Cisco live!

What You Make Possible

TOMORROW starts here.

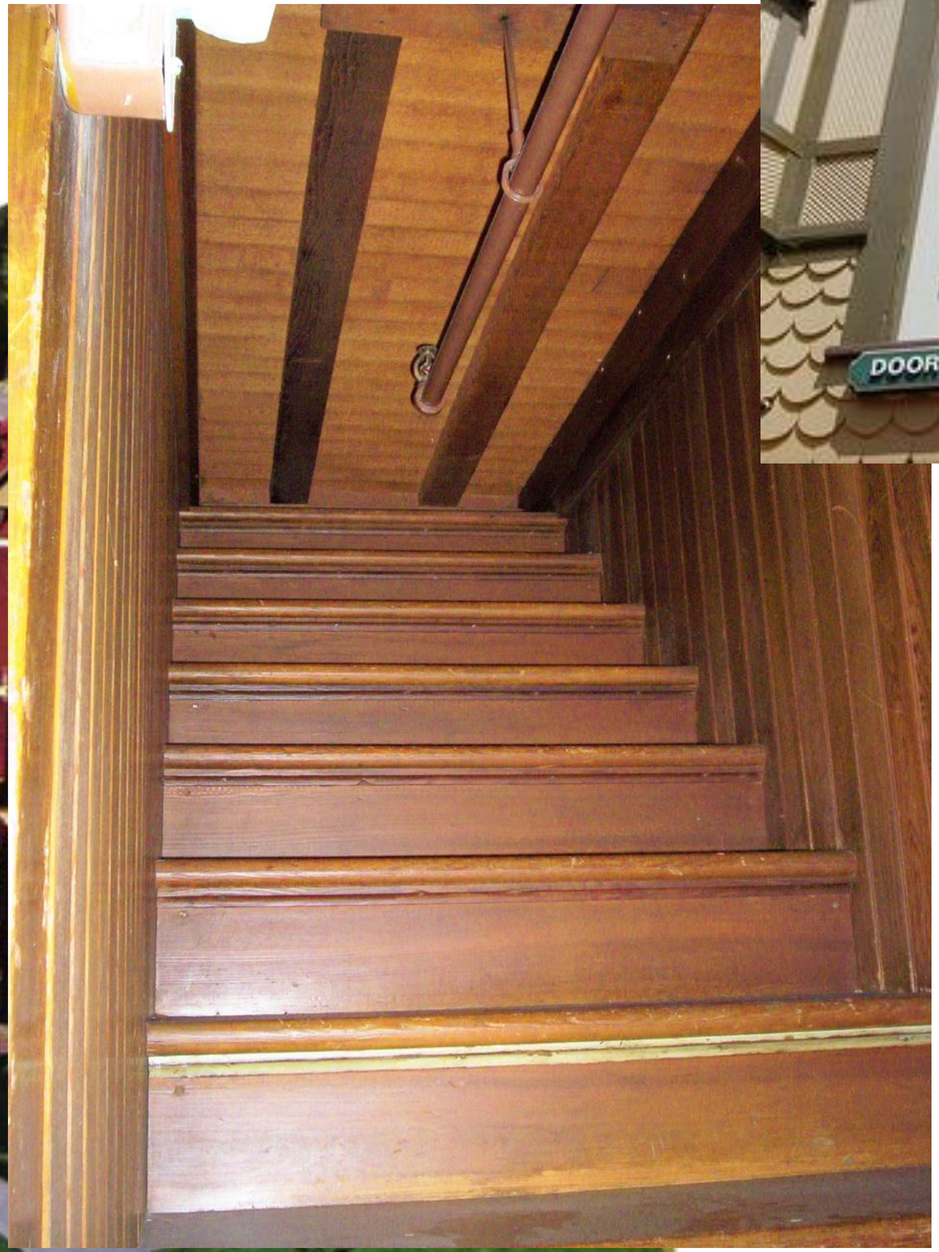
cisco

Implementing Enterprise TelePresence and Video Communication Solution

BRKEVT-2615

TOMORROW starts here.

Winchester Mystery House



The Vision

"Every visual communications device, able to talk to any visual communications device, in any location, over any network, in any configuration."

Fundamental to a Pervasive Video World

Quality

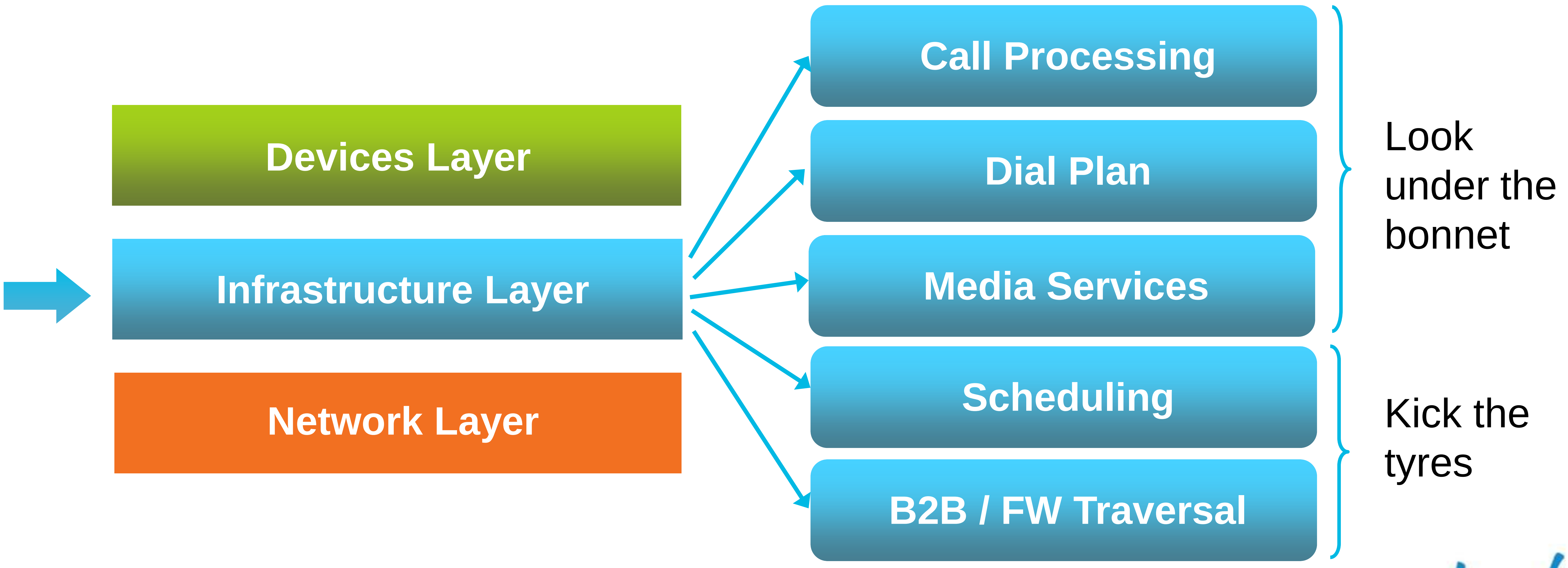
Availability

Simplicity

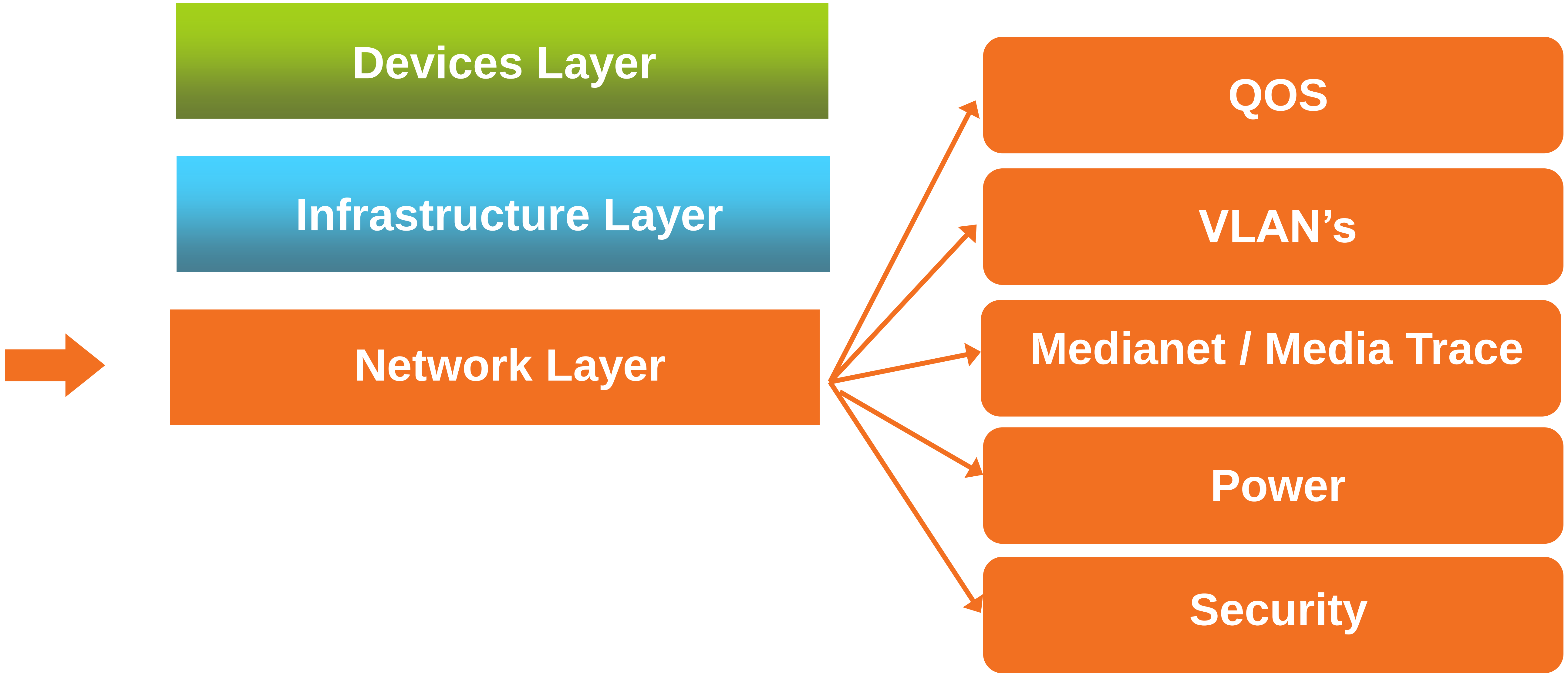
Reliability

Agenda

- Key to achieve the vision of "Any to Any communication" is getting the infrastructure layer right:



Does the Network Matter ?



http://www.cisco.com/en/US/docs/solutions/Enterprise/WAN_and_MAN/QoS_SRND_40/QoSCampus_40.html#wp1098008

From Video “Silo” to Pervasive Video

Video “Silo”	Pervasive Video
Point to Point	Many to Many
Fixed Room and Location	Any Device Anywhere
Connecting Rooms	Connecting People
Communicate Internally	Collaborate Globally Across Organisations
Largely Proprietary Mechanisms	Open Standards and Interoperability

Let’s Begin

Firewall Traversal
Inter-Company Strategy

Dial Plan
Integration between CUCM and VCS Routing Features

Scheduling
Point to Point and Multipoint Scheduling across VCS and CUCM platforms

Design & Architecture
Overall Design Strategy for Implementing Integrated Enterprise TelePresence & Video

Call Processing
Best Practice for Multisite Environment

Conferencing
Multipoint Facility to Unify Different Endpoint Types on the TelePresence Immersion Curve

Fundamentals, Design & Architecture

Call Processing

Dial Plan

Conferencing

Scheduling

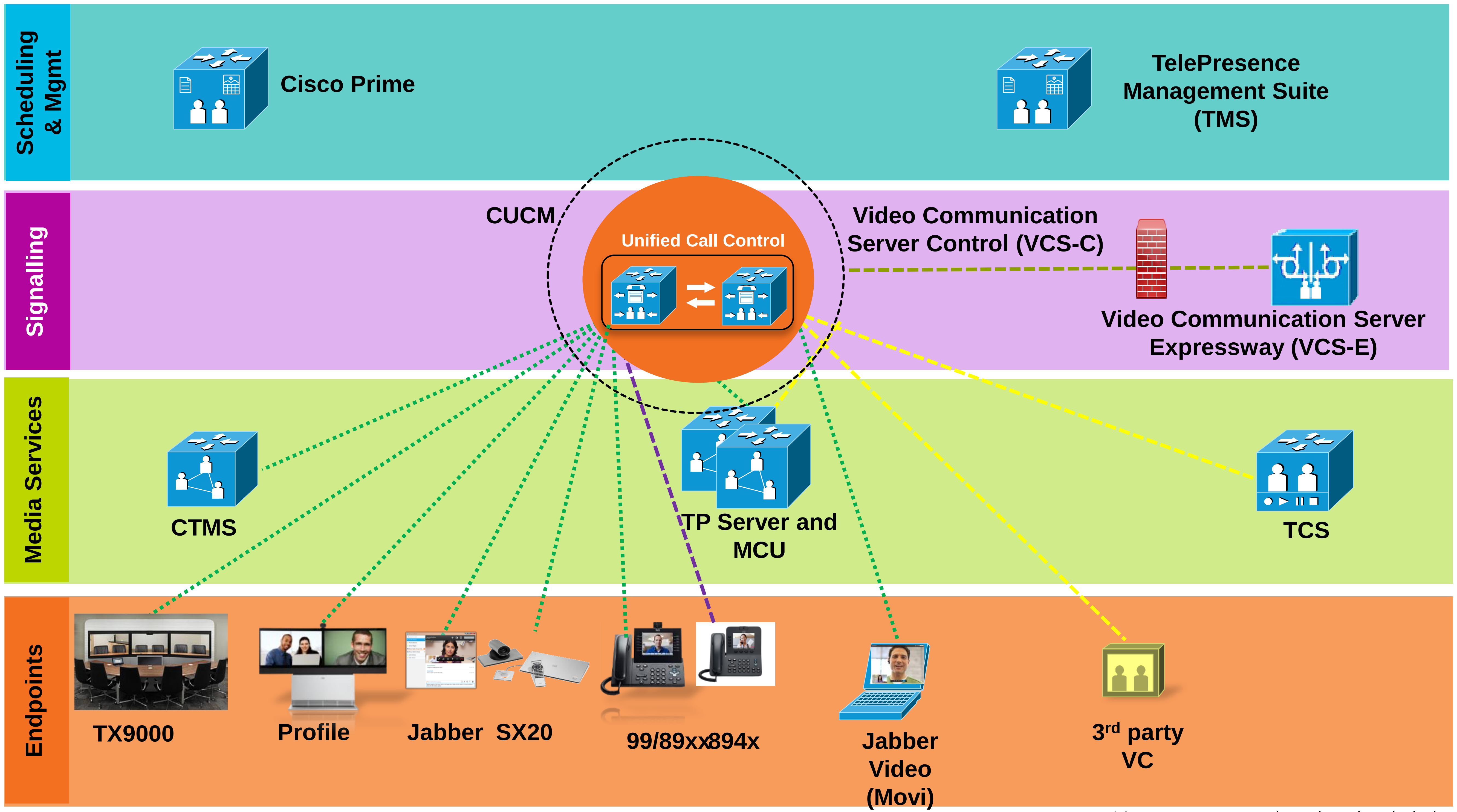
Business to Business

Fundamentals Acronyms

For Your Reference

Acronym	
CSF	Client Services Framework
CTMS	Cisco TelePresence Multipoint Switch
CTRS	Cisco TelePresence Recording Server
CTS	Cisco TelePresence System
CUCM	Cisco Unified Communications Manager
TP	TelePresence
VC	Video Conferencing
VCS	Video Communication Server

Architecture Overview



SIP H.323 SIP or H.323 SCCP H.460

** Icons are representative only and not inclusive of the full set of endpoints and infrastructure*

Fundamentals – Key Infrastructure Pieces

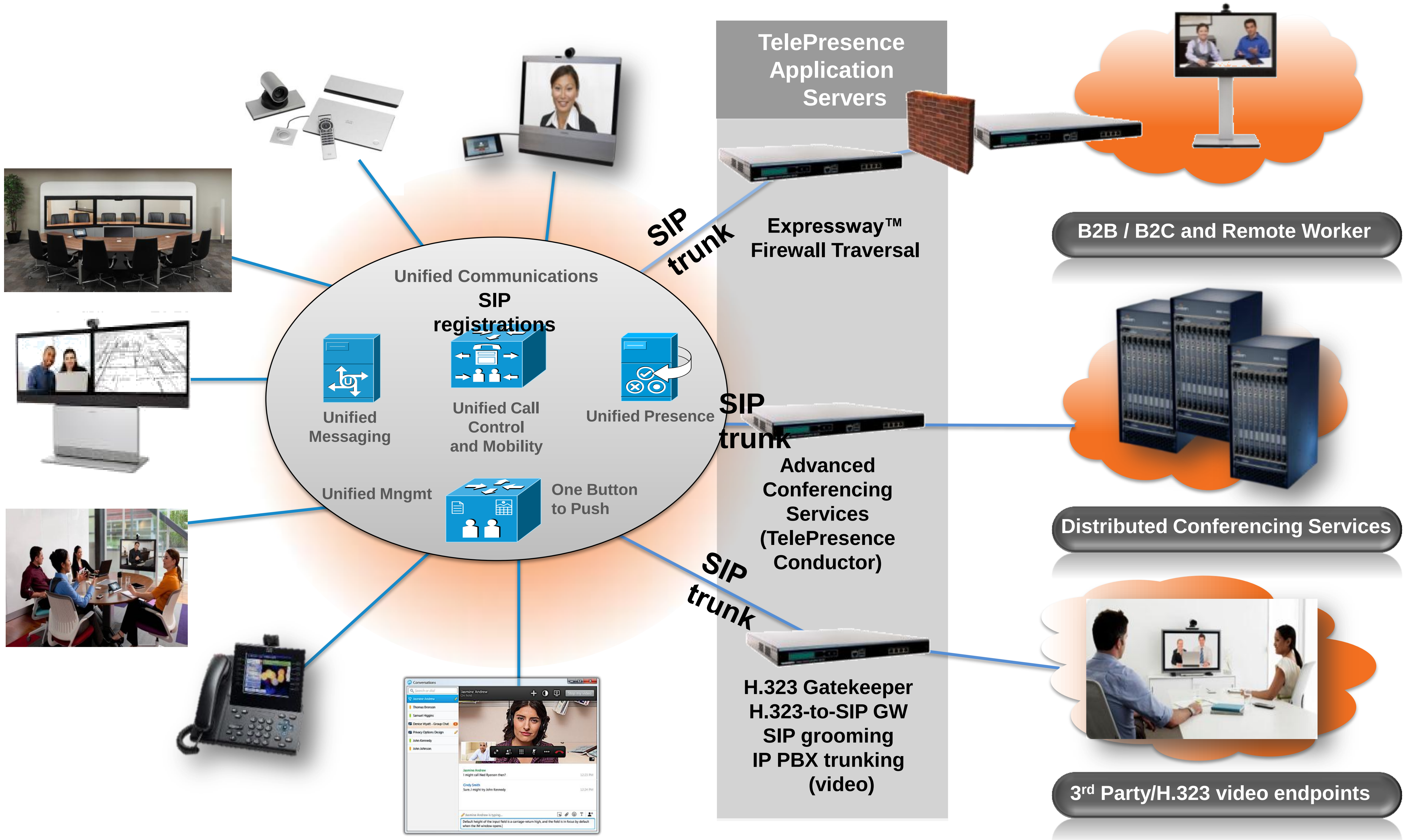
Unified Call Control

- Registration and call control for CTS, IP phones, UC clients (Jabber) mobility clients, SX, EX, MX, C Series Endpoints
- 60,000 lines per cluster, no real limitation on concurrent calls
- Supports a variety of voice and video protocols, SCCP, SIP, H.323 & MGCP
- Proven robustness, reliability and scalability

- Advanced video and TelePresence capabilities
- Standards based registration, call control and firewall traversal for SIP & H.323 Devices and Infrastructure devices
- Capacity per appliance:
  - 2,500 registrations
  - Up to 500 non traversal calls
  - Up to 100 traversal calls
- Up to 6 VCS Control in a cluster

(4 active, 2 standby)

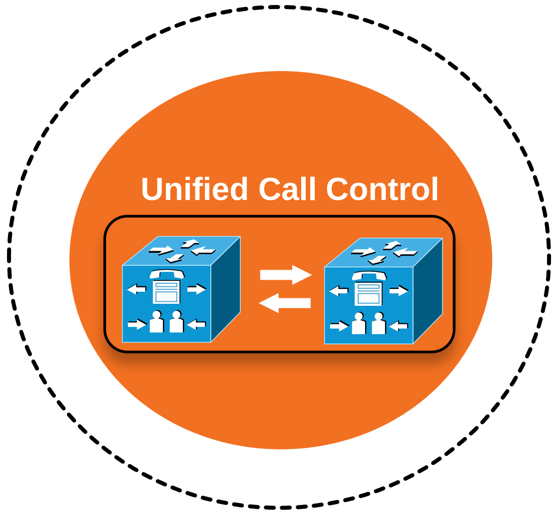
Evolution of Signalling & Call Control Architecture



Where to Register? CUCM or VCS?

