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# DESIGNING AND DEPLOYING IP VIDEO TELEPHONY NETWORKS

**Session VVT-2100**

**Networkers Solution Forum Argentina**

**Presenter:**

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# Recuerde siempre:

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- Apagar su teléfono móvil/pager, o usar el modo “silencioso”.



- Completar la evaluación de esta sesión y entregarla a los asistentes de sala.



- Ser puntual para asistir a todas las actividades de entrenamiento, almuerzos y eventos sociales para un desarrollo óptimo de la agenda.



- Completar la evaluación general incluida en su mochila y entregarla el miércoles 8 de Junio en los mostradores de registración. Al entregarla recibirá un regalo recordatorio del evento.

# Questions This Presentation Will Address

- **What is video telephony?**
- **What do I do with my existing H.323 equipment?**
- **How does CM4.1 control video endpoints?**
- **How do I design my network for QoS with video telephony?**
- **How does the addition of video effect my existing IP telephony deployment?**

# Goals of This Session

- **Understand how voice and video have been unified with Cisco CallManager 4.x**
- **Understand the integration of SCCP and H.323 devices - terminals, mcu's, gateways**
- **Understand the design options and capabilities for video telephony**
- **Learn how to configure Cisco CallManager and H.323 video gatekeepers, MCUs and gateways to be able to route calls back and forth to each other**

# Agenda

- **Video Telephony Fundamentals**
- **Endpoint, MCU and Gateway Integration**
- **Centralized Design and Deployment**
- **Distributed Design and Deployment**
- **Configuration**

# Introduction

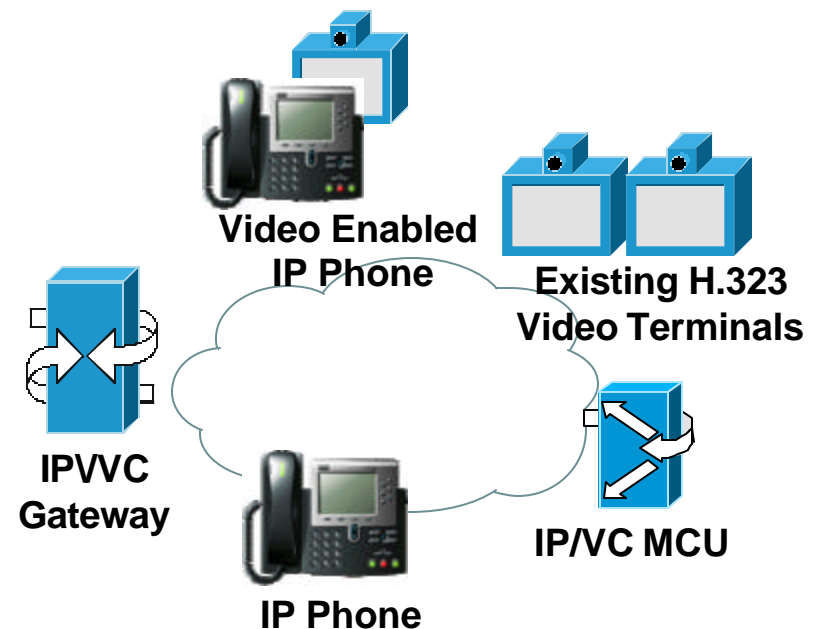
## Video Is a Phone Call

### User friendly

- **Unified voice and video dial plans**  
CDR, QoS, CAC, AAR
- **Single number for voice and video**
- **Video calls dialed like voice calls**
- **Video calls have same services as voice calls**  
Basic call, mute, hold, park, transfer, conference  
Forward, XML services
- **Single point of management and administration—Proven scalability**

### Preserves customer investments

- **Calls can be made to H.323 or SCCP video/audio terminals**



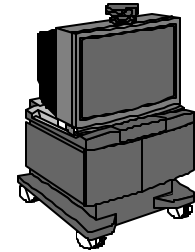
# Introduction

## Video Telephony Solution Components

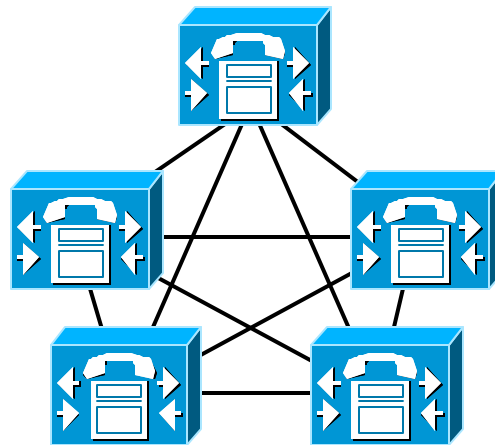


3<sup>rd</sup>-party SCCP  
Video Endpoints

H.323-based  
Video Endpoints



VT Advantage

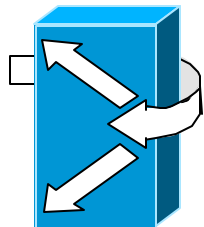
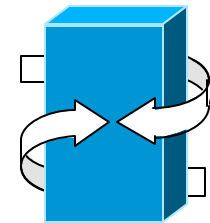


CallManager 4.x



H.323 Gatekeeper

H.320 Gateways



SCCP-based  
Conference  
Bridges

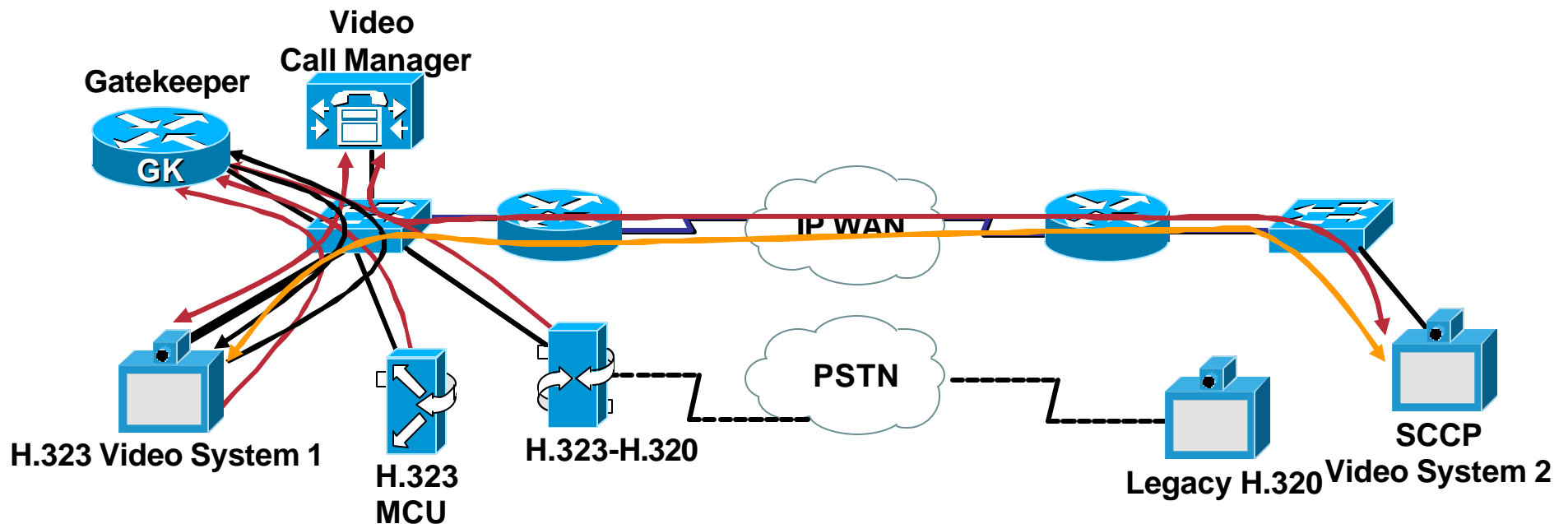
H.323-based  
Conference  
Bridges



# How Does It Work?

## Video Call Manager Example

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- All H.323 video devices register with gatekeeper (IP Address, E.164, H323-ID)
- Video System 1 dials video system 2 (using E.164), gatekeeper forwards call request to VCM based on “default-technology” registration
- Video system 1 sends call setup to VCM and VCM resolves address
- Video system 1 connects (video and audio) directly to video system 2; H.245 Control traffic is routed through VCM



# Introduction

## Cisco CallManager 4.1(3) Video Features

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- **Cisco Callmanager 4.0 Video Enhancements**

- H.323 and SCCP Protocol Stacks modified to support video capabilities**

- Added support for H.261, H.263+, Wideband Video, G.728, G.722, and FECC**

- New Cisco VT Advantage + IP Phone endpoint solution**

- SCCP protocol stack added to Cisco IP/VC 3511 and 3540 MCUs**

- Support for H.323 utilizing IP/VC 3511 and 3540 MCU's**

- Video Dial Plan Management/Call Routing (Partitions, CAC, Calling Search Spaces, Route Patterns, Translation Patterns, Shared Line Appearances, Hunt Groups, Auto-Alternate Routing, Call Forwarding, etc.)**

- Video statistics enabled**

- **What did Cisco Callmanager 4.1 Enhance over Call Manager 4.0?**

- SCCP H.264 support**

- Mid-Call Video for CVTA**

- Video Display Mode for IPVC 4.0**

- Video conference Participant Information for IPVC 4.0**

- Dynamic H.323 Addressing (E.164 addressing for GK controlled endpoints)**

# Introduction

## Deployment Models Supported

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- **Basically, any deployment model supported for voice is now also applicable for video, including:**

**Single-Site Deployments**

**Centralized Call Processing**

**Distributed Call Processing**

**V<sup>3</sup>PN / Telecommuter environments**

**Integration with existing H.323 and H.320 videoconferencing endpoints, MCUs, gateways and gatekeepers**

**Utilizing CM4.1 and IP/VC 4.0**

# Agenda

- **Video Telephony Fundamentals**
- **Endpoint, MCU and Gateway Integration**
- **Centralized Design and Deployment**
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# SCCP Endpoints

## Tandberg Devices SCCP Release 1.X

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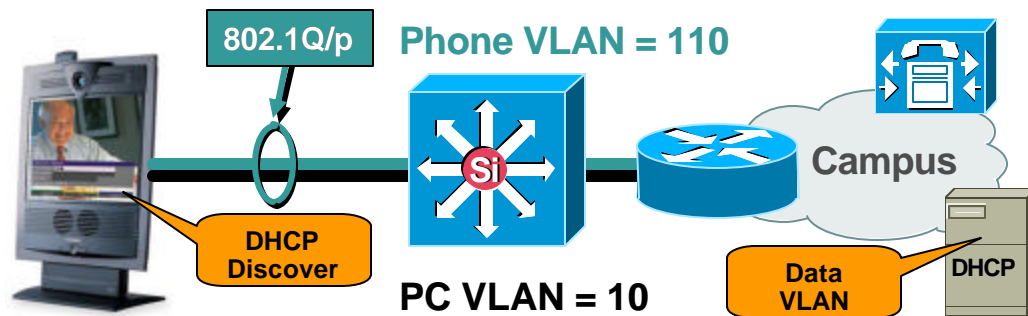
- DHCP Option 150 Support
- No CDP support
- Downloads Config via TFTP  
(e.g. SEPXXXXXXXXXXXXX.cnf.xml)
  - Software versions are not downloaded via TFTP  
(Tandberg SCCP Upgrade Tool available from Tandberg)
  - Software release key obtained from Tandberg
- CallManager QED Device Patch Installer  
(available from Tandberg) required
- Functions just like a SCCP IP Phone
  - SoftKeys, Settings, Messages, Directories  
(Received/Placed/Missed/Corporate), XML  
Services\*
- H.261, H.263+, G.711 and Far-End Camera Control
- T-1000 and T-550 models only (additional models will be supported in the future)

\* Many XML services not yet supported, such as Extension Mobility for instance

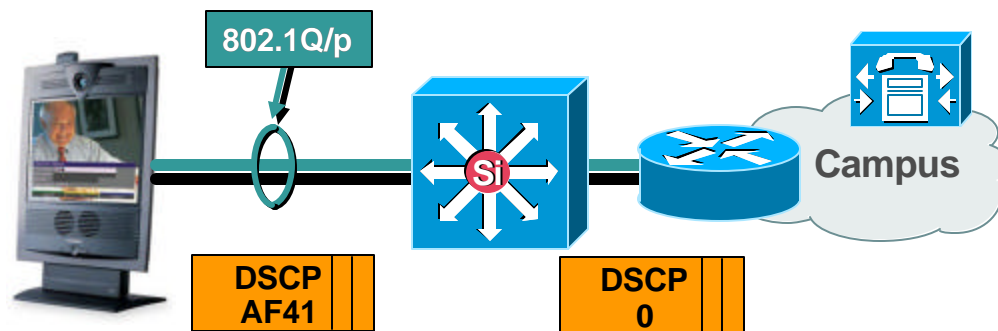
# SCCP Endpoints

## No CDP or 802.1Q/p Support in Third-Party Devices

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- Device will come up in the PC VLAN  
Option 150 must be enabled in the PC VLAN DHCP scope or else alternate TFTP must be used



- Devices mark DSCP correctly, but 802.1p COS will be empty  
If switch is set to trust COS, DSCP will be rewritten to 0  
Must set switch port to trust DSCP instead, or use an ACL

# SCCP Endpoints

## Cisco VT Advantage

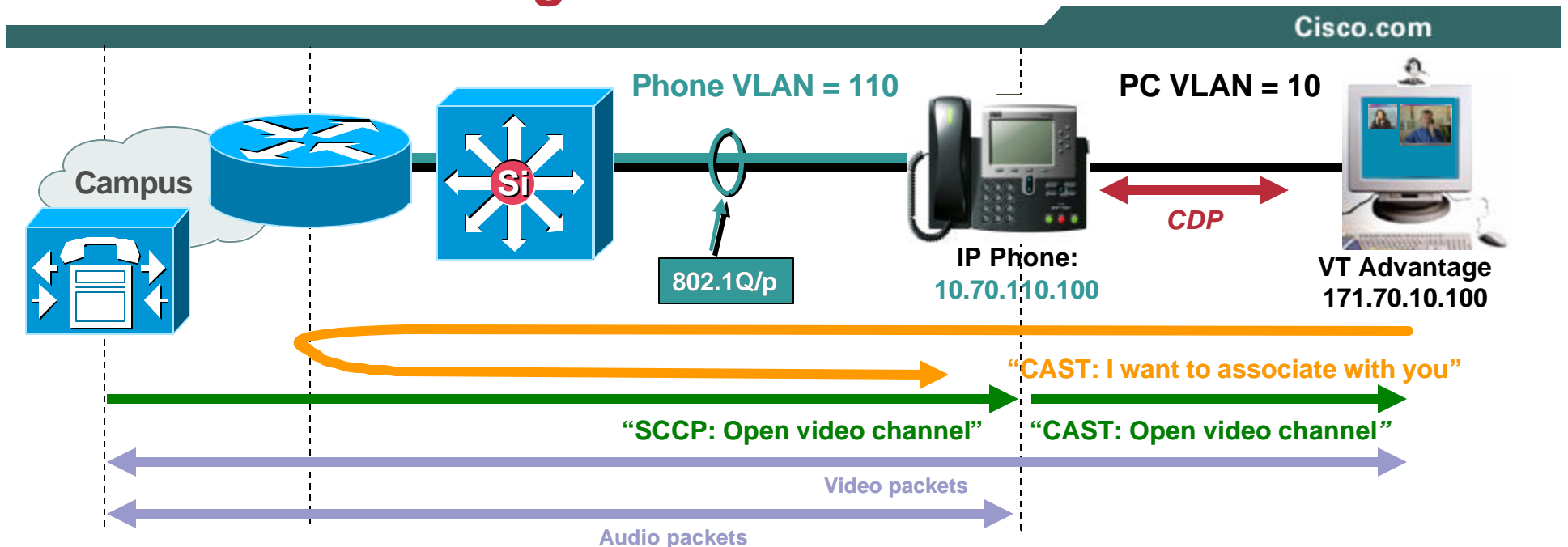
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- Supported on 7940/7960 firmware version 6.0(4) and 7970 firmware version 6.0(2)
- Video-Capabilities Enabled/Disabled per-phone in CallManager Administration
- VT Advantage automatically “associates” with IP Phone. All dialing and supplementary services done through phone
- CDP installed on PC Ethernet NIC. Must be physically connected to PC port on back of IP Phone (e.g. no wireless, no associating from a different network jack)
- Cisco USB Camera required (e.g. No 3<sup>rd</sup>-party cameras)
- H.263, WideBand Video Codec, G.729, G.711 and Wideband Audio Codec

# SCCP Endpoints

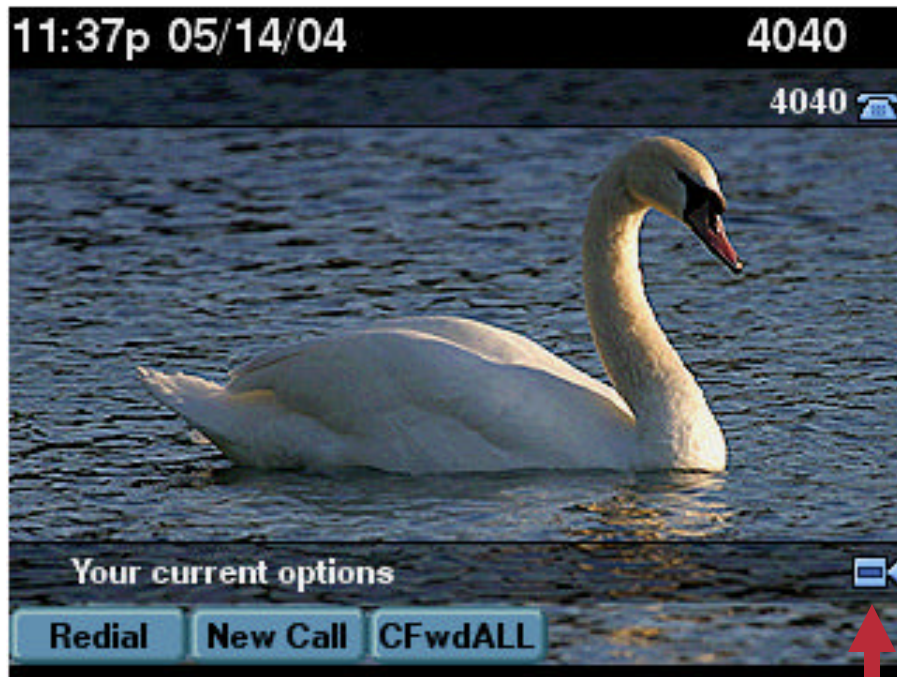
## How VT Advantage Works



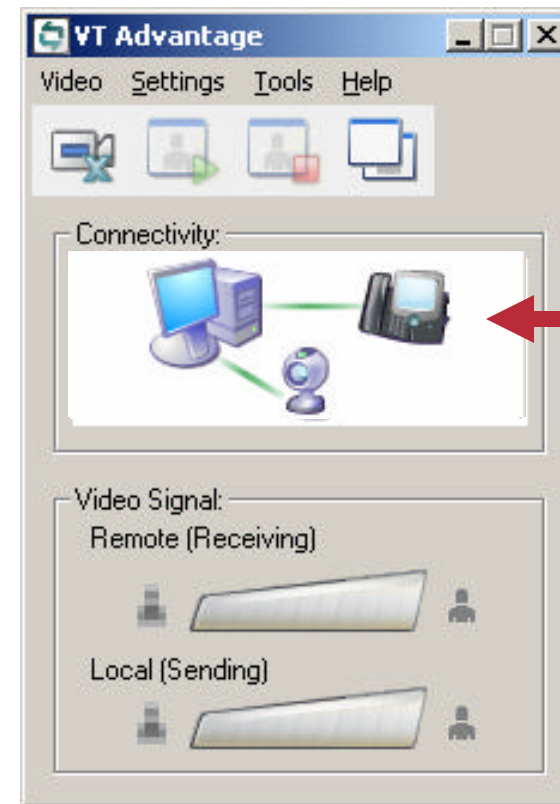
- 1 Phone and PC exchange CDP. Phone begins listening for CAST messages on TCP port 4224 from IP address of CDP neighbor
- 2 PC initiates CAST messages to phone over TCP/IP. CAST packets are routed up to layer-3 boundary between VLANs. Firewalls and/or ACLs must permit TCP port 4224
- 3 Phone acts as SCCP proxy between VT Advantage and CallManager. CallManager tells phone to open video channels per call. Phone proxies those messages to PC via CAST protocol
- 4 Phone sends/receives audio. PC sends/receives video. Audio and video marked DSCP AF41. Switch port must be set to trust DSCP (or use an ACL) instead of trust COS or else VT Advantage packets will be rewritten to DSCP 0

