Call Recording and Monitoring

Revised: March 1, 2018

Call monitoring and recording solutions provide a way to monitor and record audio and video calls that traverse various components in a Unified Communications and Collaboration solution, such as Cisco IP Phones, Cisco Unified Border Element devices, or Cisco switches. These recordings can then be used by call centers and other enterprise functions for various purposes such as compliance, transcription, speech analysis, podcasting, and blogging. This chapter provides an overview of various call recording solutions available for Cisco Unified Communications and Collaboration solutions for both audio and video calls. The chapter also outlines basic design considerations for call recording solutions embedded within a Cisco Unified Communications and Collaboration solution.

What's New in This Chapter

Table 23-1 lists the topics that are new in this chapter or that have changed significantly from previous releases of this document.

Table 23-1 New or Changed Information Since the Previous Release of This Document

New or Revised Topic	Described in:	Revision Date
Cisco MediaSense has reached end of sale (EoS) and has been removed from this chapter.	For information on Cisco MediaSense, refer to previous versions of the SRND, available at https://www.cisco.com/go/srnd.	March 1, 2018
Cisco TelePresence Content Server (TCS) has reached end of sale (EoS) and has been removed from this chapter.	For information on Cisco TCS, refer to previous versions of the SRND, available at https://www.cisco.com/go/srnd.	March 1, 2018

Types of Monitoring and Recording Solutions

This section describes the following types of call recording and monitoring solutions:

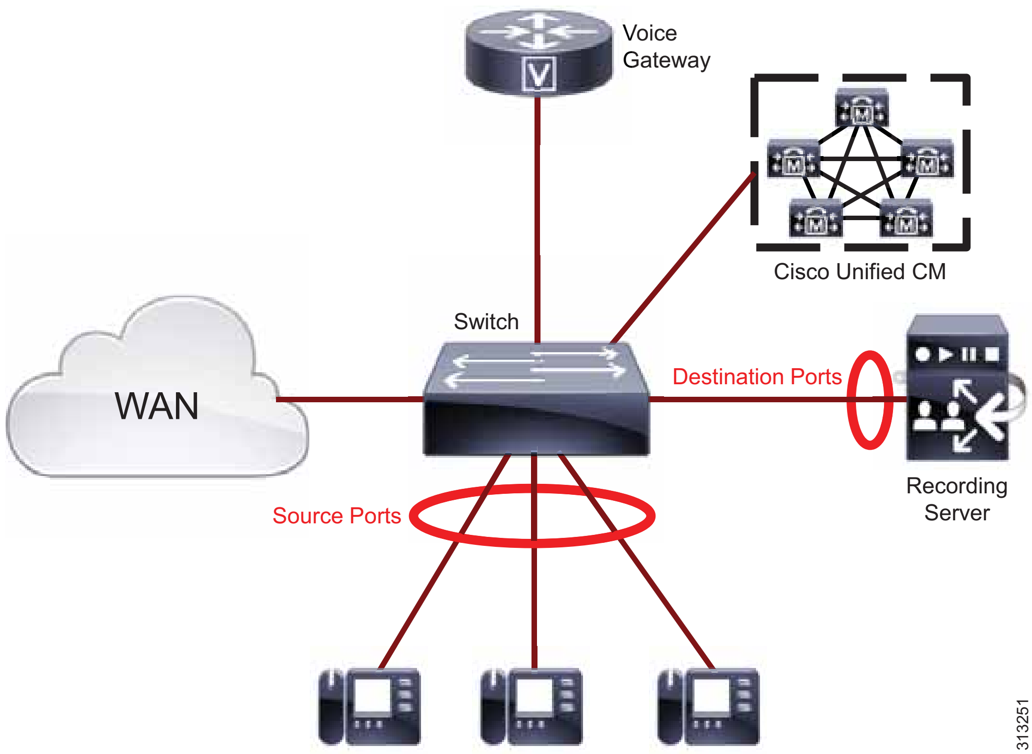
- SPAN-Based Solutions, page 23-2
- Unified CM Silent Monitoring, page 23-4
  - Unified CM Network-Based Recording, page 23-4
  - Unified CM Network-Based Recording with Built-in Bridge, page 23-6
  - Cisco Unified CM Network-Based Recording with a Gateway, page 23-7
- Agent Desktop, page 23-10