Register to VCS when these features are required

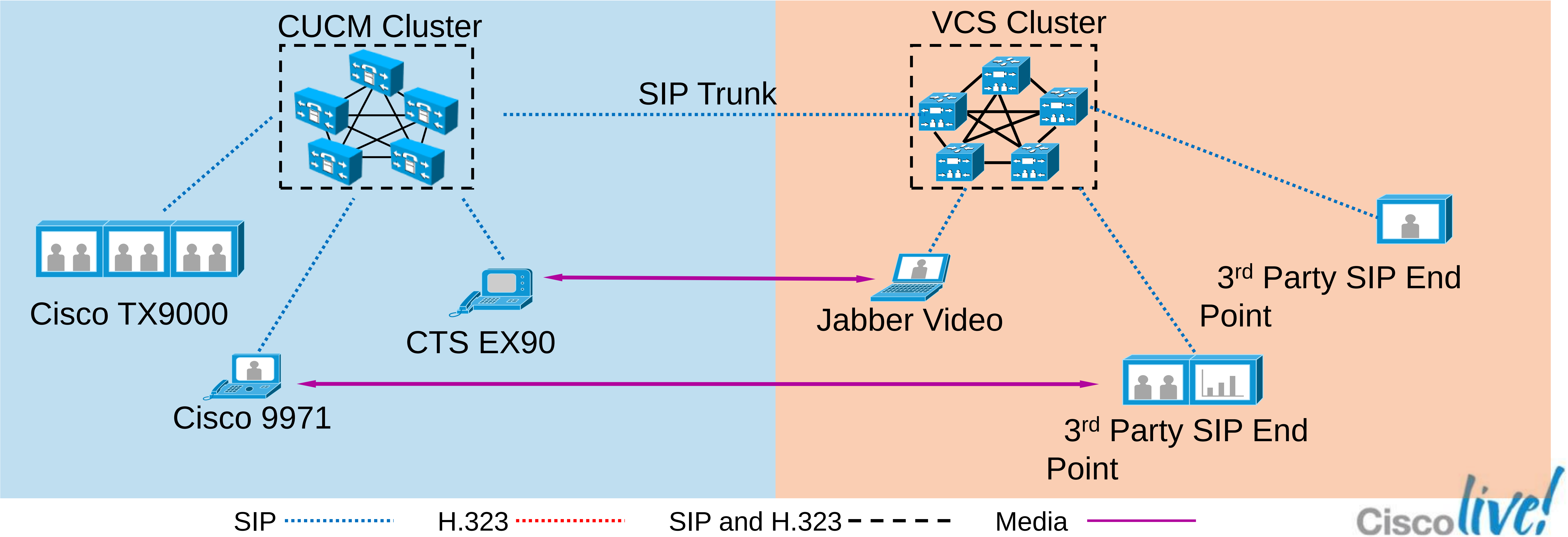
- H.323 registration
- SIP/H.323 Interworking
- IPv6
- Secure Firewall Traversal using H.460.18/19
- Cisco Jabber Video



Design & Architecture Considerations

Media Flow – SIP to SIP

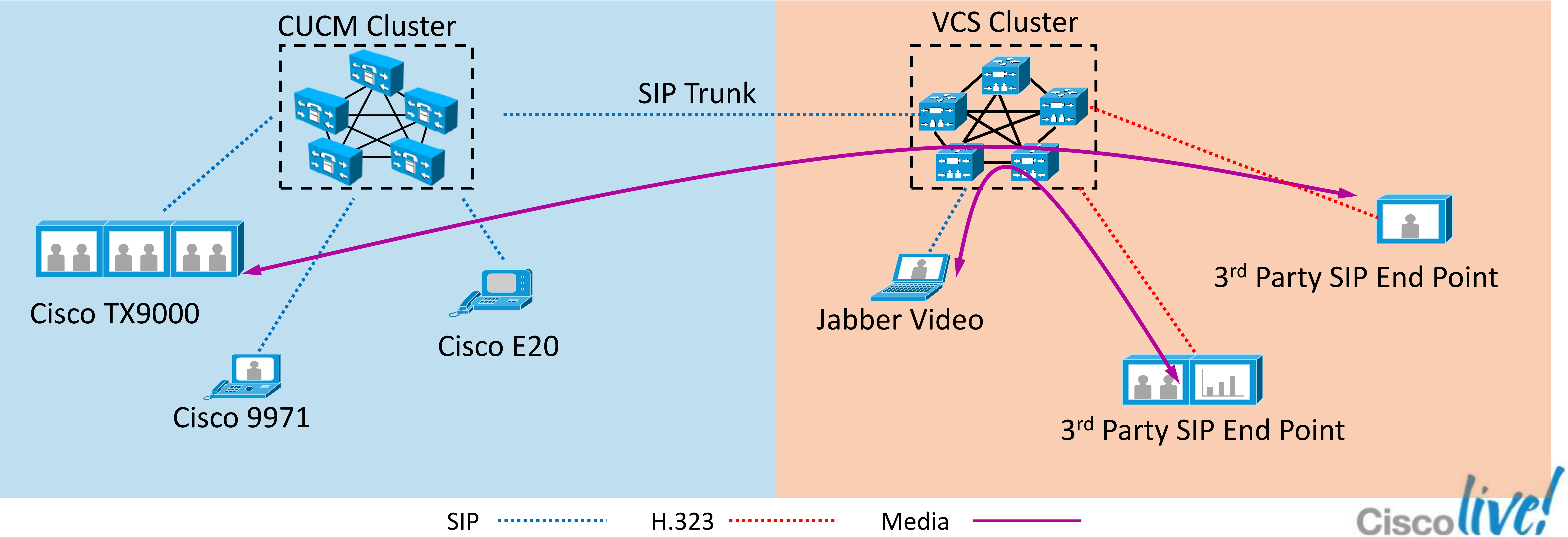
- No signalling protocol interworking required
- Media flows directly between terminating endpoints
- Referred to as a Non-traversal call



Design & Architecture Considerations

Media Flow – SIP to H.323 Interworking

- Requires VCS for signal and media interworking
- Media flows through VCS – geographic placement is critical
- Each VCS supports up to 100 interworked calls – referred to as a Traversal call



Fundamentals, Design & Architecture

Call Processing

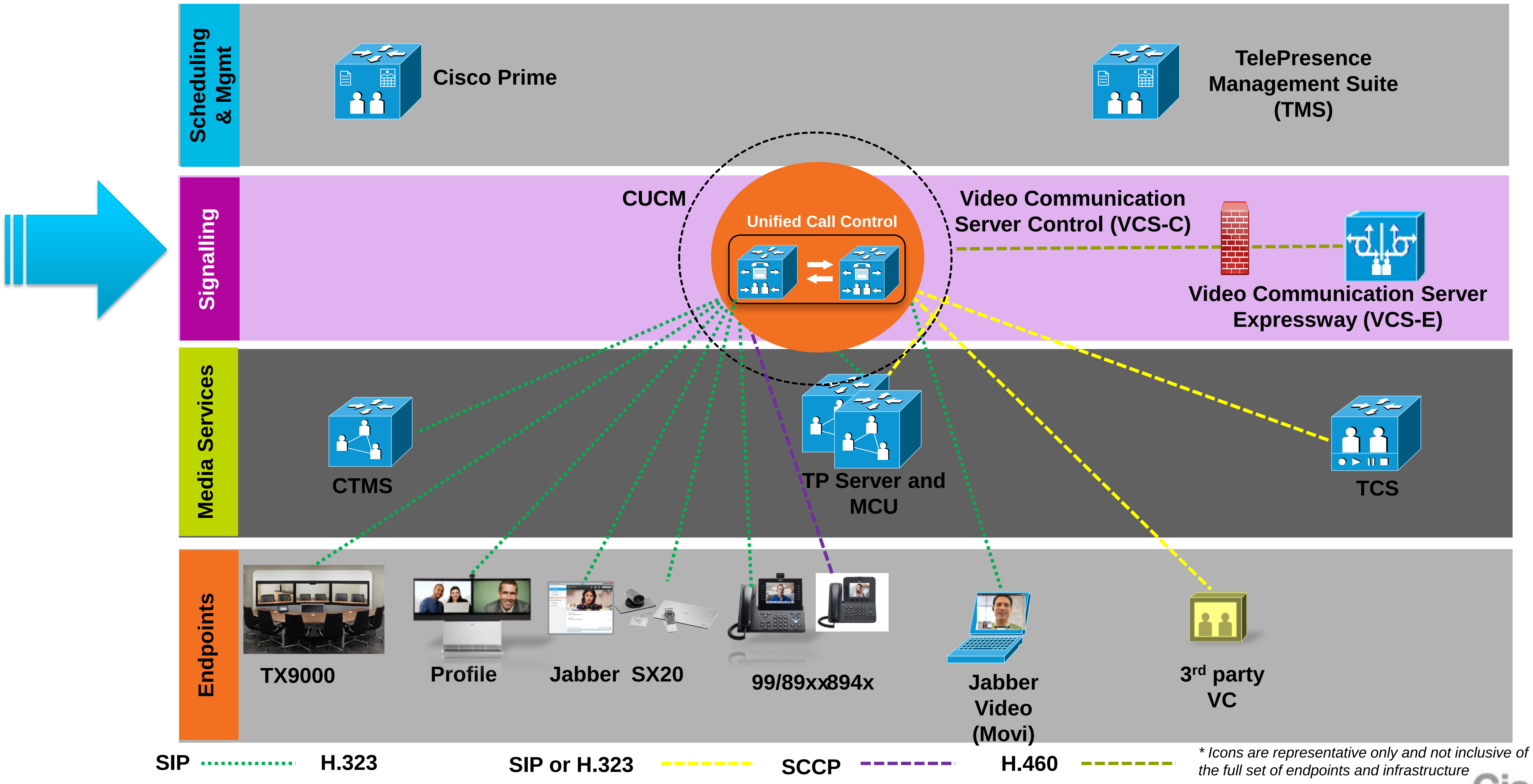
Dial Plan

Conferencing

Scheduling

Business to Business

Signalling and Call Control Layer



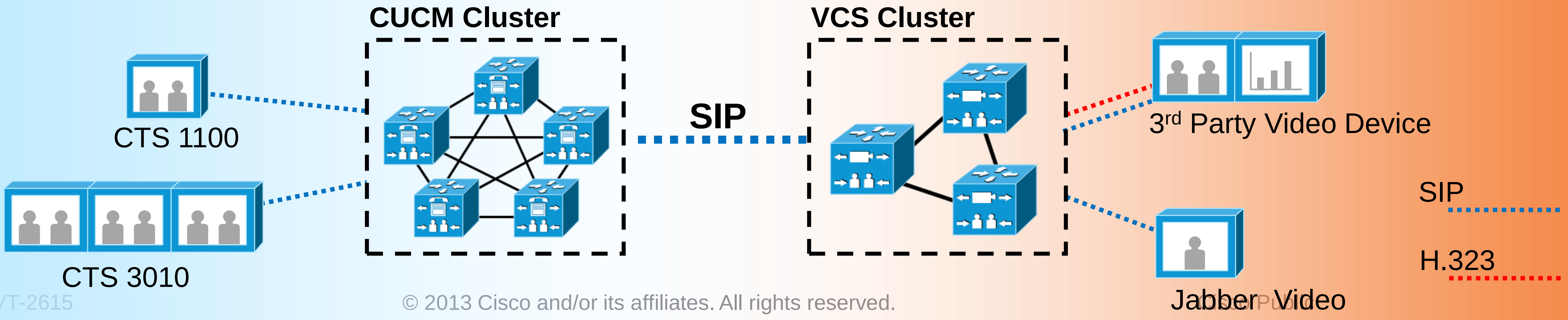
** Icons are representative only and not inclusive of the full set of endpoints and infrastructure*

Call Control

- Connecting CUCM and VCS Clusters

CUCM SIP Trunk connects to VCS Neighbour Zone

- SIP trunk connects CUCM call control with VCS call control
- H.323, SCCP, MGCP translated to SIP before being sent to VCS call control cluster
- Encryption supported
- BFCP Support



Call Processing

How to connect CUCM and VCS clusters

CUCM to VCS:

- Intercluster SIP trunks using DNS SRV
- Intercluster SIP trunks using IP addr

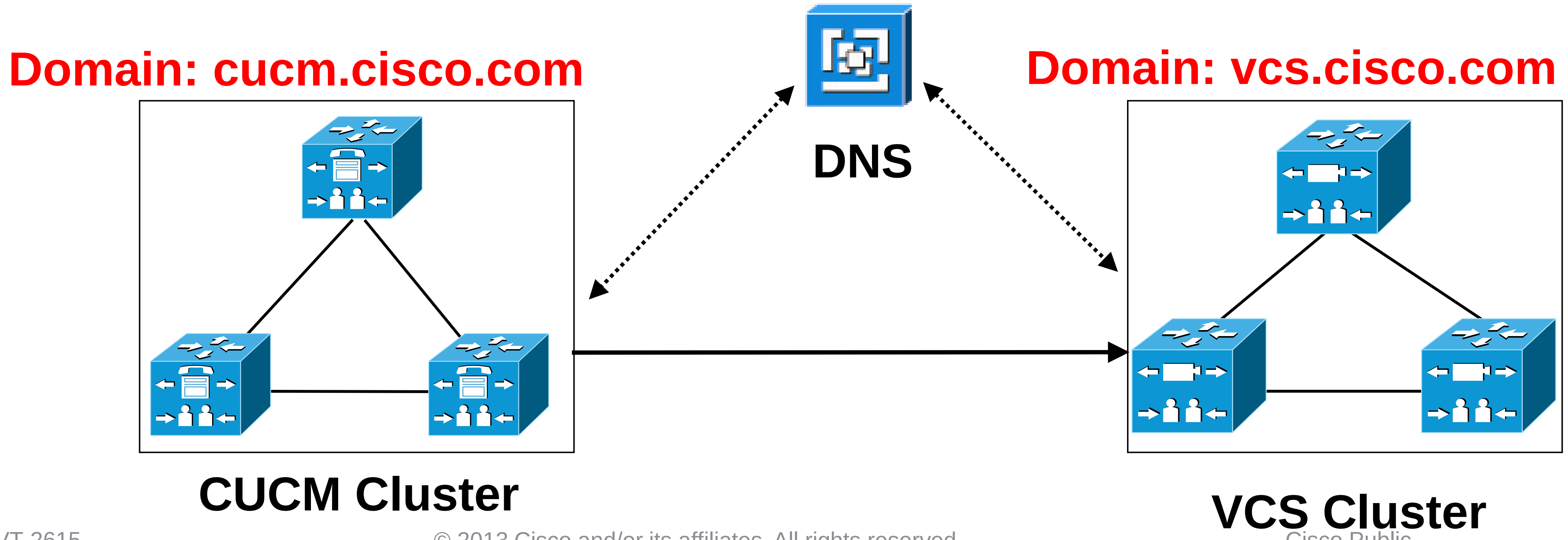
VCS to CUCM:

- Cluster trunking using DNS
- Cluster trunking using Neighbour zone

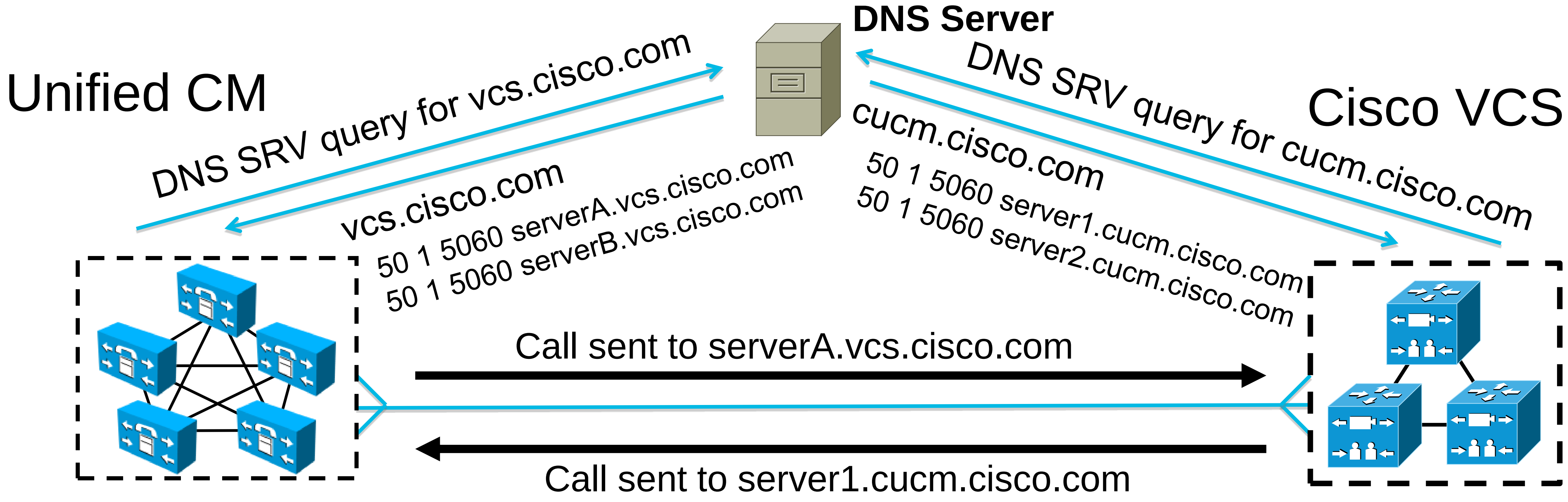
CUCM to VCS

Option 1: Cluster trunking using DNS SRV

- The VCS cluster needs to be addressable via a DNS SRV record
- Each peer should be set with an equal priority and weight in SRV
- Configure the SIP Trunk on the CUCM with following information:
  - Destination address: <Domain of VCS cluster> (defined as SRV)
  - Destination address is an SRV: Select this check box



SIP Trunk with DNS SRV



_sip._tcp.cucm.cisco.com	**IN**	**SRV**	**50**	**1**	**5060**	**server1.cucm.cisco.com**
	IN	**SRV**	**50**	**1**	**5060**	**server2.cucm.cisco.com**
_sip._tcp.vcs.cisco.com	**IN**	**SRV**	**50**	**1**	**5060**	**serverA.vcs.cisco.com**
	IN	**SRV**	**50**	**1**	**5060**	**serverB.vcs.cisco.com**
server1.cucm.cisco.com	**IN**	**A**	**10.10.10.1**			
server2.cucm.cisco.com	**IN**	**A**	**10.10.10.2**			
serverA.vcs.cisco.com	**IN**	**A**	**10.10.20.1**			
serverB.vcs.cisco.com	**IN**	**A**	**10.10.20.2**			

Weight

Priority

- SIPS or SIP service
- TCP or UDP protocol
- DNS SRV records provide load balancing and redundancy
- DNS server needs to be highly available
- Options ping for reachability

DNS SRV

Format SRV records for SIP and H.323 (RFC 2782)

For Your Reference

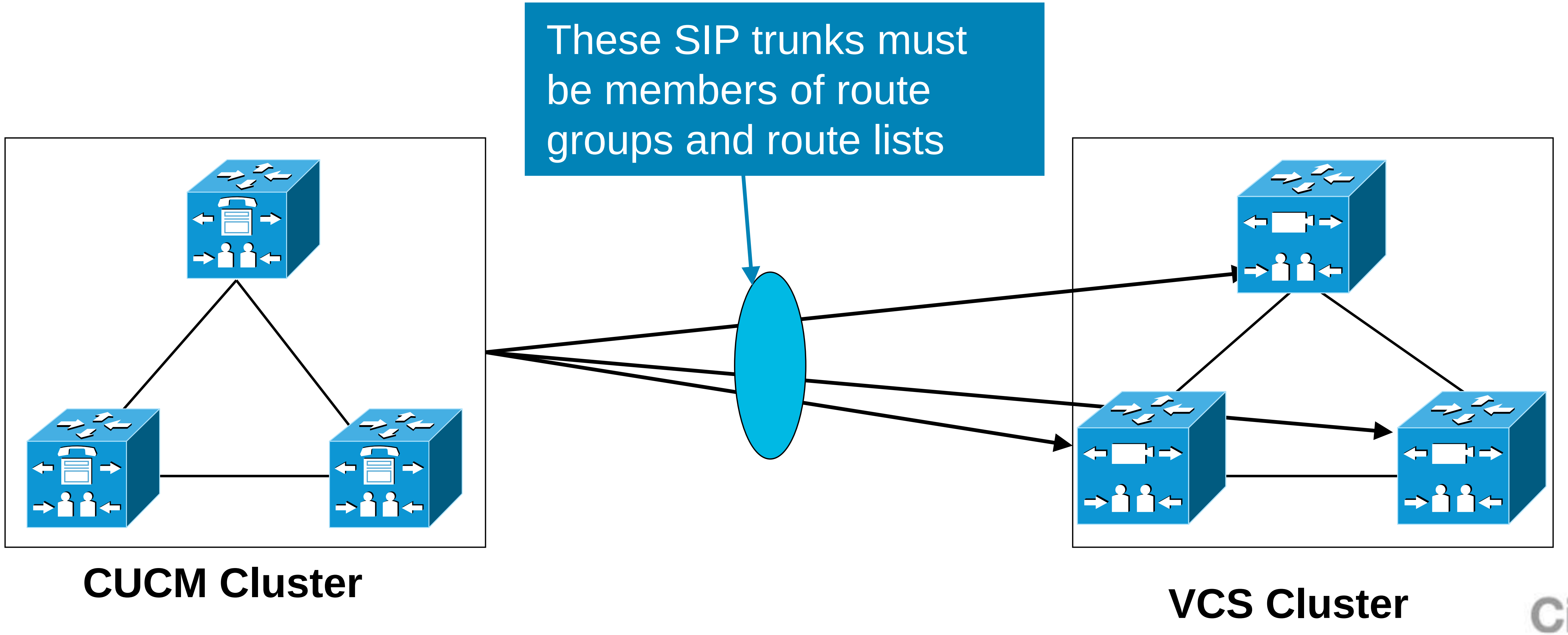
_sip._tcp.example.com 86400 IN SRV 10 60 5060 vcs.cisco.com

- **_sip** — Name of the **Service**
  - sip
  - sips
  - h323
- **_tcp.example.com** — **Protocol** and domain name (TCP, UDP...)
- **86400** — **DNS Time-To-Live**: how much time the server caches the record before it flushes the cache
- **IN** — **DNS Class.** Always "IN"
- **10** — **Priority:** Lowest priority means "preferred". If connection fails, client fallback to the higher priority record
- **60** — **Weight:** for records with same Priority, it is used for load-balancing
- **5060** — **Port:** TCP or UDP port for the service
- **vcs.cisco.com** — **Targed:** hostname or IP Address for the host Providing the service

CUCM to VCS

Option 2: Cluster trunking using IP addresses

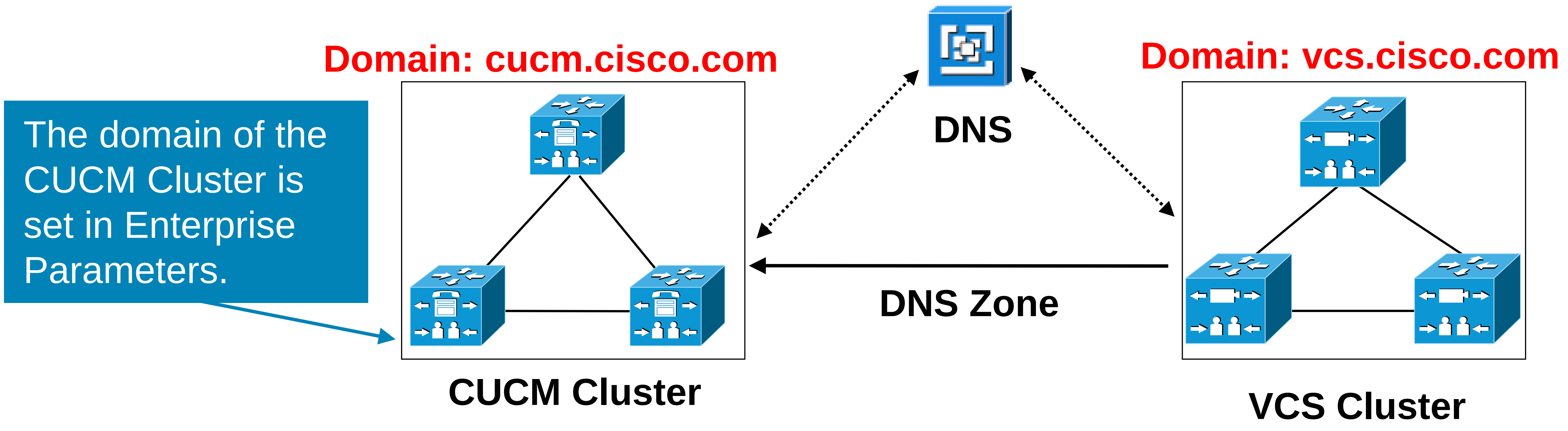
- Can also set up multiple SIP trunks to each peer in the VCS cluster.
- Configure the SIP Trunks on the CUCM with following information:
  – Destination address: <IP address of VCS> or <DNS address of VCS>



VCS to CUCM

Option 1: Cluster trunking using DNS Zone

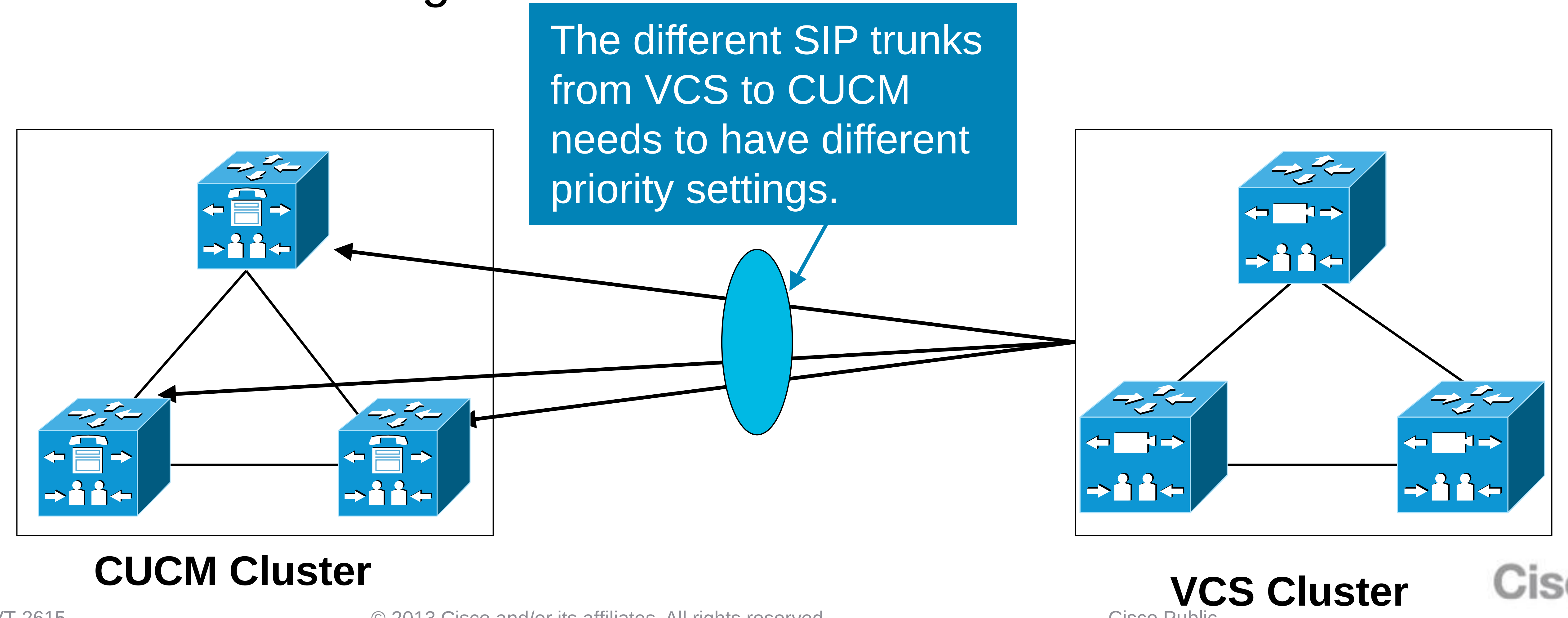
- Create a SIP DNS Zone from the VCS to the CUCM cluster.
- For routing of calls DNS and SRV lookups are utilised.
- The VCS and the CUCM can not be part of the same subdomain
  - Need to have different SRV records



