Call Admission Control Design for the Enterprise WAN

BRKUCT-2010

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Cisco Networkers 2007

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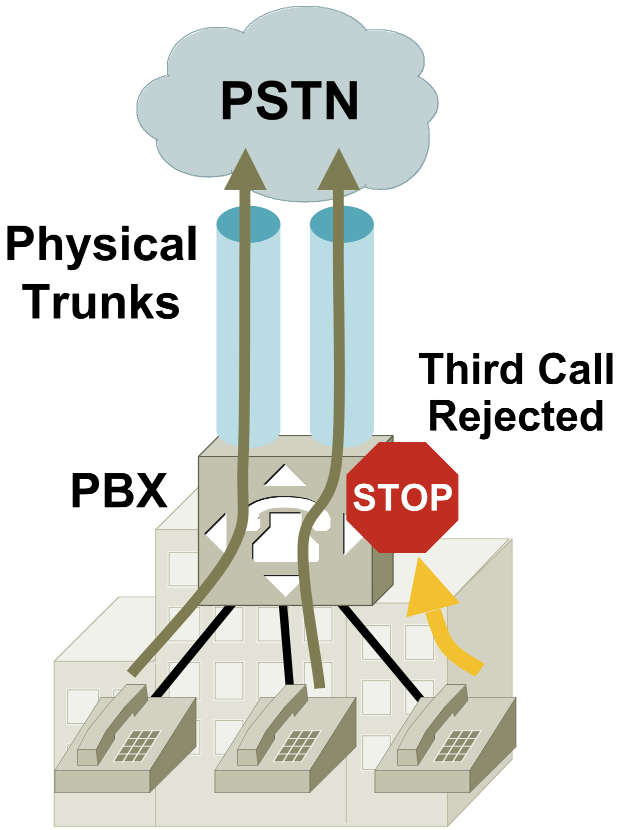
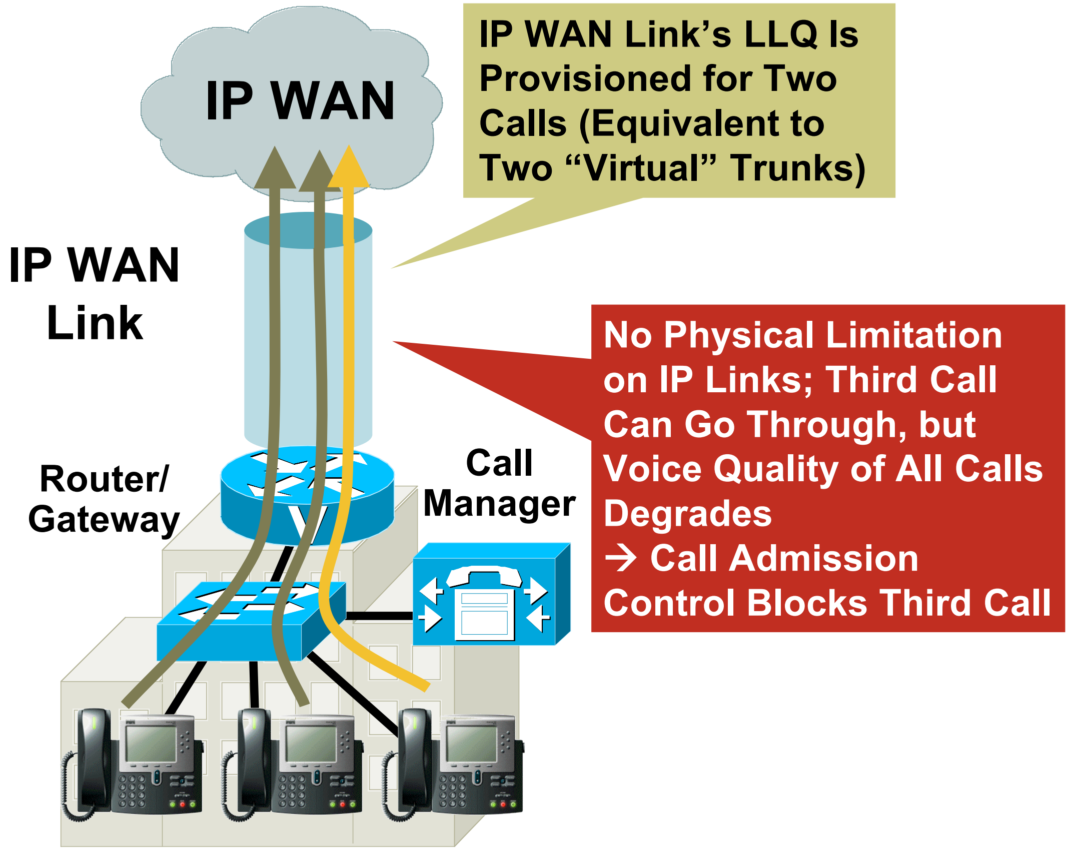
Introduction
Why Is Call Admission Control (CAC) Needed?

Circuit-Switched Networks



Packet-Switched Networks



Introduction
Session Agenda and Scope

- Call Admission Control Principles

  What Are the Two Main CAC Principles?

  What Are Their Advantages and Limitations?

- Call Admission Control Elements

  What CAC Mechanisms Are Available for a Cisco Unified Communications System?

  How Do They Work?

- Call Admission Control Design

  What Are the Different WAN Topologies?

  How Do You Apply the CAC Mechanisms to a Real Enterprise Network Using Cisco Unified Communications?

Introduction
Prerequisites

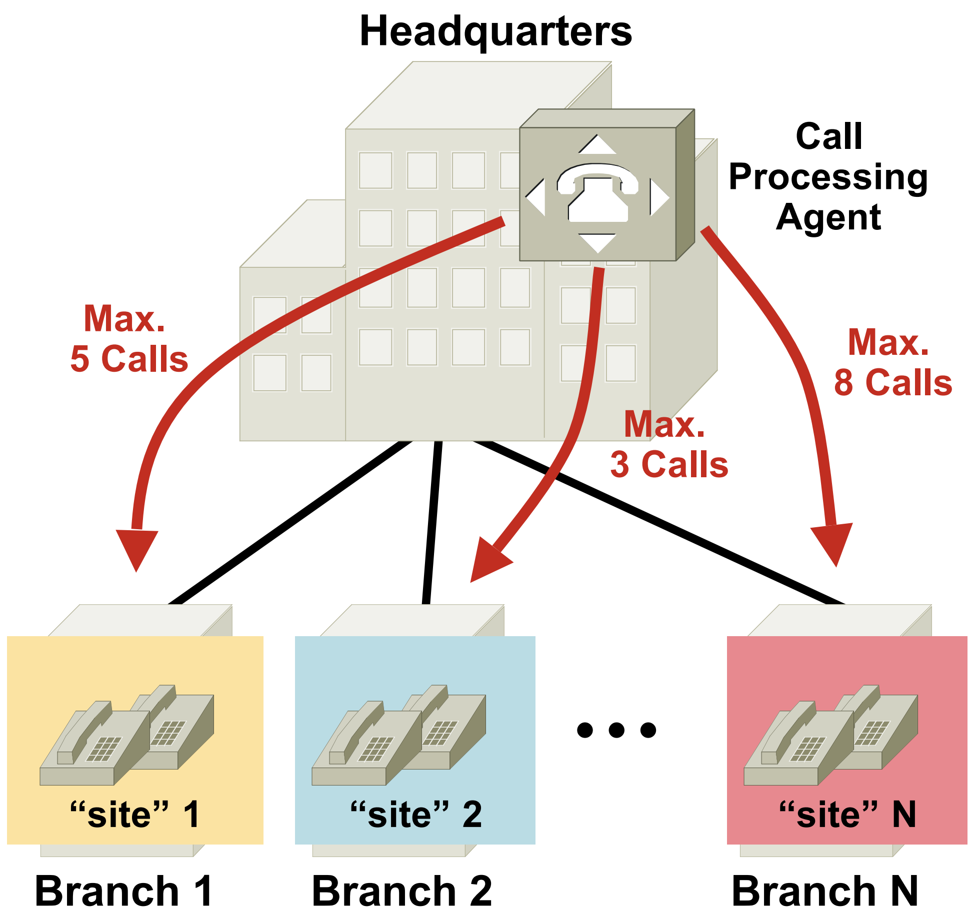
This Session Assumes Knowledge of:

- H.323
- Quality of service
- Marking and classification of traffic
- Cisco Unified Communications deployment models

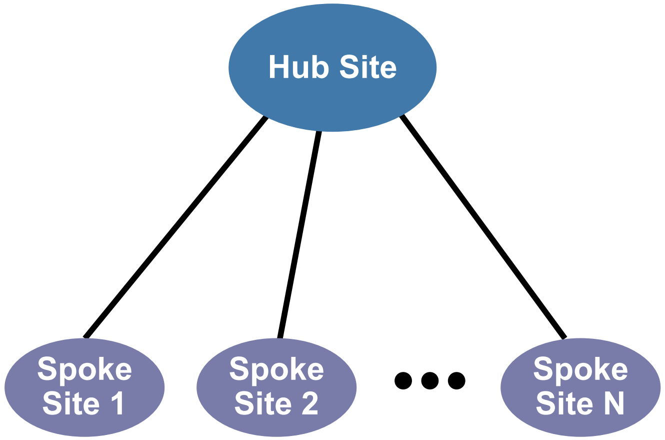
Call Admission Control Principles

Topology-Unaware CAC

- Based on static configuration in call processing agent
- Define the logical “site” to match physical branch office
- Configure a max number of calls or max amount of bandwidth



Topology-Unaware CAC



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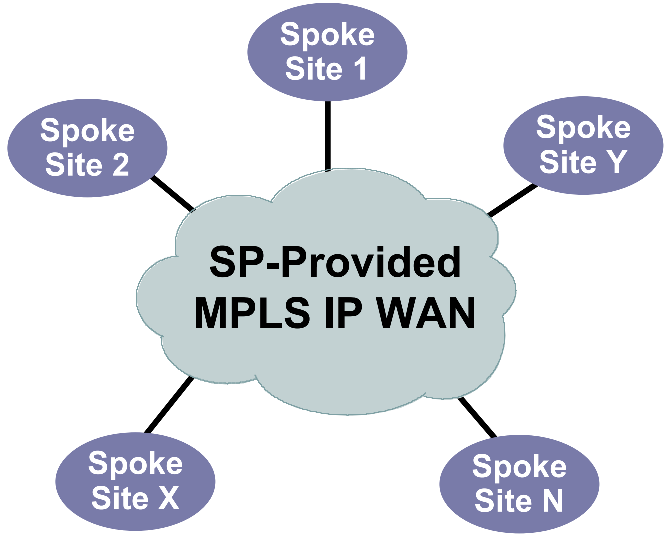
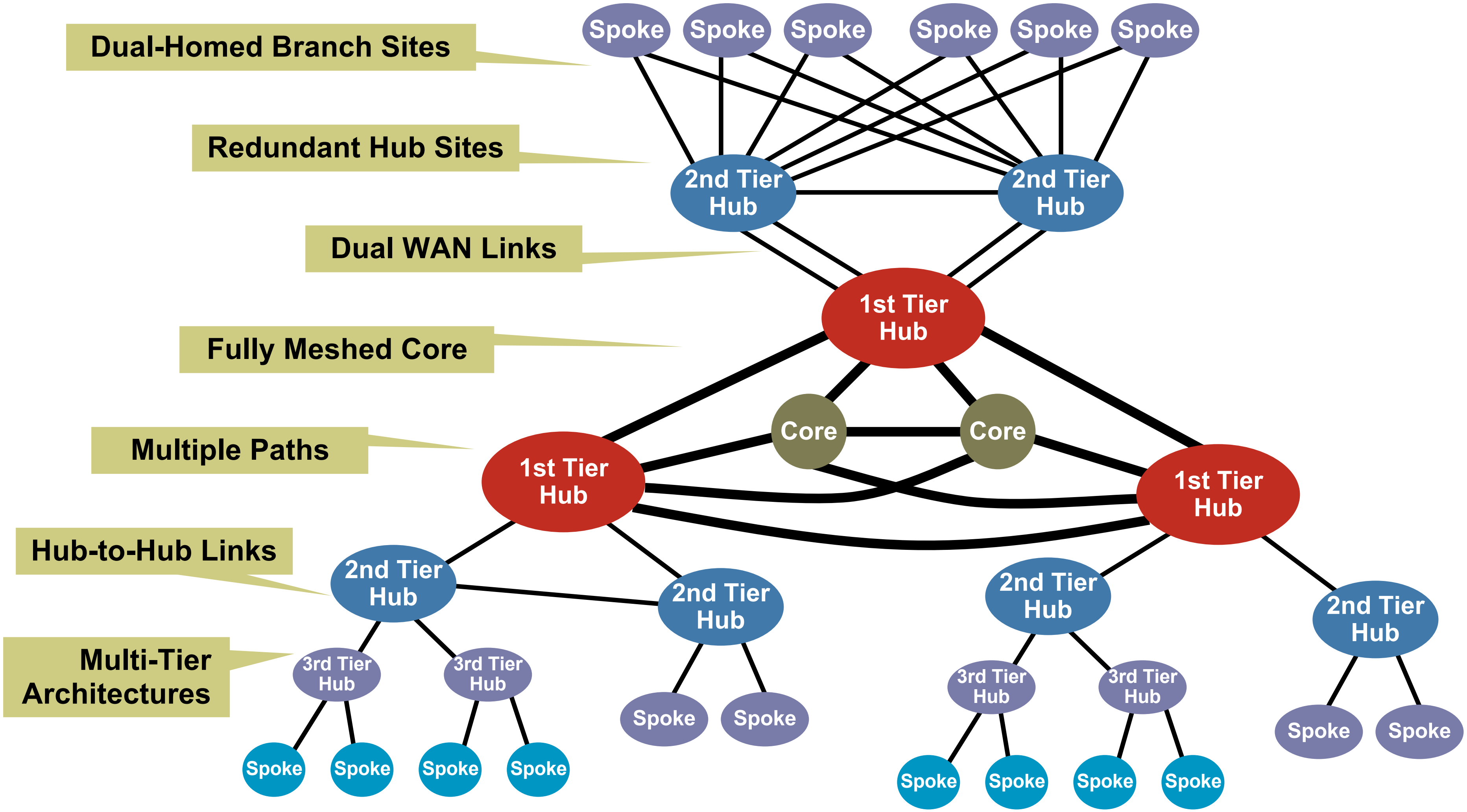


Limited to:

- Simple hub-and-spoke topologies
- Simple MPLS-based topologies

Topology-Unaware CAC
"Real" Network Topology Aspects



Call Admission Control Principles
Topology-unaware CAC Limitations: Example #1

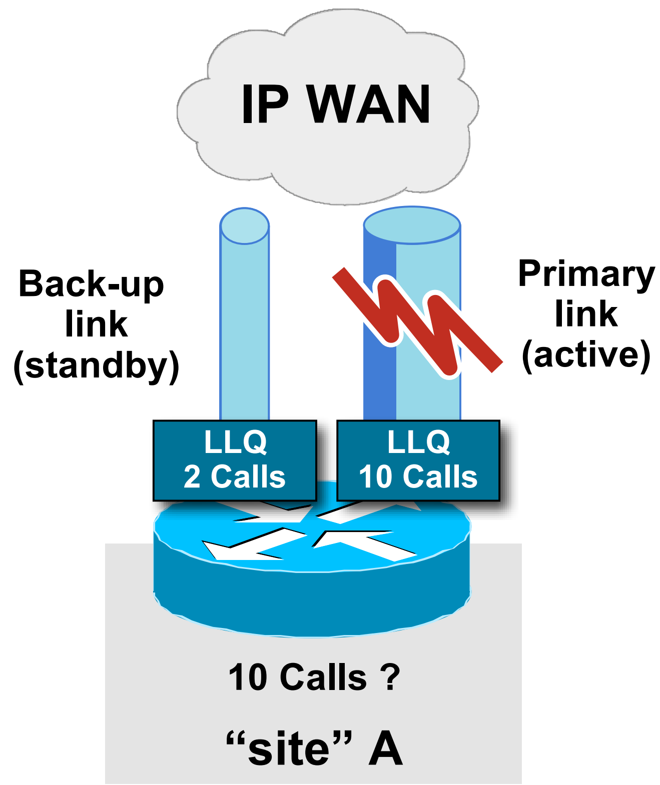
- How do you configure “site” A using a topology-unaware CAC mechanism?
- Allow 10 calls?

    PQ on backup link is over-run during failures

- Allow 2 calls?

    PQ on primary link is under-utilized during normal operation

Primary/Backup Dual Links



Call Admission Control Principles
Topology-unaware CAC Limitations: Example #2

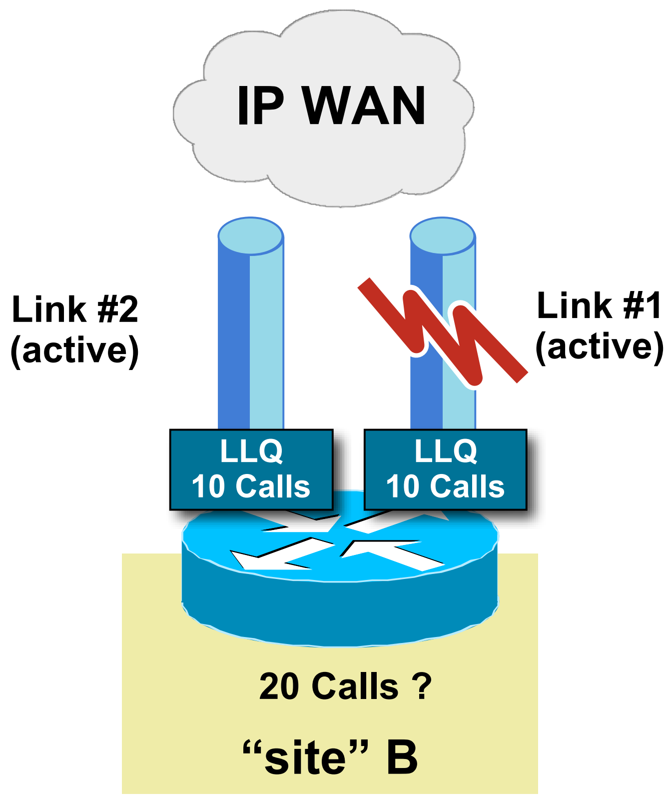
- How do you configure “site” B using a topology-unaware CAC mechanism?
- Allow 20 calls?

  PQ is over-run during failures of one link

  How do you ensure “perfect” load-balancing?
- Allow 10 calls?

  PQ’s are under-utilized during normal operation

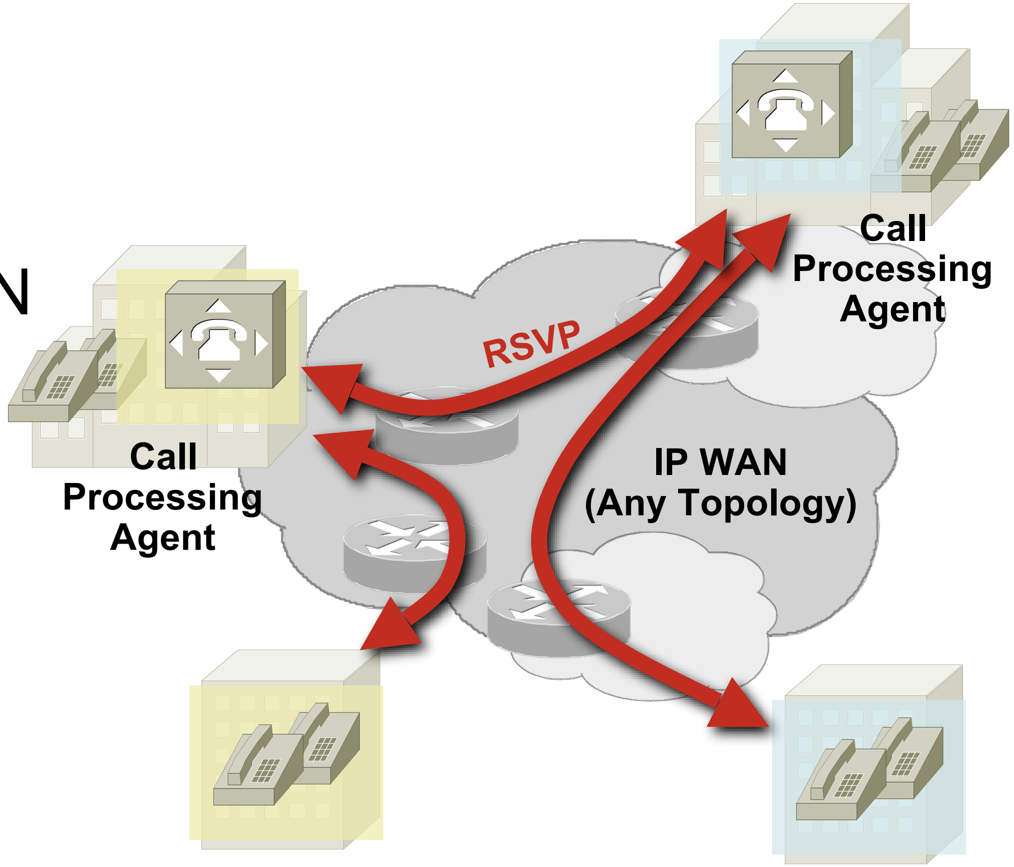
Dual Links with Load Balancing



Call Admission Control Principles
Topology-aware CAC

- Based on communication between call processing agents and the network on available resources
- Can be applied to any WAN network topology
- Dynamically adjusts to topology changes
- Requires a signaling protocol

→ RSVP (Resource ReSerVation Protocol) is the first industry standard for QoS signaling



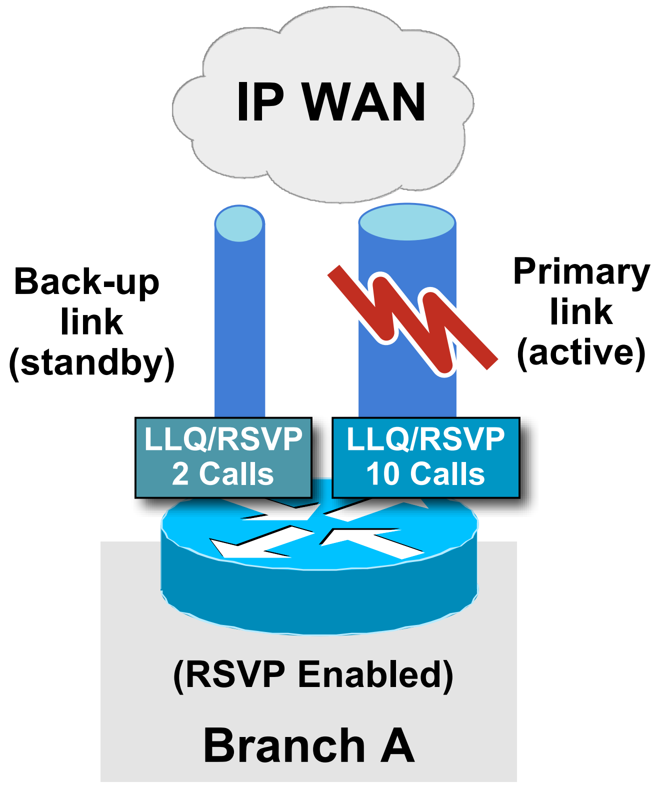
Call Admission Control Principles
Example #1: RSVP Solution

- Configure Branch A to use RSVP CAC in the call processing agent
- Enable RSVP on both WAN router interfaces, matching available LLQ bandwidth
- Number of RSVP-admitted calls now varies depending on which link is used

  10 on primary, 2 on backup

- Policies determine what to do when CAC fails for a call

Primary/Backup Dual Links



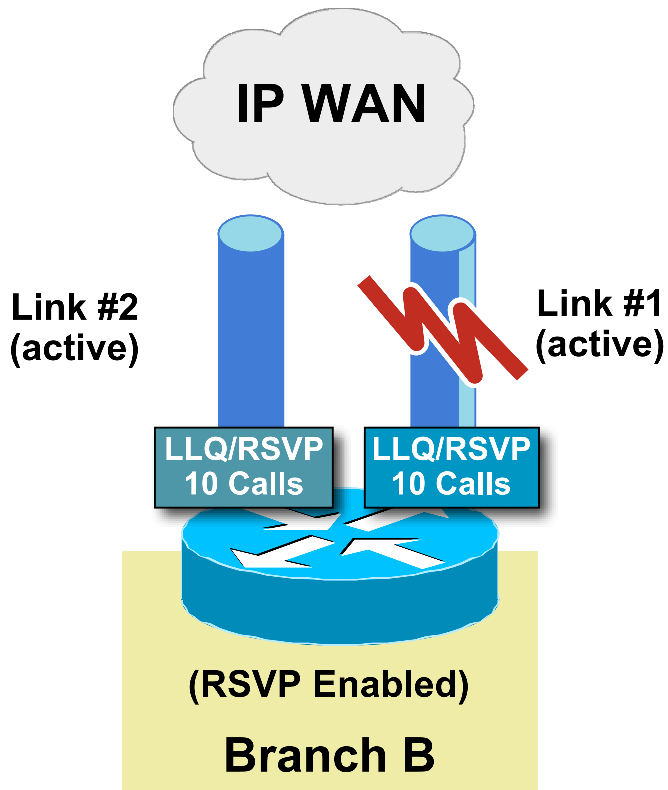
Call Admission Control Principles
Example #2: RSVP Solution

- Configure Branch B to use RSVP CAC in the call processing agent
- Enable RSVP on both WAN router interfaces, matching available LLQ bandwidth
- Number of RSVP-admitted calls now varies depending on how many links are up

  10 with one link up

  Up to 20 with two links up *(depending on how routing protocol selects links)*

Dual Links with Load Balancing



Call Admission Control Principles
Summary

Topology-Unaware CAC

- Based on static configuration within call processing agent
- Does not react to network topology changes (e.g. link failures)
- Limited to simple topologies
- Limited to a single call processing agent
- Examples: Cisco CallManager® "static" locations, Cisco IOS® gatekeeper zones

Topology-Aware CAC

- Based on real network resources
- Requires a signaling protocol (RSVP)
- Reacts to network topology changes
- No topology limitation
- Can be used by different call processing agents
- Examples: Cisco CallManager RSVP-enabled locations, Cisco multiservice IP-IP gateway