SPAN-Based Solutions

Recording solutions based on a Switched Port Analyzer (SPAN) use the packet sniffing technology for recording calls. SPAN is a method of monitoring network traffic. When SPAN is enabled on a switch port or VLAN, the switch sends a copy of all network packets traversing that port or VLAN to another port where a recording or monitoring server (such as Cisco Unified Workforce Optimization Quality Management or a third-party recording server, for example) analyzes those packets. It detects and decodes the VoIP RTP packets embedded in the network traffic and stores them as audio on a storage device. SPAN can be enabled on the ports connected to a Cisco Voice Gateway or Cisco IP Phones, as required. For example, for recording internal calls between IP phones, SPAN should be enabled on switch ports connected to the IP phones.

Figure 23-1 illustrates a SPAN-based recording solution deployment for recording internal calls. The ports marked as source ports connected to IP phones are mirrored to the destination port connected to the recording server.

Figure 23-1 SPAN-Based Recording Call Flow for Internal Calls



Several Cisco partners provide SPAN-based recording servers and applications for Cisco Unified Communications and Collaboration solutions. For technical details, refer to the specific partner product information in the *Cisco Developer Network Marketplace Solutions Catalog*, available at

https://marketplace.cisco.com/catalog/search?utf8=%E2%9C%93&x=48&y=6&search%5Btechnology_category_ids%5D=1900

In addition, network traffic flow needs to be considered for appropriate bandwidth provisioning when port mirroring is enabled.

SPAN-Based Recording and Virtualization

This section reviews some common SPAN-based deployments with virtualization enabled and lists some of the limitations. VMware provides support for the SPAN feature on VMware vSphere Distributed Switch (VDS) starting with vSphere 5.0.

In a virtualized setup, some of the Unified Communications applications, contact center applications, and the port analyzer application may be deployed on virtual machines on the same host or on different hosts. There are some limitations to SPAN-based recording solutions in a virtualized setup. For example, the following features are not supported for deployments of Cisco Unified Contact Center Enterprise (Unified CCE) with virtualization:

- Remote silent monitoring
- SPAN-based silent monitoring and recording on Cisco Unified Computing System (UCS) B-Series chassis

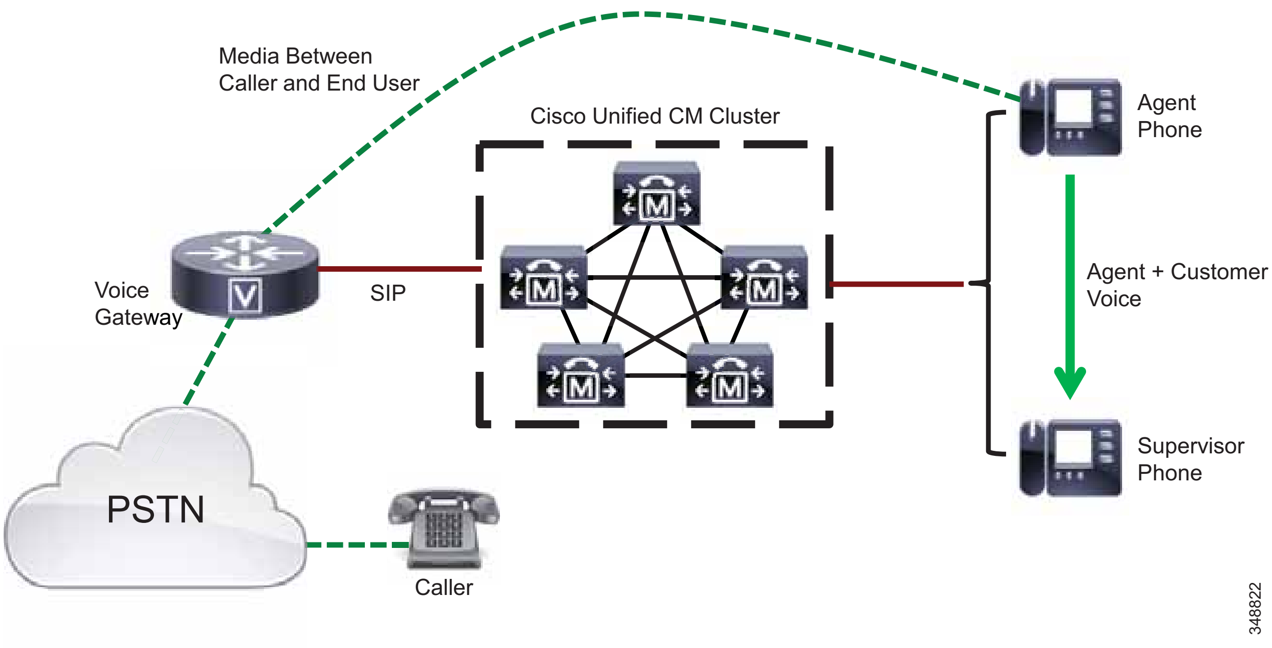
Note SPAN-based silent monitoring and recording is not supported on the UCS B-Series chassis.