# SCCP Endpoints

## Cisco VT Advantage



- Camera icon on phone indicates that Video Capabilities are enabled

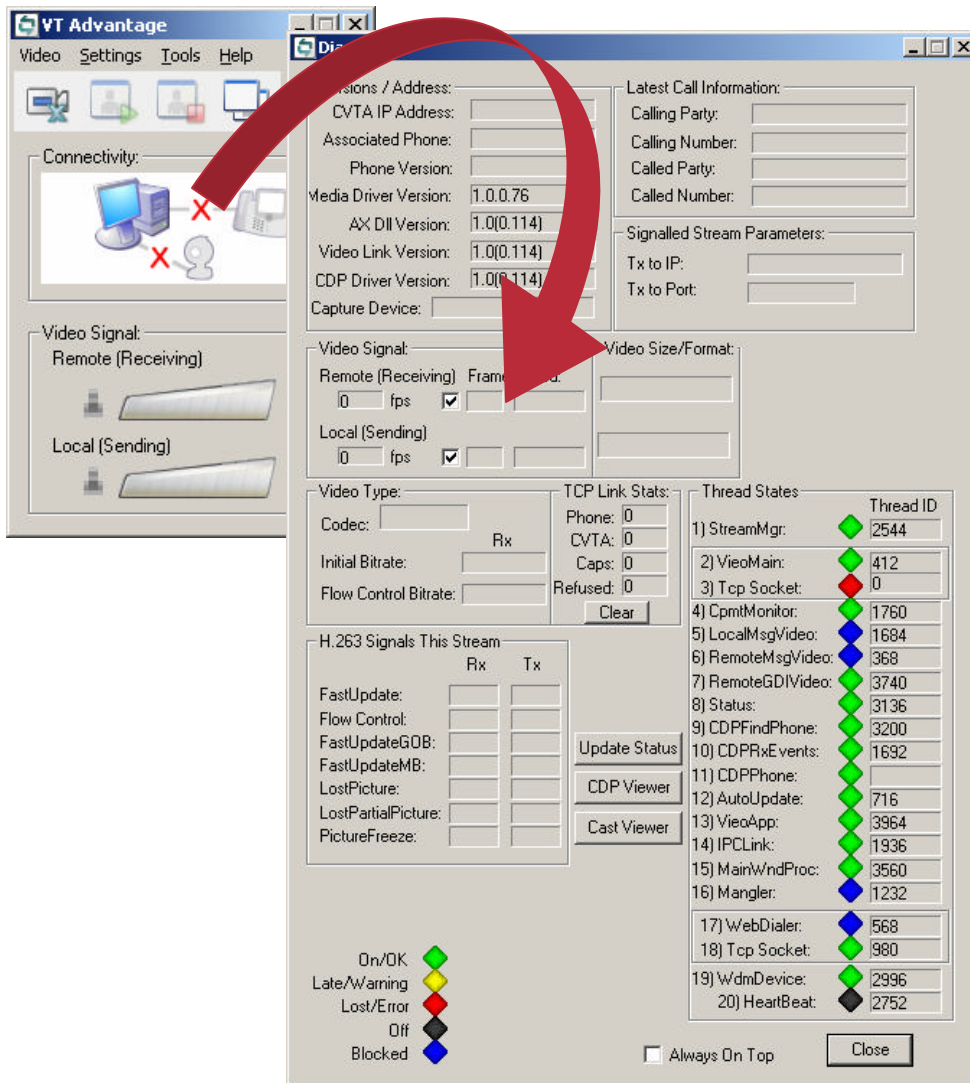


- VT Advantage user interface indicates status of association



# SCCP Endpoints

## Cisco VT Advantage



- **Double-right-click on VT Advantage user interface to view Diagnostics screen**

**Shows you detailed status of association, call status, CDP, and CAST protocol trace**

# H.323 Endpoints

## Tandberg Devices H.323 Release E3.X

Cisco.com



- Release E3.X supports CallManager in two ways:

Implemented Empty Capabilities Set support so that it can be placed on Hold, Transferred, Conferenced, etc.

Does not require a Gatekeeper. Can point straight to CallManager

- Supports Cisco IOS Gatekeeper Clustering (e.g. Alternate Gatekeeper)
- Configured in CallManager as an H.323 Client

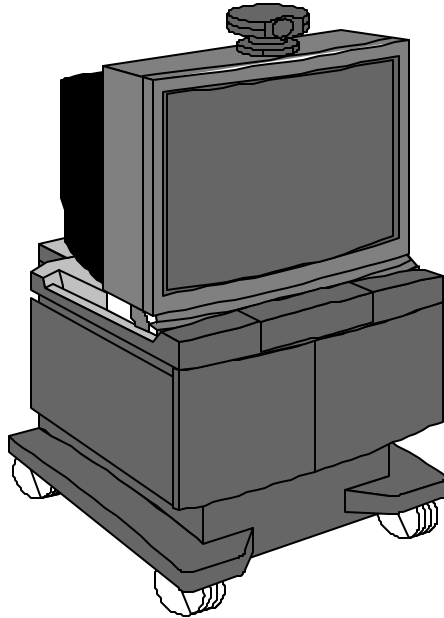
H.323 Clients are only allowed to make one call, so if built-in MCU functionality is required, configure it in CallManager as an H.323 Gateway instead

Uses port 1720

- H.261, H.263+, G.728, G.711, G.722 and Far-End Camera Control

# H.323 Endpoints

## Polycom, Sony et al



- **Pretty much any endpoint on the market will work, with certain caveats:**
  - Endpoint must support Empty Capabilities Set (ECS) in order to be placed on hold, transferred, conferenced or parked**
  - CallManager does NOT support some of the latest endpoint-specific non-standard or recently ratified features such as PictureTel iPower, Dual-Video, Encryption, etc.**
  - DTMF \*may\* NOT work because many H.323 devices pass DTMF in-band. CallManager uses out-of-band H.245 alpha-numeric DTMF**
  - H.261, H.263+, G.728, G.711, G.722 and Far-End Camera Control**
- **Working closely with Polycom, Tandberg and Sony to resolve the above caveats**
- **Endpoints can register to Gatekeeper or point directly to CallManager as a Gateway**

# H.323 Endpoint Integration

- **2 Methods:**

1. **Endpoint defined in CallManager, no GK requirement**

**Not widely available**

2. **Endpoint registers to gatekeeper, 2 options for dial plan and call routing:**

**Use default tech-prefix routing**

**—or—**

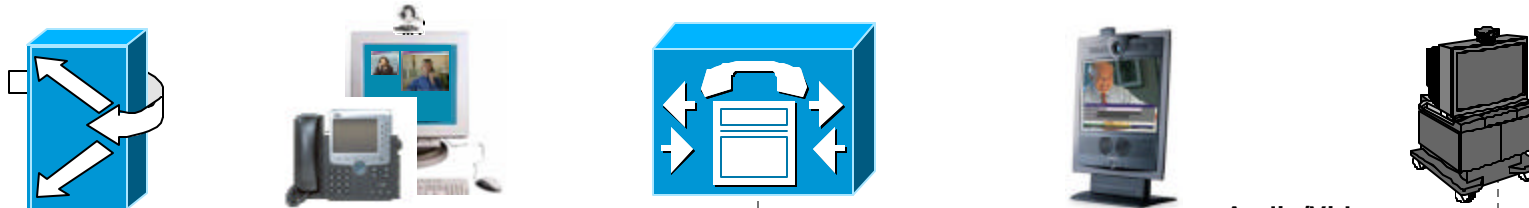
**Use Route-Patterns on CM**

**CM functions in H.245 routed mode, GK is direct mode**

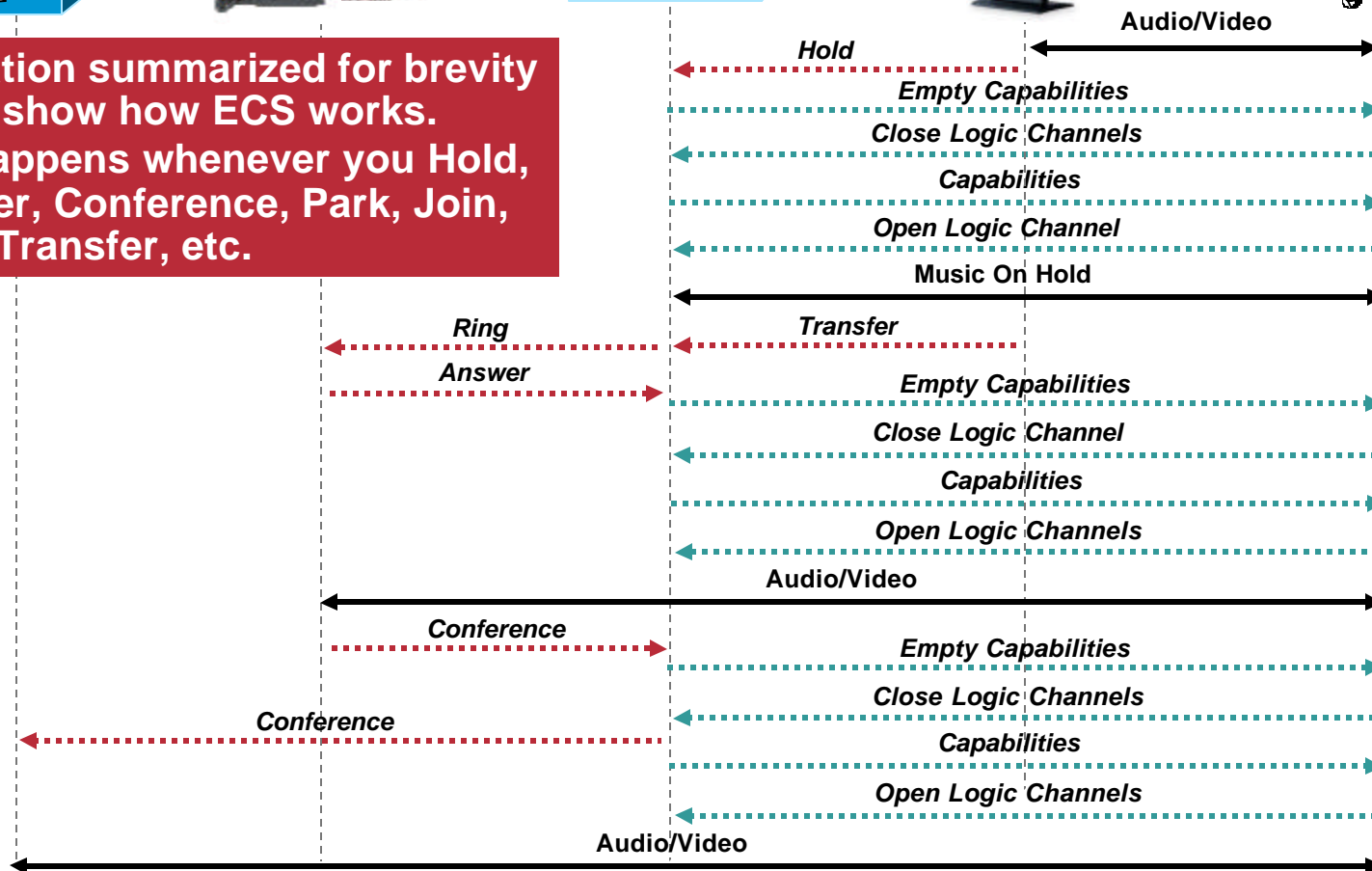
**(more on this shortly)**

# H.323 Endpoints

## Empty Capabilities Set



- Illustration summarized for brevity just to show how ECS works.
- ECS happens whenever you Hold, Transfer, Conference, Park, Join, Direct Transfer, etc.



# H.323 Endpoints

## Empty Capabilities Set

Note: Not a comprehensive list, just some that we've specifically tested against

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	Empty Caps Supported
All Tandberg Endpoint Models	Yes
Tandberg MCU/Gateway	No
Cisco IP/VC 3511/3540 MCUs	Yes
Cisco IP/VC 3510 MCUs	No
Cisco IP/VC 3526/3540 Gateways	Planned
Cisco IP/VC 3520/3525/3530 Gateways	No
Polycom VSX 3000/7000 →	Planned
Polycom ViewStation128 FX	No
Polycom ViaVideo, ViaVideo II	No, Planned
Polycom iPower Series (a.k.a. PictureTel)	No
Polycom MGC MCUs/GWs (a.k.a. Accord)	Planned
Sony PCS Endpoints	Planned
Microsoft NetMeeting	No
VCON ViGO	Yes

- Workaround for devices that don't support ECS to prevent them from being placed on Hold, Transferred, etc.
  1. Checkmark the "MTP Required" option for that device
  2. Assign it a Media Resource Group List that does NOT have any MTPs available in any of its MRGs
  3. Set the "Fail Call if MTP Allocation Fails" service parameter to FALSE
- Result = Softkeys will be disabled when you call that device

# H.323-Based Endpoints with Older ECS Implementations

- **No empty capability set support means that hold, park, transfer, conference, divert, or any other feature that breaks and reestablishes the media will not work**
- **Early implementations of empty capability set were overly restrictive; these implementations allow all of the above features to work except the call will be limited to whatever bandwidth the caller used to set the call up**
- **In some call scenarios after the feature is invoked the video may not be available or the bandwidth will be lower than it should be**

# H.323-Based Endpoints with More Recent ECS Implementations

- **More recent implementations of Empty Capability Set provide full support for these features**
- **To test an H.323 video endpoint to see which class it falls into do the following:**
  - Have an audio only SCCP phone call the endpoint (it is important that the audio phone initiates the call); hit transfer; call a SCCP video endpoint; hit transfer
  - If the transfer fails (wait for 20 seconds to make sure the call will stay up) then the H.323 endpoint does not support empty capability set
  - If the transfer succeeds but is audio only the endpoint supports the earlier style of implementation of empty capability set
  - If the transfer succeeds and has video then the endpoint has the more recent style of implementation
- **Make sure regions, locations and gatekeepers are not restricting the call**
- **Workaround for devices that don't support ECS to prevent them from being placed on Hold, Transferred, etc. :**
  1. Checkmark the "MTP Required" option for that device
  2. Assign it a Media Resource Group List that does NOT have any MTPs available in any of its MRGs
  3. Set the "Fail Call if MTP Allocation Fails" service parameter to FALSE

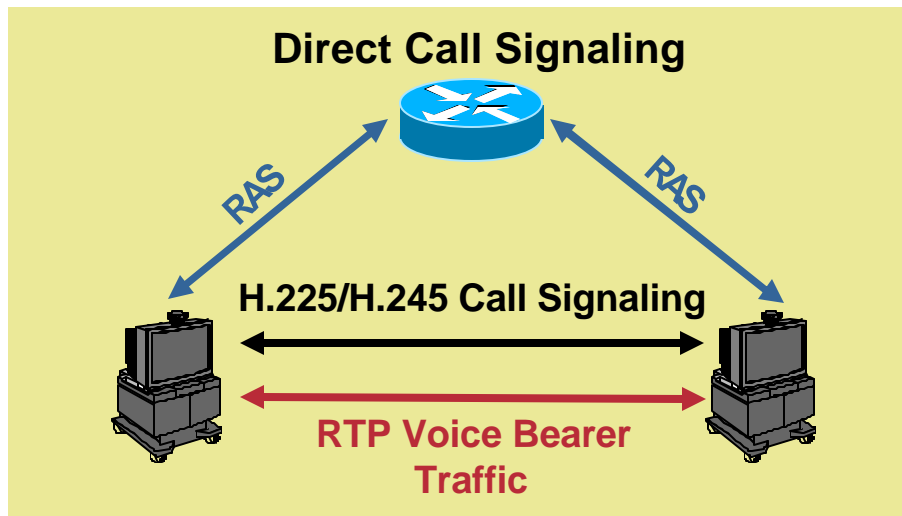


# Gatekeeper Integration

## Decision Points for Gatekeeper Implementations:

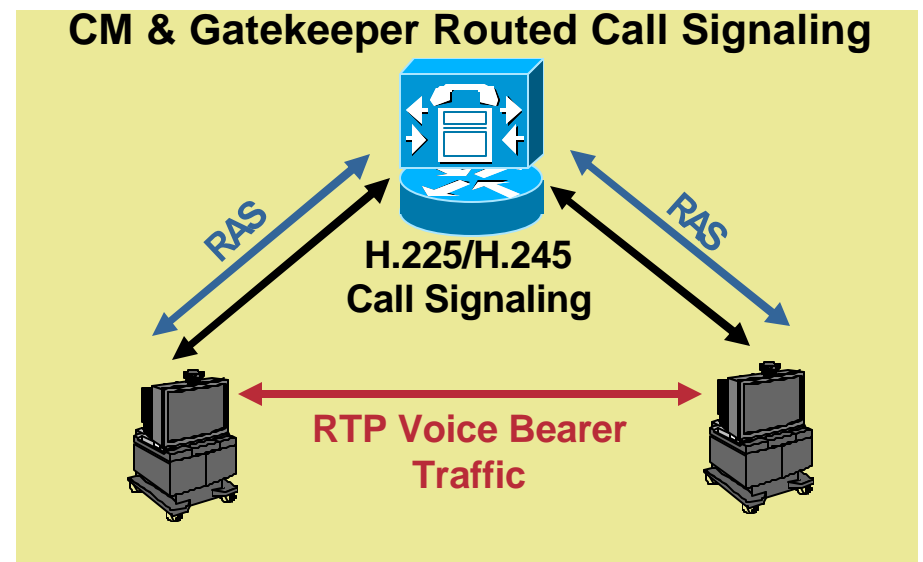
- **Utilizing Via-Zones?**
- **Redundancy**
  - HSRP or Alt-GK?
- **Scalability**
  - Load-sharing, directory gatekeepers?
- **Policy**
  - Dial plan restrictions between zones, device types
  - Is there any way that GK would resolve dialed digits without sending the call setup info to CM?
  - These decisions may lead to implementations of GK that require multiple sets of gatekeepers