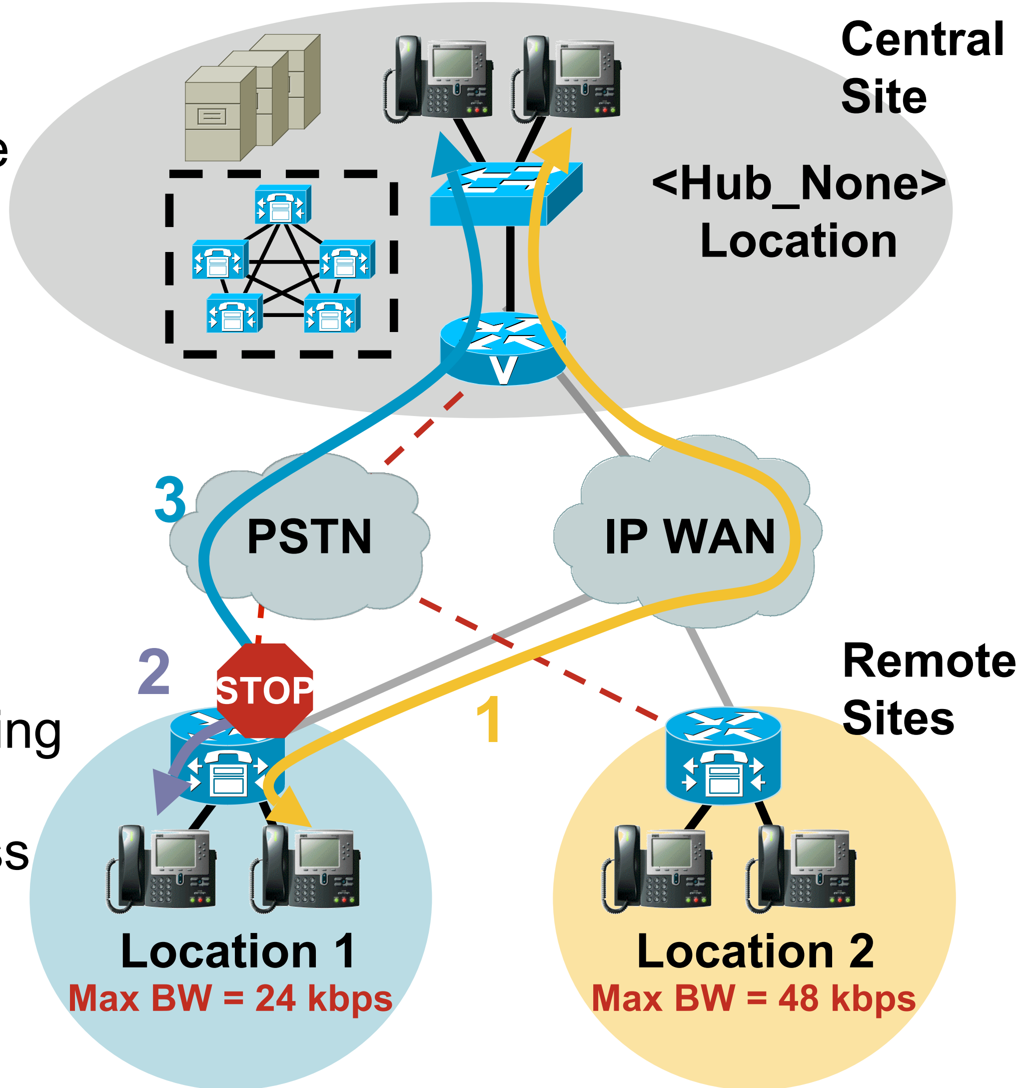
Call Admission Control Elements

Call Admission Control Elements
Agenda

- Topology-Unaware CAC
  - Cisco CallManager “Static” Locations
  - Gatekeeper Zones
- Resource Reservation Protocol (RSVP)
- Topology-Aware CAC
  - Cisco CallManager RSVP-Enabled Locations
  - Gatekeeper and IP-IP Gateway with RSVP

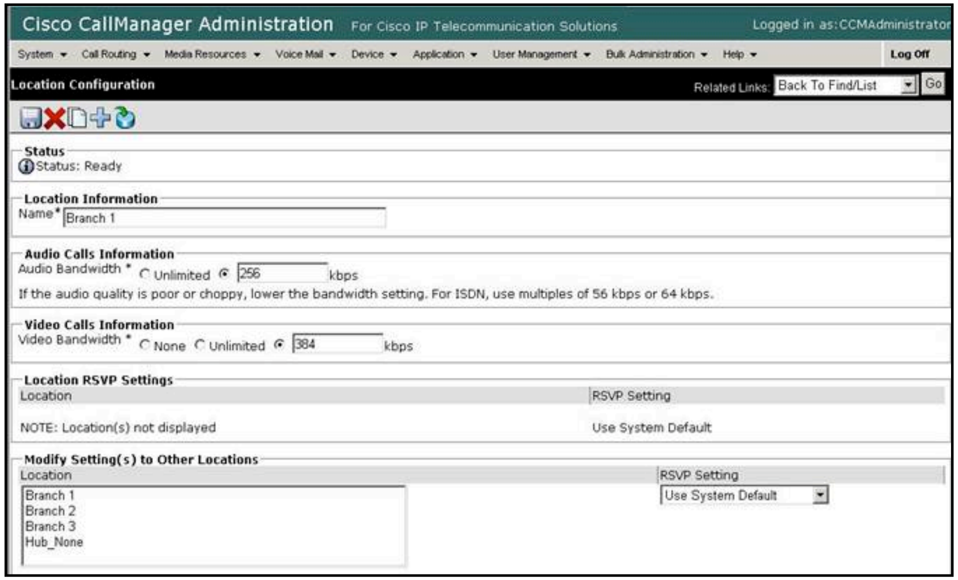
Cisco CallManager Static Locations
Concept

- Prevent WAN link over-subscription by limiting voice bandwidth
- Assign bandwidth limit for voice per location
- When resources are insufficient, phone gets fast-busy tone and a message is displayed
- If Automated Alternate Routing (AAR) is enabled, the call is automatically rerouted across the PSTN



Cisco CallManager Static Locations
Configuration

- Audio is represented as bit-rate + IP 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) which includes the audio portion
- The audio bandwidth setting does not pertain to the audio channel of a video call



Cisco CallManager Static Locations
Notes

- If transcoders are needed (e.g., in presence of G.711-only devices), the transcoder must be colocated with the G.711-only device; Cisco CallManager always assumes that a transcoded call uses G.729 across the WAN
- AAR is only invoked in case of call admission control rejection (while the WAN link is up); in case of WAN failure, AAR is not invoked
- The location setting on CTI route points is only used by Cisco CallManager if an application registers to handle media with that route point

Cisco CallManager Static Locations

Bandwidth Provisioning

For More Details, Refer to the QoS SRND and IP Telephony SRND At:
http://www.cisco.com/go/srnd

Provision LLQ PQ with These Values

	CCM Location	L3 Bandwidth	L2 Bandwidth (Frame Relay)
G.711 Audio	80 Kbps (64K + Header)	80 Kbps (64K + Header)	81.6 Kbps (80K + L2 Hdr)
G.729 Audio	24 Kbps (8K + Header)	24 Kbps (8K + Header)	25.6 Kbps (24K + L2 Hdr)
384K Video	384 Kbps (64K + 320K)	460 Kbps (384K + est. L2/L3 Headers)	

Call Admission Control Elements
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- Topology-Unaware CAC
    - Cisco CallManager "Static" Locations
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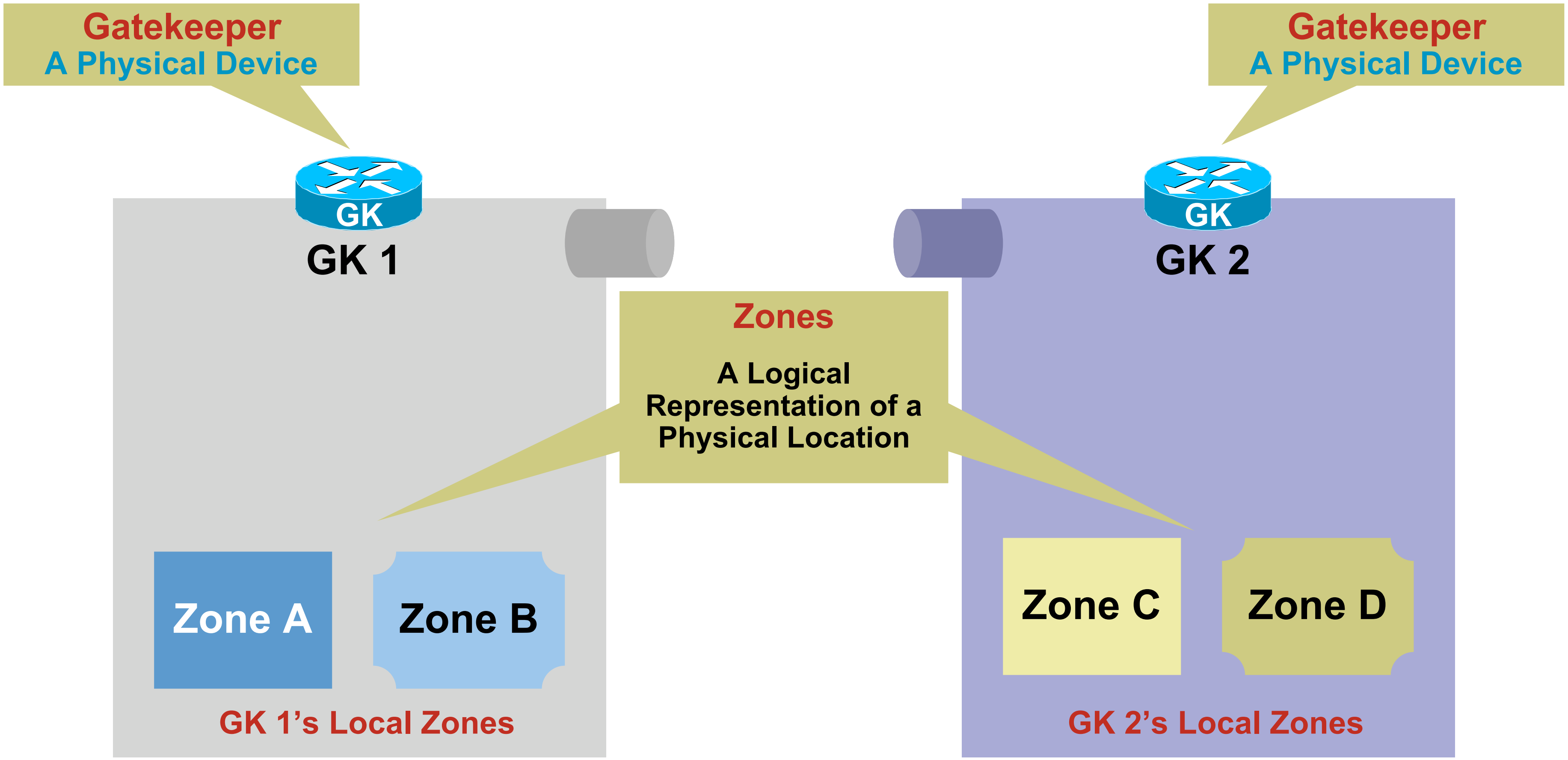
Gatekeeper Zones
Basics

- Cisco IOS feature, based on H.323 RAS protocol
- Can be used between Cisco CallManager clusters, H.323 gateways and H.323 endpoints
- Provides CAC using concept of zones and associated bandwidth counters
- Static configuration approach limits supported topologies (mainly hub-and-spoke)


```
gatekeeper
 zone local A abc.com 10.10.10.10
 zone local B abc.com
 zone remote C abc.com 10.10.20.20
 zone remote D abc.com
 bandwidth interzone zone A 384
 bandwidth interzone zone B 256
 bandwidth remote 512
```

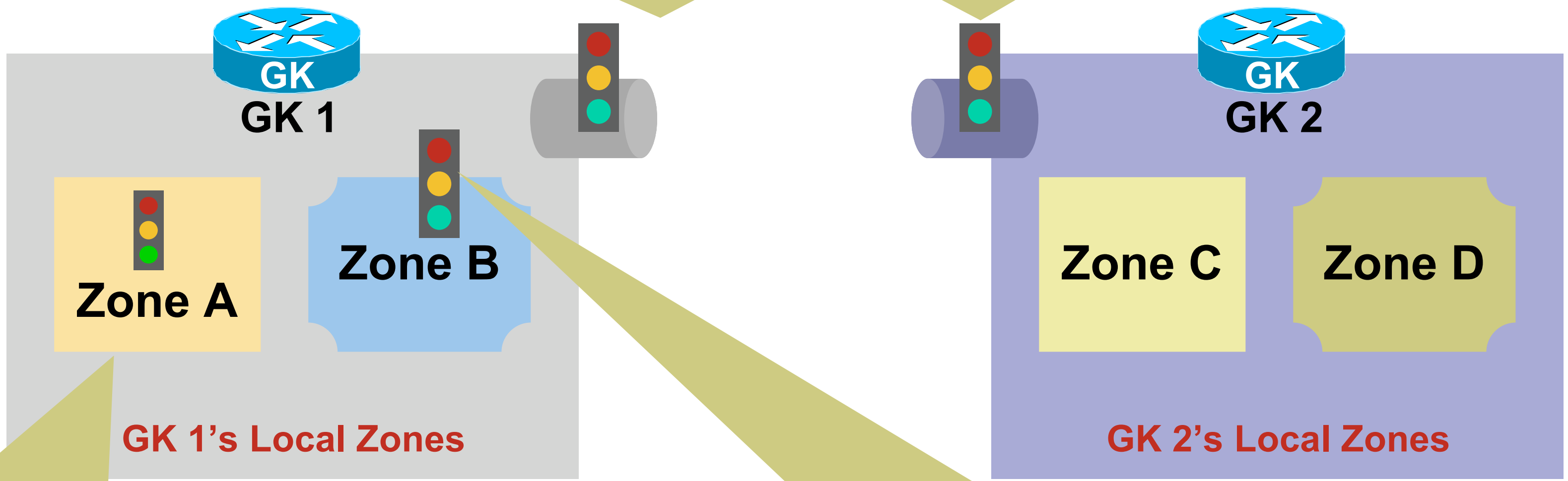

Gatekeeper Zones
Zone Concept



Gatekeeper Zones
Bandwidth Configuration

bandwidth ***session*** **zone** *xyz max-bw*
This Is the Maximum Bandwidth Allowed per Session

bandwidth ***remote max-bw***
The Total Bandwidth Allowed in/out of the Physical GK

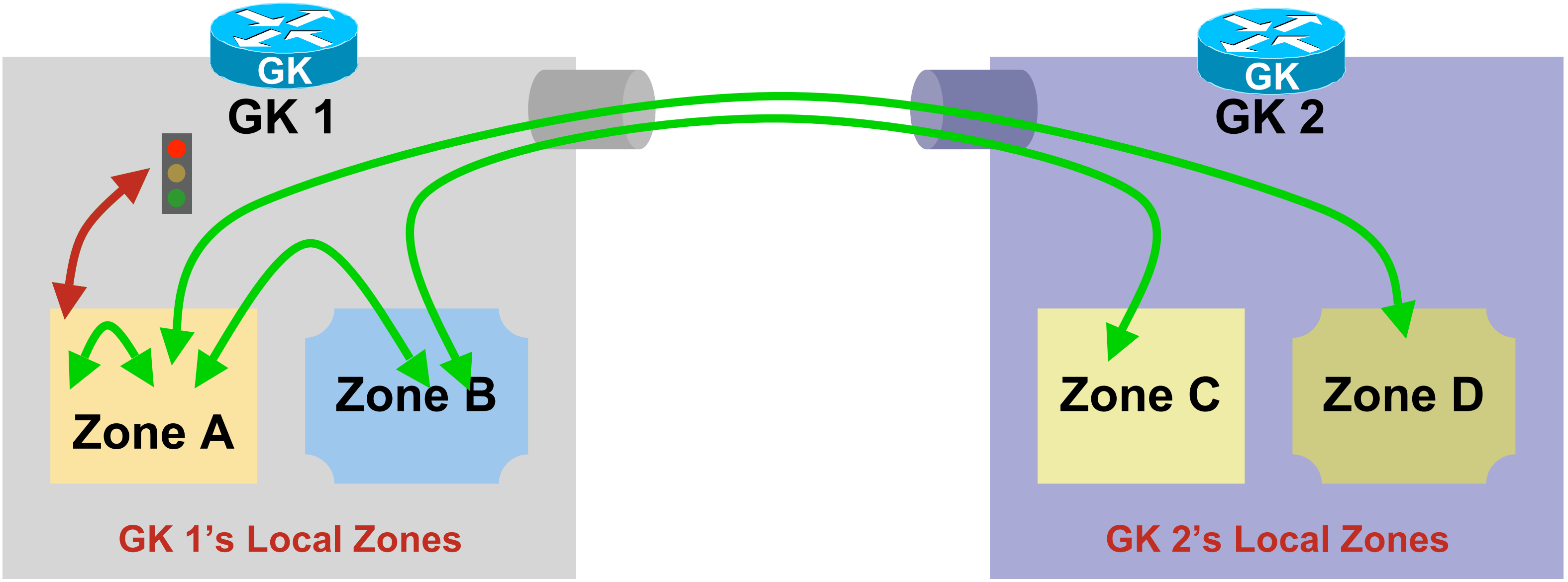


bandwidth ***total*** **zone** *xyz max-bw*
The Total Bandwidth Allowed Within a Zone as Well as in/out of the Zone

bandwidth ***interzone*** **zone** *xyz max-bw*
This Is the Total Bandwidth Allowed in/out of the Zone

Gatekeeper Zones
Bandwidth Calculations

Assume Requested Bandwidth for Each Call Equals 16K



GK1
Remote = 32K In Use = 32

Zone A
InterZone = 32K In Use = 32
Total = 48K In Use = 48 X

Zone B
InterZone = 48K In Use = 32
Total = 48K In Use = 32

Session = 16K

Blue Text Represents Configured Bandwidth

GK2
Remote = 48K In Use = 32

Zone C
InterZone = 32K In Use = 16
Total = 32K In Use = 16

Zone D
InterZone = 32K In Use = 16
Total = 32K In Use = 16

Session = 16K

Gatekeeper Zones
Bandwidth Provisioning

For More Details, Refer to the Qos SRND and IP Telephony SRND At:
http://www.cisco.com/go/srnd

Provision LLQ PQ with These Values

	Gatekeeper	L3 Bandwidth	L2 Bandwidth (Frame Relay)
G.711 Audio	128 Kbps (64K x 2)	80 Kbps (64K + Header)	81.6 Kbps (80K + L2 Hdr)
G.729 Audio	16 Kbps (8K x 2)	24 Kbps (8K + Header)	25.6 Kbps (24K + L2 Hdr)
384K Video	768 Kbps (384K x 2)	460 Kbps (384K + est. L2/L3 Headers)	

Call Admission Control Elements
Agenda

- Topology-Unaware CAC

  Cisco CallManager “Static” Locations

  Gatekeeper Zones

- Resource Reservation Protocol (RSVP)
- Topology-Aware CAC

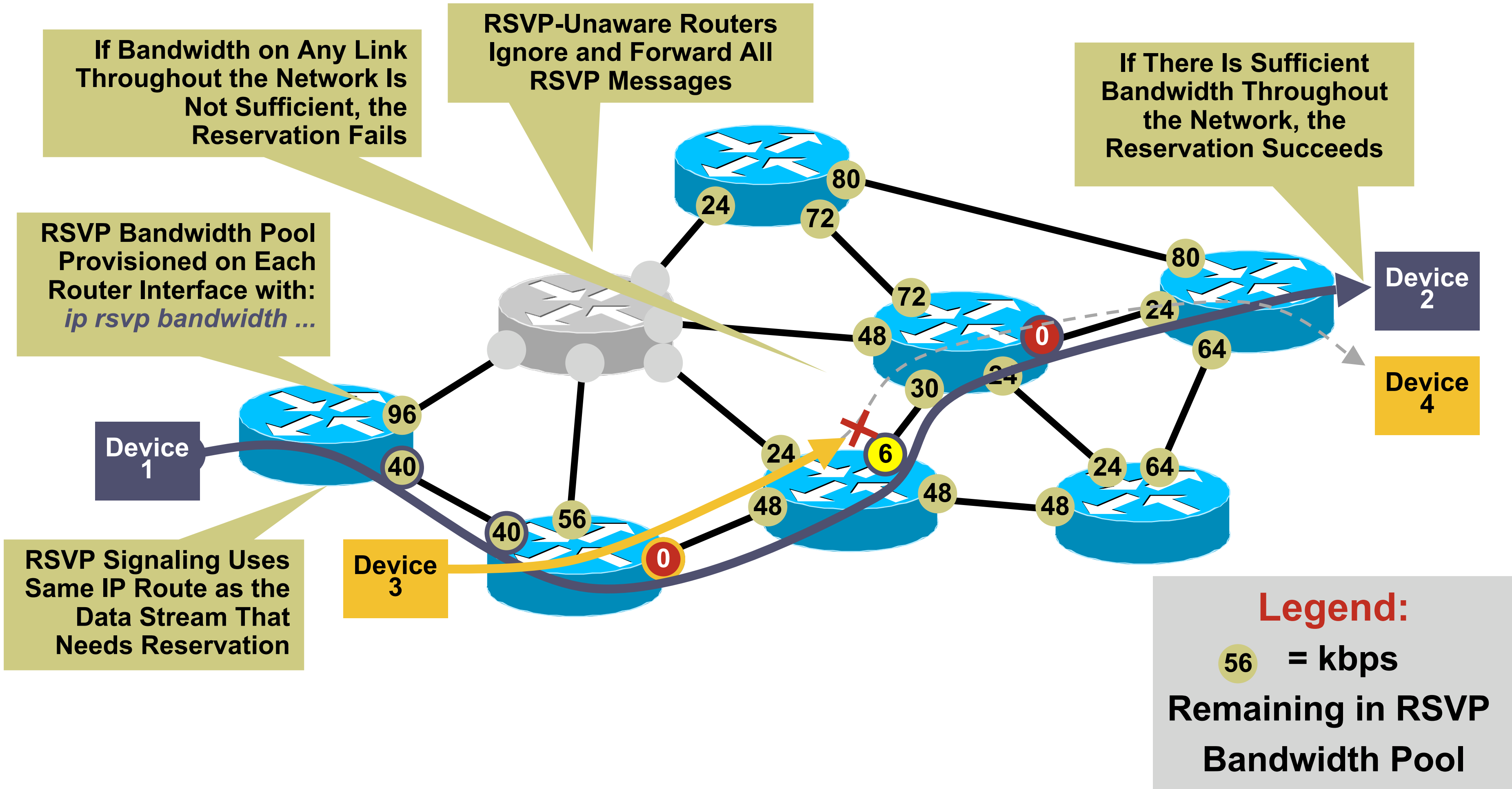
  Cisco CallManager RSVP-Enabled Locations

  Gatekeeper and IP-IP Gateway with RSVP

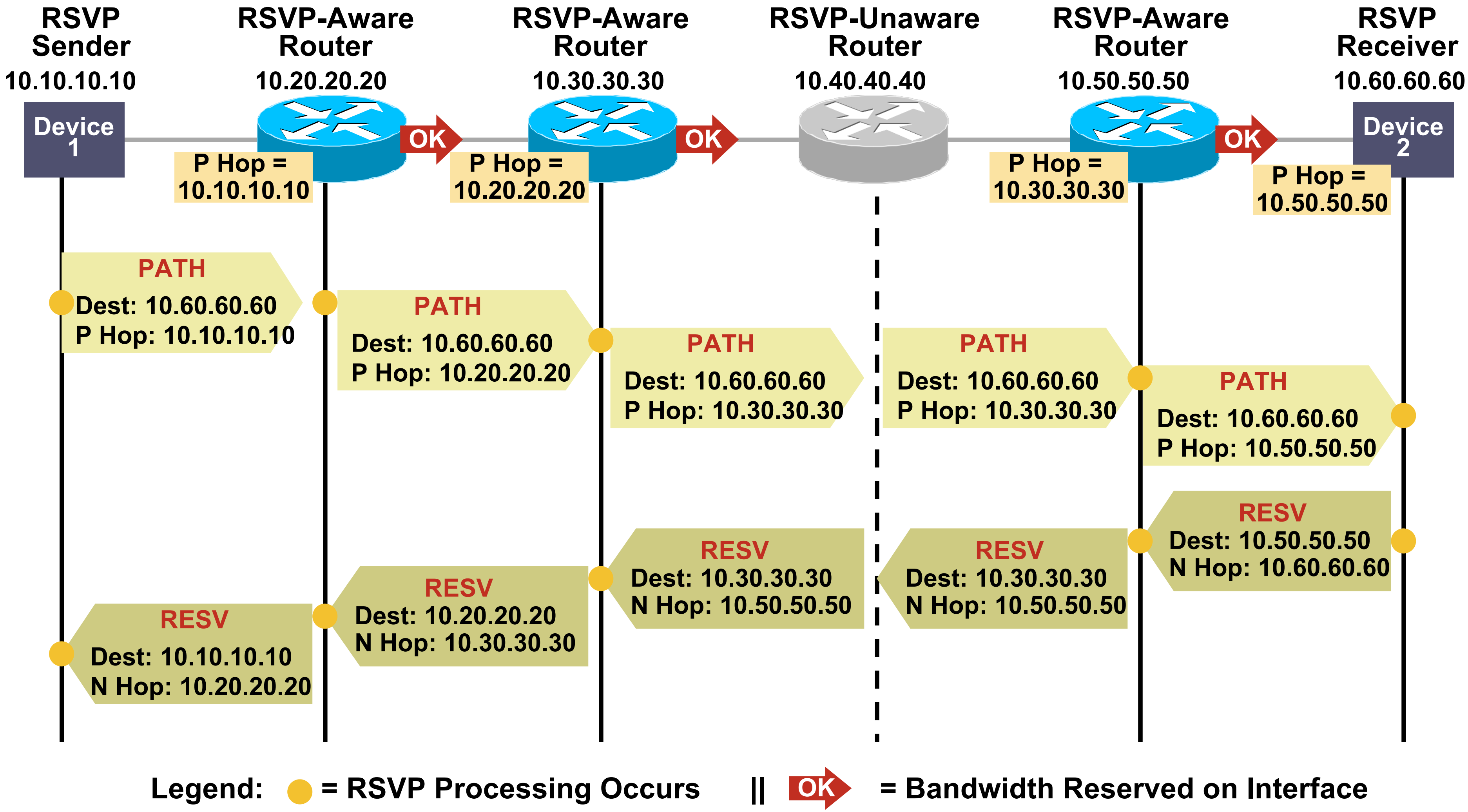
RSVP
(Resource ReSerVation Protocol): Principles

