CHAPTER 4

# Design Considerations

## QoS Recommendations

Retailers have many business applications that have different requirements for priority when traffic congestion occurs. A strategic QoS deployment will allow for an enhanced customer/retailer user experience. Typically, Enterprise retailers are not cognizant of all of the business applications that traverse the network as many applications get deployed by non IT departments or by IT departments that fail to involve Network staff for QoS considerations. As a result, some applications will greatly disrupt the performance of other well behaved applications when they are anonymously deployed. The ability of the retailer to successfully plan, implement and manage Enterprise scale QoS deployments, given the lack of its own application knowledge, is compounded by the complexity and technical knowledge required by the network staff to fully comprehend QoS. Improperly configured routers and switches are as big a threat to performance as rogue applications. Network staff should prioritize traffic by exclusion, meaning that they classify and prioritize the known important applications explicitly (e.g., voice, video and POS, leaving remaining traffic to participate in the best effort queue.

Retailers and service providers are encouraged to adopt RFC 4594 provisioning recommendations with the aim of improving QoS consistency, compatibility, and interoperability. Since these are guidelines and not standards, modifications can be made to these recommendations as specific needs or constraints require. A summary of Cisco's implementation of RFC 4594 is presented in Table 4-1.

*Table 4-1 Cisco Differentiated Services (DiffServ) QoS Recommendations for Medianets*

| Application Class | Per-Hop Behavior | Admission Control | Queuing and Dropping |
|---|---|---|---|
| VoIP Telephony | EF | Required | Priority Queue (PQ) |
| Broadcast Video | CS5 | Required | (Optional) PQ |
| Real-Time Interactive | CS4 | Required | (Optional) PQ |
| Multimedia Conferencing | AF4 | Required | BW Queue + DSCP WRED |
| Multimedia Streaming | AF3 | Recommended | BW Queue + DSCP WRED |
| Network Control | CS6 | | BW Queue |
| Call Signaling | CS3 | | BW Queue |
| Ops/Admin/Mgmt (OAM) | CS2 | | BW Queue |
| Transactional Data | AF2 | | BW Queue + DSCP WRED |
| Bulk Data | AF1 | | BW Queue + DSCP WRED |
| Best Effort | DF | | Default Queue + RED |
| Scavenger | CS1 | | Min BW Queue |

The method of QoS used in the testing lab was based on the Cisco Enterprise Quality of Service reference design as shown in Table 3 - Cisco Enterprise Quality of Service.

For more information on QoS, see the following:

- Cisco Enterprise Quality-of-Service

  http://www.cisco.com/en/US/solutions/ns340/ns414/ns742/ns817/landing_voice_video.html
- QoS Design Recommendations for Medianets

  http://www.cisco.com/en/US/docs/solutions/Enterprise/Video/qosmrn.html
- Cisco Telepresence Design Guide - QoS

  http://www.cisco.com/en/US/docs/solutions/Enterprise/Video/tpqos.html
- Network Ports Used by Cisco Unified Personal Communicator

  http://www.cisco.com/en/US/docs/voice_ip_comm/cupc/7_0/english/release/notes/ol15710.html

# Bandwidth Considerations

Each of the video options in the solution have differing bandwidth requirements which can dramatically affect the design and deployment scenarios. Table 4-2 provides a brief overview of the different requirements for each product.

*Table 4-2 Bandwidth Requirements*

| Product / Platform | Bandwidth Audio & Video | Resolution | Frame Rate | Notes |
|---|---|---|---|---|
| TelePresence-500 Best | 4128Kbps | 1920x1080p | 30fps | Best Quality |
| TelePresence-500 Better | 3628Kbps | 1920x1080p | 30fps | Better Quality |

