Presence in Unified Communications and The new world of “TelePresence”

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Agenda

- What is Presence
- SIP/SIMPLE
- Cisco Presence Server
- What is Telepresence?
- Design Rules (as we know them today)

What: Presence Awareness

- What is “Presence”?
  - Information about a person’s willingness and availability to communicate
- Examples of presence in action today
  - IM “Buddy List” status indication
  - “Busy” tone on traditional phone
  - Contact Center Agent status
- Publish / Subscribe
  - Clients publish presence information to other users who are called subscribers
- MPOP, Federation and “Presence by Observation”
  - Combining presence information from multiple devices and making this information available for other applications

SIP & SIMPLE

SIP / SIMPLE

- SIP for Instant Messaging and Presence Leveraging Extensions
- SIMPLE is defined in additional RFC documents

  RFC 3428, Session Initiation Protocol (SIP) Extension for Instant Messaging
- RFC 3856, A Presence Event Package for the Session Initiation Protocol (SIP)
- RFC 3857, A Watcher Information Event Template Package for the Session Initiation Protocol (SIP)
- RFC 3858, An Extensible Markup Language (XML) Based Format for Watcher Information

Presence Components

ENTITIES

- PRESENCE SERVER

  accepts, stores, and distributes PRESENCE INFORMATION

- PRESENTITY (presence entity)

  provides PRESENCE INFORMATION to a PRESENCE SERVER

- PRESENCE USER AGENT

  means for a PRINCIPAL to manipulate one or more PRESENTITIES

- WATCHER

  requests PRESENCE INFORMATION about a PRESENTITY from the PRESENCE SERVICE

ACTIONS

- SUBSCRIPTION

  the information kept by the PRESENCE SERVER about a SUBSCRIBER's request to be notified of changes in the PRESENCE INFORMATION of one or more PRESENTITIES

- NOTIFICATION

  a message sent from the PRESENCE SERVICE to a SUBSCRIBER when there is a change in the PRESENCE INFORMATION of some PRESENTITY

- PUBLICATION

  An unsolicited message sent from the USER AGENT whenever a status change occurs

Presence Definitions

- IM/Presence Federation

  Model in which presence data and IM are shared openly between two different presence servers that manage different domains, similar to the email model we use today.

- Persona

  Modeled after a human user that may have any number of devices or applications with presence information concerning the user. A persona also has associated rules/policy that apply to modify/limit access or use of the presence information.

- Reachability

  Overall status of the persona determined by matching the presence state to the defined reachability rules (vacation, out-of-office, busy, interuptible but busy, available, unavailable, Do Not Disturb (DND), unknown)

- Visibility

  A specific view of presence information that is available to a watcher. This is governed by applying the rules.

Presence Definitions

- Authorization

  Process of determining what presence information (if any) a specific watcher is allowed to access about a user.

- Composition

  Process that produces a “raw” presence document based on the set of presence that was collected. Composition is governed by rules defined in the composition policy, which are linked to the authorization. These rules may be complex, and consider aspects such as correlation, conflict resolution, merging and splitting.

Presence Definitions

- Presence Rules/Filtering

    Privacy-based: Process by which information in the “raw” presence document is removed or transformed for the purpose of withholding sensitive information about the presentity. These rules are defined by the presentity, and may be applied to a particular watcher or set of watchers, or based on other types of input.

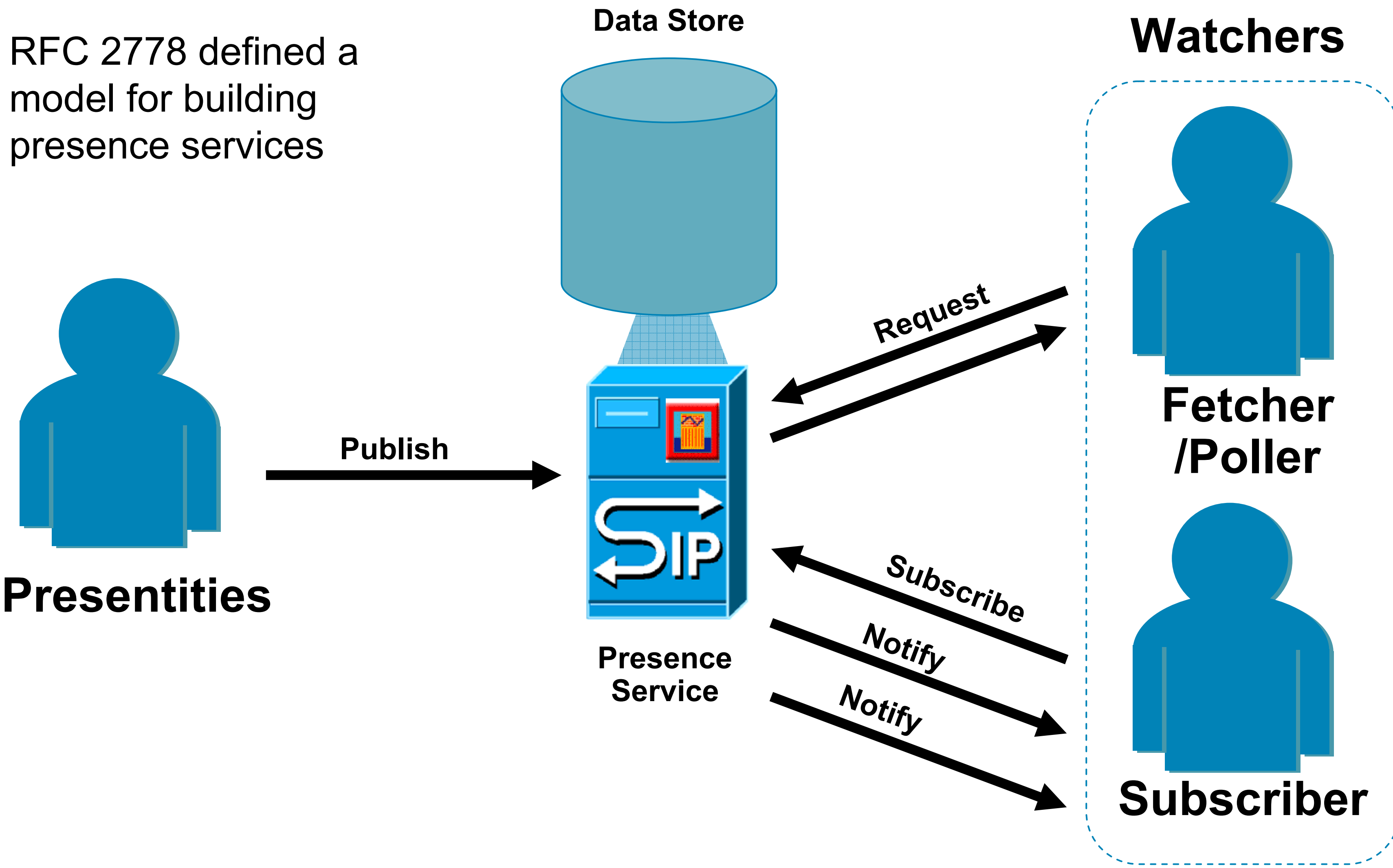
    Watcher-based: Process by which further information is removed from the document based on input from the watcher on what type of information it is interested in.

- Presence Routing Rules

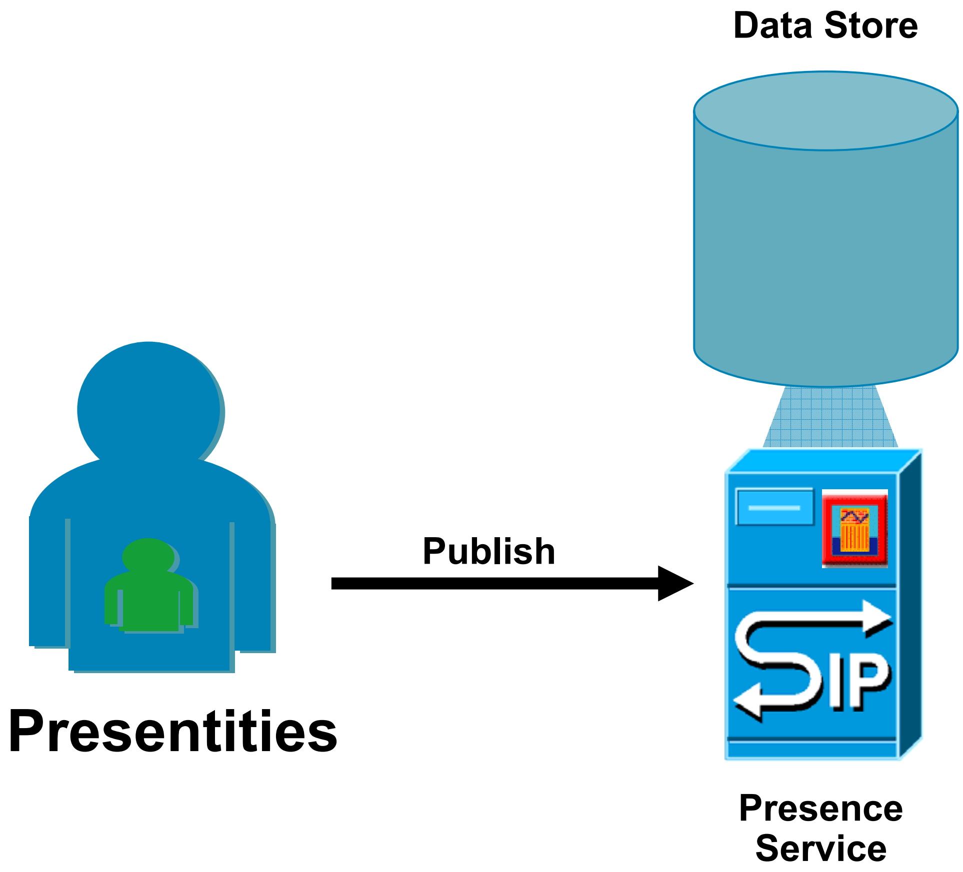
    Rules that affect the routing of multi-modal communication, that are presence enabled.

RFC 2778: A Model for Presence and IM

RFC 2778 defined a model for building presence services



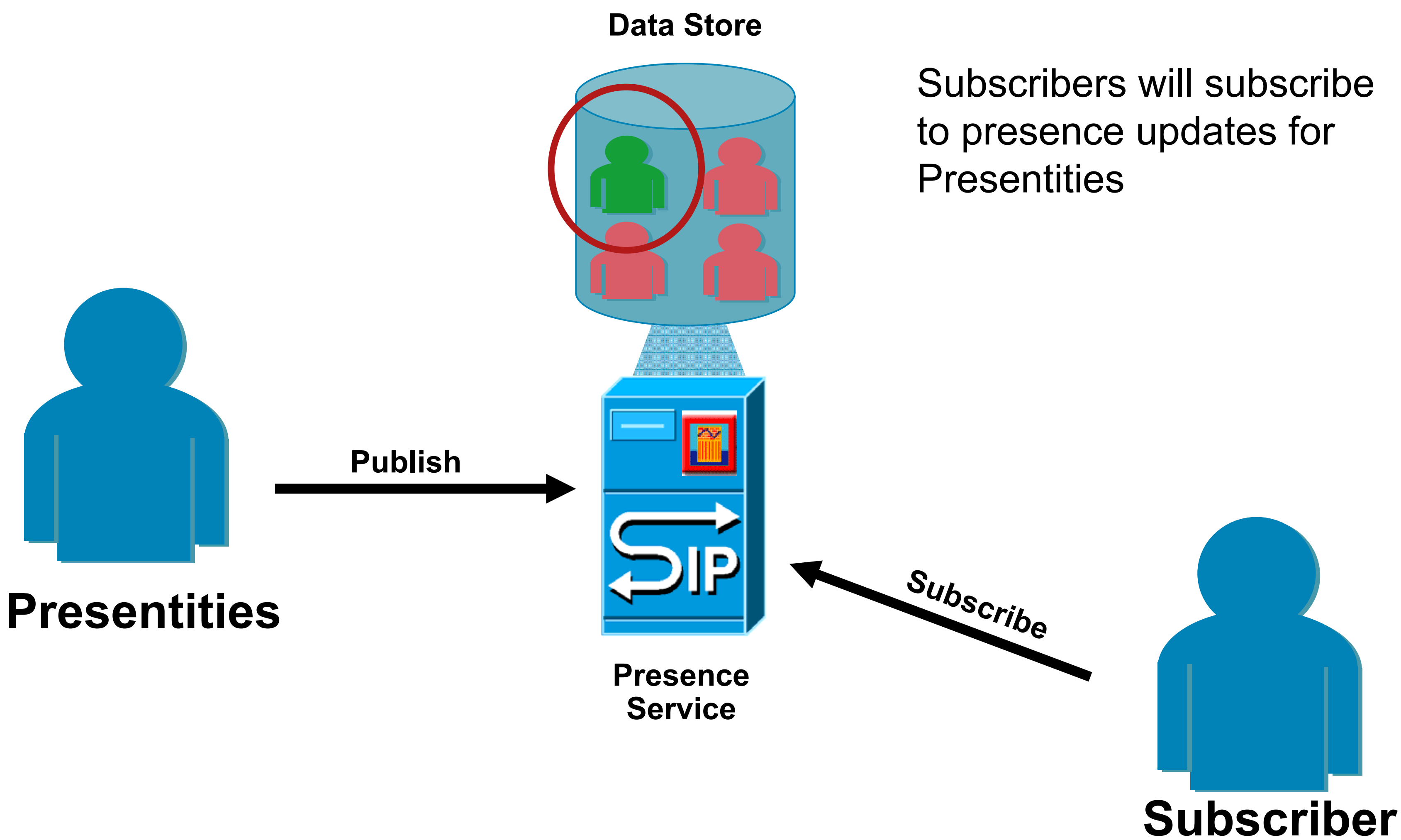
RFC 2778: A Model for Presence and IM



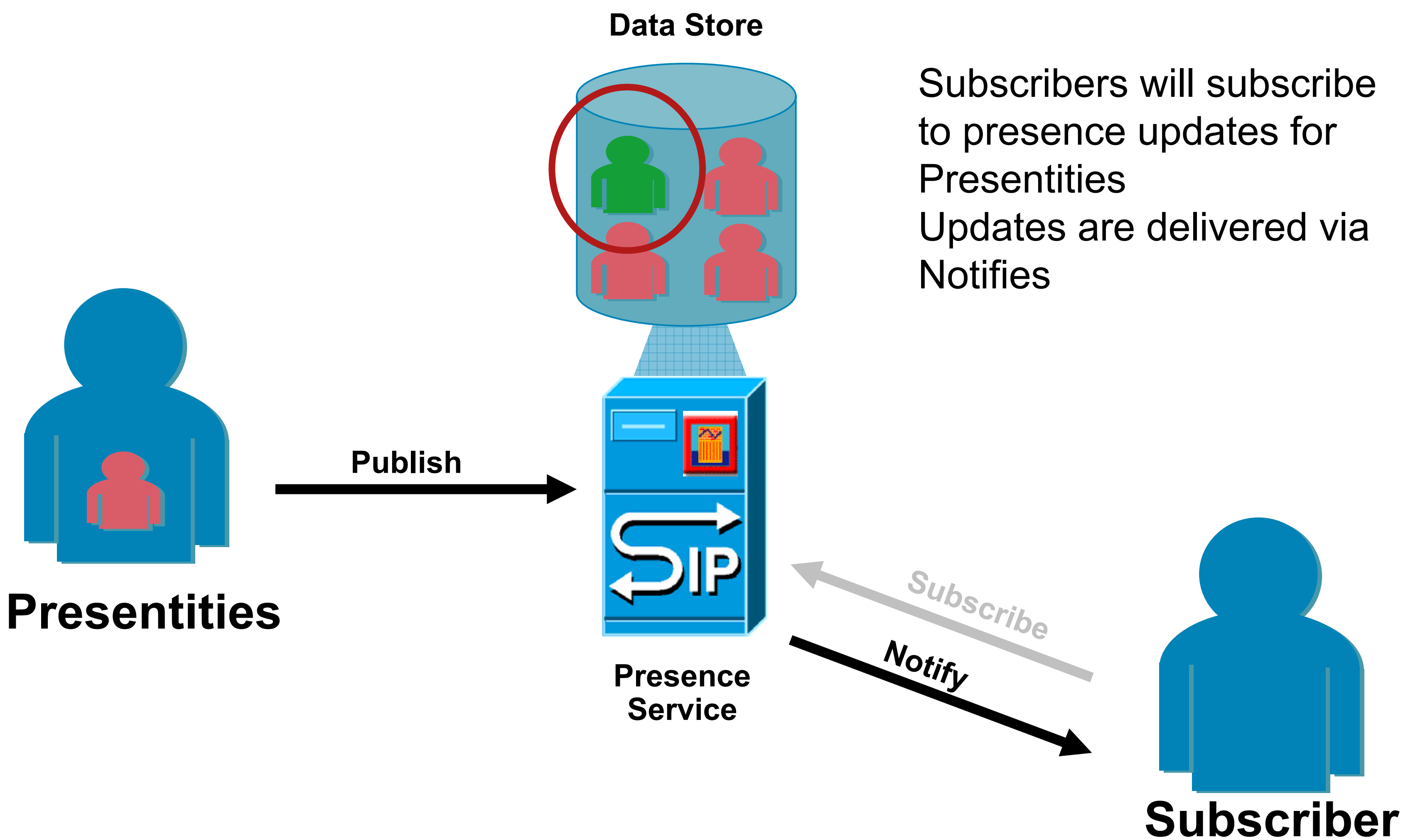
- Presentities publish and changes to their status to the presence service.
- This Information will be held in the presence service data store.