# Direct and Gatekeeper Call Routed Signaling Models



- RAS signaling between H.323 device and G/K
- H.225 and H.245 signaling between gateways

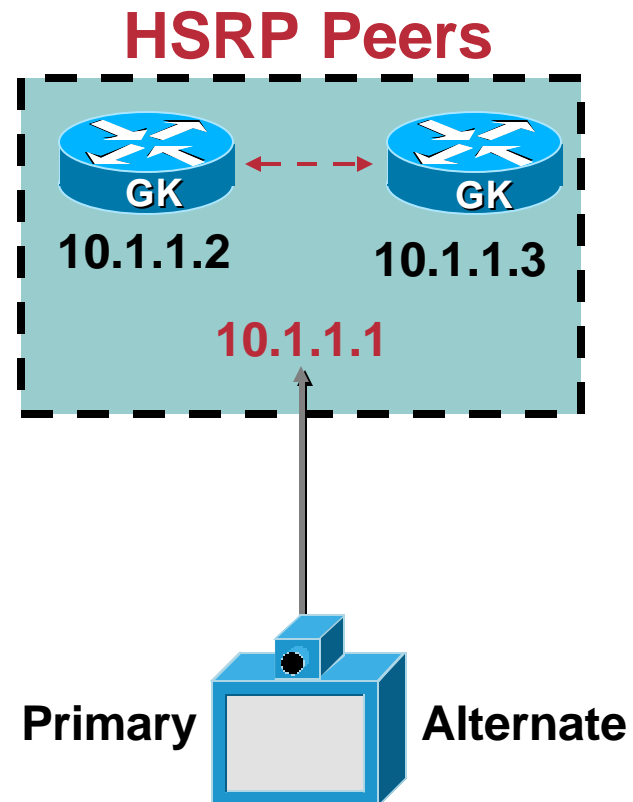
- RAS signaling between H.323 device and G/K
- When deploying CM and IOS GK, CM functions as a routed mode call processing platform
- **CM thus must support capabilities exchange parameters, such as codec, defined in call setup**



# Gatekeeper Redundancy HSRP

## Gatekeeper HSRP Characteristics

- Endpoints point to the HSRP virtual address
- Appears as one virtual Gatekeeper
- HSRP standby is “asleep” until primary fails\*
- Up to 3,000 endpoints per Gatekeeper
- HSRP peers must be in the same IP subnet



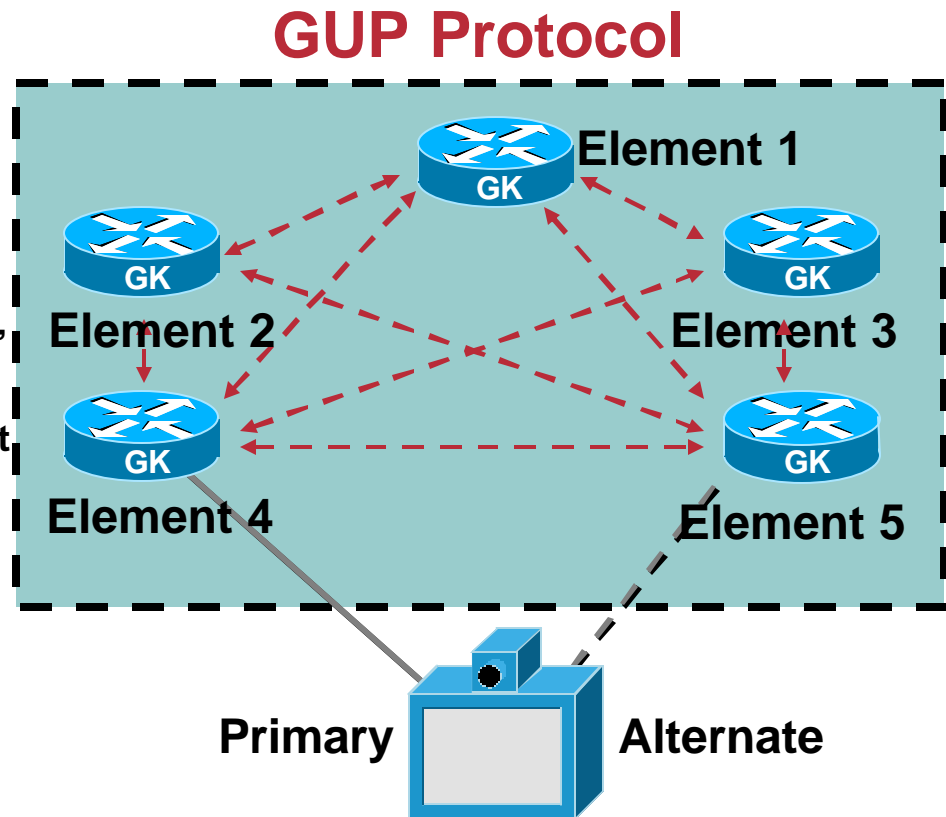
**Note: HSRP Standby Does Not Keep Active State**

# Gatekeeper Redundancy

## Gatekeeper Clustering

### Gatekeeper Cluster Characteristics

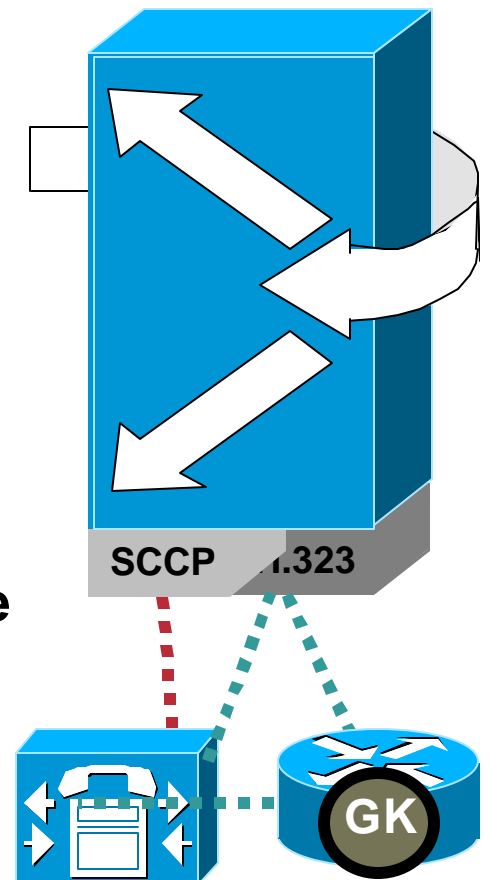
- Endpoints point to their primary gatekeeper, notified of the alternates during registration\*
- Up to 3,000 endpoints per gatekeeper
- Maximum of 7,500 endpoints in a cluster
- Maximum of five gatekeeper “Elements” in a cluster
- Gatekeeper elements may be in different subnets
- Load balancing supported
  - Memory and CPU utilization
  - Number of active calls
  - Registered endpoints



**Note: Not All H.323 Video Endpoints Support alt-GK at this Time**

# MCU Integration

- **SCCP - Ad-Hoc & Meet-Me video:**
  - Define the MCU in CM for Ad-hoc conferencing
  - Add meet-me numbers for Meet-me Conferencing
- **H.323 - Scheduled video:**
  - Relies on defining services on the MCU to define types of conferences
  - Define a route-pattern and assign it to the H.225 trunk
- **In all cases, assign the resources/DN's within the dial plan to the appropriate partitions, locations, etc.**
- **Consider using the <NULL> location for services such as the MCUs and gateways that are common for all users and are centralized**



# How Does the MCU Work with IP Video Telephony using SCCP?

- **New IP/VC software with SCCP provides MCU resource for Cisco CallManager 4.x**
- **Conferencing resources provides IP telephony-like functionality for videoconferencing**
- **Allows customers to create point-point calls then seamlessly expand into videoconference format**
- **These MCU resources are defined directly in CM, require additional dial plan configurations and meet-me number definitions**
- **Same telephony button setup experience:**
  - CONFR**—push, dial tone, add participants, repeat
  - MeetMe**—push, enter number, create or join conference

# How does the MCU Work with IP Video Telephony Using H.323?

- MCUs register to GK as a terminal type gateway
- Service prefixes configured on the MCU may or may not be part of the registration message
- If they are, there exists a possibility of H.323 endpoints being able to dial directly to the MCU service

To avoid this, there are several considerations:

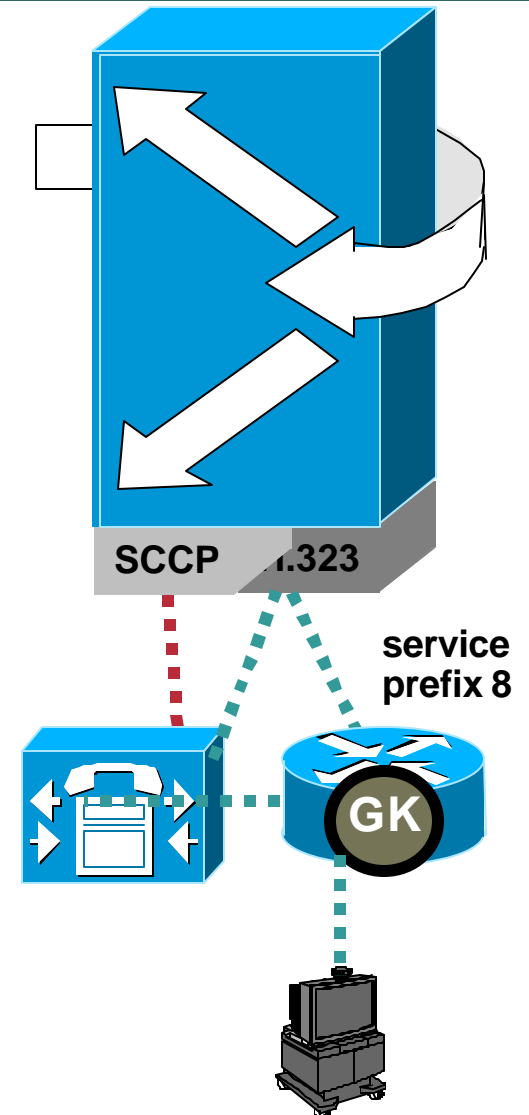
Do not allow for services to be registered

Register MCUs to one set of GKs, endpoints to another

Use translation patterns to mask the MCU service numbers from the dialed numbers the user is given

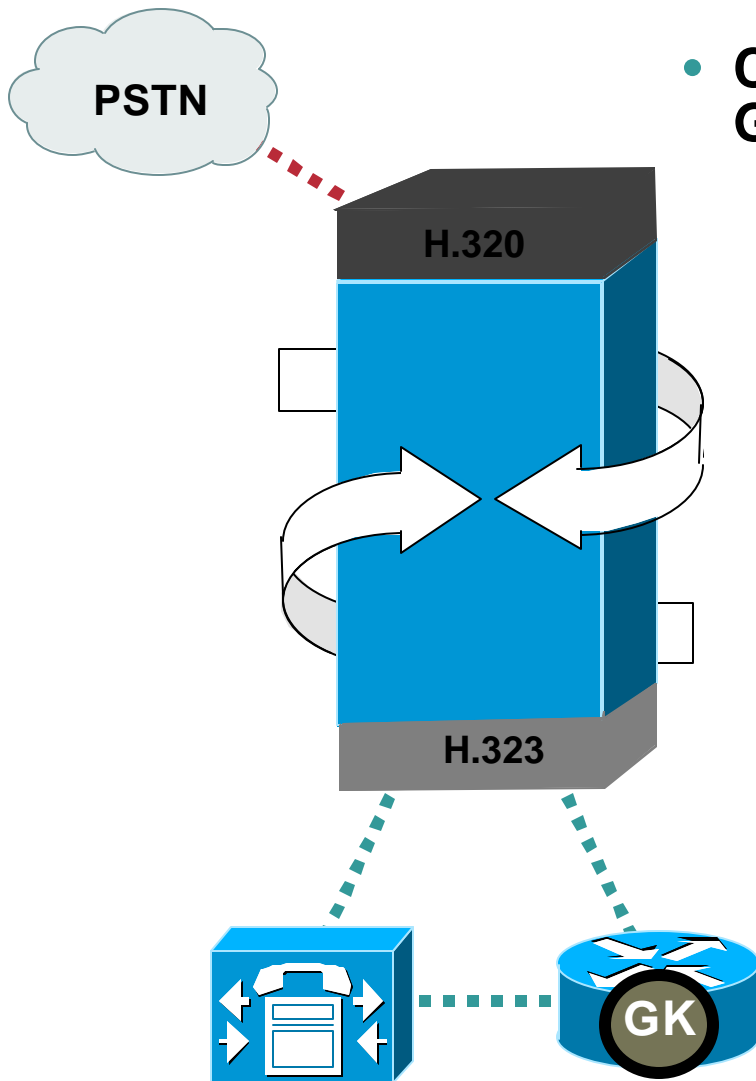
Use dial-out only services on the MCU

Use Via-Zones to force dial plan resolution to CallManager



# H.320 Gateways

## Cisco IP/VC 35xx Release 2.0



- **Configured in CallManager as an H.323 Gateway or reached via an H.225 Trunk**

**Can be added to a Route Group/Route List(s)**

**Gateways listen on TCP port 1820 by default. CallManager uses port 1720 by default. You can change either side**

**Direct Inward Dial (DID) and IVR modes supported**

**Gateway must register with Gatekeeper for inbound calls**

**Alternate Gatekeeper supported**

**Alternate Endpoints not supported**

**Empty Capabilities Set not supported**

**RAI/RAC supported for outbound calls if CallManager uses H.225 Trunk to Gatekeeper to reach the Gateway**



# Gateway Integration

**SCCP gateways - voice only**

**H.323 gateways for video -**

- **Typically use 8# for service prefix (9# for audio only gateways)**

- **2 options for implementation:**

**Define in CM**

**Allows for use of CSS, hunt groups, etc.—full mgmt by CM**

**-or-**

**Define route patterns and use GK to register gwys**

**Allows for use of RAI/RAC messaging for gwy congestion, but limited to providing CM mgmt to the trunk only (CSS, etc.)**

**-or-**

**Both**

**Implementation needs to be carefully designed such that a gateway with inadequate number of B channels available will not be used in hunt group**

# What about SIP?

- **CM4.1 supports SIP as a trunk protocol only - not as a line side protocol**
- **To incorporate SIP, use a SIP trunk from CM to the SIP call control platform**
- **Call Routing between platforms is across the trunk, utilizing the call control platform dial plans**
- **Supports Voice only - not video. Placing a video call results in an audio only call if 'Retry as Audio' enabled, else call is denied**

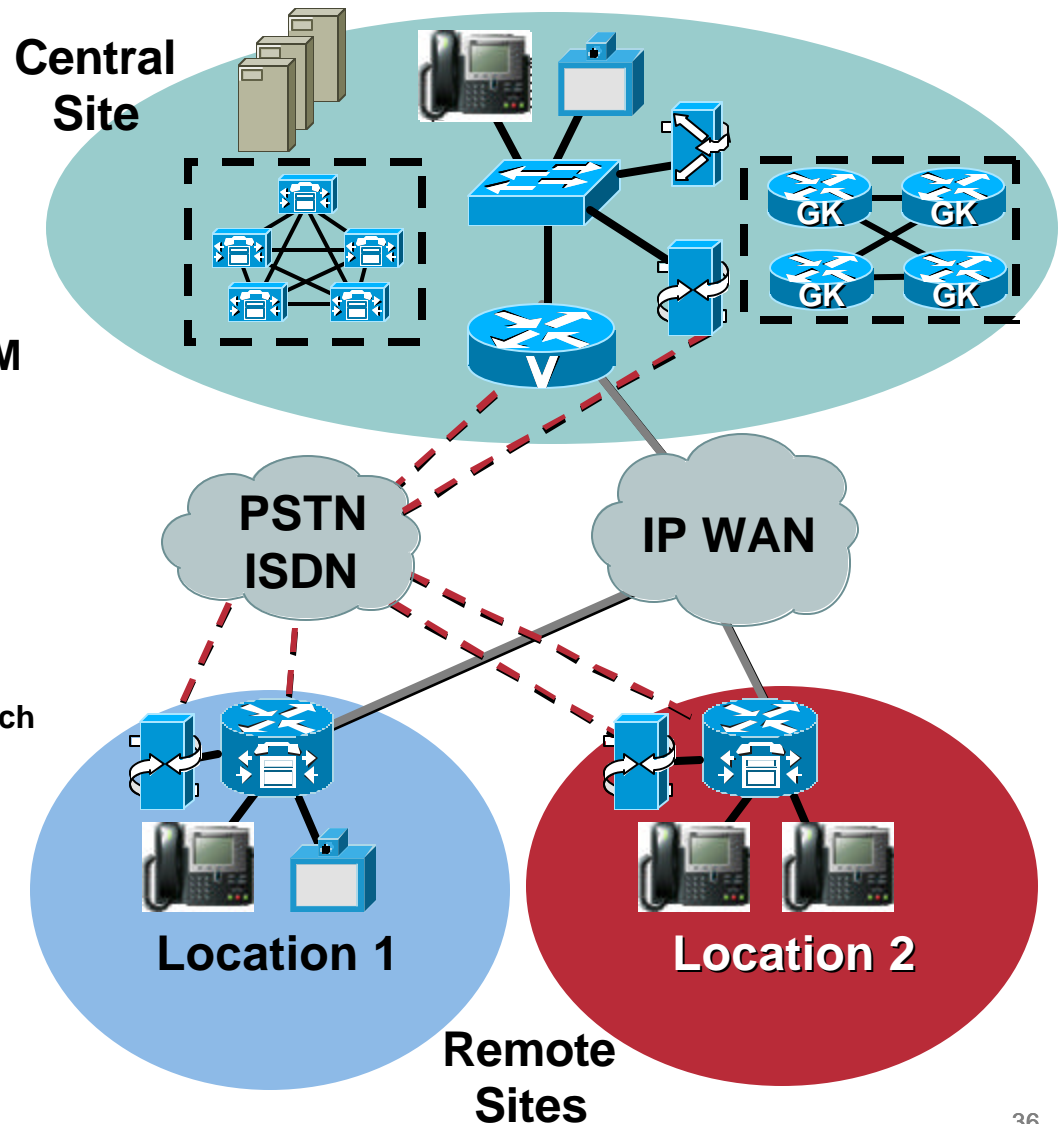
**For future releases of CM that support SIP line side protocols, an existing CM video telephony deployment will simply add SIP as a protocol, and support call routing between varying devices as it does for SCCP and H.323 today**

# Agenda

- Video Telephony Fundamentals
- Endpoint, MCU and Gateway Integration
- **Centralized Design and Deployment**
- Distributed Design and Deployment
- Configuration