*Table 4-2 Bandwidth Requirements (continued)*

| | | | | |
|---|---|---|---|---|
| TelePresence-500 Good | 3128Kbps | 1920x1080p | 30fps | Good Quality |
| TelePresence-500 Best | 2378Kbps | 1280x720p | 30fps | Best Quality |
| TelePresence-500 Better | 1628Kbps | 1280x720p | 30fps | Better Quality |
| TelePresence-500 Good | 1128Kbps | 1280x720p | 30fps | Good Quality |
| TelePresence-500 Lite | 936Kbps | 1280x720p | 30fps | Extended Reach |
| Video Advantage | 384-1500Kbps | 352x288,320x240, 176x144, and 160x120 | up to 30fps | |
| CUPC Video | 384-512Kbps | 352x288 / 176x144 | up to 30fps / 15fps | |
| Cisco 9900 Series Phones | up to 1000Kbps | up to 640x480 | 30fps | |
| Cisco 7985 Series Phones | up to 768Kbps | 352x240 | 30fps | |
| Cisco WebEx | 384Kbps | 320x240 | 7-15fps | |

Table 4-2 represents raw bandwidth requirements and also needs to accommodate additional IP overhead.

For more information on Cisco TelePresence resolution and bandwidth, refer to the following URL: http://www.cisco.com/en/US/docs/solutions/Enterprise/Video/tpover.html#wp1043742

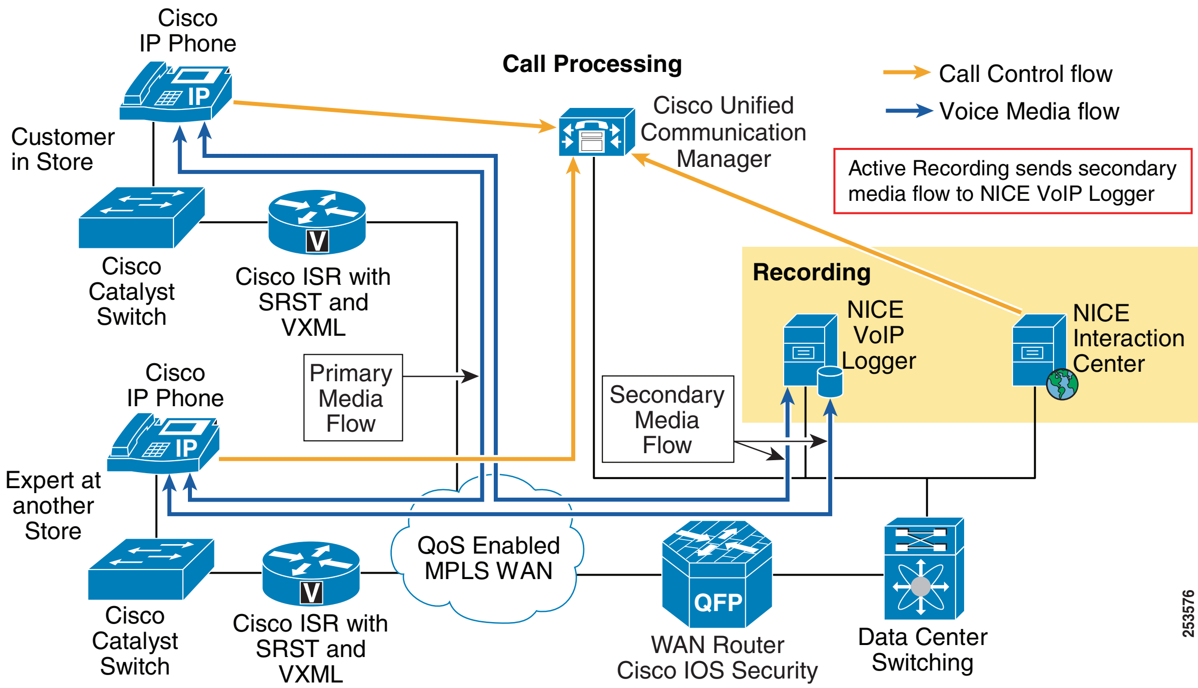
# NICE Perform Release 3.2

NICE Perform can support many methods of recording including distributed, centralized and active recording. This solution validated the functionality of both centralized recording using a trusted ready point and monitor port as well as active recording, a feature on newer Cisco 7975, 7965 and 7945 series phones.