- IETF standard
    - RFC2205, RFC2207, RFC2208, RFC2209, RFC2210, and others
- Topology-aware CAC signaling protocol
    - Works with any WAN topology
- Uses existing routing protocols
    - Dynamically adjusts to link failures and topology changes
- Unidirectional reservations
    - Reservations are receiver-initiated
- Maintains “soft state” in RSVP-enabled routers
- Operates transparently across non-RSVP routers
    - Allows for partial or gradual deployment across network

RSVP Principles



RSVP
Path and Resv Messages



RSVP
Other Messages

- PathTear

  Tears down Path state

- ResvTear

  Tears down Resv state

- PathErr

  Signals errors within Path message

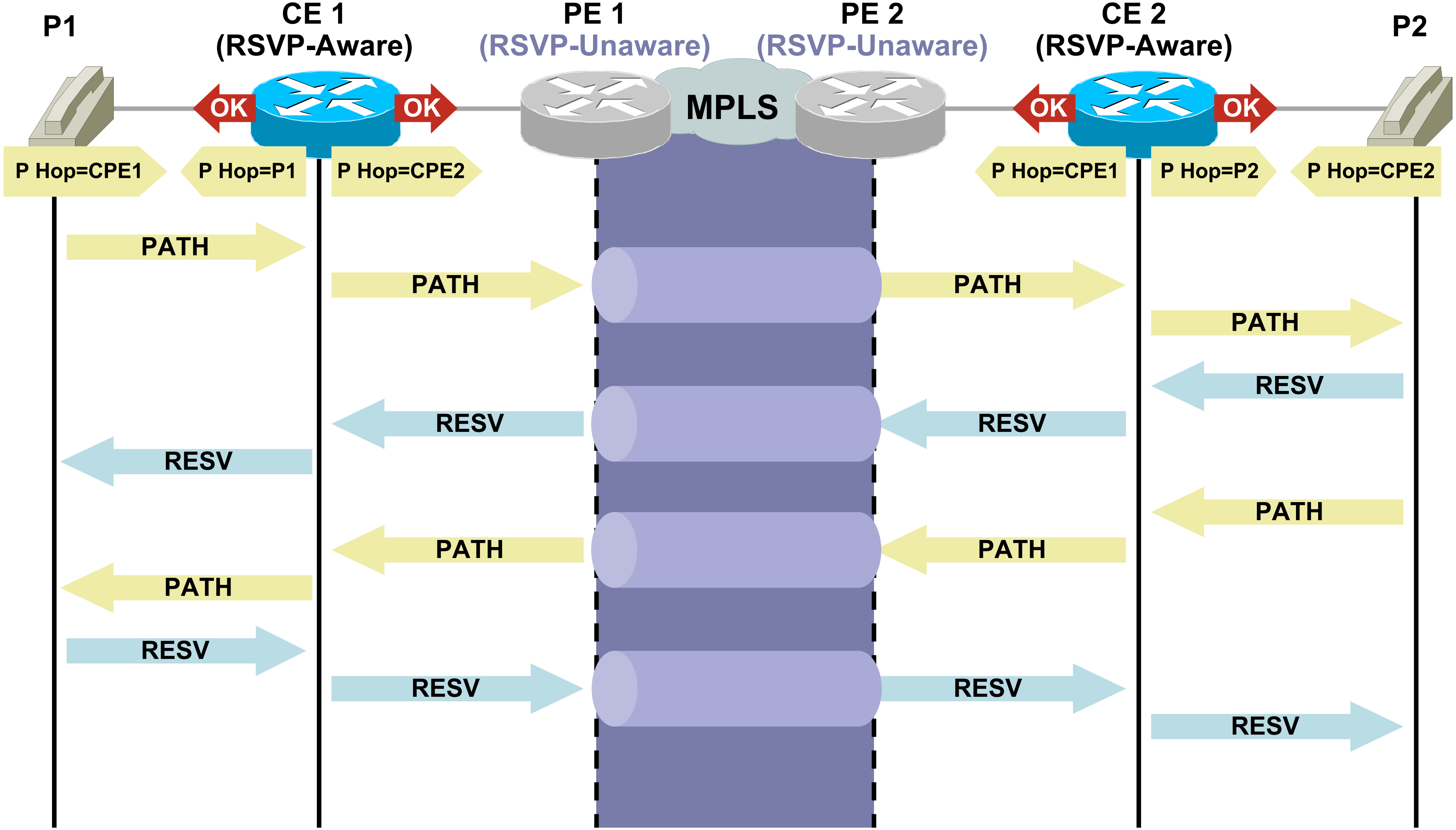
- ResvErr

  Signals admission control failure or other error

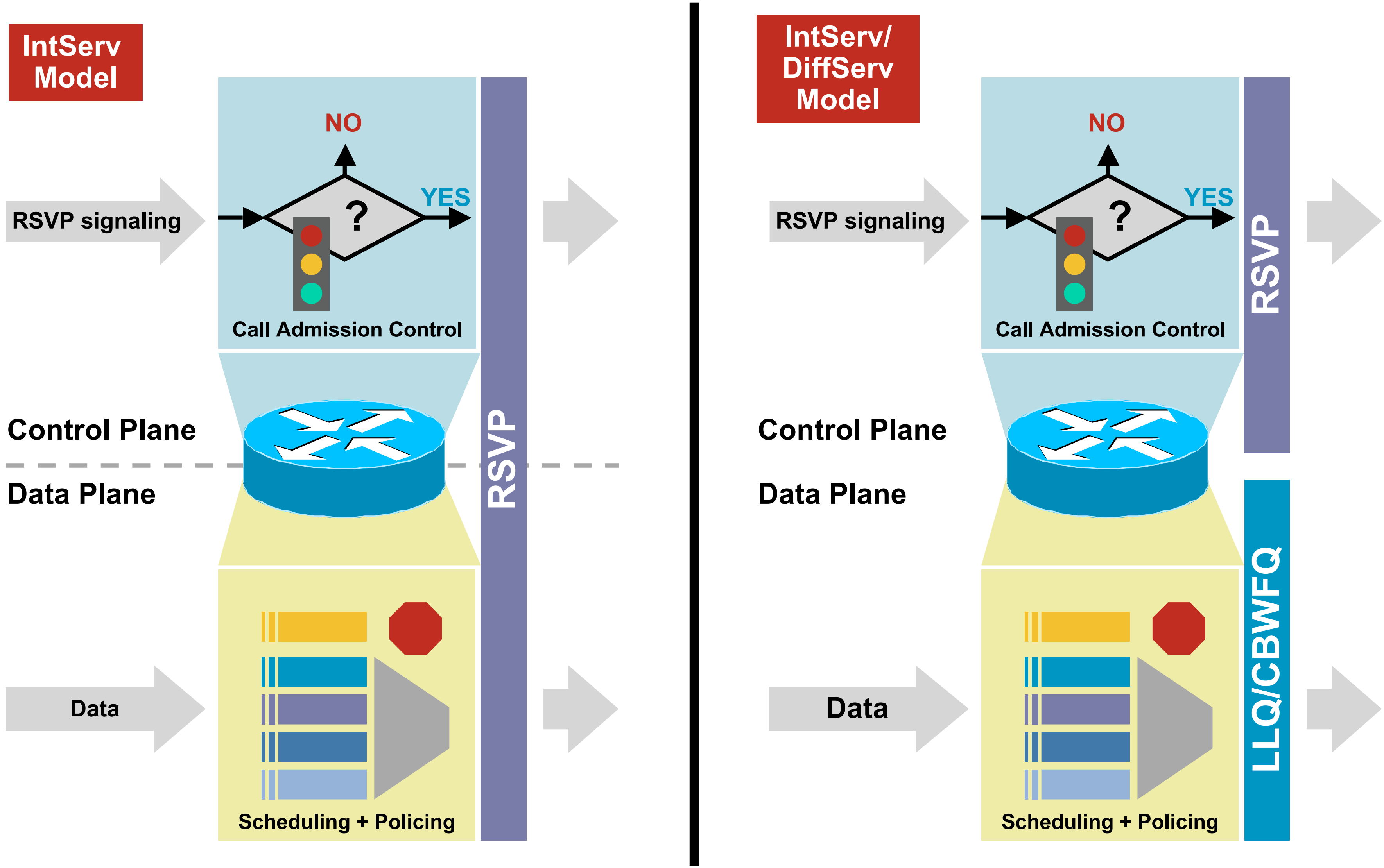
- ResvConf

  Confirms end-to-end reservation (optional)

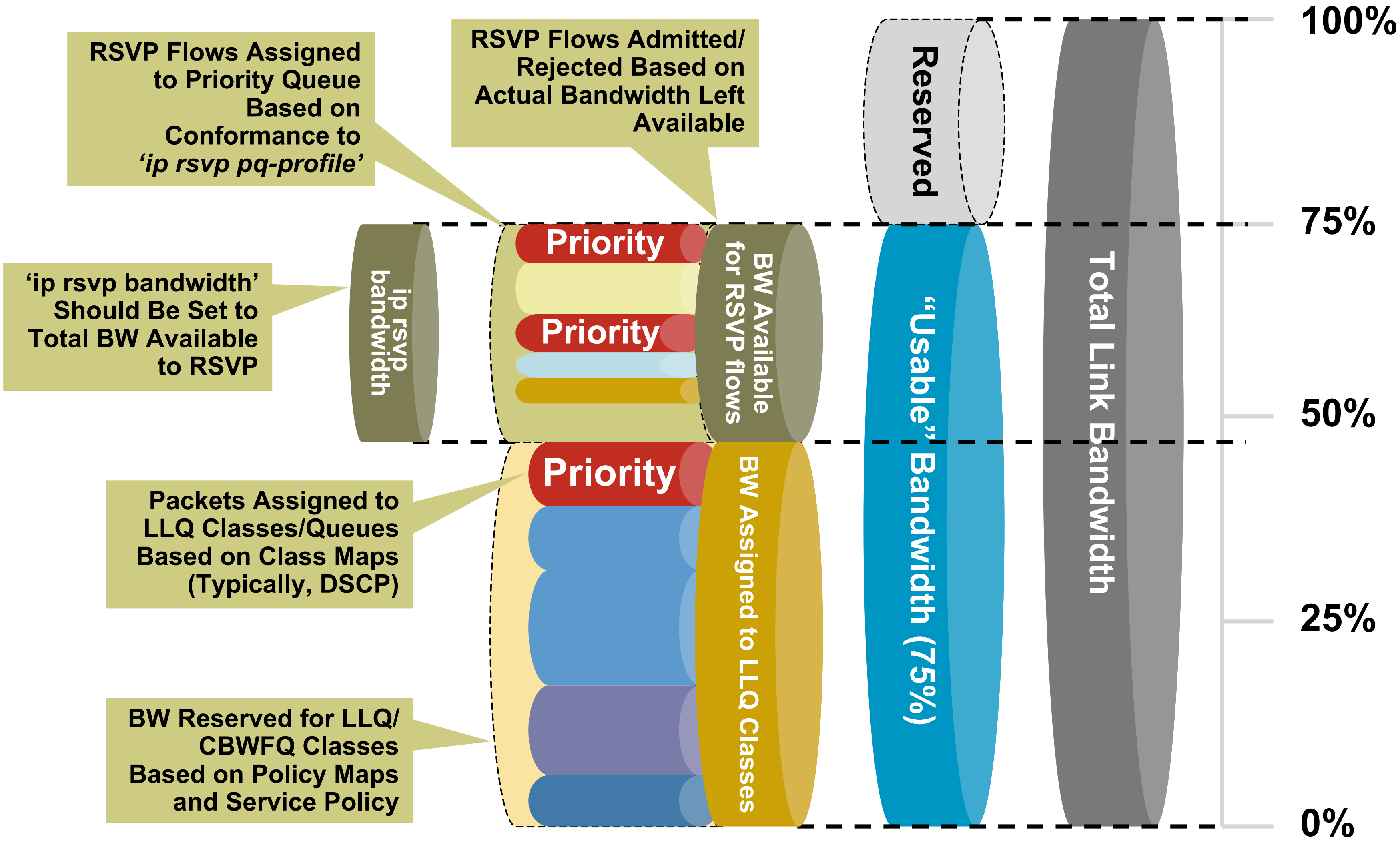
RSVP
Example: Phone Call Across MPLS-Based WAN



RSVP
IntServ vs. IntServ/DiffServ Operation Models IOS



RSVP
IntServ Cisco IOS Model—Interface Queuing

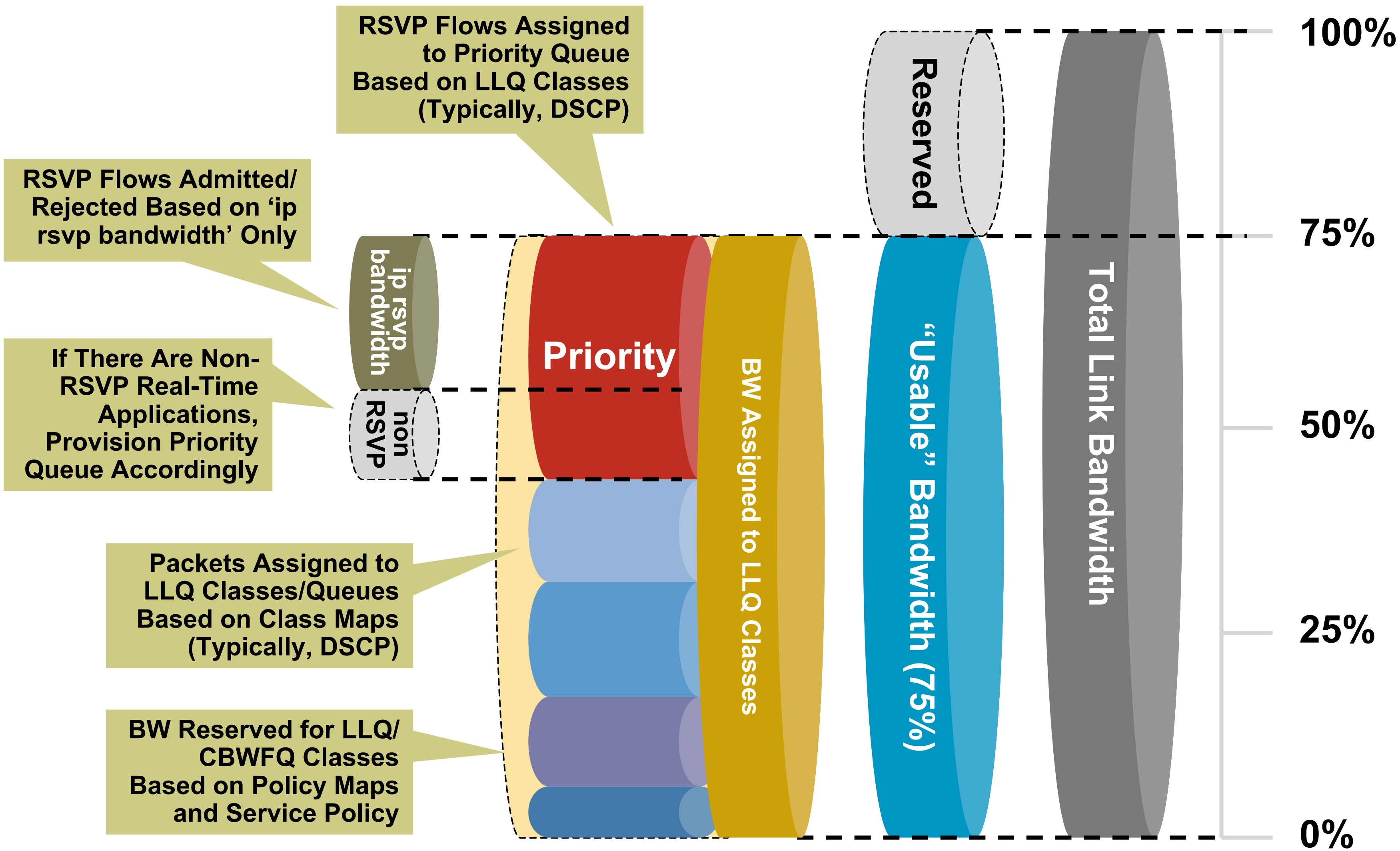


RSVP
IntServ Cisco IOS Model: Notes

```
ip rsvp resource-provider wfq [interface | pvc]
no ip rsvp data-packet classification
ip rsvp pq-profile 12288 592 110
```

- LLQ/CBWFQ classes can be configured as usual and bandwidth allocated to them on the interface
- No bandwidth is reserved with ip rsvp bandwidth: need to leave corresponding amount of bandwidth unallocated to LLQ/CBWFQ classes
- Reservations accepted/rejected based on value configured in ip rsvp bandwidth and on actual bandwidth left unallocated on the interface
- RSVP traffic assigned to priority queue if it conforms to profile defined in ip rsvp pq-profile

RSVP
IntServ/DiffServ Cisco IOS Model—Interface Queuing

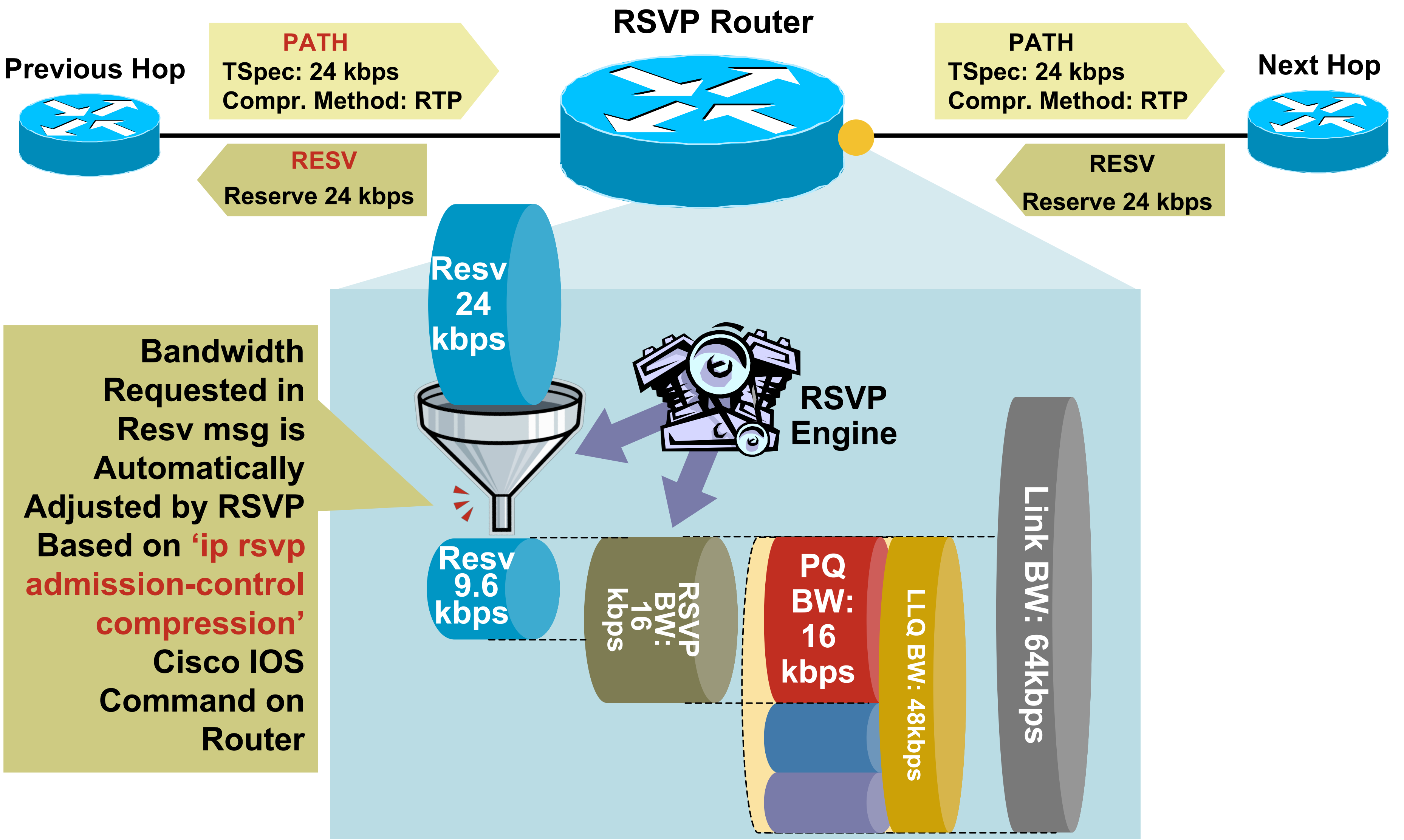


RSVP
IntServ/DiffServ Cisco IOS Model: Notes

```
ip rsvp resource-provider none
ip rsvp data-packet classification none
```

- LLQ/CBWFQ classes can be configured as usual and bandwidth allocated to them on the interface
- No bandwidth is reserved with ip rsvp bandwidth
- Reservations accepted/rejected based exclusively on value configured in ip rsvp bandwidth
- RSVP traffic assigned to queues based on LLQ rules (RSVP is not involved in classification)
- Must ensure that non-RSVP real-time applications, if present, use a different CAC mechanism to avoid overrunning the priority queue

RSVP
Support for Compressed RTP (cRTP)



RSVP
Bandwidth Provisioning

- Bandwidth requested by each call is determined by the TSpec object in the RSVP reservation
- RSVP reservation occurs prior to media negotiation, hence Cisco CallManager uses worst-case assumption based on the CODEC indicated in the region configuration

  (This is to avoid having to adjust the reservation ‘up’ and thus risk a post-ring failure)

- Reservation is adjusted ‘down’ to correct value once call is established and media negotiation has occurred

→ Need to over-provision ‘ip rsvp bandwidth’ and PQ in order to accommodate larger initial reservation

RSVP
Bandwidth Provisioning—Requested TSpecs

- Voice calls (initially assume 10ms samples):

  G.711: request 96kbps
  (drops to 80kbps after call setup, with default 20ms sampling)

  G.729: request 40kbps
  (drops to 24kbps after call setup, with default 20ms sampling)

- Video calls—video stream reservation:

  Call rate ≤ 256kbps: request call rate + 20%
  (drops to (call rate – audio bandwidth) + 20% after call setup)

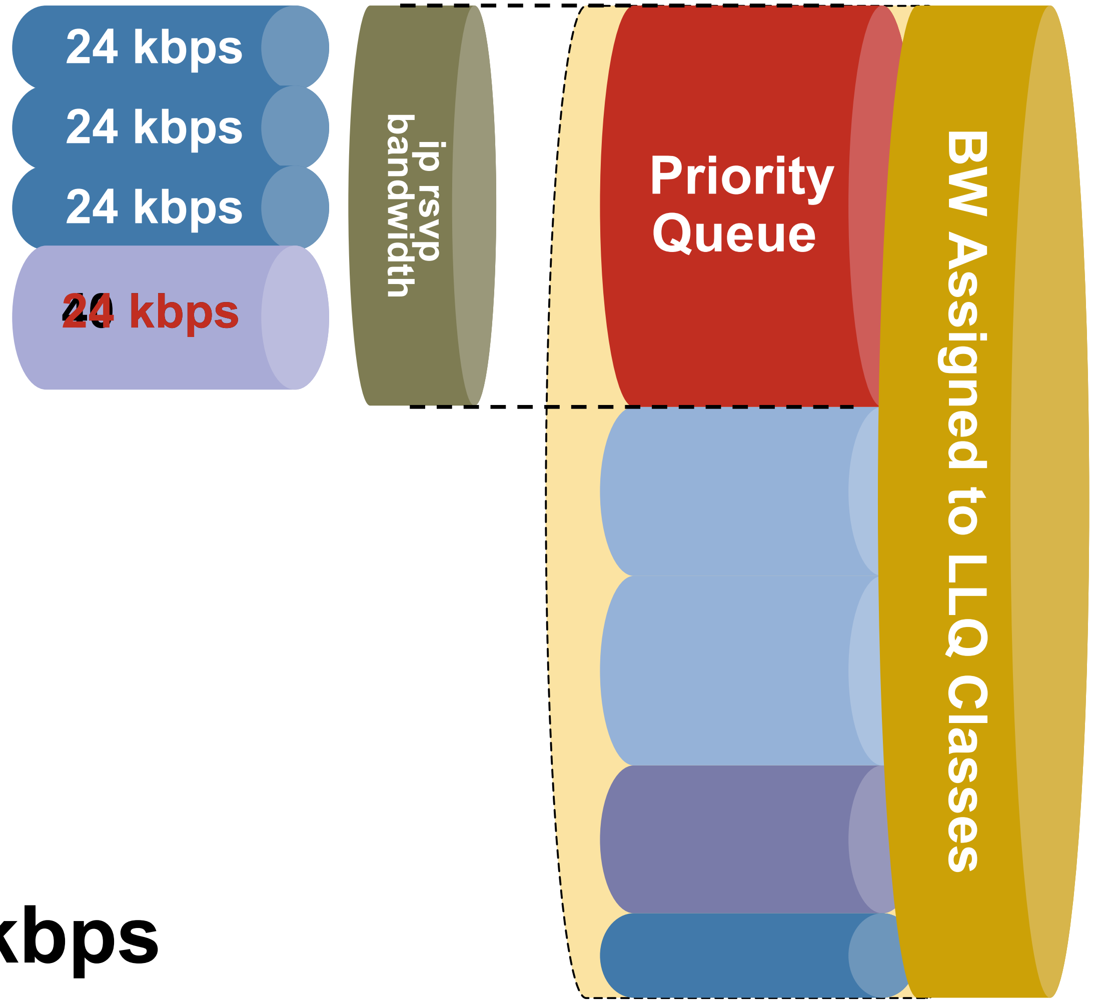
  Call rate > 256kbps: request call rate + 7%
  (drops to (call rate – audio bandwidth) + 7% after call setup)

- Video calls—audio stream reservation:

