervisor transfer of a monitoring call.

Figure 5: Supervisor Transfers the Monitoring Call



During a monitoring call, the supervisor transfers the monitoring call, and the following steps take place:

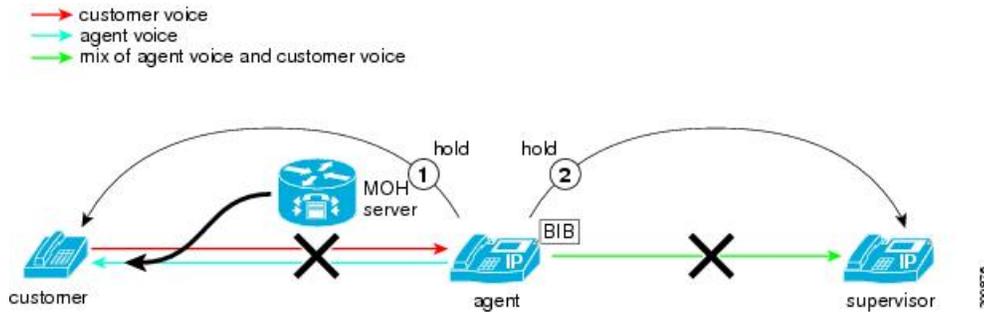
- Supervisor 1 presses the Transfer softkey and dials the phone number of supervisor 2.
- Supervisor 2 answers the call.
- Supervisor 1 completes the transfer by pressing the Transfer softkey again.
- The monitoring call transfers to supervisor 2. Supervisor 2 starts to receive the mix of the agent voice and the customer voice.

Agent Cannot Control a Monitoring Call

The agent does not have direct control over the monitoring call; however, the agent action on the primary call causes a corresponding action for the monitoring call.

The following figure illustrates the scenario where the agent puts the customer on hold while the supervisor is monitoring the agent.

Figure 6: Agent Does Not Control the Monitoring Call



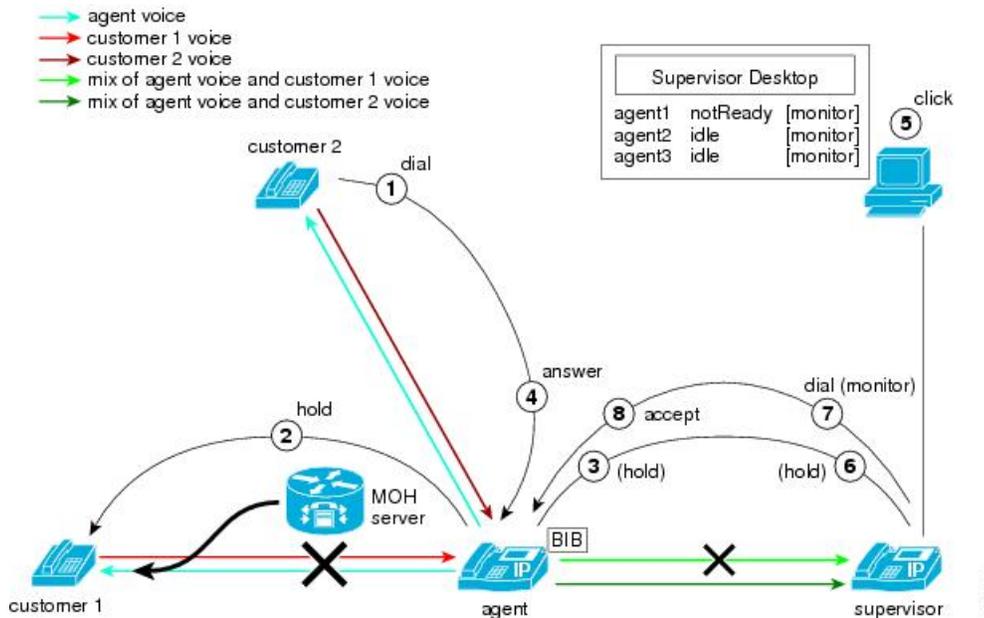
While an agent is being monitored, the agent puts the customer on hold, and the following steps take place:

1. The agent puts the customer on hold. MOH gets inserted and played to the customer.
2. Cisco Unified Communications Manager automatically puts the supervisor on hold. No MOH gets inserted to the supervisor.

Multiple Monitoring Sessions

The following figure illustrates the call flows during multiple monitoring sessions.

Figure 7: Multiple Monitoring Sessions



During multiple monitoring sessions, the following steps take place:

1. Customer 2 calls the agent while the agent is in a call with customer 1 and supervisor is monitoring the agent call with customer 1.
2. The agent puts customer 1 on hold; MOH gets inserted to customer 1.
3. Cisco Unified Communications Manager puts the supervisor on hold. No MOH gets inserted to the supervisor.

4. The agent answers the call from customer 2.
5. The supervisor initiates a second monitoring request for the agent call with customer 2.
6. The supervisor phone goes off hook and makes the second monitoring call to the agent.
7. The agent IP phone (BIB of the agent IP phone) automatically accepts the monitoring call. The mix of agent voice and customer 2 voice gets sent to the supervisor phone.

Barging or Monitoring an Agent Call

If the agent call is being monitored, the barge-in call from a shared line fails.

If the agent call has already been barged, the monitoring request gets rejected with a No Resource Available error.

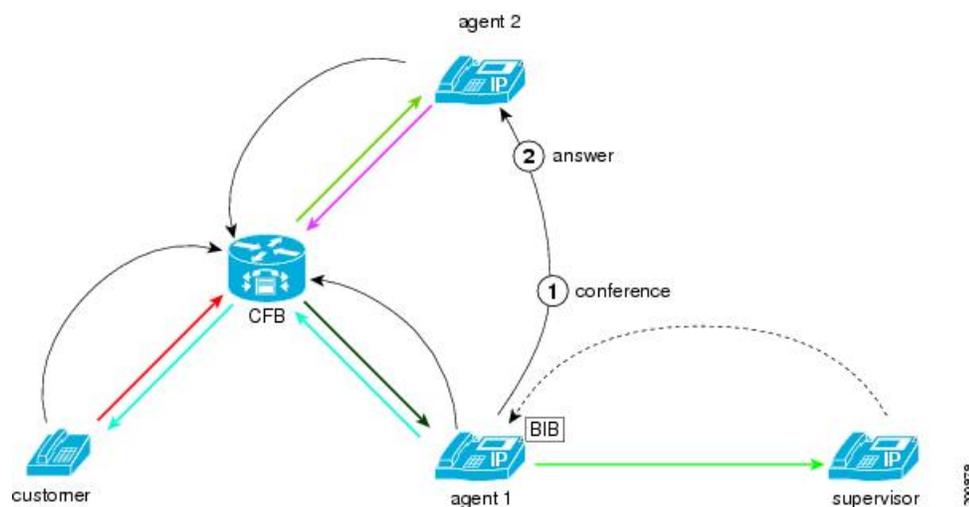
Monitoring an Agent in a Conference

An agent in a call center sometime needs to bring in another party into the conversation with the customer.

The following figure illustrates a case where agent 1 starts an ad hoc conference to include agent 2 in the conversation with the customer. The supervisor for agent 1 monitors the original call with the customer.

During the setting-up process, the media of the monitoring call briefly disconnect. After the conference completes, the supervisor can hear all the parties that are included in the conference.

Figure 8: Monitoring an Agent in a Conference



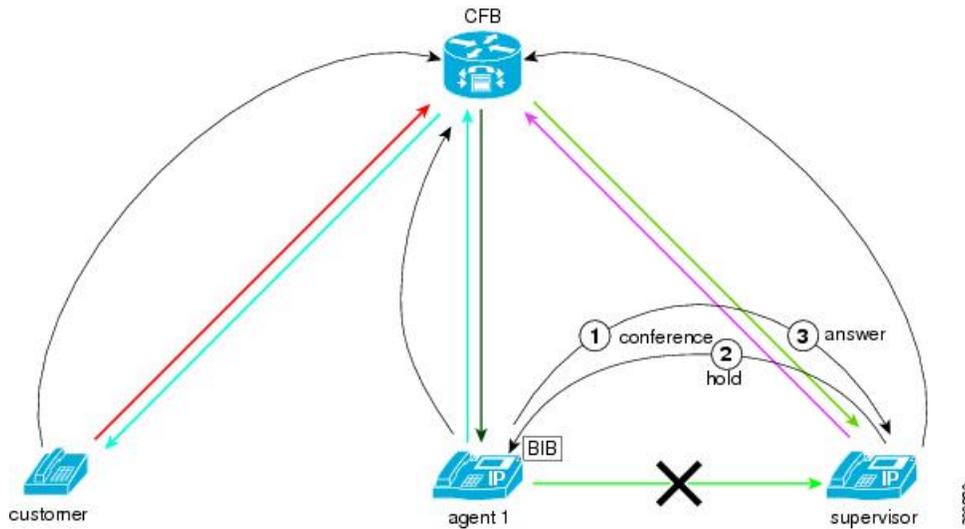
Agent Conferences in the Supervisor

The agent may create a conference with the supervisor while that supervisor is monitoring that agent.

The supervisor must put the monitoring call on hold before joining the conference.

This figure shows the final connection when the supervisor puts the monitoring call on hold and joins the conference. The monitoring session remains in the Hold state while the supervisor participates in the conference. After the supervisor leaves the conference, the supervisor can then resume the monitoring session.

Figure 9: Agent Conferences in the Supervisor

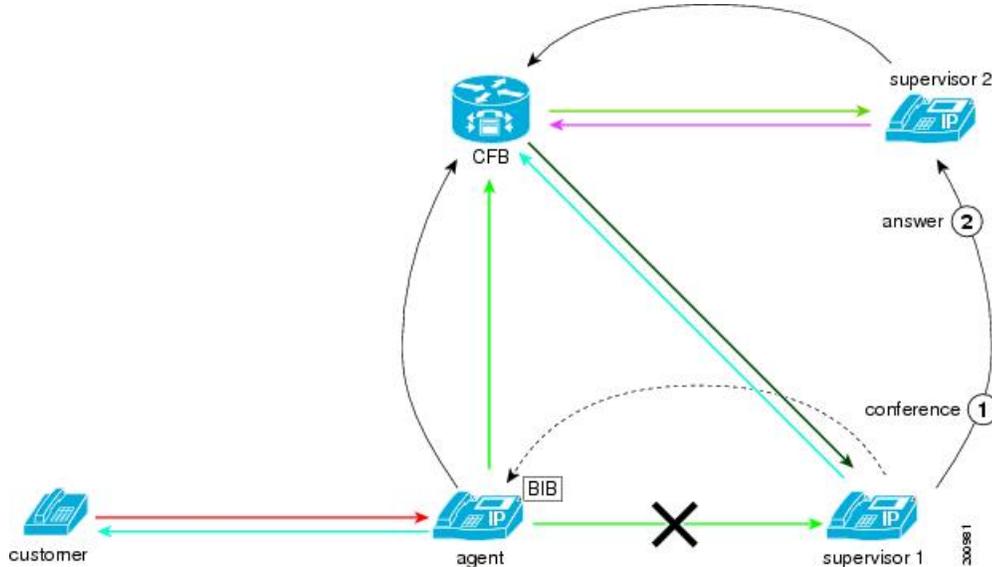


Supervisor Conferences In Another Supervisor

A supervisor can conference in another supervisor for the monitoring session.

The following figure illustrates this scenario.

Figure 10: Supervisor Conferences In Another Supervisor



In the example shown, supervisor 1, who started a monitoring call to the agent, conferences in supervisor 2 to the monitoring call. The customer and agent can still hear each other and are not aware of either of the monitoring supervisors. Both supervisor 1 and the supervisor 2 can hear the conversation of the agent with the customer. The two supervisors can hear each other.

Whisper Coaching

Whisper coaching allows supervisors to talk to agents during a monitoring session. To start a whisper coaching session, a monitoring session must already be in progress. A whisper coaching session is initiated by any CTI application (JTAPI/TAPI), such as Cisco Unified Contact Center Enterprise or Cisco Unified Contact Center Express.

Supervisors can toggle between monitoring and whisper coaching sessions as needed using a CTI application.

To enable whisper coaching, follow the instructions to enable monitoring. No additional configuration is needed for whisper coaching.

Secure Silent Monitoring

Secure silent monitoring allows encrypted media (sRTP) calls to be monitored. Monitoring calls are always established using the highest level of security determined by the capabilities of the agent phone regardless of the security status of the call being observed. The highest level of security is maintained by exchanging the secure media key in any call between the customer, agent, and supervisor. Monitoring calls using secured media carries approximately 4000 bits per second of additional bandwidth overhead, same as standard secure media (sRTP) calls.

If the agent phone has encryption enabled, then the supervisor phone must have encryption enabled to silently monitor agent calls. If the agent phone has encryption enabled, but the supervisor phone does not, then requests to monitor the agent are rejected.

Secure Silent Monitoring and Transcoders

Cisco Unified Communications Manager transcoders do not support encrypted media. If a transcoder is required to establish the secure silent monitoring session between supervisor and agent, and the agent's phone is enabled for encrypted media, the monitoring session is not allowed. A tone is played to the supervisor and cause code 57 (Bearer capability not authorized) is used to disconnect the call. The CTI application notifies the supervisor that the monitoring session was disconnected.

Secure Silent Monitoring Icons

The following table describes the secure icons.

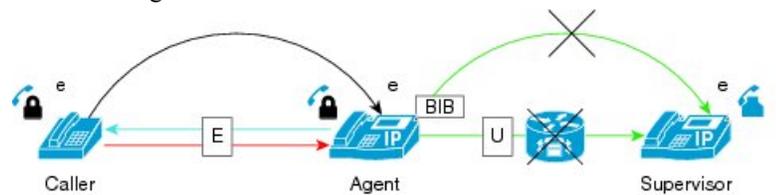


Table 2: Secure Silent Monitoring Icon Definitions

Icon	Description
e	Indicates that both devices can make and receive secure calls.
E	Indicates that the call media is encrypted using sRTP.
Lock Symbol	Indicates that the call is secure.

Secure Silent Monitoring Usage Scenarios

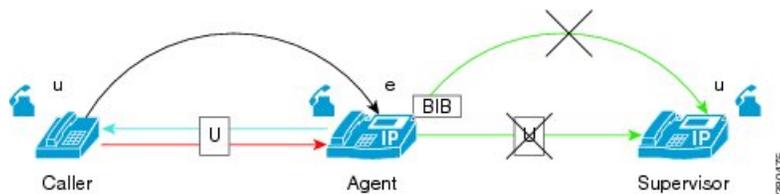
Unsecure Silent Monitoring Call

Supervisor's phone is enabled for secure media, but the agent is not, so the silent monitoring call is unsecure.