Unified CM Silent Monitoring

The Unified CM Silent Monitoring feature allows a supervisor to listen to a conversation between an agent and a customer with neither the agent nor the customer aware of the supervisor's presence on the call. During call monitoring, the agent phone combines the two voice RTP streams (one for the agent and one for the customer) on the agent phone and sends the resulting stream to the supervisor phone. In addition, whisper coaching allows the supervisor to talk to the agent during the call monitoring session. Call monitoring and whisper coaching can be invoked by call center applications through the JTAPI or TAPI interfaces of Unified CM.

Figure 23-2 illustrates the basic setup for Cisco Unified CM Silent Monitoring.

Figure 23-2 Unified CM Silent Monitoring Architecture



Unified CM Network-Based Recording

The Unified CM network-based recording feature allows system administrators to record conversations between calling and called parties. Network-based recording allows for forking media using either the built-in bridge (BIB) of a supported IP phone model or a SIP gateway of a supported version and configuration. The administrator can set a preference to one forking device type or the other; however, if the preferred forking device is not available, Unified CM automatically fails over to the other method. For example, if an IP phone has recording enabled with **Phone Preferred** but there is no recording resource available (the phone does not have a built-in bridge), the gateway would be used for call recording.

Regardless of the media forking devices used by Unified CM for call recording, Unified CM always provides the metadata about the near-end and far-end parties of the recorded calls to the recording server. The metadata resides in the FROM header of the SIP Invite and other SIP messages that are sent between Unified CM and the recording server.

For details about Unified CM silent call monitoring and call recording features, refer to the latest version of the *Feature Configuration Guide for Cisco Unified Communications Manager*, available at

https://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-installation-and-configuration-guides-list.html