## Active Recording

When the customer talks to the expert agent, the Cisco Unified Communications Manager (CUCM) sets up an additional call between the agent's phone and the NICE VoIP Logger. The voice itself is replicated at the phone's BIB (Built in Bridge) and sent to the VoIP Logger IP address. Figure 4-1 shows the call flows for call control and the media flows between devices and the NICE VoIP logger.

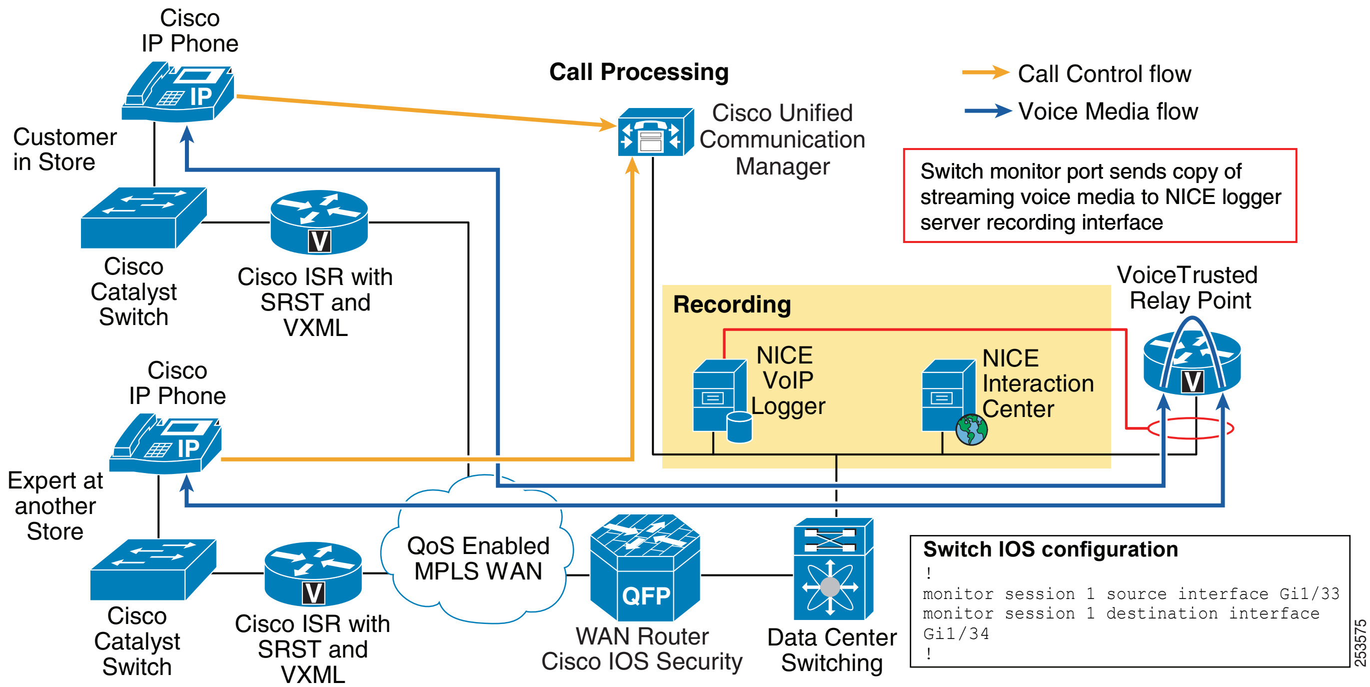
*Figure 4-1 Active Recording to NICE VoIP Logger in Data Center*




# Centralized Recording

When the customer talks to the expert agent, the Cisco Unified Communications Manager (CUCM) routes all calls for these stations through a trusted relay point. The voice itself is then replicated at the switches interface port of the trusted relay point and sent to the VoIP Logger monitoring interface. Figure 4-2 shows the call flows for call control and the media flows between devices.