RFC 2778: A Model for Presence and IM

Subscribers will subscribe to presence updates for Presentities



RFC 2778: A Model for Presence and IM

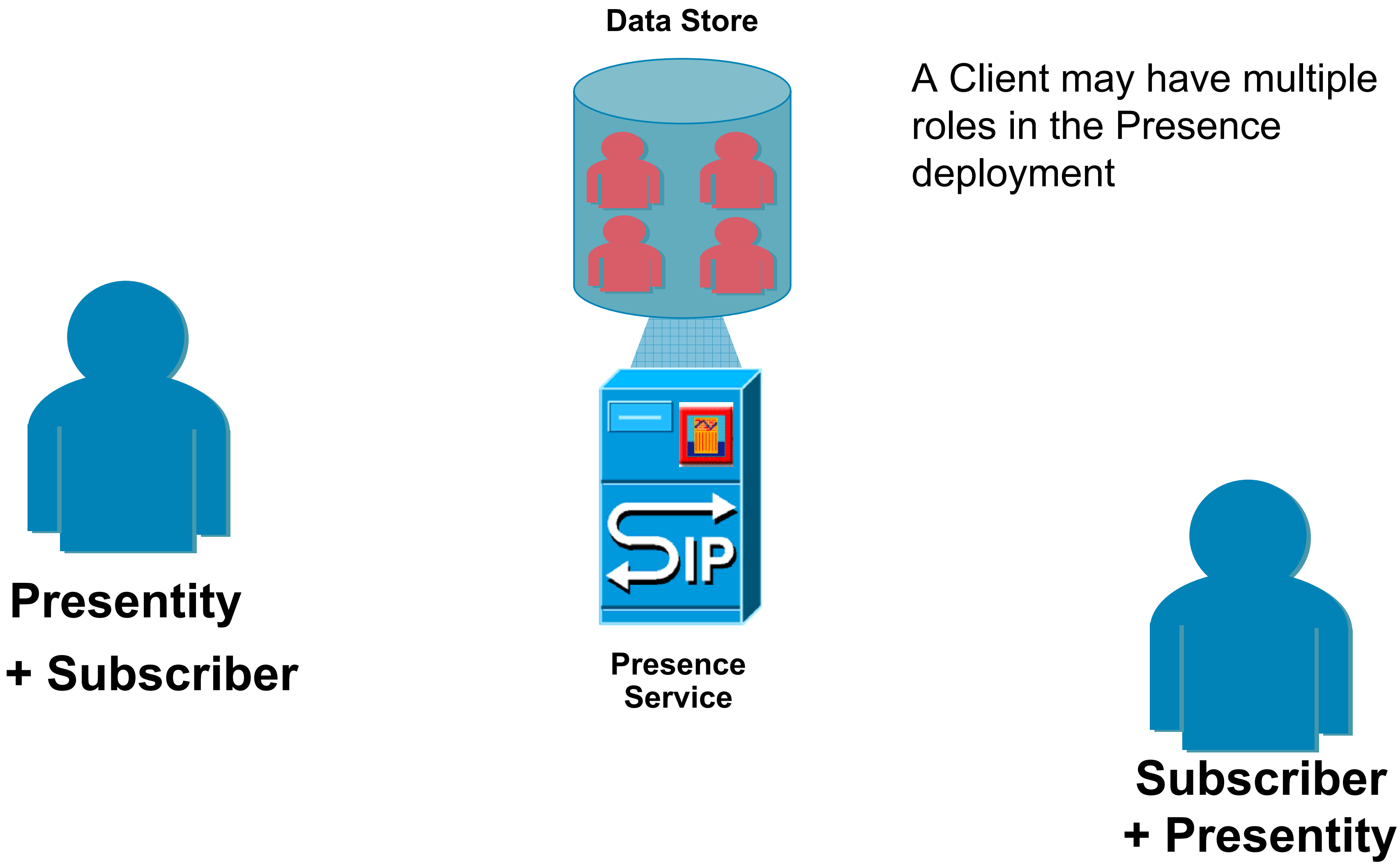


Subscribers will subscribe to presence updates for Presentities

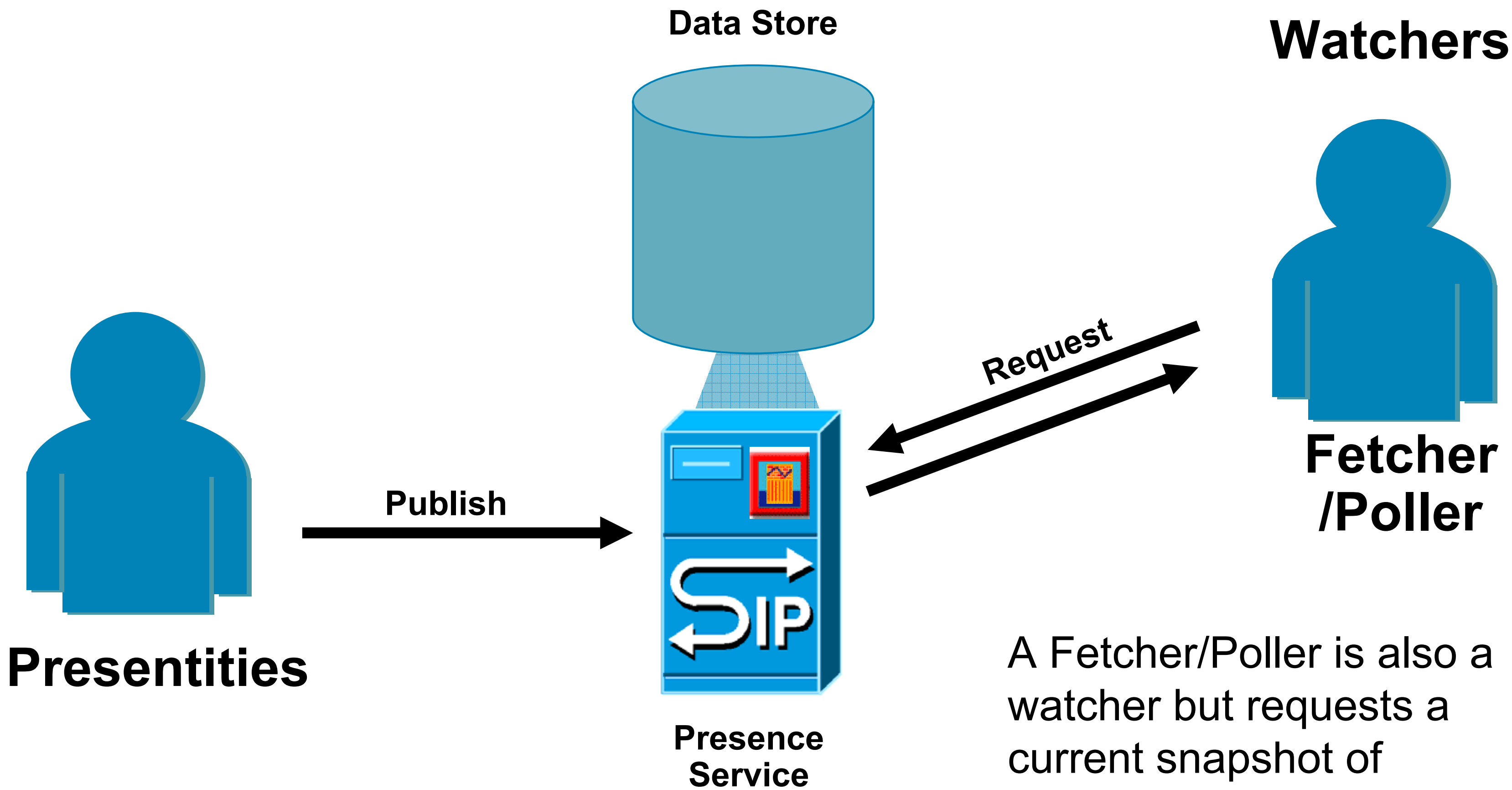
Updates are delivered via Notifies

RFC 2778: A Model for Presence and IM

A Client may have multiple roles in the Presence deployment

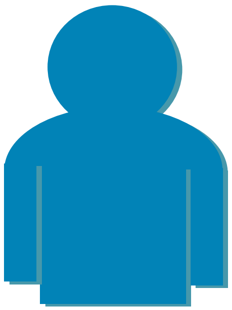


RFC 2778: A Model for Presence and IM

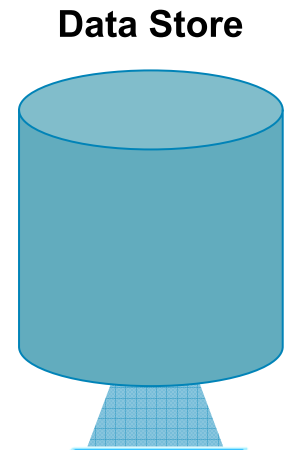


A Fetcher/Poller is also a watcher but requests a current snapshot of presentites status

A Model for Presence and IM

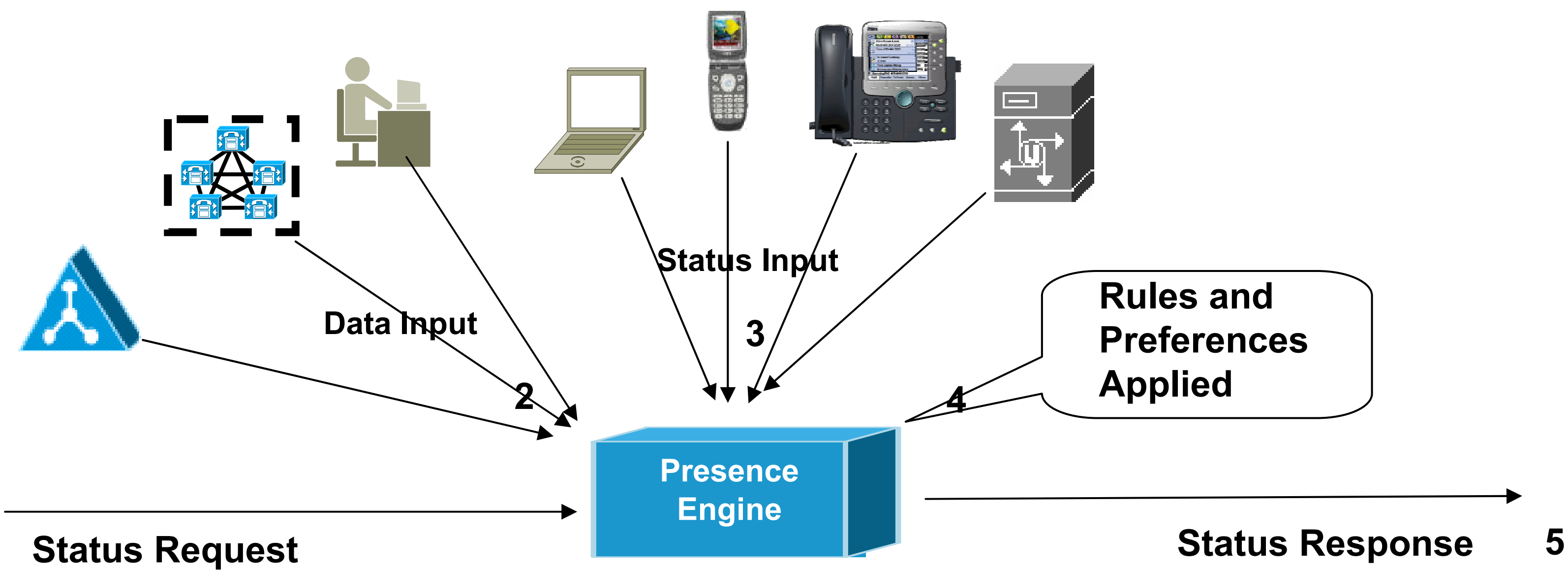


Presentities can be different device types



A Presence service can combining presence information from multiple devices and making this information available for other applications

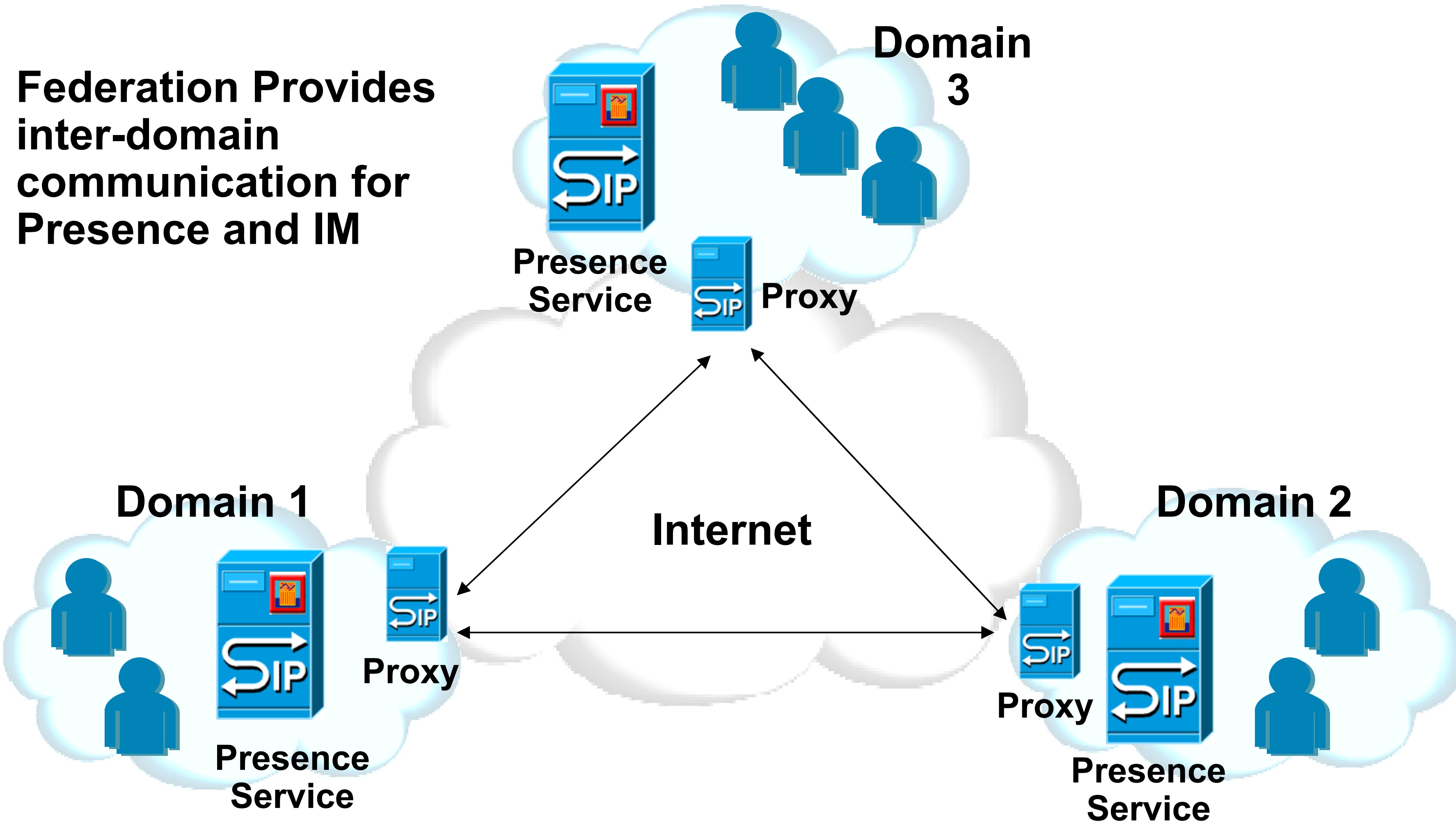
Presence Service Functions



1. Receives Status Request
2. Accesses Databases: Unifies all data associated to a user, including sourced (directories), provisioned (application specific) and dynamic (preferences)
3. Aggregates Status : Collects all status of a user to provide a single point for the definitive status of that user, including all devices/apps of the user
4. Applies Policy: Negotiates Subscriber data with Watcher request to yield appropriate presence
5. Sends Status Response

Federation

Federation Provides inter-domain communication for Presence and IM



Cisco Presence Server

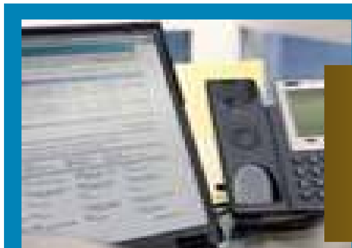
Cisco Unified Presence Server Functionality

1. Provides enhanced User-based Presence capabilities leveraging dynamicsoft Presence technology
2. Supports Rich Presence services for both Cisco enterprise products and customer enterprise desktop applications
3. Provides IP phone Messenger Application
4. Provides the infrastructure for the Cisco Unified Client

Cisco Unified Presence Server:
Cisco Unified Personal Communicator

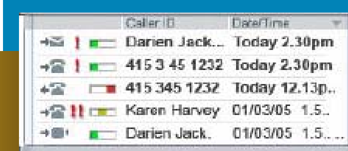
Powerful productivity tools in a single, easy-to-use desktop software application

CallManager 5.0
IP Communications System

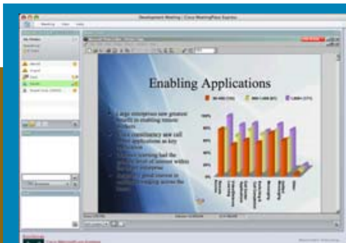


Unity Connection 1.1
Intelligent Voice Messaging

- Adaptive User Interface
- Presence-enabled
- Call, Collaborate, Escalate
- Desktop Video Calling



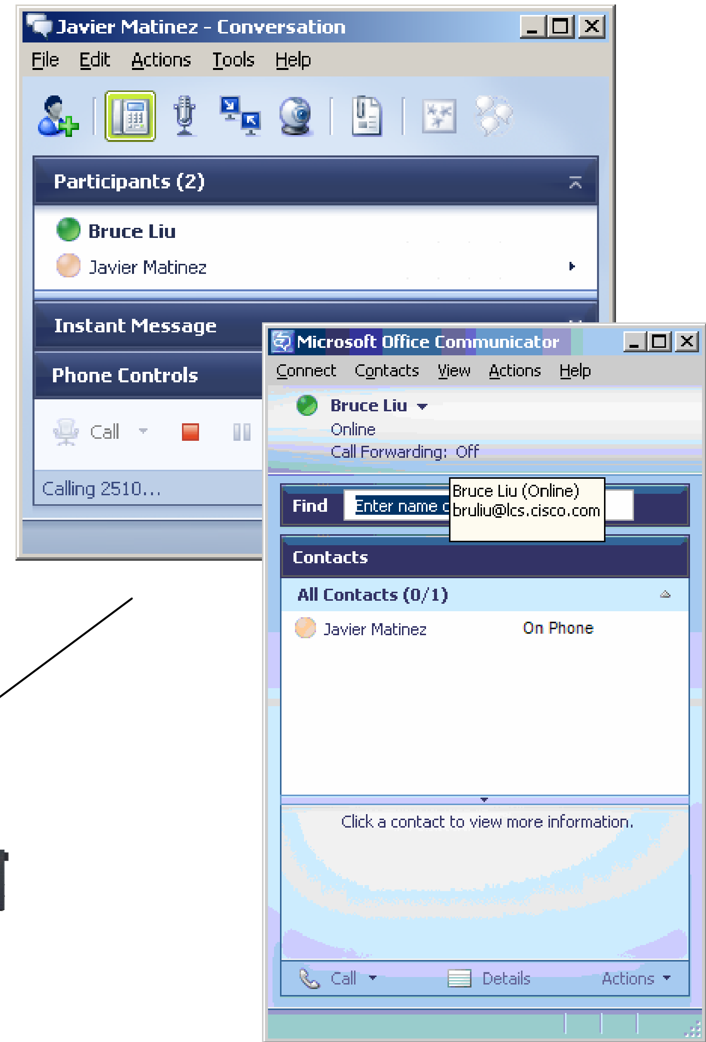
Presence Application Services
Reachability, State, Services



MeetingPlace Express
Web Collaboration

Cisco Unified Presence Server:
LCS 2005 and MOC integration

- Implements CSTA to CTI bridge to integrate with existing LCS 2005 interfaces
- Provides click-to-dial, phone hook status reporting and general phone control from MOC client
- Will migrate over time to a pure SIP solution for better scalability



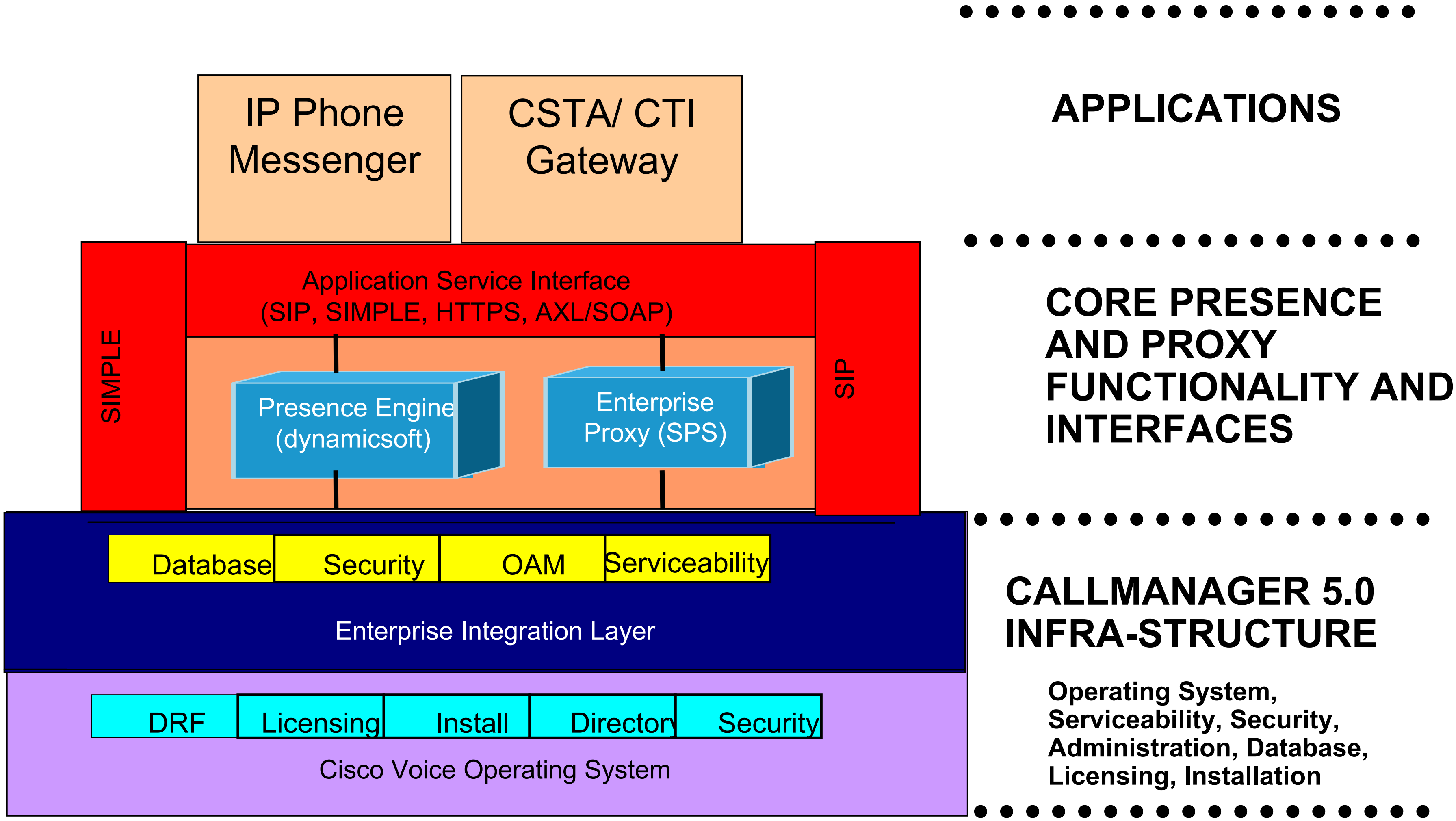
MS Office Communicator user with Cisco IP Phone

Cisco Unified Presence Server:
Lotus Sametime 7.5 Integration

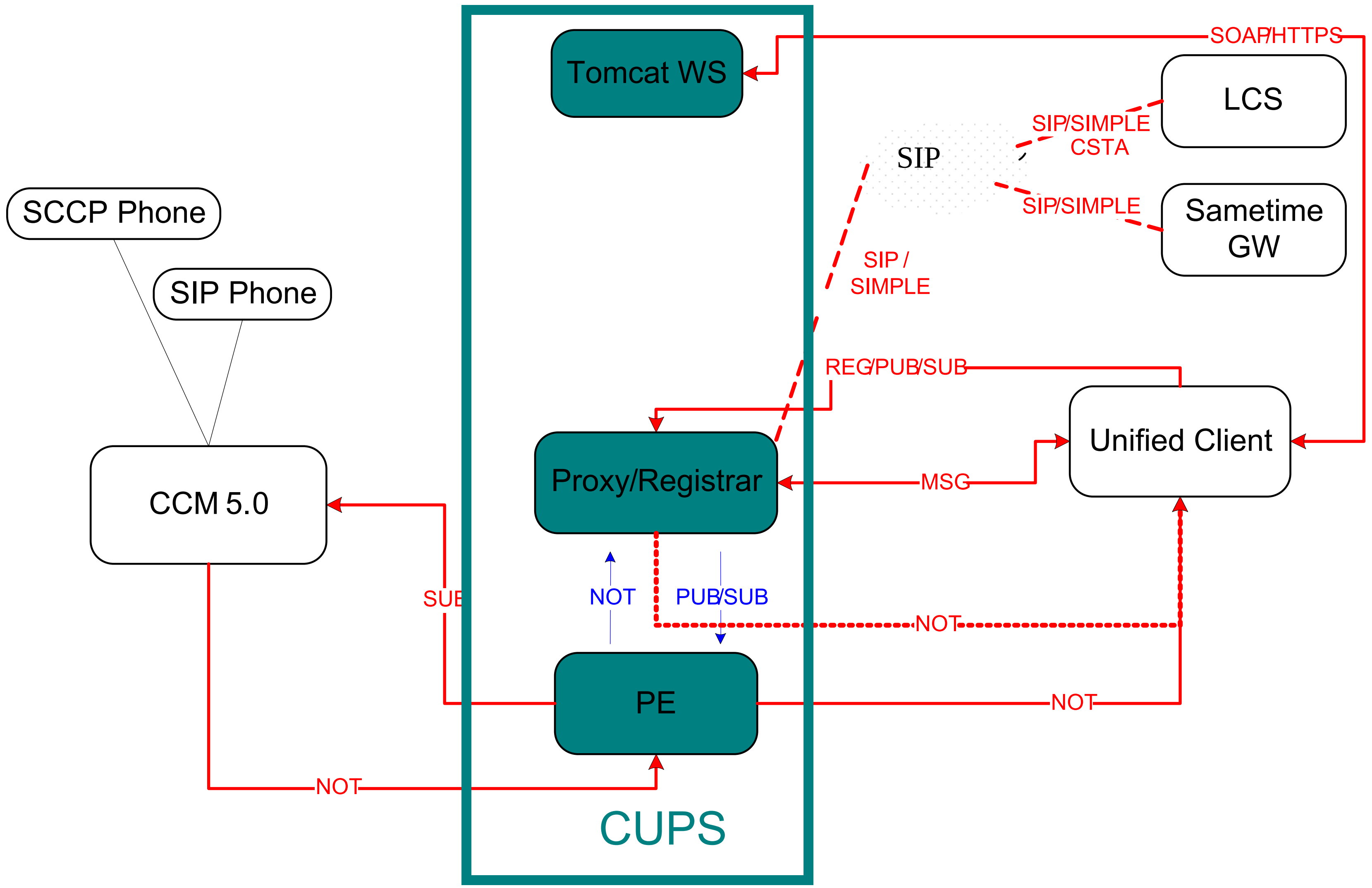
- Connection to Sametime 7.5 via Sametime 7.5 Real-Time Collaboration Gateway
- Provides click-to-dial, phone hook status reporting from Sametime client
- Uses SIP/SIMPLE connection today.