VCS to CUCM

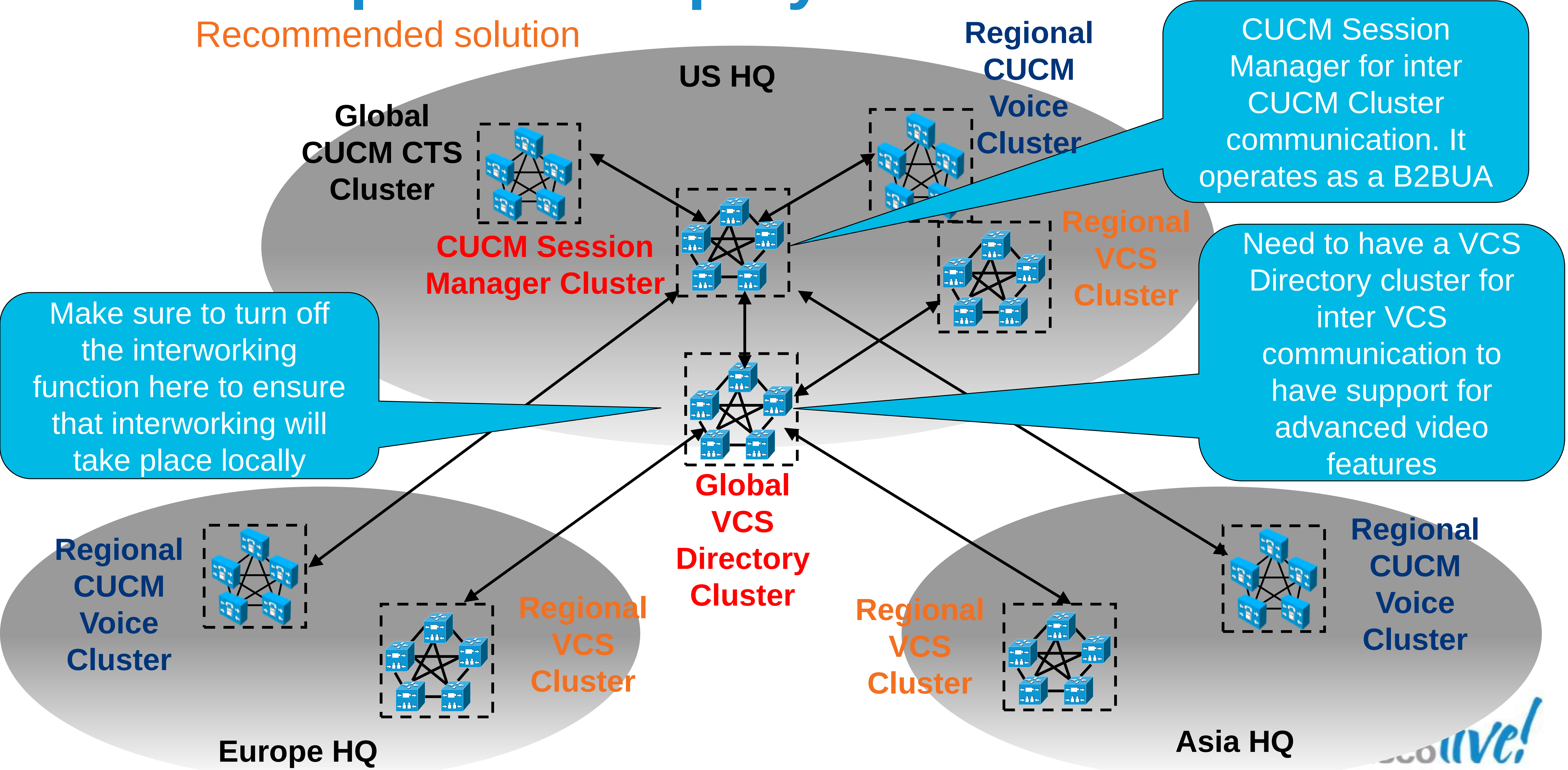
Option 2: Cluster trunking user Neighbouring Zone

- Create a SIP trunk from the VCS to each CUCM in the CUCM cluster.
- Use search rules to query only one neighbour in a priority order
  - To avoid forking to all CUCMs



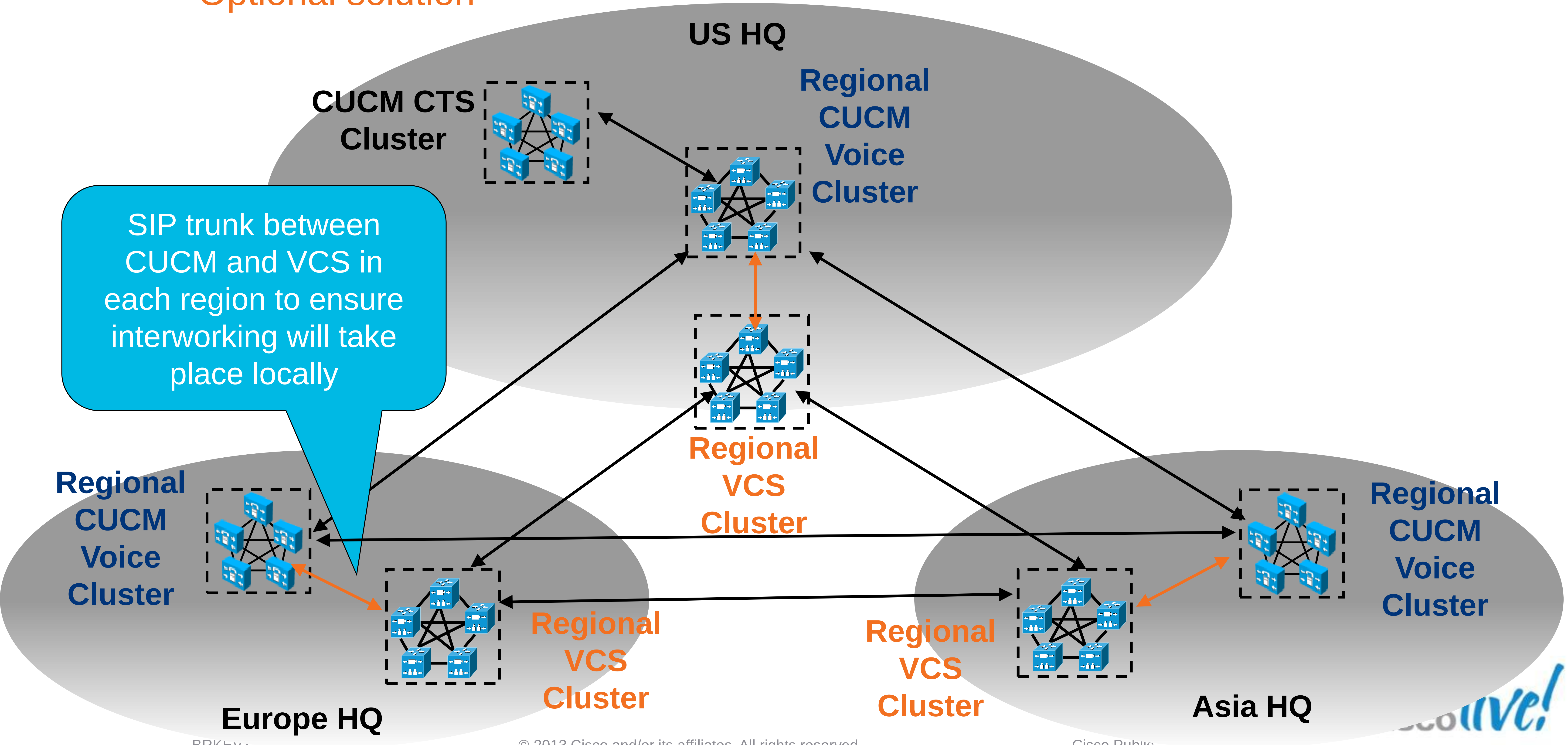
Multiple Site Deployment

Recommended solution



Multiple Site Deployment

Optional solution



Fundamentals, Design & Architecture

Call Processing

Dial Plan

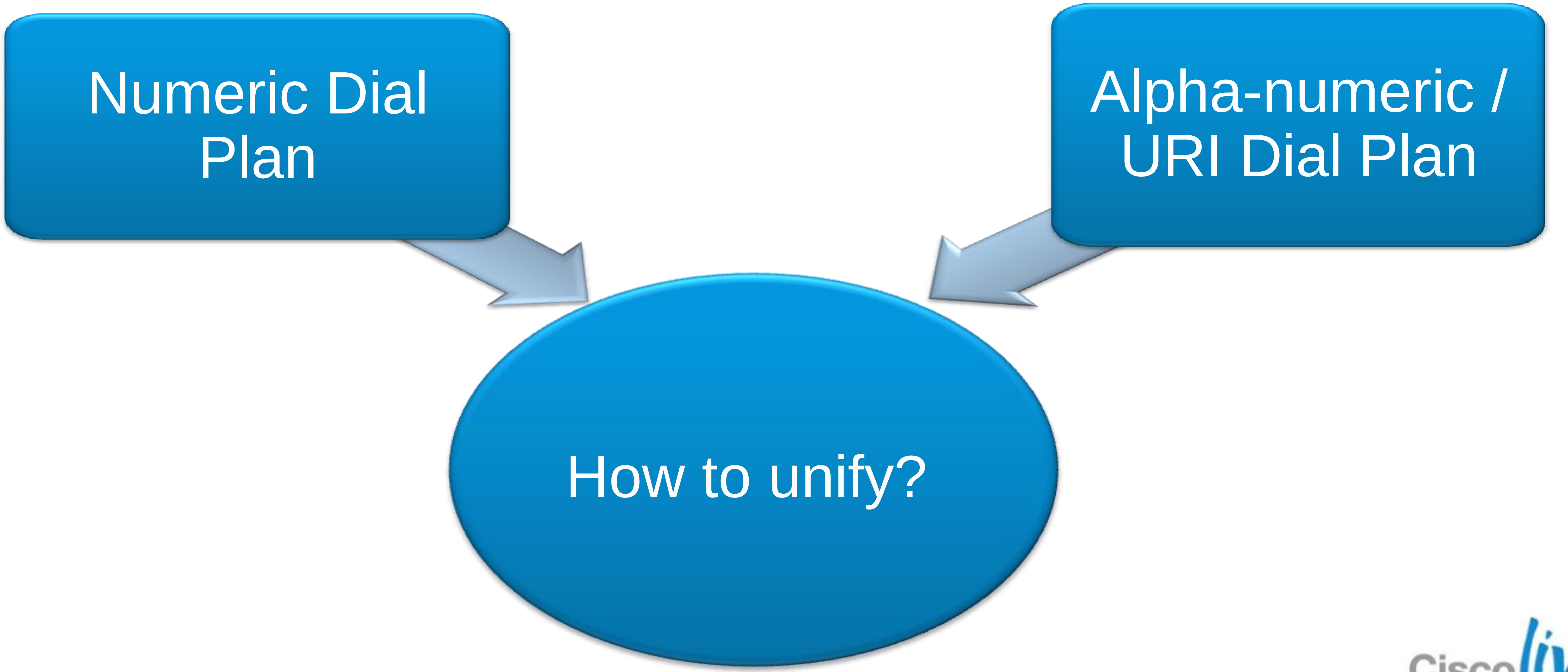
Conferencing

Scheduling

Business to Business

Dial Plan

Two major dial plan types



Call Control

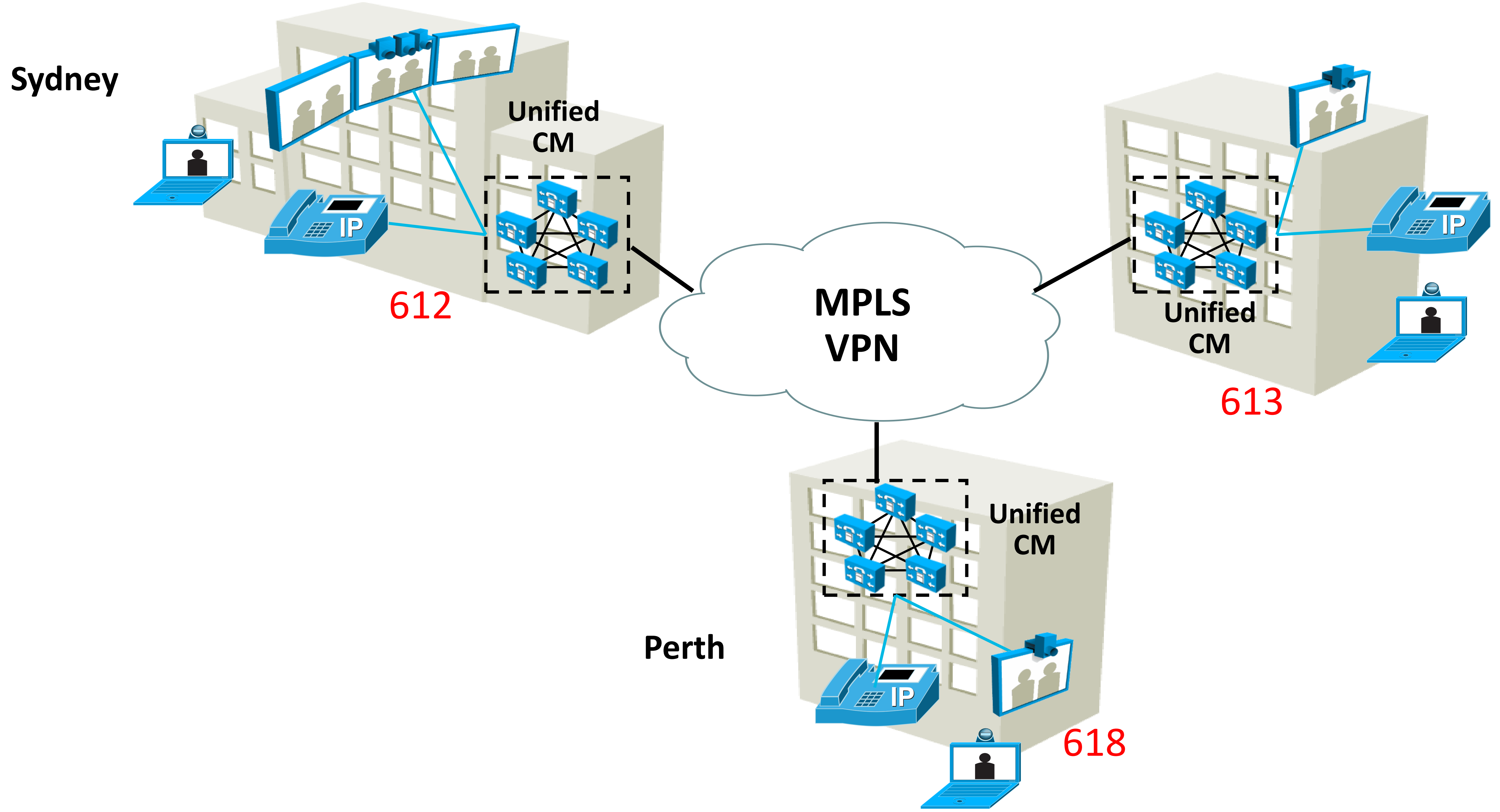
- Dial Plan – E.164 and URI’s
  - Both are relevant
    - E.164 addresses allow easy integration with PSTN and audio-only endpoints
    - URI addresses allow easier B2B communications by using domain names and are generally more intuitive for end users to operate
  - Past: Users typically had an audio device and a video device
  - Trending/Future: Users will have a single device that does both
  - As video becomes pervasive, devices will need to support both address schemes.

Call Control

Dial Plan – E.164 and URI's

Address Scheme	Example	Cisco Unified CM Registration	VCS Registration
E.164	61892145400	Supported as Directory Number (DN)	H.323 E.164 Registration
E.164 Based URI	61892145400@cisco.com	Supported – 9.X / SIP URI	Supported H.323id / SIP URI
Alphanumerical URI	john.doe@cisco.com	Supported – 9.X / SIP URI	Supported H.323id / SIP URI

Typical E.164 Site Code Assignment



Overall Numeric Dialling Strategy

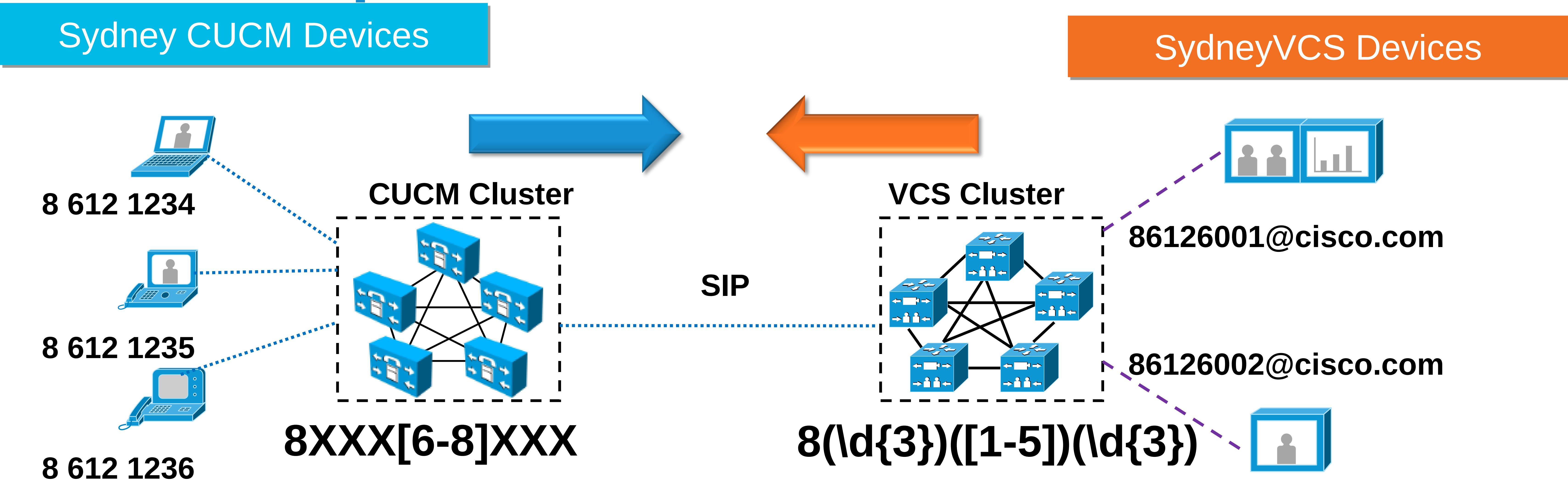
Scalable over Large Amount of Sites and Extensions

8 +

Site	Site Code	Cluster	Extension
Sydney	612	CUCM	[1-5]XXX
		VCS	[6-8]XXX
Melbourne	613	CUCM	[1-5]XXX
		VCS	[6-8]XXX
Perth	618	CUCM	[1-5]XXX
		VCS	[6-8]XXX

- Inter-site calls use an escape code (8)
- Abbreviated dialing within a site (four digits site code).
- Overlapping extensions allowed at different sites.

Escape Codes between CUCM and VCS



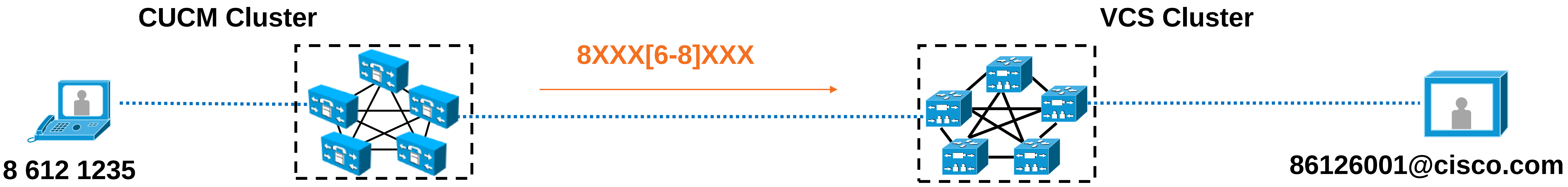
CUCM Route Pattern:

Called # with 8
+ site code
+ [6-8] as the starting digit in the 4 digit ext
= routed to the VCS

VCS Search Rule:

Called # with 8
+ site code
+ [1-5] as the starting digit in the 4 digit short code
= routed to the CUCM

Calling from CUCM to VCS



Dials 86126001

1. The CUCM routes 8XXX[6-8]XXX to the VCS
2. The CUCM uses DNS to locate the destination address of the VCS Cluster
3. Called URI = 86126001@cisco.com, the CUCM adds the OTLD (@cisco.com)

1. VCS to transform called URI to 86126001@cisco.com
2. The VCS will find a local match and send the call to the VC endpoint.

Calling from VCS to CUCM

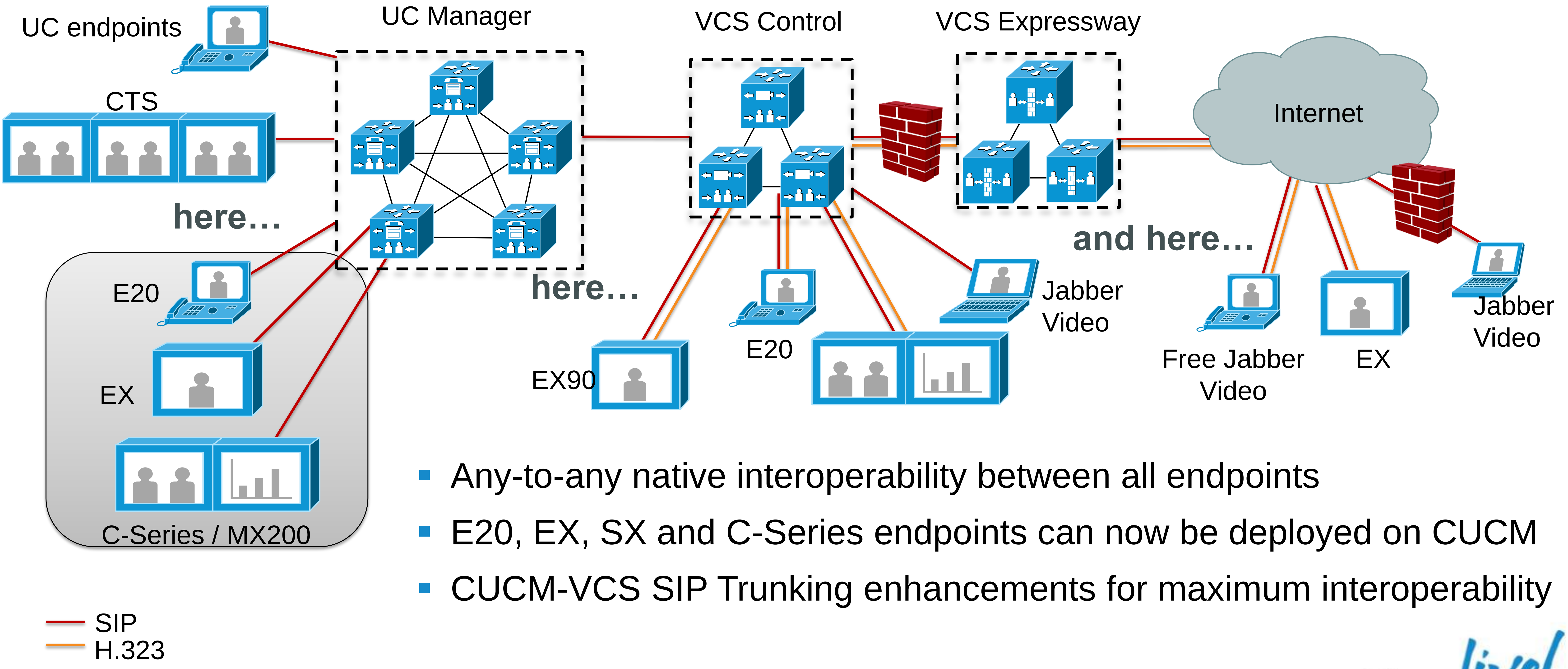
Dials 86121235@cisco.com

1. No local match for this number on VCS.
2. Match found on the DNS Zone to the CUCM Cluster.
3. The VCS uses DNS to locate the destination address of the CUCM Cluster

1. The call will arrive at CUCM as 86121235@cisco.com
2. The CUCM will find a local match for 86121235
3. Call routed to the destination endpoint.