  Same as for voice calls

RSVP
Example of Bandwidth Provisioning

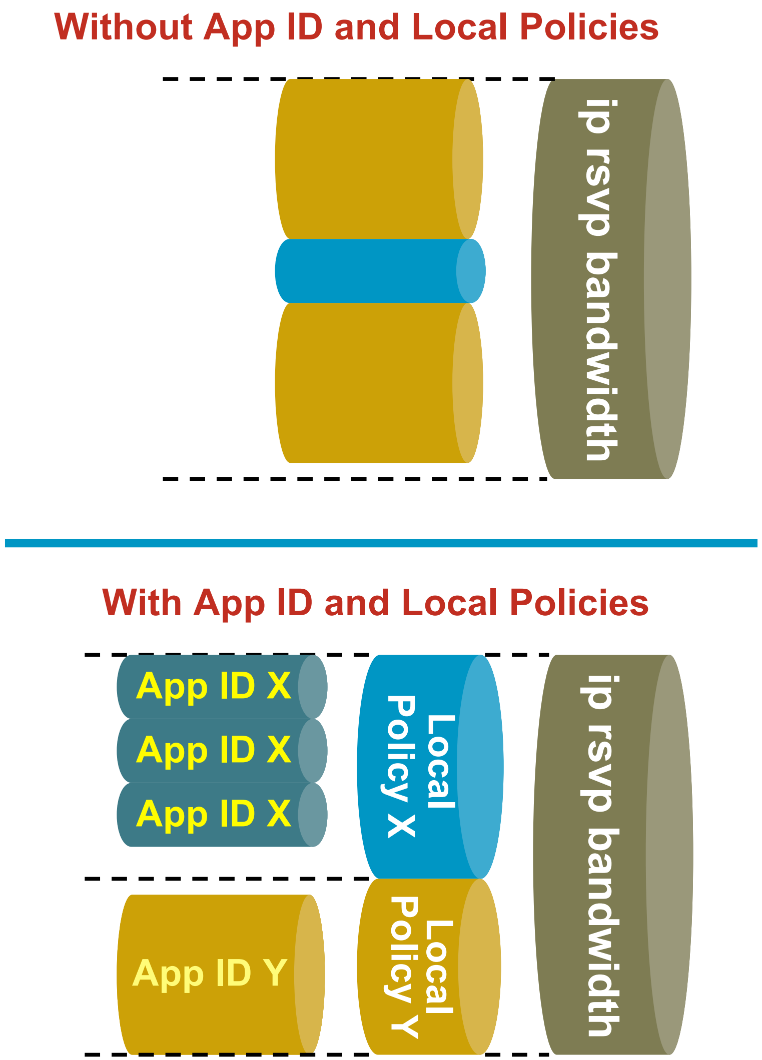
- Need to leave enough "room" for the Nth reservation to succeed using Cisco CallManager's worst-case assumption
- For four calls at G.729, provision RSVP bandwidth and PQ as:



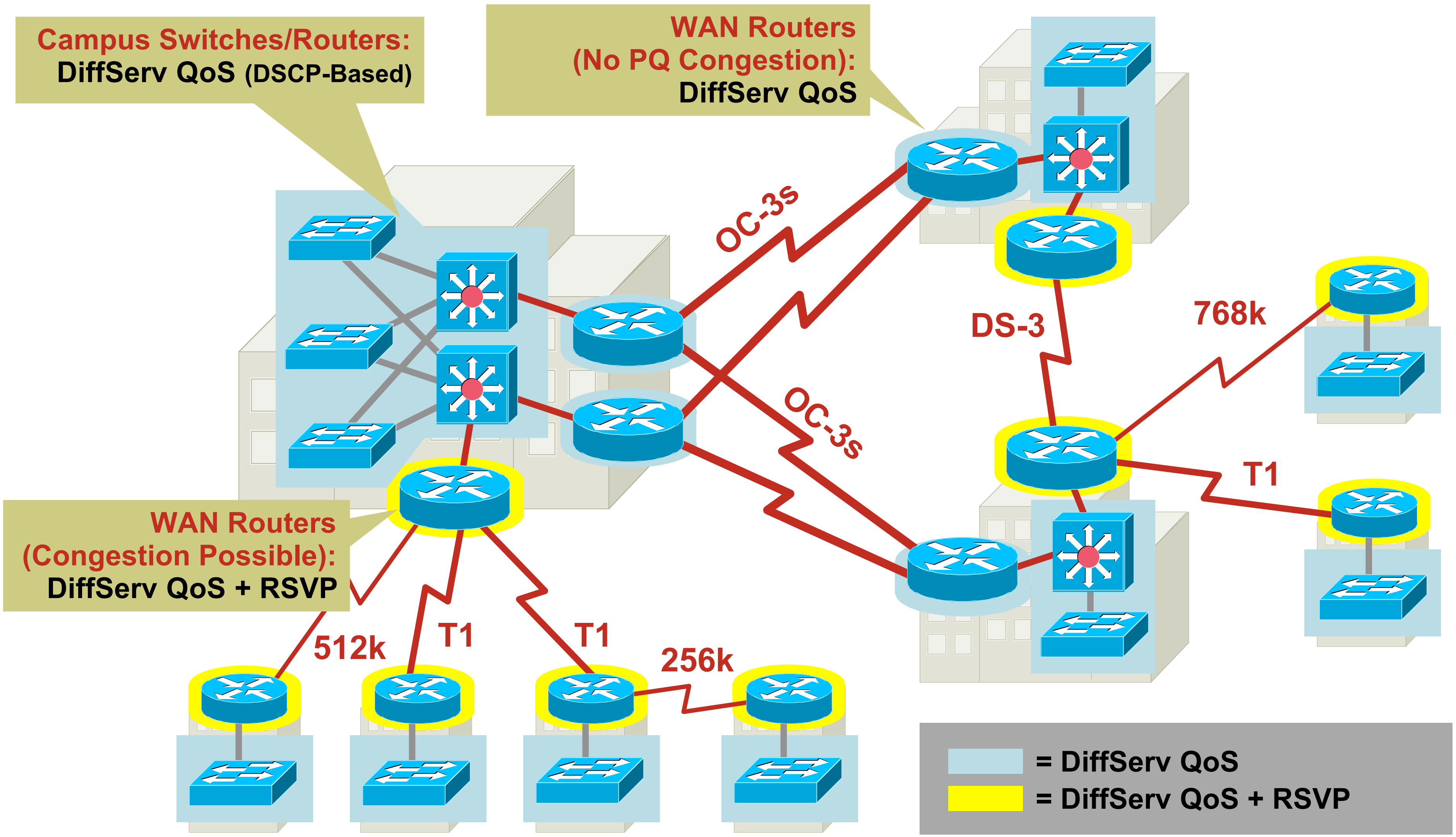
(24*3 + 40) = 112 kbps

RSVP
Application ID (App ID) Object

- Introduced in RFC2872 as part of RSVP Policy Element—associates RSVP reservation with a specific application and sub-application
- Allows routers to admit reservations based on the application requesting bandwidth
- Protects bandwidth resources across applications
- In Cisco IOS, app IDs are associated to RSVP local policies which define corresponding behavior
- Usage example: in presence of voice and video calls, prevent video calls from using all available bandwidth



RSVP
Where to Enable RSVP?



RSVP
Design Best Practices

- The IntServ/DiffServ operation model is the recommended choice for enterprise customers
- Easy mapping onto existing DiffServ QoS configurations (LLQ classes based on DSCP values)
- Least impact on router CPU
- In Cisco CallManager-based deployments, all traffic destined to the PQ should be controlled by RSVP
- RSVP bandwidth accounts for L3 overhead, while PQ bandwidth also accounts for L2 overhead

→ Set 'ip rsvp bandwidth' = PQ size - L2 overhead

RSVP
Design Best Practices (Cont.)

- Enable RSVP application ID support if you need to limit the maximum amount of bandwidth used by video calls
- Bundle interfaces, including MLPPP, ATM-IMA, and FRF.16, should have the RSVP bandwidth configured to the size of one physical link

Caveats:

- RSVP is currently not available on tunnel interfaces
- RSVP is currently not available on the Cisco Catalyst® Switching Platforms

Call Admission Control Elements
Agenda

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  - Cisco CallManager “Static” Locations
  - Gatekeeper Zones
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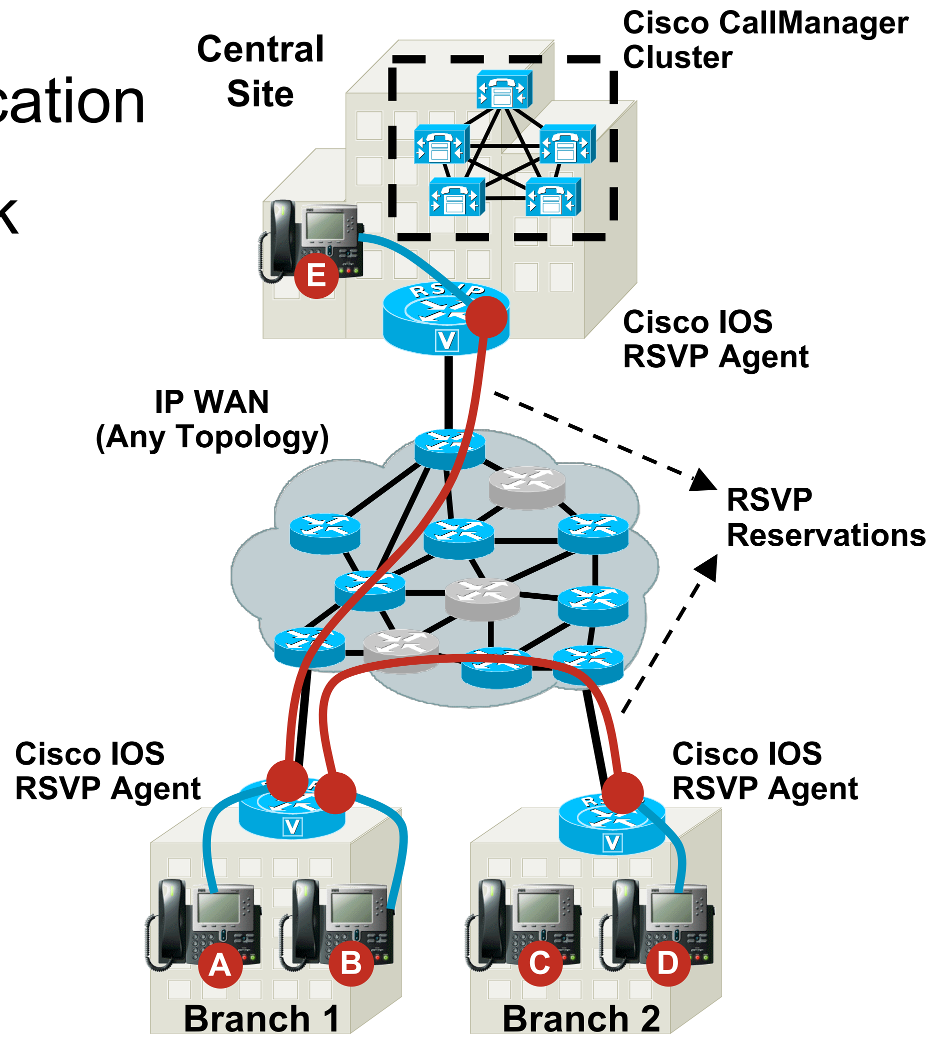
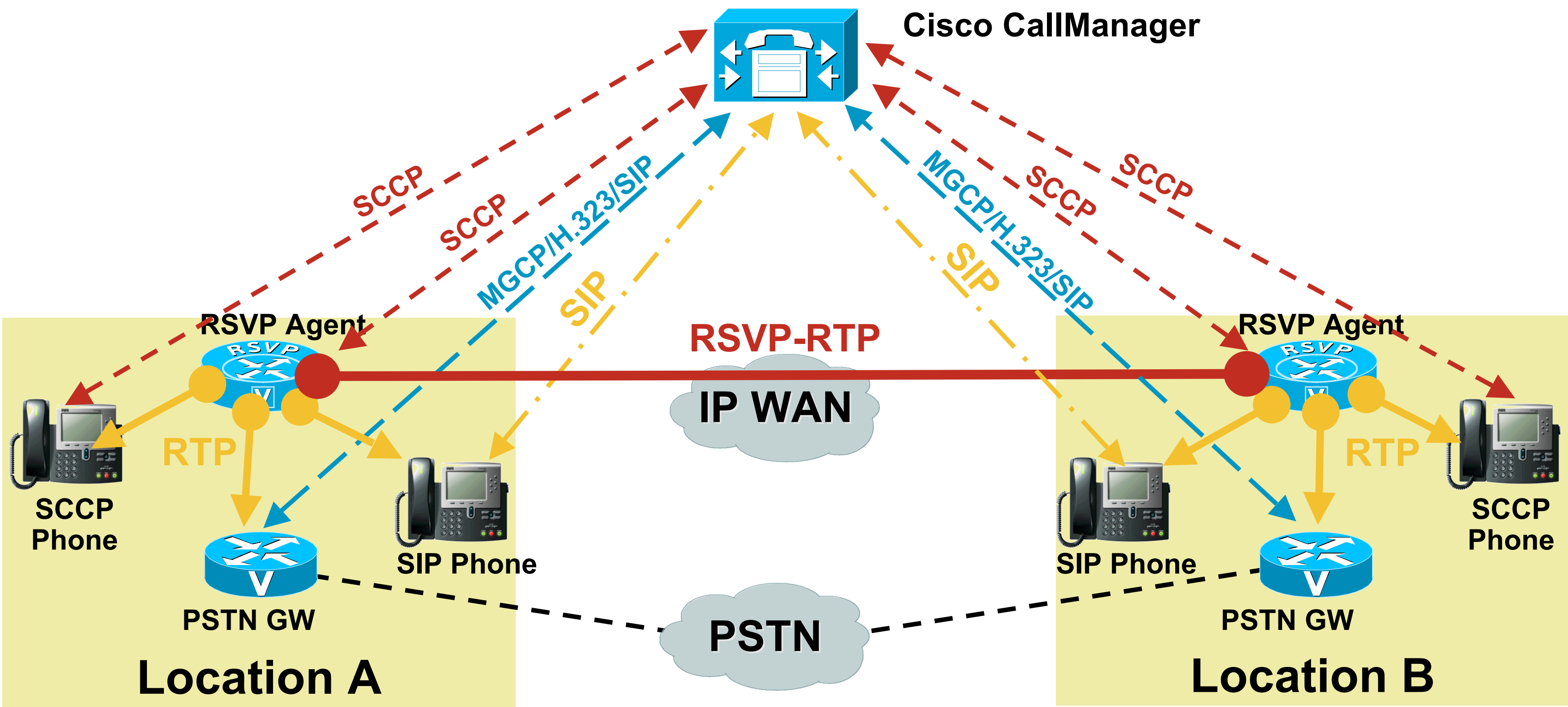
CallManager RSVP-Enabled Locations
The RSVP Agent Concept

- Enable RSVP at each location
- Applicable to any network topology
- RSVP agent acts as a proxy to make bandwidth reservations



Cisco CallManager and RSVP
Protocol Flows with RSVP Agent



Solution Requires:

- Cisco CallManager 5.0(2) or later
- Cisco 26xxXM, 2691, 37xx, 28xx, 38xx routers
- Cisco IOS 12.4(6)T or later

CallManager RSVP-Enabled Locations

Policy Recommendations

System > Service Parameters > Cisco CallManager:

Clusterwide Parameters (System - RSVP)

Default inter-location RSVP Policy *	Mandatory	No Reservation
RSVP Retry Timer *	60	60
Mandatory RSVP Mid-call Retry Counter *	1	1
Mandatory RSVP mid call error handle option *	Call fails following retry counter exceeded	Call becomes best effort

- "Mandatory" RSVP policies are recommended (equivalent to current location behavior) → call fails or reverts to AAR when RSVP reservation fails
- "Mandatory (video desired)" allows a video call to proceed as audio-only if not enough bandwidth
- For mid-call failures (e.g., blind transfer, decrease of bandwidth in the network), shorten retry times
- Call admission control protects network resources, but does not replace proper traffic engineering

Cisco CallManager and RSVP
RSVP Agent Scalability

- Requires either SRST license or IP-IP gateway license
- Entitlement:

  SRST: depends on router performance and licensed sessions

  IPIPGW: depends on router performance only
- Note: platform capacity estimated assuming router dedicated to RSVP agent and 75% CPU utilization; addition of other applications will reduce supported sessions

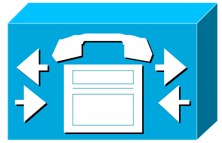
Platform Capacity Tables (IOS 12.4(6)T):

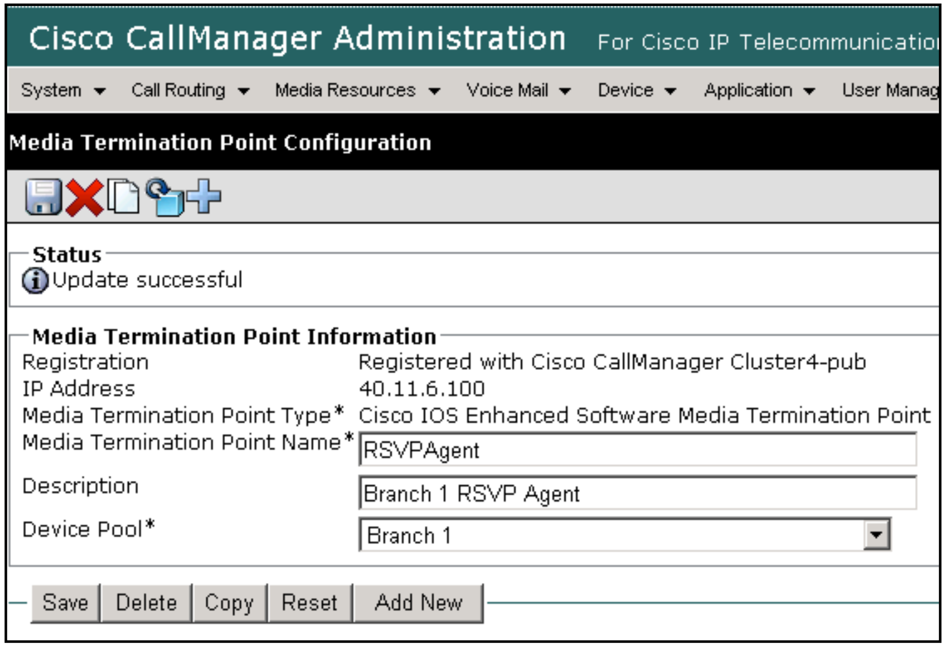
Platform	Max. # Sessions
2611XM	40
2621XM	50
2651XM	65
2691	150
2801	130
2811	180

Platform	Max. # Sessions
2821	240
2851	300
3725	250
3745	320
3825	400
3845	536

Cisco CallManager and RSVP
RSVP Agent Configuration Example


```
interface Loopback0
 ip address 40.11.6.100 255.255.255.255
!
sccp local Loopback0
sccp ccm 20.11.1.50 identifier 1 priority 1 version 5.0.1
sccp ccm 20.11.1.51 identifier 2 priority 2 version 5.0.1
sccp
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate ccm 2 priority 2
 associate profile 1 register RSVPAgent
 switchover method immediate
 switchback method guard timeout 7200
!
dspfarm profile 1 mtp
 codec pass-through
 codec g729ar8
 rsvp
 maximum sessions software 100
 associate application SCCP
```



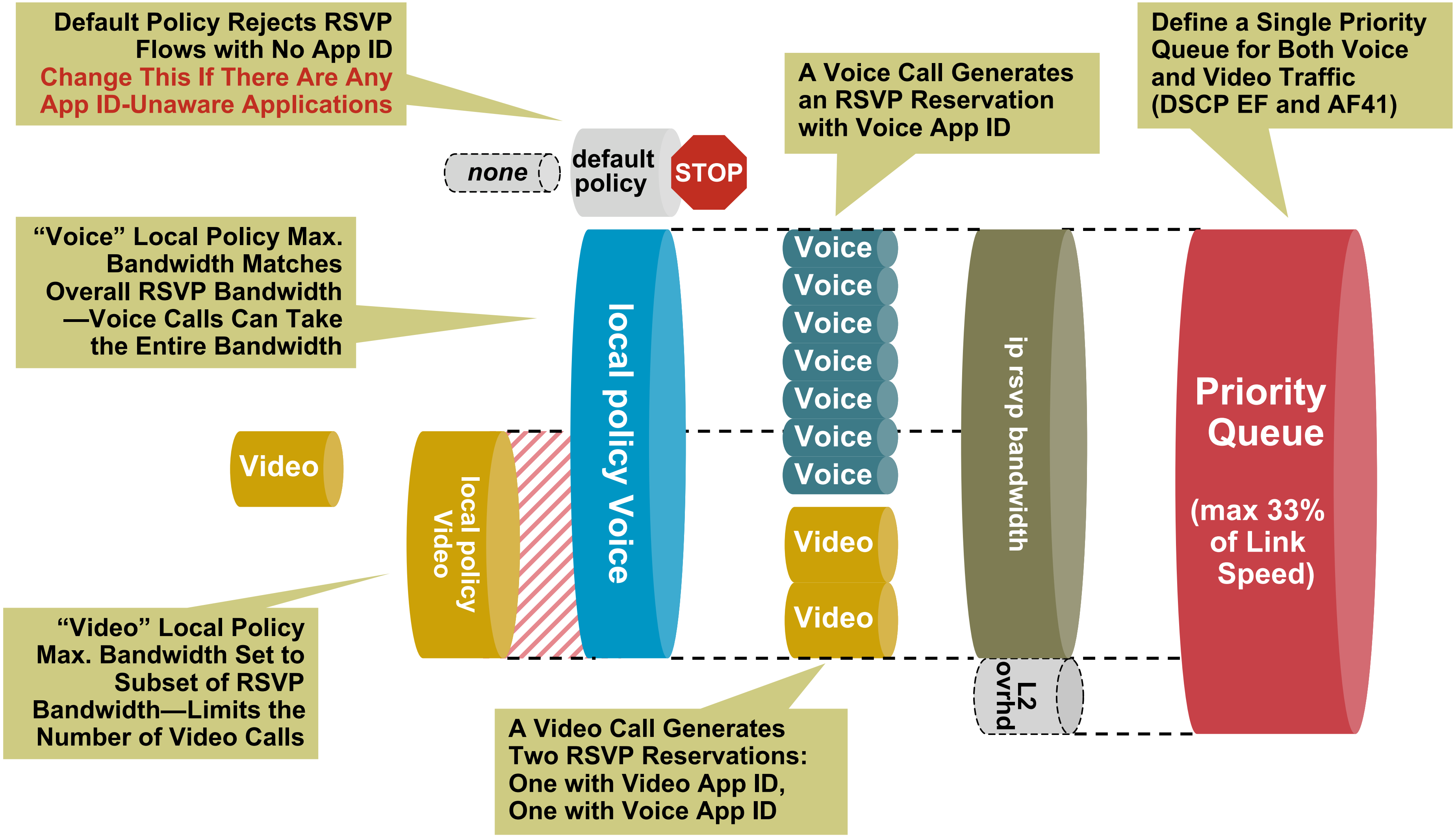


Cisco CallManager and RSVP
How CallManager Uses RSVP Application IDs

- Cisco CallManager uses two RSVP application IDs:
  - “AudioStream” : used for audio streams of voice and video calls
  - “VideoStream” : used for video streams of video calls
- Current implementation allows you to limit the maximum number of video calls across a link
- Voice calls are guaranteed a minimum but can “expand” to full capacity of the priority queue if bandwidth is available
- Video calls (both streams) are marked AF41, voice calls marked EF
  - Need to use a single class-map statement for both when configuring LLQ

Cisco CallManager and RSVP

RSVP Local Policies for Cisco CallManager App ID



Cisco CallManager and RSVP
Application ID Configuration Example

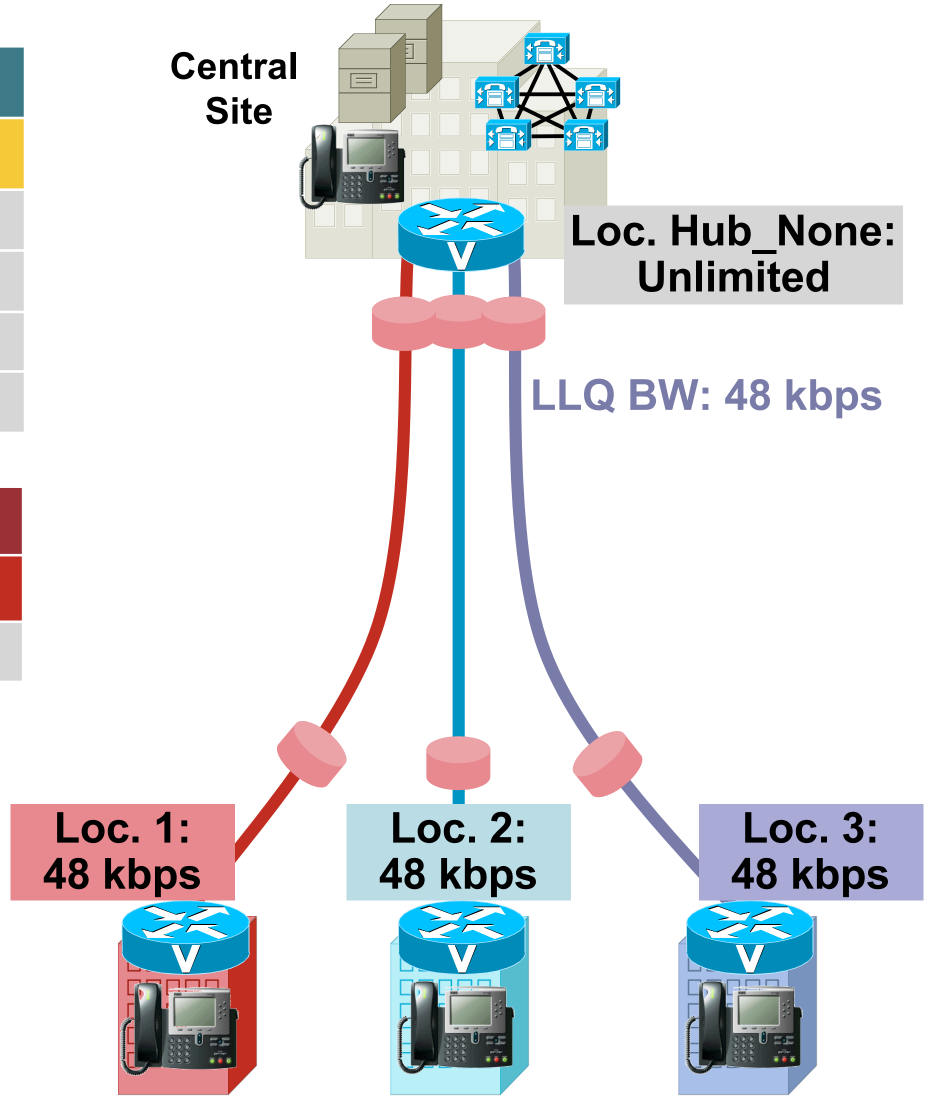
```
class-map match-any IPC-RTP
  match ip dscp ef                              ! Place both voice traffic (DSCP EF) and
  match ip dscp af41                            ! video traffic (DSCP AF41) in same class!
policy-map VoiceVideo-Policy
  class IPC-RTP
    priority percent 33                         ! Define single priority queue
!
! Match Cisco CallManager Application ID strings
ip rsvp policy identity rsvp-video policy-locator .*VideoStream.*
ip rsvp policy identity rsvp-voice policy-locator .*AudioStream.*
!
interface Serial0/0/1:0
  ip address 10.2.101.5 255.255.255.252
  service-policy output VoiceVideo-Policy
  ip rsvp bandwidth 506                         ! Overall RSVP bandwidth pool is 1/3 (33%)
  ip rsvp data-packet classification none
  ip rsvp resource-provider none
  ip rsvp policy local identity rsvp-voice
    maximum bandwidth group 506                 ! Voice streams may use the entire BW pool
    forward all
  ip rsvp policy local identity rsvp-video
    maximum bandwidth group 320                 ! Video streams are limited to 320 kbps
    forward all
```