# Centralized Call Processing AND Centralized Gatekeeper for H.323 Legacy and SCCP Video

- Cisco CallManager and GK located at single site, endpoints distributed
- H.323 endpoints register to GK or CM
- SCCP endpoints register to CM
- CM locations and region CAC
- Centralized or distributed gateways?  
DID or IVR?
- Dial plan design  
Call routing, partitions, calling search spaces



# Fundamentals - Dial Plan and QoS

- **SCCP Resources - endpoints, MCU's**
- **H.323 Endpoints**
- **QoS for video via ACLs**
- **External Devices**
  - H.323 MCU's - scheduled conferencing**
  - H.323 Gateways - PSTN/ISDN access, AAR**

# SCCP Dial Plan

**CM utilizes the same logic for class of service, call routing, etc., for video as it does for audio**

- **Shared Line Appearances**

**SCCP devices can share lines with full features**

**H.323 devices can share lines, but loss of features such as hold, etc, which require specific interface features not found on H.323 devices**

- **Call Forwarding**

**Across regions can be difficult, depending on H.245 and H.225 caps exchanges**

**H.323 devices must be on but not answered, or CF fails**

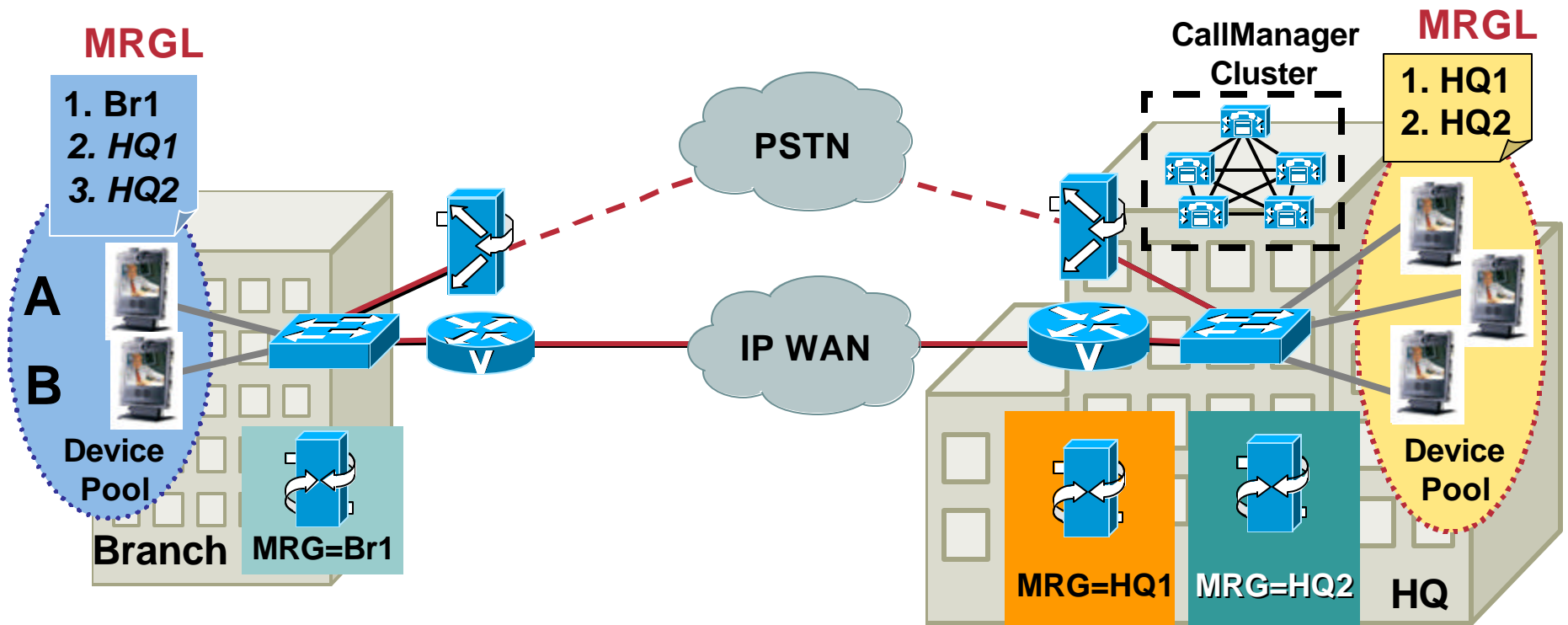
- **Hunt Groups**

**SCCP and H.323 devices can be in a hunt group**

**If H323 device is in hunt, but is OFF, hunt terminates - use 'broadcast' to avoid this condition**

# SCCP Media Resources

## Distributed Conferencing Resources



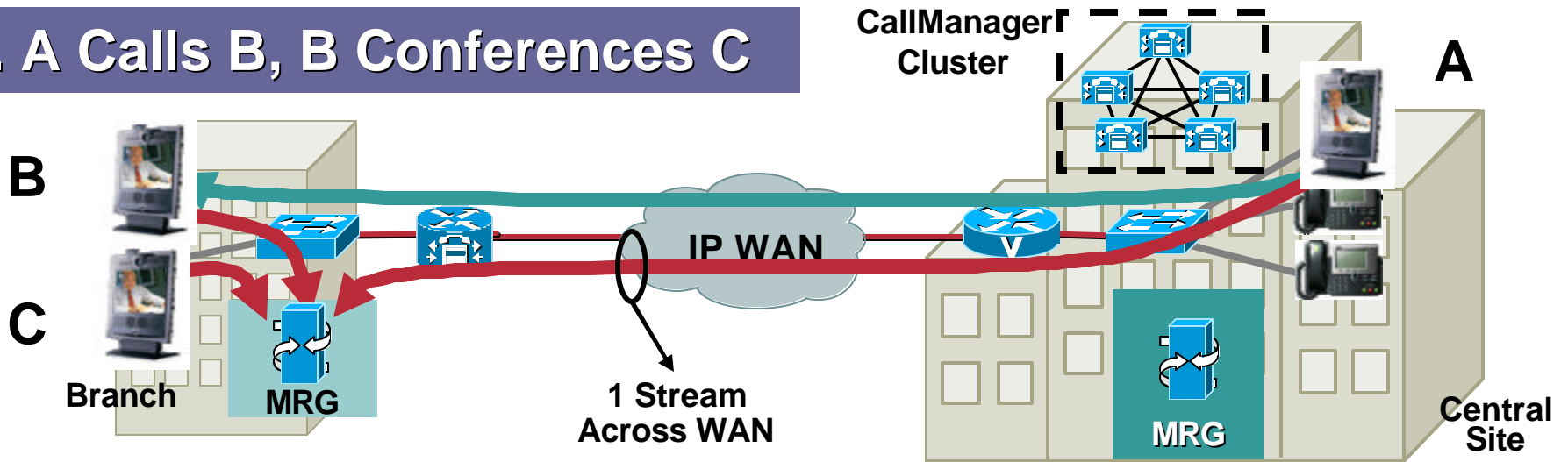
- Conference between A, B —no video across WAN
- MCU, Gateway resources at branch
- **Transcoding/Transrating resources are 'owned' and managed by the MCU**
- **No conferencing during WAN failures**

**MRG = Media Resource Group**  
**MRGL = Media Resource Group List**

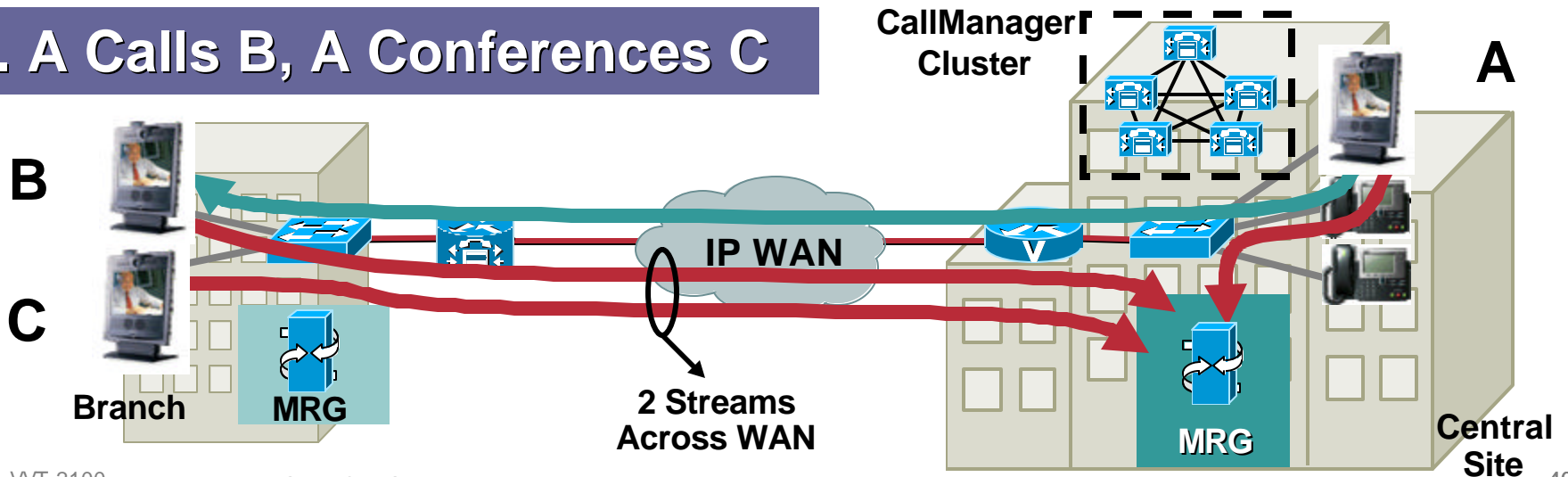
# SCCP Media Resources “Conference Initiator” Concept

Overprovision WAN  
to Allow for This...

## 1. A Calls B, B Conferences C



## 2. A Calls B, A Conferences C



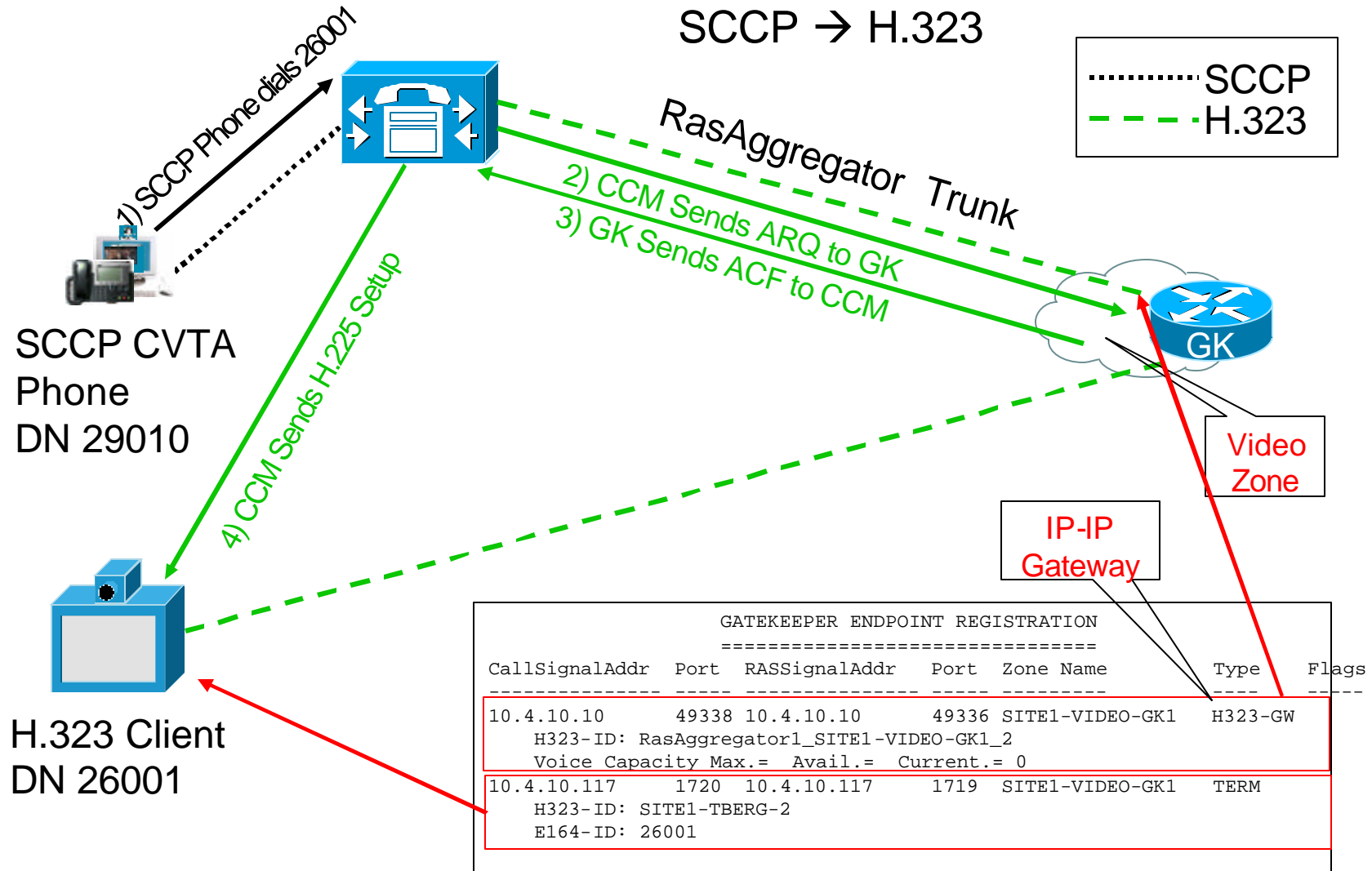


# Deploying GK for H.323 Endpoints

- **In CM4.0, H.323 endpoints are identified by IP address  
H.323 trunk can not support DNS or E.164 currently for endpoint definitions**
- **In CM 4.1, the IOS GK utilizes the RAS aggregator trunk to define the GK, and endpoints are defined by E.164 address**
- **With CM 4.1 registered as IP-IP gateway and via-zone configuration on GK, all dial plan resolution is forced to CM**

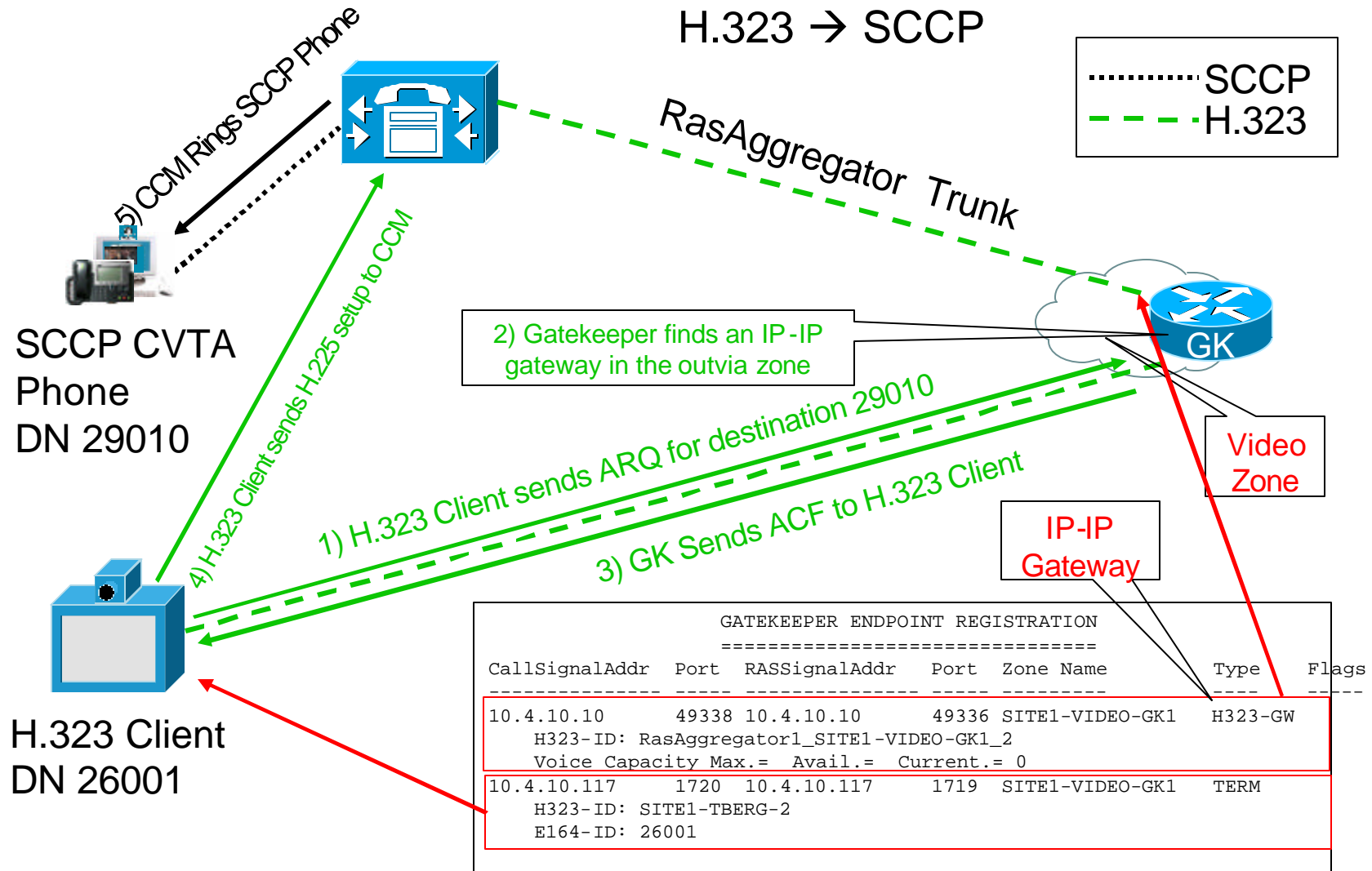
# Dynamic H.323 Addressing

## Call Routing SCCP → H.323



# Dynamic H.323 Addressing

## Call Routing H.323 → SCCP



# GK Deployment Options

- **Design GK deployment for redundancy, scalability, and call routing via CM**
- **CM and IP/VC support Alt-GK, while some endpoints do not, requiring the use of HSRP**
- **To ensure that endpoints can not dial directly to MCU or Gateway resources, but need to route to CM, requires a separate GK deployment OR the use of via-zones**
  - Simply putting MCU and gateways into a separate zone will not block call routing between endpoints and the conferences or gateways
  - Inbound gateway calls would be forced to CM
- **One design option utilizes one set of gatekeepers for endpoints utilizing HSRP, and another set of GKs for MCUs and gateways utilizing Alt-GK**
  - CM has two trunks—one to each GK
- **The use of Via-Zones resolves multiple issues, including mobility and call routing**