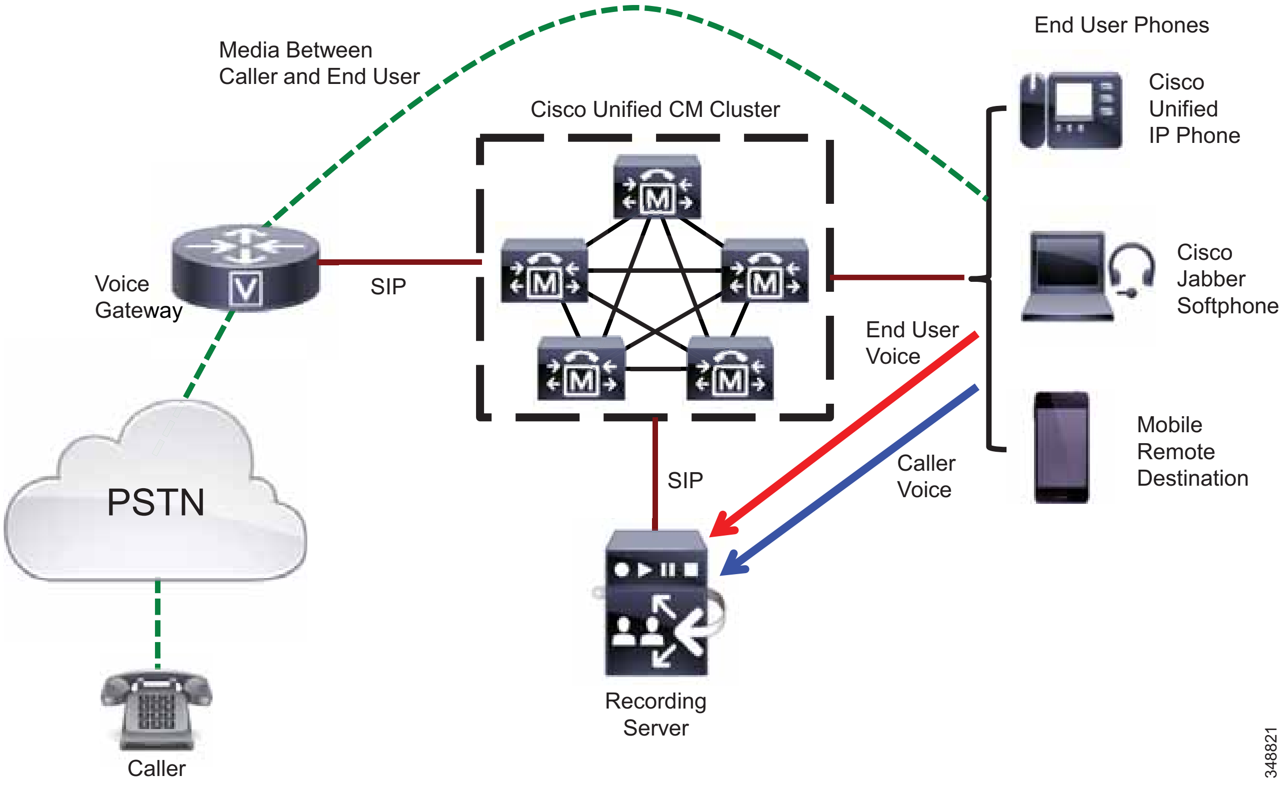
Cisco Unified CM network-based recording supports automatic and selective recordings for each individual line instance. This is accomplished by assigning a Recording Profile to each instance of a line where recording is required. This allows for recording on a single line of a multi-line device or a single instance of a shared line. In automatic recording, Unified CM automatically records every call that is connected on the endpoint. In selective recording, the user or an external application via JTAPI/CTI has to explicitly request Unified CM to start the recording for the call on the endpoint. Users can make the recording request by pressing the Start Recording button on the endpoint or by sending the recording request from the JTAPI or TAPI application. To start the recording, Unified CM sends the request to the forking device to fork the media of the conversation to the recording server, where the media is recorded.

Note If you have enabled both call recording and Multilevel Precedence and Preemption (MLPP), the lines that use both features will generate two additional call legs. Therefore, you must set the busy trigger for those lines to 3.

Unified CM Network-Based Recording with Built-in Bridge

Cisco Unified CM network-based recording with BIB uses the IP phone's built-in bridge to enable call recording. (See Figure 23-3.) During call recording, the agent phone forks the two streams to the recording server. The two streams, one for the called party's voice and one for the calling party's voice, get recorded separately. If a single stream is desired, customers can use third-party applications to mix the recorded streams to produce the conversation.