*Figure 4-2 Passive Recording via Trusted Relay Point in Data Center*



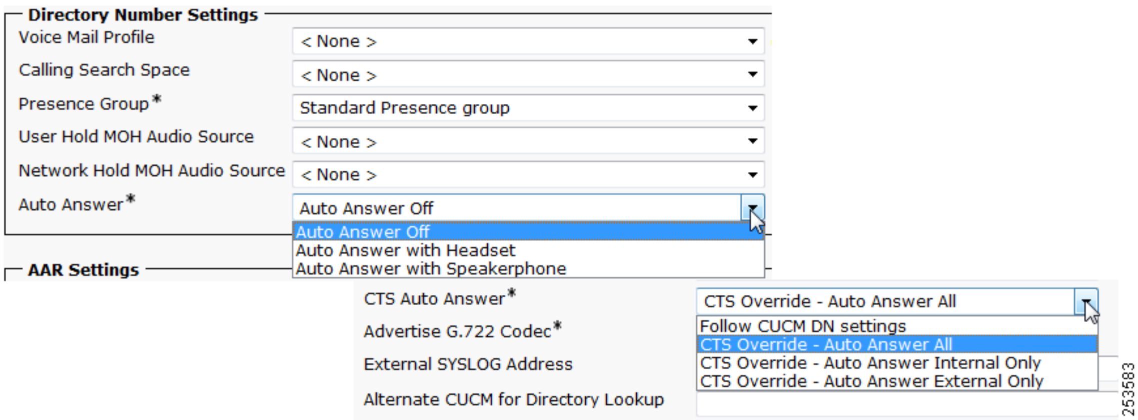

# Lessons Learned

## TelePresence Lessons Learned

The video portion of a TelePresence call will not work with a regular deployment when following the configuration guide *Cisco Unified Communications Manager Configuration Guide for the Cisco TelePresence Systems* http://www.cisco.com/en/US/partner/docs/telepresence/cucm_cts/1_6/configuration/guide/cucm_cts_1_6.html.

In the above guide, the directory number used by the CTS unit and associated 7975 handset is a shared line. Within the installation steps, the directory number is specified to be configured as **Auto Answer with Speakerphone** in the Auto Answer drop-down menu. This configuration causes problems when calls are routed over SIP trunks to other systems (e.g., between Call Manager clusters to CVP queue, etc.). The Cisco 7975 phone will auto answer before the CTS unit as the transferred SIP call will appear as an audio-only call. In order to have the CTS unit answer the call and perform a reinvite, adding video back to the call between the endpoints, the directory number must be configured with **Auto Answer off** and in the CTS unit Product Specific Configuration Layout section of the CTS device configuration set the CTS Auto Answer drop-down menu to **CTS Override - Auto Answer All**. See Figure 4-3.

*Figure 4-3 Configuring Directory Number*




## Trusted Relay Point for Calls

The Cisco Unified Communications Manager enables the insertion of trusted relay points (TRPs). The insertion of TRPs into the media path is most often used in a network virtualization environment and when QoS enforcement is needed. In the Cisco Virtual Expert Management solution, the TRP is used to reroute the media stream from the call endpoints and force them to flow through the TRP. On the switch port where the TRP connects a SPAN session is created to mirror all traffic to the NICE recording server. This is a cost-effective way to implement a centralized recording solution for non-encrypted audio calls on devices that do not support duplicate audio streams (e.g., Cisco 7985, 7960, etc) like the newer Cisco 7975 phones. With proper decode codec support on the recording server, even TelePresence calls can be recorded. A TRP can be configured on Cisco ISR routers with Voice IOS software. A typical TRP configuration in IOS would look as follows:

```
!
sccp local FastEthernet0/0
sccp ccm 192.168.45.182 identifier 1 version 7.0
sccp
!
sccp ccm group 1
associate ccm 1 priority 1
associate profile 1 register MTP-01
!
dspfarm profile 1 mtp
codec g711ulaw
codec pass-through
maximum sessions software 110
associate application SCCP
!
```