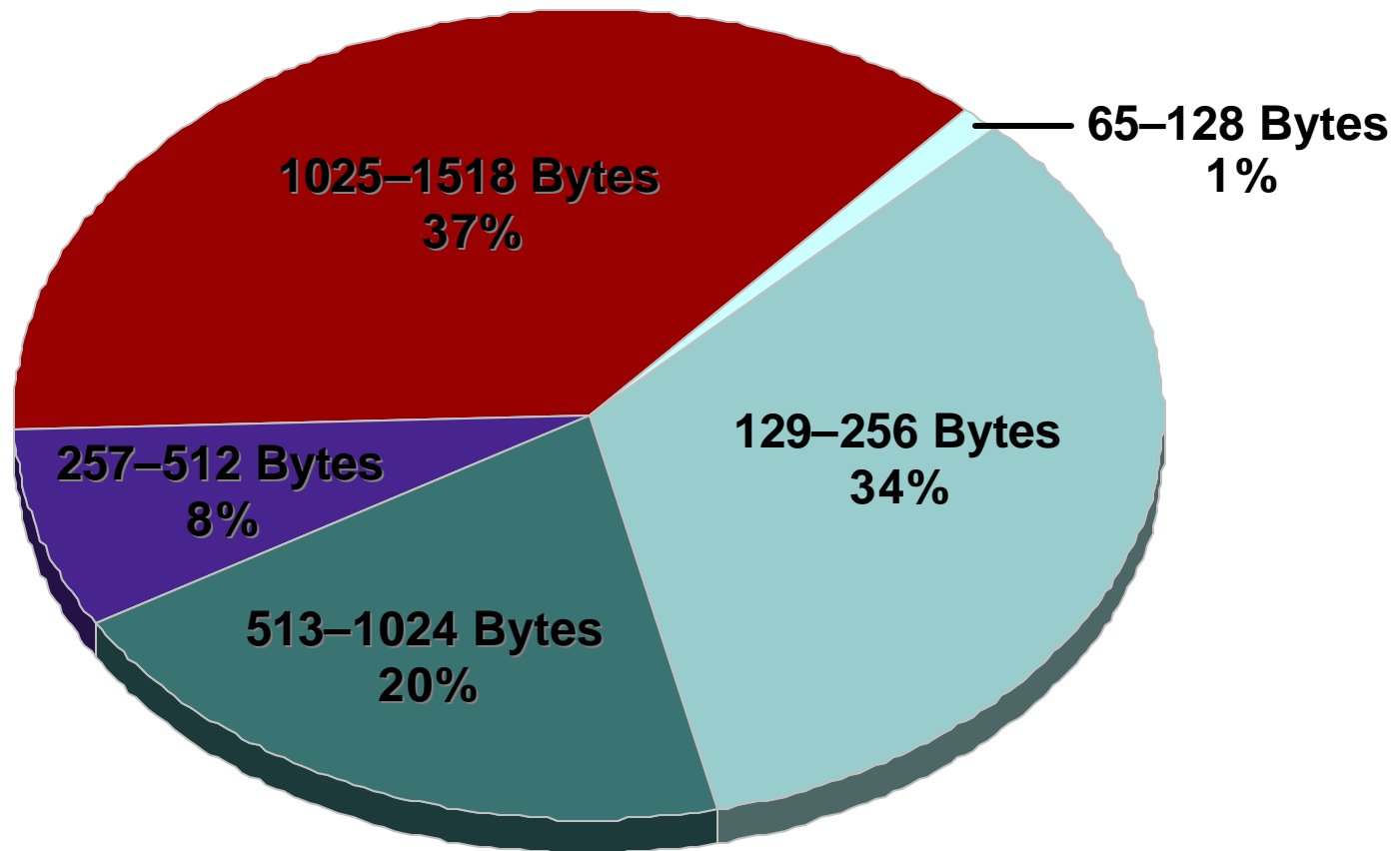
# Video QoS & CAC

**To deploy QoS for video, utilize Low-Latency Queueing configurations where video is placed in a separate PQ, or in a CBWFQ, and the audio channel of a video call is placed in the same class as the video channel**

**Use CM Regions and Locations for inter-cluster CAC:**

- **Audio is represented as bit-rate + overhead (i.e. 24k for G.729, 80k for G.711)**
- **Video is represented as bit-rate only (i.e. 384k for a 384k call) and includes the audio portion**

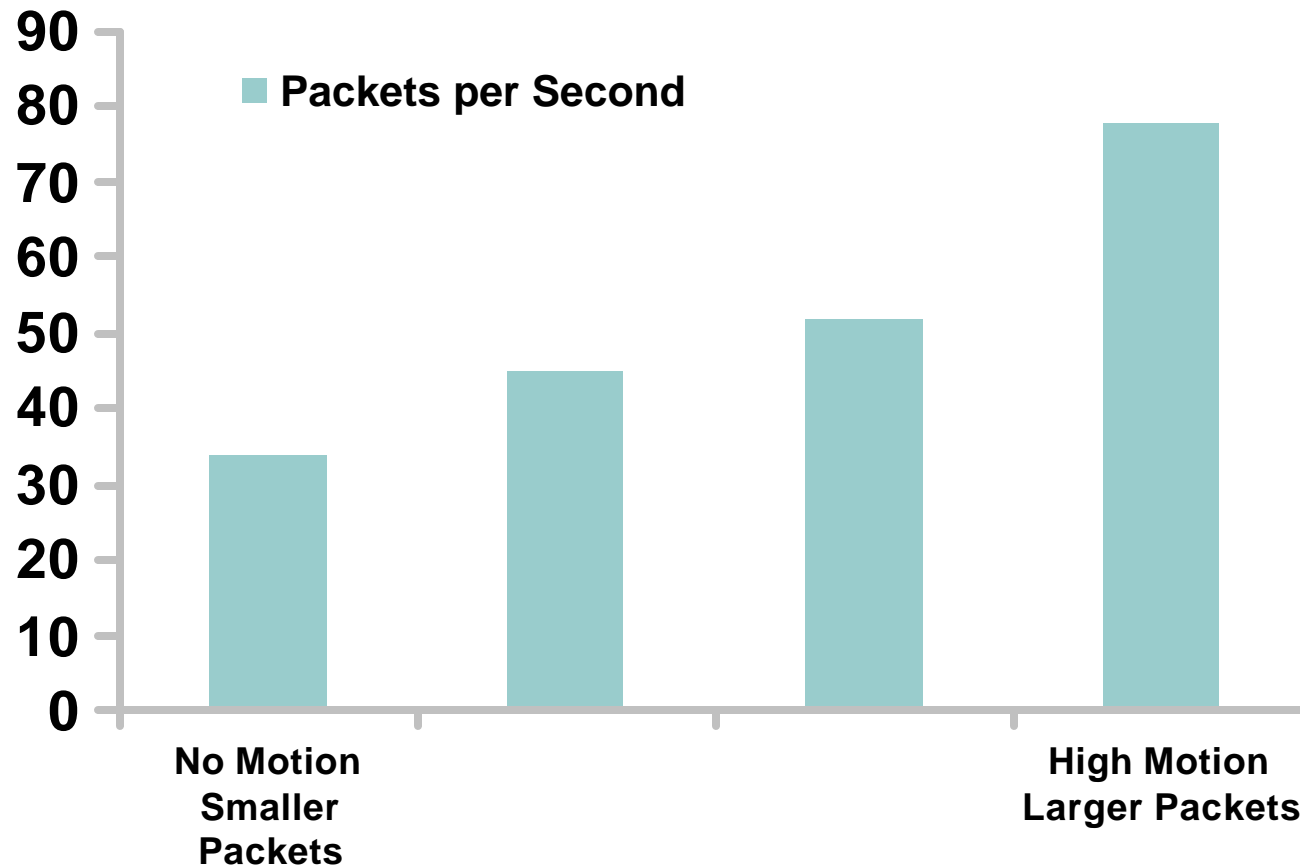
# Video Conferencing Traffic Packet Size Breakdown (CIF)



**384kbps Video call = 320kbps video, 64kbps audio  
With overhead (approx 20%) equates to 420kbps**

# Video Characteristics

## 384kbps Video Call



**Due to the nature of the encoding algorithm,  
video bearer traffic is bursty**

# Recommended QoS Traffic Classifications

L2 CoS	L3 Classification			Application
	IP Prec.	PHB	DSCP	
7	7	-	56-63	Reserved
6	6	-	48-55	Reserved
5	5	EF	46	Voice Media
4	4	AF41	34	Videoconferencing*
3	3	CS3	24	Voice Signaling**
2	2	AF2y	18,20,22	High Priority Data
1	1	AF1y	10,14,16	Medium Priority Data
0	0	BE	0	Best Effort Data

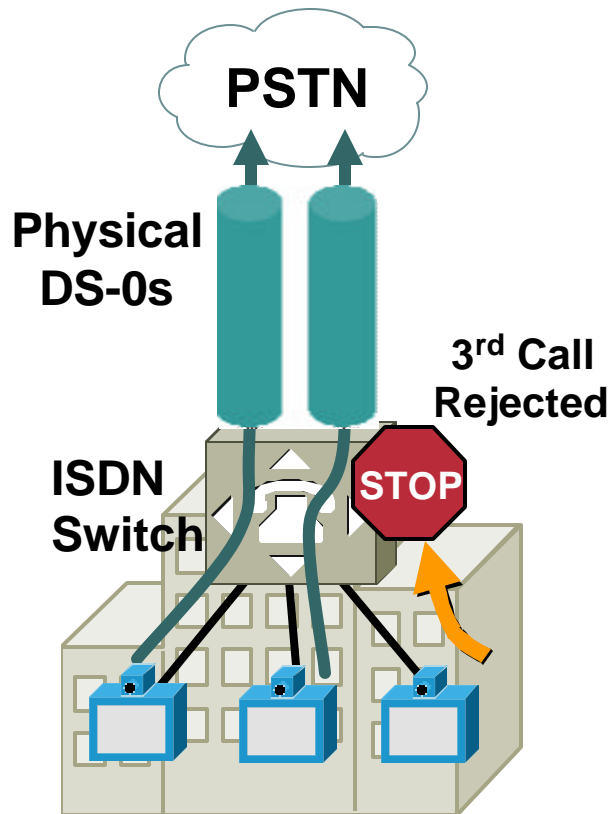
\*Including Audio and Signaling for H.323

\*\* Including Signaling for SCCP Video

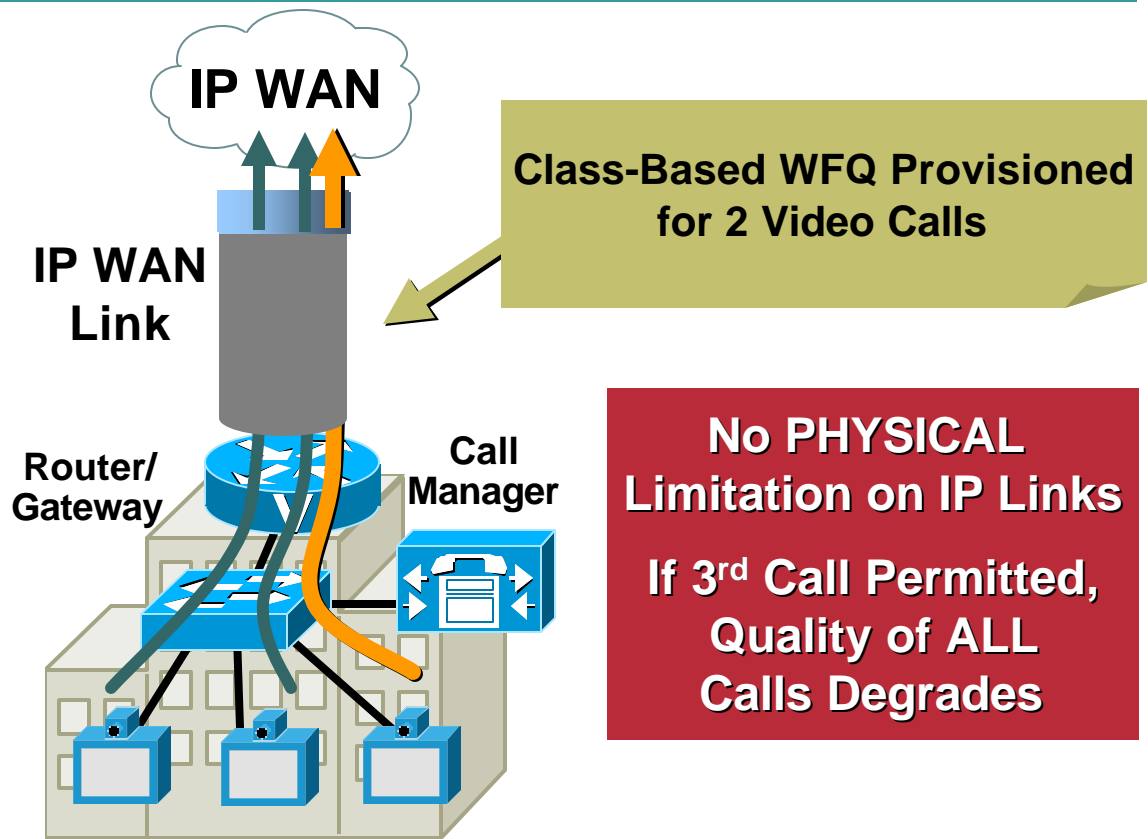


# Call Admission Control

## Circuit-Switched Networks



## Packet-Switched Networks



**No PHYSICAL  
Limitation on IP Links**  
**If 3rd Call Permitted,  
Quality of ALL  
Calls Degrades**

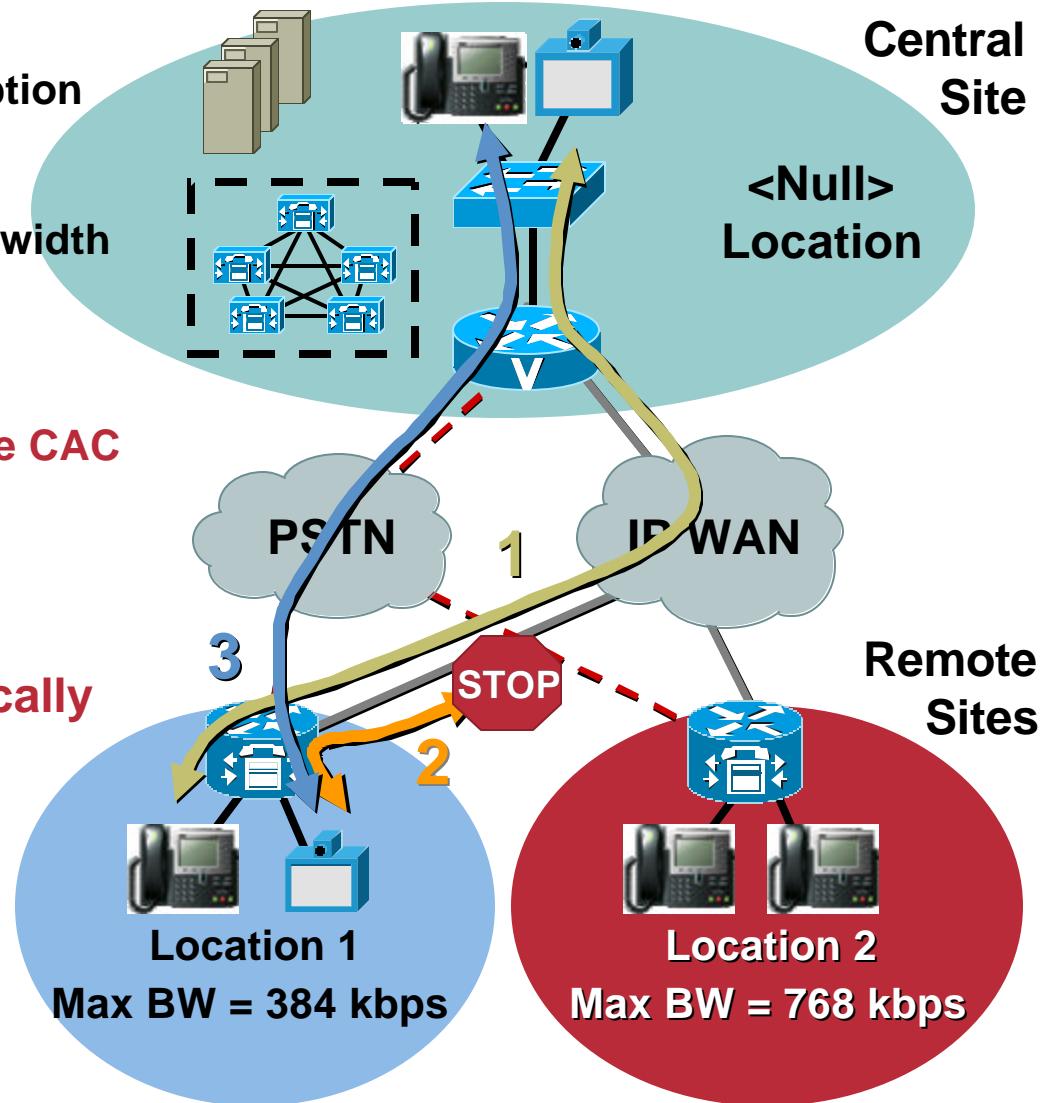
**Call Adm. Control Limits # of Calls Between Sites**

# Cisco CallManager CAC Within a Cluster

- Prevent WAN link over-subscription by limiting voice and video bandwidth
- Use **regions** to set per call bandwidth limit
- Assign aggregate bandwidth limit **per location**
- Not like H.323—how do I provide CAC for call transfer, etc.?????