Figure 23-3 Unified CM Network-Based Recording Using a Phone's Built-in Bridge



For a list of Cisco Unified IP Phones that support call monitoring and recording with Unified CM, refer to the *Unified CM Silent Monitoring/Recording Supported Device Matrix*, available at

https://developer.cisco.com/site/uc-manager-sip/documents/supported/

Cisco Unified CM Network-Based Recording with a Gateway

When a call passes through a recording gateway, Cisco Unified CM network-based recording can utilize the gateway's media forking capability for call recording. When an external call is connected with an end user on the phone, Unified CM requests the gateway to fork the media of the conversations to the recording server through the UC Gateway Services API running on the gateway. The forked media consists of two RTP streams, one for end user voice and one for caller voice, and the recording server captures the streams separately. When a recording-enabled gateway is part of a call, several recording scenarios are possible, including external calls connected with end users on Cisco Unified IP Phones, Cisco Softphone (Cisco Jabber, for example) running on a PC, mobile phones as remote destinations, CTI ports, and Extend and Connect destinations. Essentially, once an external call terminates on the voice gateway that Unified CM is registered with, the entire conversation of the call from the caller's perspective can be recorded, no matter where the call goes inside the enterprise.

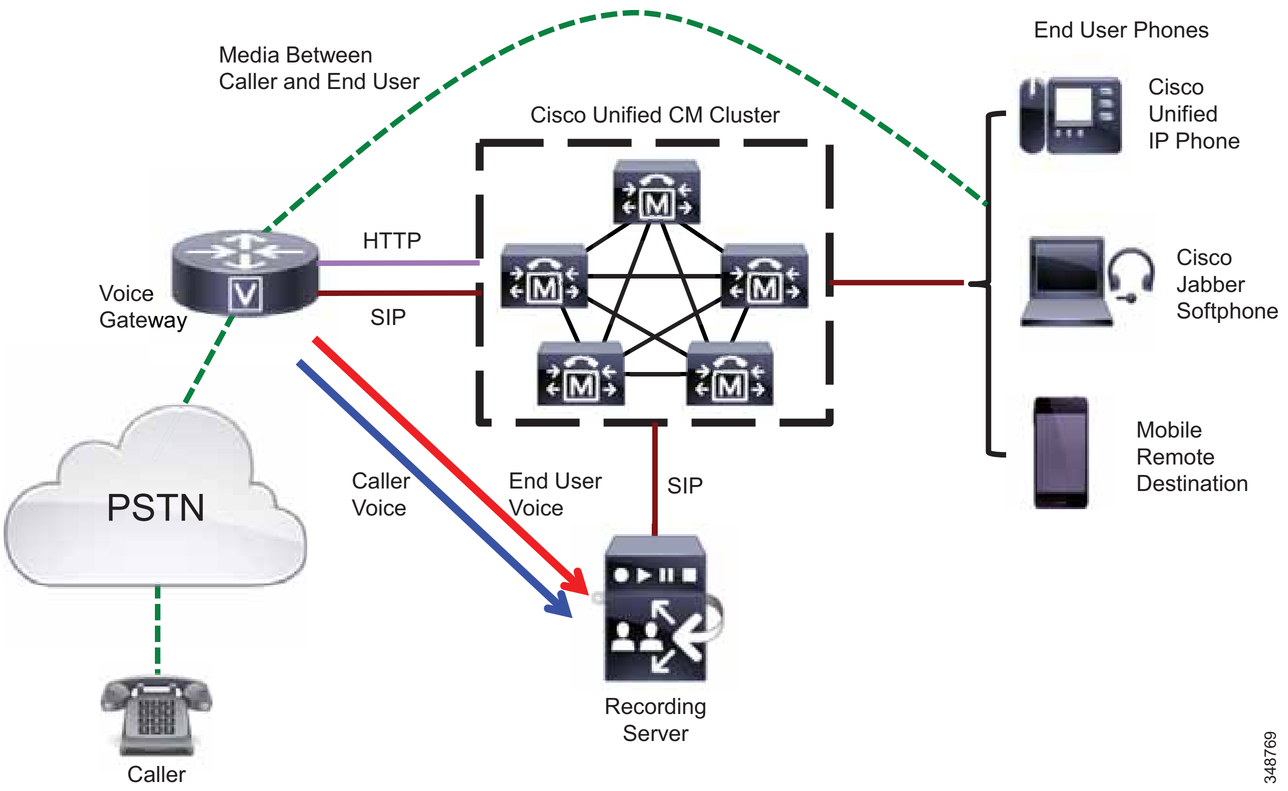
Cisco Unified CM network-based recording supports additional call types other than the ones described above. For details, refer to the latest version of the *Feature Configuration Guide for Cisco Unified Communications Manager*, available at

https://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-installation-and-configuration-guides-list.html

Note Invoking media forking from a voice gateway produces two RTP streams, and if silent monitoring is required, the application is responsible for mixing the streams.