Lotus Sametime 7.5 user with Cisco IP Phone

Cisco Unified Presence Server:
Internal Architecture



Overview – System Message Diagram



CUPS by the numbers

- Today we can have a primary and secondary server (not clustered).
- Scale to support 2,500 hardphone controlling clients (on each server). Is improving in subsequent releases. The limitation is the CTI link from CUPS to the Callmanager.
- Have traffic profiles for registration and different call flows available if required.

Cisco Unified Presence Server

SUBSCRIBE Calling Search Space

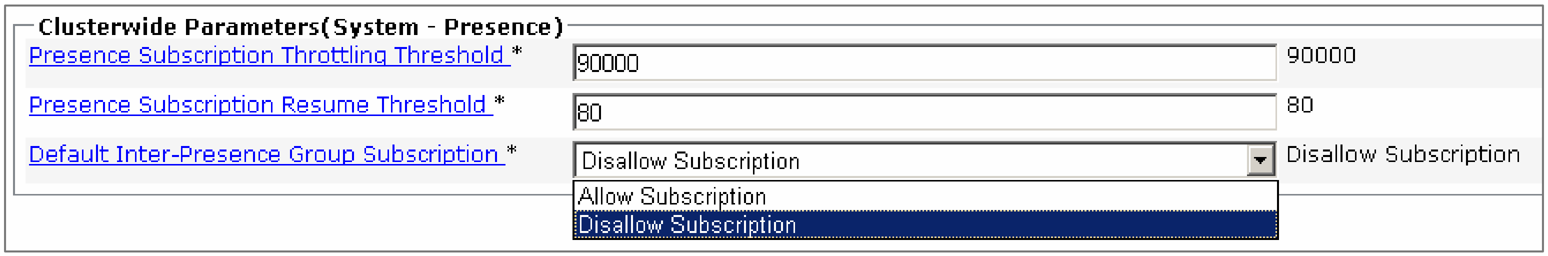
- SUBSCRIBE messages with presence event package (Event: presence) are “routed” just like regular calls
- SUBSCRIBE CSS is associated with the watcher and lists the partitions the watcher is allowed to “see”
- Allows the presence feature to work transparently through translation patterns and normal CSS functionality
- Devices and users can be assigned a SUBSCRIBE Calling Search Space

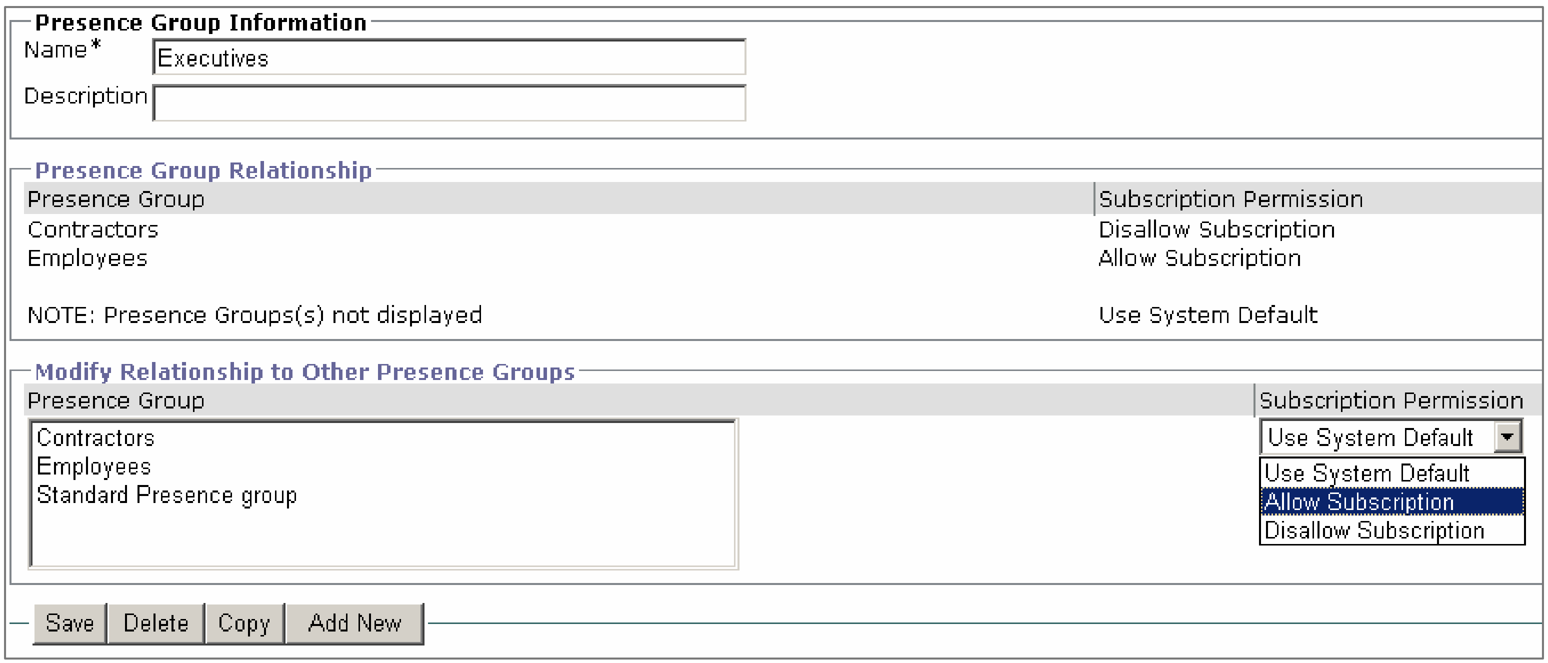
Presence Groups

- Controls the destinations that watchers can monitor
- Devices, directory numbers, and users can be assigned to a Presence Group
  - By default all users get assigned Standard Presence Group
- Inter-Presence Group Subscribe Policy

Presence Policy Configuration

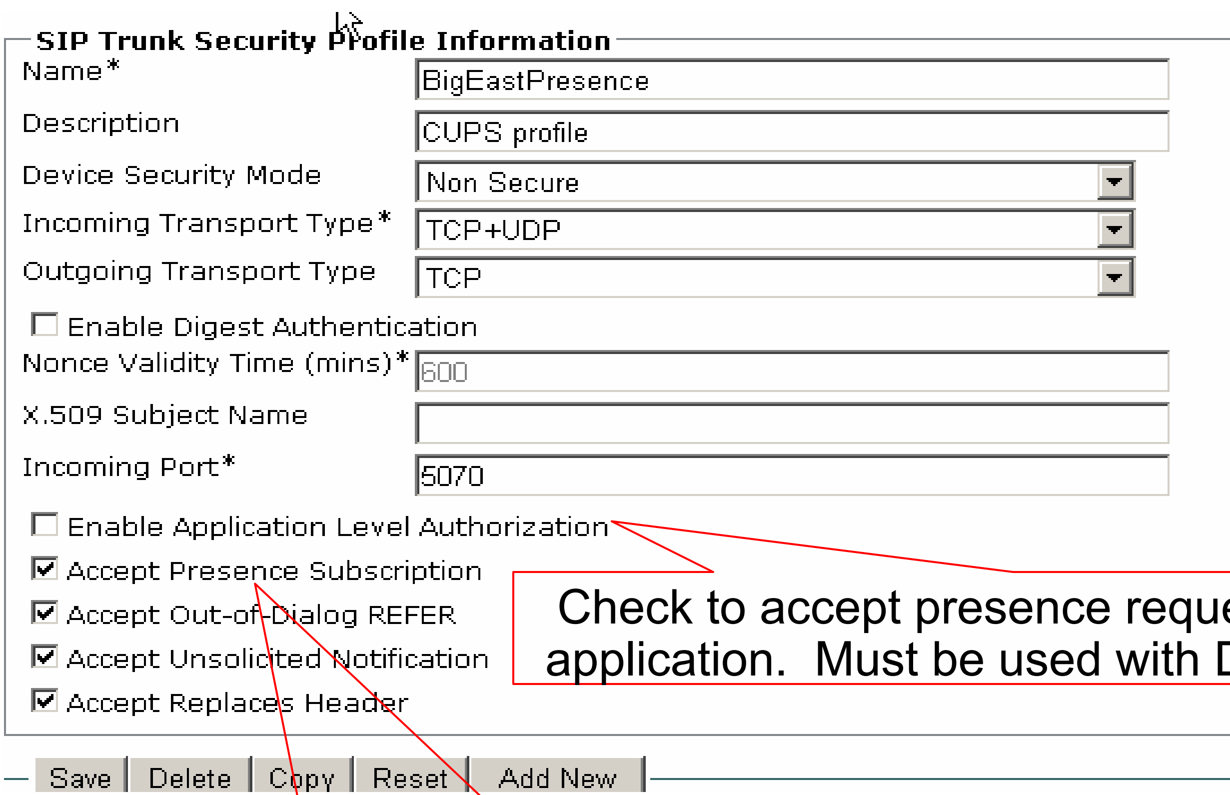
- Inter-Presence Group subscribe policies:





Presence Policy Configuration (continued)

- SIP Trunk Security Profile



Check to accept presence requests from an external application. Must be used with Digest Authentication.

Check to enable incoming presence requests

Cisco Unified Presence Server:
IP Phone Messenger

- Unified Client users see other user's IP phone's on/off hook states
- Users can send or reply to messages from their IP Phones using predefined templates or
composing text messages
- Users can call back IM senders by hitting 1 button.
- Implements presence enabled contact list on the phone
- Will also integrate with other IM clients and presence sources beyond CUPS 1.0

Summary of Presence Server

- IP Phone Messenger

  Integrated IM capability within Cisco IP Phones

- Proxy Functionality

  Based on Cisco SIP Proxy Server providing proxy of presence functions only

- Presence Engine Functionality

  Data store and Presence Aggregator providing enhanced user based presence capabilities

- SIMPLE Network Interface

  IETF Standard interface to pass/receive Presence information

- Cisco Unified Personal Communicator Support

  Provides configuration profiles for LDAP, Proxy, MeetingPlace, Unity, and CTI Gateway using SOAP interface

- Click To Dial / Phone Monitoring interoperability with Microsoft LCS 2005 / Office Communicator

  CSTA to CTI gateway to support functionality of MOC

Cisco Unified Presence Server 1.0:
Key SIP Related RFCs and Drafts

- RFC 3261 – core SIP rfc
- RFC 3263 – SIP Servers
- RFC 3265 – Subscribe / Notify
- RFC 3325 – Asserted Identity
- RFC 2782 – DNS SRV
- RFC 2778/2779 – SIMPLE
- RFC 3856 – Presence Event Package
- RFC 3857 – Watcher Info
- RFC 3858 – Watcher Info Format
- RFC 3863 – PDIF
- RFC 3903 – PUBLISH Method
- RFC 3428 – MESSAGE Method
- RFC 3680 –Registration Event Package

draft-ietf-simple-event-list, draft-ietf-simple-rpid, draft-ietf-simple-prescaps, draft-levy-sip-diversion, draft-dcsgroup-sip-privacy

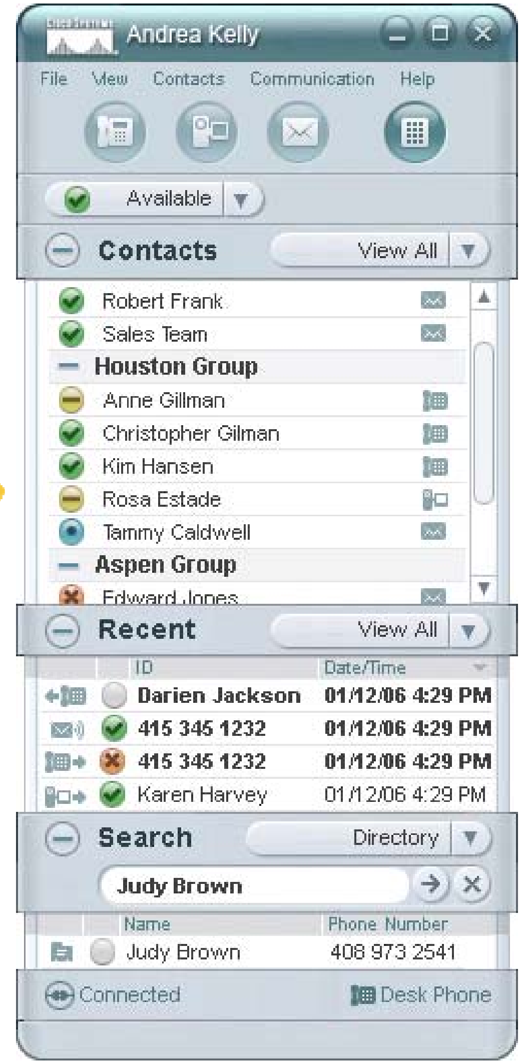
Cisco Unified Personal Communicator

All-in-One Communication Tool

- Voice, Video, and Email
- Presence
- Preferred Device
- Voice Mail Playback
- Contact lists

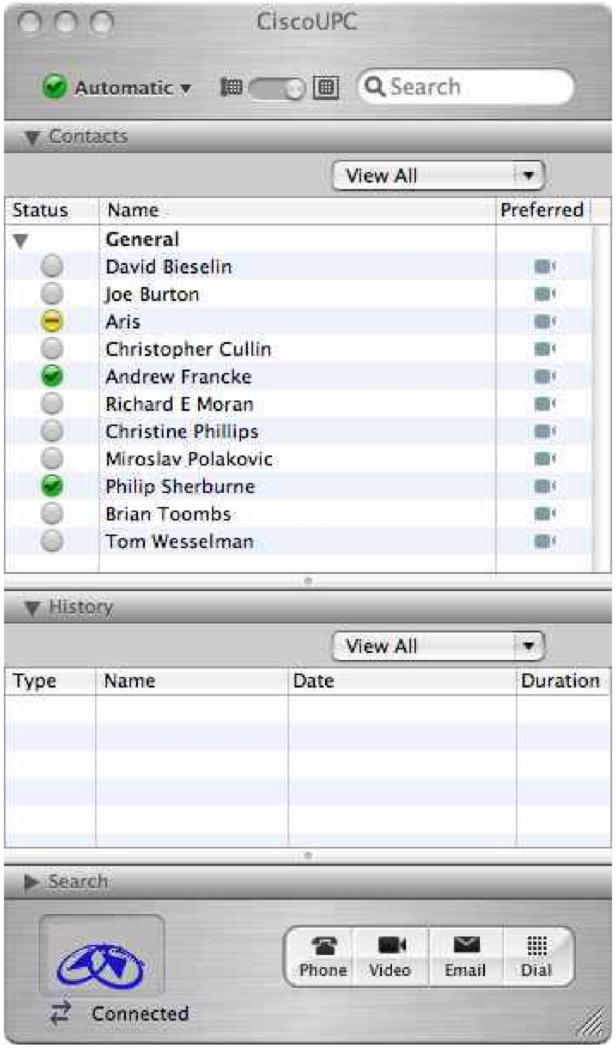
Unified Personal Communicator

- Document Sharing
- Conferencing
- Directory Searches
- Communication Logs
- Mobility

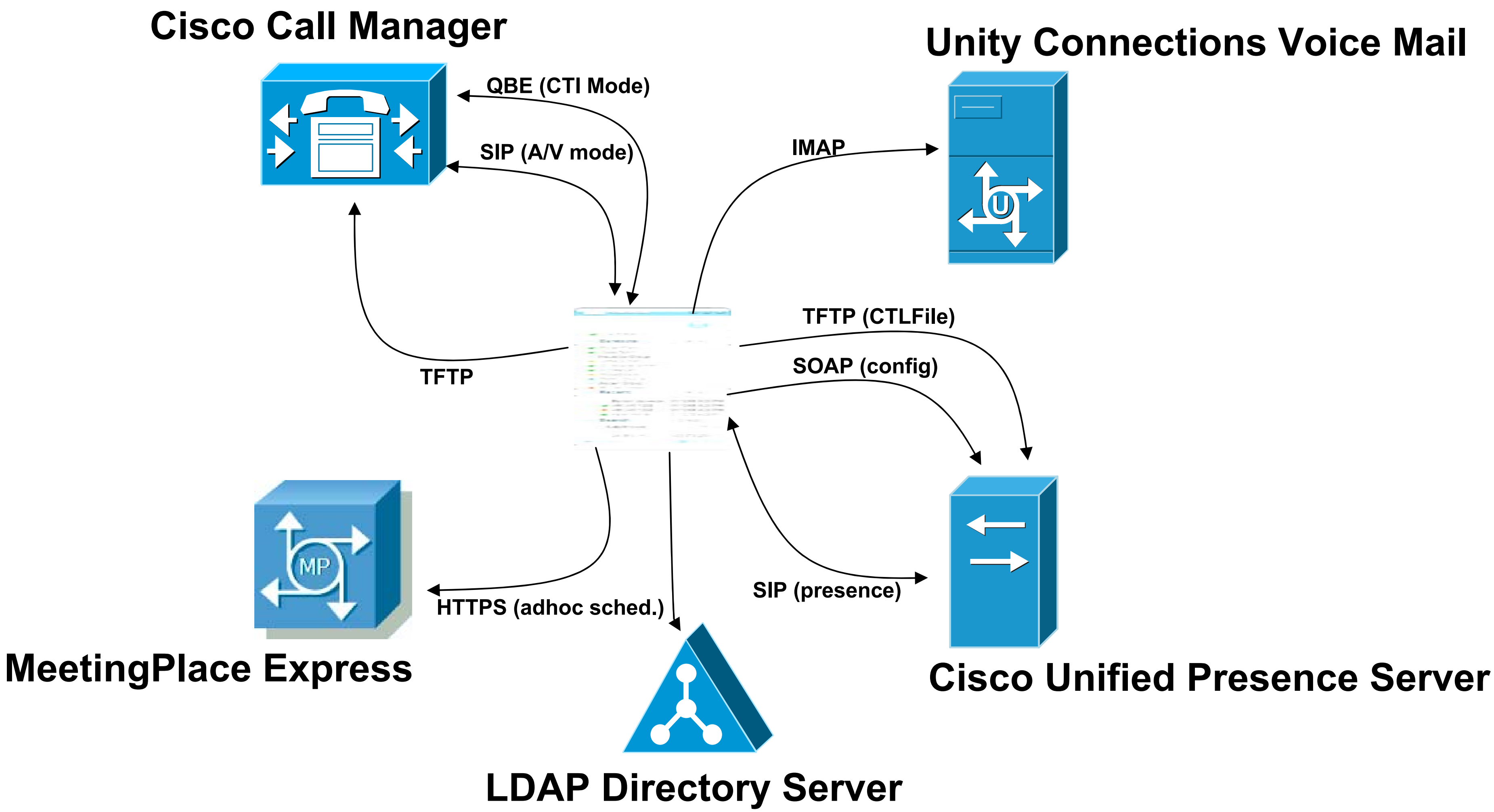


Cisco Unified Personal Communicator General User Interface (Mac OSX platform)