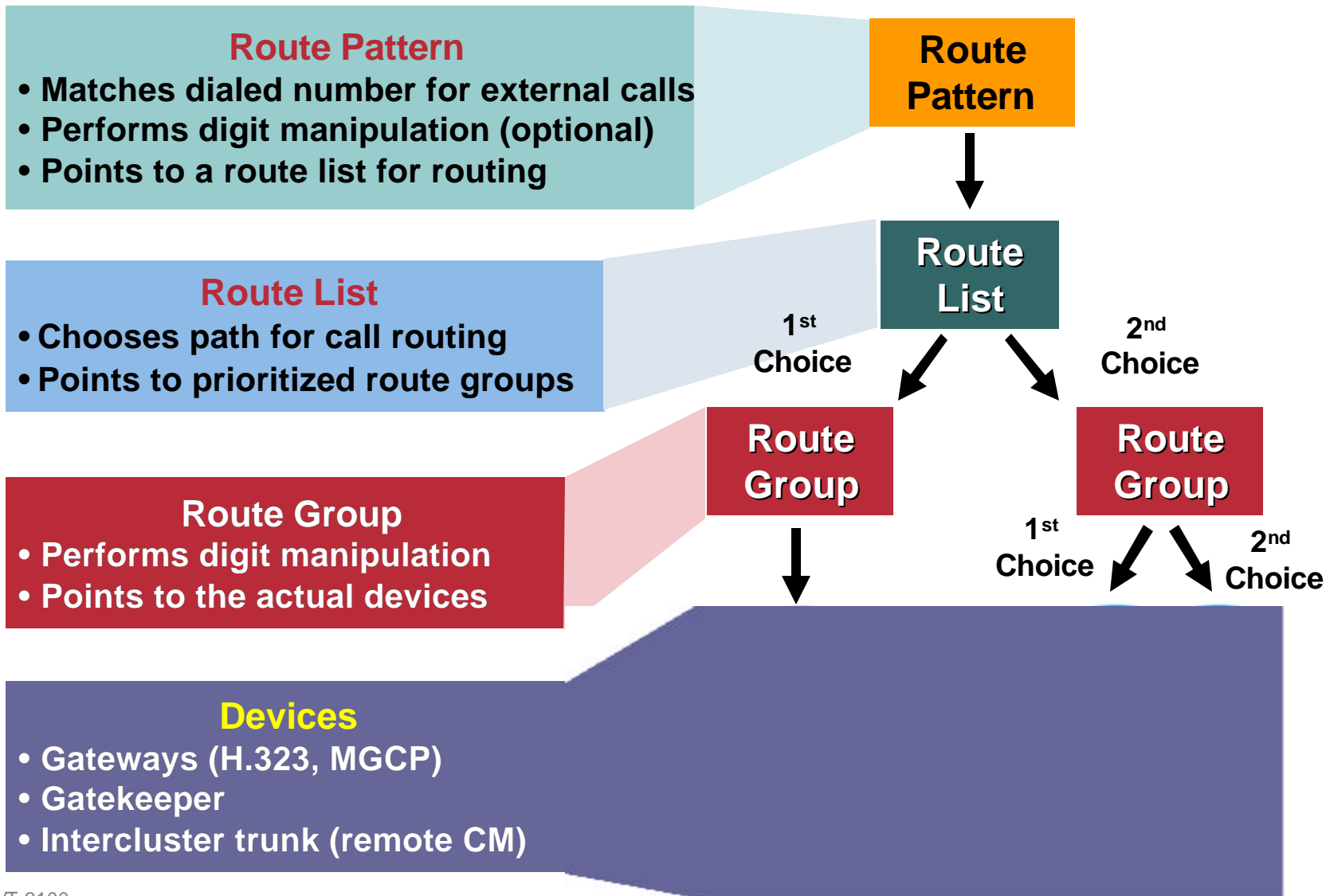
- **When resources are insufficient, call is automatically prefixed to auto-alternate route via gateway/PSTN—Works for voice and video\* calls**

**\*When IP/VC GW Is Present; Without Video Gwy, Reroute as Audio Only**



# Dial Plan

## External Route Elements in CallManager



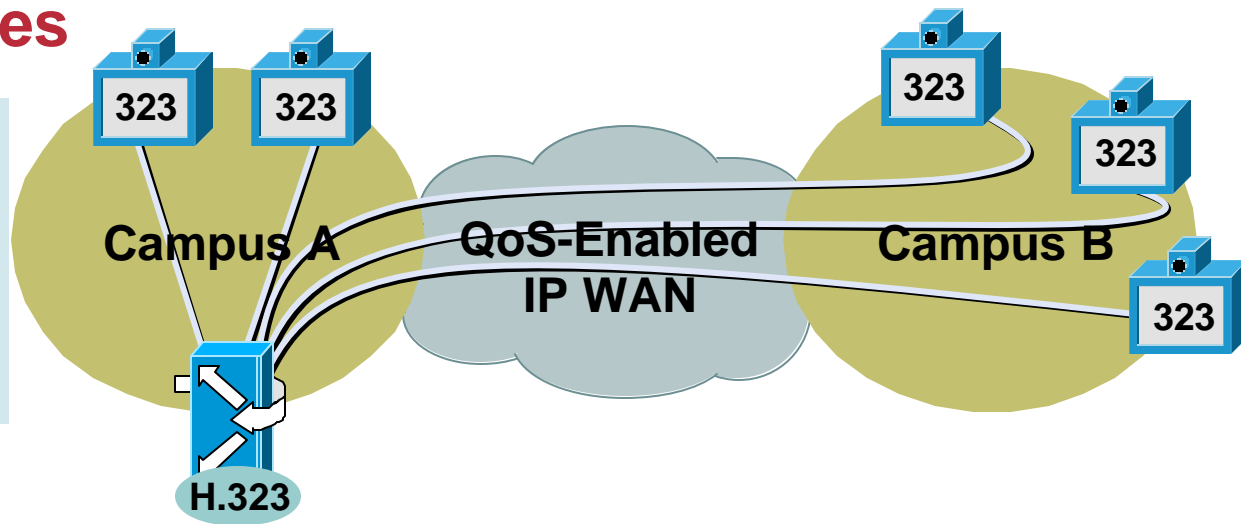
# Dial Plan Design

- **Deploying gateways**
  - H.323 gateways for video, registered to gatekeeper
- **For inbound calls, DID or IVR? Distributed or centralized?**
- **Gatekeepers requires H.225 trunk, route pattern for outbound calls**
- **Assign voice and video gateways to separate partitions and use a common prefix (i.e., 9) or use separate prefixes (9 for voice, 8 for video)**
- **Digit stripping for inbound calls**
- **Deploying AAR**
  - How do the dialed digits get altered to allow for PSTN/ISDN access?  
H.323
  - H.245 routed mode—No use of CanMapAlias required!**

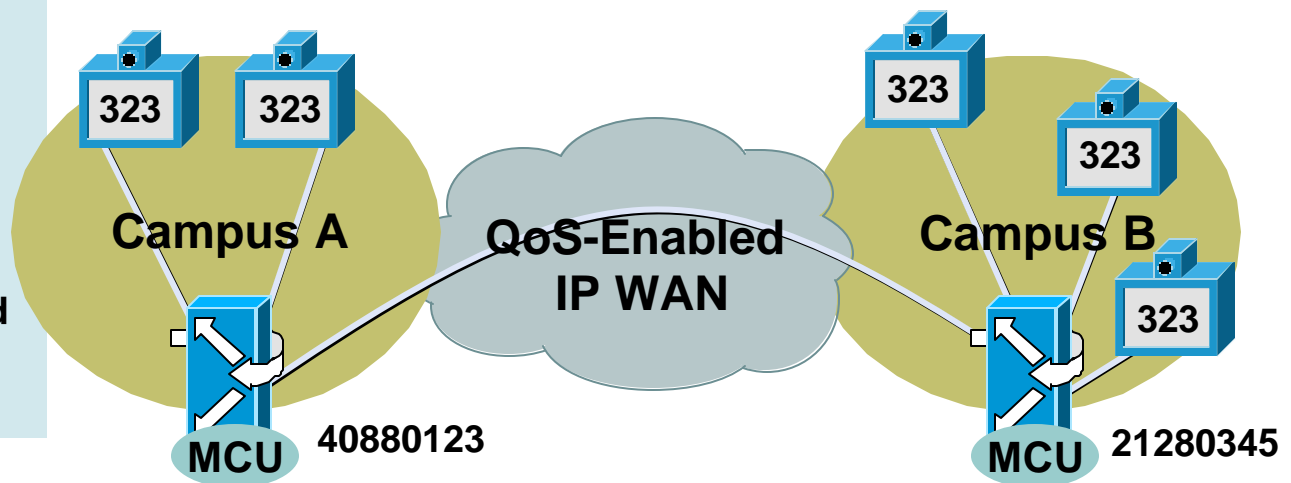
# H.323 MCU Resources

## Bandwidth Issues

- Low-speed WAN Links
- Remote sites are limited to the amount of bandwidth provisioned for videoconferencing



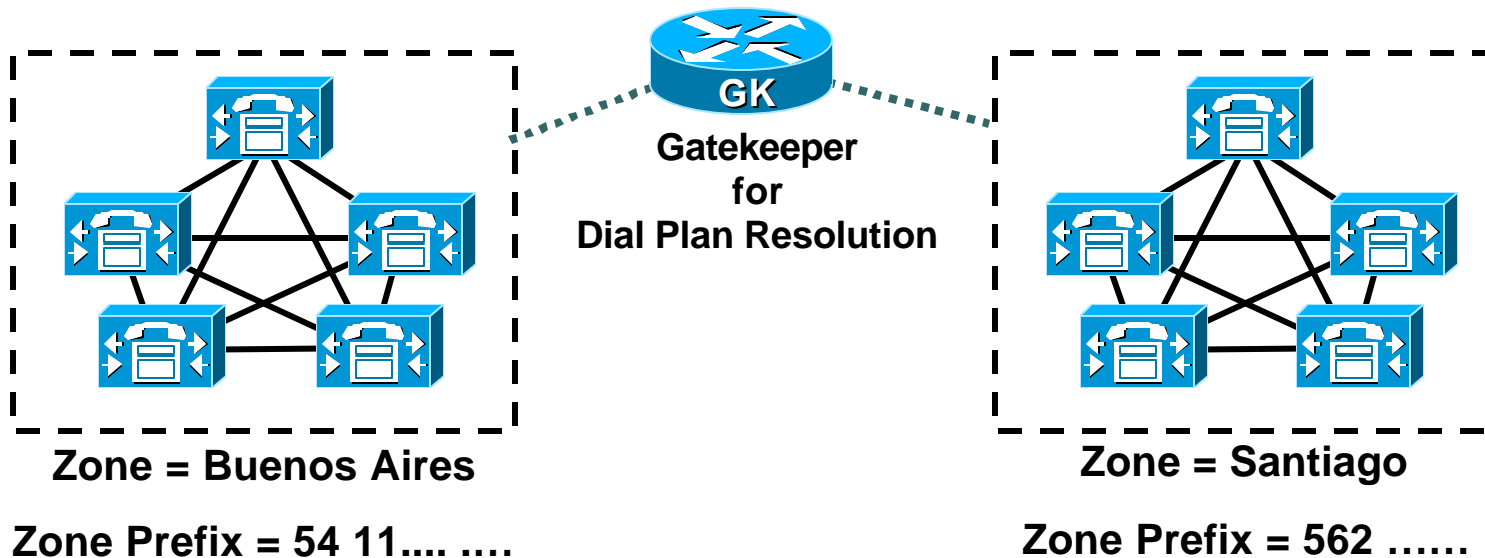
- Distribute MCUs to large sites
- Video terminals use their local MCU
- Single site multipoint calls stay local
- WAN bandwidth is limited to one call at the conference bandwidth



# Agenda

- **Video Telephony Fundamentals**
- **Endpoint, MCU and Gateway Integration**
- **Centralized Design and Deployment**
- **Distributed Design and Deployment**
- **Configuration**

# Intercluster Trunk Call Routing Using GK



- Cisco CallManager registers with gatekeeper and has H.323 “Trunk” configured as a route to/from the gatekeeper
- Up to 100 local zones per gatekeeper, one Cisco CallManager cluster per zone
- Gatekeeper is configured with prefix for call routing between multiple CallManager clusters
- Configure GK with zone prefixes to match dial plan of each cluster, utilizing simple, consolidated dial plan

# Cisco CallManager CAC Between Clusters



- Gatekeeper provides Call Admission Control (CAC) between Cisco CallManager clusters
- Cisco CallManager registers with gatekeeper and has H.323 “Trunk” configured as a route to/from the gatekeeper
- Up to 100 local zones per gatekeeper, one Cisco CallManager cluster per zone
- If gatekeeper rejects call request, Cisco CallManager can auto-alternate route via route lists/route groups
- Utilizes Alt-Gk for resiliency



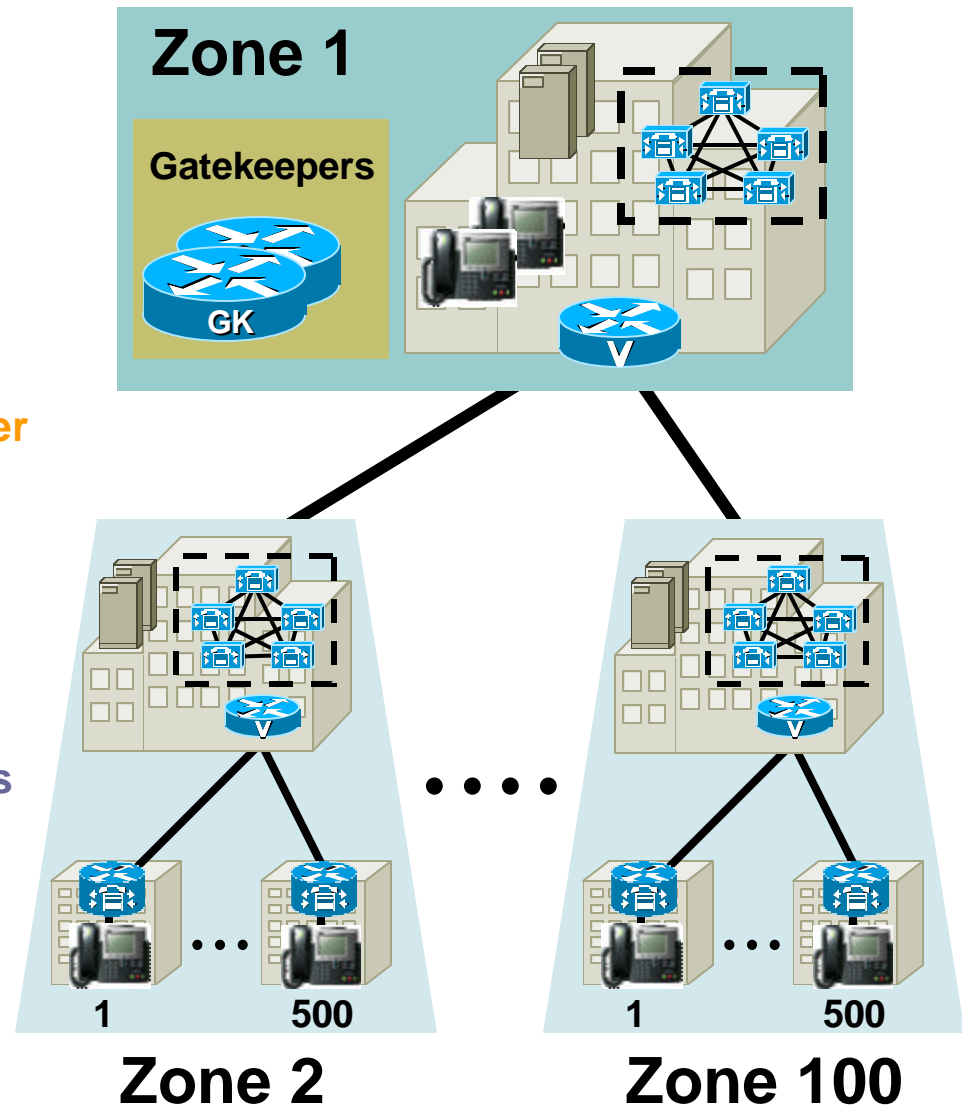
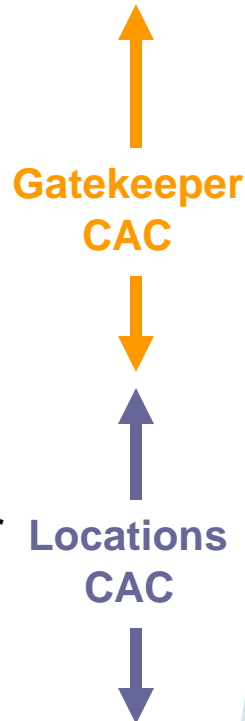
# Combined (Two-Tier) CAC Within and Between Clusters

- Hub-and-spoke topology
- Gatekeeper CAC used between regional sites

1 CallManager cluster per zone  
100 local zones per Gatekeeper

- Locations CAC used for branch sites

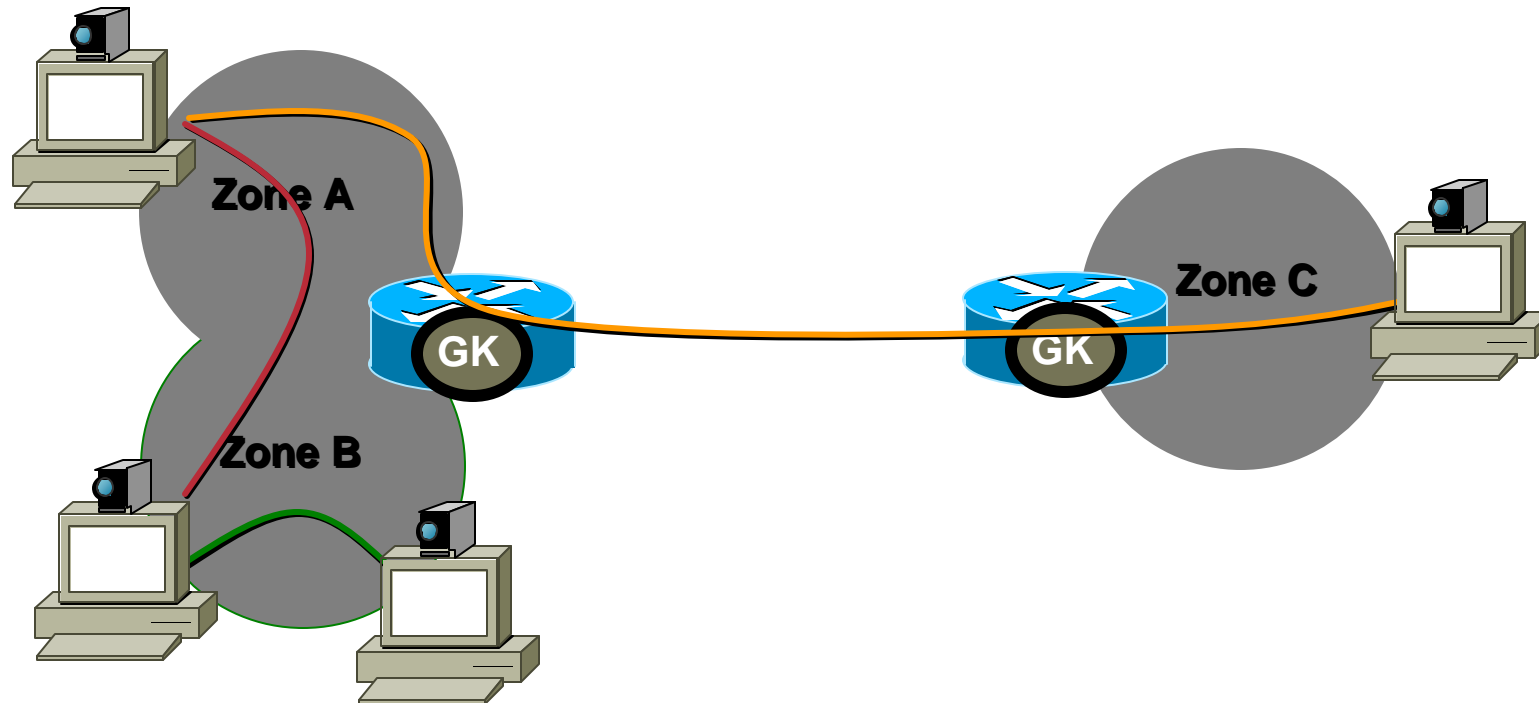
500 locations per cluster



# Call Admission Control

## Gatekeeper Bandwidth Commands Explained

Cisco.com



1. **Interzone** = Bandwidth of all calls for a local zone to/from all other zones
2. **Remote** = Aggregate bandwidth of all local zone(s) to/from any remote zones
3. **Total** = Bandwidth of all calls within an individual zone
4. **Session** = Bandwidth allowed on a per call basis