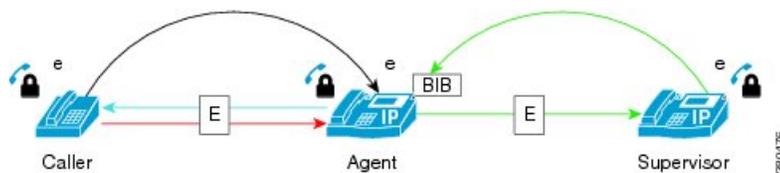
Secure Silent Monitoring Call

Supervisor's phone is not enabled for secure media, but the agent phone is enabled for secure media, so the silent monitoring call is rejected. Supervisor's phone does not meet or exceed security capabilities of agent device.



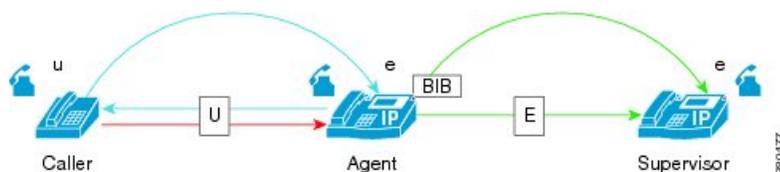
Encrypted Silent Monitoring Call

All phones support secure media, so all calls are encrypted.



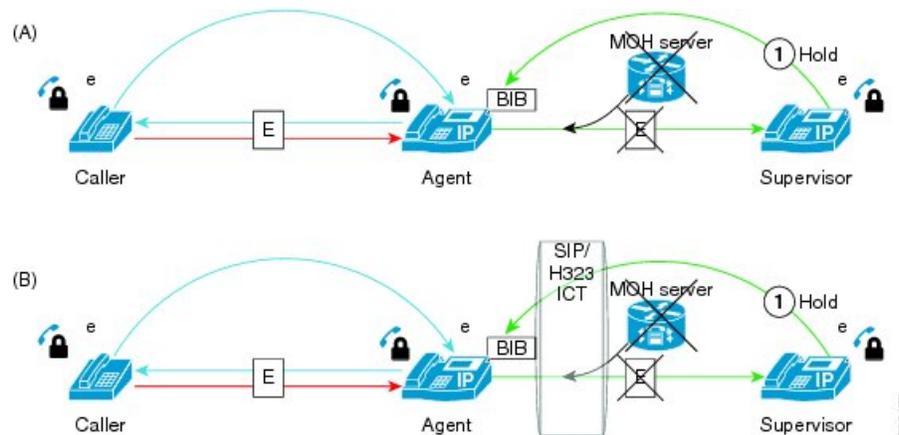
Unencrypted Silent Monitoring Call

Agent and supervisor phones both support secure media, so the silent monitoring call is encrypted, while the customer-agent call is unencrypted.



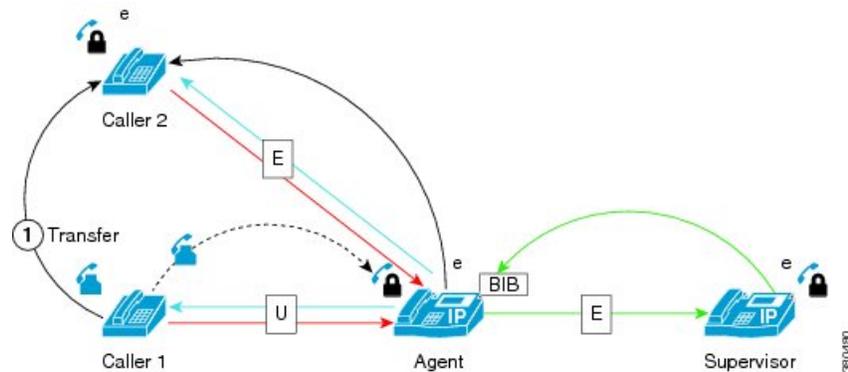
Supervisor Places Monitoring Call On Hold

Supervisor placing the monitoring call on hold has no affect on the caller-agent call. Monitoring session resumes when the supervisor resumes the call.



Call Security Upgrades as a Result of the Caller Transfer

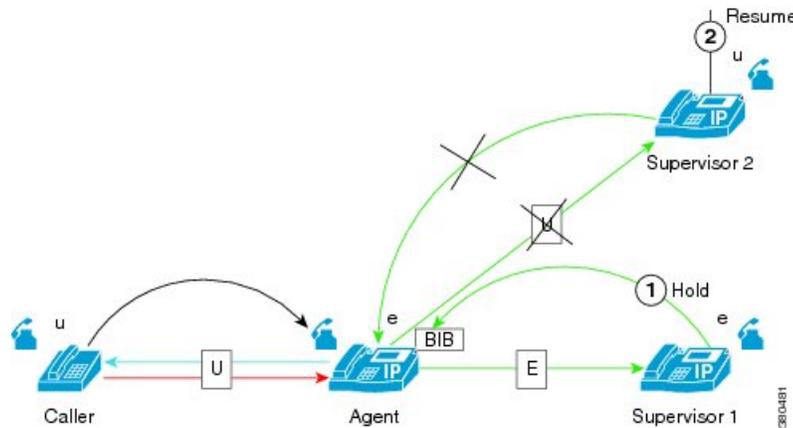
Cisco Unified Communications Manager determines if a call should be secured by examining the device capabilities of call-participants. When all call participant devices support sRTP, Cisco Unified Communications Manager automatically secures the call. When one or more call participant devices are not enabled for secure media using Secure Real Time Transfer Protocol (sRTP), the call is unsecured. If the agent device is enabled for sRTP, the Supervisor device must also be enabled for sRTP to establish a monitoring session. When caller 1 transfers to caller 2, the security of the call upgrades to secure media because caller 2's phone is enabled for sRTP. The Secure Icons appear on the agent and caller 2 phones. The monitoring call is not affected as the supervisor's phone is enabled for sRTP. The same scenario applies when the caller i-diverts to another destination or if caller 2 resumes the call via a shared line.



Transfer A Secure Monitoring Session to a Nonsecure Device

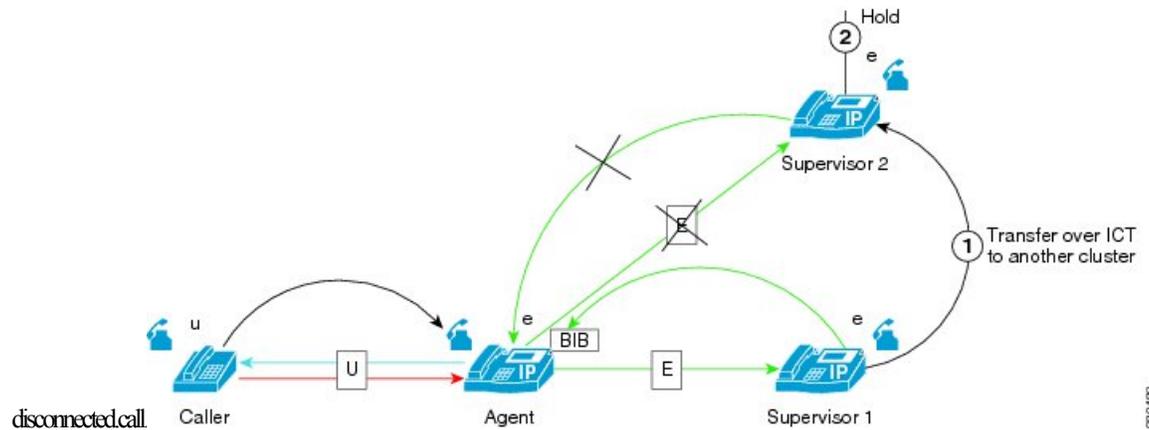
- Supervisor 1 monitors the agent in a secure monitoring session
- Supervisor 1 then transfers the monitoring call to supervisor 2
- Since supervisor 2's device is not enabled for encrypted media, the monitoring call is disconnected.
- Tone is played to supervisor 2 and cause code 57 (Bearer capability not authorized) is used to disconnect this call.
- CTI notifies the application connected to the original supervisor who started the monitoring session (supervisor1) that the monitoring session was disconnected.

- If the monitoring call is held, then resumed using a shared line on another phone not enabled for encryption, the same behavior results



Supervisor Transfers Monitoring Call Across Inter-Cluster Trunk

Cisco Unified Communications Manager does not support transferring secure monitoring calls over an Inter-Cluster trunk. The secure monitoring call is



Secure Monitoring a Conference Call

1. Caller 1 and agent 1 are engaged in a secure call
2. Supervisor establishes a secure monitoring session
3. Caller 2 joins the call from a non-secure device, so call security is downgraded to unsecure.

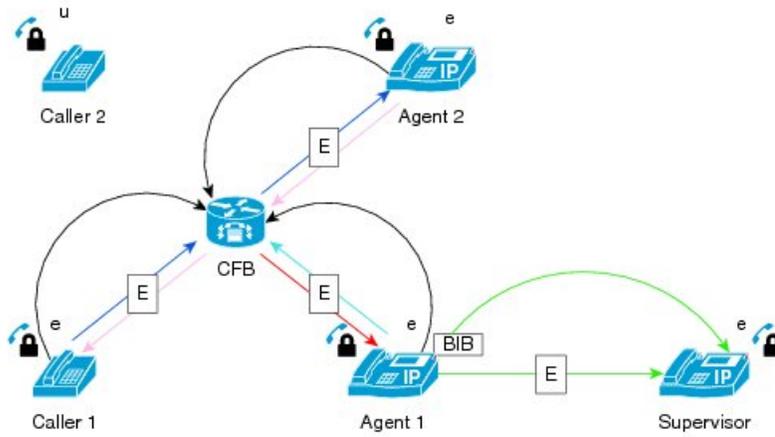
The secure silent monitoring session remains “encrypted” because agent 1 and supervisor both support encrypted media

4. Caller 2 leaves the call

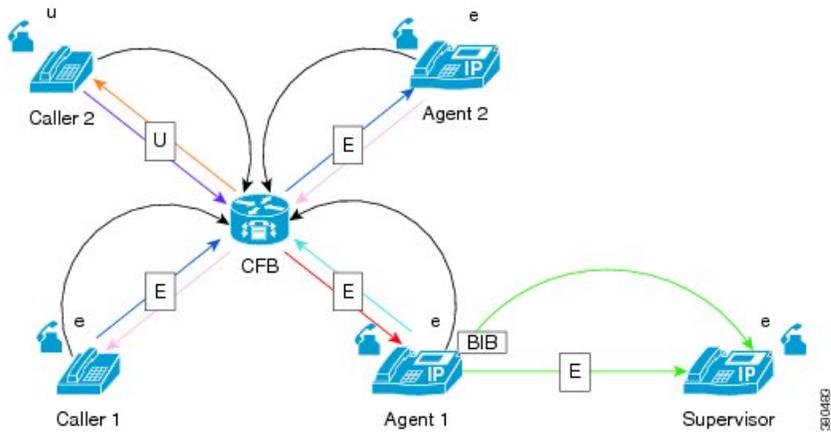
All remaining participants support encryption, so call security is upgraded automatically to use encrypted media

5. The original silent monitoring session was established as a secure call, so it remains unaffected

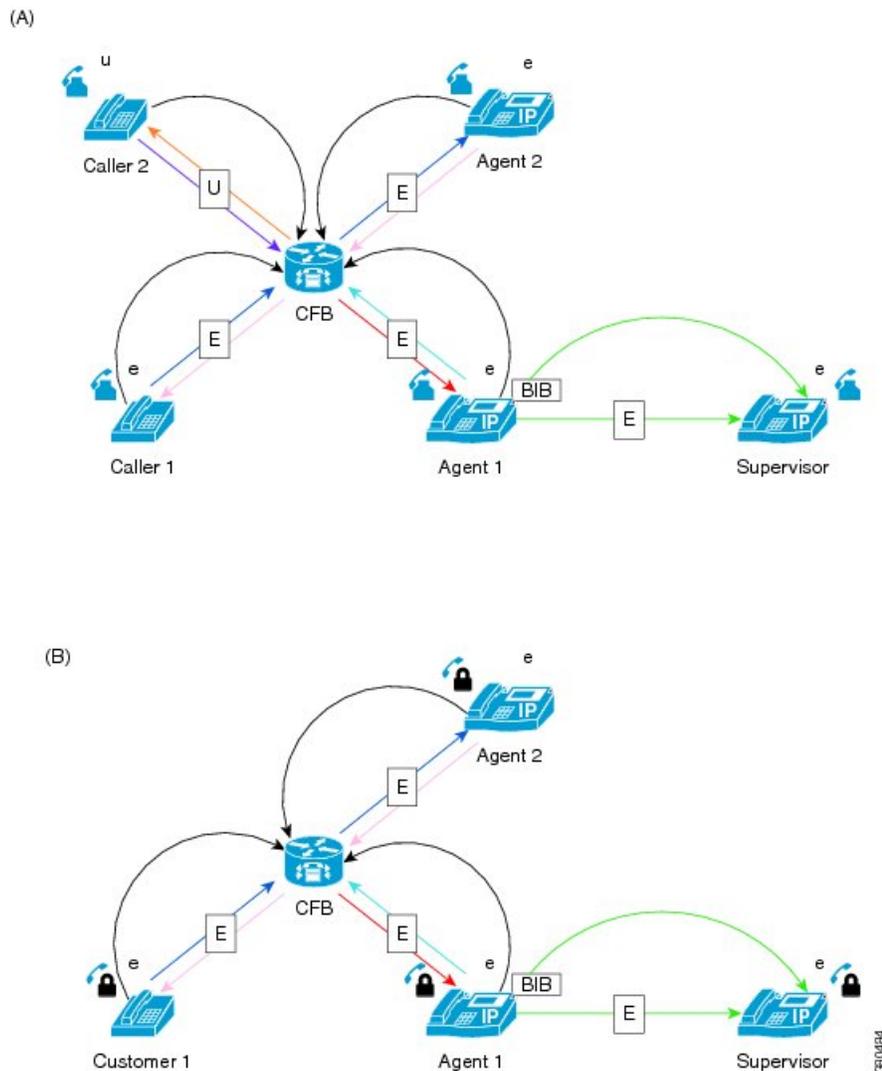
(A)



(B)



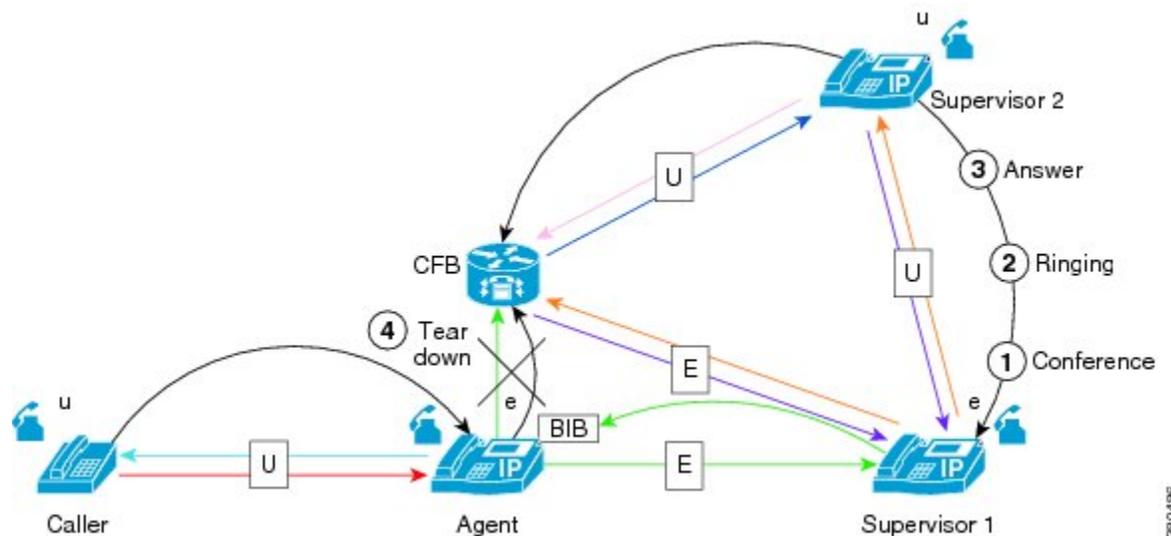
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Transfer Secure Silent Monitoring Calls

Supervisors can conference and transfer secure silent monitoring calls. Transfer of secure silent monitoring is possible if the transfer participant phone meets or exceeds the security capabilities of the Agent being monitored.