- The interface has the same components, but uses the standard OSX elements



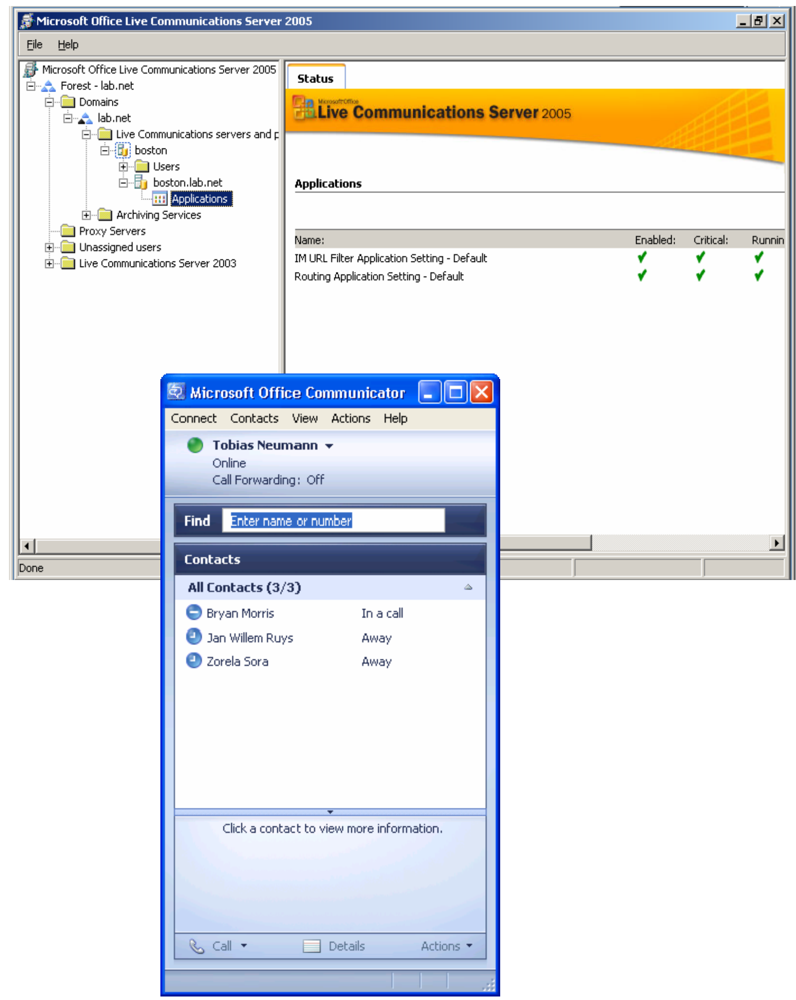
Cisco Unified Personal Communicator
High Level Network Overview



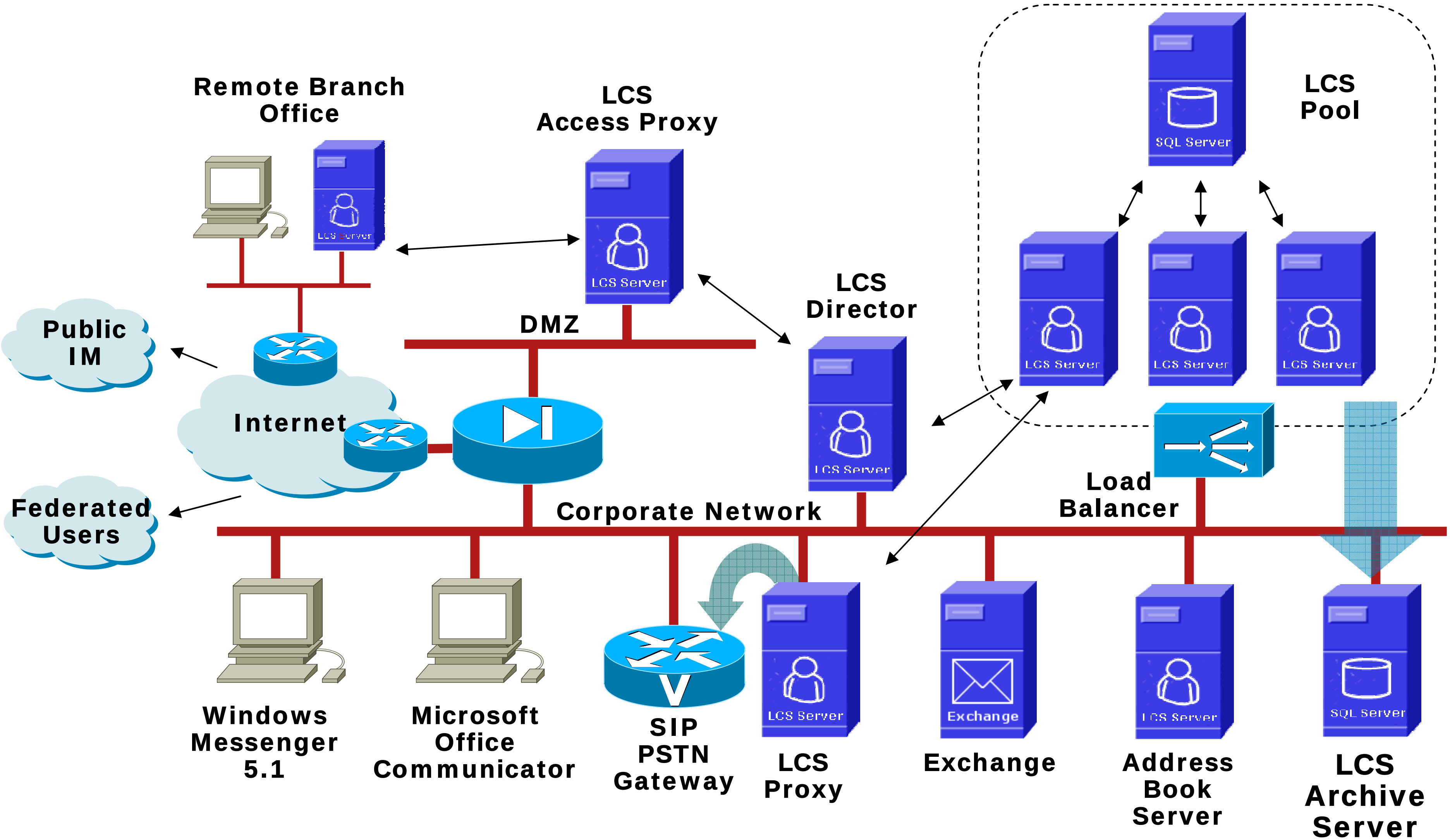
Microsoft Office Communicator

How do Microsoft describe Microsoft Live Communication Server

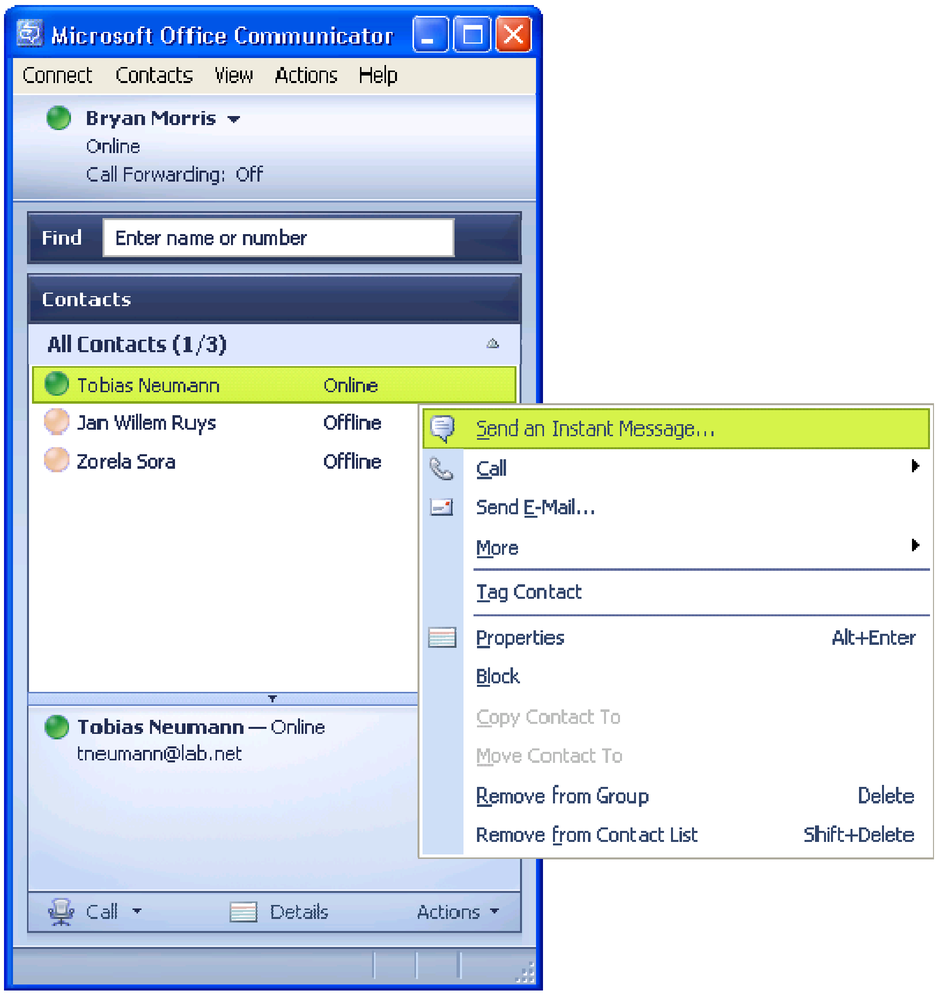
- Microsoft® Office Live Communications Server 2005 provides a stable, extensible, enterprise-ready IM (instant messaging) and presence awareness platform based on the SIP (Session Initiation Protocol) and SIMPLE (SIP IM and Presence Leveraging Extensions) standards. Live Communications Server 2005 also supports audio and video exchange, application sharing, and data collaboration on a peer-to-peer basis.
- LCS runs on Windows 2003 Server
- LCS may require MS SQL Server
- LCS does require Active Directory



Example Deployment

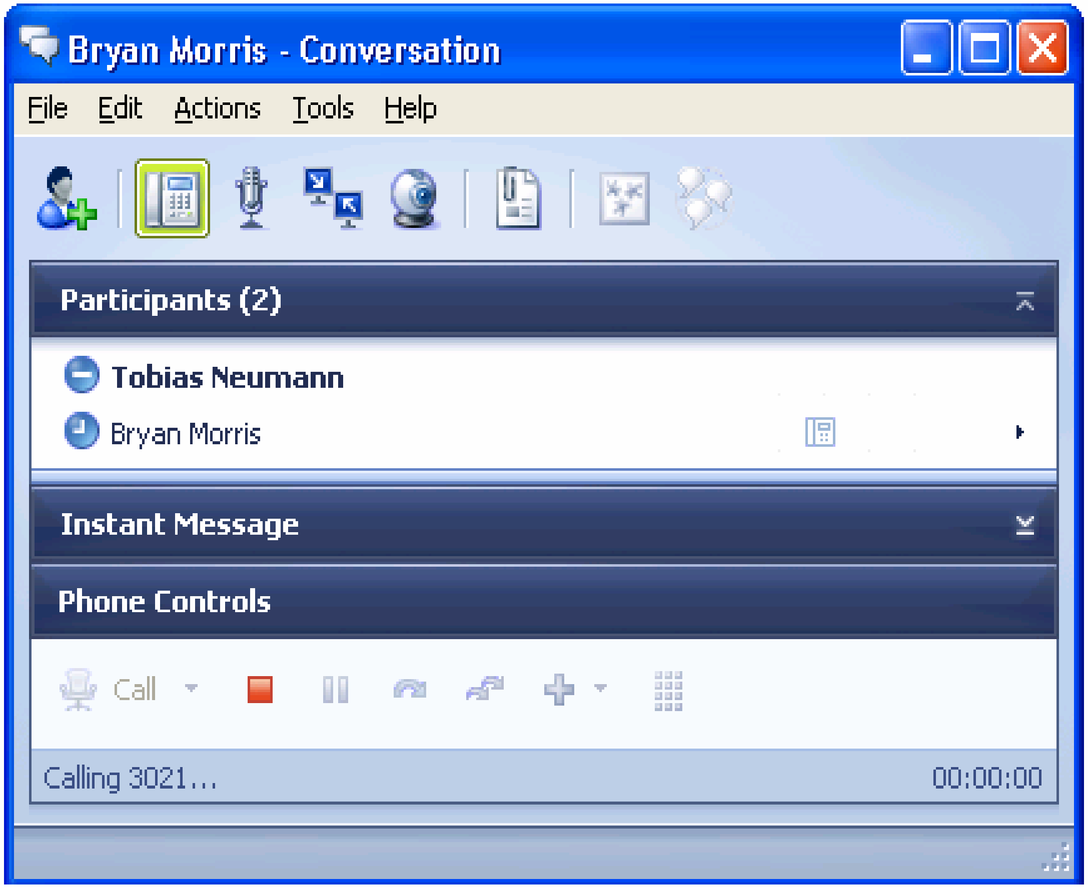


Microsoft Office Communicator



- A user can use Office Communicator to start an audio / video conversation with another user using the Microphone and Speaker on their workstation
- Alternatively MOC can be used to remotely control a physical telephone using SIP/CSTA

MOC Telephone Integration

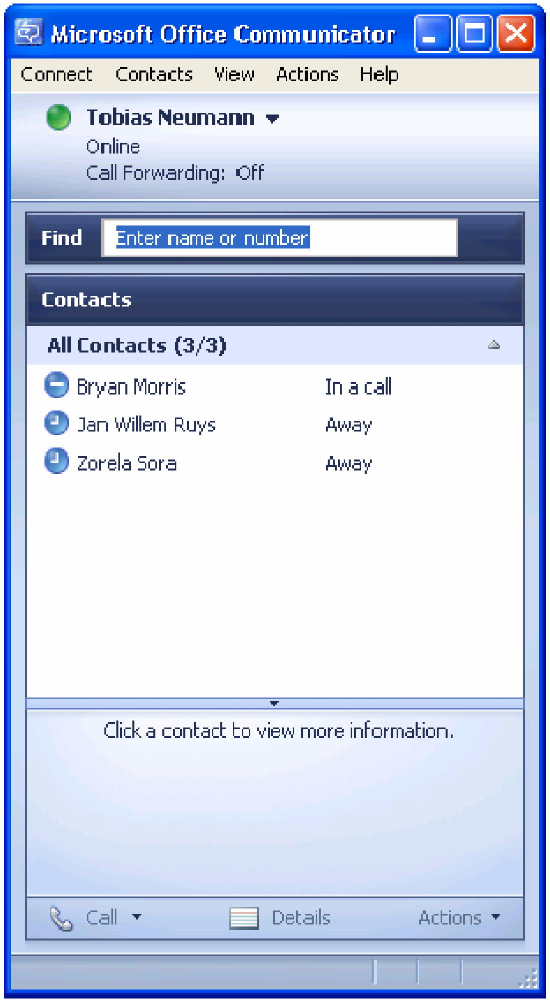


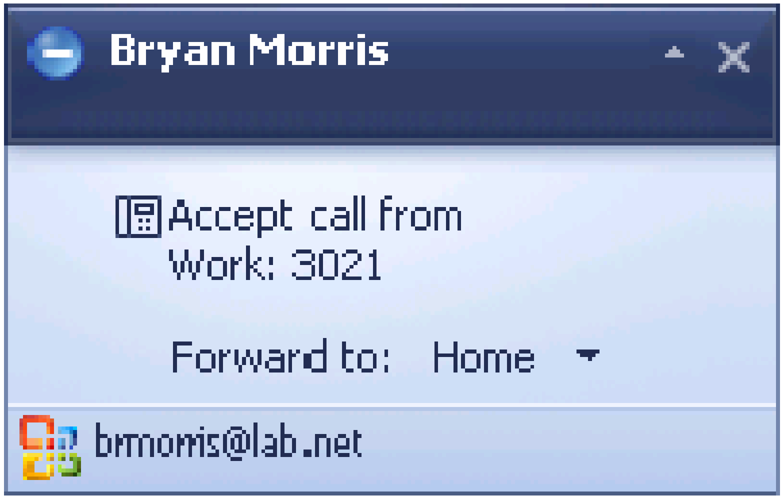
- Make Call/Click to Call
- Answer Call
- Clear/Hang-up Call
- Deflect Call
- Hold Call
- Single Step Transfer
- Retrieve call
- Generate DigitDTMF
- Reconnect Call
- Set Forwarding
- Set Do Not Disturb
- Get Forwarding
- Get Do Not Disturb

MOC VOIP Feature

MOC TDM/IP PBX

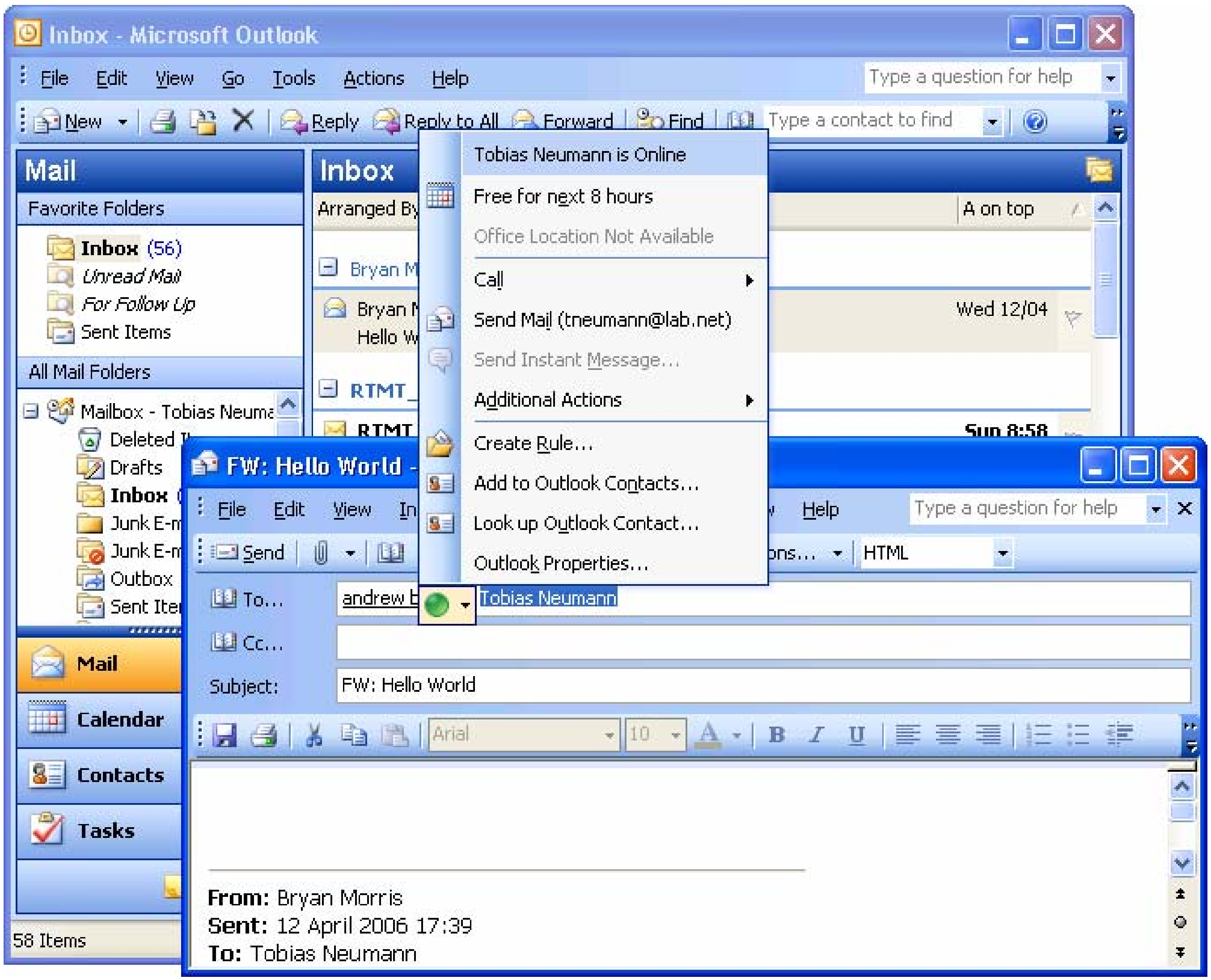
MOC Incoming Call Management, Forwarding and Mobility



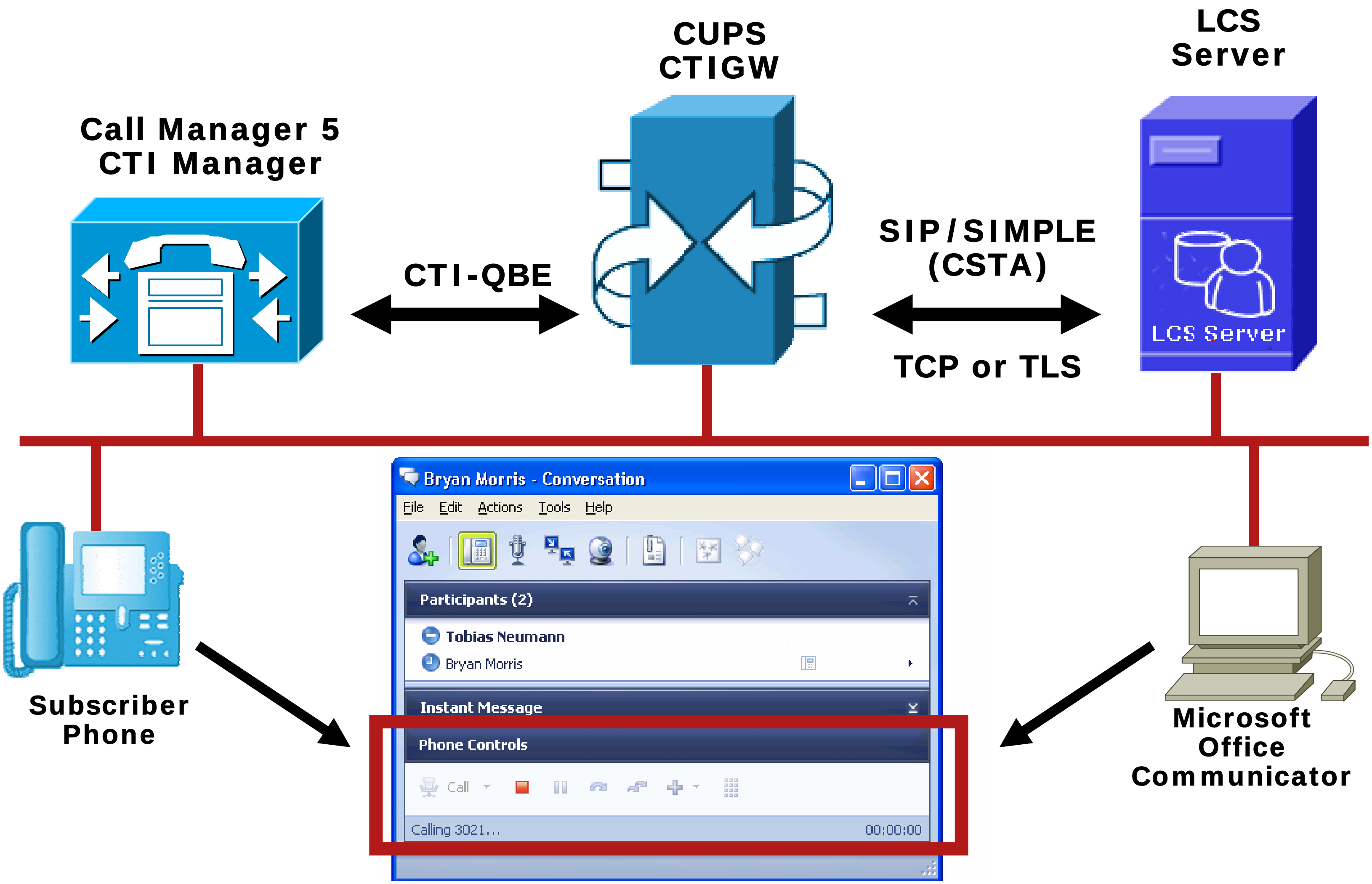


- When Phone integration is configured MOC will alert a subscriber of incoming calls.