# Call Admission Control

## What Values to Use

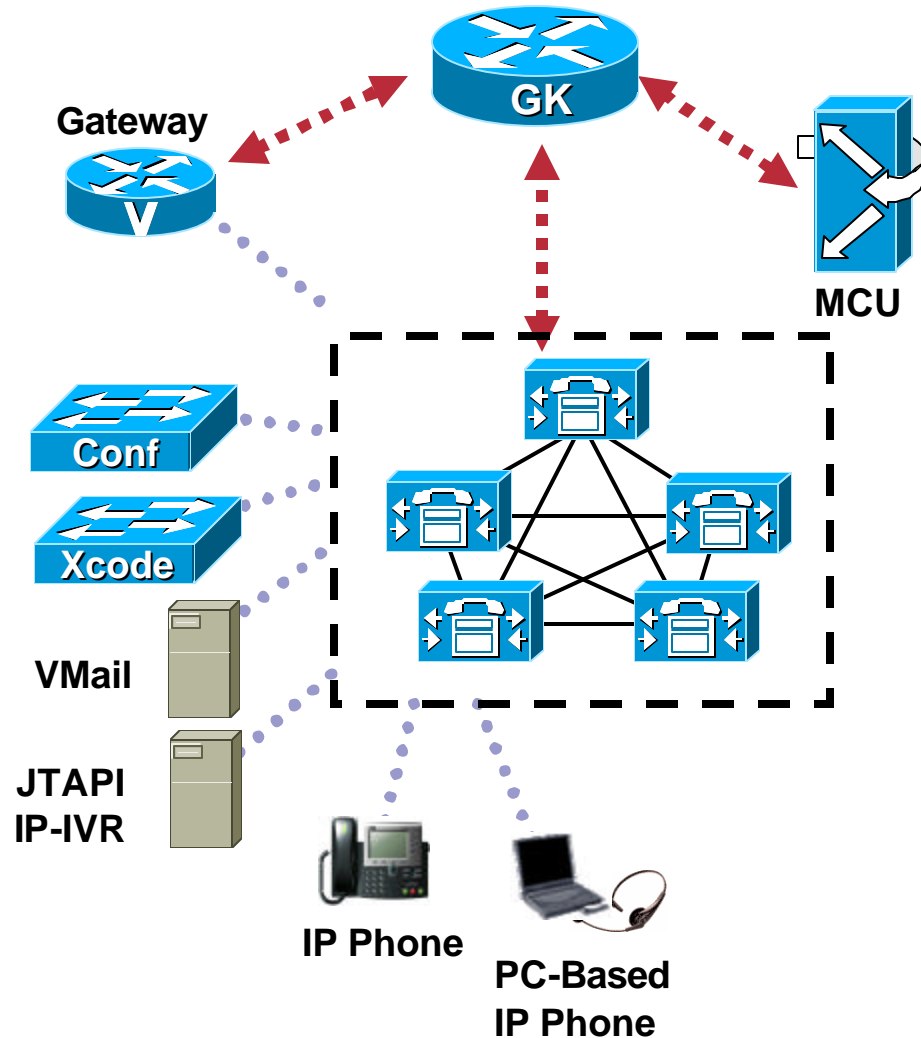
	CallManager Region	CallManager Location	H.323 Gatekeeper	LLQ Class
Audio-Only Calls Configuration	Audio Codec Only	Audio Bit-Rate + Layer-3 Overhead	2X the Bit-Rate <sup>1</sup>	Bit-Rate + all Layer2 and 3 overhead
Video Calls Configuration	Audio Codec + Video Call Bandwidth	Video Call Bandwidth	2X the Bit-Rate	Bit-Rate + 20% overhead
Example G.711 Call	G.711	80kbps	128kbps <sup>1</sup>	80kbps
Example 384kbps Video Call	G.711 audio codec and 384kbps video bandwidth	384kbps	768kbps	460kbps

<sup>1</sup> This behavior changed in CallManager 3.2(2)c and IOS release 12.2(2)XA. Prior to that, CallManager asked for bit-rate + layer-3 overhead, and IOS Gateways asked for 64kbps no matter what type of call it was

# Agenda

- **Video Telephony Fundamentals**
- **Endpoint, MCU and Gateway Integration**
- **Centralized Design and Deployment**
- **Distributed Design and Deployment**
- **Configuration**

# Configuring Cisco CallManager, Cisco IOS GK



- Configure regions, locations
- Configure Gatekeeper device in CCM
  - Choose "H.225" or use auto-discovery
- Configure H.225 trunk
- Configure endpoints
- Configure SCCP MCUs, MRGLs
- Configure Meet-Me numbers
- Configure Gateways that register to CM
- Configure route list/AAR
- Define route patterns, CSS, etc.

# SCCP Endpoints

## Configuring the SCCP Tandberg in CallManager

System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration For Cisco IP Telephony Solutions

### Phone Configuration

[Add a new phone](#)  
[Back to Find/List Phones](#)

Directory Numbers

Phone: New  
Status: Ready  
[Insert](#)

Phone Configuration (Model = TANDBERG Video Endpoint)

Device Information

MAC Address\*

Description

Owner User ID  [\(Select User ID\)](#)

Device Pool\*  [\(View details\)](#)

Calling Search Space

IAR Calling Search Space

Media Resource Group List

User Hold Audio Source

Location

Privacy

Retry Video Call as Audio

Phone Button Template Information

Phone Button Template\*  [\(View button list\)](#)

Softkey Template Information

Softkey Template

Cisco IP Phone - External Data Locations (leave blank to use default)

Information

Directory

Messages

Services

Authentication Server

Proxy Server

Idle

Idle Timer (seconds)

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain

MLPP Indication Not available on this device

MLPP Preemption Not available on this device

Product Specific Configuration

Network Settings Access\*

\* indicates a required item.

[Back to top of page](#)  
[Back to Find/List Phones](#)

System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration For Cisco IP Telephony Solutions

### Directory Number Configuration

[Configure Device \(SEPAAAB0CC0EEFF\)](#)

Associated With

Directory Number: New  
Status: Ready  
Note: Any update to this Directory Number automatically resets the associated devices  
[Add](#)

Directory Number

Directory Number\*

Partition

Directory Number Settings

Voice Mail Profile   
(Choose <None> to use default)

Calling Search Space

AAR Group

User Hold Audio Source

Auto Answer

Call Forward and Pickup Settings

	Voice Mail Destination	Calling Search Space
Forward All	<input type="checkbox"/>	<input type="text" value="&lt; None &gt;"/>
Forward Busy	<input type="checkbox"/>	<input type="text" value="&lt; None &gt;"/>
Forward No Answer	<input type="checkbox"/>	<input type="text" value="&lt; None &gt;"/>
No Answer Ring Duration	<input type="text" value=""/>	(seconds)

Line Settings for this Device

Display (Internal Caller ID)

Line Text Label

External Phone Number Mask

Message Waiting Lamp Policy

Ring Setting (Phone Idle)

Ring Setting (Phone Active)\*\*

Multiple Call / Call Waiting Settings

Maximum Number of Calls\*  (1 - 200)

Busy Trigger\*  (<= Max. Calls)

Forwarded Call Information Display

Caller Name  Caller Number

Redirected Number  Dialed Number

\* indicates required item; changes to Line or Directory Number settings require restart.  
\*\* Ring Setting (Phone Active) applies to this line when any line on the phone has a call in progress.

Note:  
If you are using a language other than English for Display (Internal Caller ID) or Line Text Label text, make sure the correct character set (shown below) is selected. Text displays incorrectly if the wrong character set is selected. (English characters are included in all character sets.)

Character Set

# SCCP Endpoints

## Configuring a VT Advantage Client in CallManager

System | Route Plan | Service | Feature | Device | User | Application | Help

Cisco CallManager Administration  
For Cisco IP Telephony Solutions

Phone Configuration [Add a new phone](#)  
[Back to Find/List Phones](#)

Directory Numbers  
Lines can be added after the new phone is inserted in the database.

Phone: New  
Status: Ready  
[View](#)

Phone Configuration (Model = Cisco 7900)

Device Information

MAC Address\*

Description

Owner User ID  (Select user ID)

Device Pool\*  (View details)

Calling Search Space

AAR Calling Search Space

Media Resource Group List

User Hold Audio Source

Network Hold Audio Source

Location

User Locale

Network Locale

Device Security Mode

Built In Bridge

Privacy

Retry Video Call as Audio

Phone Button Template Information

Softkey Template Information

Expansion Module Information

Firmware Load Information (leave blank to use default)

Cisco IP Phone - External Data Locations (leave blank to use default)

Multilevel Precedence and Preemption (MLPP) Information

Product Specific Configuration

Disable Speakerphone

Disable Speakerphone and headset

Forwarding Delay\*

PC Port\*

Settings Access\*

Gratuitous ARP\*

PC Voice VLAN Access\*

Video Capabilities\*

Auto Line Select\*

Web Access\*

\* indicates a required item.

[Back to top of page](#)  
[Back to Find/List Phones](#)

### Relevant Video Configuration Options:

**Retry Video Call as Audio**  
Checked by default  
*Beware of bug between 4.0(1) and 4.0(1)sr1*  
Uncheck it to allow for AAR

**PC Port**  
Enabled by default  
Must be enabled for VT Advantage to connect to IP Phone

**Video Capabilities**  
Disabled by default  
Enable to allow VT Advantage to associate with IP Phone

# H.323 Endpoints

## Configuring H.323 Clients in CallManager 4.0

Cisco.com

System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration  
For Cisco IP Telephony Solutions

CISCO SYSTEMS

### Phone Configuration

[Add a new phone](#)  
[Back to Find/List Phones](#)

**Directory Numbers**  
Lines can be added after the new phone is inserted in the database.

**Phone: New**  
Status: Ready

**Phone Configuration (Model = H.323 Client)**

**Device Information**

Device Name\*

Description

Owner User ID  (Select User ID)

Device Pool\*  (View details)

Calling Search Space

AAR Calling Search Space

Media Resource Group List

Location

Signaling Port

Retry Video Call as Audio

Wait for Far End H.245 Terminal Capability Set

**H.323 Information**

Outgoing Caller ID Pattern

Calling Party Selection

Calling Party Presentation

Display IE Delivery

Redirecting Number IE Delivery - Outbound

Redirecting Number IE Delivery - Inbound

Media Termination Point Required

**Multilevel Precedence and Preemption (MLPP) Information**

MLPP Domain  (e.g., "0000FF")

MLPP Indication Not available on this device

MLPP Preemption Not available on this device

\* indicates a required item.

[Back to top of page](#)  
[Back to Find/List Phones](#)

System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration  
For Cisco IP Telephony Solutions

CISCO SYSTEMS

### Directory Number Configuration

[Configure Device \(10.1.1.100\)](#)

Associated With

**Directory Number: New**  
Status: Ready  
Note: Any update to this Directory Number automatically resets the associated devices.

**Directory Number**

Directory Number\*

Partition

**Directory Number Settings**

Voice Mail Profile   
(Choose <None> to use default)

Calling Search Space

AAR Group

Auto Answer Not available on this device.

**Call Forward and Pickup Settings**

	Voice Mail Destination	Calling Search Space
Forward All <input type="checkbox"/>	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Forward Busy <input type="checkbox"/>	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Forward No Answer <input type="checkbox"/>	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>

No Answer Ring Duration  (seconds)

Call Pickup Group

**MLPP Alternate Party Settings**

Target (Destination)

Calling Search Space

No Answer Ring Duration  (seconds)

**Line Settings for this Device**

Display (Internal Caller ID)

Line Text Label Not available on this device.

External Phone Number Mask

Message Waiting Lamp Policy Not available on this device.

Ring Setting (Phone Idle) Not available on this device.

Ring Setting (Phone Active)\*\* Not available on this device.

**Multiple Call / Call Waiting Settings**

Maximum Number of Calls\*  (1 - 2)

Busy Trigger\*  (<= Max. Calls)

**Forwarded Call Information Display**

Caller Name  Caller Number

Redirected Number  Dialed Number

\* Indicates required item; changes to Line or Directory Number settings require restart.

Note:  
If you are using a language other than English for Display (Internal Caller ID) or Line Text Label text, make sure the correct character set (shown below) is selected. Text displays incorrectly if the wrong character set is selected. (English characters are included in all character sets.)

Character Set



# H.323 Endpoints

## Configuring H.323 Clients in CallManager 4.1

Cisco.com

System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration  
For Cisco IP Telephony Solutions

Phone Configuration

Phone: SITE2-VSX-1 (Site2 VSX 7000 Video)  
Registration: Unknown  
IP Address: 10.4.21.11  
Status: Ready

Copy Update Delete Reset Phone

Phone Configuration (Model = H.323 Client)

Device Information

Device Name\* SITE2-VSX-1  
Description Site2 VSX 7000 Video  
Owner User ID (Select User ID)  
Device Pool\* DP-Video-WB (View details)  
Calling Search Space international\_css  
AAR Calling Search Space <None >  
Media Resource Group List MRGL\_site2\_Video  
Location site2  
Signaling Port\* 1720

Retry Video Call as Audio  
 Ignore Presentation Indicators (internal calls only)  
 Wait for Far End H.245 Terminal Capability Set

IP Address no longer needed

This will register a RasAggregator Trunk in the Gatekeeper MCU Zone

### Gatekeeper Information

Gatekeeper Name\*\* 10.4.20.3  
E.164\*\* 26002  
Technology Prefix\*\* 1#\*  
Zone\*\* SITE2-VIDEO-GK1  
 Gatekeeper Controlled H.323 Client

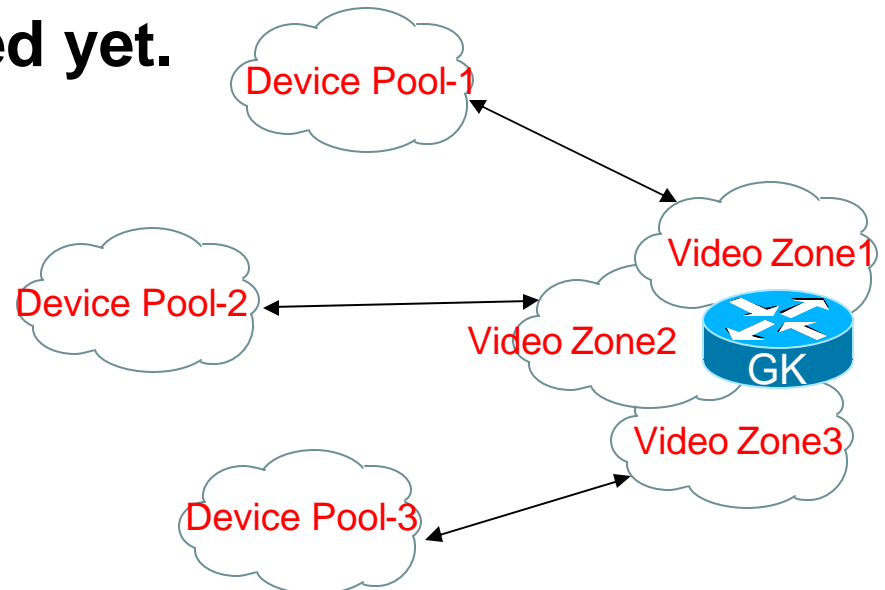
- Supports Dynamic DHCP Addressing
- Must have a Gatekeeper configured in CCM
- E.164 has to match endpoint configuration
  - CCM 4.1(2) must also match the CCM DN
  - See CSCef71775
- Video endpoints dedicated Gatekeeper Zone
- Must Group Endpoints with the following
  - Device Pool
  - Gatekeeper
  - Technology Prefix
  - Zone

# Dynamic H.323 Addressing

## Caveats

- **Must Group all GK Controlled H.323 Clients in the same Device Pool with the same Gatekeeper Zone**
- **In order to load balance these endpoints within the CCM cluster more Gatekeeper zones need to be created.**
- **NOT tested or documented yet.**

Gatekeeper Information	
Gatekeeper Name**	10.4.20.3
E.164**	26001
Technology Prefix**	1#
Zone**	SITE2-VIDEO-GK1
<input checked="" type="checkbox"/> Gatekeeper Controlled H.323 Client	



# Call Admission Control

## CallManager Regions

Cisco.com

System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration  
For Cisco IP Telephony Solutions

CISCO SYSTEMS

### Region Configuration

[Add a New Region](#)  
[Back to Find/List Regions](#)  
[Dependency Records](#)

Region: **San Jose**  
Status: Update completed

Update Delete Restart Devices

#### Region Information

Region Name\*

#### Call Information

The maximum audio codec/video bandwidth supported within this region and between 2 other regions are:

Region	Audio Codec	Video Call Bandwidth
Dallas	<input type="text" value="G.711"/>	<input type="radio"/> None <input checked="" type="radio"/> <input type="text" value="384"/> kbps
San Francisco	<input type="text" value="G.729"/>	<input type="radio"/> None <input checked="" type="radio"/> <input type="text" value="128"/> kbps
San Jose (Within this Region)	<input type="text" value="G.711"/>	<input type="radio"/> None <input checked="" type="radio"/> <input type="text" value="768"/> kbps

Items per page  First previous Next Last

\* indicates required item

- Audio is represented by codec while video is represented by speed. Both really mean the same thing: the maximum bit-rate allowed
- Video bandwidth includes audio (i.e. 320kbps + 64kbps)
- Audio codec also applies to video calls

**!!!VERY IMPORTANT!!!**

Video endpoints typically only support G.728, G.711 and G.722

Audio endpoints typically only support G.729 and G.711

**Effects what audio codec is used for video calls as well!**

# Call Admission Control

## CallManager Locations

Cisco.com

The screenshot shows the Cisco CallManager Administration interface. At the top, there is a navigation menu with links for System, Route Plan, Service, Feature, Device, User, Application, and Help. Below the menu is the Cisco CallManager Administration logo and the Cisco Systems logo. The main heading is "Location Configuration". To the right of the heading are three links: "Add a New Location", "Back to Find/List Locations", and "Dependency Records". The configuration is for a location named "San Francisco". The status is "Update completed". There are four buttons: "Copy", "Update", "Delete", and "Resync Bandwidth". The "Location Information" section shows "Location Name\*" as "San Francisco". The "Audio Calls Information" section shows "Audio Bandwidth\*" with radio buttons for "Unlimited" and "48" kbps. Below this is a note: "If the audio quality is poor or choppy, lower the bandwidth setting. For ISDN use multiples of 56 kbps or 64 kbps." The "Video Calls Information" section shows "Video Bandwidth\*" with radio buttons for "None", "Unlimited", and "128" kbps. A note at the bottom states: "\* indicates required item".

- Audio is represented as bit-rate + overhead (i.e. 24k for G.729, 80k for G.711)
- Video is represented as bit-rate only (i.e. 384k for a 384k call) and includes the audio portion

**!!!VERY IMPORTANT!!!**

The audio bandwidth setting does not pertain to the audio channel of a video call

- Kept separate for a very good reason: voice should have its own dedicated bucket, separate from video, rather than having them fight over one big bucket

Matches the way it works at Layer-2 in Low-Latency Queueing configurations where video is placed in a separate PQ, or in a CBWFQ, and the audio channel of a video call is placed in the same class as the video channel

# SCCP Conference Bridges

## Configuring the Conference Bridge SCCP Protocol

Cisco.com

The image displays the IP/VC Administrator interface for an MC03A MCU. The main window shows the 'Basics' configuration page with 'MCU Mode' set to 'MCU'. A dropdown menu is open, showing values 0, 15, and 30. A red arrow points from this menu to the 'Edit SCCP Protocol Configuration' dialog box.