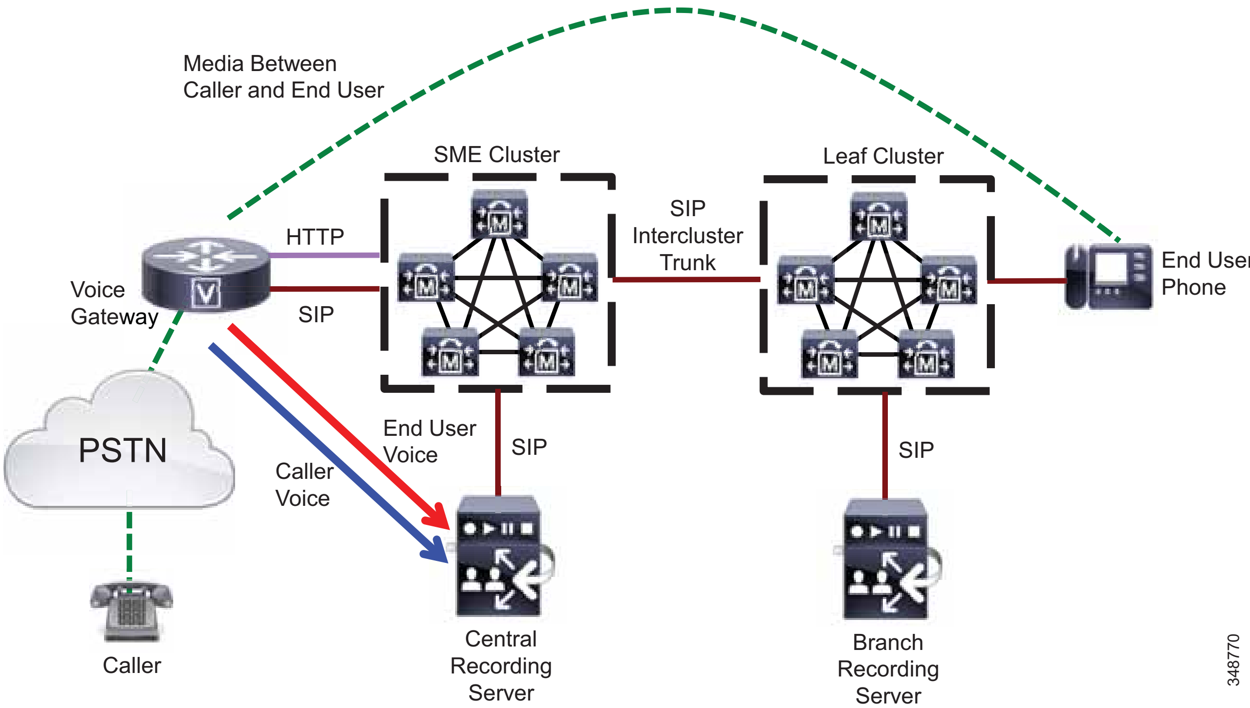
Figure 23-4 illustrates the basic setup for Cisco Unified CM network-based recording using gateways. Cisco Unified CM and the voice gateway are connected through a recording-enabled SIP trunk. Unified CM registers with the UC Gateway Services API running on the gateway through its HTTP interface. This enables Unified CM to receive call event notifications for all calls passing through the gateway and to decide when to start or stop the recording. Depending on the recording option configured, when a gateway call is connected with an end user on the phone, Unified CM might notify the gateway immediately to fork the media or wait for the user indication to start the recording before notifying the gateway. Unified CM notifies the gateway to stop forking the media upon user indication to stop the recording, or the gateway automatically stops the recording upon call termination. The requests to start or stop the recording are sent over the HTTP interface using the Extended Media Forking (XMF) API.