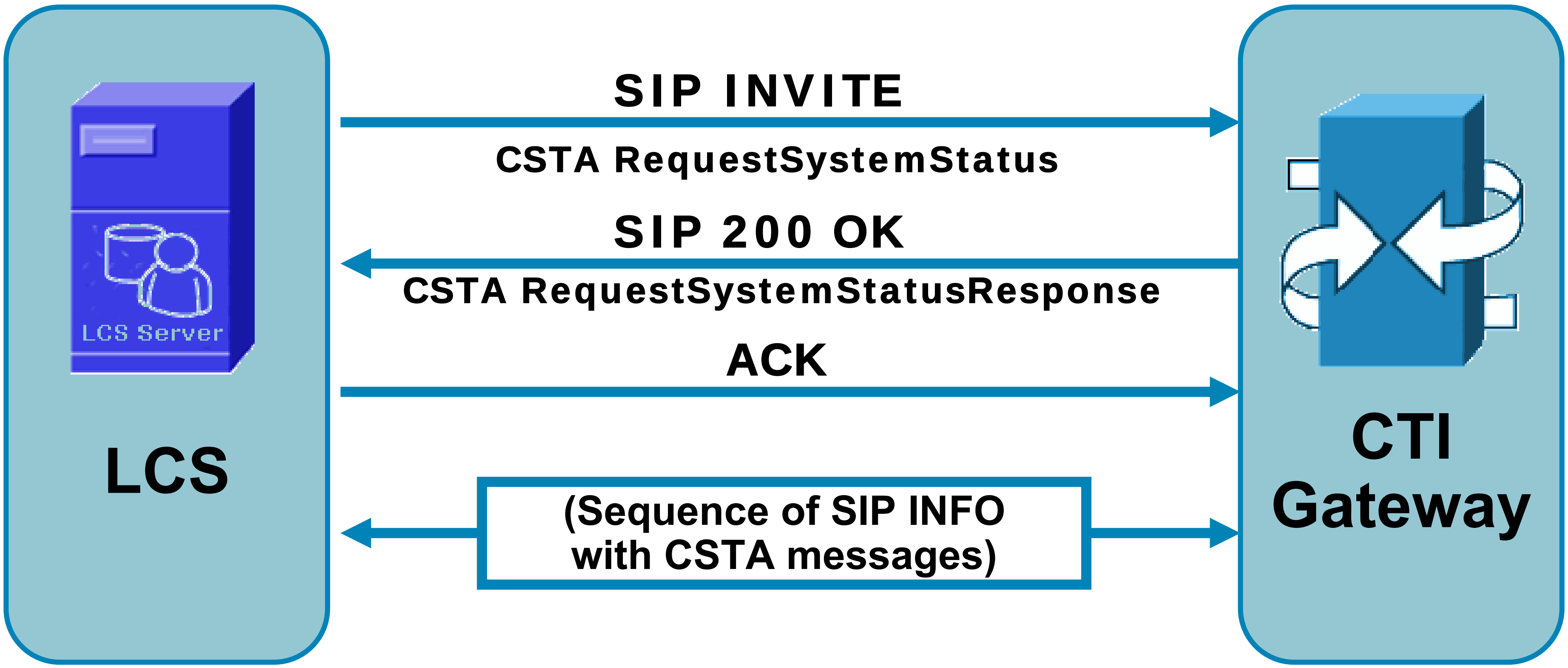
Application integration



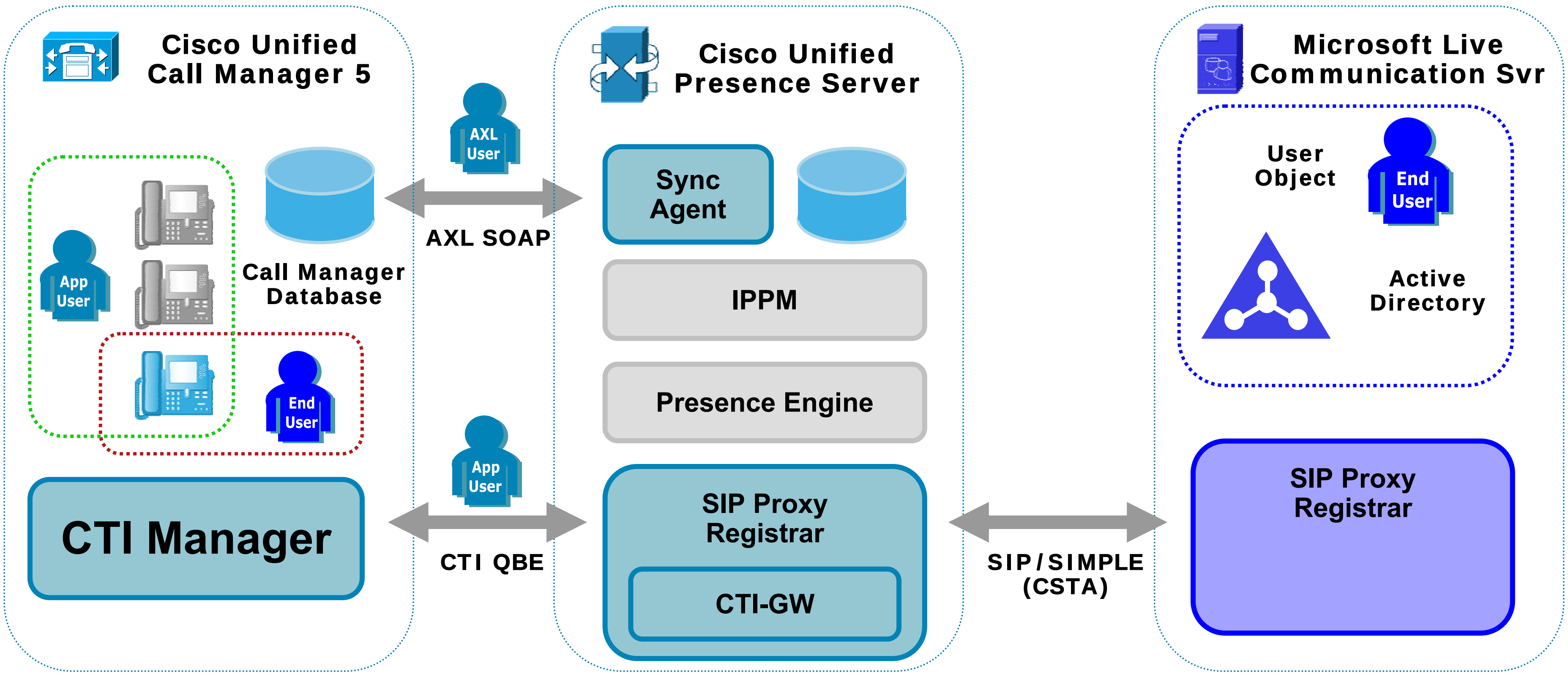
MOC/LOC Telephone Integration



CSTA Communication



CUPS – CTIGW



- App User is a member of
  Standard CTI Enabled User Group
  Standard allow control of all Devices
- End User must have a device association

- The CTIGW is a module of the SIP Server.
- CUPS syncs it's database to Call Manager

- End User must exist in both Active directory and Call Manager for Phone Control.

IBM Lotus Sametime 7.5

How do IBM Lotus describe Sametime 7.5

- IBM Lotus Sametime provides instant, anytime access to people and information through three on demand concepts: presence awareness, business instant messaging and Web conferencing. Millions of people worldwide use Lotus Sametime capabilities every day to gain instant access to people and information, bring together geographically dispersed teams and improve individual and team productivity.
- Lotus Sametime now uses audio integration from leading teleconferencing and telecommunications providers to offer a single interface to both audio and Web conferencing, as well as click-to-call functionality directly from the Lotus Sametime Connect Client.

QuickTime™ and a
TIFF (LZW) decompressor
are needed to see this picture.

QuickTime™ and a
TIFF (Uncompressed) decompressor
are needed to see this picture.

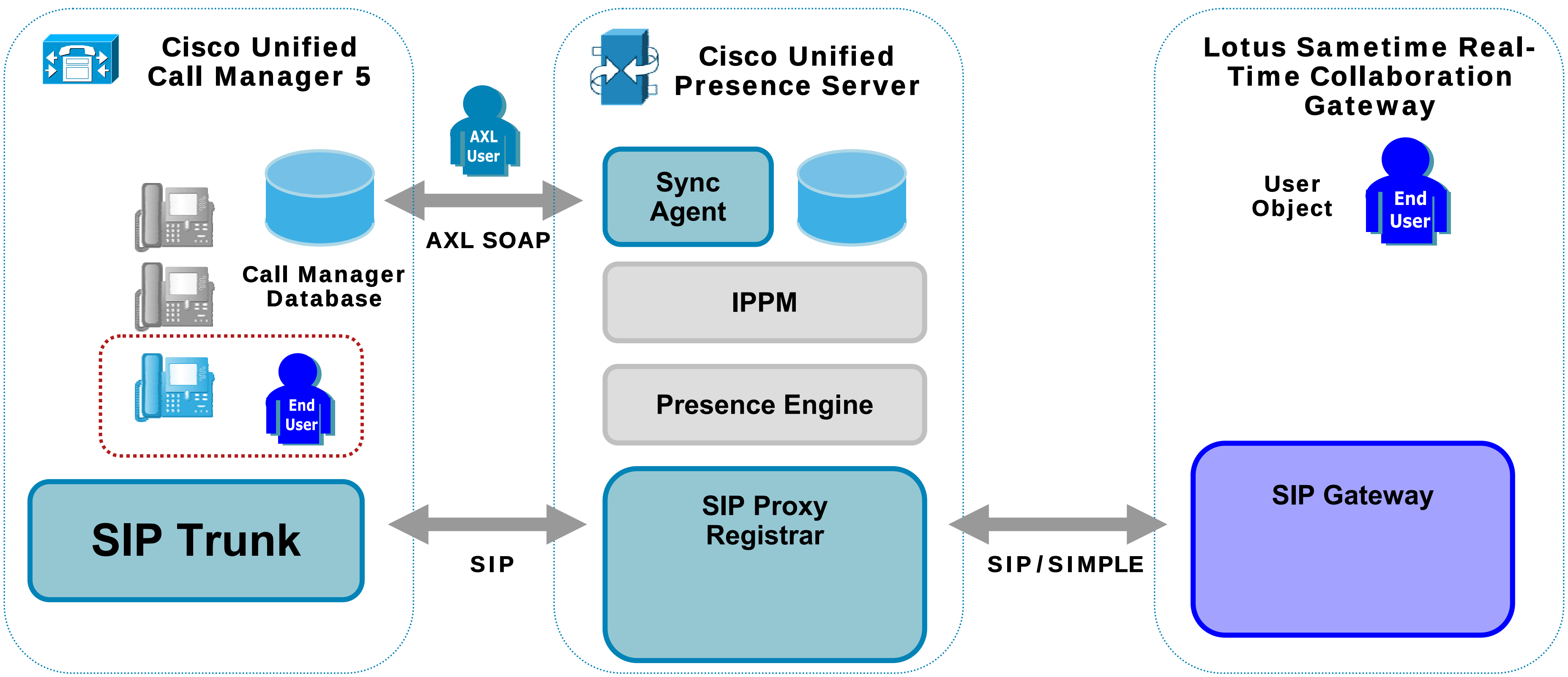
IBM Lotus Sametime 7.5 Features

QuickTime™ and a
TIFF (LZW) decompressor
are needed to see this picture.

QuickTime™ and a
TIFF (Uncompressed) decompressor
are needed to see this picture.

- The provide multiple methods of communication including:
  - Instant Messaging
  - Voice (Softphone)
  - Video (using PC Camera)
  - Web Conferencing
  - Application Sharing
  - White Boarding

CUPS – Sametime



•SIP/SIMPLE Connection

• CUPS syncs it's database to Call Manager

TelePresence

Cisco TelePresence

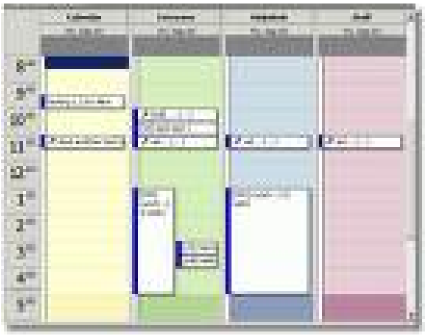
What It Is Today—The Cisco TelePresence Meeting

What It Is

What It Isn't

Cisco TelePresence – Complete Solution

Cisco TelePresence
Endpoints



Cisco TelePresence
Calendar Scheduling

Services

Cisco TelePresence
High Definition
Video Switch

Cisco TelePresence Endpoints
Network NAT/FW

Designing Cisco TelePresence

- Virtual Table

  12 participants

- Eye Contact

  Cameras built for eye contact

- Multiple Cameras

  Captures the entire room

- CD Quality Spatial Audio

  Designed for TelePresence – Left, Center, Right

- Ultra High Definition Video 1080p

  SIP & H.264 Highest quality & Lowest latency in the industry

Cisco TelePresence CTS 3000

- 3 - Cisco Codecs

  720p & 1080p

- 3 - Cisco Cameras

  1080p Unique form factor

- 3 - Cisco Microphones

  GSM Cell Phone RF filter

- Cisco Unified CallManager Integration

  Configured like IP phone

- Purpose Built Table

  Designed for TelePresence – Ethernet & Power

- 3 - Cisco 65" Displays & Graphics Projector

  Enhanced for TelePresence, Latency reduction

Cisco TelePresence CTS 1000

- 1 - Cisco Codec

  720p & 1080p
- 1- Cisco Camera

  1080p Unique form factor
- 1 - Cisco Microphone

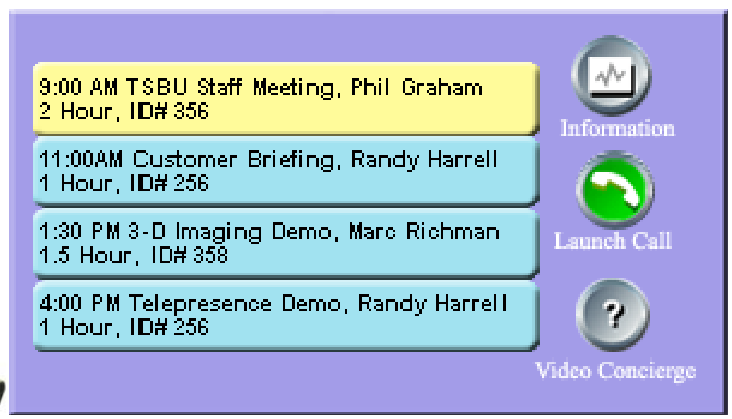
  GSM Cell Phone RF filter
- Cisco Unified CallManager Integration

  Configuration & installation is like IP phone
- 1 Cisco 65" Display & Stand

  Enhanced for TelePresence, Latency reduction

User Interface is Simple

- One button to push from Exchange Calendar
- Daily room schedule from Groupware apps
- XML application on Cisco Unified IP Phone 7970G
- Technology friendly – no 37 button remote
- Concierge services are available

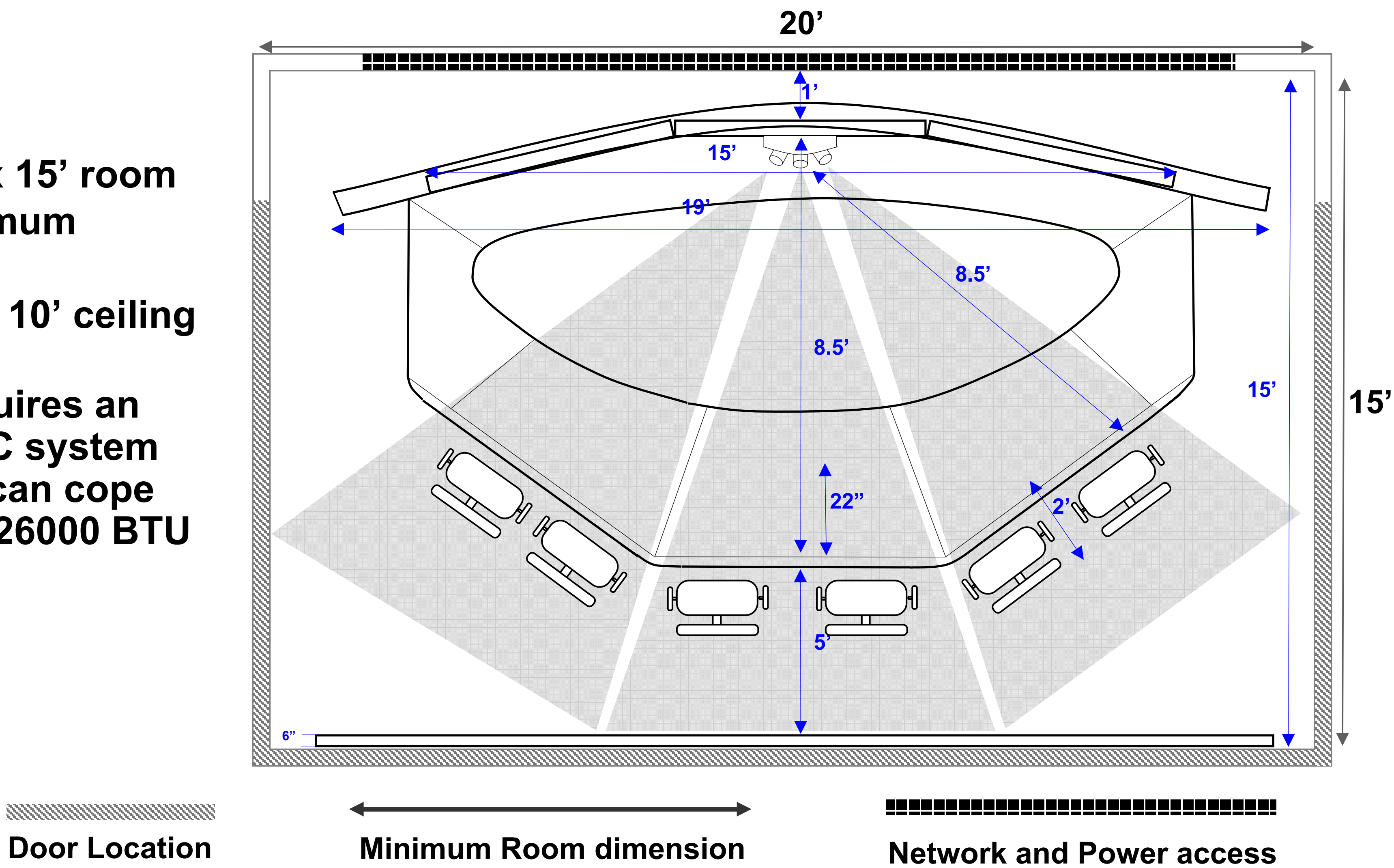


Room Design

Cisco TelePresence CTS 3000

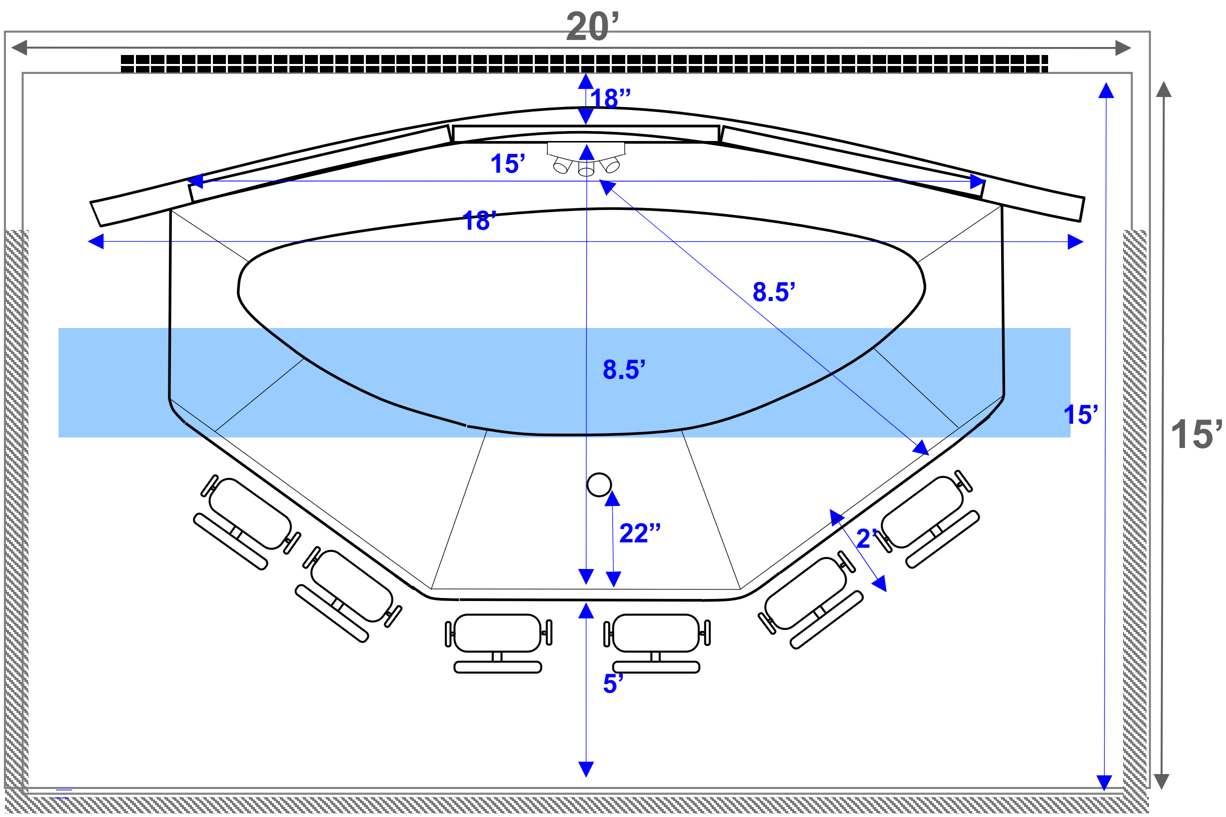
Room Specifications – "Triple"