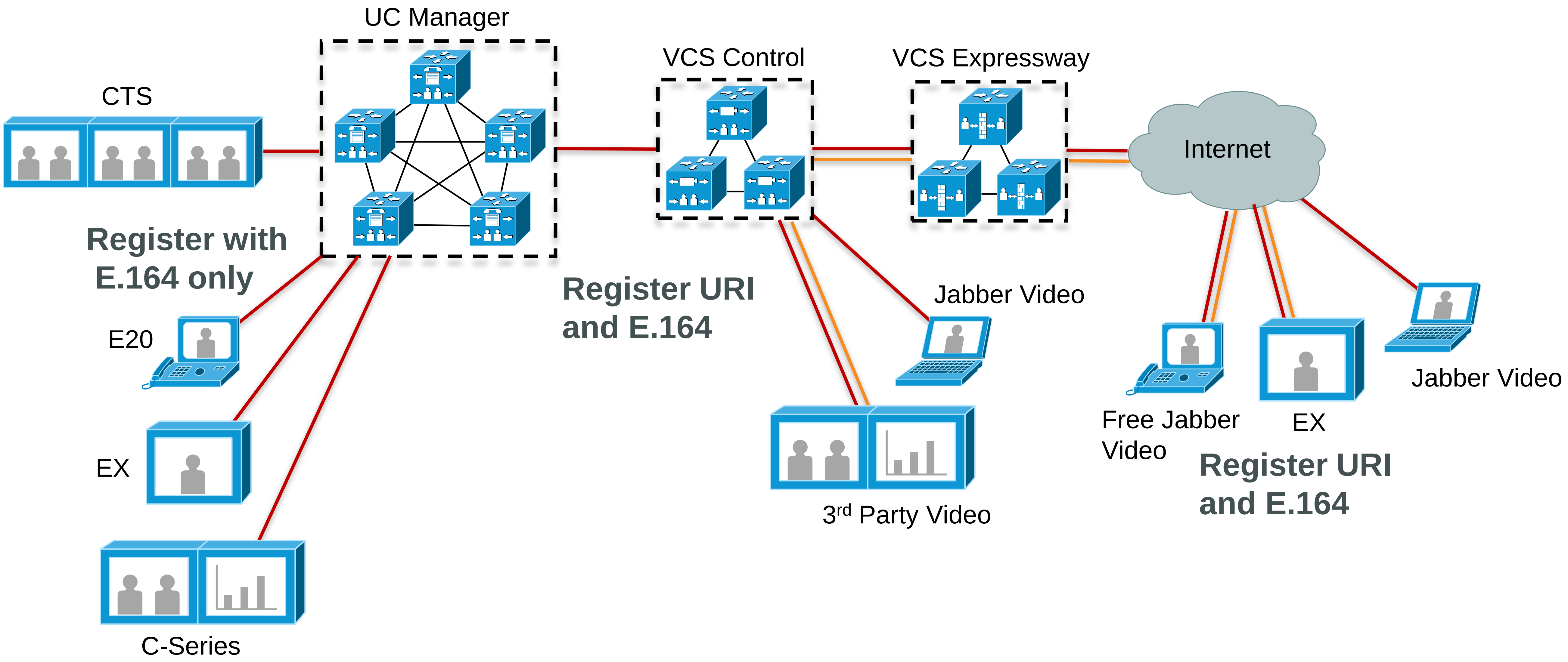
Creating a Unified Call Platform

Endpoints Can Now Be Deployed…



- Any-to-any native interoperability between all endpoints
- E20, EX, SX and C-Series endpoints can now be deployed on CUCM
- CUCM-VCS SIP Trunking enhancements for maximum interoperability

Alpha URI dialing between CUCM and VCS

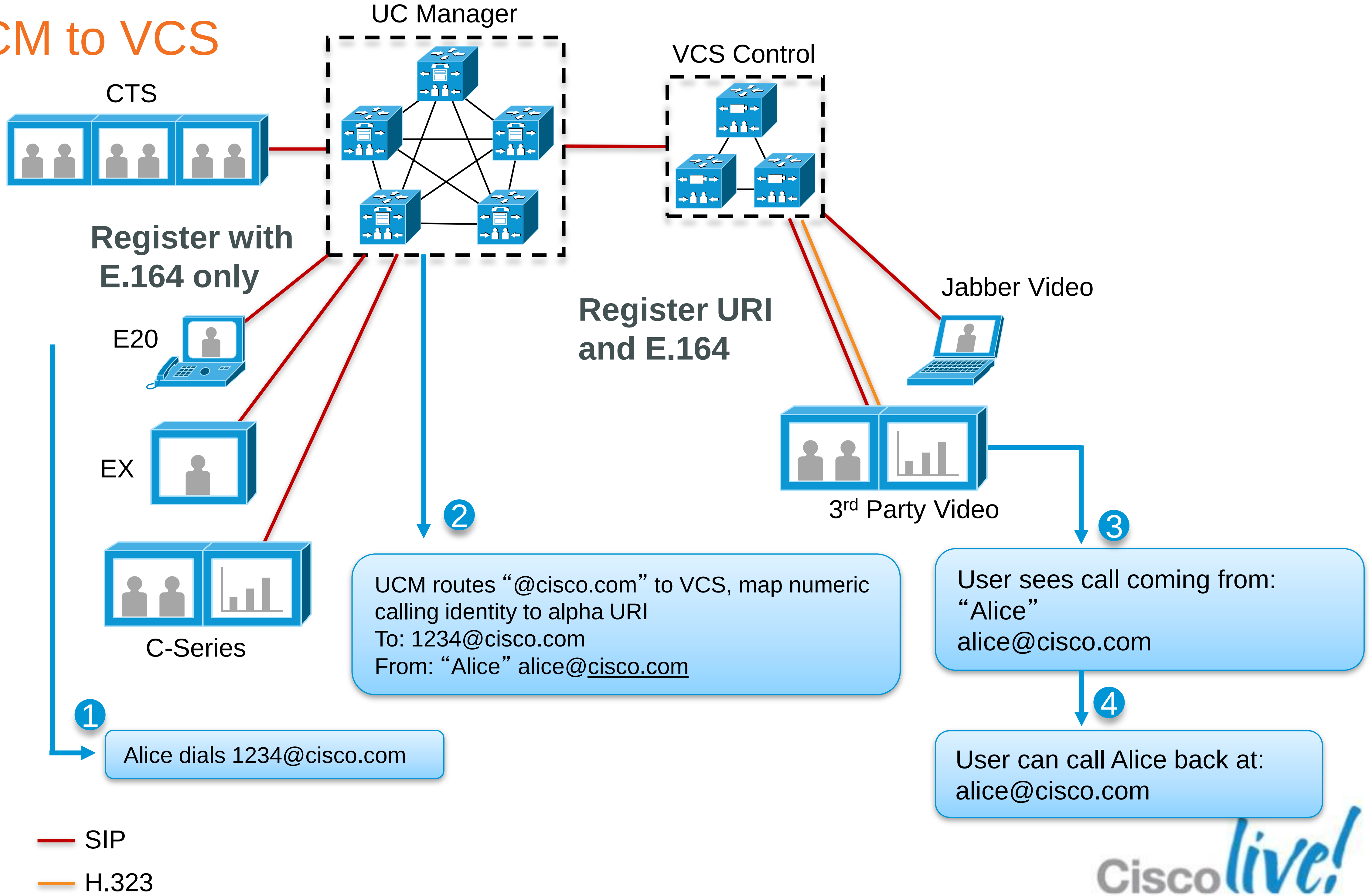


SIP
H.323

Alpha URI Dialing between CUCM and VCS

Numeric Dialing from CUCM to VCS

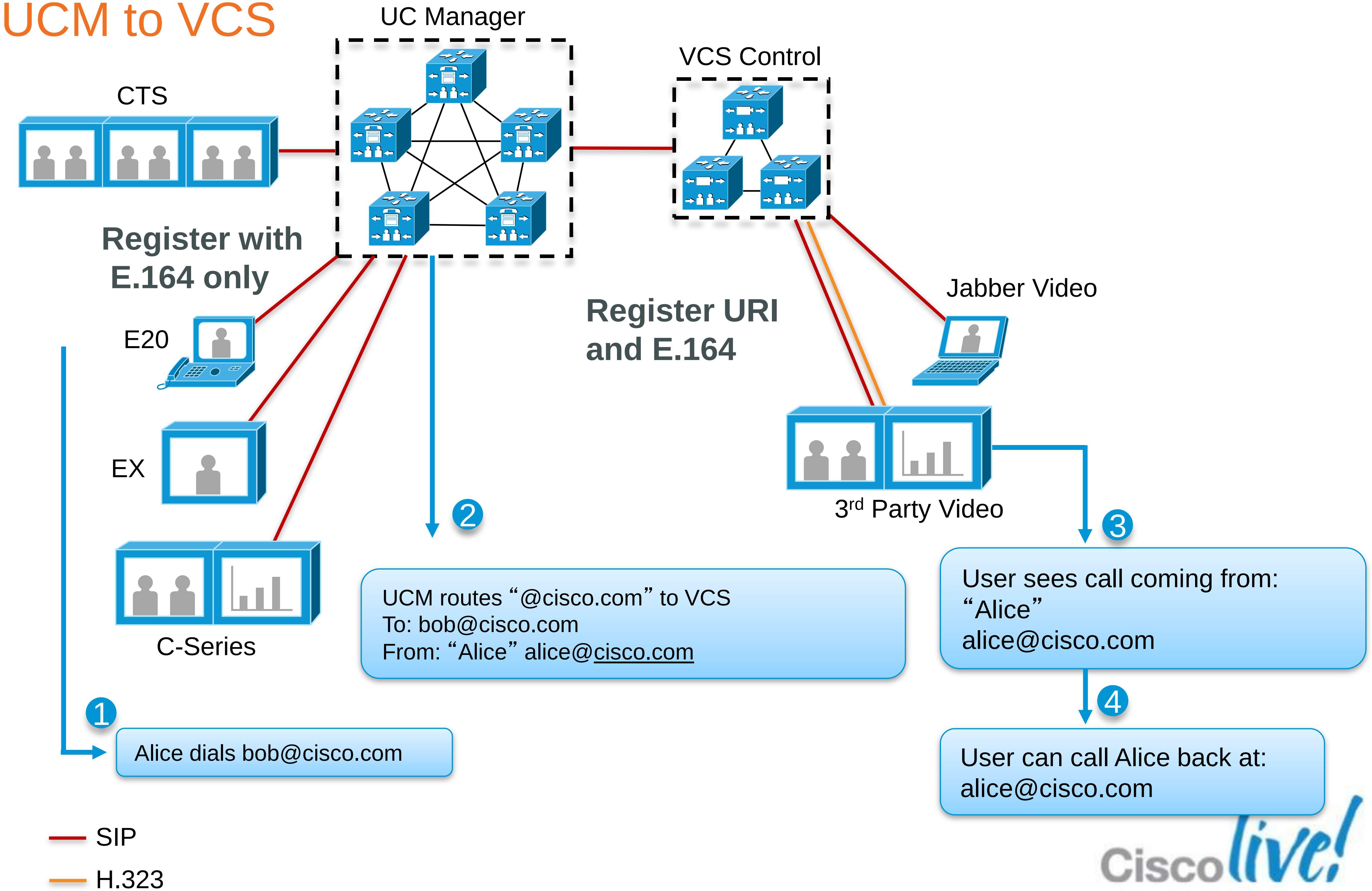
- Endpoint on CUCM can call alpha URI or numeric
- Enterprise Directory gives numeric destination
- CUCM will blend calling identity (make sure to enable alpha URI calling identity on trunk to VCS)
- numeric dialing to VCS via appropriate dial plan



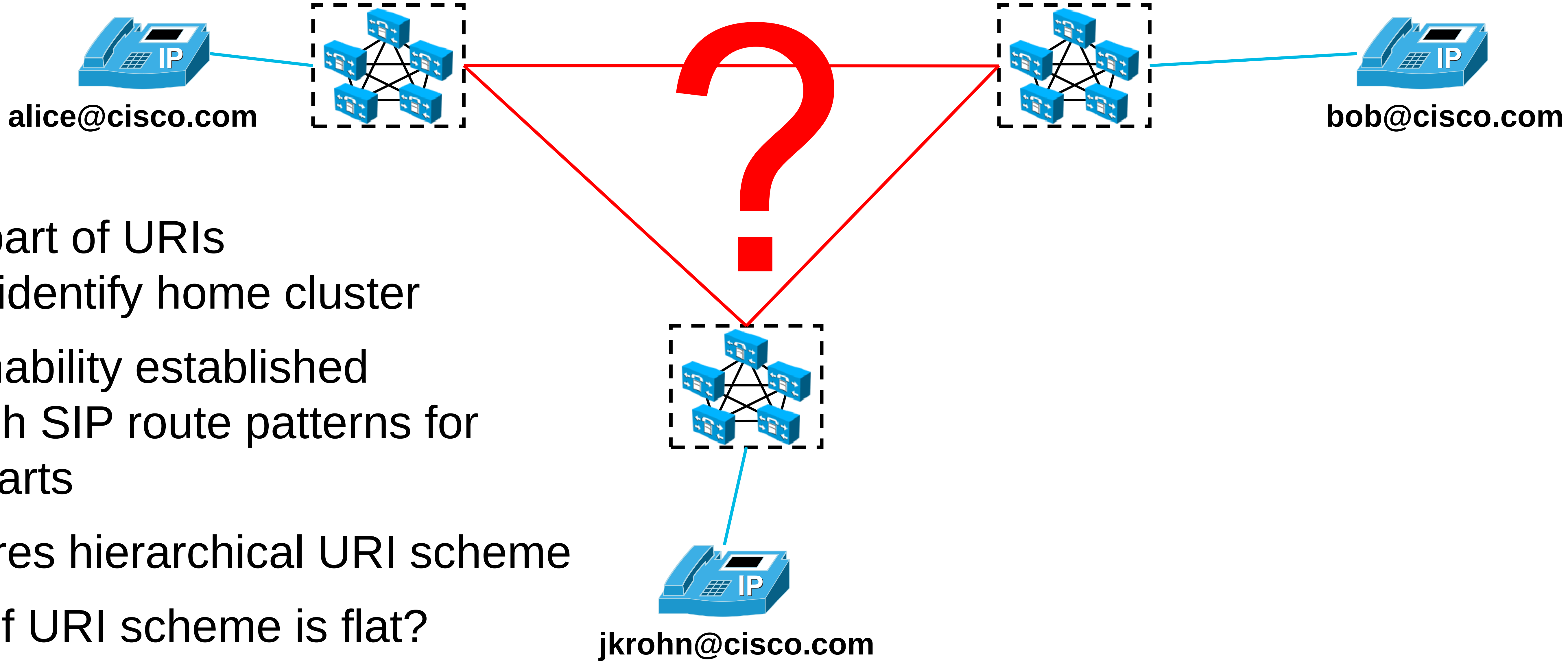
Alpha URI Dialing between CUCM and VCS

Alpha URI Dialing from CUCM to VCS

- alpha URI routing to VCS using “Default” routing via SIP route pattern “cisco.com”
- Routes all non-local cisco.com alpha URIs to VCS
- make sure to implement loop avoidance (don’t route calls coming in from trunk to trunk)



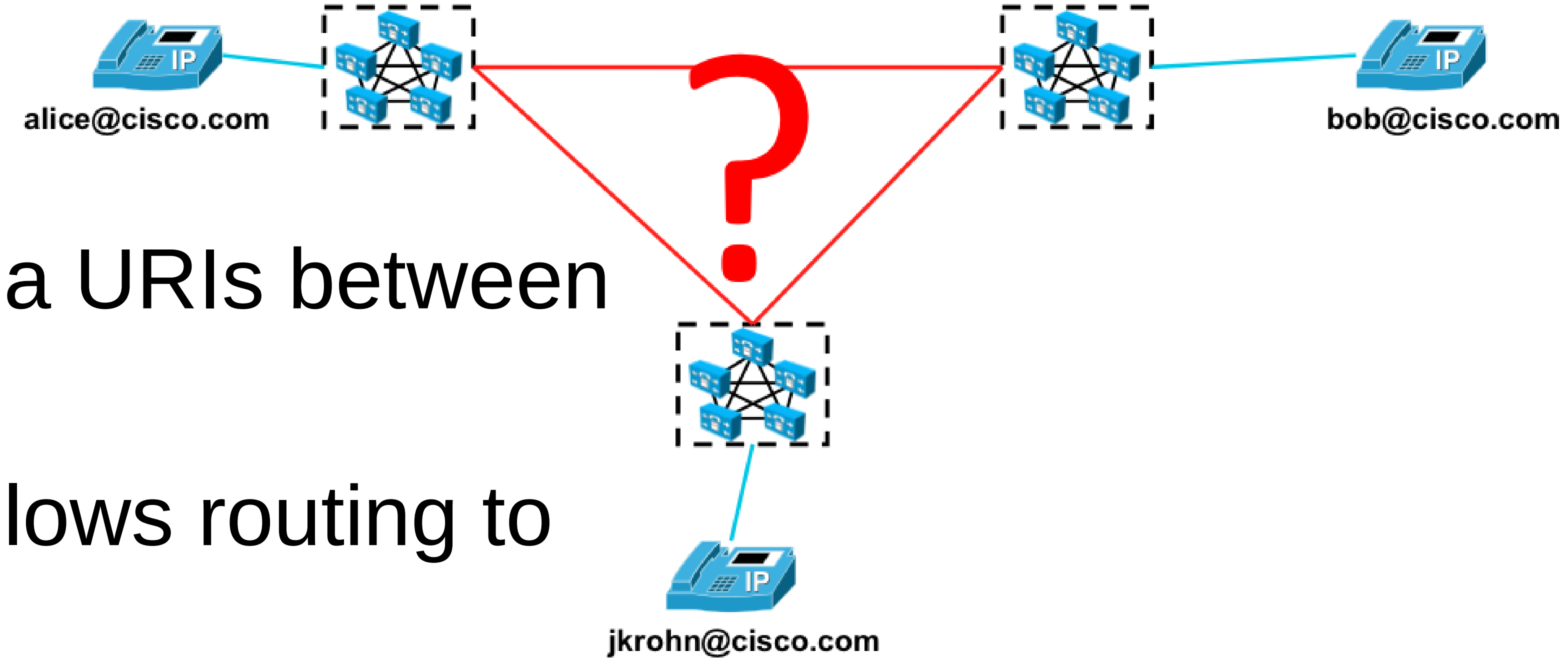
Multicluster URI routing



- Host part of URIs might identify home cluster
- Reachability established through SIP route patterns for host parts
- Requires hierarchical URI scheme
- What if URI scheme is flat?

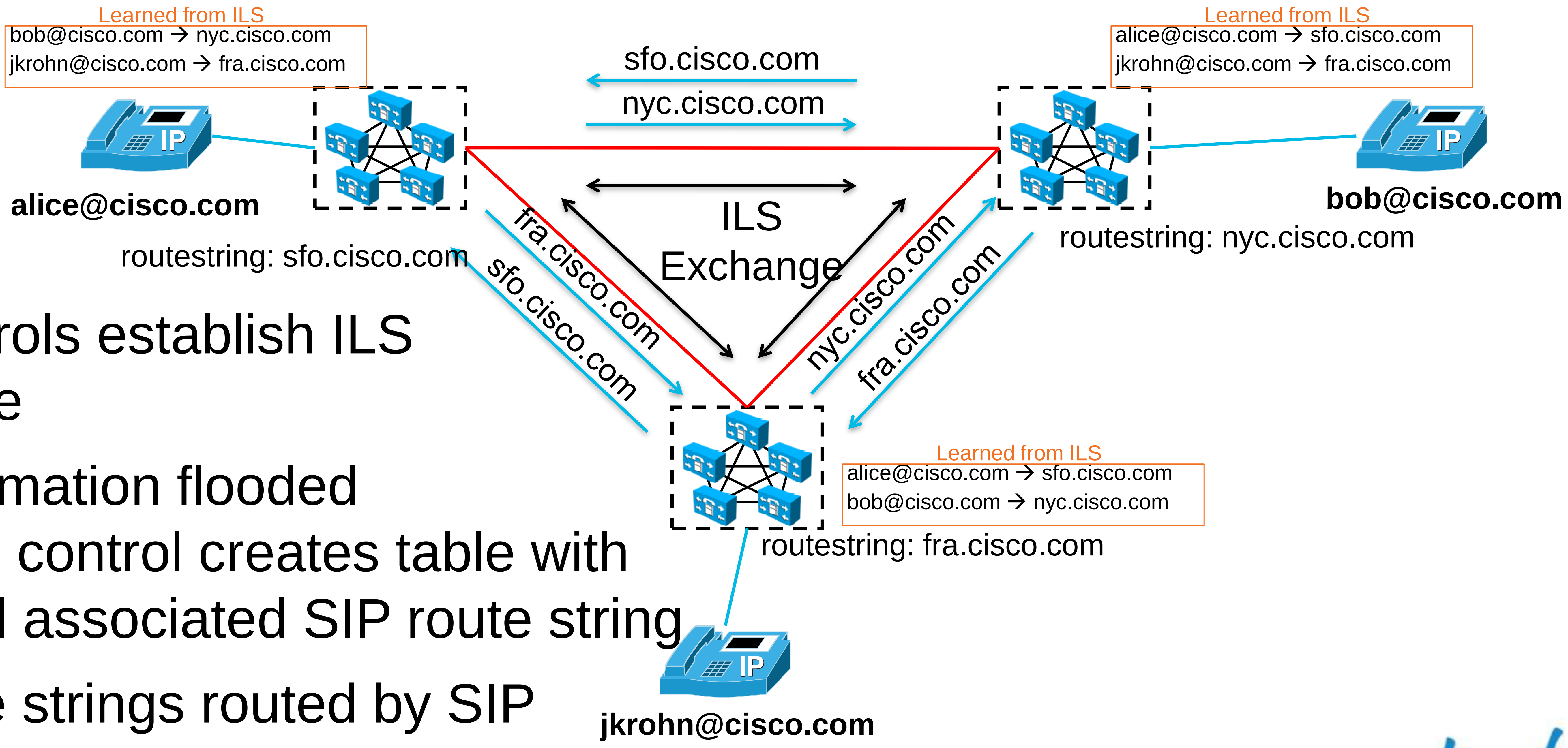
Intercluster Lookup Service (ILS)