- Supervisor 1 is monitoring Agent in a secure monitoring session.
- Supervisor 1 transfers the call to Supervisor 2, whose phone is not enabled for encryption. When the call is transferred, the secure monitoring session call is disconnected.
- No tones are played to Supervisor 2. Supervisor 1 receives a CTI notification that the monitoring session was disconnected.



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System Requirements

The following section provides the system requirements for monitoring.

Supported Devices

For more information about verification of agent phone models that are supported for call monitoring, see <http://developer.cisco.com/web/sip/wikidocs/-/wiki/Main/Unified+CM+Silent+Monitoring+Recording+Supported+Device+Matrix>

CTI Requirements

Computer Telephony Integration (CTI) provides the ability for applications to monitor calls on a per-call basis. Cisco defines the monitor target as the party that is monitored (agent) and the monitor initiator as the monitoring party (supervisor).

If a single CTI application monitors both the agent and supervisor phones, that application receives the call events that are necessary for silent monitoring. If different CTI applications monitor the agent and supervisor phones, the CTI application that monitors the agent phone must provide the call information to the application that observes the supervisor phone.

CTI applications that monitor calls must have the corresponding monitoring privileges enabled for the application-user or end-user account.

Configuration

This section provides the steps that are necessary to configure monitoring.

Turn on IP Phone Built-In-Bridge to Allow Monitoring

The built-in bridge (BIB) of the agent phone must be set to **On** to allow its calls to be monitored.

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

Use the **Device > Phone** menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.

Figure 11: Setting the Phone Built In Bridge to On

The following figure illustrates turning on the IP phone BIB to allow



monitoring.

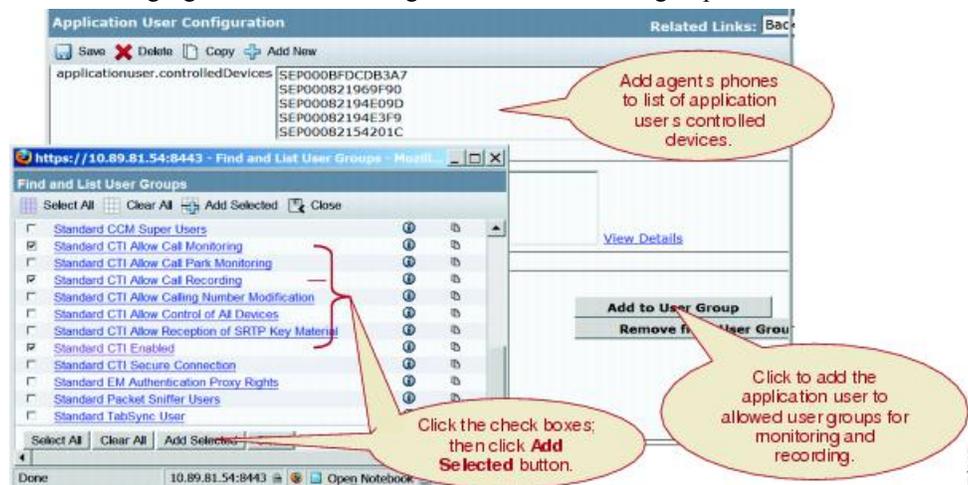
Add Supervisor to Groups That Allow Monitoring

Add the supervisor to the user groups: **Standard CTI Allow Call Monitoring user group** and the **Standard CTI Enabled user group**.

Use the **User Management** menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.

Figure 12: Adding Supervisors to Groups That Allow Monitoring

The following figure illustrates adding the user to these user groups.



Configure Monitoring Calling Search Space

The monitoring calling search space of the supervisor line appearance must include the agent line or device partition.

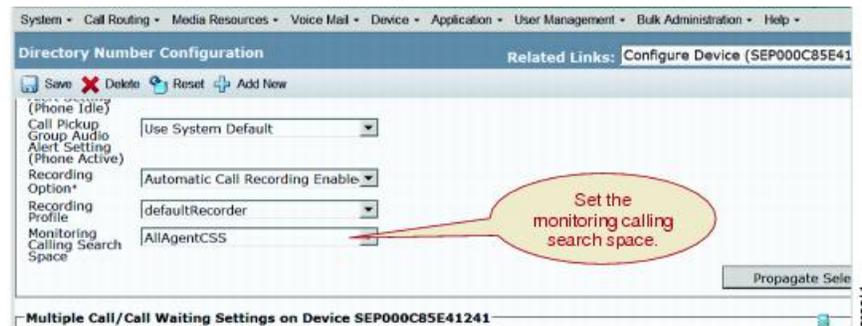
Set the monitoring calling search space in the supervisor line appearance window.

Use the **Device > Phone** menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration. When you display the Phone Configuration window for a phone, click on a line,

such as Line [1], in the Association Information pane. (You can choose either a DN that is already associated with this phone or you can add a new DN to associate with this phone.) In the Directory Number Configuration window that displays, configure the Monitoring Calling Search Space field for the chosen line on this phone.

Figure 13: Configuring DN for Monitoring Calling Search Space

The following figure illustrates how to configure the monitoring calling search space.



Configure Notification Tones for Monitoring

Set the service parameters for playing tone to **True** to allow tones to be played to the agent, the customer, or both.

CTI applications that initiate monitoring can also pass the play tone option to Cisco Unified Communications Manager. The monitoring tone plays when either the service parameter or the CTI application specifies the play tone option.

Use the **System > Service Parameters** menu option in Cisco Unified Communications Manager Administration to configure notification tones for monitoring.

Figure 14: Using Service Parameters to Configure Tones

The following figure illustrates how to use service parameters to configure tones.



Set the Monitoring Service Parameters

This section describes the service parameters that are related to the silent monitoring feature.

Notification

The following service parameters affect the playing of notification tones to the parties that are monitored by the call monitoring feature:

Clusterwide Parameters (Feature - Monitoring)

- Play Monitoring Notification Tone To Observed Target
- Play Monitoring Notification Tone To Observed Connected Parties

The default value for these service parameters is **False**. You must change the value of each parameter to **True** to enable the particular notification tone to play.

Built-In-Bridge

The following service parameter enables or disables the built-in bridge on phones:

Clusterwide Parameters (Device - Phone)

Built-in Bridge Enable

For more information about this service parameter, see "Turn on IP phone BIB to allow Monitoring".

Secure Silent Monitoring Provisioning Phone Configuration

To configure the phone for secure silent monitoring, follow these steps:

Procedure

-
- Step 1** Create a security profile for the Supervisor's phone type.
 - Step 2** Set the **Device Security Mode** in the security profile to **Encrypted**.
 - Step 3** Apply the device security profile to the Supervisor's device.
-

Recording

Call recording is a Cisco Unified Communications Manager feature that enables a recording server to archive agent conversations.

Terminology

The following terminology is used to describe recording:

Calling party

The person initiating the call; also referred to as the Customer.

Called party

The person answering the call; also referred to as the Agent.

Recorder

The application responsible for capturing and storing conversation media.

Built-in-Bridge (BiB)

Cisco IP Phone resources used to copy and forward the media streams of the Calling and Called parties to the Recorder.

Gateway

Cisco Voice Gateways and Unified Border Elements used to copy and forward the media streams of the Calling and Called parties to the Recorder.

Recording Media Source

The phone or gateway selected to copy and forward the media streams to the recorder.

Unified CM Recording Features

Call recording is one of the essential features in call centers, financial institutions and other enterprises. The recording feature sends copies of the agent and customer call media streams to the recorder. Each media stream is sent separately in an effort to best support a wide range of voice analytic applications.

Cisco Unified Communications Manager offers the following recording features:

- Allows conversations to be captured and stored for review, analysis, and/or legal compliance.
- Designed to use a combination of IP Phone and network (e.g. Gateway) resources to ensure conversations are recorded
- Mobility and off-network conversations can be captured using Network-recording; available in Unified Communications Manager release 10 or later
- Network topology friendly
- Does not require SPAN
- Provides Call Admission Control, Bandwidth Reservation, and Codec Negotiation
- Provides notification tones when legal compliance is required
- Supports Secure (sRTP) Media in Unified CM 8.0(1)
- Any conversation may be recorded; does not require contact center software

Network Based Recording

Network based recording is available as of Cisco Unified Communications Manager, Release 10.0(1). Network based recording allows you to use the gateway to record calls.

Network based recording offers the following benefits:

- Allows Cisco Unified Communications Manager to route recording calls, regardless of device, location, or geography.
- Centralizes recording policy control.
- Captures calls that are extended off the enterprise network to mobile and home office phones.
- Cisco Unified Communications Manager selects dynamically the right media source based on the call flow and call participants.
- Enhanced SIP Header and CTI metadata enables applications to track recorded calls in both single and multi-cluster environments.
- Recording Serviceability counters and alarms help compliance officers ensure that calls are recorded by monitoring the real-time status and historical performance of the feature.

Architecture

Cisco Unified Communications Manager uses either IP phone-based or network-based recording architecture to provide call recording.

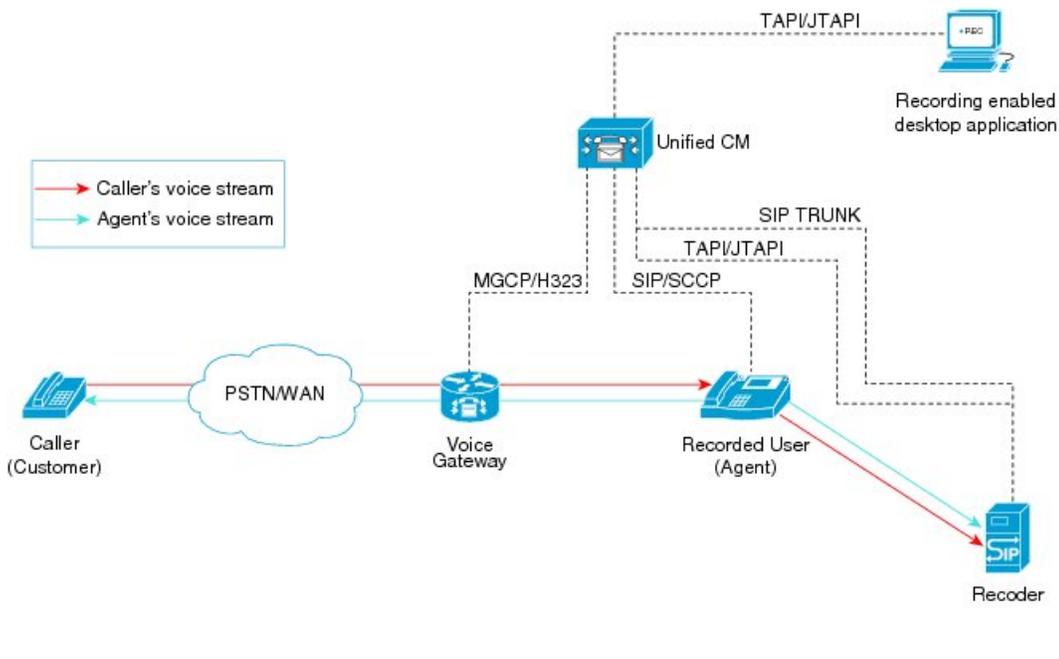
In IP phone based recording, recording media is sourced from the IP phones. The phone forks two media streams to the recording server where the media streams represent the two media streams in the call.

In network-based recording, recording media can be sourced from either the phone or the gateway. When you implement network-based recording, the gateway in your network must connect to Cisco Unified Communications Manager over a SIP trunk.

IP Phone-Based Recording Architecture

The following figure illustrates the phone based approach for recording. In the illustration, the agent phone relays two independent media streams (the agent and the customer) to the recorder.