- 20' x 15' room minimum
- 8' to 10' ceiling
- Requires an HVAC system that can cope with 26000 BTU



Room Specifications "Lighting"

- **Ships with built-in point lighting sources**
- **Designed to work in a normal conference room lighting**
- **Single source well dispersed light should be centered over the table (Shown in blue)**
- **Light arrays should not be centered over the displays**
- **Back wall behind users should be warmer earth tones. Cisco will designate color**
- **Blackout curtains should match viewing wall colors**



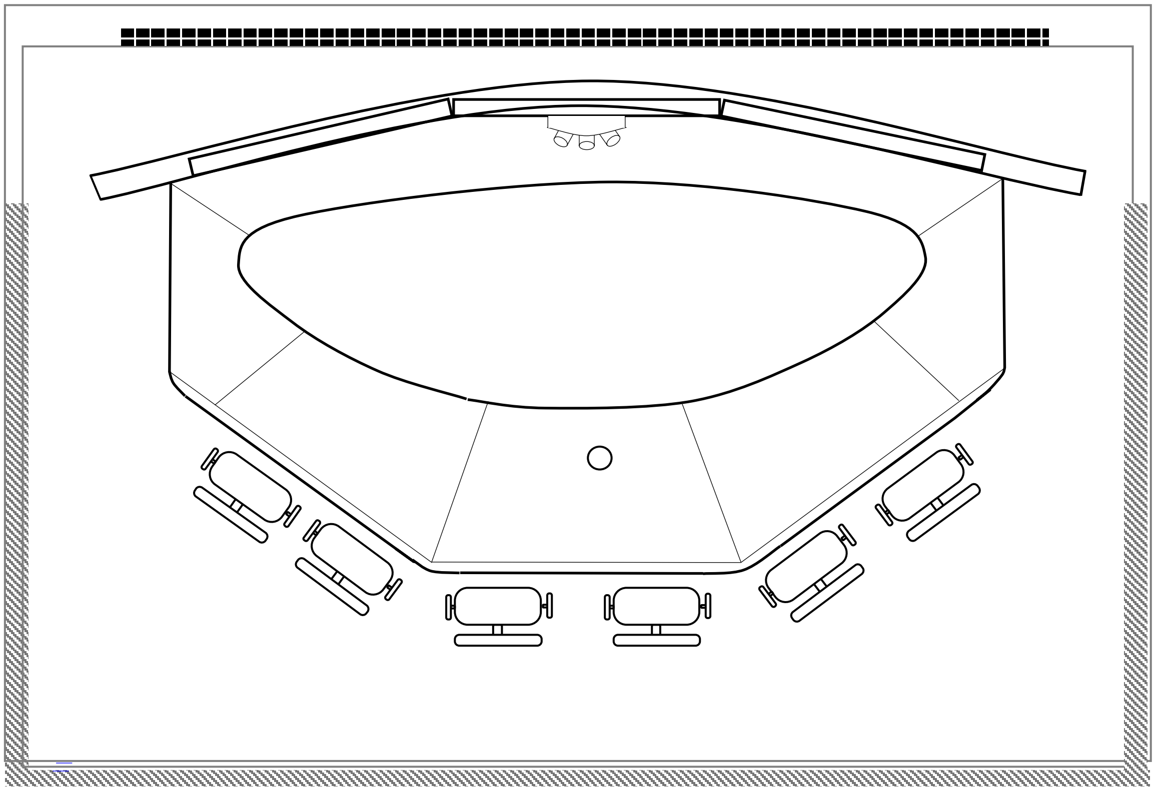
Room Specifications “Acoustics”

8’ to 10’ ceiling

Standard business office ceiling tiles

Carpeted floors

Normal drywall room surfaces



Device Interaction

Protocol and Device Interaction - Codec

- The Codec is the "Workhorse" of the Cisco TelePresence solution (Linux)
- Operates just like a Cisco IP Phone:
  - CDP and 802.1Q for VLAN assignment
  - DHCP and TFTP for configuration and firmware
  - SIP for signaling to Cisco Unified CallManager
  - XML for making/terminating scheduled and ad hoc calls
  - Secondary Codecs are "invisible" to the network. They process video and send it to the Primary Codec. Primary Codec multiplexes video into a single RTP stream

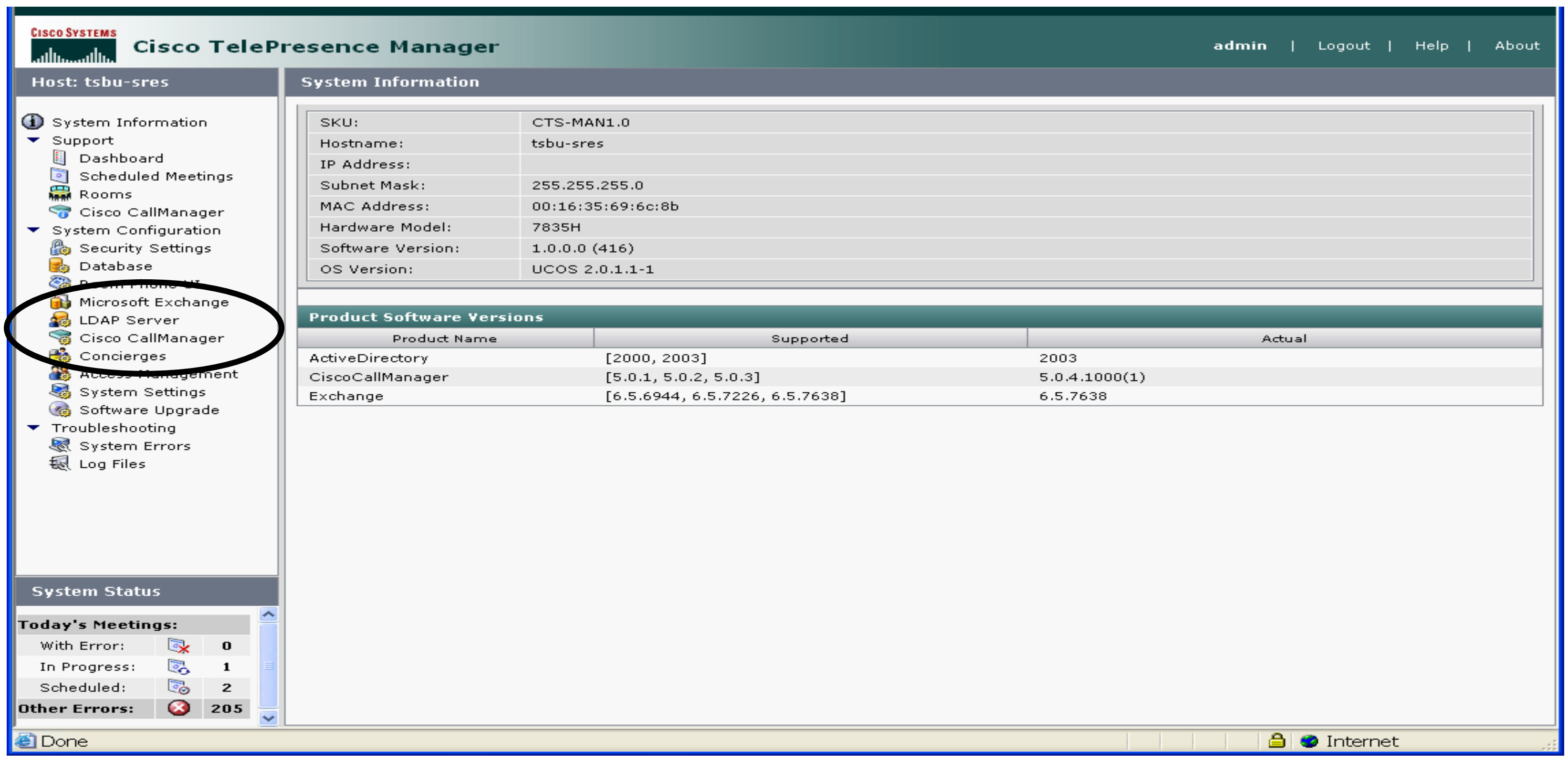
CTS-Codec
Primary

Protocol and Device Interaction – 7970/7971

- One Cisco Unified IP Phone 7970 or 7971 per Cisco TelePresence endpoint – connected to Primary Codec
- Receives 802.3af PoE from Primary Codec. Primary Codec passes CDP, 802.1Q between the phone and the network.
- XML touch screen interface used to place and receive scheduled and ad hoc calls. XML screens are generated by Primary Codec.
- Audio is captured, multiplexed, and delivered as a single standard rtp/udp stream by the Primary Codec – audio on IP phone is only used when making regular voice calls.

Protocol and Device Interaction – Cisco TelePresence Manager

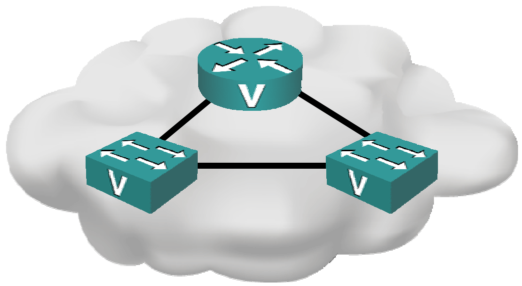
CTM provides the scheduling and management of Cisco TelePresence endpoints. Interacts with Microsoft Exchange (LDAP/WEBDAV), Cisco Unified CallManager (AXL/SOAP, JTAPI) and the Primary Codec (XML, SNMP).



Network Configuration – The Basics

- 797x IP Phone plugs into Primary Codec. Primary Codec supplies PoE to the phone. Only Primary Codec plugs into network. Secondary Codecs are invisible to the network and to Cisco Unified CallManager.
- Primary Codec uses CDP and 802.1Q for placement in the VVID. CDP and 802.1Q/p are passed between the phone and the network. Network switch sees two CDP neighbors on the port.
- Primary Codec uses same DHCP/TFTP process as Cisco Unified IP Phones.
- Primary Codec uses same SIP signaling as Cisco Unified IP Phones. Appears to CallManager as a SIP endpoint.
- Everything communicates over IP.

Network MUST be provisioned for QoS with a minimum bandwidth 45Mbps

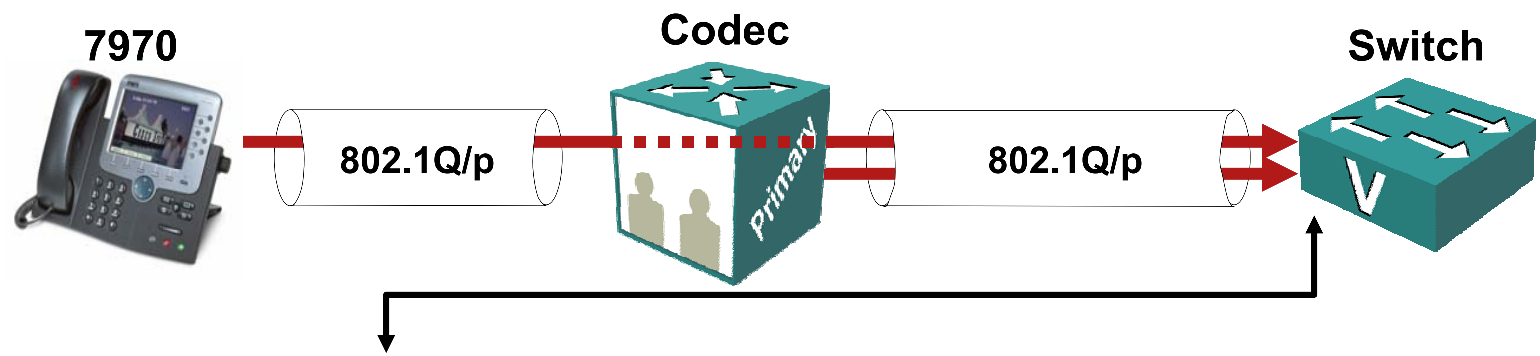


Network Configuration – Switchports and VLANs

Phone and Codec both reside on the VVLAN.

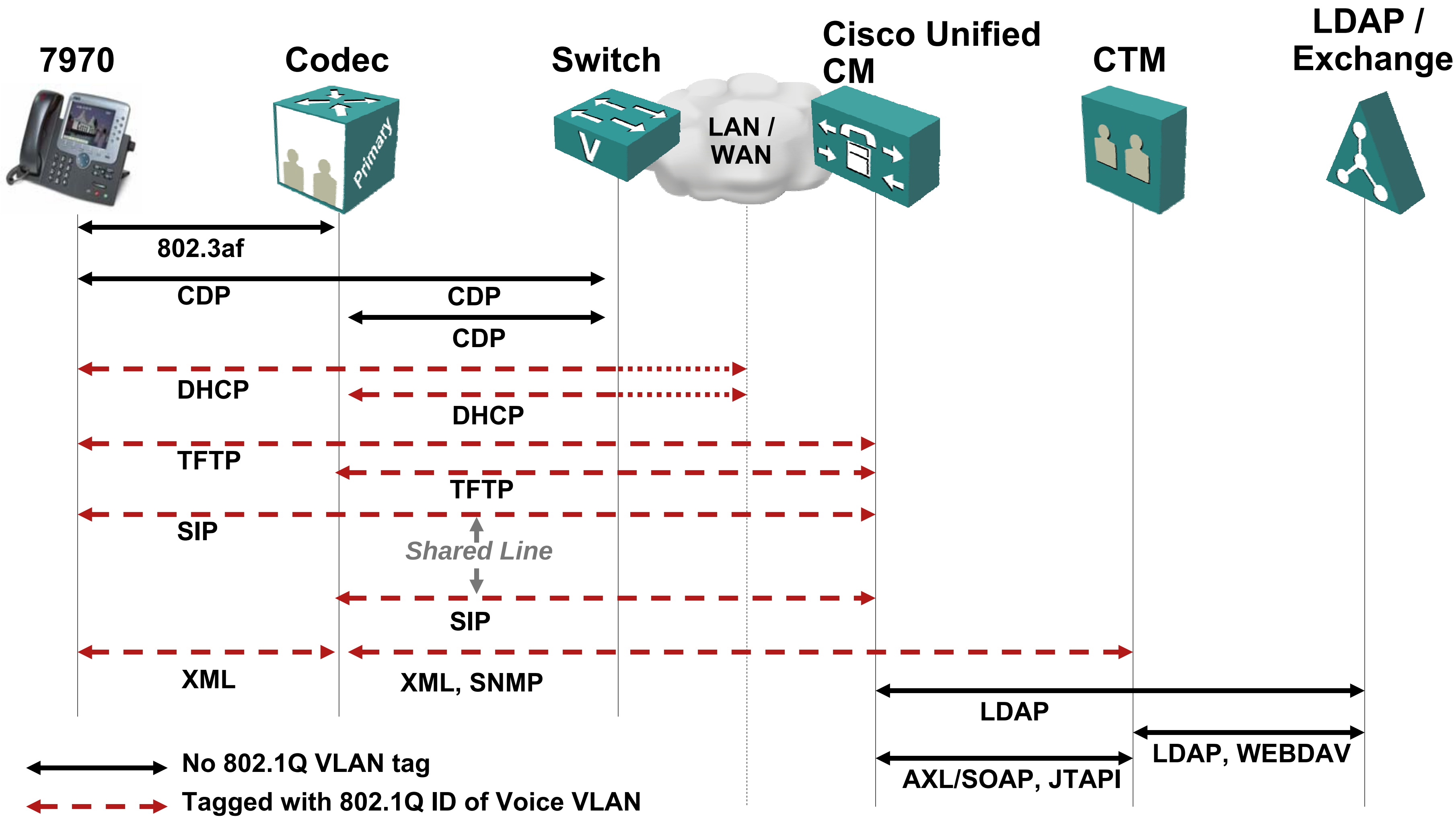
Codec passes 802.1Q tags between the phone and Network.

PC behind the phone is supported, but not necessary due to Ethernet ports in furniture. PC port can be disabled if desired.

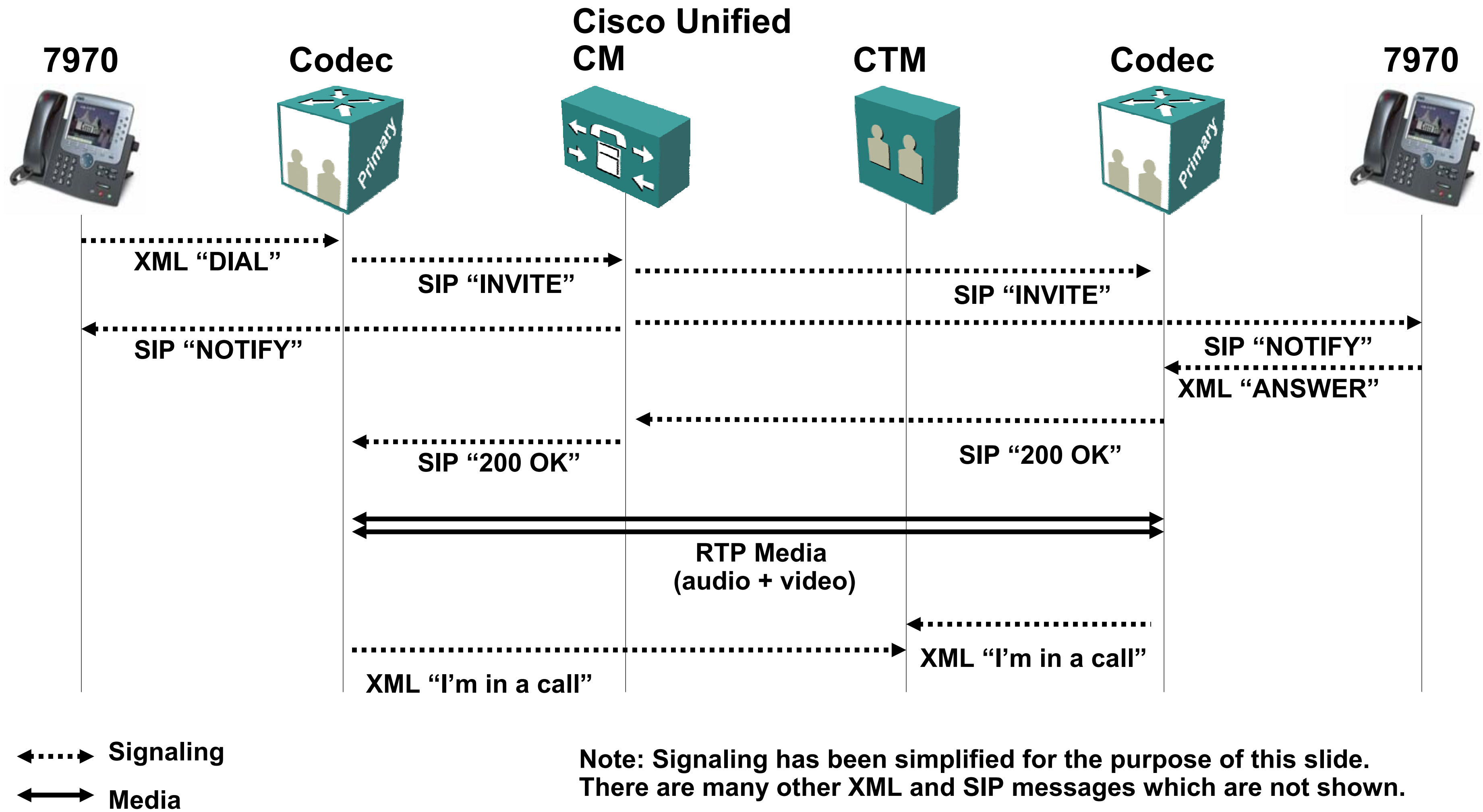


Example:
```
Console(config)#interface Gigabit 0/16
Console(config-if)#switchport mode access
Console(config-if)#switchport access vlan 261
Console(config-if)#switchport voice vlan 262
Console(config-if)#spanning-tree portfast
Console(config-if)#mls qos trust {dscp | cos}
Console(config-if)#service-policy input {telepresence}
! See later for more details on QoS recommendations
```

Protocol Interaction (Signaling Paths)



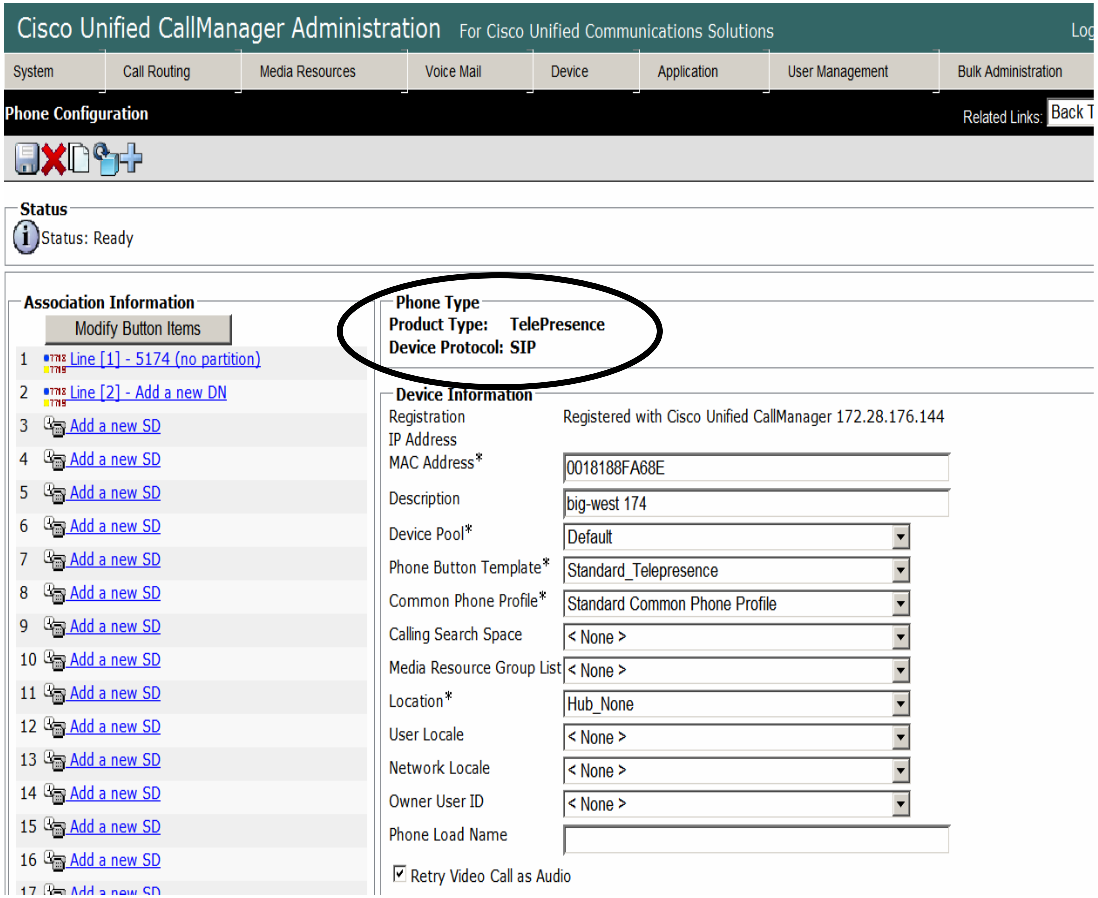
Protocol Interaction (Media Paths)



Note: Signaling has been simplified for the purpose of this slide. There are many other XML and SIP messages which are not shown.