Fundamental idea



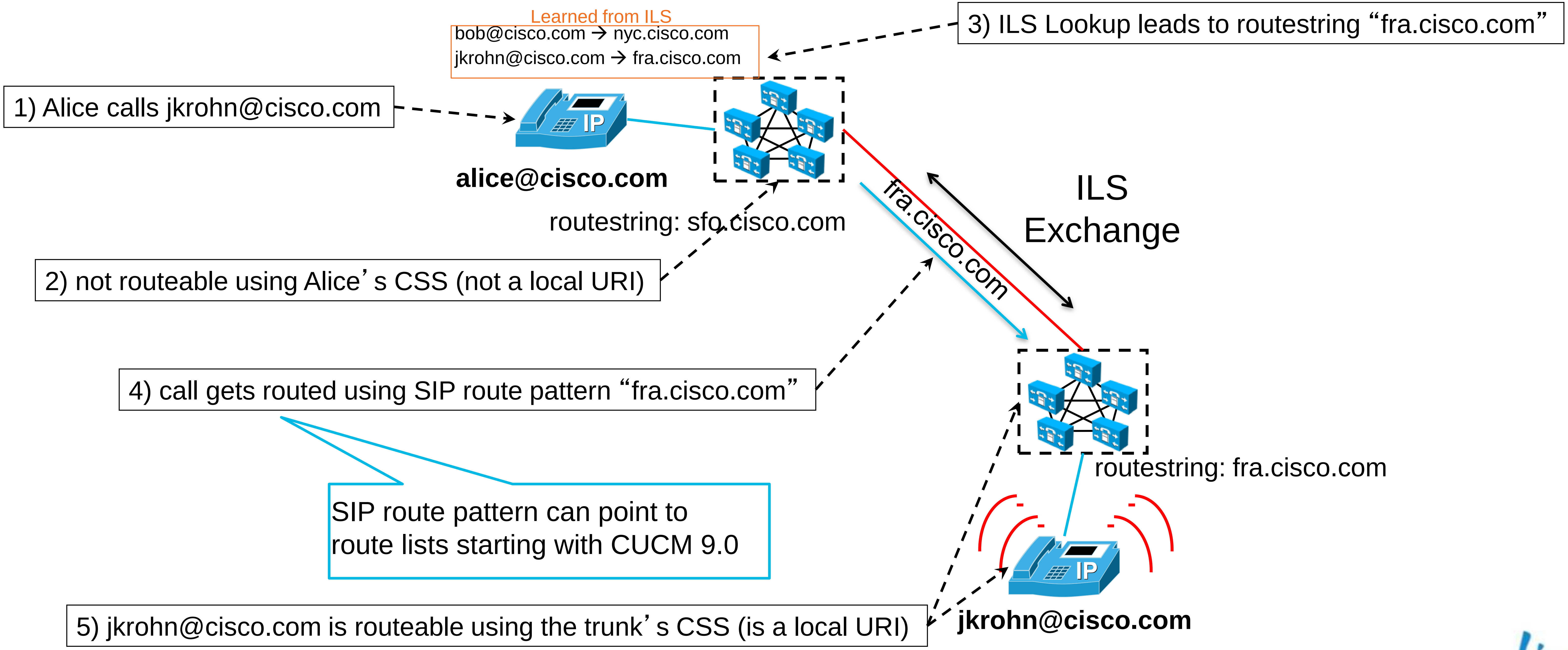
- Need mechanism that
  - allows propagation of individual alpha URIs between call controls
  - binds alpha URI with attribute that allows routing to URI's home cluster
- ILS
  - each call control replicates it's alpha URIs to it's neighbours
  - each call control also announces "SIP route string" together with the alpha URIs
  - "SIP route string" can be routed based on SIP route patterns → intercluster routing of alpha URIs not based on URIs' host part, but on SIP route string

ILS Learning

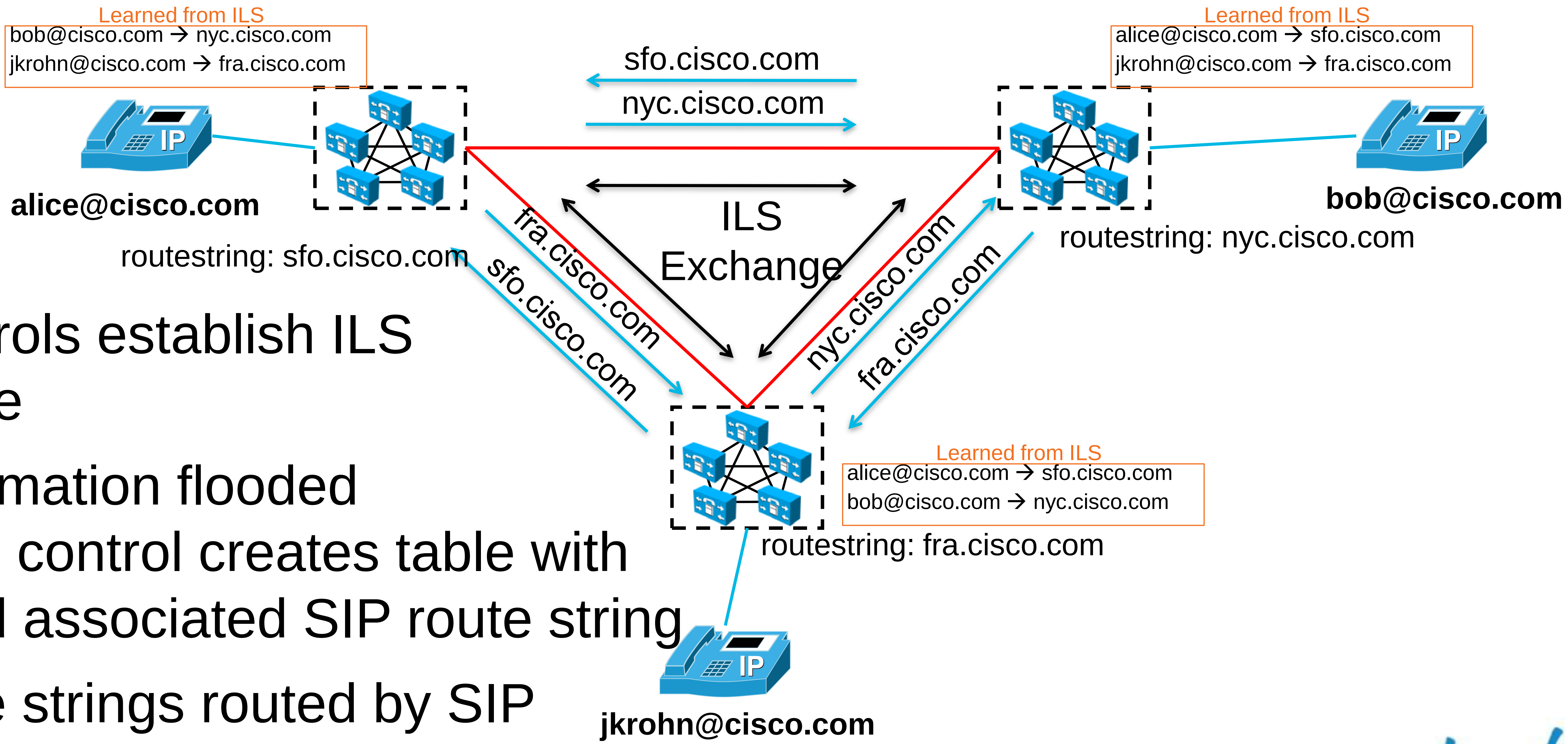


- Call controls establish ILS Exchange
- URI information flooded
  Each call control creates table with URIs and associated SIP route string
- SIP route strings routed by SIP route patterns

Routing Alpha URI Using ILS Information



ILS Learning



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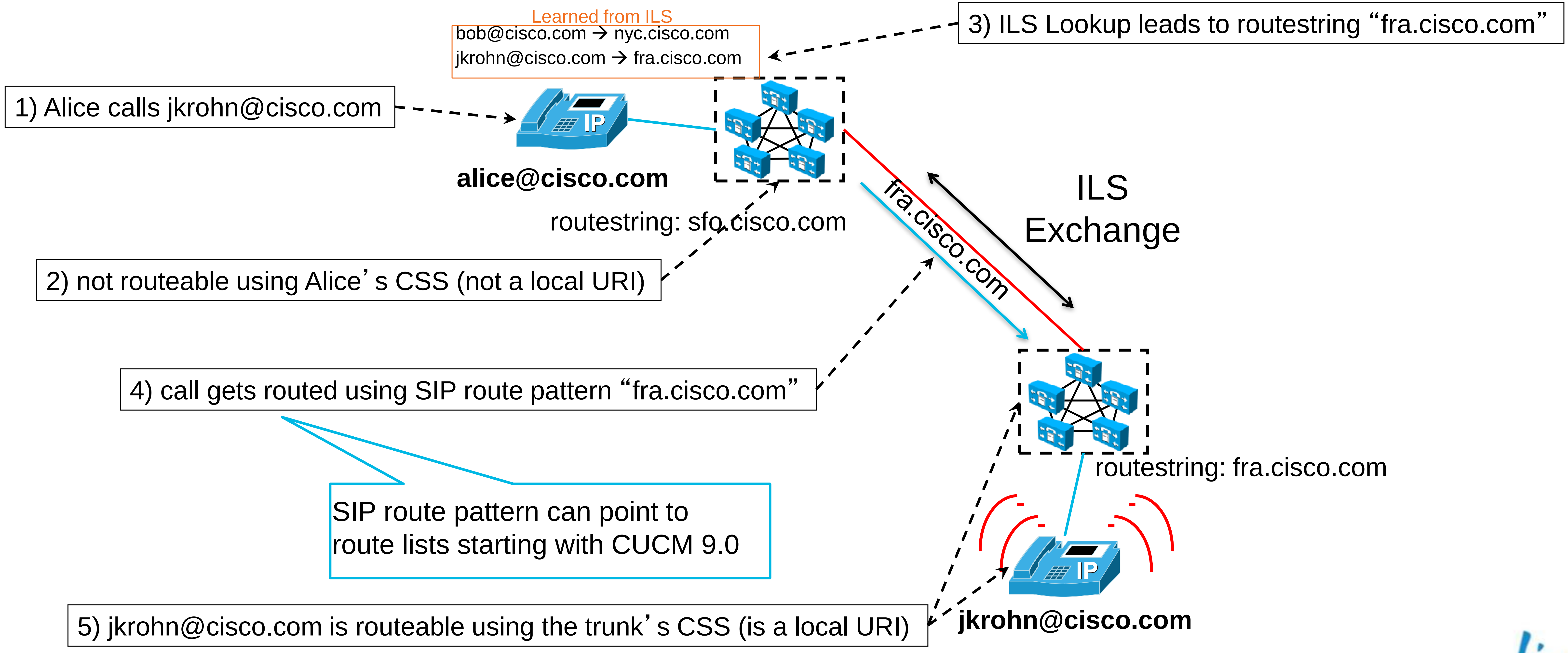
Routing Alpha URI Using ILS Information



Fundamentals, Design & Architecture

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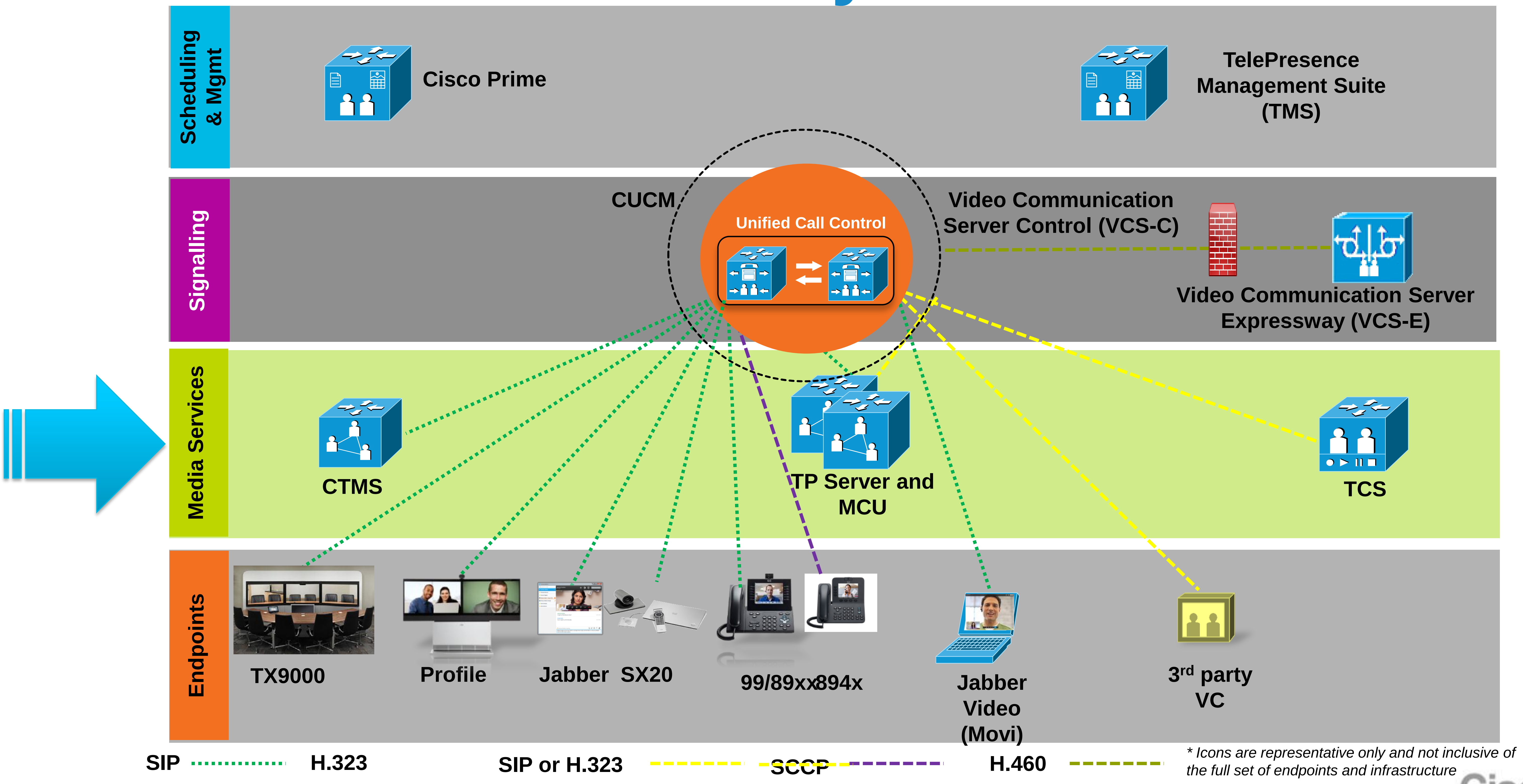
Media Services Layer



Fundamentals

H.323 MGCP SIP SCCP ISDN TIP

▪ SIP and TIP – Whats the difference?

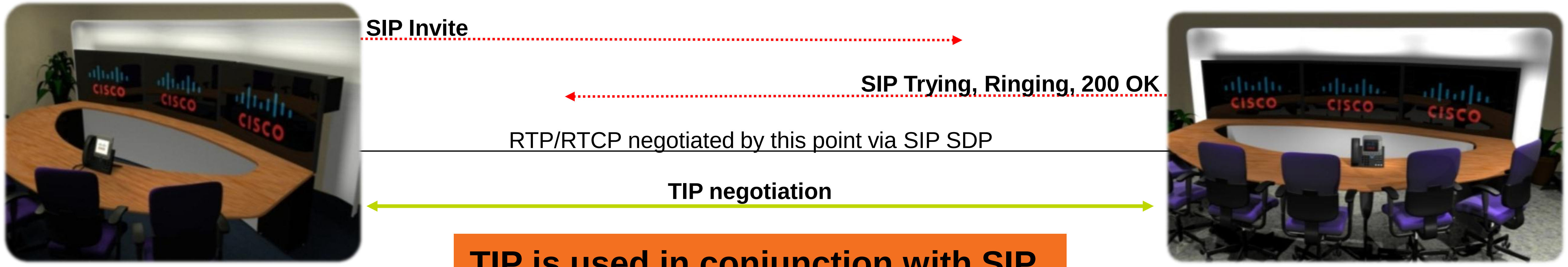
– SIP

- Responsible for negotiating the RTP and RTCP IP addresses and ports

- RTP/RTCP channels are used not just for media, but also for the TIP signalling messages

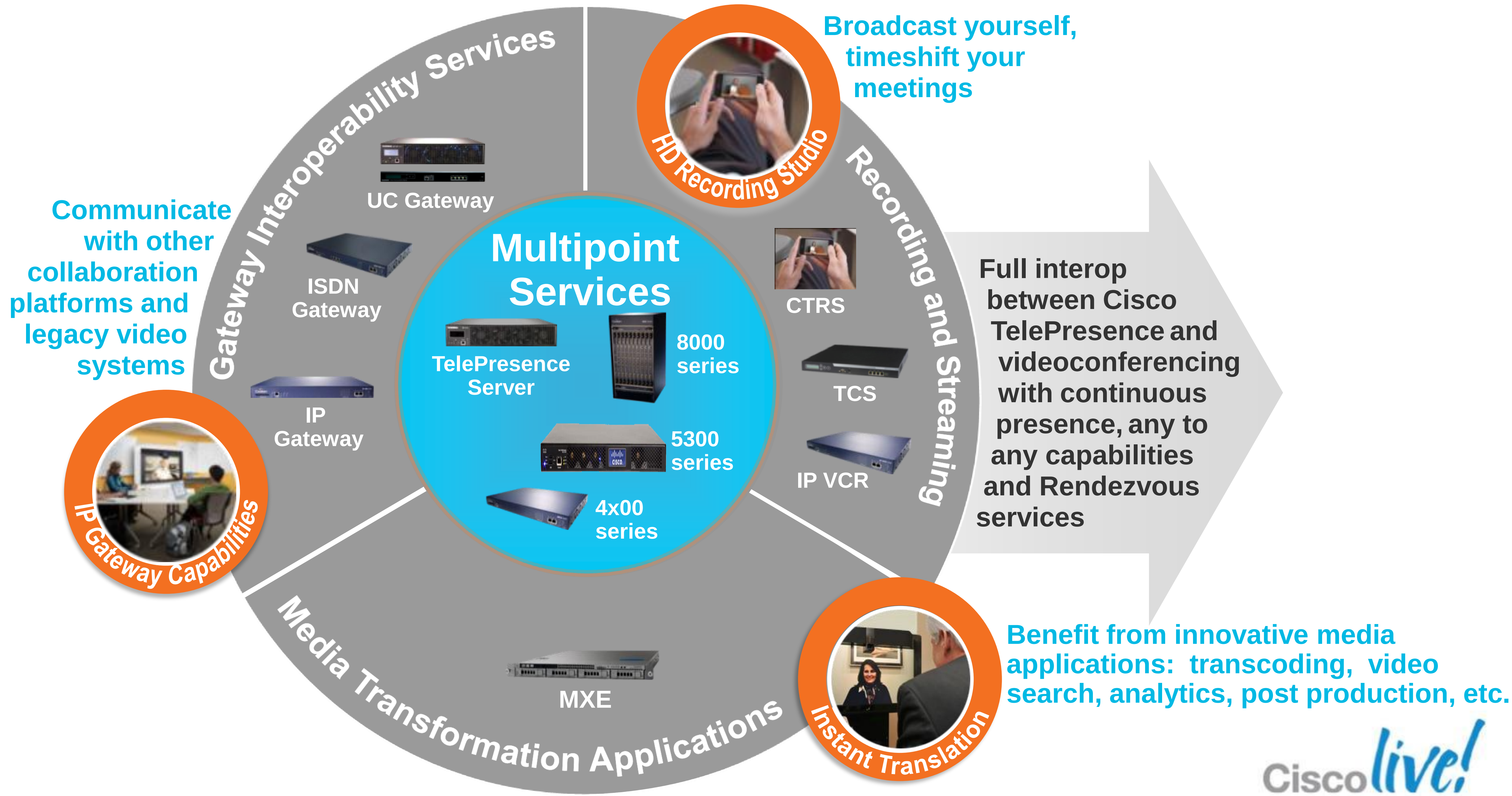
TIP

- Developed to overcome challenges with Multiscreen Multichannel TP systems
- Evolved from CTX-MUX protocol
- IMTC have standardised TIP
- Relies on an initial call negotiation using SIP
- TIP will take over after SIP call setup
- Re-negotiates
  - •number of video and audio streams
  - •multiplexing of multiple media streams



TIP is used in conjunction with SIP

Services Offered at Media Services Layer



Conferencing

Most Critical Media Services

- Multipoint Meeting = Conference
- Used to connect if there are more than 2 participants
    - •Most often just video terminals (endpoints)
    - •Sometimes also used in other use cases (e.g. recording)
- Different Visual Experiences (ActivePresence, Advanced CP, Voice Switched)
- Different Conference types (Adhoc, Scheduled & Rendezvous)
- Transcoding (MCU's) vs. Switching (CTMS)

Conferencing – Different Visual Experiences

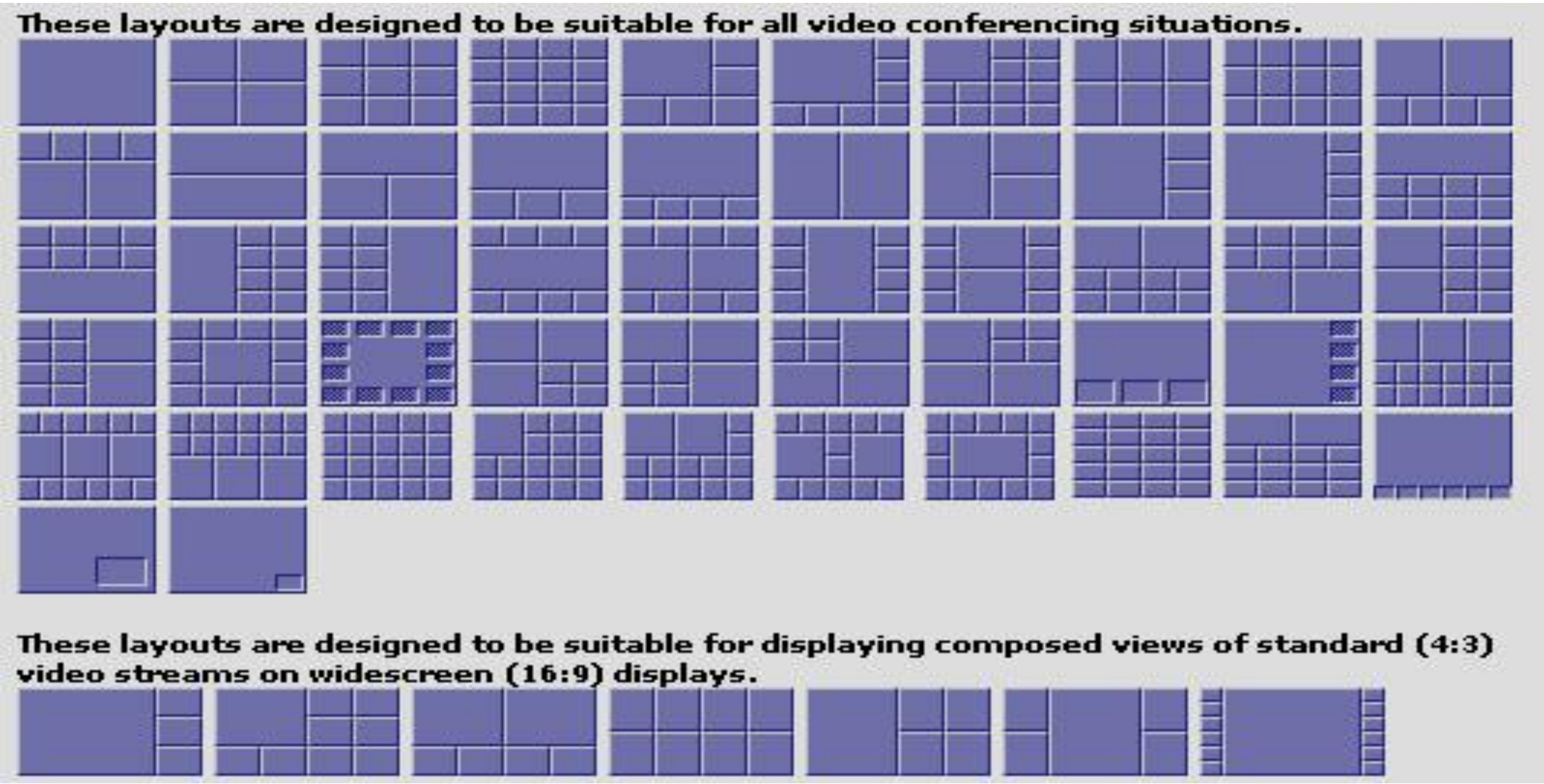
Terminology – Voice Activated Switching

- In the past seen on CTS Immersive systems.
- Maintains FULL screen active speaker.
- Screen segments are switched to the active speaker
- Multi-screen systems will see active speaker and some of the previous speakers on the other screens

Conferencing – Different Visual Experiences

Terminology – Advanced Continuous Presence

Continous Presence Layouts



Conferencing – Different Visual Experiences

- Terminology - ActivePresence

Issue

How do you maintain an immersive experience in a multipoint meeting with many endpoints?