Migration from Static Locations
Initial Deployment

Locations Bandwidth	
Location	**Bandwidth**
Hub_None	Unlimited
Loc. 1	48
Loc. 2	48
Loc. 3	48

RSVP Policy		
Locations Pair		**Policy**
Any	Any	None

- One location per spoke site
- Devices at hub site in the Hub_None location



Migration from Static Locations

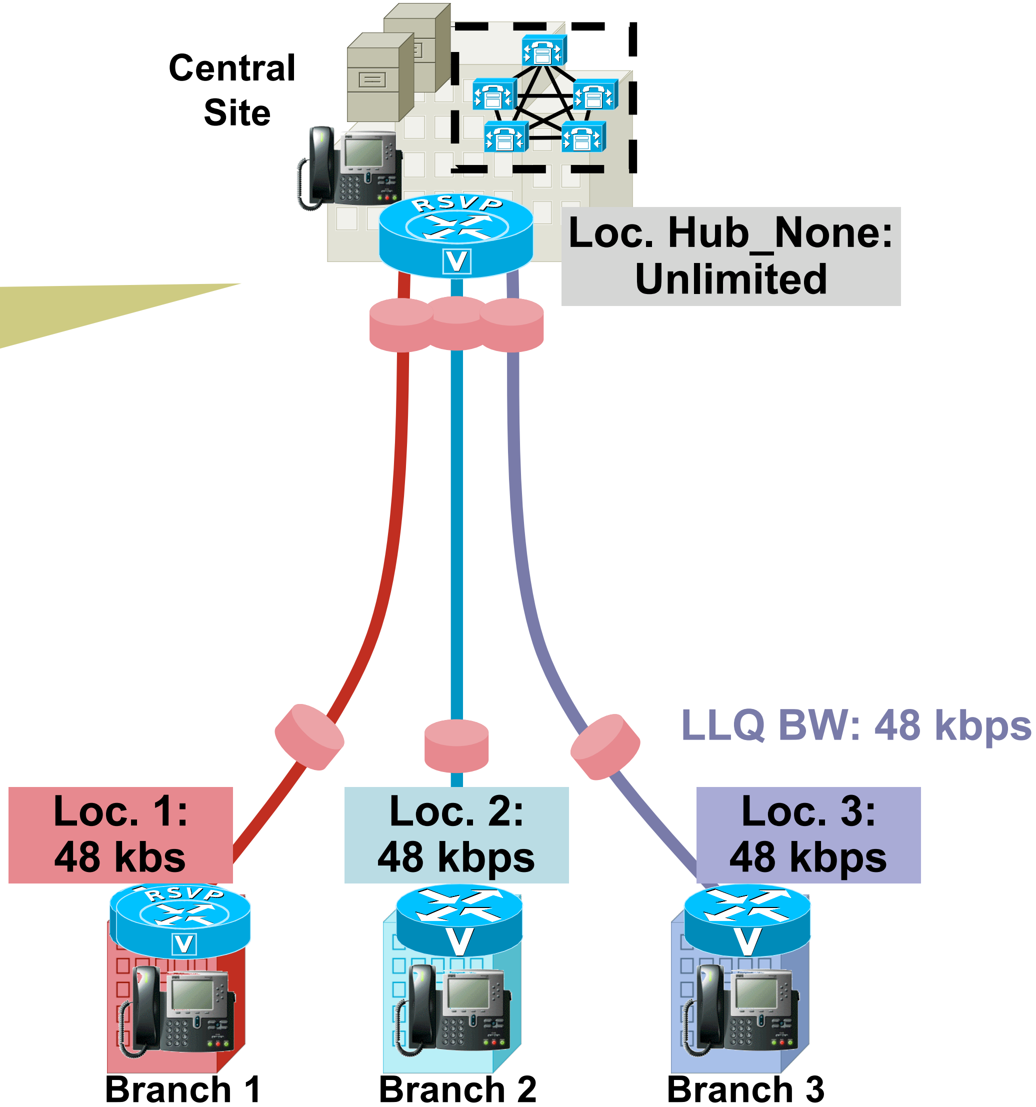
Migrating the First Location

- Install an RSVP agent in the location to be migrated
- Add the agent to the MRGLs of all devices at that location
- Install a second RSVP agent at location Hub_None
- Add the second agent to the MRGLs of all devices at locations that aren't going to be migrated yet
- Be sure the second agent isn't in the "none" MRG or the MRGL of the location that is being migrated
- Configure RSVP policy (mandatory/optional etc.) for locations pair: location to be migrated to other locations

Migration from Static Locations

Migrating the First Location

The RSVP Agent at the Hub_None Location Must Be Included in the MRGL of Locations 2 and 3 as Well as Hub_None



RSVP Policy		
Locations Pair		Policy
Within Any Location		None
Loc. 1	Any	RSVP
All Other Locations	All Other Locations	None

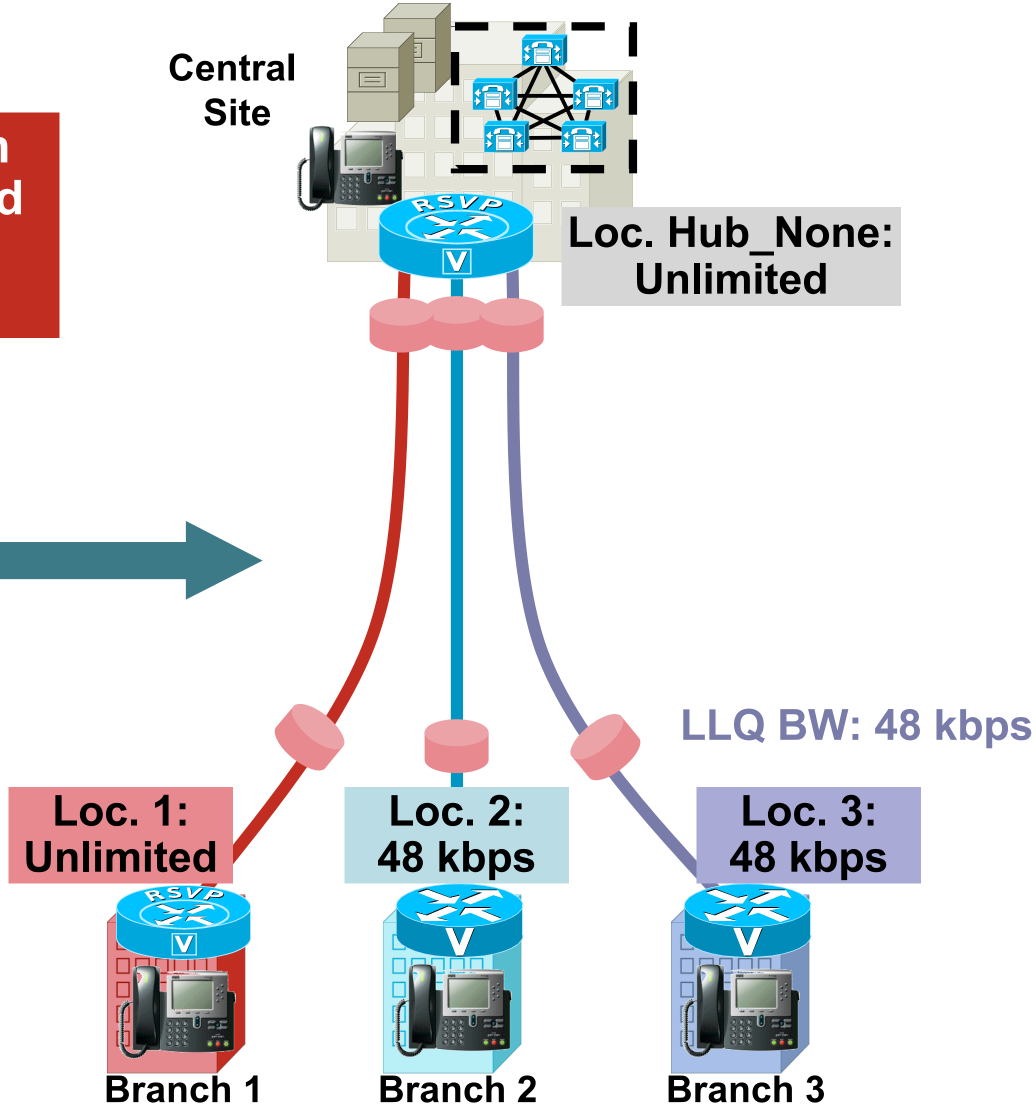
Migration from Static Locations

Locations and RSVP CAC Coexistence

Setting Location-CAC Bandwidth to Anything Other Than Unlimited Is Strongly Discouraged when RSVP-CAC Is Enabled

Step 2:
If the Location Check Succeeds then RSVP Based CAC Is Invoked

Step 1:
Location-Based CAC Is Checked First



Migration from Static Locations

Migrating the Second Through Nth Location

- Install an RSVP agent in the location to be migrated
- Add the agent to the MRGLs of all devices at that location
- Remove the Hub_None agent from the MRGLs of all devices at that location
- Configure RSVP policy (mandatory/optional etc.) for locations pair: location to be migrated to other locations
- Media flow between RSVP CAC enabled sites now does not have to flow through the hub location

CallManager RSVP-Enabled Locations

Best Practices and Caveats

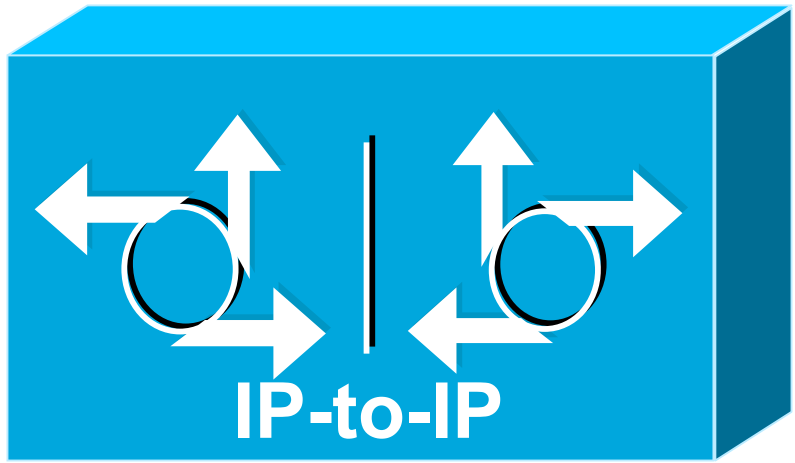
- Configure cluster-wide RSVP policy as “mandatory” or “mandatory (video desired)”
- Provision static location bandwidth as “unlimited” if location is RSVP enabled
- Configure “call fails following retry counter exceeded” as the mid-call retry option
- No device mobility; manual configuration is needed to reflect the device is moving from one location to another
- Applied to intra-cluster call only (see later for multi-cluster ideas)

Call Admission Control Elements
Agenda

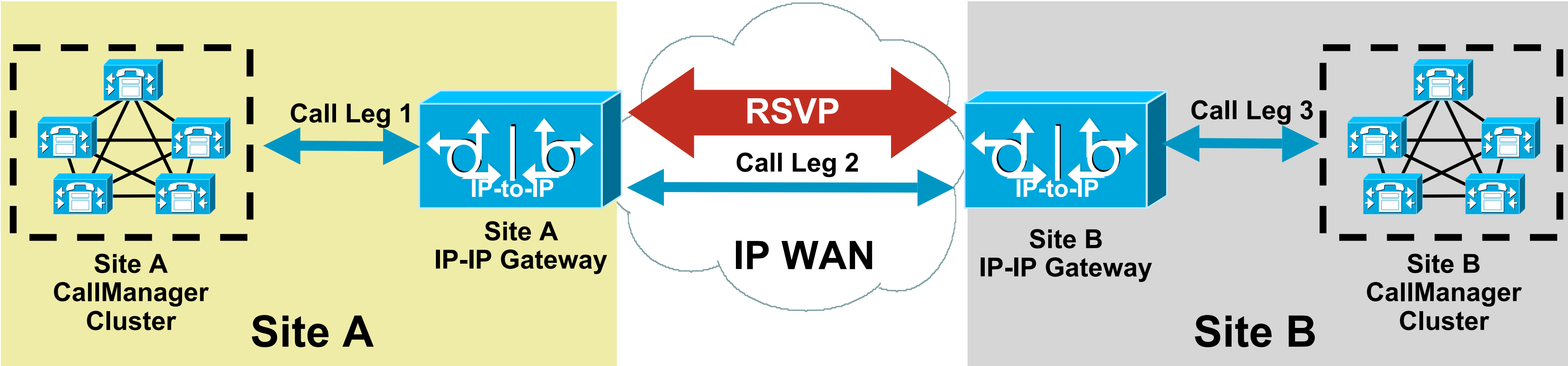
- Topology-Unaware CAC
  - Cisco CallManager “Static” Locations
  - Gatekeeper Zones
- Resource Reservation Protocol (RSVP)
- Topology-Aware CAC
  - Cisco CallManager RSVP-Enabled Locations
  - Gatekeeper and IP-IP Gateway with RSVP

Gatekeeper and IP-IP Gateway with RSVP
Introduction to IP-IP Gateway

- Normally used as a demarcation point
- Generates two IP call legs
- Can initiate RSVP reservations → topology-aware call admission control
- Available in Cisco IOS 12.3(7)T or later



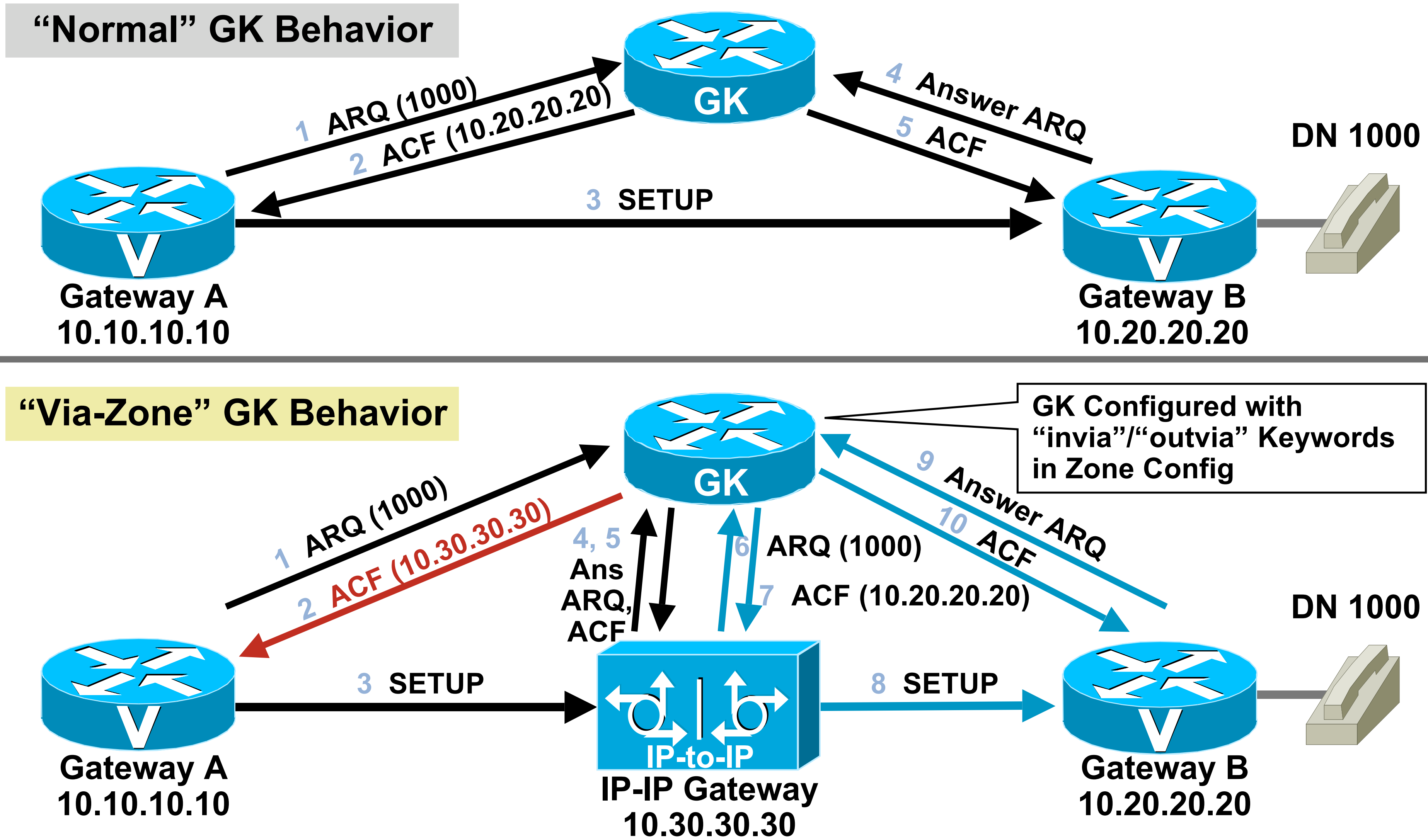
Gatekeeper and IP-IP Gateway with RSVP
Simple Example: Cisco CallManager and IP-IP Gateway



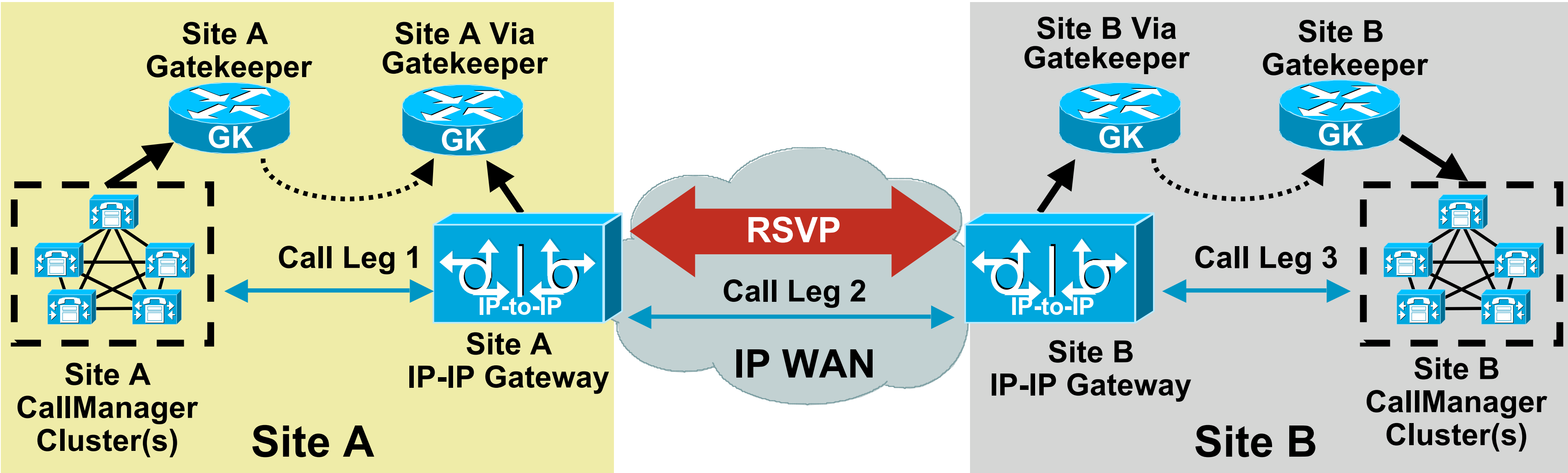
- Define non-GK controlled intercluster trunk between Cisco CallManager and IP-to-IP gateway
- Multiple signaling and media call legs
- RSVP reservations between sites

Gatekeeper and IP-IP Gateway with RSVP

"Via-Zone" Gatekeeper Concept

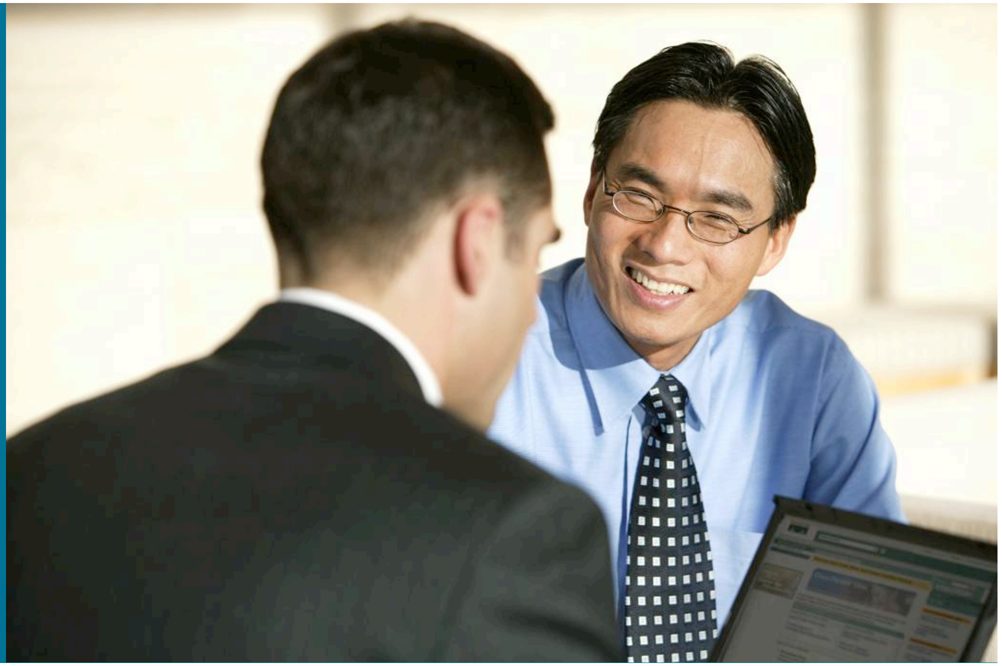


Gatekeeper and IP-IP Gateway with RSVP
Example with Via-Zone GK



- Define GK-controlled intercluster trunk between CCM and gatekeeper
- Via-zone gatekeeper
- Invia and outvia keywords in zone configuration

Call Admission Control Design

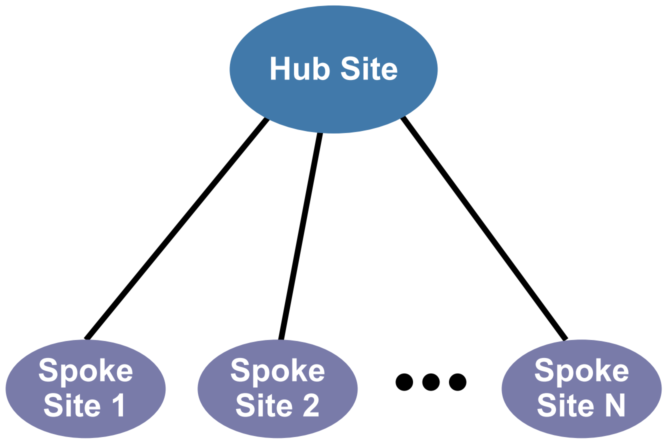


Call Admission Control Design
Agenda

- Simple Hub-and-Spoke Topologies
  - Centralized Cisco CallManager Deployments
  - Distributed Cisco CallManager Deployments
- Two-tier Hub-and-Spoke Topologies
- Simple MPLS Topologies
- Generic Topologies

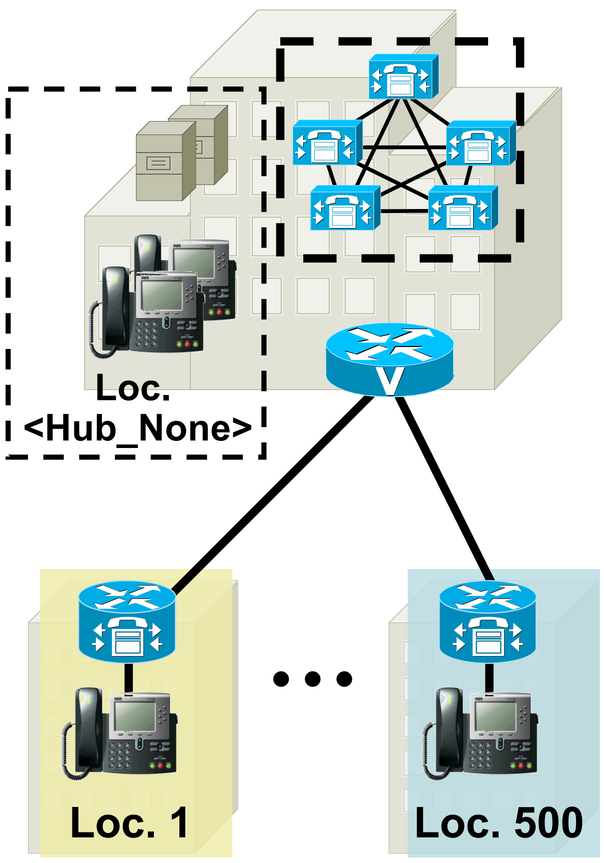
Simple Hub-and-Spoke Topologies

- Also known as “star topologies”
- All spoke-to-spoke traffic must transit through hub
- No direct connections between any two spoke sites
- Layer 2 technologies:
    - Frame Relay
    - ATM
    - FR/ATM service interworking
    - Leased lines