Figure 23-4 Network-Based Recording with a Gateway



With Unified CM network-based recording with a gateway, the end user phone and the media forking device (voice gateway) are decoupled. They can register to the same Unified CM cluster (as shown in Figure 23-4) or to separate Unified CM clusters. Therefore, this solution could be deployed in a multi-cluster environment such as Cisco Unified CM Session Management Edition (SME). Figure 23-5 illustrates an example of deploying Unified CM network-based recording with SME, where the voice gateway registers to the SME cluster and the end user phone registers to the leaf cluster. The SME cluster and leaf cluster are connected by a SIP intercluster trunk (ICT) with the gateway recording option enabled on both sides. Thus, the recording invocation requests and responses can be sent between SME and leaf clusters. Also, customers have the option to deploy the recording server centrally in the SME cluster with the voice gateway or to distribute the recording servers in all the leaf clusters.

Figure 23-5 Cisco Unified CM Network-Based Recording Deployment with SME



When deploying Unified CM network-based recording with a gateway, observe the following guidelines:

- Network-based recording with a gateway is supported on a variety of platforms including Cisco Integrated Services Routers (ISRs) (for example, ISR 4K) and Cisco Aggregation Services Routers (ASRs). For detailed requirements, refer to the latest version of the *Feature Configuration Guide for Cisco Unified Communications Manager*, available at

  https://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-installation-and-configuration-guides-list.html

- Only SIP is supported between the voice gateway and Cisco Unified CM, and SIP proxy servers are not supported.
- For inter-cluster recording, only a SIP trunk is supported to interconnect the clusters.
- Secure recording is not supported.
- IPv6 is not supported.

Agent Desktop

Agent desktop monitoring and recording solutions are specific to contact center deployments that enable supervisors to do silent monitoring and initiate call recording when needed. Several agent desktop monitoring and recording solutions are available, such as:

- Cisco Agent Desktop (CAD) Silent Monitoring and Recording
- Cisco Remote Silent Monitoring (RSM)

These solutions are described in detail in the latest version of the following documents:

- *Solution Design Guide for Cisco Unified Contact Center Enterprise*, available at

  https://www.cisco.com/c/en/us/support/customer-collaboration/unified-contact-center-enterprise/products-implementation-design-guides-list.html

- *Solution Design Guide for Cisco Unified Contact Center Express*, available at

  https://www.cisco.com/c/en/us/support/customer-collaboration/unified-contact-center-express/products-implementation-design-guides-list.html

Capacity Planning for Monitoring and Recording

Enabling any type of monitoring and/or call recording impacts the overall Unified Communications system capacity. Some silent monitoring and recording solutions (such as the silent monitoring and recording feature based on Unified CM) consume resources from Unified CM, whereas other solutions such as SPAN or desktop silent monitoring and recording do not. Consider the following points when doing capacity planning for Unified Communications systems with call recording enabled:

- With Unified CM call recording, each recorded call adds two calls to the call processing component BHCA capacity. Forking media from an IP phone or voice gateway consumes resources from Unified CM or the voice gateway, respectively.
- Bandwidth requirements increase when media forking is enabled on IP phones or Cisco Unified Border Element devices to send forked media to the recording server. In case of agent desktop monitoring and recording, the bandwidth utilization can be bursty, depending on how many calls are being monitored or recorded at a given time.
- Call recording using a Cisco Unified Border Element doubles the weight of a call. Thus, call capacity would be cut in half if all calls passing through the Cisco Unified Border Element were recorded.
- Memory utilization on Cisco Unified Border Element increases for each call that is recorded.
- In cases where CTI applications interact with Cisco Unified CM to invoke recording and monitoring, you should consider the Unified CM cluster deployment model and load-balance the CTI applications across the cluster.

Due to the complexity associated with sizing, all deployments must be sized with the Cisco Collaboration Sizing Tool, available to Cisco employees and partners only (with proper login authentication) at

https://cucst.cloudapps.cisco.com/landing