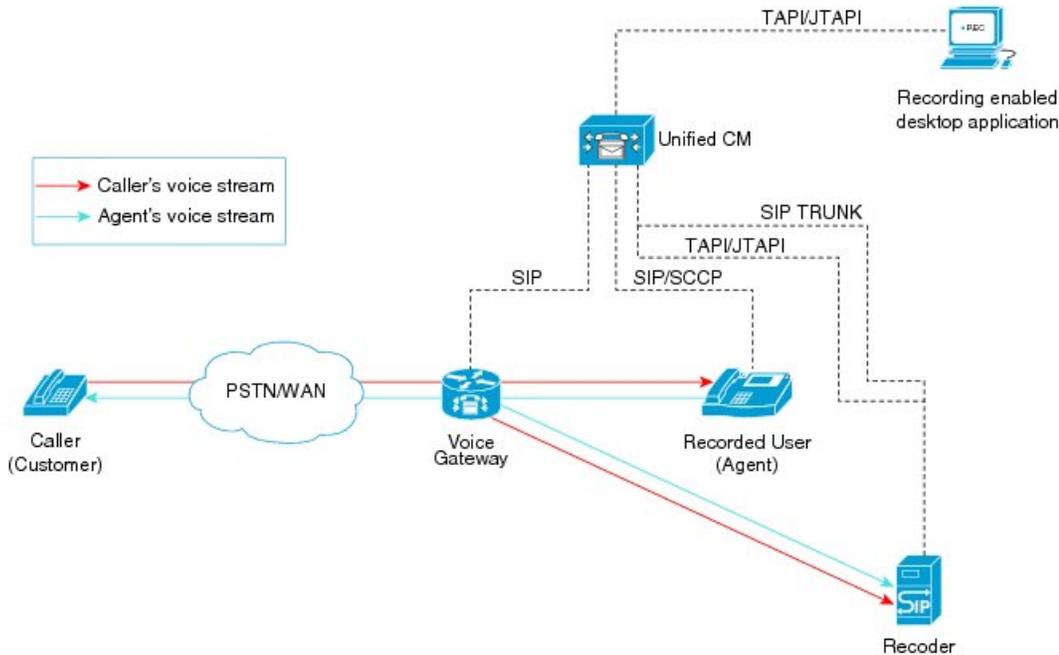
Figure 15: IP Phone-Based Recording Session



Network-Based Recording Architecture

The following diagram illustrates the network based approach for recording. In the illustration, the gateway relays two independent media streams (the agent and the customer) to the recorder.

Figure 16: Network-Based Recording Session



IP Phone Based Recording - How It Works

In IP phone based call recording, Cisco Unified Communications Manager forks media from an IP phone over to a recording servers. IP Phone based recording supports the following recording modes: Automatic, Selective Silent, and Selective User.

The following describes a typical call flow for IP phone based recording. In this example, call recording is configured to happen automatically.

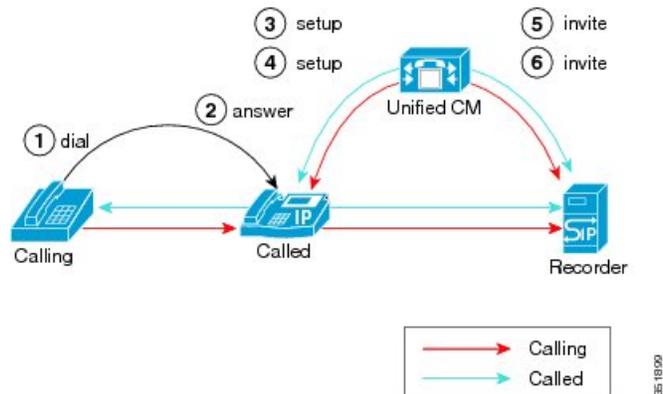
- A call is received and answered on a line that is configured for automatic recording.
- Cisco Unified Communications Manager automatically sends two call setup messages to the BiB (Called) device. The first setup message is for the called party stream. The second setup message is for the calling party stream.
- Cisco Unified Communications Manager INVITES the recorder to both calls via SIP Trunk.
- The recorder accepts both calls.
- The phone BiB forks two call media streams to the recorder.



Note An optional recording tone can be configured to play to either the calling party, called party, or both. when a call is being monitored and recorded simultaneously, the recording tone settings override the monitoring tone settings

The following figure illustrates an IP phone based recording session that is configured for automatic recording.

Figure 17: IP Phone Based Recording



Network Based Recording - How it Works

With network based recording call recording media can be sourced from either the IP phone or from a gateway that is connected to Cisco Unified Communications Manager over a SIP trunk. Network based recording supports the following recording modes: Automatic, Selective Silent, and Selective User.

Following is a typical call flow for network based recording:

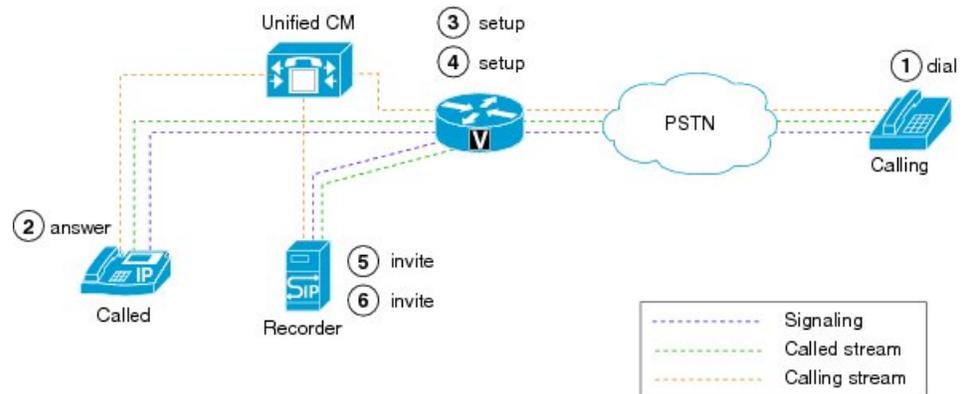
- A call is received and answered on a line that is configured for automatic silent recording where the gateway is selected as the recording media source.
- Cisco Unified Communications Manager automatically sends two call setup messages to the gateway.
- Cisco Unified Communications Manager INVITES the recorder to both calls via SIP Trunk.
- The recorder accepts both calls. and receives two RTP streams from the gateway.
- The gateway forks both media streams to the recorder.



Note An optional recording tone can be configured to play to the Calling, Called, or both parties. When the call is both monitored and recorded, recording tone settings override monitoring tone settings.

The following figure illustrates how network based recording works.

Figure 18: Network Based Recording



Recording Media Source Selection

When configuring recording, you must specify whether you want to use the phone or the gateway as the preferred media source for call recording. In selecting the recording media source, Cisco Unified Communications Manager takes into account the preferred media source that is configured for the user being recorded, the call media type, and whether a gateway is available in the call path.

Selecting the Recording Media Source

The following table summarizes how Cisco Unified Communications Manager selects the recording media source for calls. For example, if the gateway is chosen as the preferred recording media source, the media type is unsecured and there is a gateway in the call path, Cisco Unified Communications Manager will attempt to use the first gateway in the call path to record the media.



Note If the selected recording media source is unavailable, Cisco Unified Communications Manager will attempt to use an alternate source.

Table 3: Recording Media Source Selection

Preferred Media Source	Media Type	Gateway is in call path?	Selected Media Source
Gateway	Unsecure (RTP)	Yes	Gateway **
		No	Phone***
	Secure (sRTP)	Yes	Phone****
		No	Phone***

Preferred Media Source	Media Type	Gateway is in call path?	Selected Media Source
Phone	Unsecure (RTP)	Yes	Phone*
		No	Phone*
	Secure (sRTP)	Yes	Phone*
		No	Phone*

* Administrator chooses phone, phone is in the call path, phone is used for recording.

** Administrator chooses gateway, gateway is in call flow, gateway is used for recording.

*** Administrator selects gateway, but gateway is not in call flow, phone is used for recording.

**** Administrator selects gateway, gateway is in call flow, but media is secure, phone is used for recording.

Alternate Recording Source Selection

The following table displays the order that Cisco Unified Communications Manager follows when selecting an alternate source for the selected recording media in the event that the selected recording media source is unavailable. For example, if gateway is the selected media source, Cisco Unified Communications Manager will attempt to use the first gateway in the call path for recording. If that option fails, Cisco Unified Communications Manager will attempt to use the last gateway in the call path. Else, Cisco Unified Communications Manager will use the phone for call recording.

Table 4: Alternate Recording Media Source

Recording Source Selection	Gateway Preferred	Phone Preferred
First attempt	First gateway in call path	Phone
Second attempt	Last gateway in call path	First gateway in call path
Third attempt	Phone	Last gateway in call path

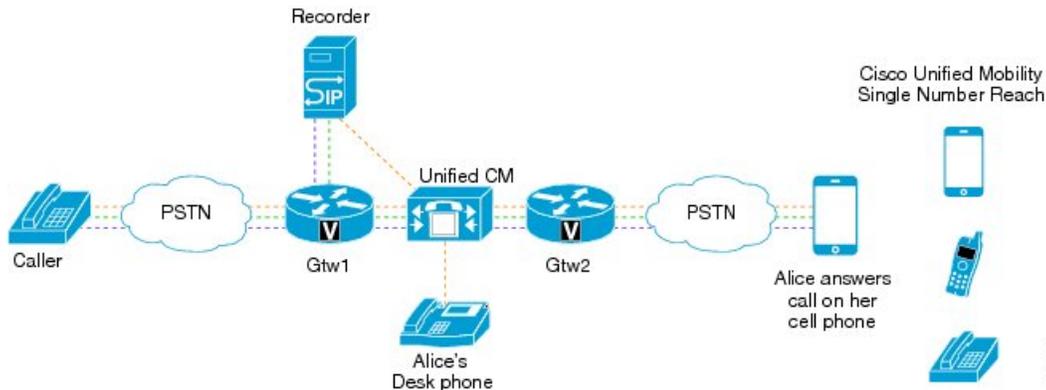
Gateway Selection

When you select the gateway as the preferred source, Cisco Unified Communications Manager attempts to use the first gateway in the call path for recording source media. This gateway may be the ingress or the egress gateway.

Single Number Reach Example

An external caller phones over the PSTN and Gateway 1 to Alice's desk phone. However, Alice is not in the office and has Single Number Reach configured on her desk phone so that Cisco Unified Communications Manager extends the call out of the enterprise network and back out across the PSTN to Alice's mobile phone. Because the gateway between the external caller and Cisco Unified Communications Manager is the first gateway in the call path, recording media is sourced from that gateway.

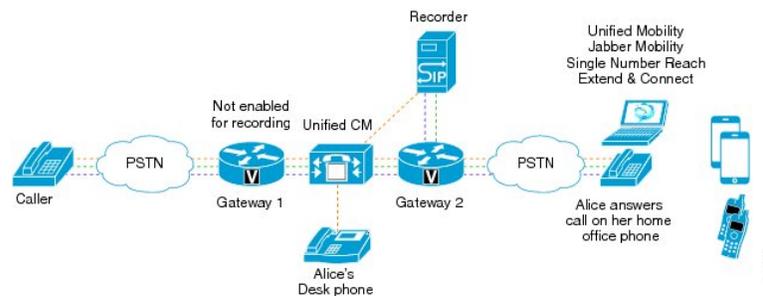
Figure 19: Single Number Reach Example



Cisco Extend & Connect Example

Bob and Alice both work for company X. Bob uses his Cisco IP Phone to call Alice's desk phone. However, Alice is working from home and has Cisco Jabber Extend & Connect configured. Cisco Unified Communications Manager extends the call off-network to Alice's home office phone and Alice answers the call on her home phone. Since Gateway 1 is not enabled for recording, Gateway 2 is used.

Figure 20: Cisco Extend & Connect Example



Recording Operation Modes

The following types of call recording modes exist:

Automatic silent recording

Automatic silent recording records all calls on a line appearance. Cisco Unified Communications Manager invokes the recording session automatically with no visual indication on the phone that an active recording session is established.

Selective silent call recording

Selective silent recording allows calls to be recorded as needed. A supervisor can start or stop the recording session via CTI-enabled desktop. Alternatively, a recording server can invoke the session based on predefined business rules and events. In Selective silent call recording, there is no visual indication on the phone that an active recording session is established.

For the two types of selective recording, selective silent recording is the default recording mode.

Selective user call recording

Selective user call recording allows calls to be recorded as needed. An agent can choose which calls to record and invoke the recording session via CTI-enabled desktop, or by a Softkey/Programmable Line Key. When selective user call recording is used, the Cisco IP phone displays recording session status messages. This feature is available in Cisco Unified Communications Manager 9.0(1) or later.

As of Release 10.0(1), mobility users can start or stop a recording session via DTMF (*86).

Regardless of the recording mode, each recording session delivers two unmodified RTP streams to the recording server over a SIP trunk. The independent media streams allow contact centers to take better advantage of speech analytic technologies that look for speech patterns, key words, and vocal behaviours that might indicate a problem during the call.



Note Silent selective and user selective call recording sessions can not be run simultaneously. When you try to invoke one of the selective recording modes while the other is already in use, the phone or CTI application displays a "Recording already started" message.

Automatic Silent Recording

In automatic call recording, Cisco Unified Communications Manager establishes the recording session automatically when the call connects. The following describes the call flow:

- A call is received and answered on a line that is configured for automatic silent recording.
- Cisco Unified Communications Manager automatically sends two call setup messages to the BiB (Called) device. The first setup message is for the called party stream. The second setup message is for the calling party stream.
- Cisco Unified Communications Manager INVITES the recorder to both calls via SIP Trunk.
- The recorder accepts both calls and receives two RTP streams from the phone BiB.



Note An optional recording tone can be configured to play to either the calling party, called party, or both. when a call is being monitored and recorded simultaneously, the recording tone settings override the monitoring tone settings

For an illustration of automatic silent recording, see the example in [IP Phone Based Recording - How It Works, on page 23](#).

Selective Silent Recording

Selective silent recording is typically used in a call center environment to allow a supervisor to record an agent call. In selective silent recording mode, a CTI-enabled application running on a supervisor desktop is typically used to start and stop call recording for selected agent-customer calls. The call recording status does not display on the Cisco IP phone.

A typical call flow for selective silent recording is as follows:

- A call is received and answered on a line that is configured for selective recording.
- A supervisor working from a CTI-enabled desktop initiates the recording session. Alternatively, a recording server based on predetermined business rules.

- Cisco Unified Communications Manager automatically sends two call setup messages to the BiB (Called) device: one for the called party media stream and one for the calling party media stream.
- Cisco Unified Communications Manager INVITES the recorder to both calls via SIP trunk.
- The recorder accepts both calls and receives two RTP streams from the device BiB.