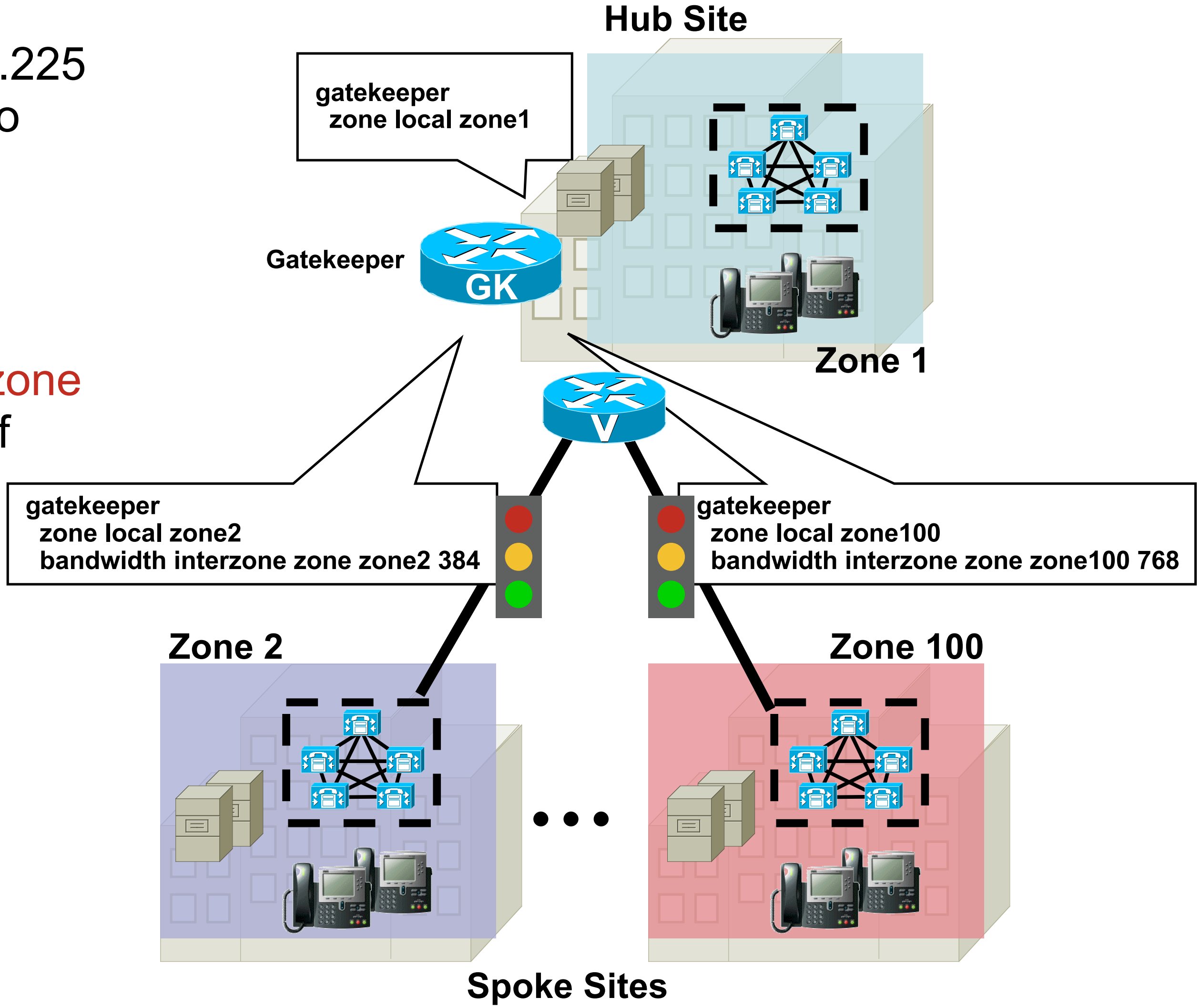
Simple Hub-and-Spoke Topologies
Centralized Deployments

- Use static locations: one location per spoke site
- Devices at hub site in <Hub_None> location
- Up to 500 locations per Cisco CallManager cluster
- If more than one Cisco CallManager cluster at hub site, use intercluster trunks (leave in <Hub_None> location)
- Location needs to be updated if device moves to a different site



Simple Hub-and-Spoke Topologies
Distributed Deployments

- Use GK-controlled H.225 Trunks on each Cisco CallManager cluster
- 1 GK zone per Cisco CallManager cluster
- Use bandwidth interzone to limit traffic in/out of each site
- Up to 100 Cisco CallManager clusters per GK
- > 100 clusters with additional GKs and directory GK



Call Admission Control Design
Agenda

- Simple Hub-and-Spoke Topologies
- Two-tier Hub-and-Spoke Topologies
  - Centralized Cisco CallManager Deployments
  - Distributed Cisco CallManager Deployments
- Simple MPLS Topologies
- Generic Topologies

Two-Tier Hub-and-Spoke Topologies

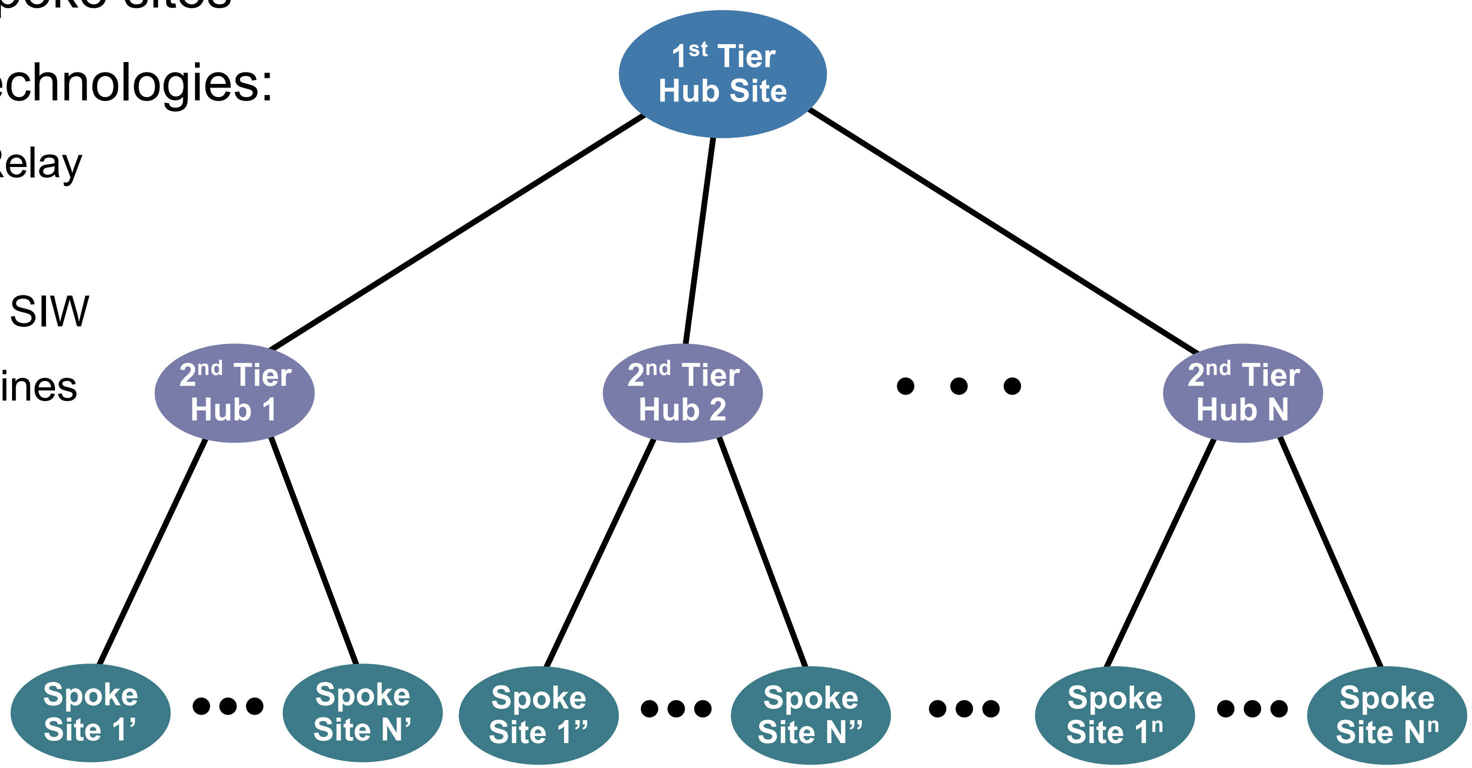
- All spoke-to-spoke traffic must transit through 1st/2nd tier hubs
- No direct connections between any two spoke sites
- Layer 2 technologies:

  Frame Relay

  ATM

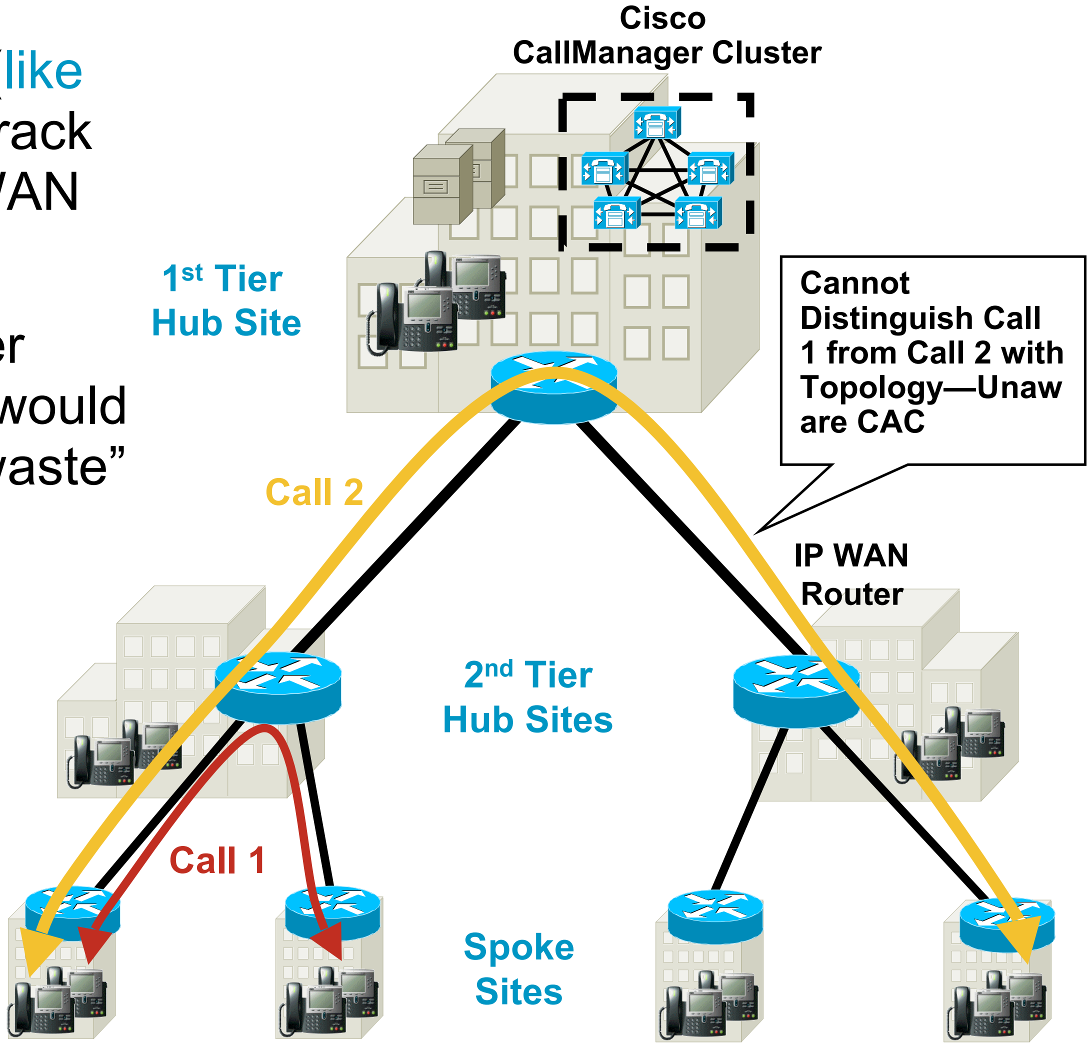
  FR/ATM SIW

  Leased lines



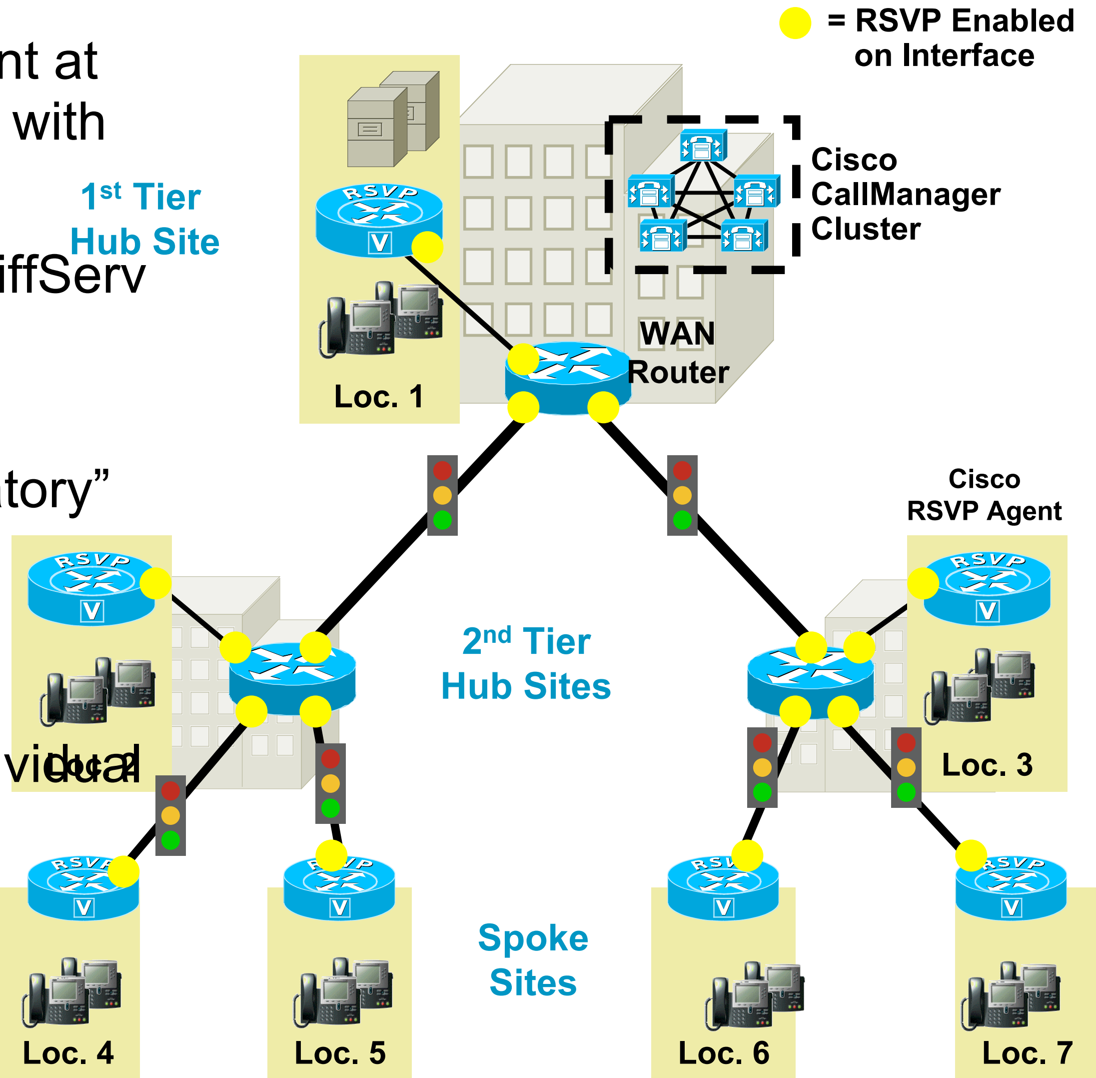
Two-Tier Hub-and-Spoke Topologies
Centralized Deployments: the Issue

- Topology-unaware CAC (like static Locations) cannot track bandwidth of individual WAN links
- Deploying it with a two-tier hub-and-spoke topology would lead to high bandwidth “waste”
- Topology awareness is needed in this case: → use RSVP-enabled locations



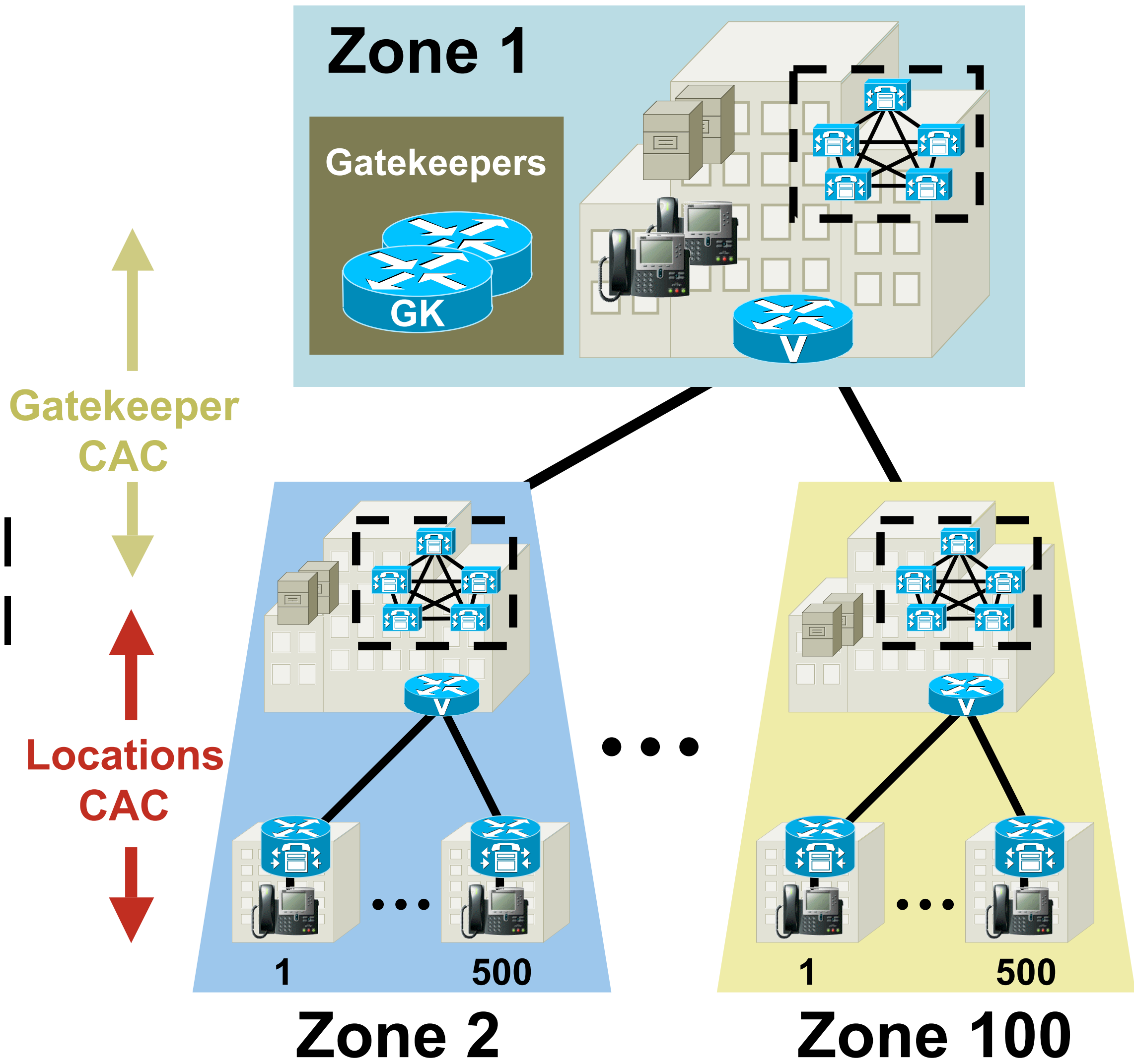
Two-Tier Hub-and-Spoke Topologies
Centralized Deployments: Solution

- Deploy Cisco RSVP agent at each site (may co-reside with WAN router)
- Enable RSVP IntServ/ DiffServ mode on WAN routers
- Configure RSVP "mandatory" policy between locations in Cisco CallManager
- CAC occurs on each individual WAN link



Two-Tier Hub-and-Spoke Topologies
Distributed Deployments

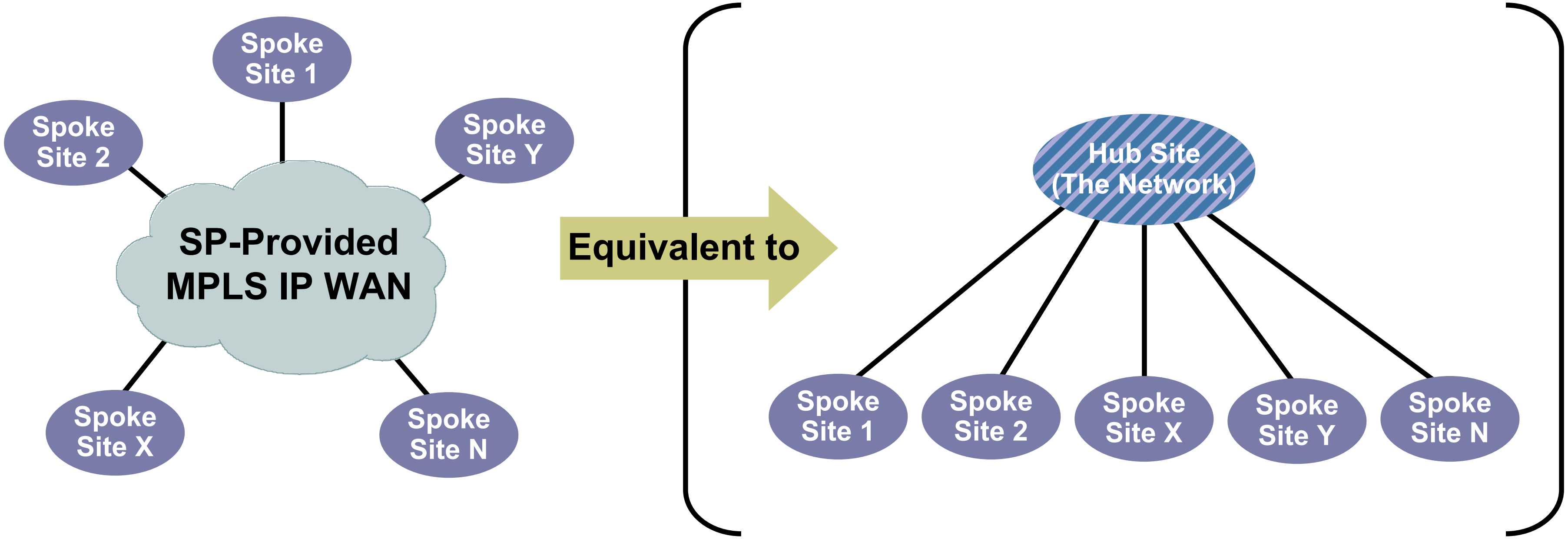
- Use gatekeeper call admission control between first and second tier hub sites
- Use locations call admission control between second tier hub sites and spoke sites



Call Admission Control Design
Agenda

- Simple Hub-and-Spoke Topologies
- Two-tier Hub-and-Spoke Topologies
- Simple MPLS Topologies
    - Co-existence of MPLS and Frame Relay/ATM
    - Centralized Cisco CallManager Deployments
    - Distributed Cisco CallManager Deployments
- Generic Topologies

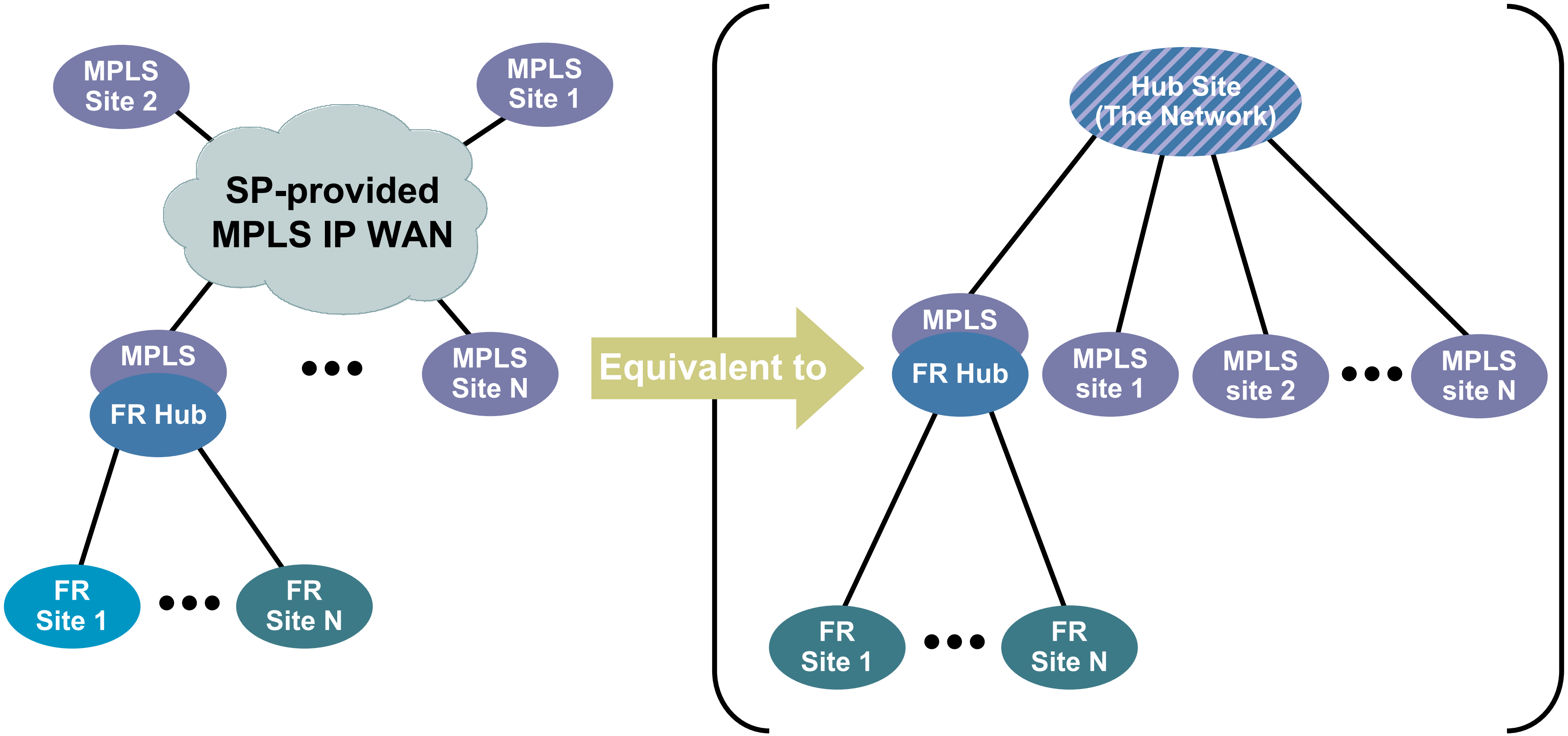
MPLS-Based Topologies



- MPLS WAN is provided by a service provider
- As seen by the enterprise network, every site is one IP "hop" away
- Equivalent to a full mesh, or to a "hubless" hub-and-spoke