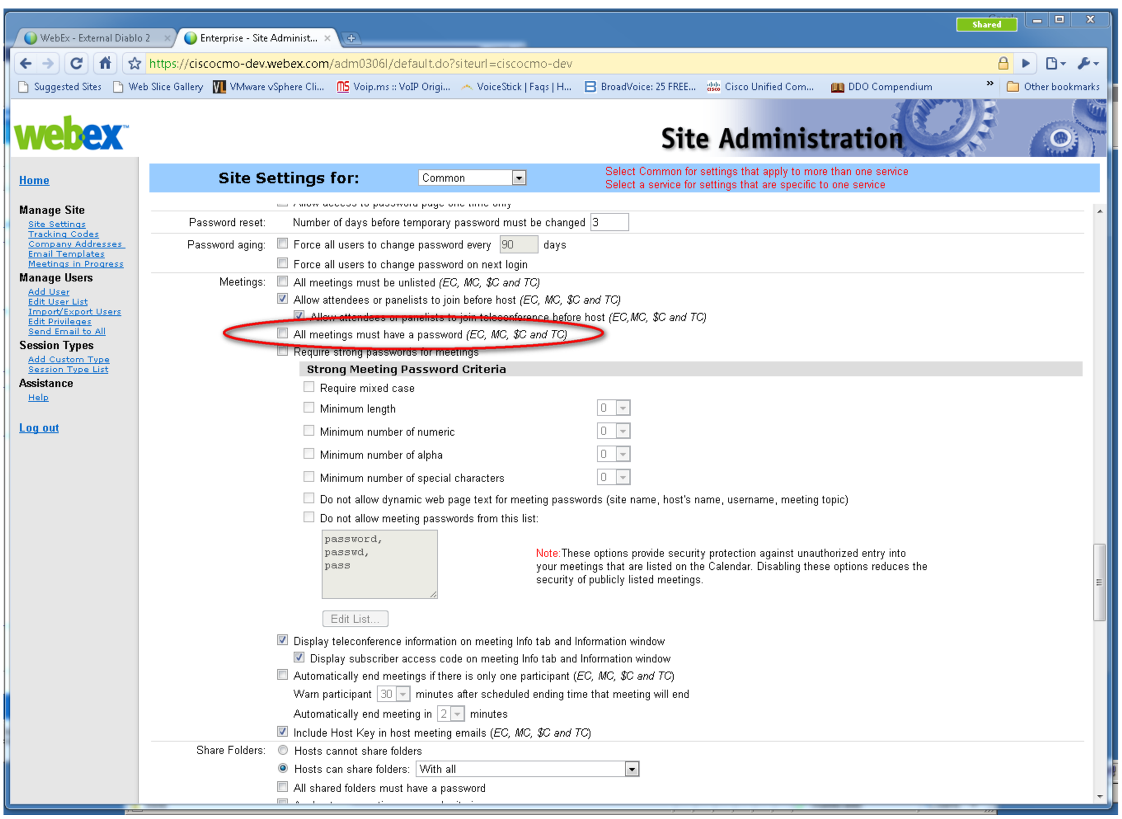
Once the TRP is configured, a Media Termination Point (MTP) is added to the CallManager under the Media resources menu. The MTP name must match the register name specified on the TRP. After the TRP is configured and registered, each phone can be configured to use the TRP individually or based on a device pool.

For more information on configuring TRPs and MTPs, see Media Resources in *Cisco Unified Communications Manager System Guide for Cisco Unified Communications Manager Business Edition, Release 7.1(2)* at the following URL:
http://www.cisco.com/en/US/docs/voice_ip_comm/cucmbe/admin/7_1_2/ccmsys/a05mtp.html#wpxref35934

# CUPC Instant Meetings

One of the primary advantages of using CUPC client over other IM services is the ability to start an instant meeting with the other party of a VEM session. When a customer call is passed from the expert queue to the Expert Agent the CUPC clients on both systems share call information and enable the start of an instant meeting using either Cisco MeetingPlace or Cisco WebEx Meeting. When using Cisco WebEx Meeting there are several configuration items that need to be taken into consideration. The CUPC client does not use the stations proxy configuration settings when launching a meeting. If your enterprise uses a proxy when connecting to Internet sites, special considerations need to be taken for connecting to the WebEx meeting servers. Additionally, version 7.02 and later of the CUPC client is not able to dynamically generate a meeting password for instant WebEx meetings. Consequently, when using CUPC the requirement for a meeting password in the WebEx site administration must be disabled to use WebEx meetings for collaboration in the Virtual Expert solution. See Figure 4-4.