Cisco TelePresence Endpoint Configuration in Cisco Unified CallManager

- Access the Cisco Unified CallManager Administration at https://cm-server-name where cm-server-name is the name or IP address of the server.
- Add the Cisco TelePresence device pack if needed. This adds “TelePresence” to the list of phone models.
- Add the Primary Codec by MAC address just like you would a phone.



Cisco TelePresence Codec Configuration for Cisco Unified CallManager (Cont.)

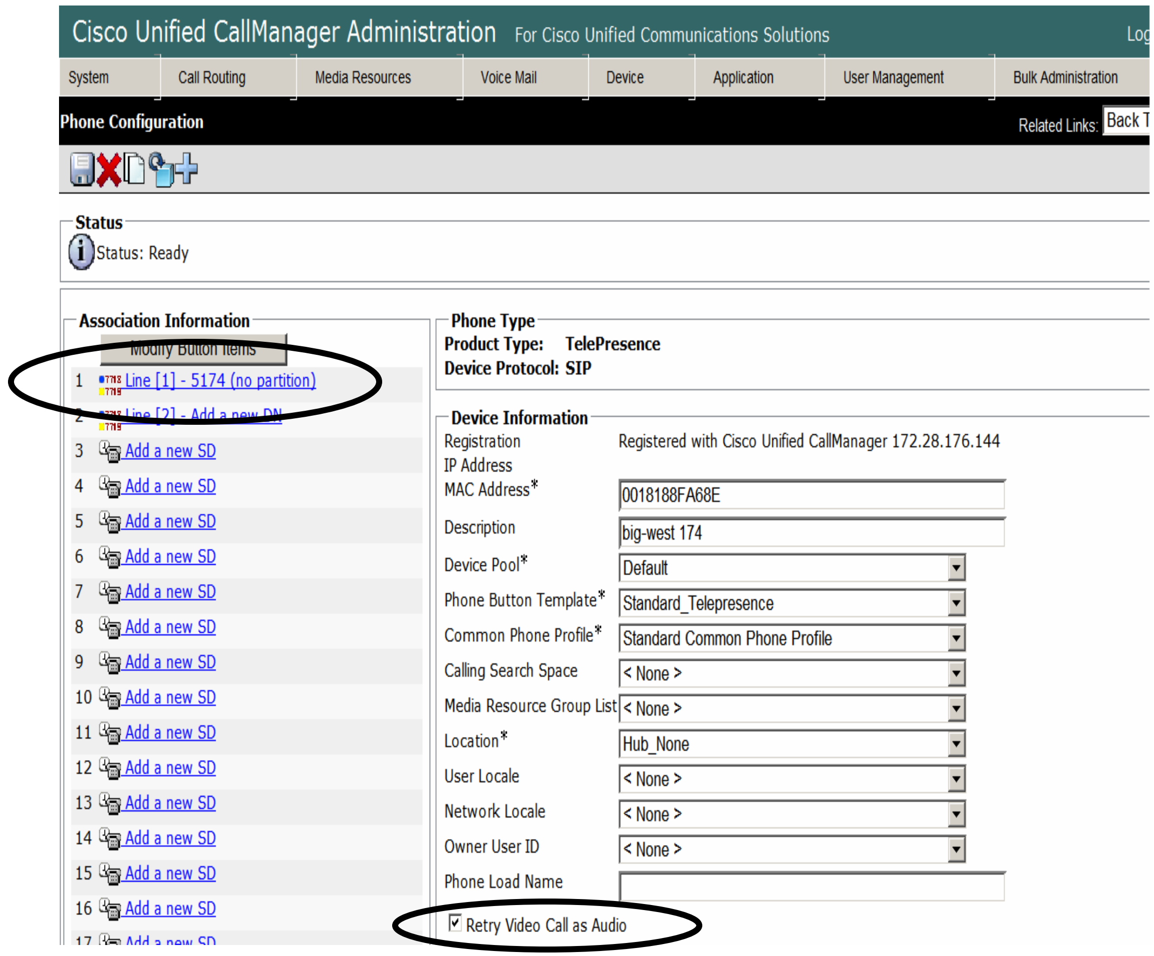
Select the appropriate Cisco TelePresence System type (1000, 2000, 3000)

Assign a Directory Number to the Cisco TelePresence device

Retry Video as Audio setting is optional

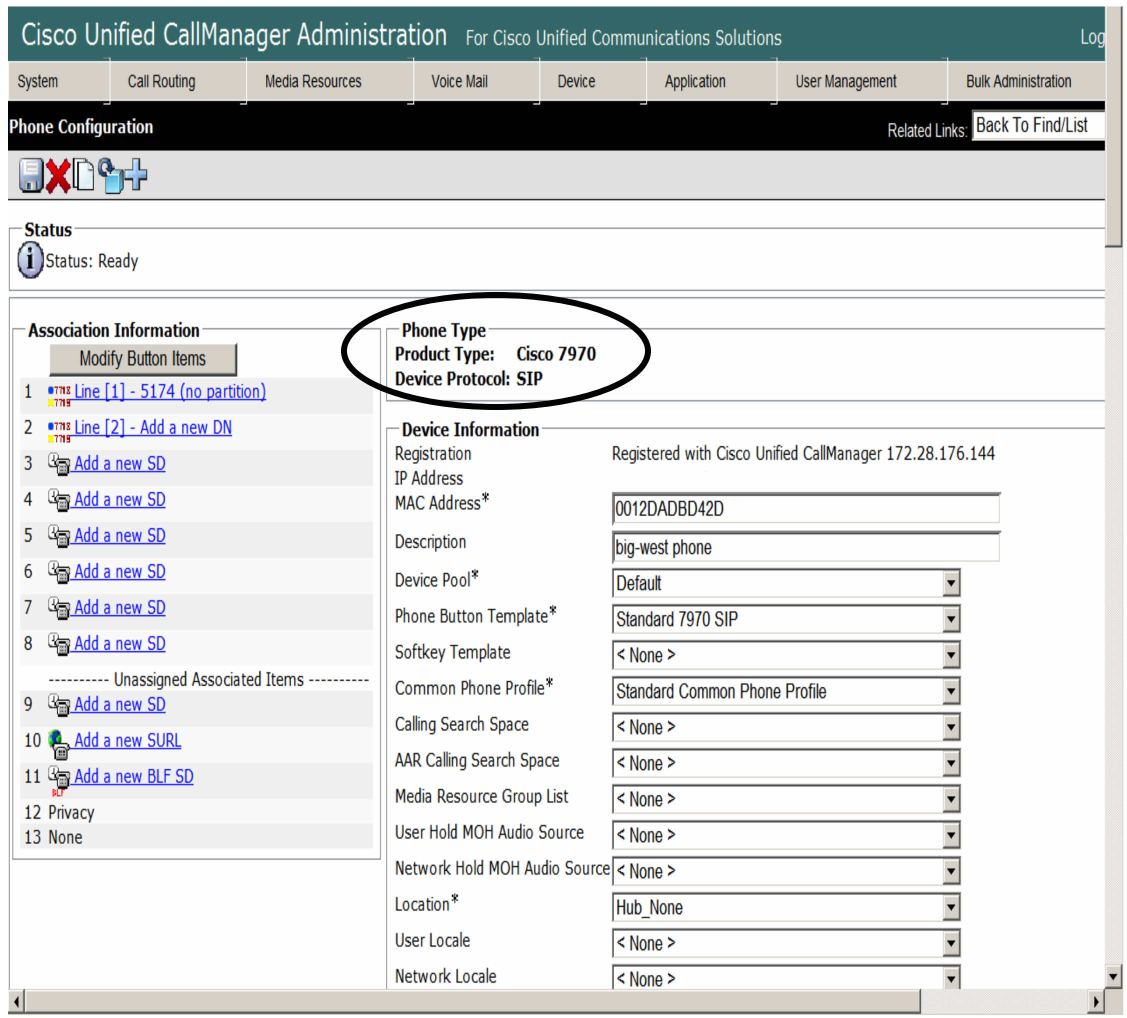
Verify that Allow Control of Device from CTI is selected (used by CTM to monitor the endpoints)

Choose a Video Quality setting (i.e. highest, high, low, lowest)



Cisco TelePresence 7970 Phone Configuration for Unified CallManager

- Access the Cisco Unified CallManager Administration at https://cm-server-name where cm-server-name is the name or IP address of the server
- Add the Cisco 797X as a SIP phone
- Ensure that Speakerphone and Headset are disabled
- Verify that Allow Control of Device from CTI is selected (used by CTM to monitor the endpoints)

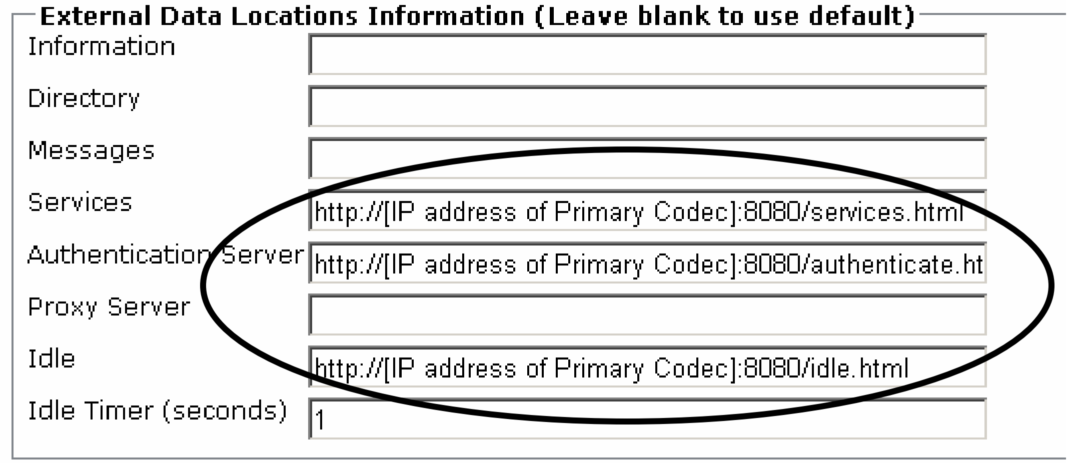


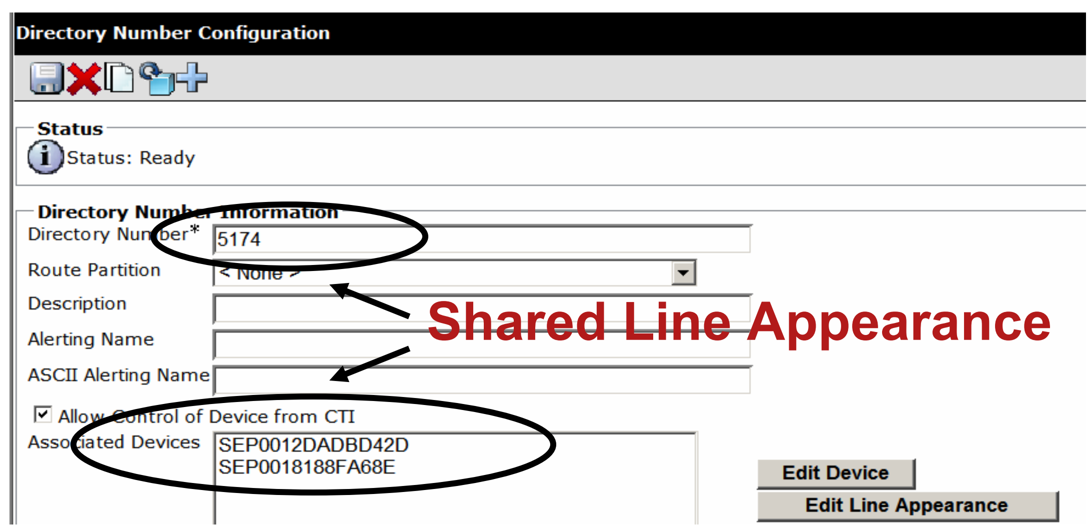
Cisco TelePresence 7970 Phone Configuration for Unified CallManager (Cont.)

Services, Authentication Server, and Idle External Data Location must point to IP address of the Primary Codec. Idle Timer = 1

Assign the same Directory Number to the primary line of the 7970 IP Phone that was assigned to the Cisco TelePresence device (shared line appearance)

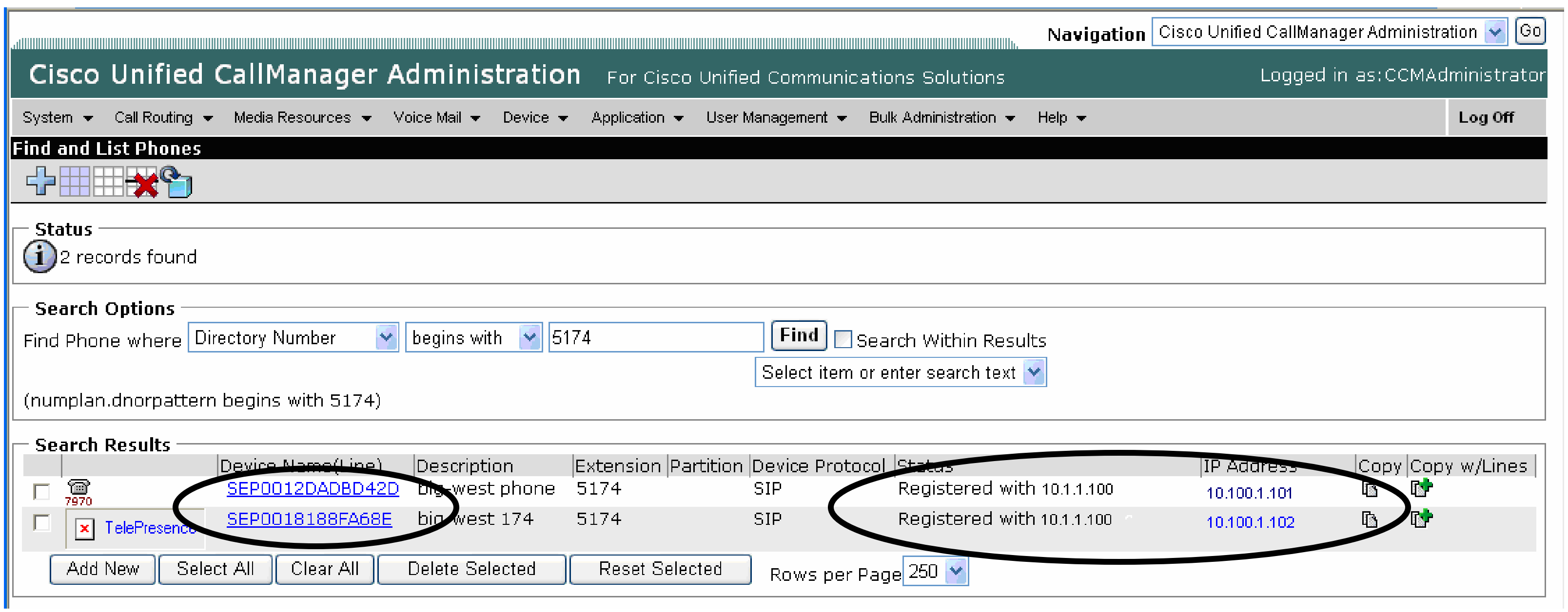
Max Calls = 2, Busy Trigger = 1





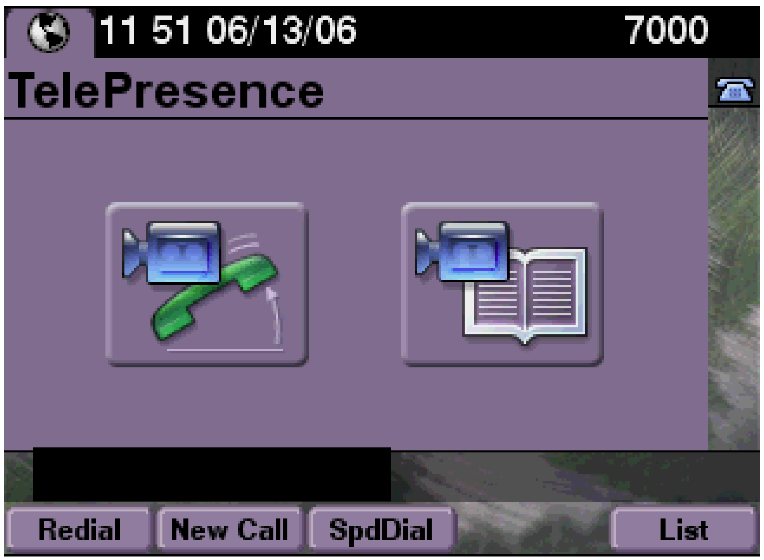
Configuration Verification

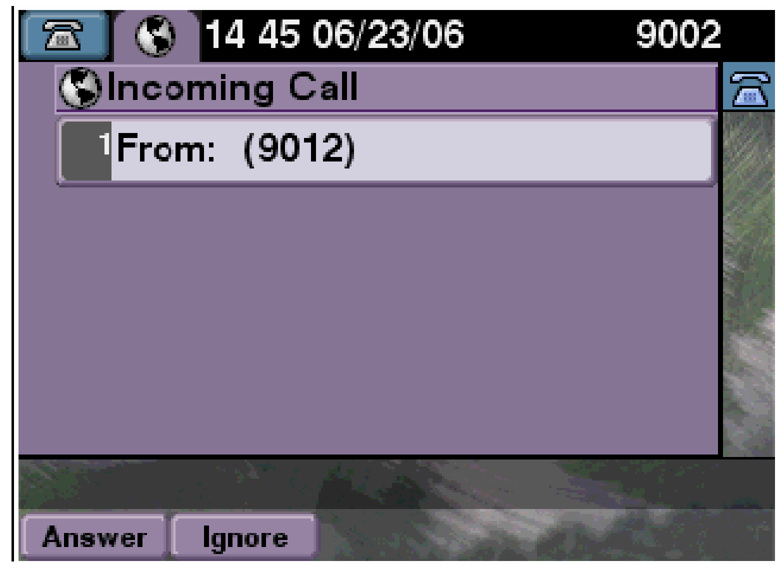
Codec and 7970 IP Phone register with Cisco Unified CallManager



Starting a Meeting

Cisco TelePresence Phone User Interface





Redial   New Call   SpdDial

Answer   Ignore

Meeting in Progress

Options Available

Hld/Resm End Call Tones Volume List

QoS Design

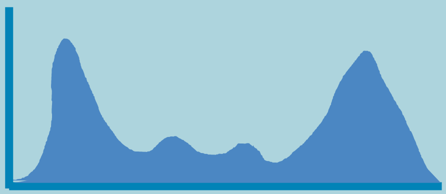
Overview of Cisco TelePresence Network Requirements

- Latency ≤ 250 ms
- Jitter ≤ 10 ms
- Loss ≤ 0.04%

One-Way Requirements

- Bandwidth 4 Mbps per screen
  + voice (64 kbps streams)
  + auxiliary streams
  + burst allowance
  + L2 overhead
- Call Admission Control must be enabled (within Cisco CallManager)