As more endpoints join it becomes harder to maintain an immersive experience

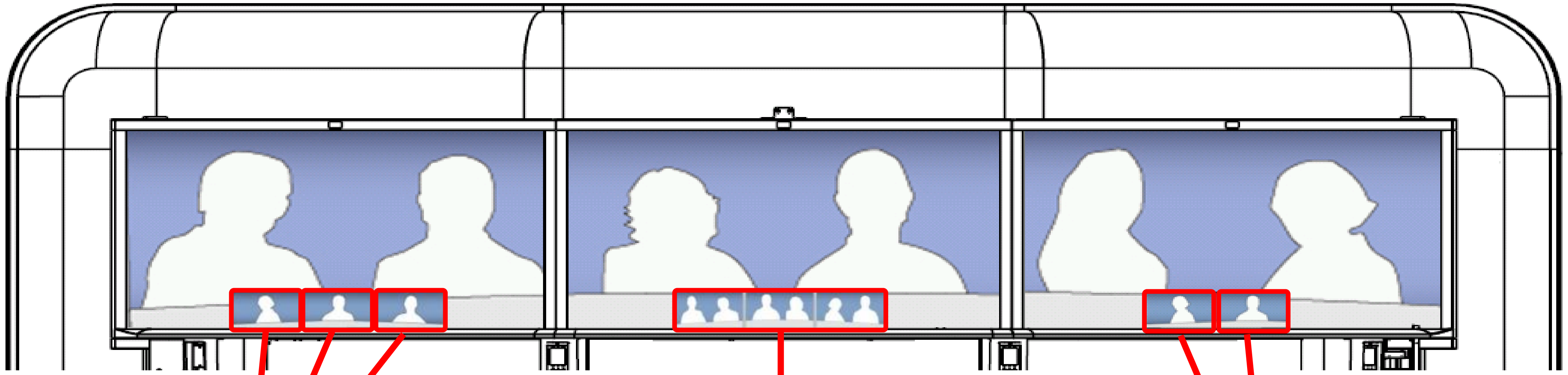
Answer

ActivePresence: Preserve ability to see multiple endpoints, but show active speaker full-screen to maintain focus

ActivePresence: A specific layout/behaviour in multipoint meetings that allows full screen, life size depictions for the current speaker while still showing the other participants in the meeting

Conferencing – Different Visual Experiences

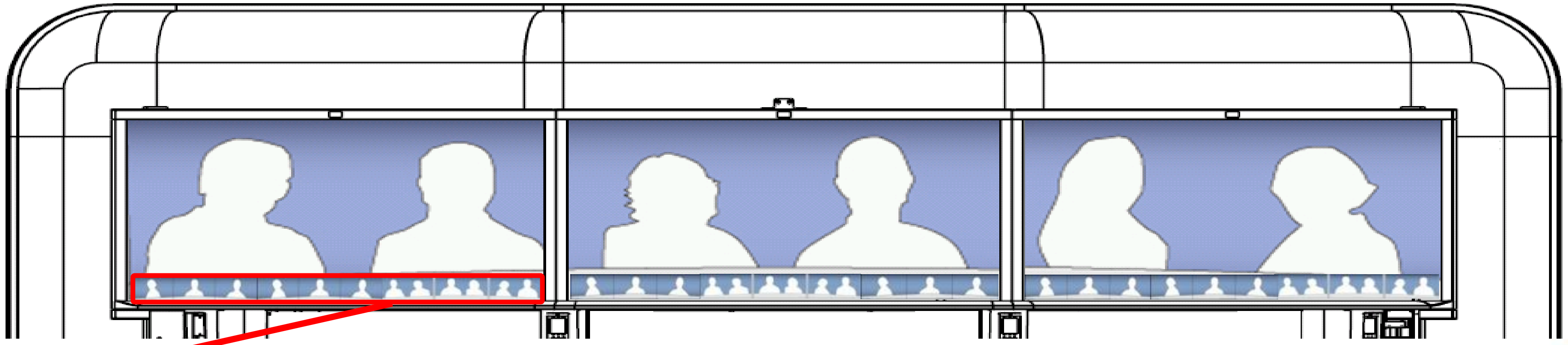
Cisco ActivePresence™



Single Screen Endpoints

Multi-screen Endpoints

Single Screen Endpoints



Up to 9 Active Presence windows on a single screen at a time. (27 total on a triple screen)

Conferencing

Terminology – Conference Types

- Ad hoc Conference
  - Impromptu meetings, they are not scheduled beforehand ,nor require an administrator to initiate them. Suitable for smaller, on-the-fly, meetings. A point-to-point call escalated to a multipoint call is considered ad hoc.
- Rendezvous Conference
  - Also called meet-me/permanent/static conferences, requires endpoints to dial in to a pre-determined number. Often used for recurring group meetings which involve different endpoints each time.
- Scheduled Conference
  - Provides a guarantee that endpoints and multipoint resources will be available at a certain time. Endpoints join manually or are automatically connected by the multipoint resource.

Conferencing

Transcoding vs. Switching

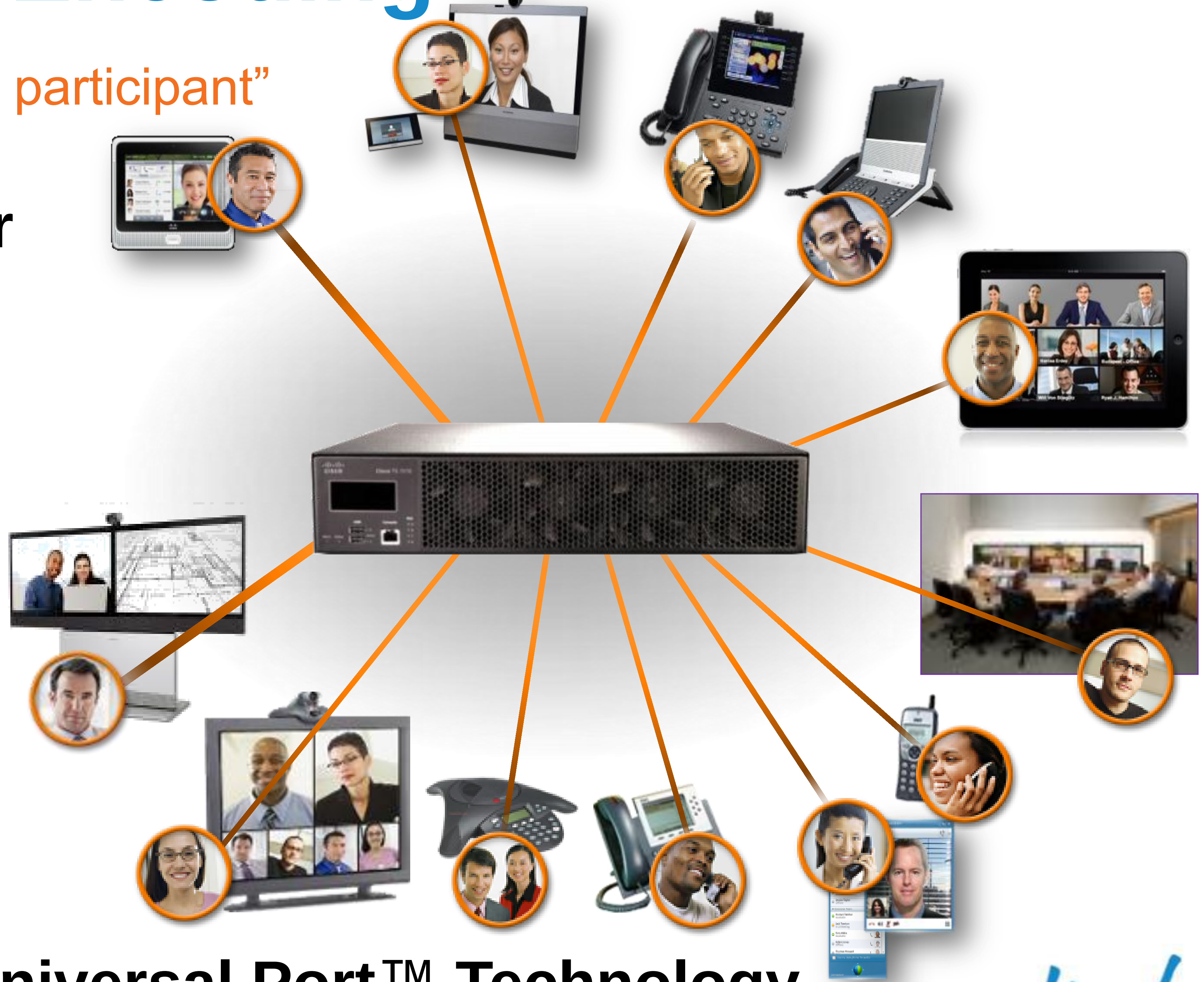
<u>Transcoding</u> • Active Presence • Universal Port Encoding • Individual transcoding per user • Customise layouts	• Slightly higher latency • Size of meetings is limited by DSP hardware • Higher cost per port • Specialised video hardware	 MCU  TelePresence Server
<u>Switching</u> • Very low latency (<10ms) • Ability to scale higher • Lower Cost per port	• Limited to basic full-screen video switching (No Active Presence) • All endpoints <u>must</u> support and agree on single resolution/frame rate • Interoperability requires additional hardware (transcoding)	 CTMS

Cisco Universal Encoding™

"The best experience possible for every participant"

Independent Port for Each User

- Active Presence / Advanced CP
- Optimal layout of multiple images
- Enabling "any to any" conferencing
- Mixed video and audio participants can join conferences
- Highest Multivendor Support



Universal Port™ Technology

Conferencing

Conferencing Platform Form Factors

Transcoding

Immersive

TelePresence Server

7010

16 ports at 720p30 or
12 ports at 1080p30

8710

16 ports 720p30 or
12 ports 1080p30

MCU

4203, 4205, 4210, 4215, 4220

6 to 40 ports at 720p15/480p30

4501, 4505, 4510, 4515, 4520

6 to 40 ports at 720p30
3 to 20 ports at 1080p30

5310, 5320

8420

40 ports at 720p15/480p30

8510

80 ports at 480p30
10 ports at 1080p30

Switching

Immersive

CTMS

Version 1.8

UCS 210 M2

48 ports at 720p30 or 1080p30

MCS 7845-I3
MCS 7845-I2
MCS 7845-H2

48 ports at 720p30 or 1080p30

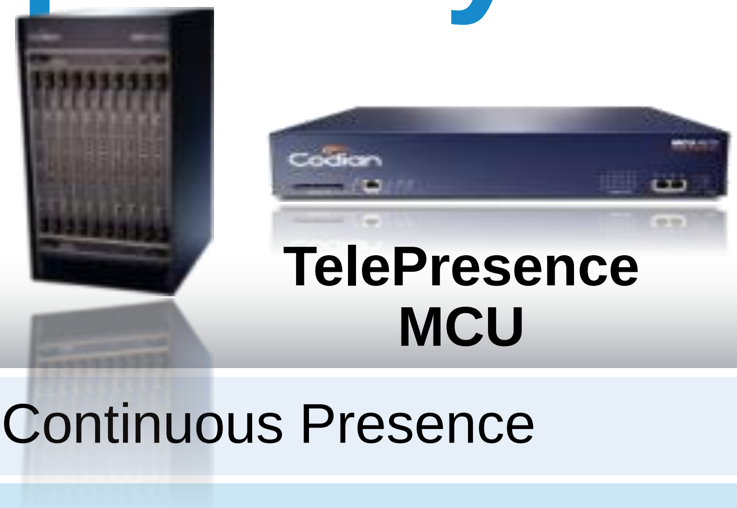
Conferencing Layout Comparison

TelePresence Server (Active Presence)

TelePresence MCU (Advanced Continuous Presence)

Cisco TelePresence Multipoint Switch (Voice Switching)

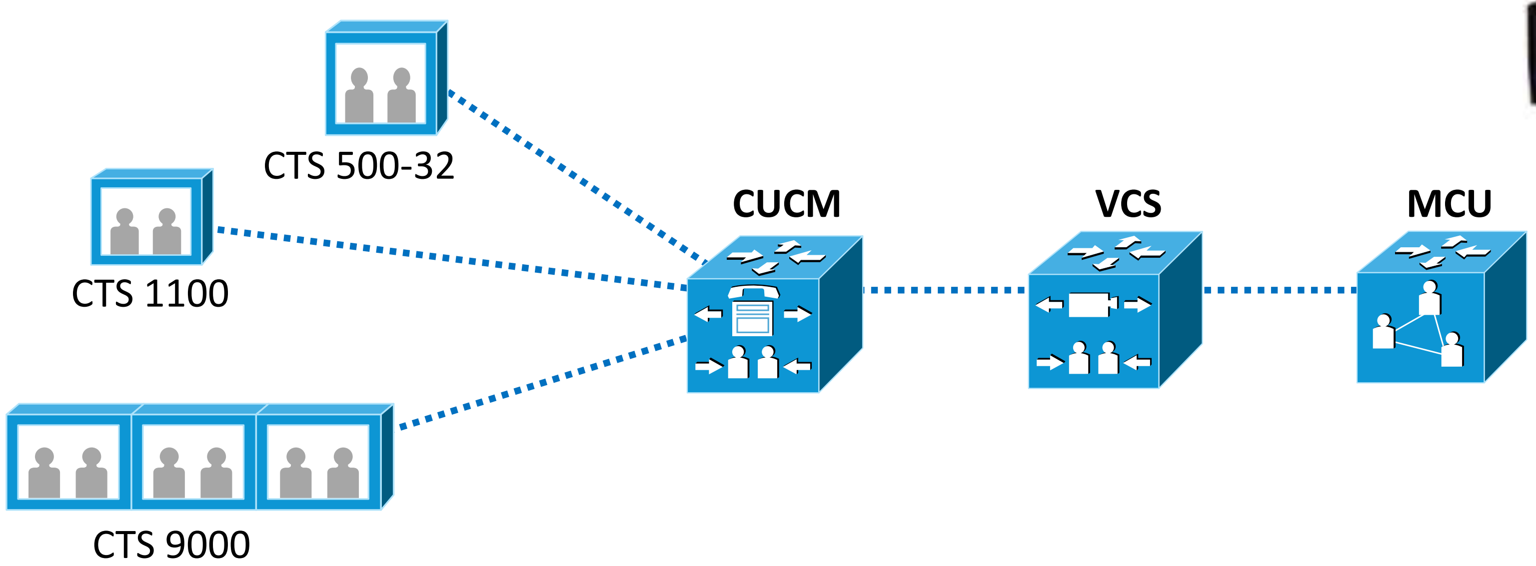
Conferencing Capability Comparison



	TelePresence Server	TelePresence MCU	Cisco TelePresence Multipoint Switch (CTMS)
User exp.	Active Presence	Continuous Presence	Voice Switching
TIP/multiscreen	Yes	No	Yes
WebEx Int.	Yes – WebEx OneTouch Two	Yes – WebEx OneTouch Two	Yes – WebEx OneTouch
Transcoding	Per Port	Per Port	No – Voice switching
Max port count in single conf.	64 (4 TP blades)@720p 48 (4 TP blades)@1080p	60 (3 HD blades)@720p or 1080p	48 per CTMS@1080p
Reg. to VCS	Ad hoc & scheduled	Ad hoc & scheduled	No
Reg. to CUCM	Ad hoc & scheduled	Ad hoc only	Ad hoc & scheduled
Use Case	▪ **Multi and single screen immersive** ▪ **Active presence**	**endpoints** ▪ **Single screen multipoint** ▪ **Continuous presence (brady bunch)**	▪ **CTS-centric** ▪ **Multi and single screen immersive**

Conferencing

- CTS with MCU



CTS 500-37 called into 4501 MCU

- CTS endpoints are supported on MCU
- In any deployment with CTS 3/9xx0 triple screen systems, TelePresence Server is still the **recommended** multipoint solution, using ActivePresence

Scaling Conferencing

Why Cisco TelePresence Conductor?

- **Conferencing using multiple MCU’s may have challenges of…**
  - Poor scalability
  - Limited resiliency
  - Ad hoc resource management and utilisation
  - Complexity of cascading conference
  - Administratively burdensome

- **Solution : TelePresence Conductor**
  - ✓ Manages the resources assigned to conferences on MCUs
  - ✓ Dialled conference aliases are agnostic of the MCU the conference is hosted on
  - ✓ Resilient solution providing service continuity if a power failure affects a VCS/MCU/TelePresence Conductor
  - ✓ Customises the conferences generated based on the aliases dialled.

TelePresence Conductor

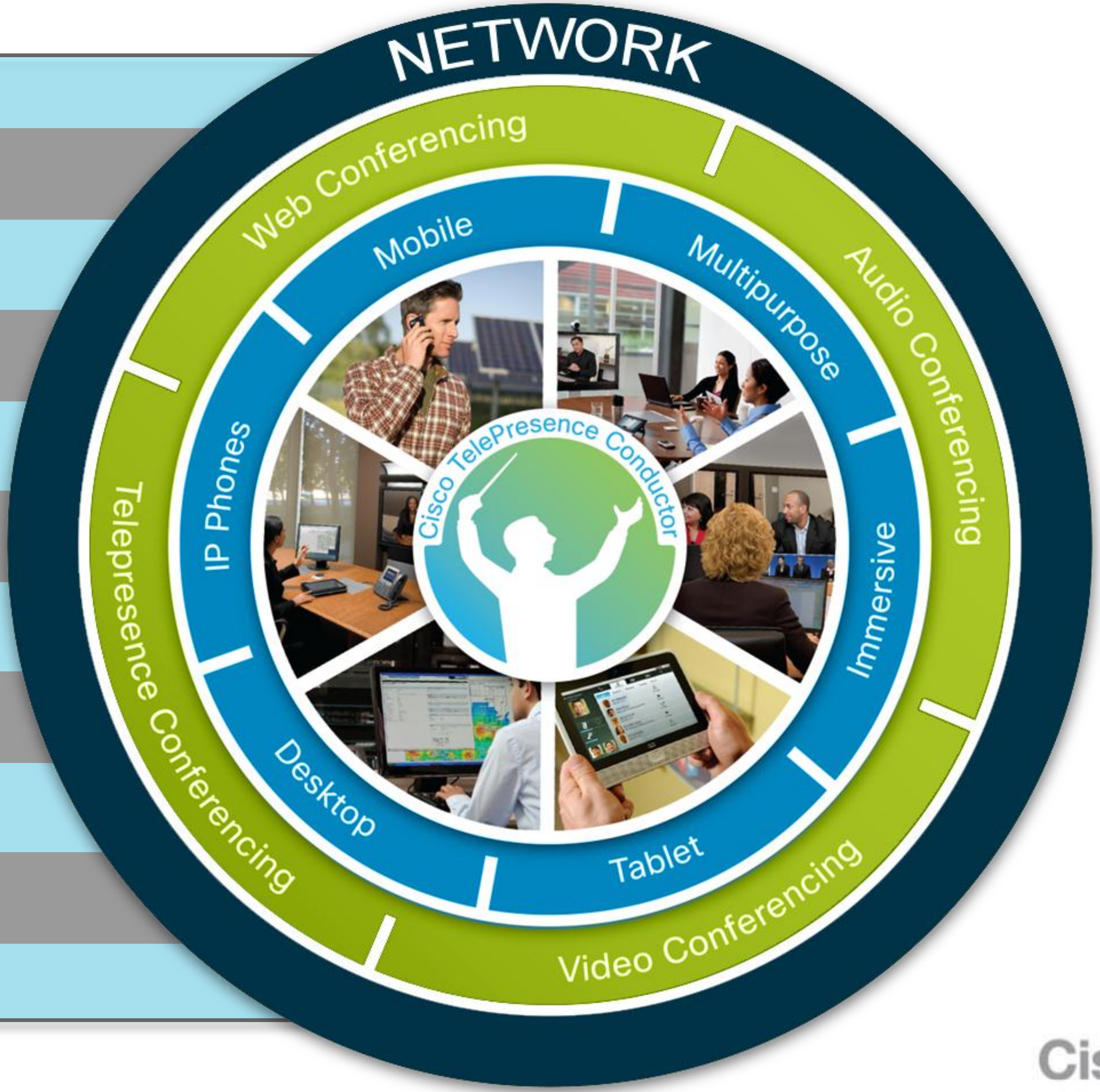
Improved User Experience
Simple to Use,

Maximise Reliability
Always Available, Zero Downtime

Extended Scale
Enhanced Conference Size, Intelligent Resource Usage

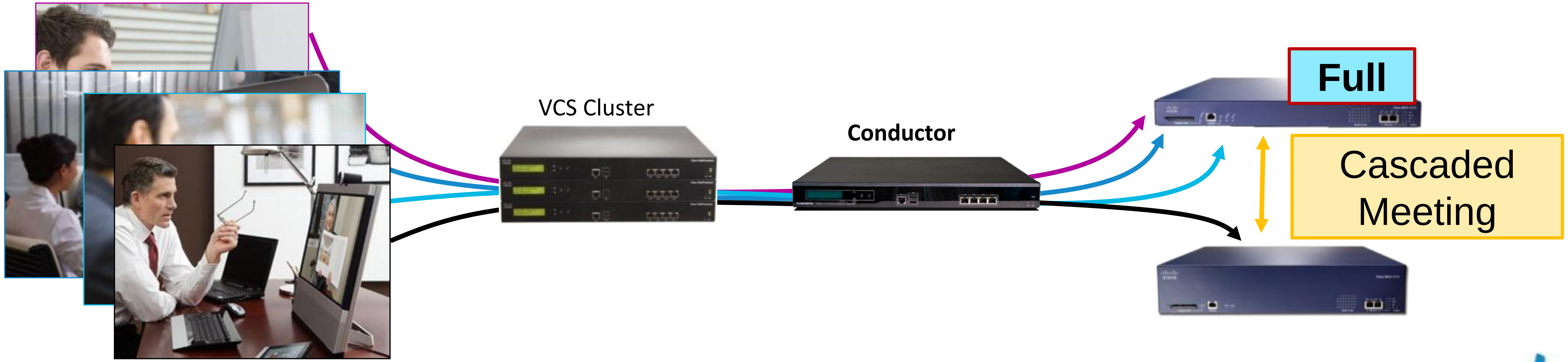
Any to Any Collaboration
Interoperable, Standards-Based

Highest Return on Investment
Minimal Operational Costs, No Forklift Upgrades



Conferencing

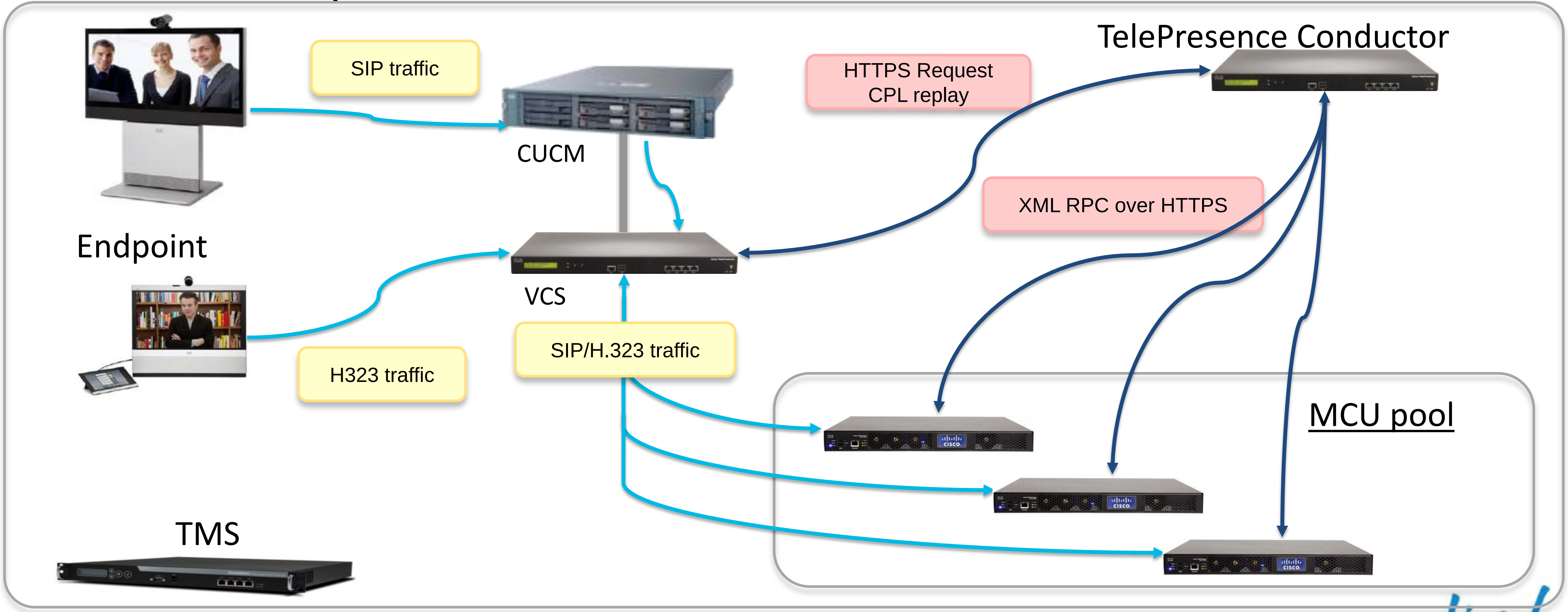
- Cisco TelePresence Conductor
- Solution with TelePresence Conductor
  - ✓ Manages MCU (42xx, 45xx, 53XX, 8420 & 8510) conference resources
  - ✓ Dialled conference aliases are agnostic of the MCU that the conference is hosted on
  - ✓ Resilient solution providing service continuity if a power failure affects a VCS/MCU/TelePresence Conductor
  - ✓ Customises the conferences generated based on the aliases dialled



Whole process is transparent to the end user

How Cisco TelePresence Conductor works

- Overall Concept



Conductor Requirements

For Your Reference

- TelePresence Conductor Solution Requirements
  - ✓ Required software versions

Parameter	Supported Software version
TelePresence VCS	X6.0 or newer version
TelePresence MCU	4.2(1.43) or newer version
TelePresence Management Suite	TMS13.1.2
Unified Communication Manager	CUCM8.5 or newer version
H.323 Endpoint / SIP UA	N/A (TelePresence Conductor solution use standard H.323/SIP call flow between Endpoint and MCU via VCS)

  - ✓ TelePresence MCU's must be under the **full** control of TelePresence Conductor (or TelePresence Conductor Cluster).
    Any other Conferences created on the MCU will be **deleted.**
  - ✓ **Disable** "Allow Bookings" in TMS for MCUs controlled by TelePresence Conductor
  - ✓ Initial TMS integration is within CCC only and for ad-hoc conferences.