MPLS-Based Topologies

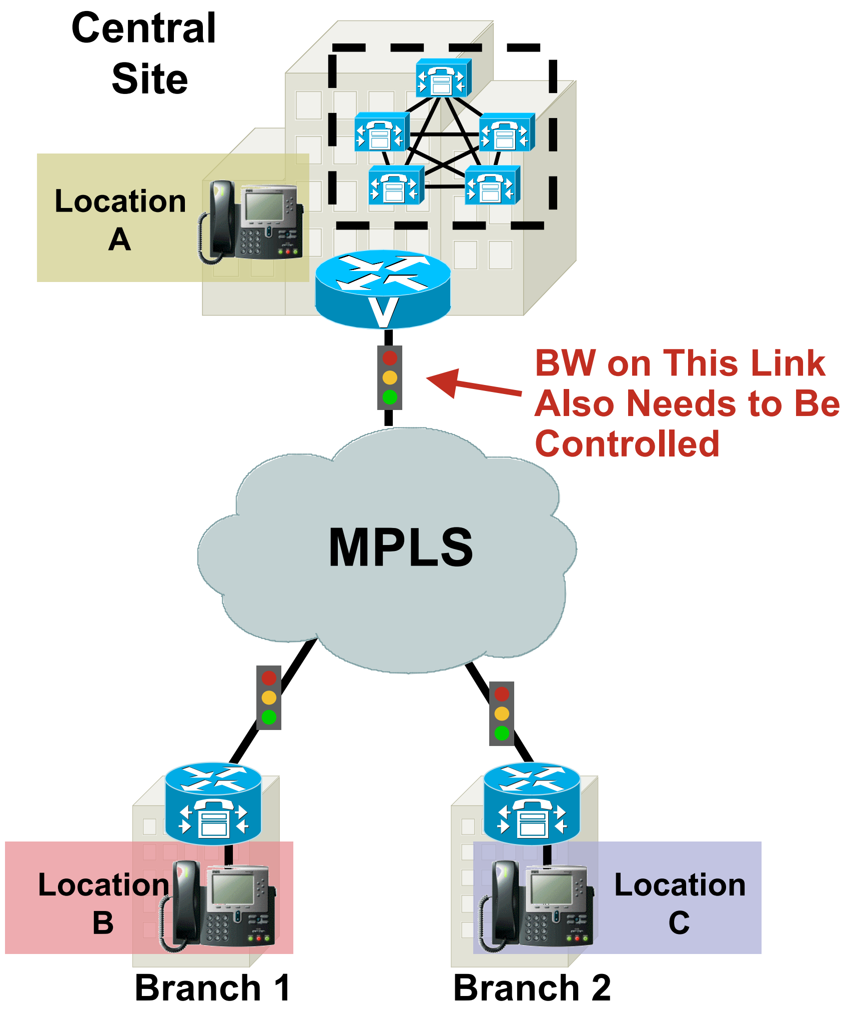
Co-Existence of MPLS and Frame Relay/ATM



Refer to the Two-Tier Hub-and-Spoke Section for Topologies Such as This One

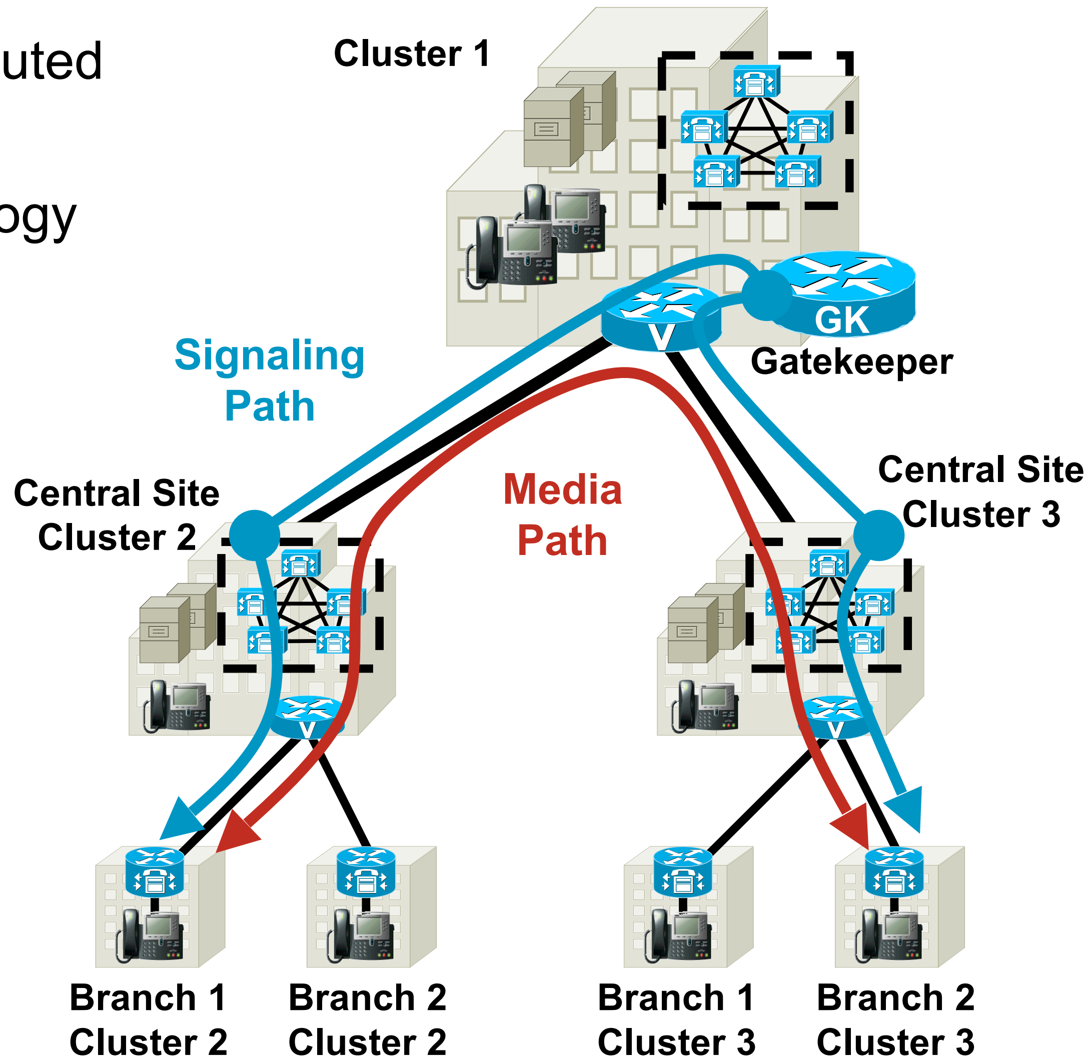
MPLS-Based Topologies
Centralized Deployments

- When the WAN topology is MPLS-based, the location-based configuration needs to be modified
- Endpoints at the central site also need to be placed in a location
- Locations now provide call admission control for each site, including the central site



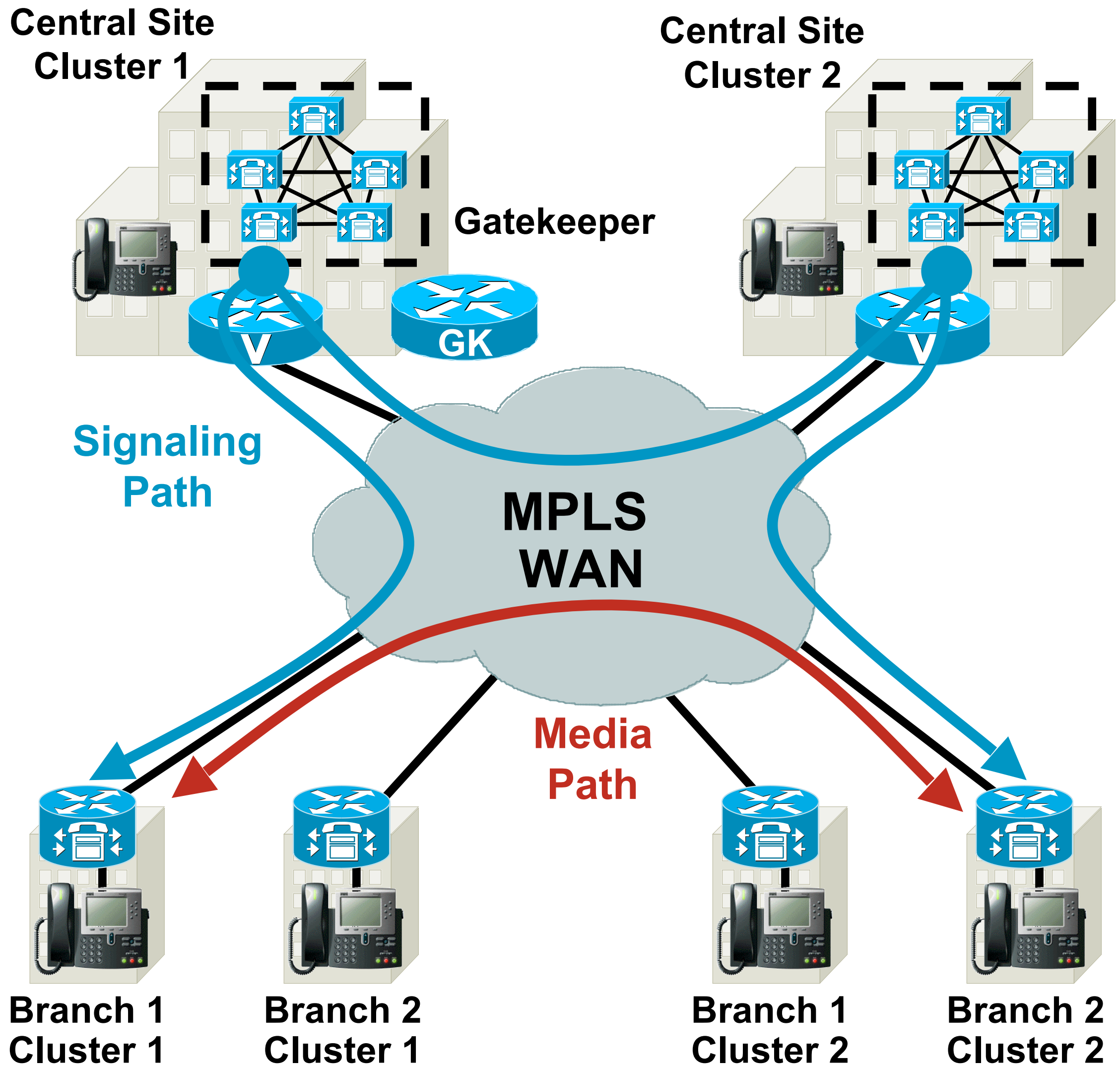
MPLS-Based Topologies
Distributed Deployments—Traditional WAN Call Flows

- Hybrid centralized/distributed deployment
- Traditional WAN technology (Frame Relay, ATM,—)
- Two-tier hub-and-spoke topology
- Both media and signaling paths "follow" the topology

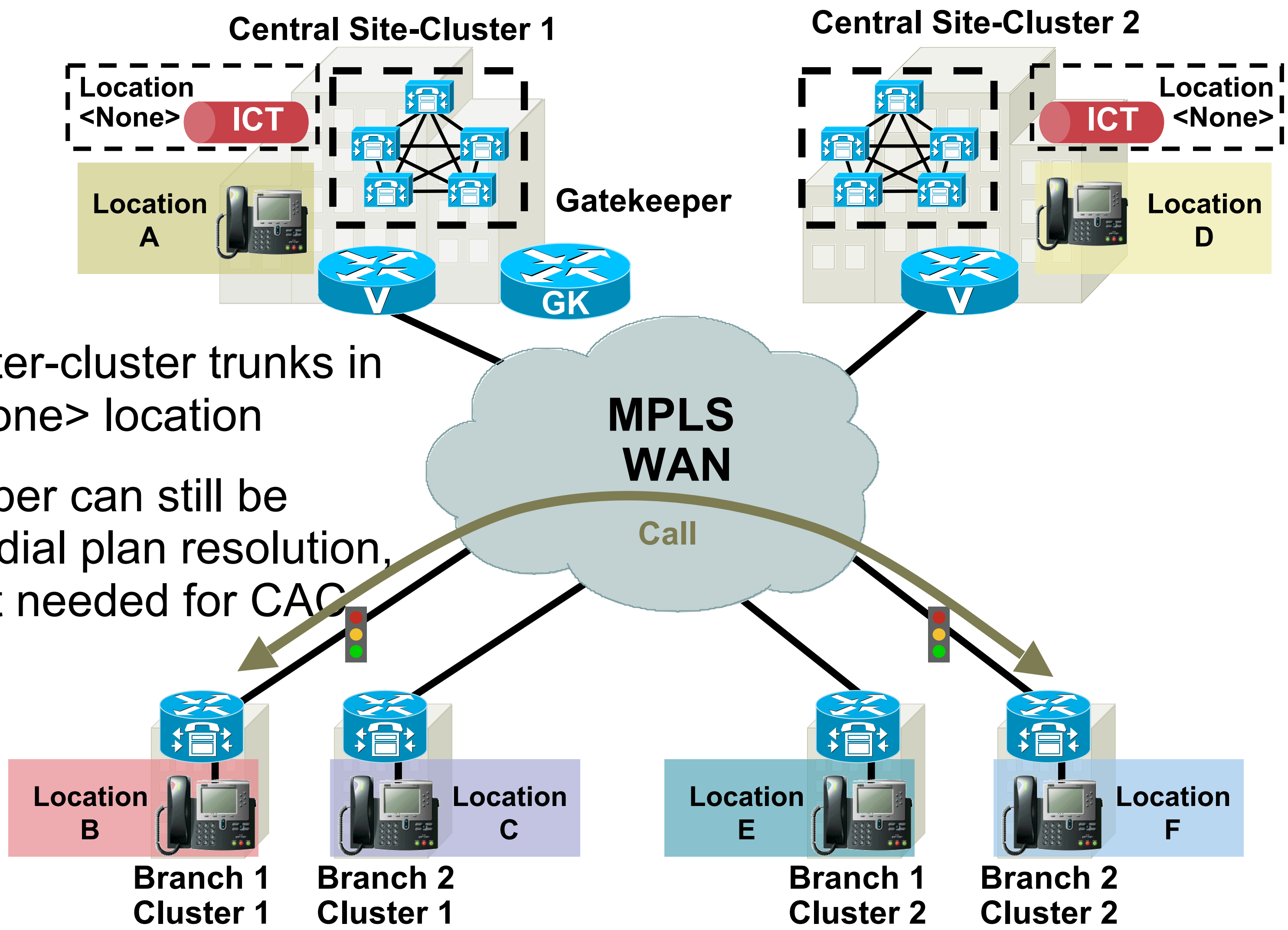


MPLS-Based Topologies
Distributed Deployments—MPLS WAN Call Flows

- Signaling path still goes through the Cisco CallManagers at the central sites
- Media path goes directly between branches
- Traditional CAC with locations + gatekeeper does not work correctly in this scenario



MPLS-Based Topologies
Distributed Deployments

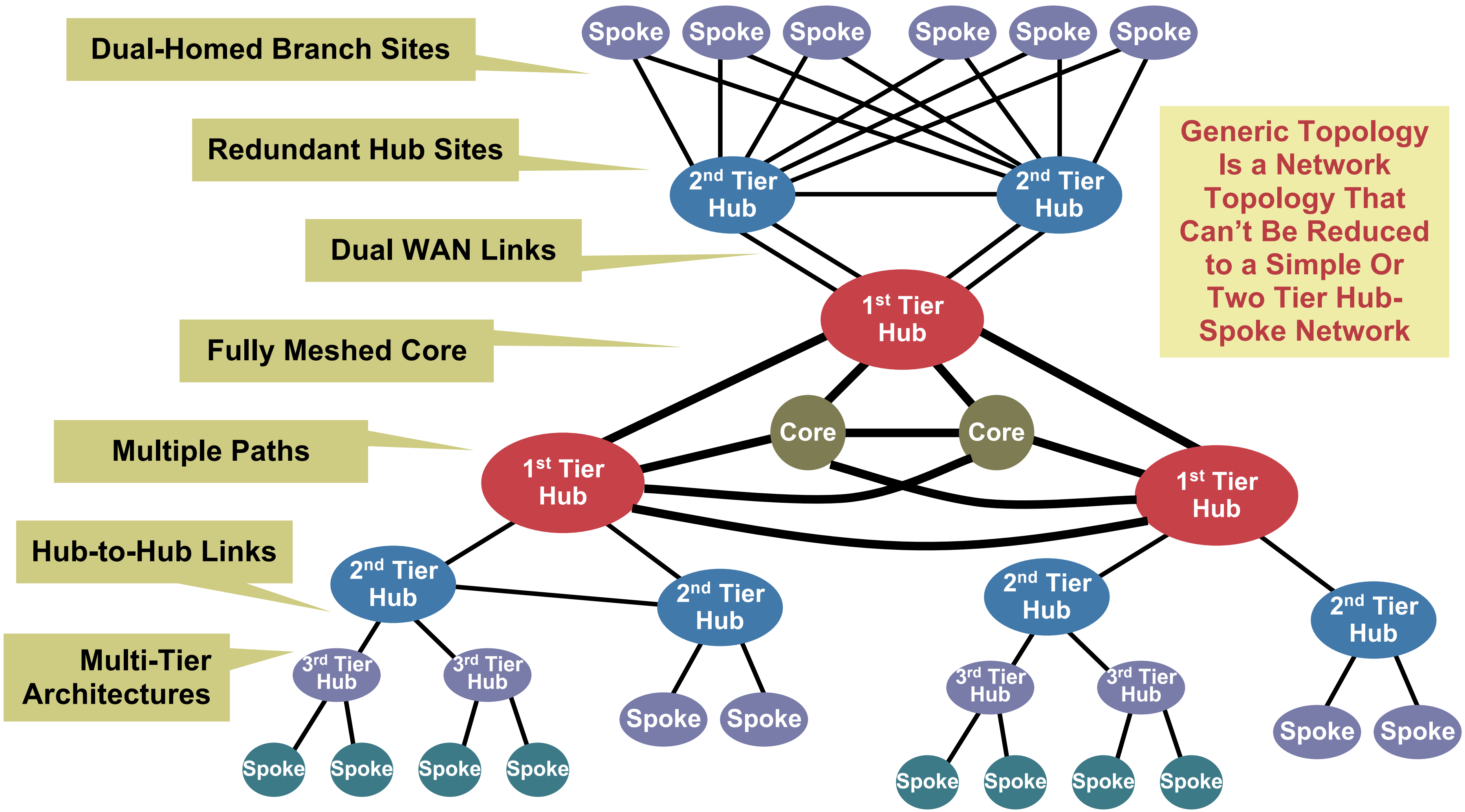


- Leave inter-cluster trunks in <Hub_None> location
- Gatekeeper can still be used for dial plan resolution, but is not needed for CAC

Call Admission Control Design
Agenda

- Simple Hub-and-Spoke Topologies
- Two-tier Hub-and-Spoke Topologies
- Simple MPLS Topologies
- Generic Topologies
    - Centralized Cisco CallManager Deployments
        - Single Cluster
        - Multiple Clusters
    - Distributed Cisco CallManager Deployments

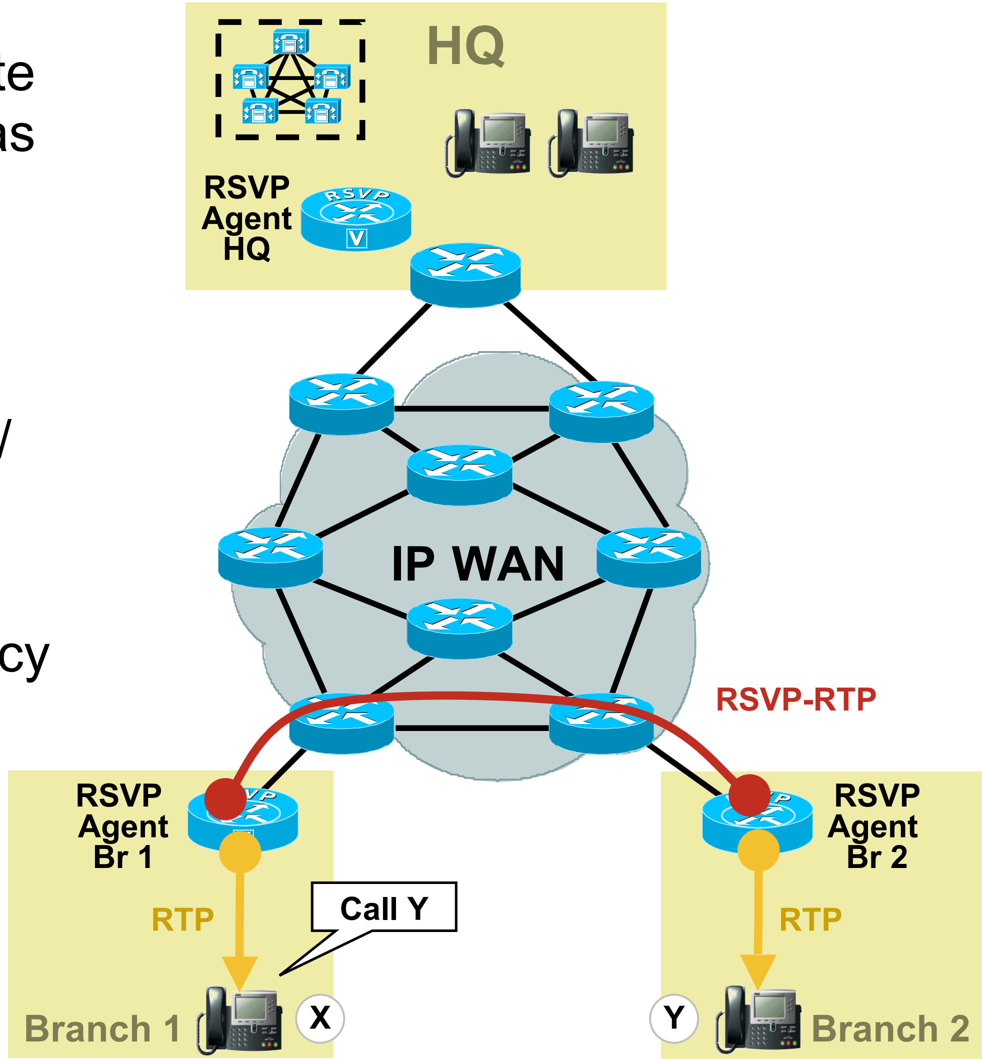
Generic Topologies Defined



Generic Topology Is a Network Topology That Can't Be Reduced to a Simple Or Two Tier Hub-Spoke Network

Generic Topologies
Centralized Deployments—Single Cluster

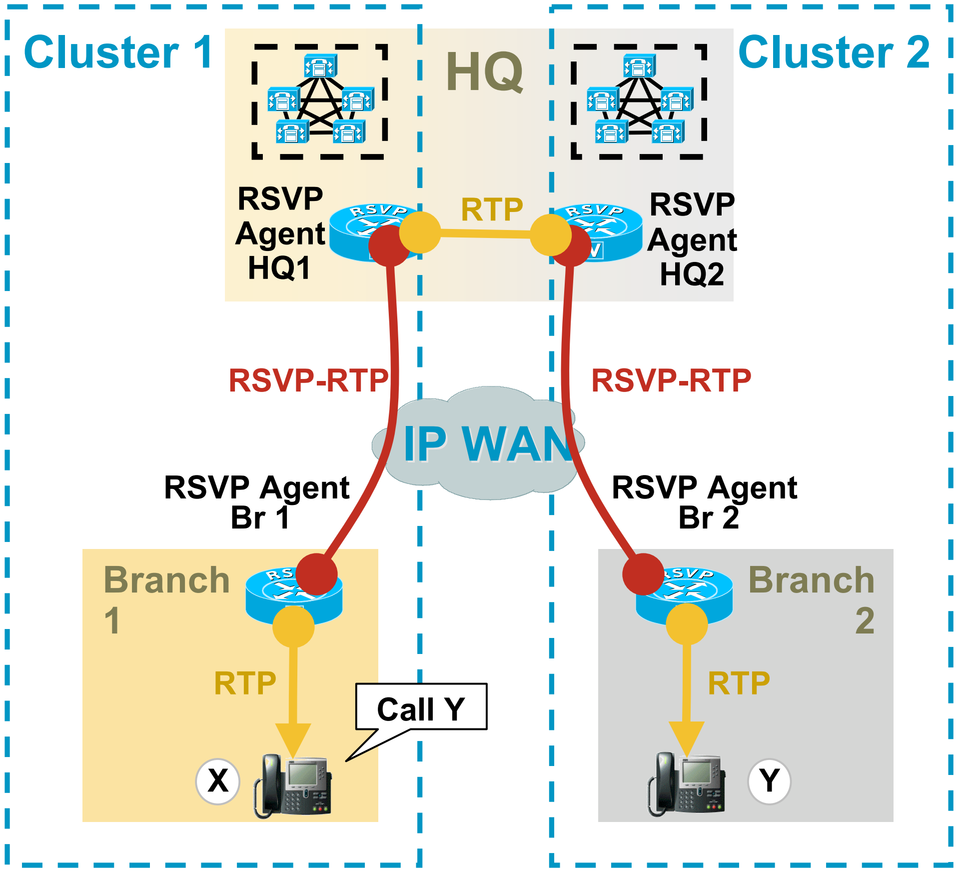
- Define a location for each site and leave bandwidth value as unlimited
- Enable RSVP agent at each site
- Assign RSVP agent in MRG/ MRGL of all devices at that site
- Configure default RSVP policy as mandatory or mandatory (video desired)