***Figure 4-4 Instant Meetings***

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# Summary

Retailers that want to be perceived as delivering value beyond low prices, such as product expertise or specialized services, can benefit from implementing the Cisco Virtual Expert Management solution. This solution performed well in face-to-face consultation through video, voice, and content sharing between stores with the most appropriate subject matter expert. Its ability to locate and seamlessly connect experts across the enterprise using skill-based routing and presence availability-enabled stores to capture the same business opportunity without deploying subject-matter experts at every store. The Cisco Virtual Expert Management solution helps retailers differentiate themselves by redefining superior customer service expertise.

# Partner Profile

## NICE Solutions for Financial Institutions

Complying with regulations, improving customer retention, and enhancing operational efficiency is critical in today's complex business environment. Achieving these goals while reducing expenses in a tough economic climate is no easy task. To empower organizations and attain these business objectives, NICE has developed a set of innovative enterprise solutions for contact centers, branches (including home agents and backoffice operations), and trading floors.

NICE's enterprise solutions deliver the powerful tools necessary to address critical business needs including the following:

- Compliance and risk management
- Customer retention and insight
- Operational efficiency

For more information about the NICE Solutions refer to the following URL:
http://www.nice.com/solutions/enterprise/index.php

### IP Phone-Based Active VoIP Recording

NICE active VoIP recording enables the delivery of centralized recording capabilities in distributed environments. All NICE Perform servers are consolidated in the data center, where all calls that take place in the organization's branches and other remote locations are recorded.

By reducing the need for costly branch set-up, administration, and management of recording servers, NICE helps to flatten the organization and enables simple, yet efficient handling of remote employees.

This IT-friendly technology makes active VoIP recording the ideal solution even for single-site operations. NICE's solution for IP-phone-based active recording for Cisco Unified Communications Manager (CUCM) is another component of NICE's extensive portfolio of active recording integrations.

# Solution Benefits

## Consolidation

NICE active recording for Cisco enables the centralization of the recording system in the data center, in a similar manner to the centralization of the CUCM servers. This allows the organization to benefit from reduced expenses and to enjoy both economies of scale and the lower support costs, thanks to simplified and consolidated administration, management, and maintenance.

## IT Friendliness

Passive VoIP recording requires the use of mirroring ("SPAN") sessions. These sessions have to be maintained for supporting moves, adds and changes of the telephony and data networks. This may conflict with organizations' IT policies. IP-phone based active recording does away with the need for mirroring sessions, thus reducing the network management load on IT staff.

## Lower Total Cost of Ownership

NICE offers a reduced footprint, industry standard servers, the highest number of recording channels per server, and advanced compression capabilities that reduce long-term storage volume and ensure lower ownership costs.

## Freedom from Size Limitations

The NICE integrated recording solution meets the recording needs of all sizes and kinds of business, from small enterprises recording a few dozen phones to large single-site and multi-site operations with tens of thousands of phones.

## Cohesive, Integrated Solution Suite

NICE meets all the organization's call recording requirements. The same system can support all recording modes—Total recording, user initiated recording-on-demand, and rule-based recording, including sampled recording for quality management in contact center environments.

NICE offers a unified solution for recording in mixed telephony environments, specifically where CUCM is serving the back office of a financial trading floor while a turret system is being utilized in its front office.

## Improved Operational Control

NICE offers organizations better control by means of centralized administration, recording and playback. All the operational and administrative activities can be performed over the network.

## High Security

Extensive privilege-based user access mechanisms provide full control of user operations, while an integral audit trail provides detailed information of user activity.

## Unlimited Storage

In addition to off-line storage capabilities, NICE's integration with leading enterprise storage management vendors enables centralized archiving with seamless on-line media access.

# Theory of Operation

## NICE Perform Architecture

The NICE Perform solution is composed of four main elements:

- Interactions Center

  The Interactions Center connects to the CUCM CTIManager using TAPI (or to the Cisco Unified Contact Center Enterprise CTI Gateway in contact center environments) for receiving call events. It implements recording rules, handles recording requests and controls the loggers.