The 'Edit SCCP Protocol Configuration' dialog box is open, showing the following settings:

- Enable
- Active SCCP service prefix: 70
- TFTP Servers table:

IP Address	Port
10.1.1.100	88
- CallManagers table:

IP Address	Port
10.1.1.101	2000
10.1.1.102	2000
- Perform MCU reset on CallManager Reset message
- Control Channel:
  - Local port base: 11000
  - Priority (0-63): 24
- Registration:
  - Retries: 3
  - Initial timeout (sec): 30
  - Consequent timeout (sec): 10
- Keep Alive:
  - Retries: 3
  - Timeout (sec): 10
- Fall Over:
  - Recovery mode: gradually
  - Recovery timer (sec): 300

Buttons: Change configuration locale, OK, Cancel, Help.

The background window shows the 'SCCP Protocol Configurations' table:

Description	Status	Servicing CallManager
SCCP	active	172.19.247.148

# SCCP Conference Bridges

## Configuring the Conference Bridge SCCP Service

The screenshot displays the IP/VC Administrator interface for an MC03A MCU. The main window shows a table of services under the 'Services' tab:

Prefix	Description	Status	Parties	Media
70	SCCP	Enabled	3 (up to 24)	Voice, Video

Three configuration dialog boxes are overlaid on the interface:

- Edit Service:** Shows 'Service prefix' as 70 and 'SCCP service' checked. The 'Service description' is 'SCCP'. 'Media types' for 'Voice', 'Video', and 'Data' are all checked.
- Edit View:** Shows 'View 1 Settings'. 'Max Layout' and 'Initial Layout' are set to 'None'. 'Enable voice activate' is checked with 'Voice Activate Method' set to 'See you see me'. 'Video Schemes Settings' table is shown below.
- Edit Video Scheme:** Shows 'Mode' as 'Basic', 'Max bit rate' as 320, 'Max frame rate' as 30, 'Video format' as 'H.263', 'Picture size' as 'CIF', and 'Picture rate (Kbps)' as 320.

Red arrows indicate the configuration flow: from the 'Services' table to 'Edit Service', then to 'Edit View', then to 'Edit Video Scheme', and finally back to the 'Services' table.

# SCCP Conference Bridges

## Configuring the Conference Bridge SCCP Service for Continuous Presence

**IP/VC Administrator**

**MC03A : MCU**

Upload Import Export Reset Refresh Setup Wizard

Status Settings Registered MPs Protocols **Services** Event Log

Prefix	Description	Status	Parties	Media
70	SCCP	Enabled	3 (up to 24)	Voice, Video

**Edit Service**

Service prefix: 70  SCCP service

Service description: SCCP

Media types:  Voice  Video  Data

**Conference Views**

	Max BW (Kbps)	Fps	Format	Pic. Size
	320	30	H.263	CIF

**Edit View**

**View 1 Settings**

Max Layout:  Change...

Initial Layout:  Change...

Use generic layout

Display participant names in border

Display participant names in theme

Enable voice activate

Voice Activate Method: See you see me

Enable auto-switch

Auto-switch Interval (sec): 15

**Video Schemes Settings**

#	Max BW (Kbps)	Fps	Format	Pic. Size	
1	320	30	H.263	CIF	Add... Edit... Delete

**Edit Video Scheme**

Mode: Basic

Max bit rate (Kbps): 320

Max frame rate: 30

Video format: H.263 H.263+

Picture size: CIF

Video quality preference: Best motion Best quality

Allow dynamic scheme

Picture rate (Kbps): 320

Warning: Applet Window

# SCCP Conference Bridges

## Configuring the Conference Bridge in CallManager

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The screenshot shows the Cisco CallManager 4.0 Administration console. The top navigation bar includes 'System', 'Route Plan', 'Service', 'Feature', 'Device', 'User', 'Application', and 'Help'. The 'Service' menu is open, showing options like 'Cisco IPMA Configuration Wizard', 'Cisco CM Attendant Console', 'Media Resource', and 'Service Parameters'. A red arrow points from the 'Service' menu to the 'Conference Bridge Configuration' page.

**Cisco CallManager 4.0 Administration**

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**Conference Bridge Configuration**

[Add a New Conference Bridge](#)  
[Meet-Me Number/Pattern Configuration](#)  
[Cisco CallManager Service Parameters](#)  
[Back to Find/List Conference Bridges](#)

**Conference Bridge: New**  
Status: Ready

Conference Bridge Type: Cisco Video Conference Bridge(IPVC-35xx)

MAC Address\*

Description

Device Pool\*

Location: < None >

**Product Specific Configuration**

**General**

DSCP for Control Messages\*: CS3(prec 3) DSCP (011000)

Local Base Port: 11000

**Registration Info**

Failover Recovery Mode\*: Graceful



# Configuring Meet-Me Numbers

System Route Plan Service Feature Device User Application Help

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For Cisco IP Telephony Solutions

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## Meet-Me Number/Pattern Configuration

[Add a New Meet-Me Number](#)  
[Conference Bridge Configuration](#)  
[Back to Find/List Meet-Me Numbers](#)

**Meet-Me Number/Pattern: 7XXXX**  
Status: Ready

Directory Number or Pattern\*

Description

Partition

\* indicates required item

# Configuring Cisco IOS Gatekeeper

- Gatekeeper release 12.2(4)T or higher required
- Configure different local zones for CallManager and videoconferencing devices
  - Enables scalable dial plans, differentiation in zone bandwidth controls (CAC mechanisms), granular proxy usage definitions and endpoint management
  - Video endpoints can't specify the zone name they want to register with, so use "zone subnet" commands to force video endpoints into the proper zone
- Be careful to turn proxy usage **off** for calls
  - Default proxy usage configuration is:**

```
PROXY USAGE CONFIGURATION :
Inbound Calls from all other zones :
  to terminals in local zone : use proxy
  to gateways in local zone  : do not use proxy
  to mcu's in local zone    : do not use proxy
Outbound Calls to all other zones :
  from terminals in local zone : use proxy
  from gateways in local zone  : do not use proxy
  from mcu's in local zone    : do not use proxy
```
- Configure "hopoff" commands for all MCU technology prefixes and optionally use the "default-technology" command for the CallManager technology prefix; this must only be done if using multiple local zones on a single gatekeeper

# Configuring Cisco IOS Gatekeeper - CM4.1

- **CM 4.1 will register to IOS GK as an IP-IP gateway, allowing a simplified gatekeeper configuration as well as H.323 endpoint mobility**

**gatekeeper**

**zone local endpoints-mcus-and-gateways domain.com in via  
callmanager out via callmanager enable-intrazone**

**zone local callmanager domain.com**

**zone prefix video-endpoints <XXXX> //E.164 zone prefix of the endpoint zone**

**gw-type-prefix 1# default-technology**

**endpoint ttl 60**

**no shutdown**

# Gatekeeper Clustering Configuration Example

## 12.2(15)T1 Enterprise MCM

```
gatekeeper
zone local CHC_Video customer.com 10.1.2.1
zone local SJC_Video2 customer.com
```

```
zone cluster local CHCVideo_Cluster CHC_Video
element CHC_Video2 10.1.3.1 1719
element CHC_Video3 10.1.1.1 1719
!
zone cluster local SJCVideo_Cluster SJC_Video2
element SJC_Video 10.1.1.1 1719
element SJC_Video3 10.1.3.1 1719
!
```

```
zone prefix SJC_Video2 40852.....
zone prefix CHC_Video 72067.....
```

```
load-balance cpu 80 memory 80
```

**Cluster Configured for Each Zone Supported in the Cluster; Elements Listed in Order Used for Backup**

**Load-Balance Thresholds Set to 80% for CPU and Memory**

# Cisco CallManager Gatekeeper Configuration

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## Gatekeeper Configuration

**Gatekeepers**  
<Add a New Gatekeeper>

**Gatekeeper: New**

Status :Ready

Insert

**Gatekeeper Information**

Host Name/IP Address*	10.1.1.10
Description	MCM Gatekeeper/Proxy
Registration Request Time To Live	60
Registration Retry Timeout	300
Enable Device	<input checked="" type="checkbox"/>

\* indicates required item

# Cisco CallManager H.225 Trunk Setup

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## Add a New Trunk

Select the type of Trunk you would like to create:

Trunk type\*

Device Protocol\*

- H.225 Trunk (Gatekeeper Controlled)
- Inter-Cluster Trunk (Gatekeeper Controlled)
- Inter-Cluster Trunk (Non-Gatekeeper Controlled)

\* indicates required item

## Add a New Trunk

Select the type of Trunk you would like to create:

Trunk type\*

Device Protocol\*

\* indicates required item

# Cisco CallManager H.225 Trunk Setup (Cont.)

System Route Plan Service Feature Device User Application Help Logout

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## Trunk Configuration

[Add a New Trunk](#)  
[Back to Find/List Trunk](#)

**Product: H.225 Trunk (Gatekeeper Controlled)**  
**Device Protocol: H.225**  
Status: Ready

**Device Information**

Device Name*	H323_Trunk
Description	H323 Trunk to MCM GK
Device Pool*	Default
Media Resource Group List	< None >
Location	< None >
AAR Group	< None >

Media Termination Point Required

**Gatekeeper Information**

Gatekeeper Name*	10.1.1.10
Terminal Type*	Gateway
Technology Prefix	1#
Zone	voice

\* indicates required item

[Back to Find/List Trunk](#)

This Defines the Default Tech-Prefix in GK

# Adding Route to GK in Cisco CallManager

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## Route Pattern Configuration

[Add a New Route Pattern](#)  
[Back to Find/List Route Patterns](#)

**Route Pattern: New**  
Status: Ready  
Note: Any update to this route pattern automatically resets the associated gateway/route list

**Pattern Definition**

Route Pattern*	3XXX
Partition	< None >
Description	Route to MCM Gatekeeper/Proxy
Numbering Plan*	North American Numbering Plan
Route Filter	< None >
Gateway/Route List*	H323_Trunk
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern
<input checked="" type="checkbox"/> Provide Outside Dial Tone	<input type="checkbox"/> Urgent Priority



# Configuring IP/VC MCUs

- **MCU registers its service prefixes with the gatekeeper**
- **You can configure services to support audio-only, or configure them to support a variety of speeds of both voice+video, and it will allow audio-only participants to join as well**
- **IP/VC 35xx MCUs only support G.711 (except for 3540 with optional daughtercard and 3511 models)**
- **Support for SCCP in 3.2 plus**
- **Continuous Presence is supported for H.323 conferences, using either SCCP or H.323 endpoints**
- **EMP module may be required for some formats or endpoints**

# H.323 Conference Bridges

## Configuring the Conference Bridge H.323 Protocol

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The image displays the IP/VC Administrator interface for configuring H.323 protocols on an MC03A MCU. The interface is divided into several sections:

- System:** Includes buttons for Upload, Import, Export, Refresh, and Setup Wizard.
- Settings:** A sidebar menu with options: Basics, Conference Mgmt, Delimiters, Themes, Alert Indications, and Advanced.
- Basics:** Shows MCU Mode set to 'MCU' and Number of SCCP ports set to '0'. A dropdown menu is open showing options 0, 15, and 30.
- Protocols:** A table lists H.323 and SCCP protocols. The H.323 protocol is selected, showing its configuration details.
- H.323 Protocol Configurations Table:**

Description	Gatekeeper Address	Status	Servicing Gatekeeper
H.323	172.19.241.20:1719	disabled	
- Edit H.323 Protocol Configuration Dialog:** A pop-up window for editing the H.323 protocol settings.
  - Activate protocol settings
  - Description: H.323
  - Gatekeeper Settings:
    - Registration name: HK.G1PVC-354DMCU
    - Gatekeeper Address: 10.1.1.1 (with 'Go to Gatekeeper...' button)
    - Gatekeeper Port: 1719
    - Strip zone prefix
    - Gatekeeper Zone Prefix: (empty)
  - Advanced Settings: (button)
  - General:
    - Enable H.229
    - Enable alternate Gatekeeper
    - Enable Fast Start
    - Enable H.245 tunneling
    - Enable generic audio capabilities (required for G.722.1 support)

Red arrows indicate the flow of configuration: from the 'Basics' section to the 'Protocols' section, then to the 'H.323 Protocol Configurations' table, and finally to the 'Edit H.323 Protocol Configuration' dialog box.

# MCU Service Table

**IP/VC Administrator** **CISCO SYSTEMS**

**MC01A : MCU**

Upload Import Export Reset Refresh Setup Wizard

Status Settings Registered MPs Protocols **Services** Event Log

Prefix	Description	Status	Parties	Media	Bandwidth (Kbps)
60	Voice Only	Enabled	6 (up to 8)	Voice	64 (0 video)
70	SCCP	Enabled	0 (up to 0)	Voice, Video	384 (320 video)
71	SCCP Rate Matched	Enabled	0 (up to 0)	Voice, Video	768 (704 video)
80	Full Screen H.264	Enabled	6 (up to 8)	Voice, Video	384 (320 video)
81	Full Screen H.263	Enabled	6 (up to 8)	Voice, Video	384 (320 video)
82	Continuous Presence	Enabled	8 (up to 16)	Voice, Video	768 (704 video)
83	Tele-Education (H.239)	Enabled	6 (up to 8)	Voice, Video	384 (320 video)
84	Tele-Education (T.120)	Enabled	6 (up to 8)	Voice, Video, Data	384 (320 video)
85	xDSL CP	Enabled	6 (up to 8)	Voice, Video	128 (110 video)
86	Full Screen (MP)	Enabled	6 (up to 8)	Voice, Video	384 (320 video)
87	Continuous Presence (MP)	Enabled	6 (up to 8)	Voice, Video	384 (320 video)
88	Full Screen (RM)	Enabled	6 (up to 8)	Voice, Video	384 (320 video)
89	Continuous Presence (RM)	Enabled	6 (up to 8)	Voice, Video	384 (320 video)
90	Continuous Presence (RM2+8)	Enabled	8 (up to 10)	Voice, Video	384 (320 video)

Total:  Add... Edit... Delete Help

# MCU H.323 Service Configuration

**Add Service**

Service prefix: 384  SCCP service

Service description:  
384K Voice Activated

Media types:  
 Voice  Video  Data

Reserved number of parties: 3  
 Maximum number of parties: 3

Input Indications Management

Voice Data

Display all conference views

OK Cancel Help

Java Applet Window

---

**Conference Views**

	Max BW (Kbps)	Fps	Format	Pic. Size
	320	30	H.263	CIF

Enable Dual Video

---

**Edit View**

**View 1 Settings**

Max Layout:  Change...

Initial Layout:  Change...

Use dynamic layout

Display current speaker border

Display participant name in frame

Enable voice activate

Voice Activate Method: See you see me

Zoom in layout:  Change...

Use Processor: Auto

Video Scheme Policy: Maximum bit rate

Video Picture Size: CIF

Active theme: [Default]

Enable auto-switch

Auto-switch Interval (sec): 15

**Video Schemes Settings**

#	Max BW (Kbps)	Fps	Format	Pic. Size	
1	320	30	H.263	CIF	

Add... Edit... Delete

OK Cancel Help

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# H.323 Conference Bridges

## Configuring the Conference Bridge in CallManager

1 of 2

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**OUTBOUND**  
**CCM ⇒ MCU**

**Gateway Configuration**

This product contains cryptographic features and is subject to United States and local country laws. Use of this product may require a license. A summary of U.S. and international requirements is available at <http://www.cisco.com>. If you require further information, contact your local sales representative.

**Product :** H.323 Gateway  
**Gateway :** New  
**Device Protocol :** H.225

Status: Ready

**Device Information**

Device Name*	10.1.1.103
Description	IPVC MCU
Device Pool*	— Not Selected —
Media Resource Group List	< None >
Location	< None >
AAR Group	< None >
Signaling Port*	2720

# H.323 Conference Bridges

## Configuring the Conference Bridge in CallManager

2 of 2

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INBOUND

MCU ⇒ CCM

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### Add a New Phone

Select the type of the phone you would like to create:

Phone type\*

Status: Ready

\* indicates required item

This will register a RasAggregator Trunk in the Gatekeeper MCU Zone

Gatekeeper Information

Gatekeeper Name\*\*

E.164\*\*

Technology Prefix\*\*

Zone\*\*

Gatekeeper Controlled H.323 Client

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### Phone Configuration

Add a new phone  
[Dependency Records](#)  
[Back to Find/List Phones](#)

Phone: SITE1-MCU-Trunk (SITE1-MCU-Trunk)  
Registration: Unknown  
IP Address: 10.4.11.10  
Status: Ready

#### Phone Configuration (Model = H.323 Client)

##### Device Information

Device Name\*

Description

Owner User ID  (Select User ID)

Device Pool\*\*  (View details)

Calling Search Space

AAR Calling Search Space

Media Resource Group List

Location

Signaling Port\*

Retry Video Call as Audio

Ignore Presentation Indicators (internal calls only)

Wait for Far End H.245 Terminal Capability Set

# Configuring Gateways

- **Depends on the type and use of the gateway**

MGCP/PSTN gateways register with Cisco CallManager

H.323/PSTN gateways can point directly to Cisco CallManager or register with the gatekeeper (depends on the deployment scenario/application: VoIP or IP telephony...)

H.323/H.320 gateways (i.e. IP/VC gateways) register with the gatekeeper (they do not know how to point directly to Cisco CallManager)

- **Any-to-any connectivity**

IP phone to PSTN calls use MGCP/PSTN gateway, or H.323/PSTN gateway

PSTN to IP phone same as above but in opposite direction

H.323 video to PSTN/H.320, or PSTN/H.320 to H.323 use IP/VC gateway

# Cisco CallManager H.323 Gateway Setup

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## Add a New Gateway

Select the type of gateway you would like to create:

Gateway type\*

Device Protocol\*

\* indicates required item



# Cisco CallManager H.323 Gateway Setup (Cont.)

System Route Plan Service Feature Device User Application Help Logout

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## Gateway Configuration

[Back to Find/List Gateways](#)

**Product : H.323 Gateway**  
**Gateway : New**  
**Device Protocol: H.225**

Status: Ready

**Device Information**

Device Name*	<input type="text" value="10.1.1.40"/>
Description	<input type="text" value="IPVC 3540 MCU"/>
Device Pool*	<input type="text" value="Default"/>
Media Resource Group List	<input type="text" value="&lt; None &gt;"/>
Location	<input type="text" value="&lt; None &gt;"/>
AAR Group	<input type="text" value="&lt; None &gt;"/>

Media Termination Point Required

# Summary

- **Cisco CallManager 4.x has unified and simplified voice and video deployments**
- **There is a clear migration strategy from traditional video deployments to a Cisco CallManager unified voice/video approach**
- **QoS is critical to successful voice AND video deployments**

# Complete Your Online Session Evaluation!

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Por favor, complete el formulario de evaluación.

**Muchas gracias.**

**Session ID: VVT-2100**

**“DESIGNING AND DEPLOYING  
IP VIDEO TELEPHONY NETWORKS”**

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