Overall Setup requirements

Cisco VCS Configuration

1. Service Policy - TelePresence Conductor
2. Search Rule for Conference Aliases configured on TelePresence Conductor
3. Neighbour zone configuration – MCU
4. Search Rule for MCU dial plan prefix

Conductor Configuration

1. MCU Pool configuration
2. MCU Service Preferences configuration
3. Conference template configuration
4. Conference aliases configuration
5. Auto-dialed participants configuration

Cisco MCU Configuration

1. H.323 Configuration
2. SIP Configuration
3. User account

Note: Reference the deployment guide for step by step instructions to configure of all components:

http://www.cisco.com/en/US/docs/telepresence/infrastructure/conductor/config_guide/Cisco_TelePresence_Conductor_Deployment_Guide_XC1-2.pdf

Solution Overview

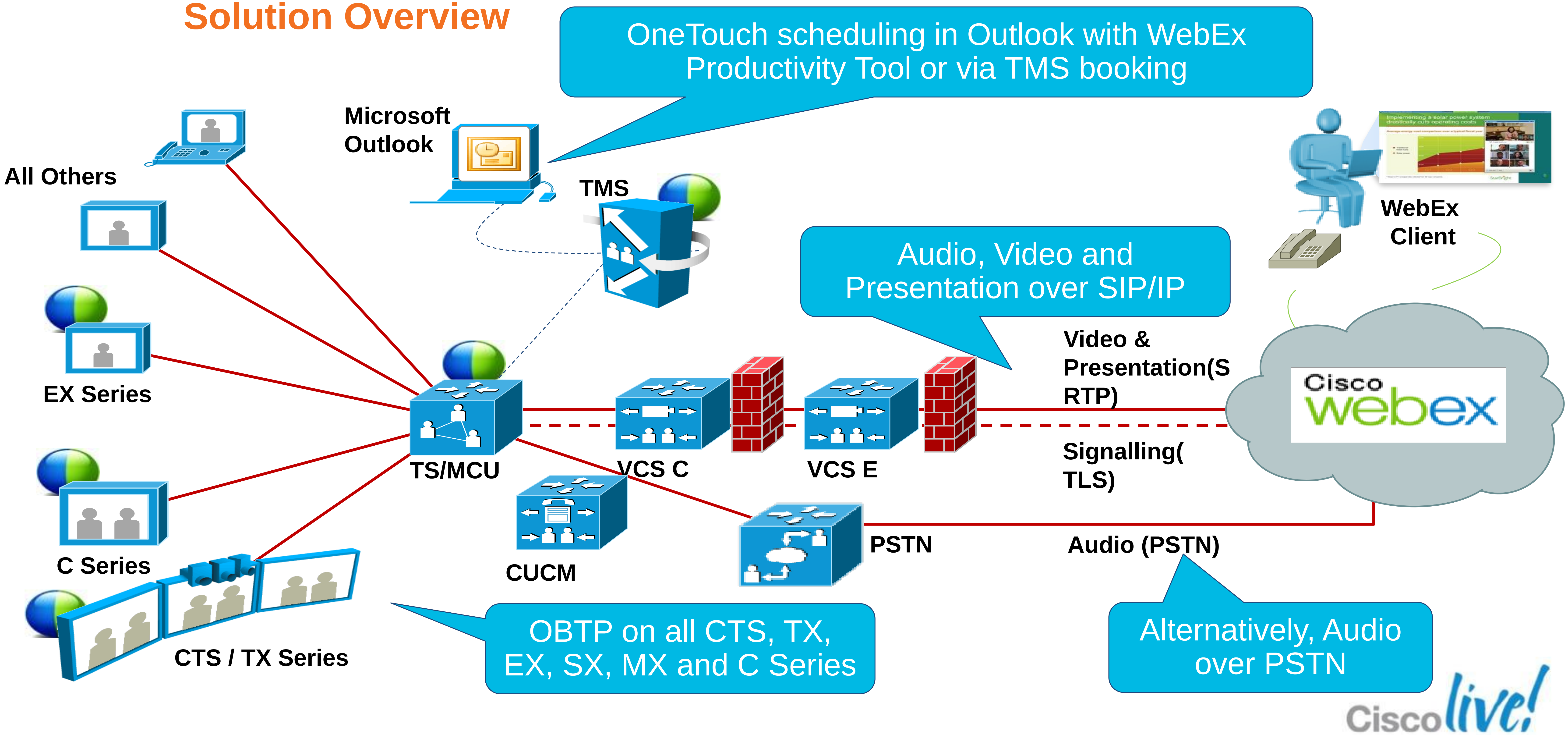
TelePresence WebEx OneTouch 2.0

Main Features

- Automatic (OneTouch) connection between TelePresence and WebEx with two way audio, video and presentation between all participants
- Easy Scheduling with Microsoft Outlook and Cisco TMS
- Standards based – any endpoint can join
- One Button to Push on select Cisco TelePresence endpoints
- VCS Expressway to WebEx Cloud
- Audio via SIP/IP using WebEx audio, or via PSTN with third party (TSP) audio providers
- WebEx Meeting Centre and mobile clients

WebEx OneTouch 2.0 Architecture

Solution Overview



Solution Components

- Conference Bridge with TelePresence MCU or TelePresence Server
- Scheduled calls using TMS and TMS Scheduling Extension (TMS-SE) with Microsoft Exchange EWS Integration or TMS Scheduling Extension API
- VCS Expressway
- WebEx Meeting Centre
- Optional Outlook Productivity Tool
- Any TelePresence endpoint supported by TelePresence MCU or TS, registered to CUCM or VCS
- WebEx Meeting Centre Client for PC or Mac. Support for mobile iPad client is also planned.

Fundamentals, Design & Architecture

Call Processing

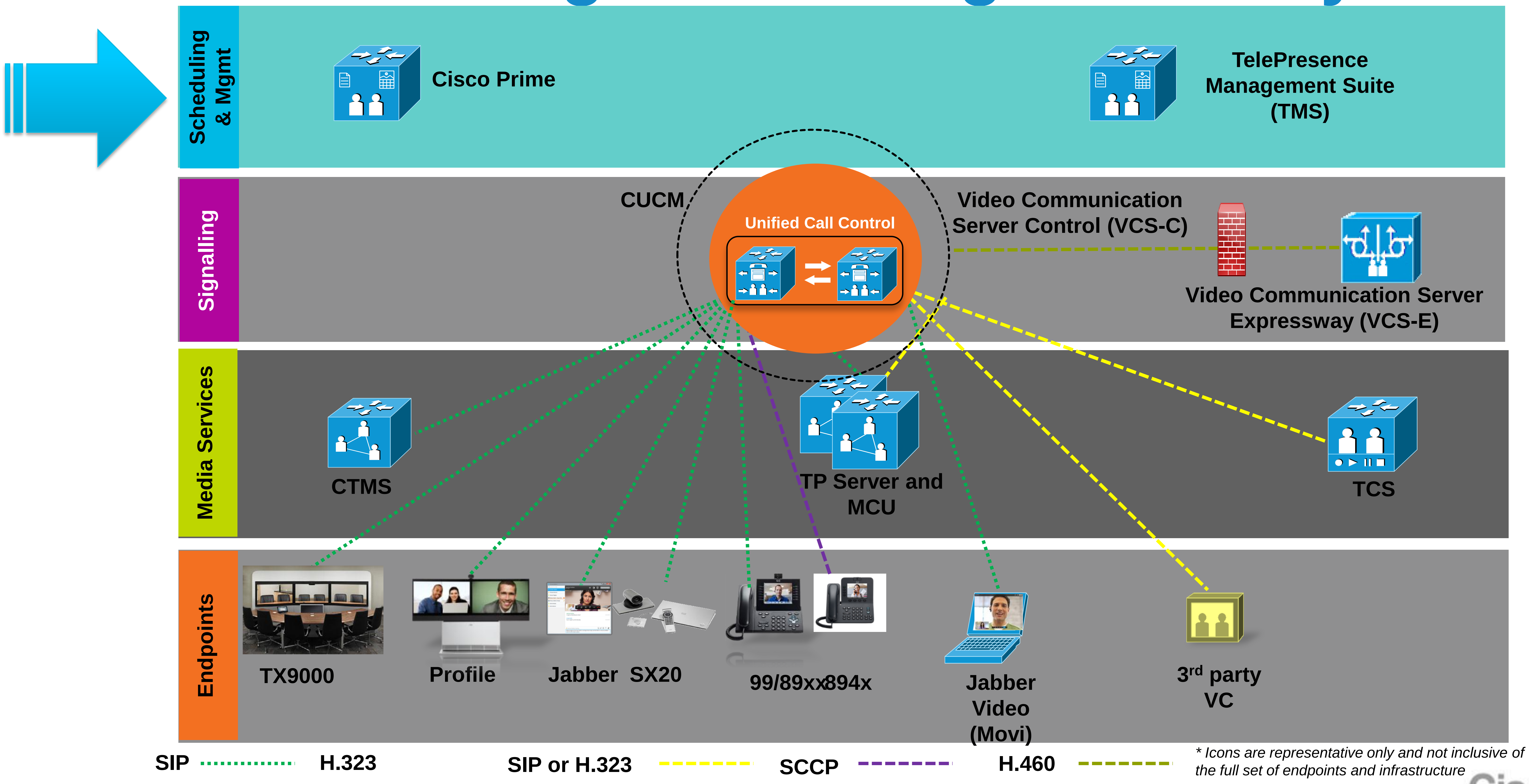
Dial Plan

Conferencing

Scheduling

Business to Business

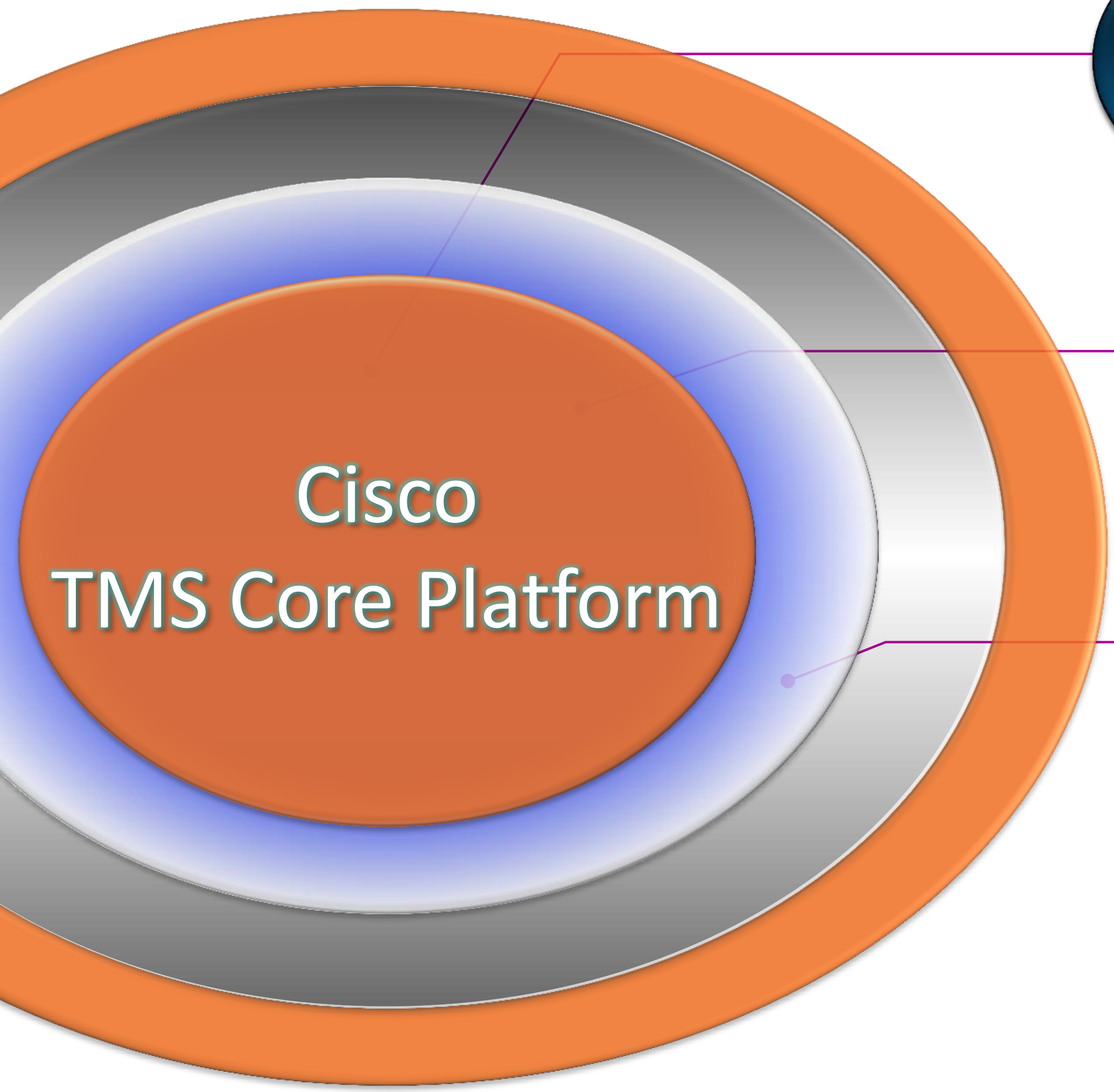
Scheduling and Management Layer



** Icons are representative only and not inclusive of the full set of endpoints and infrastructure*

Cisco TelePresence Management Suite

Centralised TelePresence Management Suite With Extensions



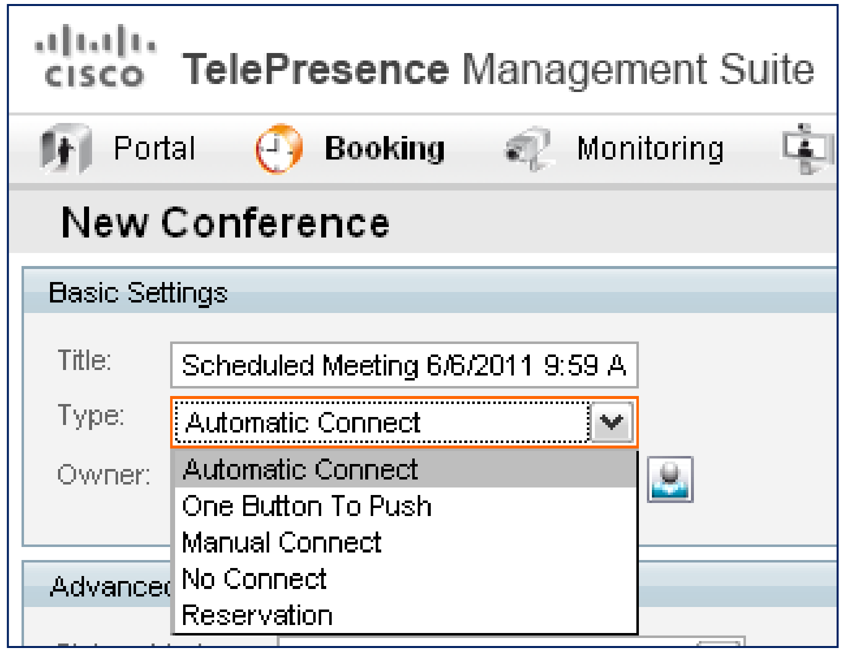
- TelePresence Management Suite Core Platform (TMS)

- TelePresence Management Suite Provisioning Extension (TMSPE)
- TelePresence Management Suite Extension for Microsoft Exchange (TMSXE)
- TelePresence Management Suite Extension for Lotus Notes (TMSXN)
- TelePresence Management Suite Extension Booking API (TMSBA)

Scheduling and Management

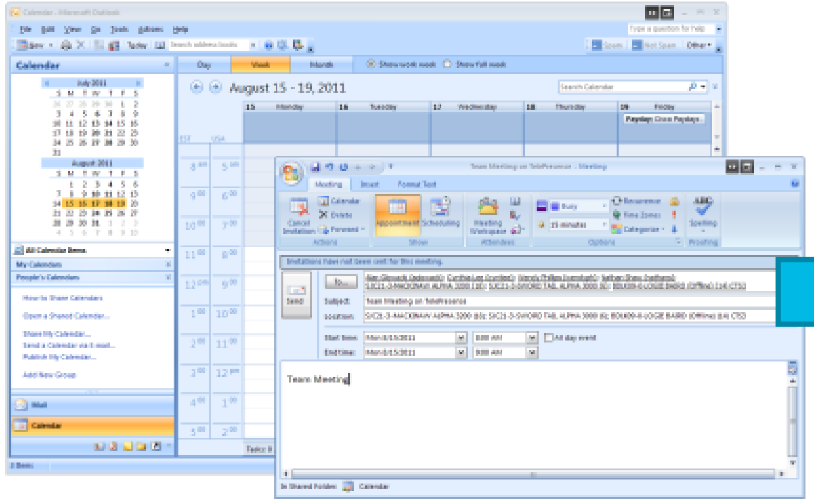
- TMS call launch options

- TMS has several call launch types: Auto, OBTP, Manual, Reservation & None
- Automatic Connect - New to CTS Systems
- CTS endpoints configured for meetings with Automatic Connect
  - CTS endpoint displays the meeting on the UI
  - User can not launch the call
  - CTS is dialed by the multipoint device at the meeting start.

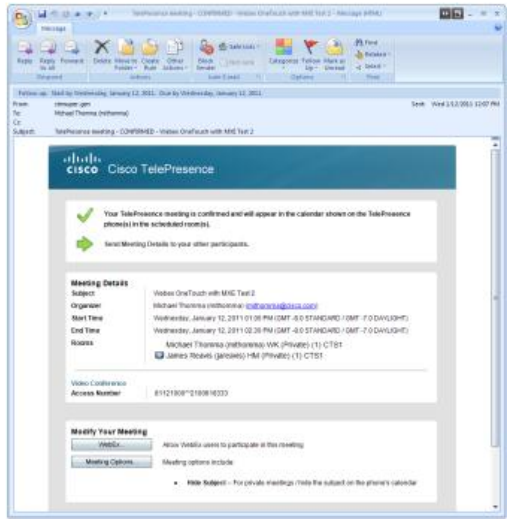


Scheduling and Management

- TMS call launch options - OBTP (One Button to Push)



User schedules TelePresence rooms in Outlook



User receives confirmation email

Participants enter their TelePresence Rooms

One button press to launch call

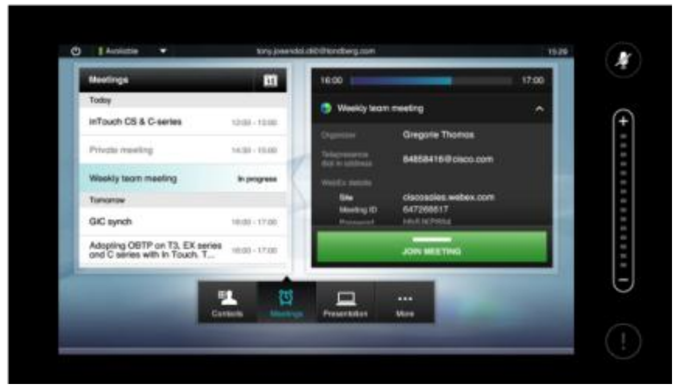
Scheduling and Management

- One Button to Push

- OBTP is available on the Cisco Touch 12, Touch 8, 797x IP phones and on-screen display (OSD) with remote control
- With TMS 13.2 and CTS 1.8, multipoint meetings can be scheduled on the TelePresence Server or MCU
- CTS and EX/MX/C series on TC5/6 can use OBTP

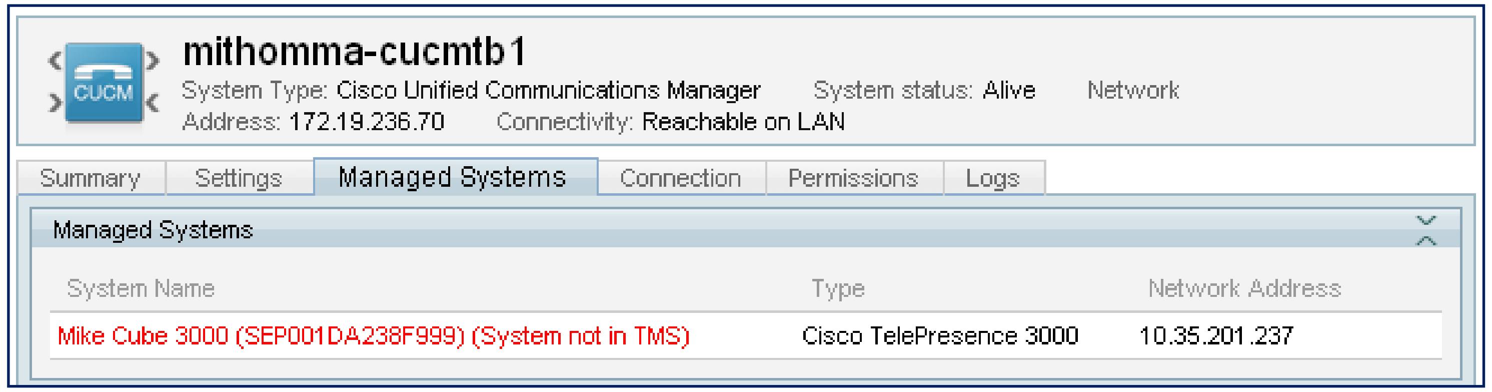
- Consistent User Experience

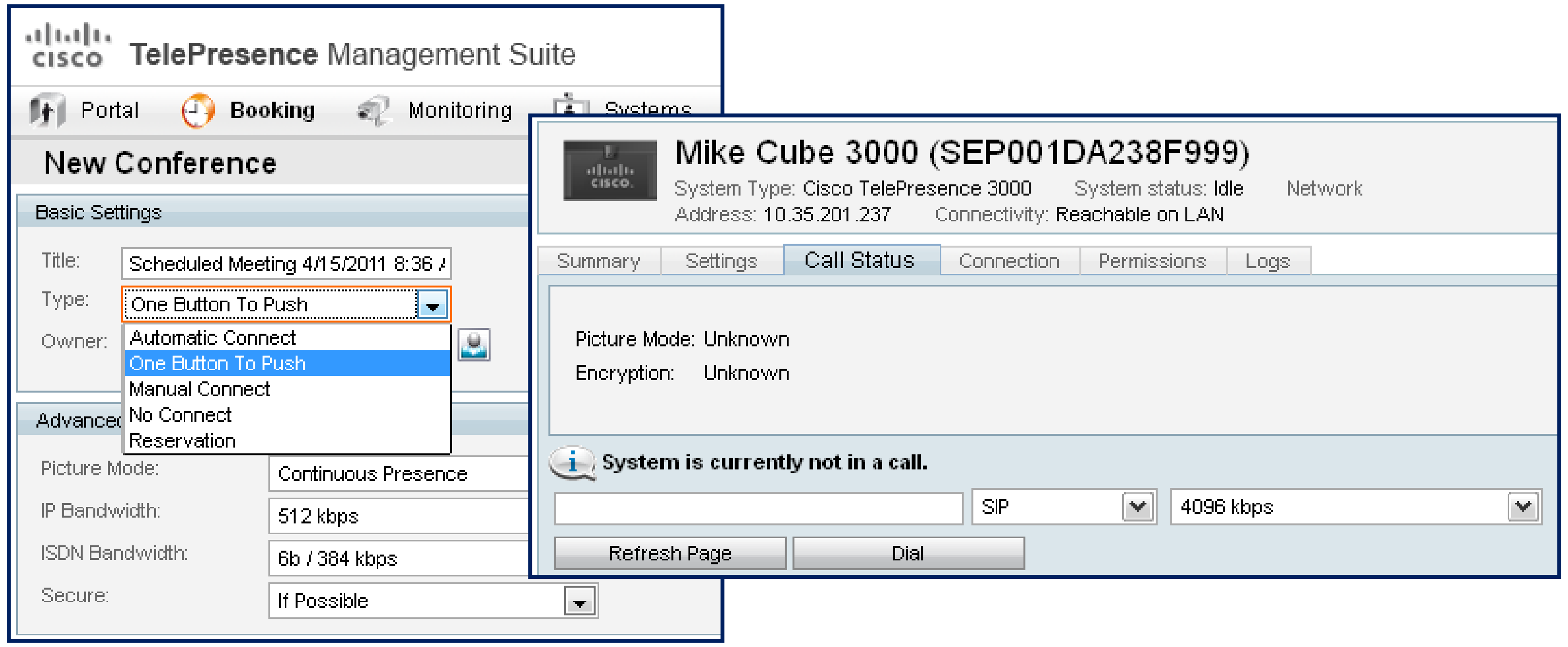
Cisco Touch 12


Cisco Touch 8

Scheduling and Management

- TMS release 13.1 or later support for scheduling CTS endpoints
- CUCM 8.5 or later is **required**. TMS will log into CUCM and return all **registered** CTS systems in CUCM
- CTS 1.7.0 or later is **required** for TMS management