- Loggers

  The VoIP loggers capture and record the voice packets.

- Database

  The Database maintains the call details and the system's administrative information.

- Application Server

  The Application Server provides access layer for the system to the end user applications. The system's elements may be consolidated in a single server or a pair of servers, or distributed among several servers, according to the scale of the solution.

Additional optional elements include screen loggers for recording the screen activity of the users, a Storage Center for managing long-term storage of the recorded data, and audio analytic servers for automated voice analysis.

Contact centers can take advantage of the advanced NICE SmartCenter solution. NICE SmartCenter provides organizations with capabilities to improve performance at the agent, operational and enterprise levels. This solution drives contact center and enterprise performance by leveraging the synergies of the combined capabilities of NICE's offering for interactions capture, quality management, interaction analytics, workforce management, performance management, coaching, and customer feedback; each the leading solution in its category, unified within a Service-Oriented Architecture (SOA) framework, providing powerful functionality with maximum flexibility.

## Phone-Based Active VoIP Recording

One of the new features Cisco Unified Communications Manager (CUCM) version 6 had introduced is an integration capability for providing IP phone-based recording. Cisco IP-phones are capable of forking the received and transmitted voice traffic in two separate Real Time Protocol (RTP) streams. NICE Perform uses SIP trunk in order to connect to the CUCM cluster. Over this SIP trunk the CUCM and the NICE Interactions Center exchange SIP messages which direct the recorded calls from the IP-phones to their destination—the VoIP logger.

## Recording Modes

The NICE-Cisco phone-based active recording integration supports the following recording modes:

- Total Recording

  Total recording is used where all the calls need to be recorded. The recording session automatically establishes when an agent answers or initiates a call.

- Interaction-based recording, including record-on-demand or quality management recording programs.

  Interaction-based recording serves for recording specific calls. NICE Perform invokes the recording session for an active call through the CUCM CTIManager using TAPI. The trigger for recording calls in interaction-based recording may be a human recording request or a recording rule, based on the call's details.

The setting of the recording mode is based on directory numbers (DNs), and mixed recording modes are supported within the same system for different DNs. The recording capability is a CUCM administered feature. The phone's DN is configured as “Automatic recording” for total recording or as “Application-invoked recording” for interaction-based recording in the CUCM administration.

### Recording Transparency and Tones

Even though the IP-phone actively participates in the recording process by sending out the audio streams, the recorded user does not receive any visual or audio indication that recording is taking place.

Note that in certain jurisdictions, a requirement exists to inform the calling or the called party by means of a specific tone that their call is being recorded. The IP-phone is capable of inserting this notification tone, ensuring that the called or the calling party (or both) is notified that recording is taking place.

### Supported Versions and Phone Models

IP-phone based active recording is supported by CUCM 6.0 and above. The recorded IP phones must be able to fork media. The supported models are Cisco third-generation IP phones: 7906G, 7911G, 7931G, 7941G, 7941G-GE, 7942G, 7945G, 7961G, 7962G,7965G, 7961G-GE, 7970G, 7971G-GE, 7975G.

For earlier CUCM versions and for other phone models, NICE offers three recording methods:

- Passive VoIP recording
- Active VoIP recording-based on NICE's VoIP Recording Gateway
- Active VoIP recording-based on NICE's VoIP Recording Agent

The VoIP Recording Gateway is a network element that filters RTP traffic and forks it, sending the forked streams to the recording system. Distributed implementation of the VoIP Recording Gateway enables consolidation of the recording system servers, and is not dependant on CUCM version or phone models.

The VoIP Recording Agent is software that runs on a PC, capable of forking the RTP packets of a Cisco IP Communicator softphone or of a daisy-chained hard IP-phone. The VoIP Recording Agent then sends the forked streams to the VoIP logger, in a similar manner as the phone-based active recording.

NICE Perform software migration paths are available once the CUCM system and phones are upgraded to support phone-based active recording. Where only a portion of the phones are of the models that support phone-based active recording, the rest of the phones can be recorded using any of the other above-mentioned methods. NICE Perform supports mixed recording methods in the same system.