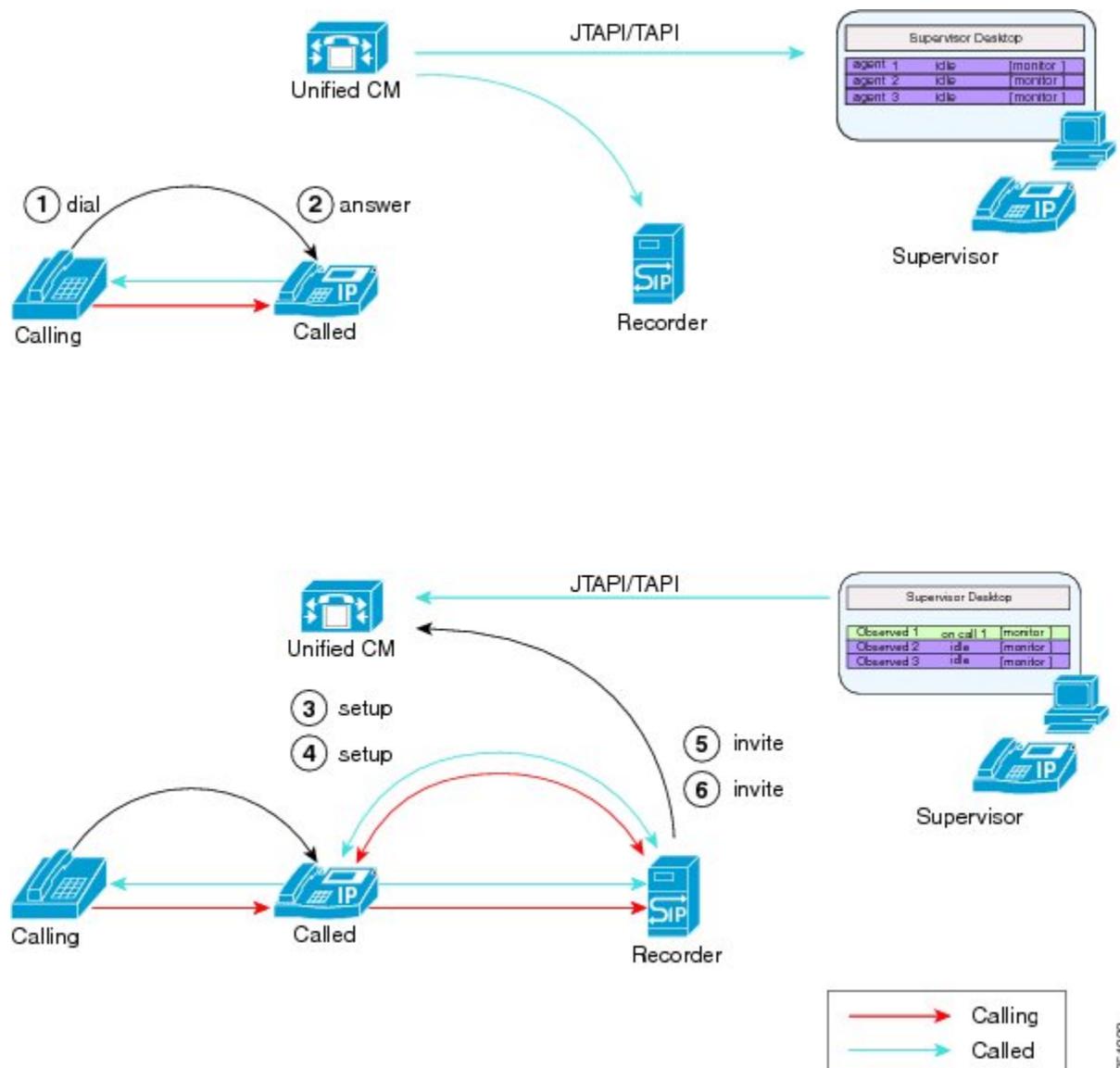


Note An optional recording tone can be configured to play to the Calling, Called, or both parties. When the call is being monitored and recorded simultaneously, the recording tone settings override the monitoring tone settings.

In this example, the supervisor manages the recording session from a CTI-enabled desktop.

The following figure illustrates a selective silent call recording session.

Figure 21: Silent Selective Call Recording Mode



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Selective User Recording

In selective user recording, an agent may start or stop a recording session using a softkey, programmable line key, or CTI-enabled application running on the desktop. In selective user recording, the call recording status displays on the Cisco IP phone.

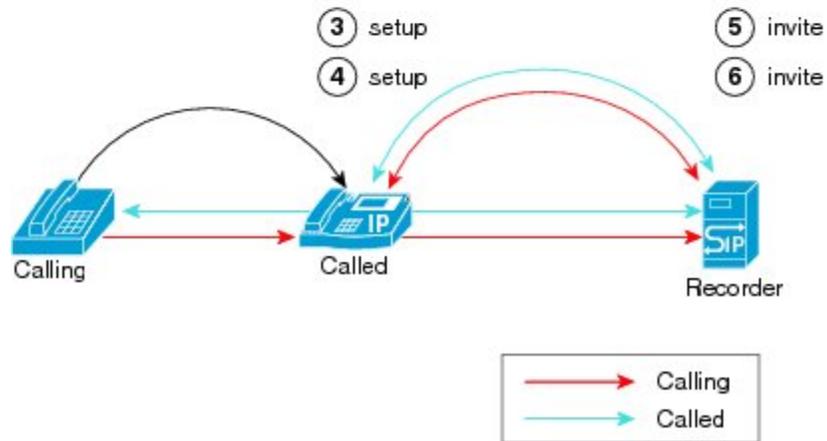
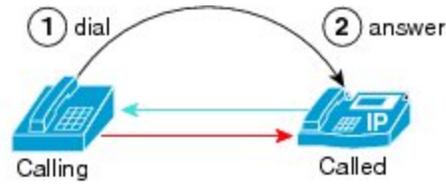
Following is a typical call flow for selective user recording:

- A call is received and answered on a line configured for selective recording.
- The called party starts the recording session by pressing the 'Record' softkey or programmable line key.
- Cisco Unified Communications Manager automatically sends two call setup messages to the BiB (Called) device: one to set up the media stream from the called party and the second to set up the media stream from the calling party.
- Cisco Unified Communications Manager sends an INVITE to the recorder via SIP trunk for both calls.
- The recorder accepts both calls and receives two RTP streams from the device BiB.
- The phone displays the status of the recording session. The Record key toggles to Stop Recording.



Note An optional recording tone can be configured to play to the Calling, Called, or both parties. When the call is being monitored and recorded, the recording tone settings override the monitoring tone settings.

Figure 22: Selective User Recording



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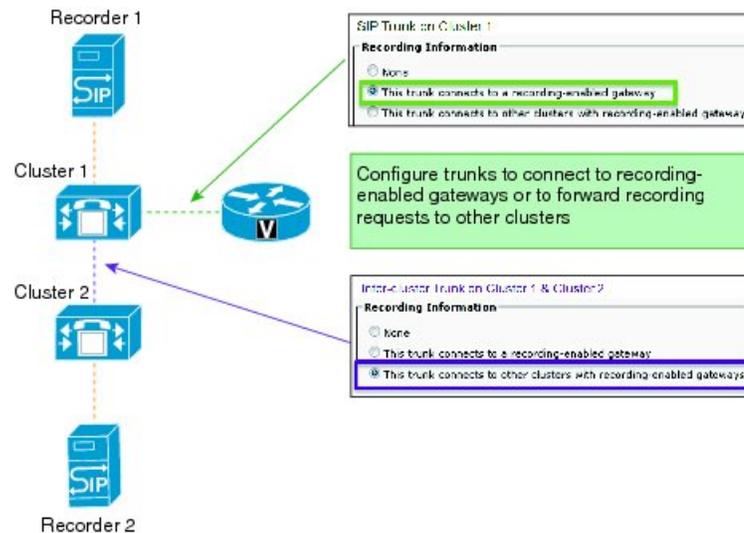
Recording In Multi-Cluster Environments

You can configure recording for multi-cluster networks. Your phones, gateways, and recorders can be connected to any cluster for recording to work. Cisco Unified Communications Manager can communicate recording requests between clusters so that recording can be centralized in a multi-cluster environment.

SIP Trunk Configuration

When you are configuring call recording in multi-cluster networks, you must set up the recording information for the trunks in your network. In Cisco Unified Communications Manager Administration, you can classify your trunks as either being connected to a recording-enabled gateway or being connected to other clusters that connect to recording-enabled gateways. If your trunks are classified as connecting to other clusters that connect to recording-enabled gateways, Cisco Unified Communications Manager forwards recording INVITES to the other clusters that connect to recording-enabled gateways. This allows you to centralize call recording in networks that span multiple clusters and to save on WAN bandwidth. You can set these details in the Recording Information section of the Trunk Configuration window.

Figure 23: Trunk Configuration for Multi-Cluster Recording

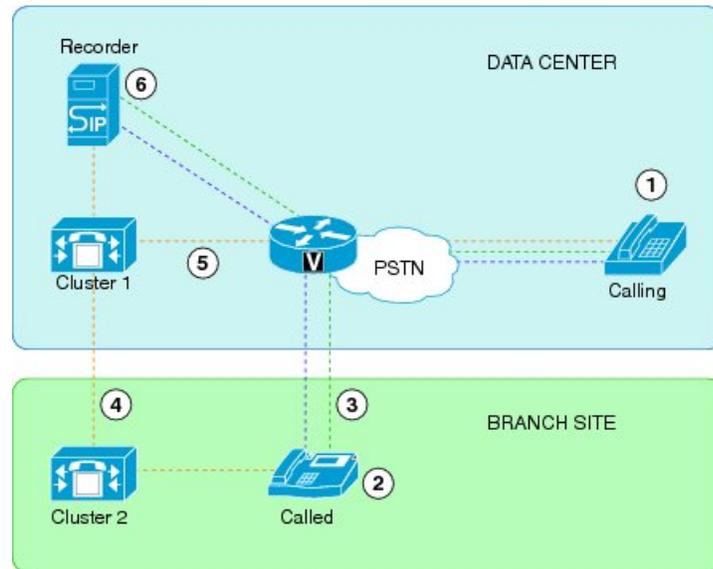


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Multi-Cluster Recording Examples

The following example illustrates a multi-site internal company that has implemented gateway recording. In this example, the gateway that is used for recording depends on whether call recording is initiated from the calling party or from the called party. In this example, the calling party starts recording so Gateway 1 from Cluster 1 is used as the recording source.

Figure 24: Multi-Cluster Recording Example



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Network Recording Use Cases

This section illustrates the following network recording use cases:

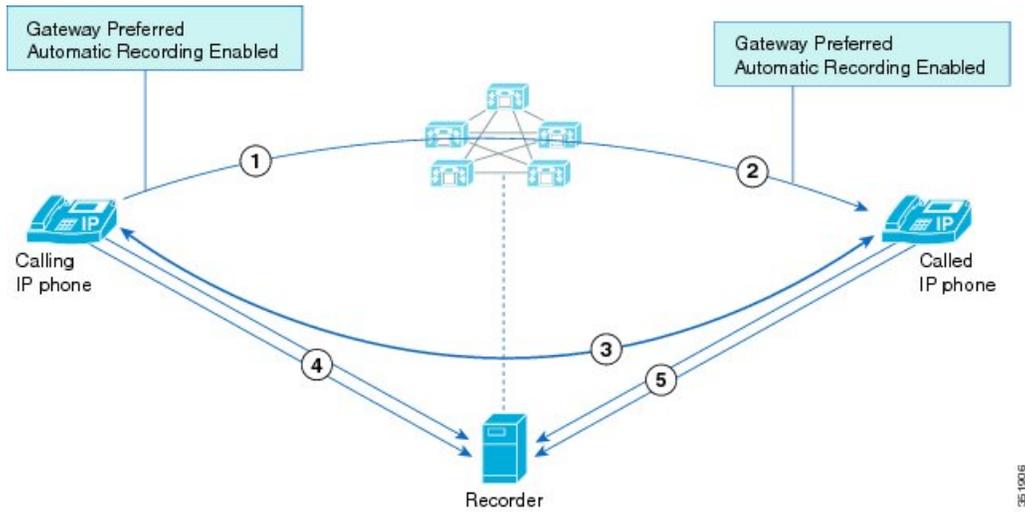
- IP phone to IP Phone both enabled for recording
- External Call to IP phone – Selective Recording
- Mid-Call – Gateway recording session stops when call is held
- Mobility – External Call to Mobile Jabber Client
- Mobility – External Call to Remote Destination Profile
- Extend & Connect – External Call to CTI Remote Device
- Inter-cluster Recording – External call from Cluster 1 to IP Phone on Cluster 2
- Inter-cluster Recording – External call from Cluster 1 is Held by User on Cluster 2

IP Phone to IP Phone Both Enabled for Recording

In this use case, Gateway is preferred, but the phone is selected.

- 1–3. An IP phone calls another IP phone
- 4–5. Two recording sessions are started automatically from both IP phones. The recording media source for both phones is Gateway Preferred. Since the gateway is not in the call flow Cisco Unified Communications Manager selects the IP Phones to fork the media to the recorder.

Figure 25: IP Phone to IP Phone Both Enabled for Recording

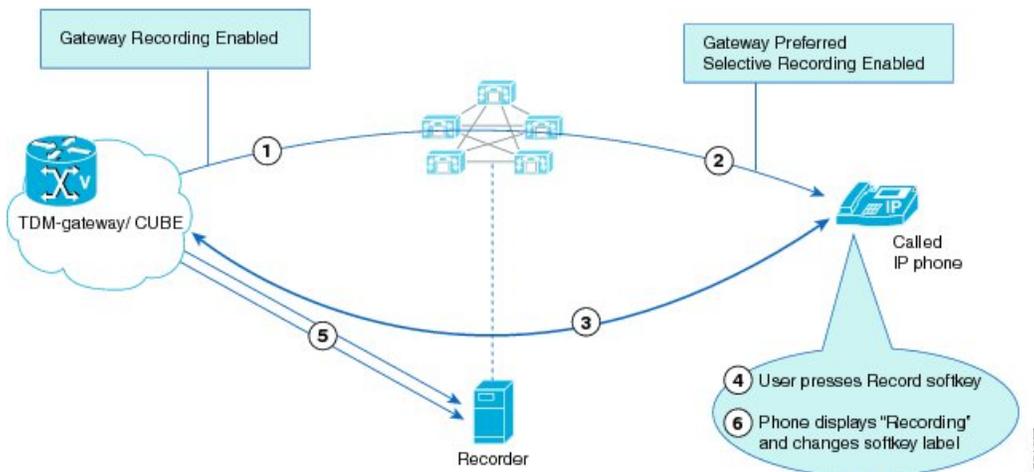


External Call to IP phone – Selective Recording

In this use case, Gateway is preferred and selected.

- 1–3. An external call is answered by a user with a Cisco IP phone.
- 4–5. The Cisco IP phone presses the Record softkey thereby starting a new recording session. The gateway is configured as the preferred recording media source and the gateway is valid for the call flow. Cisco Unified Communications Manager selects the gateway to fork the media to the recorder.
- 6. The Cisco IP phone displays "Recording..." and the softkey label changes to " StopRec".