Cisco TelePresence



- Bursty
- Drop sensitive
- Delay sensitive
- Jitter sensitive
- UDP priority

Bandwidth Requirements Breakdown: CTS-3000 at 1080p

▪ 3 primary video streams (4 Mbps each):	12 Mbps
▪ 3 primary audio streams (64 Kbps each):	192 Kbps
▪ 1 auxiliary audio stream:	64 Kbps
▪ 1 auxiliary video stream:	400 Kbps
▪ TOTAL (average at L3):	12,656 Kbps
▪ Add 20% for burst and L2 overhead:	≈ 15 Mbps

Bandwidth Requirements Breakdown: CTS-3000 at 720p

- 3 primary video streams (2 Mbps each): 6 Mbps
- 3 primary audio streams (64 Kbps each): 192 Kbps
- 1 auxiliary audio stream: 64 Kbps
- 1 auxiliary video stream: 400 Kbps
- TOTAL (average at L3): 6,656 Kbps

- Add 20% for burst and L2 overhead: ≈ 8 Mbps

Bandwidth Requirements Breakdown: CTS-1000 at 1080p

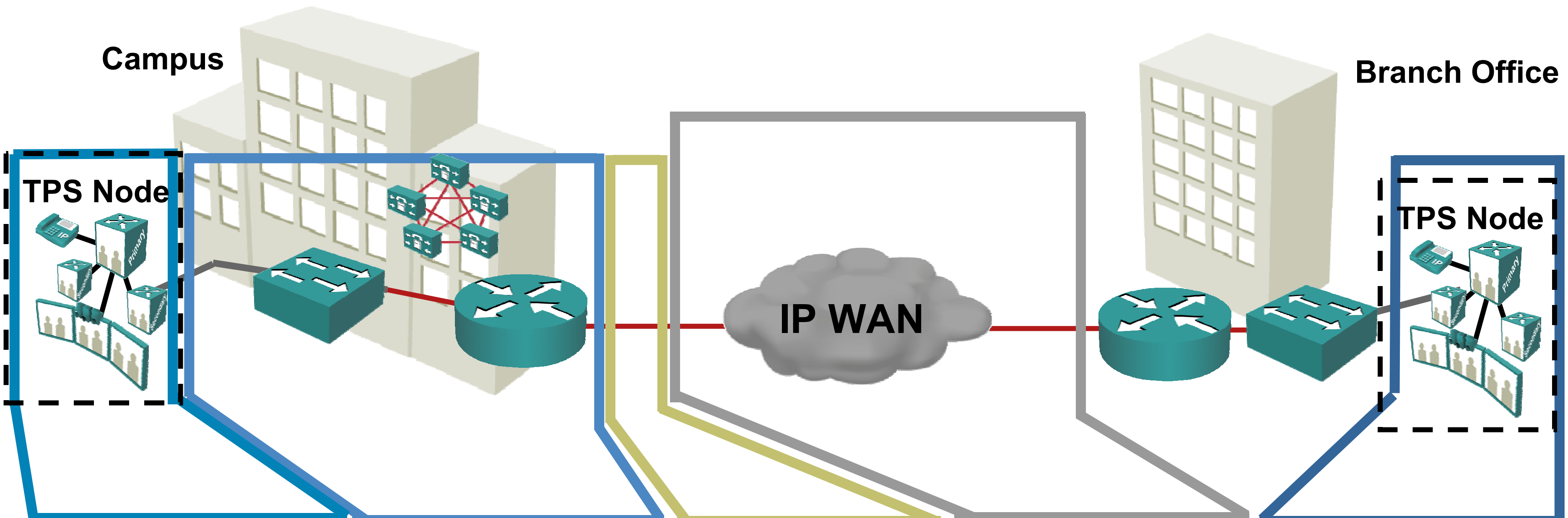
▪ 1 primary video streams (4 Mbps each):	4 Mbps
▪ 1 primary audio streams (64 Kbps each):	64 Kbps
▪ 1 auxiliary audio stream:	64 Kbps
▪ 1 auxiliary video stream:	400 Kbps
▪ TOTAL (average at L3):	4,528 Kbps
▪ Add 20% for burst and L2 overhead:	≈ 5.5 Mbps

Bandwidth Requirements Breakdown: CTS-1000 at 720p

▪ 1 primary video streams (2 Mbps each):	2 Mbps
▪ 1 primary audio streams (64 Kbps each):	64 Kbps
▪ 1 auxiliary audio stream:	64 Kbps
▪ 1 auxiliary video stream:	400 Kbps
▪ TOTAL (average at L3):	2,528 Kbps

- Add 20% for burst and L2 overhead: ≈ 3 Mbps

Cisco TelePresence Service Level Requirements: Latency



CODEC	Queuing	Serialization	Propagation and Network	Jitter Buffer + CODEC
75 ms	Variable	Variable	Fixed (6.3 µs/Km) + Network Delay (Variable)	75 ms

End-to-End Delay (Must Be ≤ 250 ms)

Classification and Marking Design: RFC 4594 Configuration Guidelines for DiffServ Classes

Application	L3 Classification		IETF
	PHB	DSCP	RFC
Network Control	CS6	48	RFC 2474
VoIP Telephony	EF	46	RFC 3246
Call Signaling	CS5	40	RFC 2474
Multimedia Conferencing	AF41	34	RFC 2597
Real-Time Interactive	CS4	32	RFC 2474
Multimedia Streaming	AF31	26	RFC 2597
Broadcast Video	CS3	24	RFC 2474
Low-Latency Data	AF21	18	RFC 2597
OAM	CS2	16	RFC 2474
High-Throughput Data	AF11	10	RFC 2597
Best Effort	DF	0	RFC 2474
Low-Priority Data	CS1	8	RFC 3662

Implementing Cisco TelePresence Classification and Marking

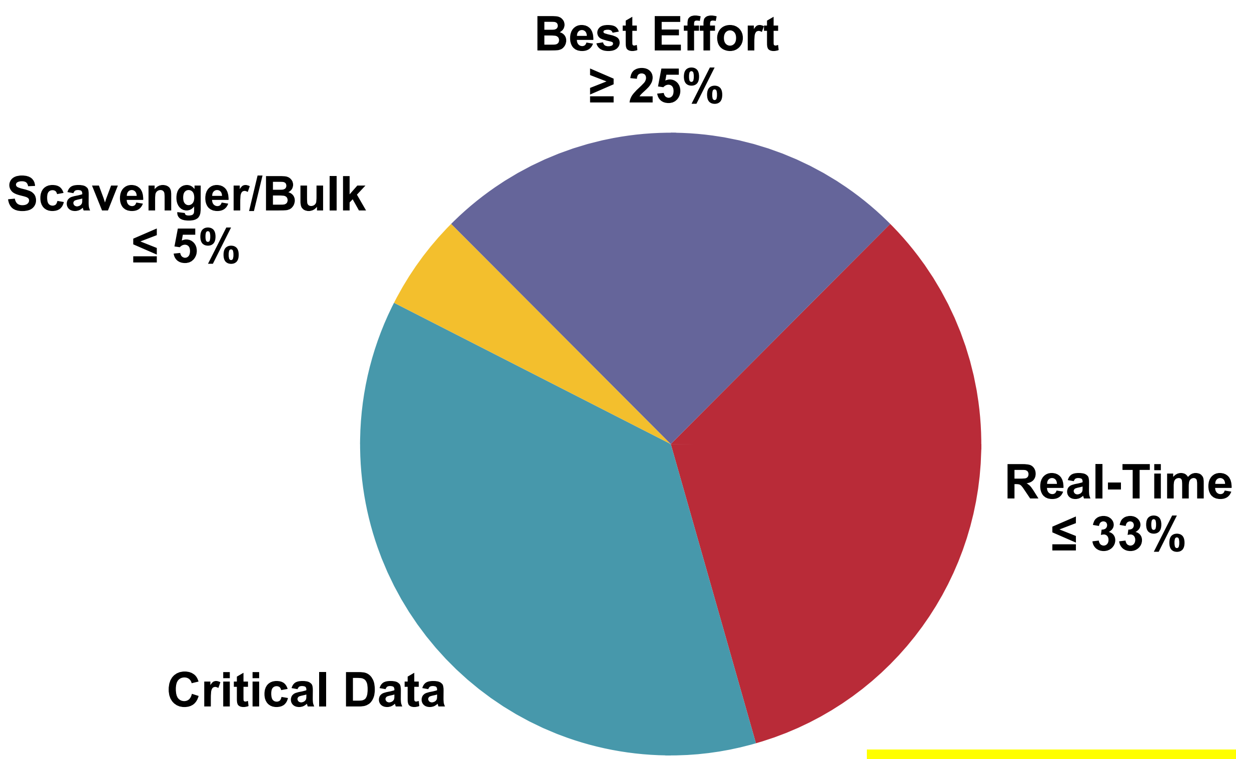
- Cisco Unified CallManager is currently unable to distinguish VT Advantage from Cisco TelePresence and will mark both (by default) to AF41.
- Cisco TelePresence needs to be remarked at the access-edge switchport (via a policy-map applied to Cisco TelePresence unit ports) to CS4.
  - Optionally policers could be deployed at the access-edge switchports to prevent physical access and trust abuse of Cisco TelePresence ports.
- Cisco TelePresence can be distinguished from VT Advantage (which will remain AF41) in downstream DiffServ policies.

Cisco TelePresence Queuing

- Optimally, assign Cisco TelePresence to a dedicated LLQ for the best SLAs (IOS supports multiple LLQs)

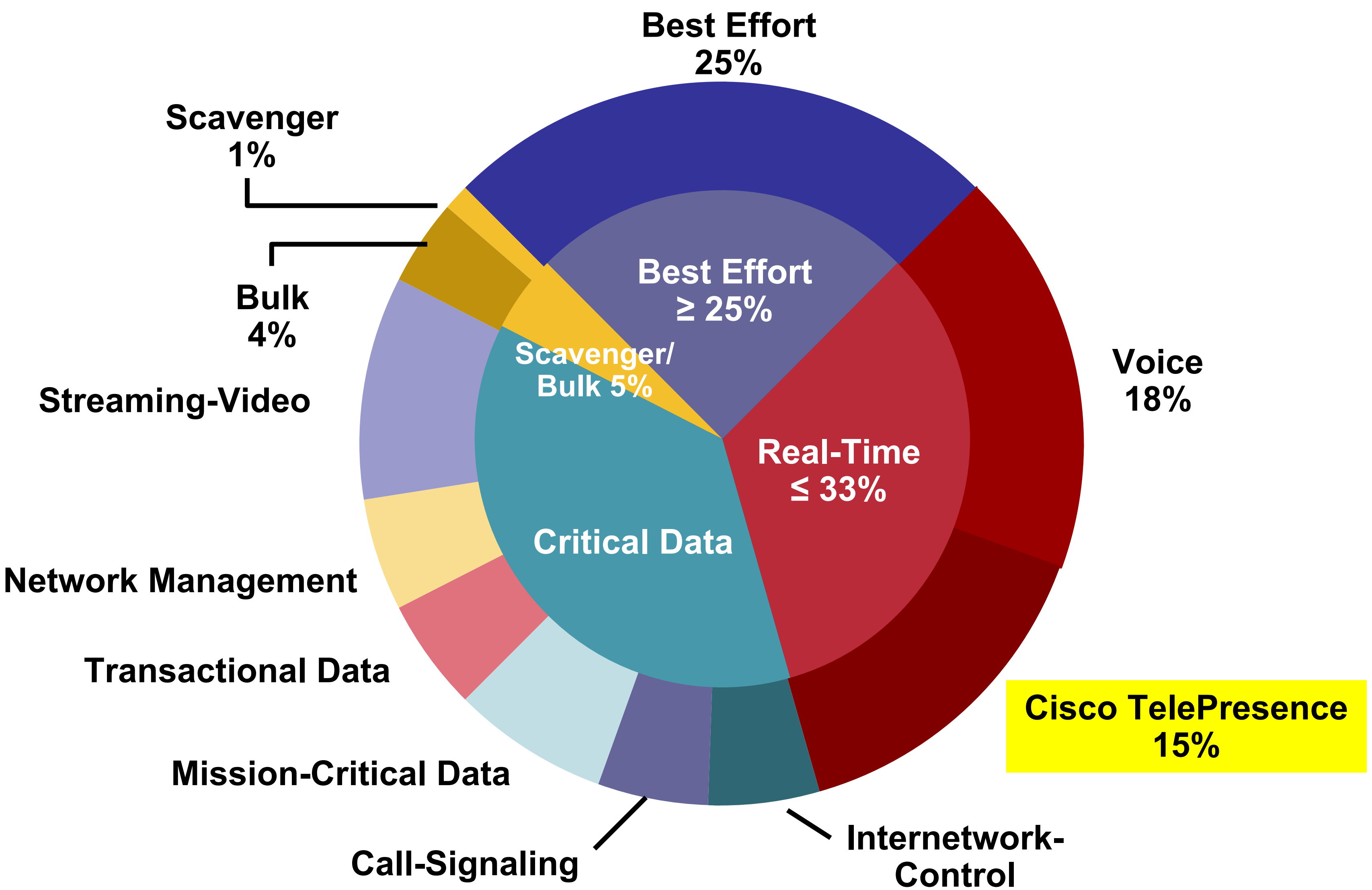
  Recommend 33% LLQ limit to preserve voice, video + data transparent convergence

Campus Queuing Design:
Realtime, Best Effort, Critical Data & Scavenger Queuing Example



Including: Voice +
Cisco TelePresence

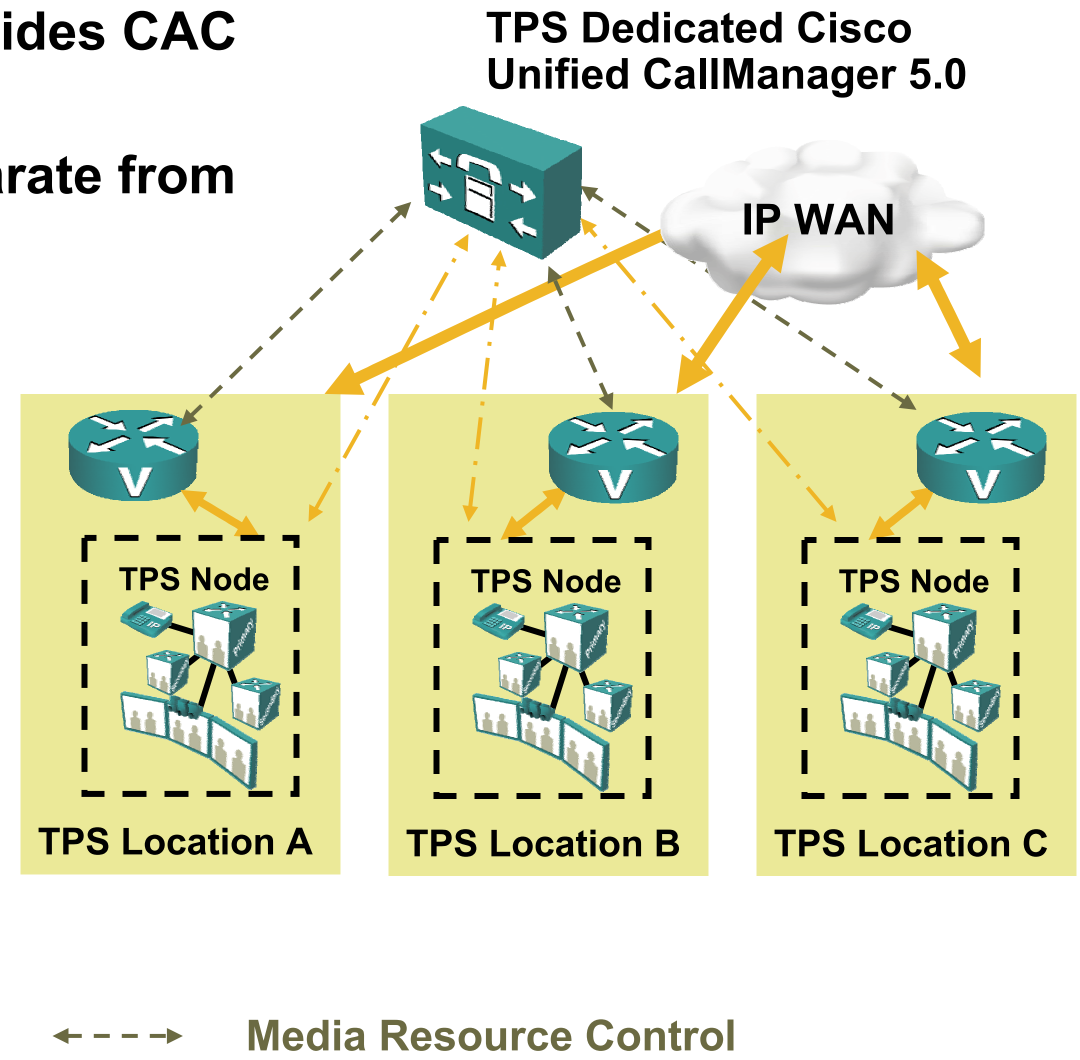
Campus and WAN/VPN Queuing Design:
Granular – but Compatible – Queuing Design Example



Cisco Locations CAC for Cisco Unified CallManager

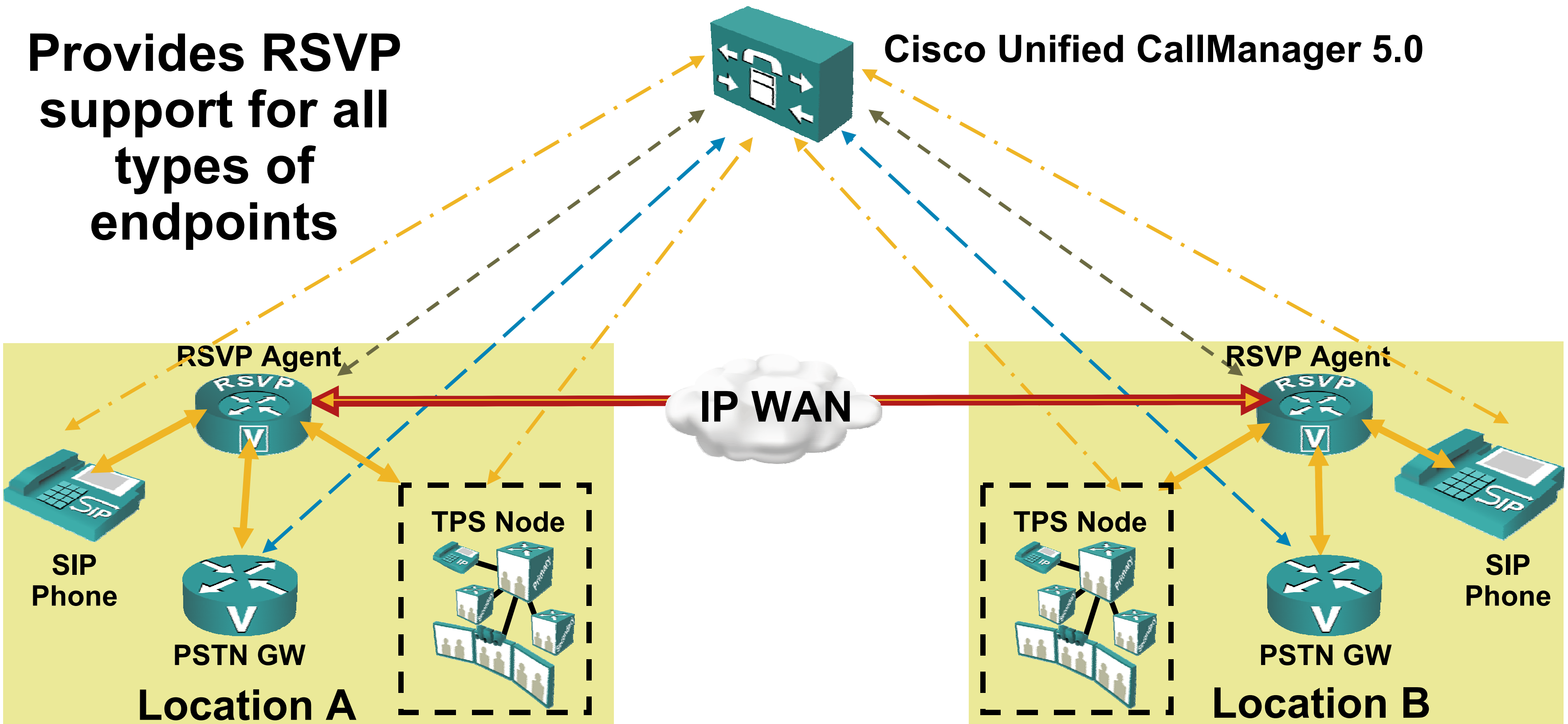
- **Locations based CAC provides CAC amongst TPS Nodes**
- **Locations for TPS are separate from VOIP and IP/VC**

Locations Bandwidth	
Location	Bandwidth
TPS Loc. A	12000
VOIP-IP/VC Loc. A	3000
TPS Loc. B	12000
VOIP-IP/VC Loc. B	3000
TPS Loc. C	12000
VOIP-IP/VC Loc. C	3000



Cisco IOS RSVP Agent for Cisco Unified CallManager

Provides RSVP support for all types of endpoints



Media Resource Control

MGCP or H.323

SIP

RSVP

RTP

Supported in IOS release 12.4(6)T on 26xx-XM, 2691, 28xx, 37xx and 38xx series platforms

Summary

- Looked at the basic tenants of “Presence” in a Unified Communications world
- Described how the Cisco Unified Presence Server is constructed and how it works
- Used the Cisco, Microsoft and IBM clients to describe the different ways they connect to the presence server and the services they offer.
- A quick look behind the gloss of the Cisco TelePresence solution……

References

References

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