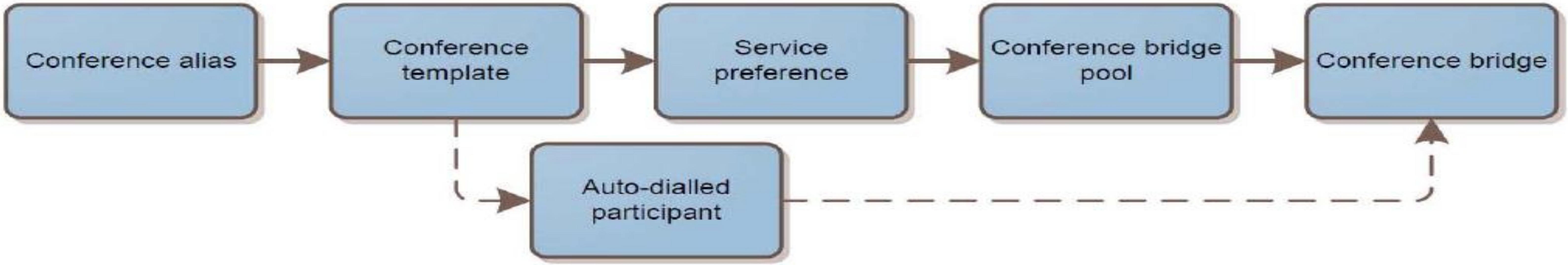




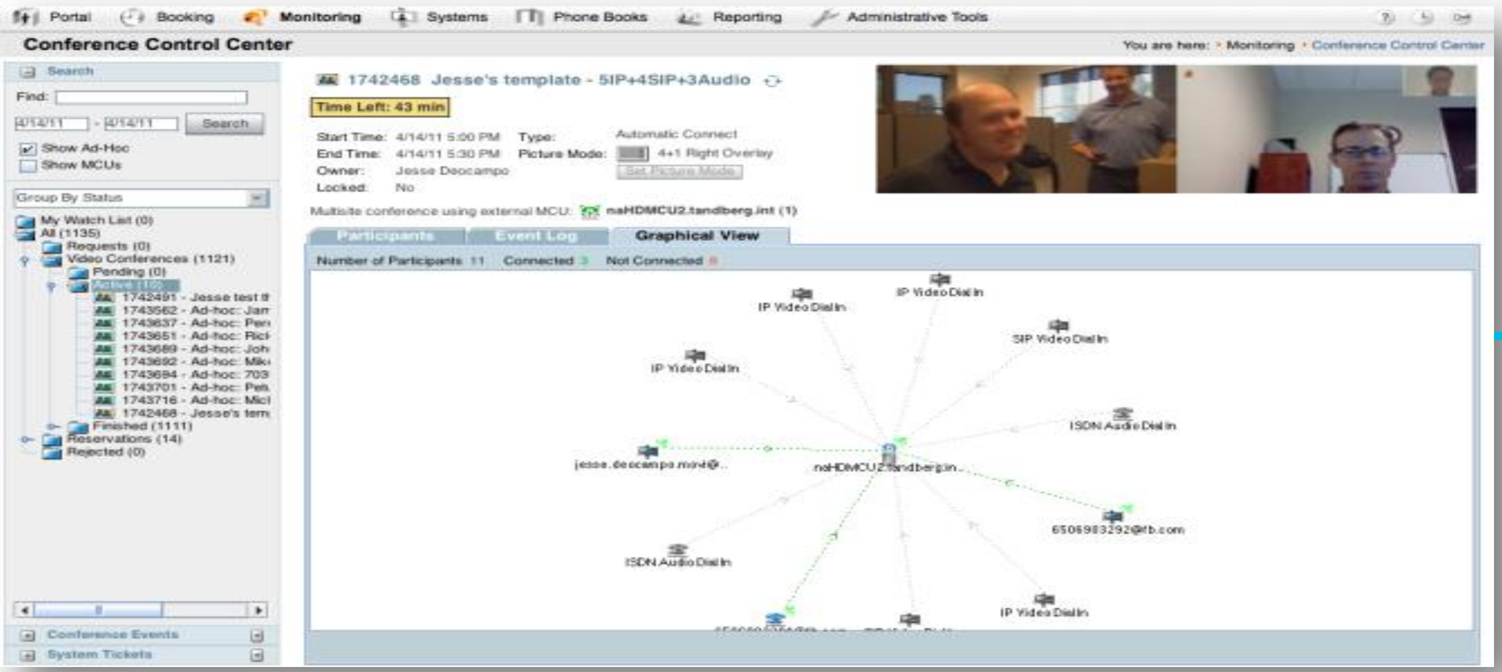
- When a CTS system is added to TMS, TMS can provide :
  - **OBTP** with TelePresence Server
  - Schedule P2P calls
  - Read system information
  - Monitor response status and call status
  - Dialing from the endpoint

Conductor Scheduling

- Normal Conductor workflow:



Taking Management to the next level



Cisco TMS 14.x

- Booking and scheduling of sessions
- Conference and participant connection control
- Phonebook management and directory import
- Infrastructure fault collection and event forwarding
- Powerful, customised reporting and analytics

Cisco Prime Collaboration Manager

- End to end visibility of all sessions
- Monitoring and troubleshooting of sessions and infrastructure devices
- Advanced event configuration and fault correlation
- Simplified out-of-the-box reports with cross launch to Cisco TMS for customised reporting
- Comprehensive inventory

Fundamentals, Design & Architecture

Call Processing

Dial Plan

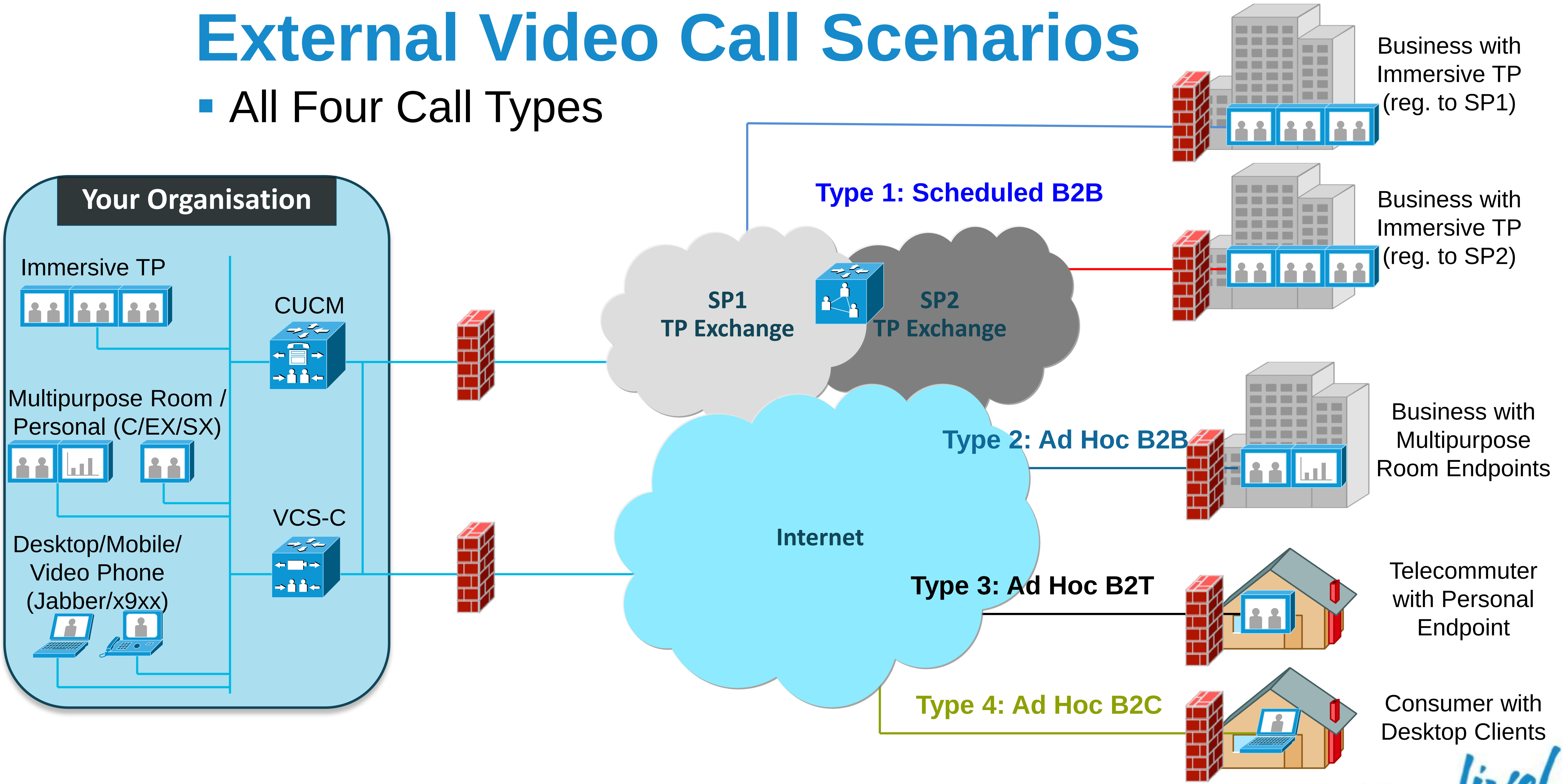
Conferencing

Scheduling

Business to Business

External Video Call Scenarios

- All Four Call Types



Summary of External Video Call Scenarios

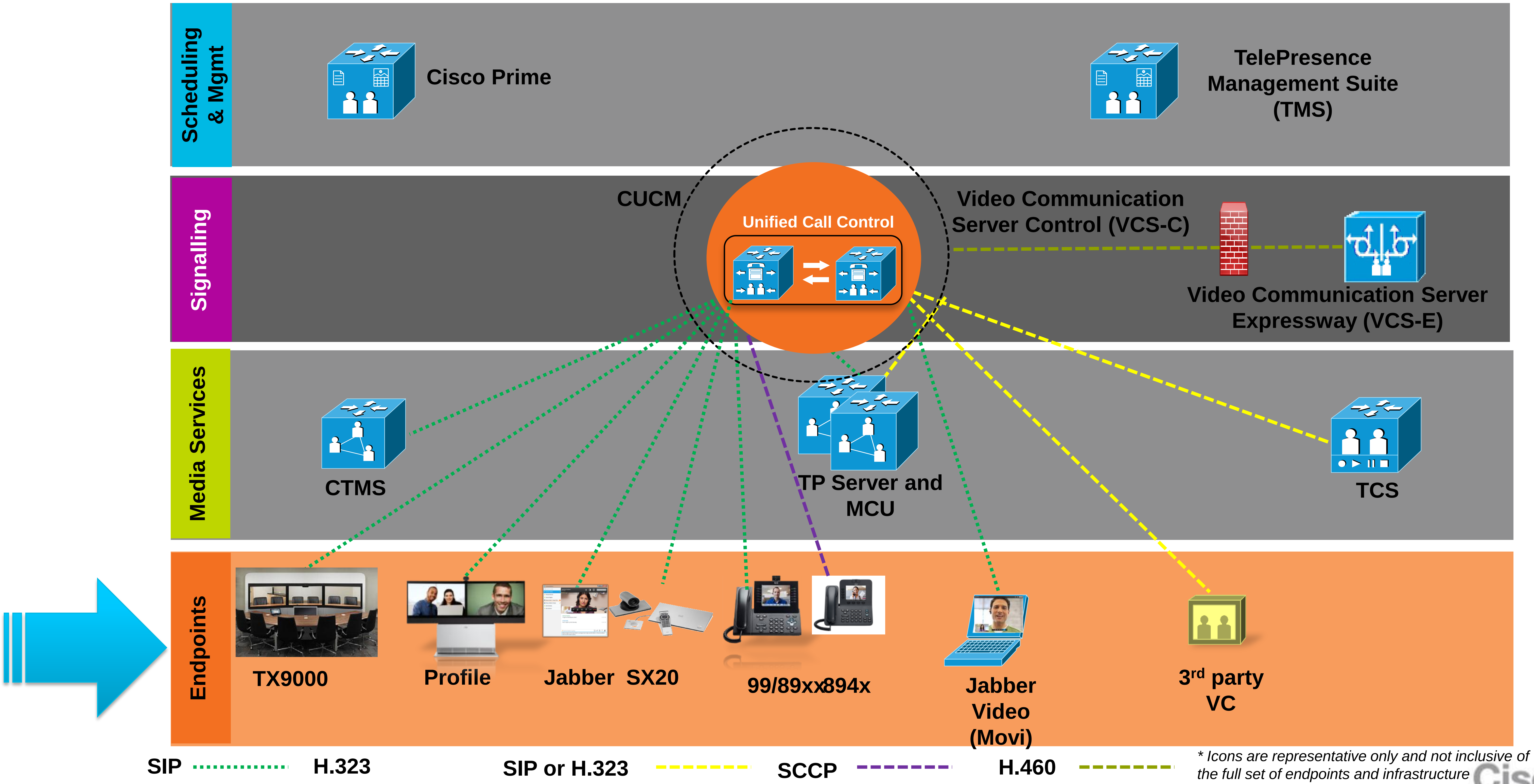
	(1) Scheduled B2B	(2) Ad hoc B2B	(3) Ad hoc B2T	(4) Ad hoc B2C
Transport	Service Provider MPLS	Internet (can be IP VPN)	Internet	Internet
Common Video Endpoints	Mostly Immersive TelePresence	Room-based or personal systems	Personal, desktop client, or video phone	Personal, desktop client, or video phones
Dial Plan	Numeric	URI / IP address	URI / IP address / Numeric	URI / Numeric
Quality of Experience	Guaranteed	Variable (over Internet)	Variable	Variable
Multipoint Calls	Provided by SP multipoint resource	Enterprise	Enterprise	Enterprise
Coverage	SP TP Exchange Customers	Nearly everyone	Nearly everyone	Nearly everyone

Dial Plans Considerations

How to Reach Other Endpoints Beyond Your Network

	Numeric / E.164	IP Address	DNS-based URI
Format	55551234	200.60.30.105	dean.ex90@company.com
Does this work on the Internet?	Requires ENUM to map E.164 number to IP, or other method	Yes	Yes
Drawbacks	Few ENUM (& other methods) implementation	IP changes? Private IP? Swiss cheese firewall?	Alpha-numeric dialing hard on numeric keypad
Requires	ENUM, or everyone using common call control platform	Endpoints that can dial IP	Endpoints that can dial URI
Cisco Video Endpoint Support	All	C-series, EX-series, MXP-series, E20, Jabber Video	C-series, EX-series, SX20, Jabber Video

Endpoint Layer



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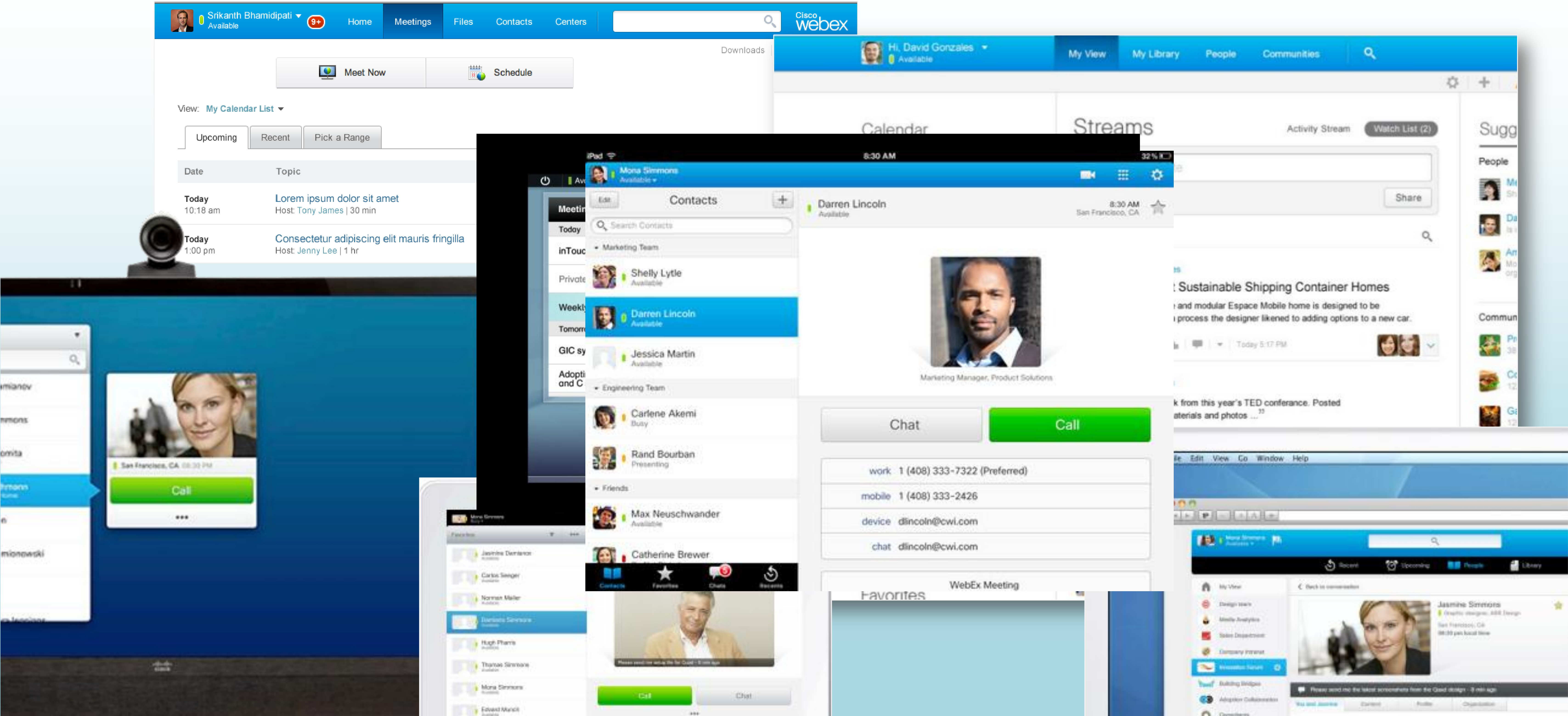
Consistent Interface Across Portfolio



Cisco TelePresence Endpoints At A Glance

Immersive

TX 9000 Series

9000 9200
Active Collaboration Room

TX 1300 Series

1300-65

1300-47

Multipurpose

PROFILE Series

Profile 65
Single or Dual

Profile 55
Single or Dual

Profile 42

MX Line

MX300 and MX200

Personal

CTS 500

500-32

EX Series

EX90 EX60

Mobility

Jabber Video for TelePresence

Solution Platforms

For custom and industry applications

INTEGRATORS

Codec C90
Codec C60
Codec C40

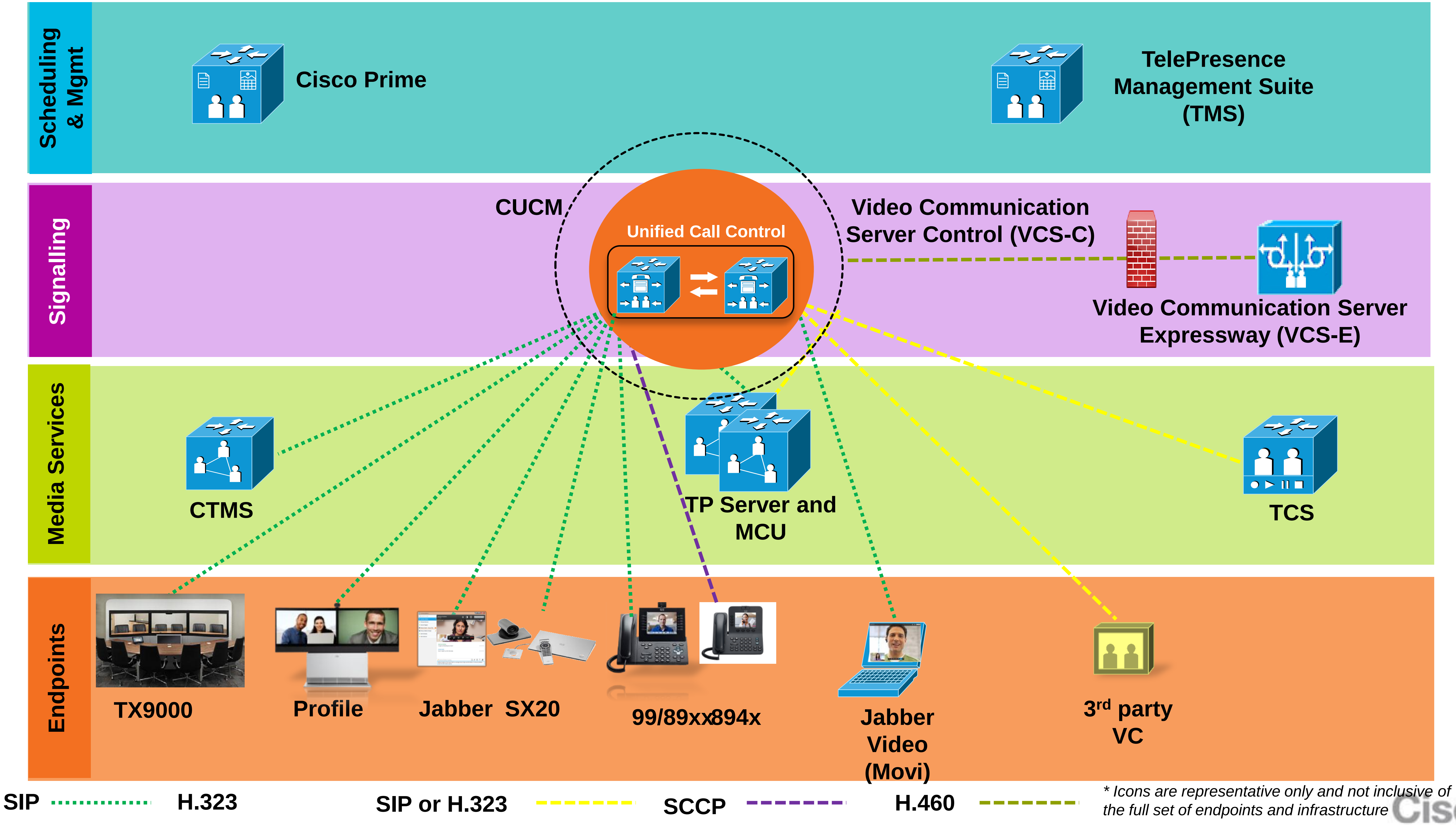
QUICK SETS

SX20 Quick Set

VERTICALS

Healthcare & Education

Wrapping It All Together



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