Figure 26: External Call to IP phone – Selective Recording



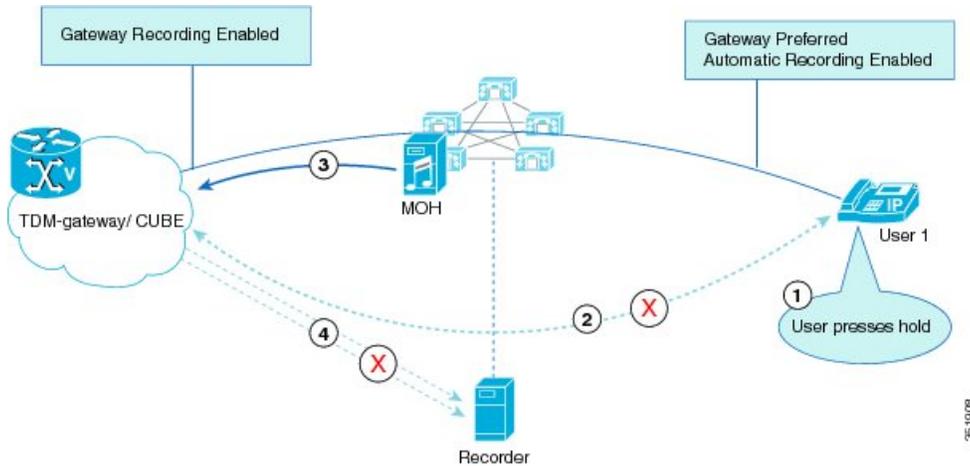
Mid-Call – Gateway Recording Session Stops When Call Is Held

This use case demonstrates the Hold function on the IP phone.

- 1–3. A call is in progress and the gateway is recording the media. User 1 places the call on hold. Cisco Unified Communications Manager plays Music-on-Hold to the caller.

- 4. Cisco Unified Communications Manager instructs the gateway to stop forking media to the recorder.

Figure 27: Mid-Call – Gateway Recording Session Stops When Call Is Held

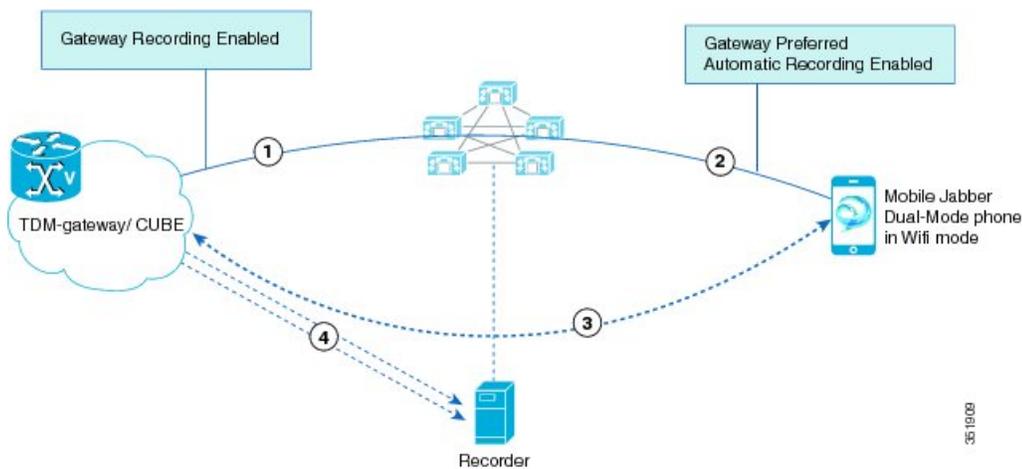


Mobility – External Call to Mobile Jabber Client

In this use case, the gateway is preferred and selected.

- 1–3. An external call is answered on a Mobile phone configured as a Dual-mode device
- 4. The recording session starts automatically for the dual mobile phone. The gateway is configured as the preferred recording source and is valid for the call flow. Cisco Unified Communications Manager selects the gateway to fork the media to the recorder.

Figure 28: Mobility – External Call to Mobile Jabber Client

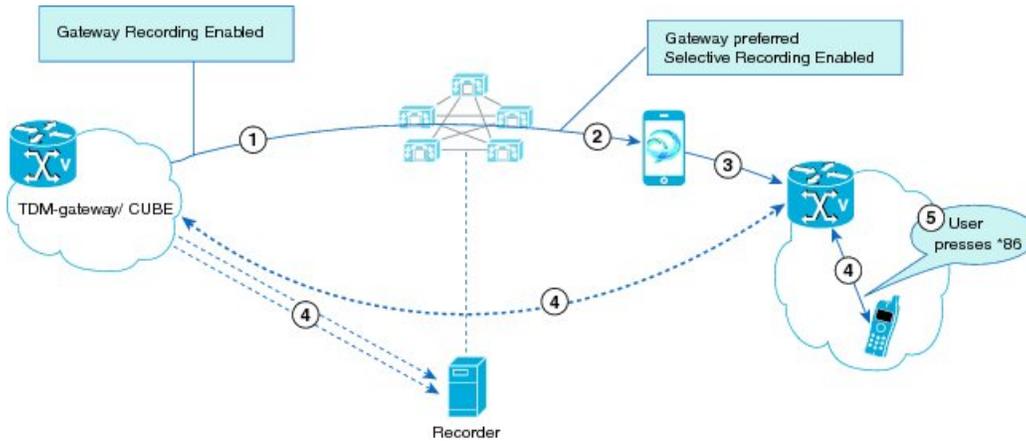


Mobility – External Call to Remote Destination Profile

In this use case, gateway is preferred and selected.

- 1–4. An external call is answered on a mobile phone using a remote destination configured on a Remote Destination Profile.
- 5–6. The mobile presses *86 to use DTMF to start the recording session. Cisco Unified Communications Manager selects the ingress gateway to fork the media to the recorder.

Figure 29: Mobility – External Call to Remote Destination Profile

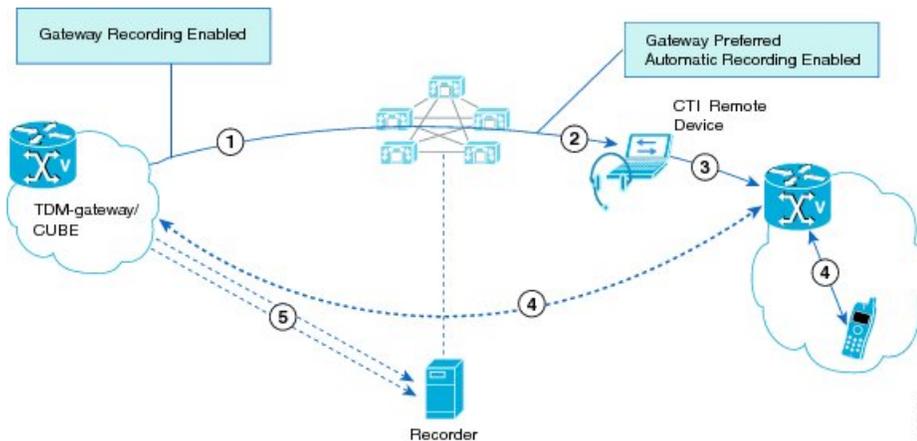


Extend & Connect – External Call to CTI Remote Device

In this use case, gateway is preferred and selected.

- 1–4. An external call is answered on an Extend & Connect remote destination that is configured as a CTI Remote Device.
- 5. The recording session starts from the CTI Remote Device. The gateway is configured as the preferred recording media source and is valid for the call flow. The ingress gateway is the first gateway in the call flow so Cisco Unified Communications Manager selects the ingress gateway to fork recording media to the recorder.

Figure 30: Extend & Connect – External Call to CTI Remote Device

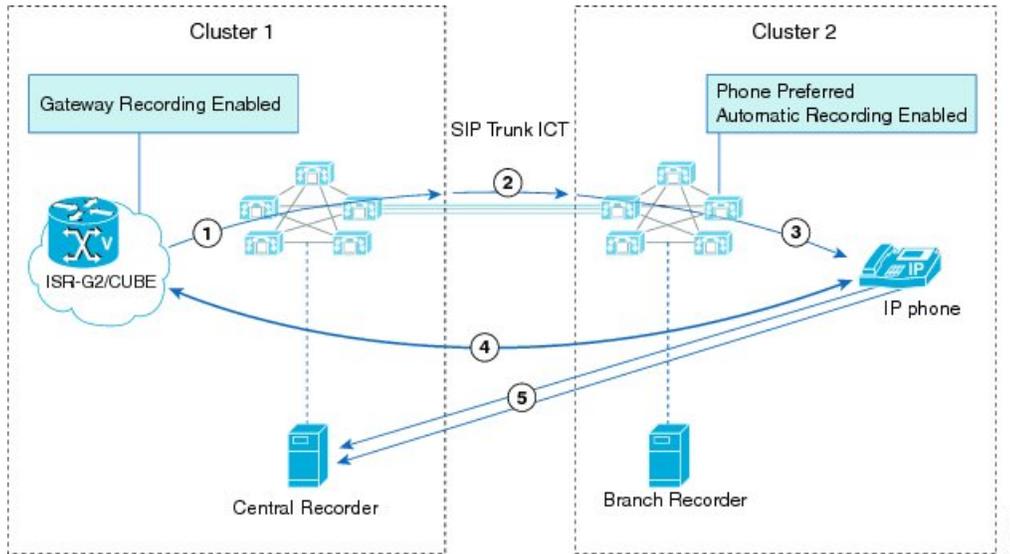


Inter-Cluster Recording – External Call From Cluster 1 to IP Phone on Cluster 2

In this use case, the gateway is selected. The Recording Profile specifies a central recorder.

- 1–4. An external call is received on a gateway in Cluster 1 and is answered by an IP phone in Cluster 2.
- 5. Recording starts automatically for the IP Phone in Cluster 2; the Phone is selected to fork the media. The recording profile assigned to the IP Phone is configured to use a central recorder.

Figure 31: Inter-Cluster Recording – External Call From Cluster 1 to IP Phone on Cluster 2

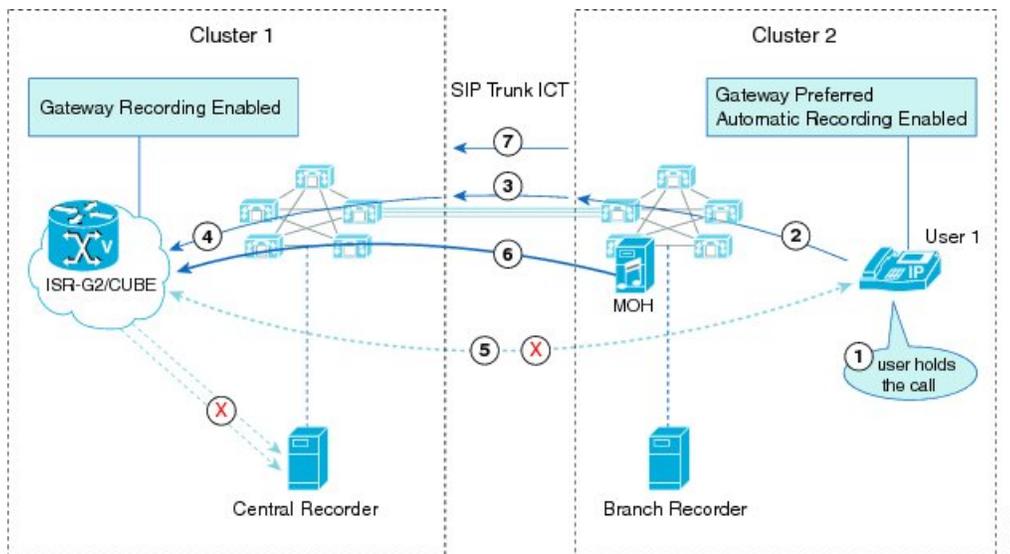


Inter-Cluster Recording – External Call From Cluster 1 Is HELD by User on Cluster 2

This use case demonstrates a mid-call Hold in an inter-cluster call.

- 1–5. An external call is underway to User 1 in Cluster 2. User 1 places the call on hold. The media stream is dropped.
- 6. Cisco Unified Communications Manager in Cluster 2 plays Music On Hold to the external caller.
- 7–8. Cisco Unified Communications Manager in Cluster 2 requests that Cluster 1 stop the recording session. Cluster 1 instructs the gateway to stop the session.

Figure 32: Inter-Cluster Recording – External Call From Cluster 1 Is HELD by User on Cluster 2



Recording Metadata

The SIP header that Cisco Unified Communications Manager sends to the recorder contains metadata that provides information on the media stream, such as device names, directory numbers, and cluster IDs. In addition, you can extract additional caller information with CTI.

You can obtain the following information for call participants from the SIP INVITE header or using CTI:

- Caller IDs
- Device names
- Directory numbers
- Cluster IDs
- Gateway IDs

In addition to the above information, you can use CTI to obtain the following information:

- Recorder profile directory number
- Recorder profile partition
- Recorder SIP trunk device name
- Near-end and far-end partitions
- Recording media source device type
- Recording media source cluster name
- Recording media source device name

Codec Selection

The rules for codec selection are as follows:

- The recording media sources send media streams with the same codec as the call that is being recorded.
- When Phone is configured as the recording media source, the recording codec is always the same as the original codec for the duration of the call.
- When Gateway is selected as the recording media source, the recording codec can be dynamically re-negotiated if the original conversation codec changes.
- If region settings for the recorder require different codecs, a transcoder is automatically inserted.



Note Some recording vendors transcode again for storage.

Recording Server Redundancy

Cisco Unified Communications Manager provides the following three options for setting up load balancing in order to provide redundancy for the recorder server.

- Option 1 - SIP redirect approach
- Option 2 - Route list approach
- Option 3 - DNS SRV records approach

Load balancing options vary depending on the vendor. Many vendors have implemented a SIP Proxy to provide server redundancy. You can use a route list with two or more SIP trunks to provide multiple SIP proxy and recording server redundancy. Refer to the vendor documentation for details on which options are supported by your recorder.

Option 1 - SIP Redirect Approach

Under this approach, the recorder or SIP Proxy issues a SIP 3XX response to redirect the Cisco Unified Communications Manager INVITE to load-balance multiple recorders.

Option 2 - Route List Approach

Under this approach, you can provide recorder load balancing by assigning multiple recorders to a route list. To do this:

- Configure your SIP trunks for each addressable recorder or recorder proxy.
- Assign your SIP trunks to a recorder route group with an assigned algorithm. The algorithm can be top-down or circular.
- Assign the recorder route group to a route list.
- Use a route pattern to direct traffic to the recorder route list.

Option 3 - DNS SRV Records

You can also use a DNS SRV record tag to populate the SIP trunk destination address. DNS SRV records can be set up with different priorities in order to provide primary and backup targets and provide load balancing across multiple targets.

DNS SRV records contain the service, protocol, domain name, TTL, class, priority, weight, port and target. Multiple SRV records may map to the same SRV record tag (service, protocol, domain name), which allows multiple records to be returned in response to a single DNS SRV query thereby providing redundancy.

Notification Tones

For legal compliance, an explicit notification in the form of a periodic tone can be made audible to Agent, Caller, or both to indicate a recording session is in progress. The tone can also be disabled. Recording tone settings override monitoring tone settings when both are enabled for the same call .

Use the following service parameters to set the default notification tone options:

- Play Recording Notification Tone To Observed Target(agent)
- Play Recording Notification Tone To Observed Connected Parties(customer)

The following figure illustrates the observed connected party and the observed target.

Figure 33: Observed Connected Party and Observed Party



Table 5: Recording Notification Tones

Notification Tone set to play to...	Agent hears tone?	Caller hears tone?	Supervisor hears tone?	Recording stream for agent contains tone?	Recording stream for caller contains tone?
No one	No	No	No	No	No
Agent (Observed Party)	Yes*	No	No***	No	No*****
Customer (Caller)	No	Yes**	No***	Yes****	No*****
Both	Yes*	Yes**	No***	Yes****	No*****

* Agent phone plays the monitoring tone locally; it does not embed the tone into the RTP voice stream sent to the remote (the Caller phone).

** Agent phone embeds a monitoring tone into the RTP voice stream sent to the remote and the remote phone plays the RTP voice stream (with tone).

*** When the remote phone is a non-Cisco Unified Communications Manager device, tones are not generated so none are recorded. If the remote phone is a Cisco Unified Communications Manager device, then it generates a tone, so the tone is included in the recording.

**** The Agent recording stream is a copy of the primary stream sent to the remote. The recording is a replica of the Caller's experience, so the tone is part of the recording.

***** When the remote phone is a non-Cisco Unified Communications Manager device, tones are not generated so none are recorded. If the remote phone is a Cisco Unified Communications Manager device, then it generates a tone, so the tone is included in the recording.

Secure Tone Interaction

Secure Tone was introduced in Cisco Unified CM 7.0(1) to provide call participants with an audible indication that the call is secured. The following rules apply to Secure Tones:

- When Secure Tone is enabled, the tone plays once to call participants at the beginning of the call.
- If Secure Tones and Monitoring Tones are both enabled, the Secure Tone plays once followed by Monitoring Tones (if the call is being monitored).
- If Secure Tones and Recording Tones are both enabled, the Secure Tone plays once, followed by Recording Tones (if the call is being recorded).

- If Secure, Monitoring, and Recording Tones are all enabled, the Secure Tone plays once followed by Recording Tones, which always take precedence over Monitoring Tones (if the call is being monitored and recorded).

Recording CDRs

Each call recording session generates one CDR for each media stream. To identify monitoring and recording CDRs, recording CDRs use the onBehalfOf fields to indicate that the calls were redirected by the recording feature. The GCI fields in the recording CDR are the same as the call that was recorded.

The original conversation ID in the recording CDR matches the Observed (Agent's) call leg that was recorded.

Recording Interactions and Limitations

The following interactions and limitations apply to call recording:

- The SIP header delivers recording metadata for each media stream. You can obtain additional metadata by using CTI interfaces for recording sessions that are managed by Cisco Unified Communications Manager.
- Cisco Unified Communications Manager supports multiple recording sessions for the same call when recording is started for both the Calling and Called parties.
- Cisco Customer Voice Portal calls may be recorded using Phone as recording media source.
- The gateway may not be a valid recording source for some call flows between two devices. For these call flows, select Phone as the recording media source. For additional detail, check the recording use cases.
- SIP Proxy servers may not be placed between Cisco Unified Communications Manager and the gateway.
- Each recording session adds two calls to the Busy Hour Call Completion (BHCC) rate with a minimal impact on CTI resources.
- If Multilevel Precedence and Preemption is configured, the **Busy Trigger** setting on the agent phone line must be at least 3.

Performance Counters and Alarms for Recording

Performance Counters

Monitor the health and status of recording media sources and sessions with real-time and historical performance counters. These counters are available in the Cisco Real-time Monitoring Tool under Cisco Call Recording.

The following table summarizes the recording performance counters that are available.

Table 6: Performance Counters Available for Call Recording

Performance Counter	Definition	Type
Recording Gateways In Service	The number of active or successful registrations to a recording-enabled gateway.	Real-time
Recording Gateways Out of Service	The number of configured recording gateways with no active registration.	Real-time
Recording Gateway Registration failures	The number of times registration to a recording-enabled gateway has failed.	Cumulative, Historical
Gateway Recording sessions active	The number of concurrent recording sessions (two streams) from the gateway to recorder.	Real-time
Gateway Recording session failures	The number of times a recording session with two streams from the gateway to the recorder has failed.	Cumulative, Historical
Phone Recording sessions active	The number of concurrent recording sessions with two streams from the phone to the recorder.	Real-time
Phone Recording session failures	The number of times a recording session with two streams from the phone to the recorder has failed.	Cumulative, Historical

Alarms

Recording feature alarms have been added to detect error conditions. The following table displays the recording alarms that are available.

Table 7: Alarms Available for Call Recording

Alarm Name	Description	Alarm Parameters	Severity
Recording Call Setup Fail	Indicates that the recording setup failed.	<ul style="list-style-type: none">Recorded device nameRecorded device DNRecorded Device Call Leg ID (Origination or destination call leg identifier for the call.)Gateway GUID (The global unique identifier, it is only available when gateway is recording media source.)Recording Media Preference (The preference set for the recording resource)Recording Media Source (The recording resource used for this recording request)Recording Cluster IDReason	Error

Alarm Name	Description	Alarm Parameters	Severity
Recording Resources Not Available	Indicates that the recording request failed due to non-availability of recording resources.	<ul style="list-style-type: none"> • Recorded device DN • Recorded Device Call Leg ID (Origination or destination call leg identifier for the call.) • Gateway GUID (The global unique identifier, it is only available when gateway is recording media source.) • Recording Media Preference (The preference set for the recording resource) • Recording Media Source (The recording resource used for this recording request) • Recording Cluster ID 	Warning

Alarm Name	Description	Alarm Parameters	Severity
Recording Already in Progress	Indicates that the recording request failed due to an existing recording request of the same name.	<ul style="list-style-type: none">• Recorded device name• Recorded device DN• Recorded Device Call Leg ID (Origination or destination call leg identifier for the call.)• Gateway GUID (The global unique identifier, it is only available when gateway is recording media source.)• Recording Media Preference (The preference set for the recording resource)• Recording Media Source (The recording resource used for this recording request)• Recording Cluster ID• Reason	

Alarm Name	Description	Alarm Parameters	Severity
Recording Invalid Call State	Indicates that the recording request failed due to internal errors.	<ul style="list-style-type: none"> • Recorded device name • Recorded device DN • Recorded Device Call Leg ID (Origination or destination call leg identifier for the call.) • Gateway GUID (The global unique identifier, it is only available when gateway is recording media source.) • Recording Media Preference (The preference set for the recording resource) • Recording Media Source (The recording resource used for this recording request) • Recording Cluster ID • Reason 	Information

Alarm Name	Description	Alarm Parameters	Severity
Recording Session Terminated Unexpectedly	Indicates that a recording in progress is terminated unexpectedly. The possible causes are trunk connected to recorder is out of service, recorder failed, or the call flow is not supported.	<ul style="list-style-type: none"> • Recorded device name • Recorded device DN • Recorded Device Call Leg ID (Origination or destination call leg identifier for the call.) • Gateway GUID (The global unique identifier, it is only available when gateway is recording media source.) • Recording Media Preference (The preference set for the recording resource) • Recording Media Source (The recording resource used for this recording request) • Recording Cluster ID • Reason 	Error
Recording Gateway Registration Rejected	Registration to recording-enabled gateway rejected after multiple attempts; gateway marked out-of-service	<ul style="list-style-type: none"> • Gateway Address • Cause 	Error
Recording Gateway Registration Timeout	No response from recording-enabled gateway after multiple attempts; timeout occurred; gateway marked out-of-service	<ul style="list-style-type: none"> • Gateway Address • Cause 	Error
Recording Gateway Out Of Service	Recording-enabled gateway closed connection to Cisco Unified Communications Manager	<ul style="list-style-type: none"> • Gateway Address • Cause 	Notice

Alarm Name	Description	Alarm Parameters	Severity
Recording Gateway In Service	Recording gateway status changed from out-of-service to in-service	<ul style="list-style-type: none"> Gateway Address 	Notice
Recording Gateway Session Failed	Recording gateway closed the recording session unexpectedly	<ul style="list-style-type: none"> Gateway Address Cause Gateway GUID 	Error

System Requirements

The following section provides the system requirements for recording.

Supported Devices

For details on which phone models are supported for call recording, see <http://developer.cisco.com/web/sip/wikidocs/-/wiki/Main/Unified+CM+Silent+Monitoring+Recording+Supported+Device+Matrix>.

Gateways Supported

- Supports both Voice gateways and Unified Border Elements (CUBE) as long as they interface with Unified CM using SIP and the Router platform supports the UC Services Interface (not supported for H323 or MGCP based calls).
- The word gateway is used interchangeably to refer to both Voice gateways and CUBE devices.
- The gateway has to be directly connected to Cisco Unified Communications Manager using a SIP trunk. Recording is not supported for SIP Proxy servers.
- ISR-Gen2 Gateways (29XX, 39XX Series) running 15.3(3)M train release are supported.
- ASR-100X Gateways running release XE 3.10 or later are supported.
- VG224 is not currently supported.

For additional details on which gateways are supported, see <https://developer.cisco.com/web/sip/wiki/-/wiki/Main/Unified+CM+Recording+Gateway+Requirements>.

Configuration

Call recording configuration involves:

1. Configure Recording Profile
 - a. Configure SIP Profile (optional)
2. Configure Recorder as SIP trunk device
3. Configure route pattern for recorder
4. Configure recorder redundancy (if supported)
5. Enable recording on a line appearance
6. Configure IP Phone as recording media source (optional)
7. Configure Gateway as recording media source (optional)
8. Configure recording notification tones (optional)



Note At least one recording media source must be configured

Configure Recording Profile

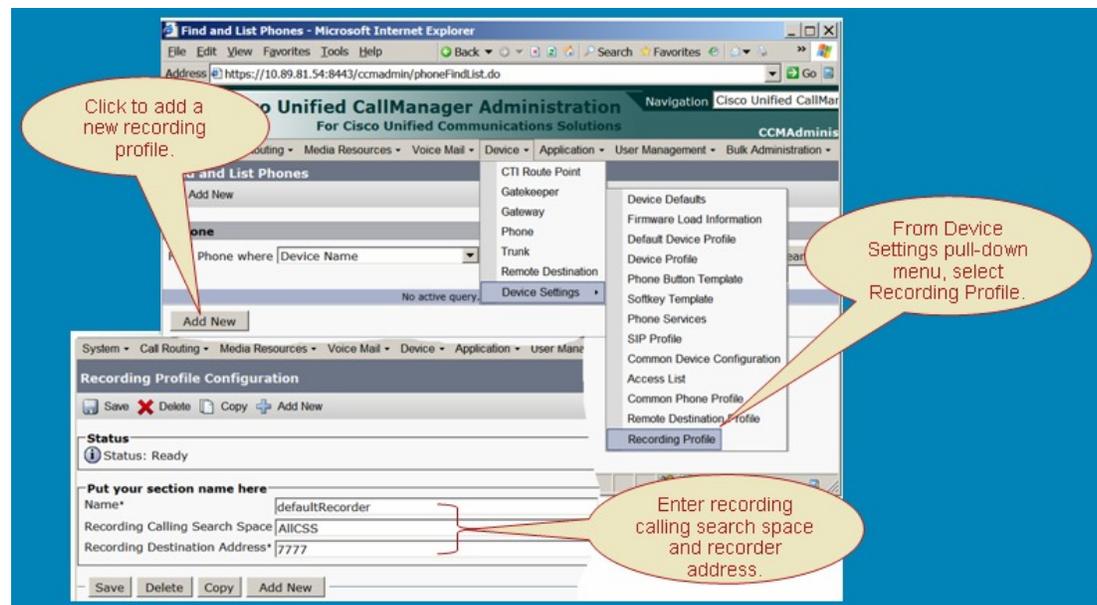
Create a recording profile from the Device Setting pull-down menu.

Enter the recording profile name, recording calling search space, and recording destination address.

Use the **Device > Device Settings > Recording Profile** menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.

The following figure illustrates creating a recording profile.

Figure 34: Creating a Recording Profile



Create SIP Profile for Recorder (Optional)

Create a SIP profile for recording. Use the **Device > Device Settings > SIP Profile** menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.

You can check the Deliver Conference Bridge Identifier check box, which delivers additional information (specifically, the b-number that identifies a conference bridge) to the recorder across the SIP trunk. If the check box is left unchecked, the far-end information for the remote conference remains empty.

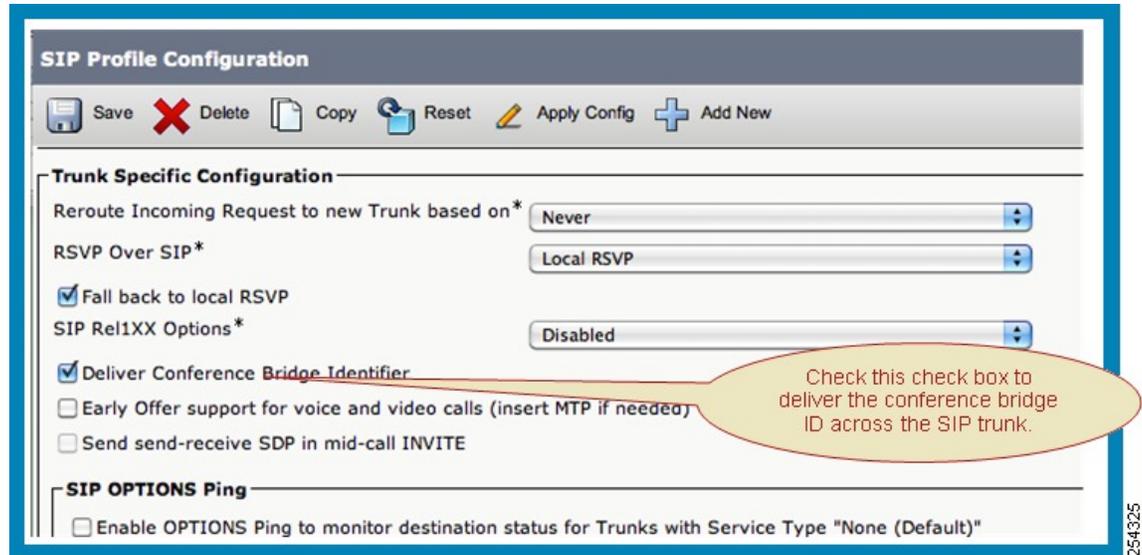
Check the Deliver Conference Bridge Identifier check box on the remote cluster SIP profile as well.



Note Checking this check box is not required for recording, but the conference bridge identifier helps to update the recorder when recording calls that involve a conference bridge. If you do not create a new SIP profile for recording, you can use the Standard SIP Profile.

See the Cisco Unified Communications Manager Administration Guide for details of configuring SIP profiles. The following figure illustrates creating a SIP profile for recording.

Figure 35: Creating a SIP Profile for Recording



Configure Recorder as SIP Trunk Device

The SIP trunk points directly to the recorder. Many recorders (such as those from Witness and Nice) consist of proxy, logger or storage and database.

The recorder accepts recording calls from Cisco Unified Communications Manager in SIP.

A directory number gets assigned to the recorder; a route pattern gets configured for the SIP trunk.

Create a SIP Trunk That Points to the Recorder

Create a SIP trunk that points to the recorder.

Enter the recorder DN, which must match a route pattern for the SIP trunk or a route list that includes the recorder.

Choose the appropriate SIP profile that you configured for recording.

Use the **Device > Trunk** menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.

The following figure illustrates creating a SIP trunk that points to the recorder.

Figure 36: Creating a SIP Trunk That Points to the Recorder

Trunk Configuration Related Links: [Back To Find/List](#) [Go](#)

Save Delete Reset Add New

SIP Information

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	10.89.63.58		

MTP Preferred Originating Codec* 711ulaw

Presence Group* Standard Presence group

SIP Trunk Security Profile* Non Secure SIP Trunk Profile

Rerouting Calling Search Space <None >

Out-Of-Dialog Refer Calling Search Space <None >

SUBSCRIBE Calling Search Space <None >

SIP Profile* Recording SIP Profile

DTMF Signaling Method* No Preference

Normalization Script

Name GetAroundNormalizationNameIssue

Enter recorder hostname or IP address.

Choose the SIP Profile for the recorder.

254326

Configure Route Pattern for Recorder

Create a route pattern for the recorder SIP trunk. The Recording Destination Address in the recording profile must match this pattern.

Select the SIP trunk that points to the recorder, or select a route list of which the recorder is a member.

Use the **Call Routing > Route/Hunt > Route Pattern** menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.

The following figure illustrates creating a route pattern for the recorder.

Figure 37: Creating a Route Pattern for the Recorder

System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help

Route Pattern Configuration Related Links:

Save Delete Copy Add New

Pattern Definition

Route Pattern* 7777

Route Partition <None >

Description CCBU Recorder

Numbering Plan -- Not Selected --

Route Filter <None >

MLPP Precedence* Default

Resource Priority Namespace Network Domain <None >

Route Class* Default

Gateway/Route List* SIPToCCBURecorder

Route Option

Route this pattern

Block this pattern No Error

Call Classification* OffNet

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Enter route pattern that matches "recording destination address" in the recording profile.

Select the SIP trunk that points to the recorder.

254327

Configure Recorder Redundancy (If Supported)

Many recorders have built-in proxy and redundancy functionalities, for example, Nice/Witness recorders.

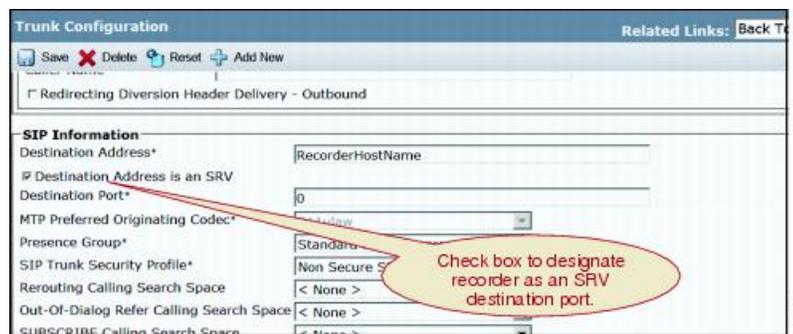
You can also use the following mechanism to achieve recorder redundancy:

- Use the SRV record for the recorder destination address in SIP trunk configuration.
- Use multiple recorders for redundancy and load balance. Create a SIP trunk for each recorder; create a route list to include the route groups that have individual SIP trunks as a member.

Use the **Device > Trunk** menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.

The following figure illustrates enabling SRV for a SIP trunk.

Figure 38: Enabling SRV for a SIP Trunk



Enable Recording for a Line Appearance

To enable recording of an agent, set the Recording Option in the line appearance of the agent to one of the following options:

- Automatic Call Recording Enabled
- Selective Call Recording Enabled

Select a pre-created recording profile from the drop-down list box. (Use **Device > Device Settings > Recording Profile** to configure a recording profile.)

Use the **Call Routing > Directory Number** menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.

If you also have Multilevel Precedence and Preemption (MLPP) configured, make sure the **Busy Trigger** is set to at least 3.

The following figure illustrates enabling recording for a line appearance.

Figure 39: Enabling Recording for a Line Appearance

Directory Number Configuration

Save Delete Reset Apply Config Add New

Party Entrance Tone* Default

Line 1 on Device SEP002155D54EC9

Display (Caller ID) Display text for a line appearance is intended for the caller.

ASCII Display (Caller ID)

Line Text Label

ASCII Line Text Label

External Phone Number Mask

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 Applies to this line when any line on the phone has a call in progress.

Call Pickup Group Audio Alert Setting(Phone Idle) Use System Default

Call Pickup Group Audio Alert Setting(Phone Active) Use System Default

Recording Option* **Selective Call Recording Enabled**

Recording Profile

Monitoring Calling Search Space

Log Merged Calls

254323

Configure Recording Media Source for a Line Appearance

Add the Record Softkey or Programmable Line Key to the Device Template (Optional)

To enable a user to start and stop recording from a Cisco IP device, add the Record softkey or programmable line key to the device template.

To add the Record softkey, use the **Device > Device Settings > Softkey Template** menu option in Cisco Unified Communications Manager Administration to create or modify a nonstandard softkey template. Configure the softkey layout for call state *connected* to have the Record softkey in the selected softkeys list.

To add the Record programmable line key, use the **Device > Device Settings > Phone Button Template** menu option in Cisco Unified Communications Manager Administration. Enter the button template name, feature, and label.

Configure IP Phone as Recording Media Source (Optional)

Turn on IP Phone BIB to allow Recording

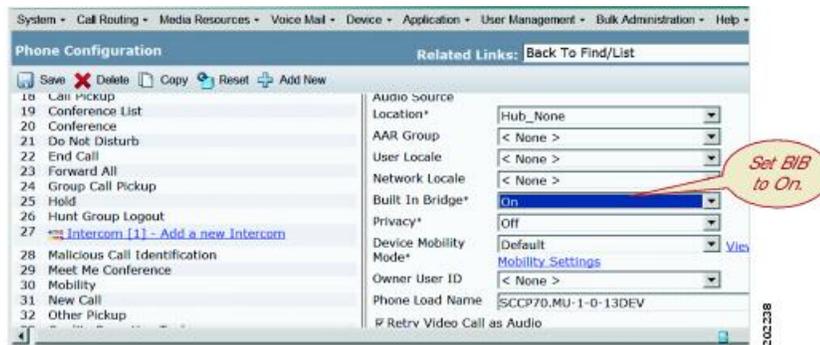
The built-in bridge (BIB) of the agent phone must be set to **On** to allow its calls to be recorded.

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

Use the **Device > Phone** menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.

Figure 40: Setting the Phone Built In Bridge to On

The following figure illustrates how to turn on the IP phone BIB to allow recording.



Configure Recorder as SIP Trunk Device

The administrator needs to enable call recording for a SIP trunk pointing to a recording enabled SIP IOS gateway, in order to use the gateway for the media forking.

Cisco Unified Communications Manager registers with the gateway as a XMF Application at initialization time or when the SIP trunk is reset, after the SIP trunk is enabled for gateway recording. The corresponding settings on the gateway for the XMF provider needs to be configured for Cisco Unified Communications Manager to register with the gateway.

Cisco Unified Communications Manager requests the gateway to start media forking with a XMF media-forking request when the recording is triggered.

Recording related configuration for a SIP trunk is grouped in the **Recording Information** section on the **SIP trunk** page. To enable the SIP trunk for gateway recording, the administrator needs to check the checkbox **This trunk connects to a recording-enabled gateway**. To enable inter-cluster gateway recording for a SIP trunk, the administrator check the checkbox **This trunk connects to other clusters with recording-enabled gateways**.

Configure Recording Notification Tones

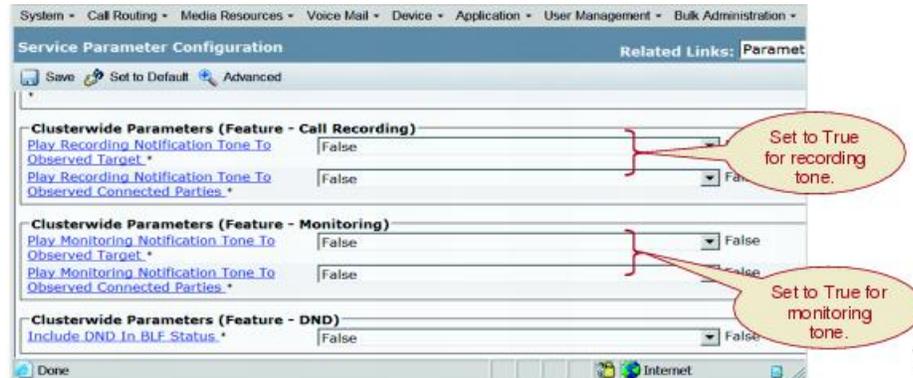
Set the service parameters for playing tone to **True** to allow tones to be played to the agent, the customer, or both.

CTI applications that initiates monitoring can also pass the play tone option to Cisco Unified Communications Manager. The recording tone plays when either the service parameter or the CTI application specifies the play tone option.

Use the **System > Service Parameters** menu option in Cisco Unified Communications Manager Administration to perform the necessary configuration.

Figure 41: Using Service Parameters to Configure Tones

The following figure illustrates using service parameters to configure tones.



Set the Recording Service Parameters

This section describes the service parameters that are related to the call recording feature.

Notification

The following service parameters affect the playing of notification tones to the parties in the call being recorded by the call recording feature:

Clusterwide Parameters (Feature - Call Recording)

- Play Recording Notification Tone To Observed Target
- Play Recording Notification Tone To Observed Connected Parties

The default value for these service parameters is **False**. You must change the value of each parameter to **True** to enable the particular notification tone to play.

Built-In-Bridge

The following service parameter enables or disables the built-in bridge on phones:

Clusterwide Parameters (Device - Phone)

Built-in Bridge Enable

For more information about this service parameter, see "Turn on IP phone BIB to allow Monitoring".

Simultaneous Monitoring and Recording



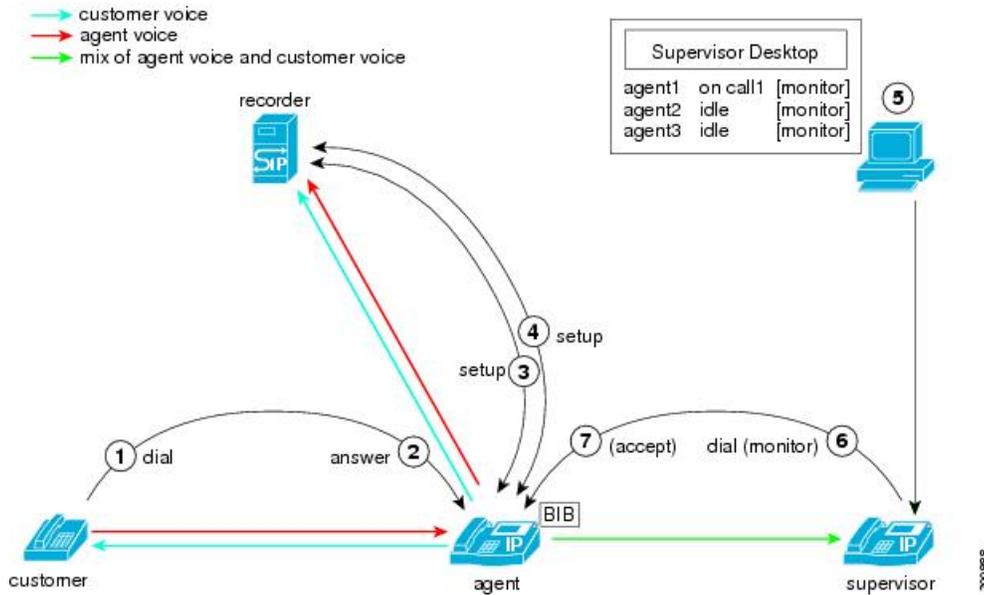
Note Recording a monitored call transferred or redirected across a cluster fails, as the DN information is not passed across the cluster for monitoring.

Recording can take place when the agent call is being monitored.

Recording and monitoring get set up independently from each other

The following figure illustrates simultaneous monitoring and recording.

Figure 42: Simultaneous Monitoring and Recording



In the case of simultaneous monitoring and recording, the following steps take place:

1. A customer calls into the call center.
2. The call routes to agent. Agent answers the call. A two-way media streaming gets set up between agent IP phone and the customer.
3. The recording call for the agent voice gets set up between agent phone and the recorder.
4. The recording call for the customer voice gets set up between agent phone and the recorder.
5. The supervisor desktop application shows that agent has an active call. On his desktop application, the supervisor clicks the monitor button for current active call of agent.
6. The supervisor IP phone gets triggered to go off hook and make a monitoring call toward agent.
7. Agent phone accepts the monitoring call. Agent phone starts to send a stream of mixed voices of the customer and the agent to the supervisor IP phone. Neither the agent nor the customer can hear the supervisor.