Generic Topologies

Centralized Deployments—Multiple Clusters

- Multiple Cisco CallManager clusters are located at the same LAN/MAN
- Define intercluster trunk in HQ location for inter-cluster communication
- Deploy RSVP agent at every site within the cluster and use RSVP CAC for call leg within the cluster
- No CAC for call leg between two clusters is needed

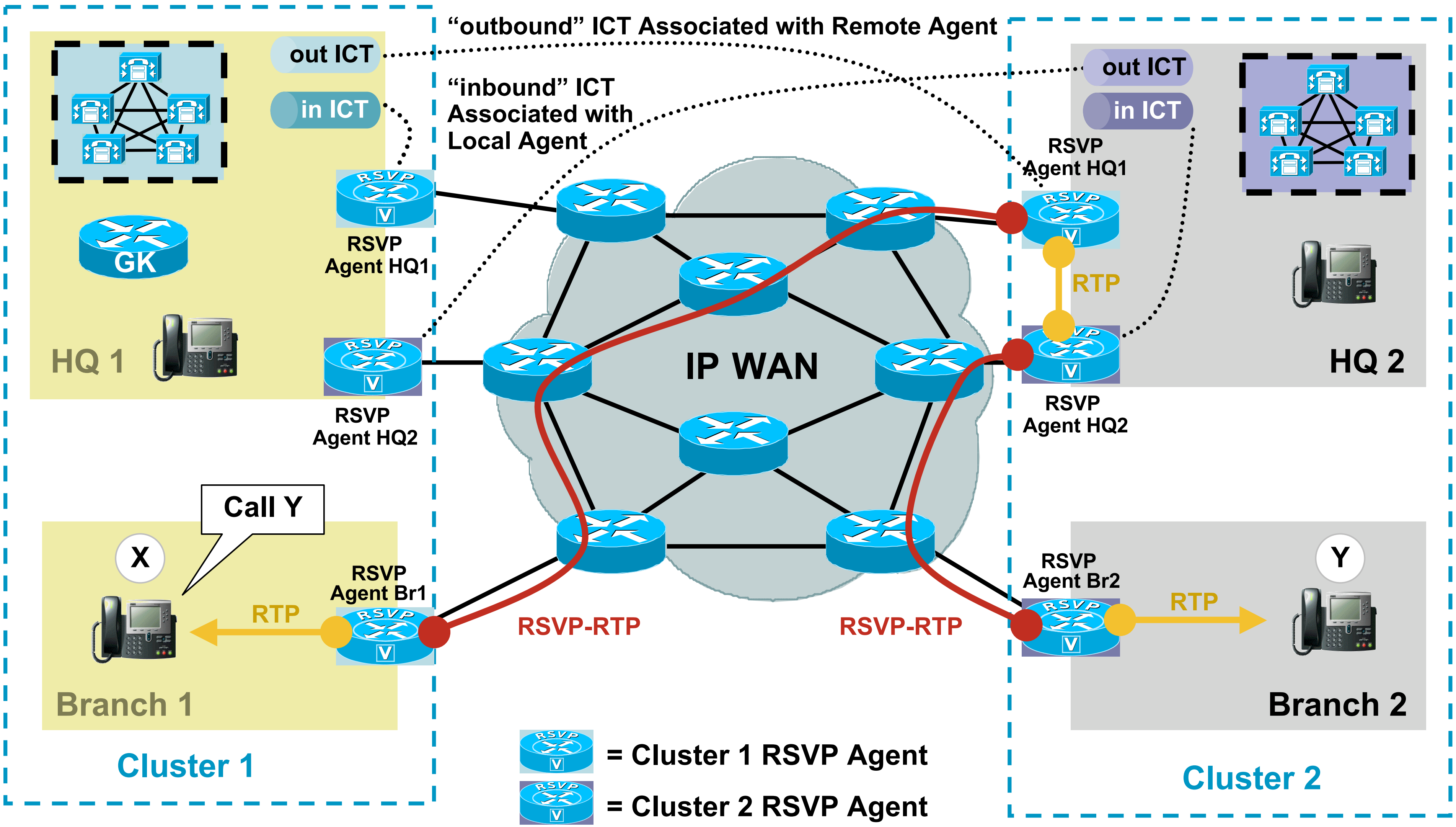


Call Admission Control Design
Agenda

- Simple Hub-and-Spoke Topologies
- Two-tier Hub-and-Spoke Topologies
- Simple MPLS Topologies
- Generic Topologies
    - Centralized Cisco CallManager Deployments
    - Distributed Cisco CallManager Deployments
    - Remote RSVP Agent Approach

Generic Topologies

Distr. Deployments—Remote RSVP Agent Approach



Generic Topologies
Distr. Deployments—Remote RSVP Agent Approach

- Approach relies on placing RSVP agents co-located with remote Cisco CallManager clusters
- Needs two trunks (ICT or H.225) per cluster, one for inbound calls and one for outbound calls
- Design not recommended beyond two–three Cisco CallManager clusters, as it requires a full mesh of agents and trunks → complexity quickly increases

Generic Topologies
Distr. Deployments—Remote RSVP Agent Approach

- Rely on originating cluster to do the bandwidth reservation for the call leg across the WAN

  Use CCM dialplan to send all outbound calls via “remote” trunk (“remote” trunk is in the same location as the remote RSVP agent)

  Use GK zone prefix to send all inbound calls via “campus” trunk (“campus” trunk is in the same location as HQ phone and HQ RSVP agent)

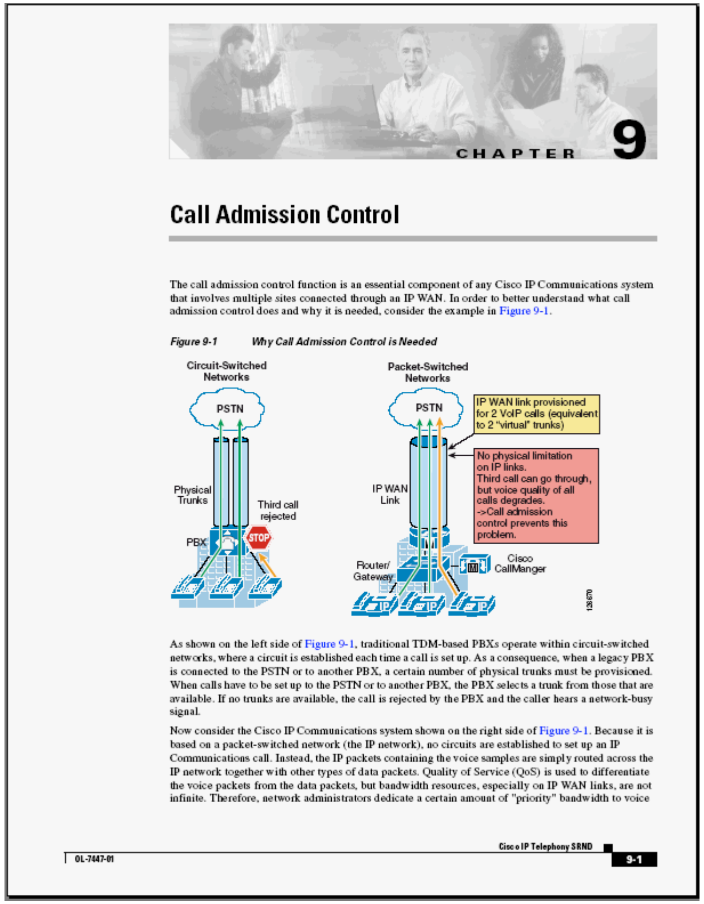
  Both trunks register with the same GK but in different zones

- Terminating cluster only needs to make a reservation if the call is destined to one of its remote branches

For More Information

- “Call Admission Control” chapter of the Cisco Unified Communications Solution Reference Network Design (SRND) for Cisco CallManager release 5.X, available online at:

www.cisco.com/go/srnd



CHAPTER 9

Call Admission Control

The call admission control function is an essential component of any Cisco IP Communications system that involves multiple sites connected through an IP WAN. In order to better understand what call admission control does and why it is needed, consider the example in Figure 9-1.

Figure 9-1 Why Call Admission Control is Needed

As shown on the left side of Figure 9-1, traditional TDM-based PBXs operate within circuit-switched networks, where a circuit is established each time a call is set up. As a consequence, when a legacy PBX is connected to the PSTN or to another PBX, a certain number of physical trunks must be provisioned. When calls have to be set up to the PSTN or to another PBX, the PBX selects a trunk from those that are available. If no trunks are available, the call is rejected by the PBX and the caller hears a network-busy signal.

Now consider the Cisco IP Communications system shown on the right side of Figure 9-1. Because it is based on a packet-switched network (the IP network), no circuits are established to set up an IP Communications call. Instead, the IP packets containing the voice samples are simply routed across the IP network together with other types of data packets. Quality of Service (QoS) is used to differentiate the voice packets from the data packets, but bandwidth resources, especially on IP WAN links, are not infinite. Therefore, network administrators dedicate a certain amount of "priority" bandwidth to voice

Cisco IP Telephony SRND

OL-7447-01 9-1

Meet the Experts
Security

- Andres Gasson
  Consulting Systems Engineer

- Christophe Paggen
  Technical Marketing Engineer

- Eric Vyncke
  Distinguished Consulting Engineer

- Erik Lenten
  Technical Marketing Engineer

- Fredéric Detienne
  CA Technical Leader

- Luc Billot
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- Olivier Dupont
  Corporate Dev Consulting Engineer

- Peter Matthews
  Technical Marketing Engineer

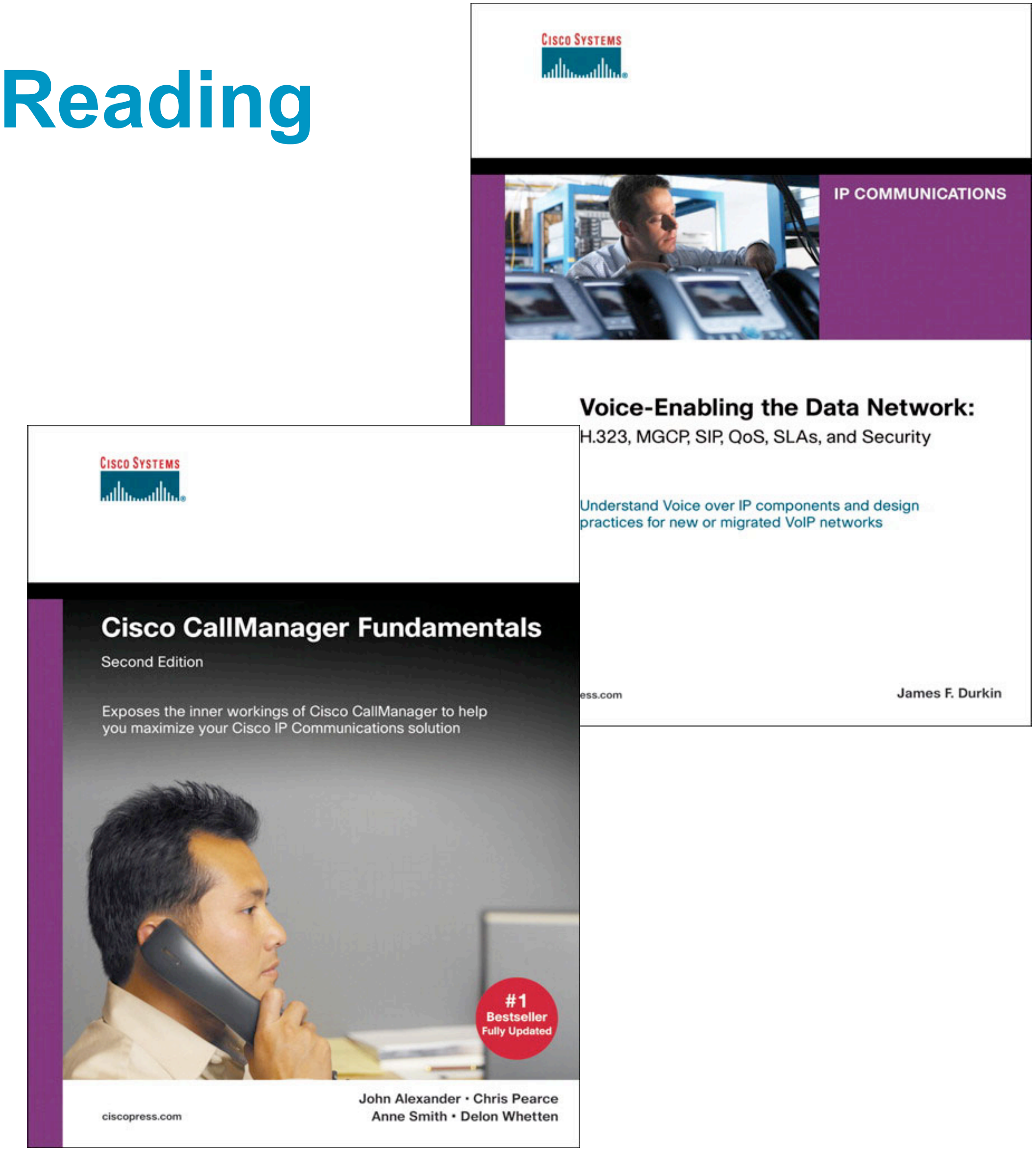
- Scott Wainner
  Distinguished System Engineer

- Steinthor Bjarnason
  Consulting Engineer

Recommended Reading

BRKUCT - 2010

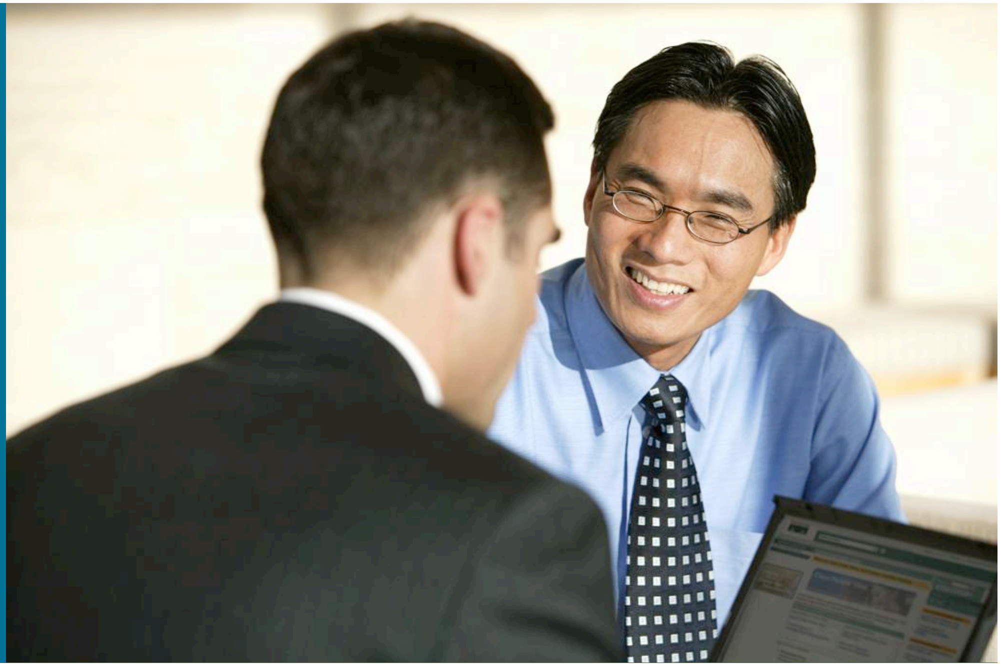
- Cisco CallManager Fundamentals
- Voice-Enabling the Data Network: H.323, MGCP, SIP, QoS, SLAs, and Security



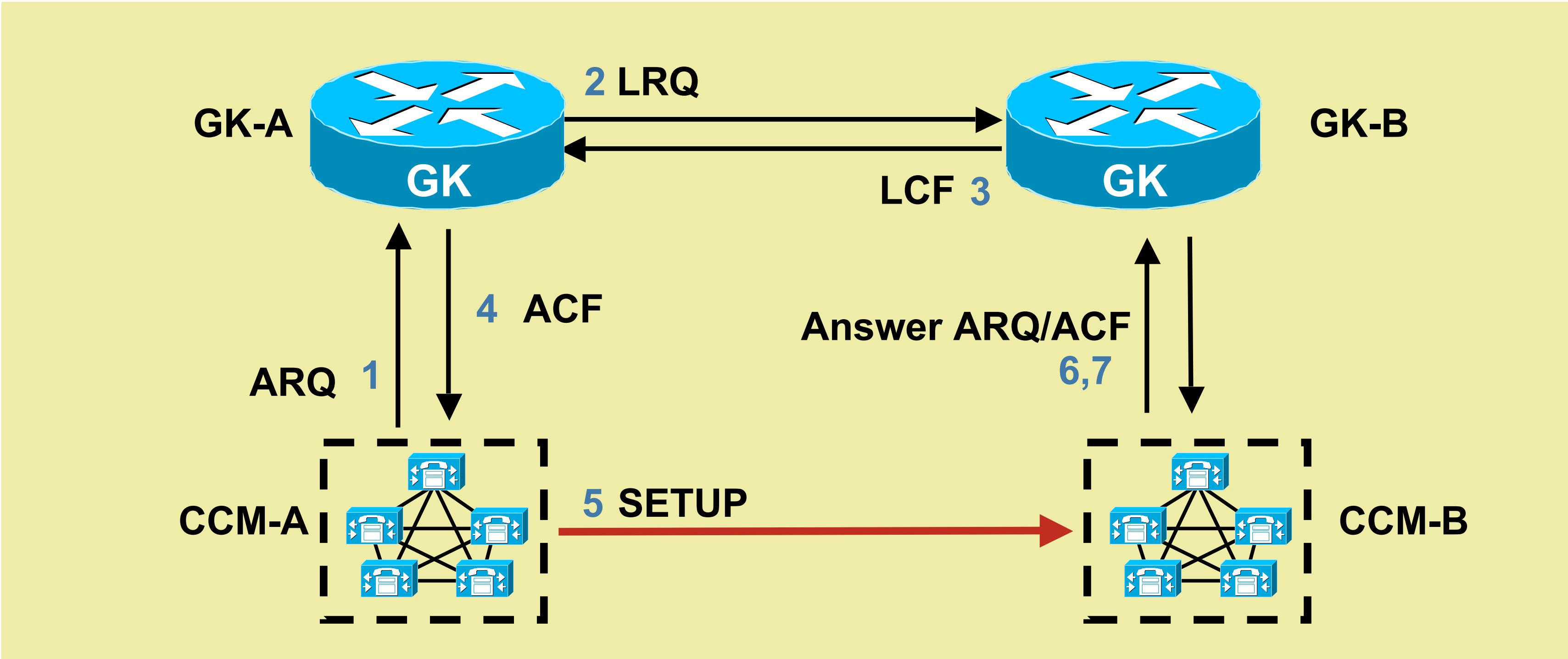
Available in the Cisco Company Store

Q and A

Appendix: Gatekeeper and IP-IP Gateway with RSVP



Gatekeeper Zones
Basic Call Setup (Accepted Call)

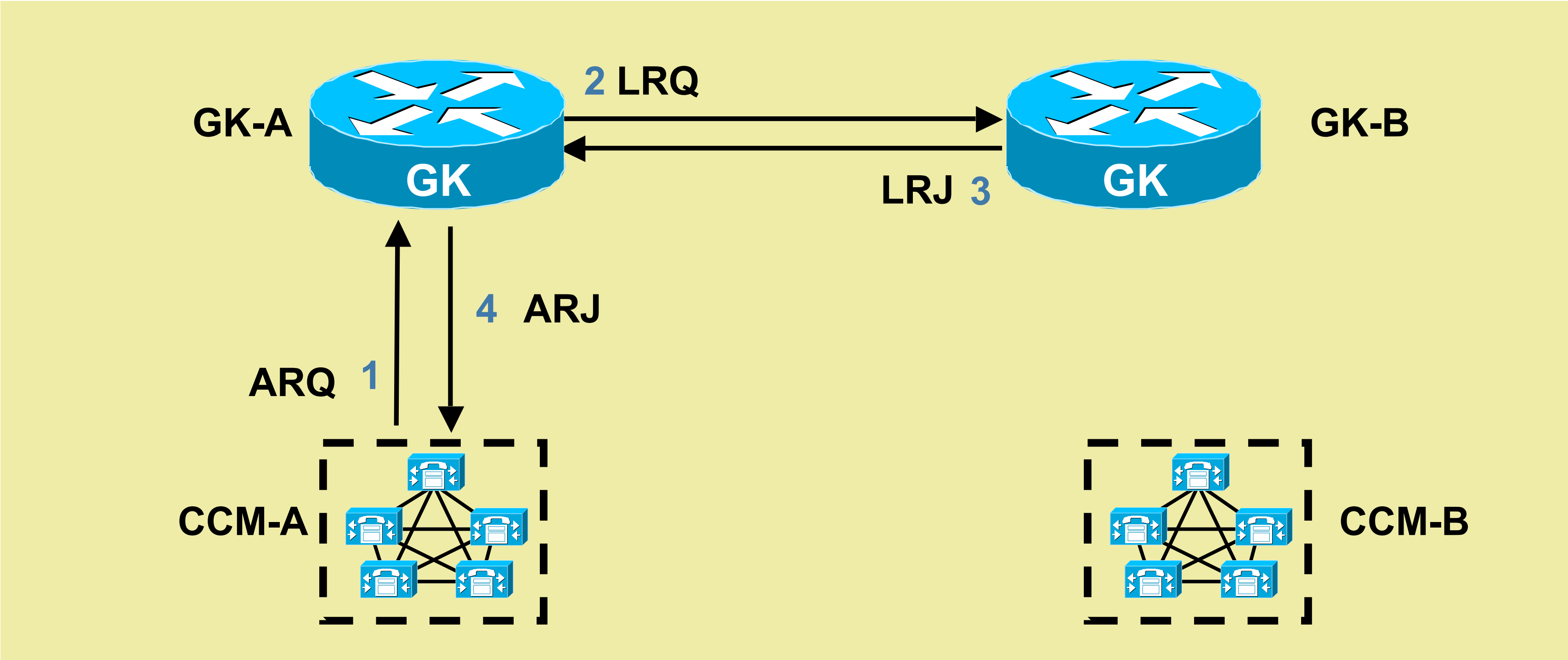


Call Admission
- AdmissionRequest (ARQ)
- AdmissionConfirm (ACF)
- AdmissionReject (ARJ)

Location Request
- LocationRequest (LRQ)
- LocationConfirm (LCF)
- LocationReject (LRJ)

Disengage
- DisengageRequest (DRQ)
- DisengageConfirm (DCF)
- DisengageReject (DRJ)

Gatekeeper Zones
Basic Call Setup (Rejected Call)

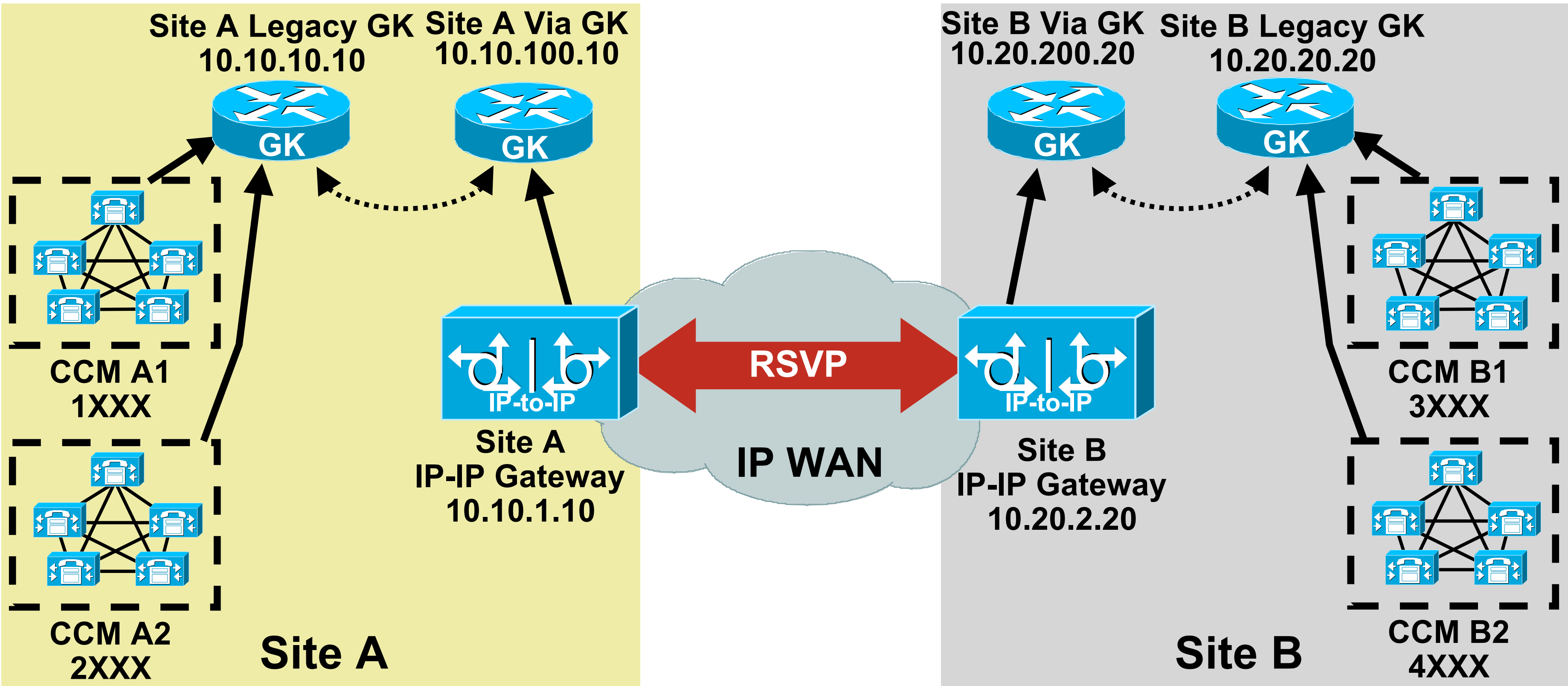


Call Admission
- AdmissionRequest (ARQ)
- AdmissionConfirm (ACF)
- AdmissionReject (ARJ)

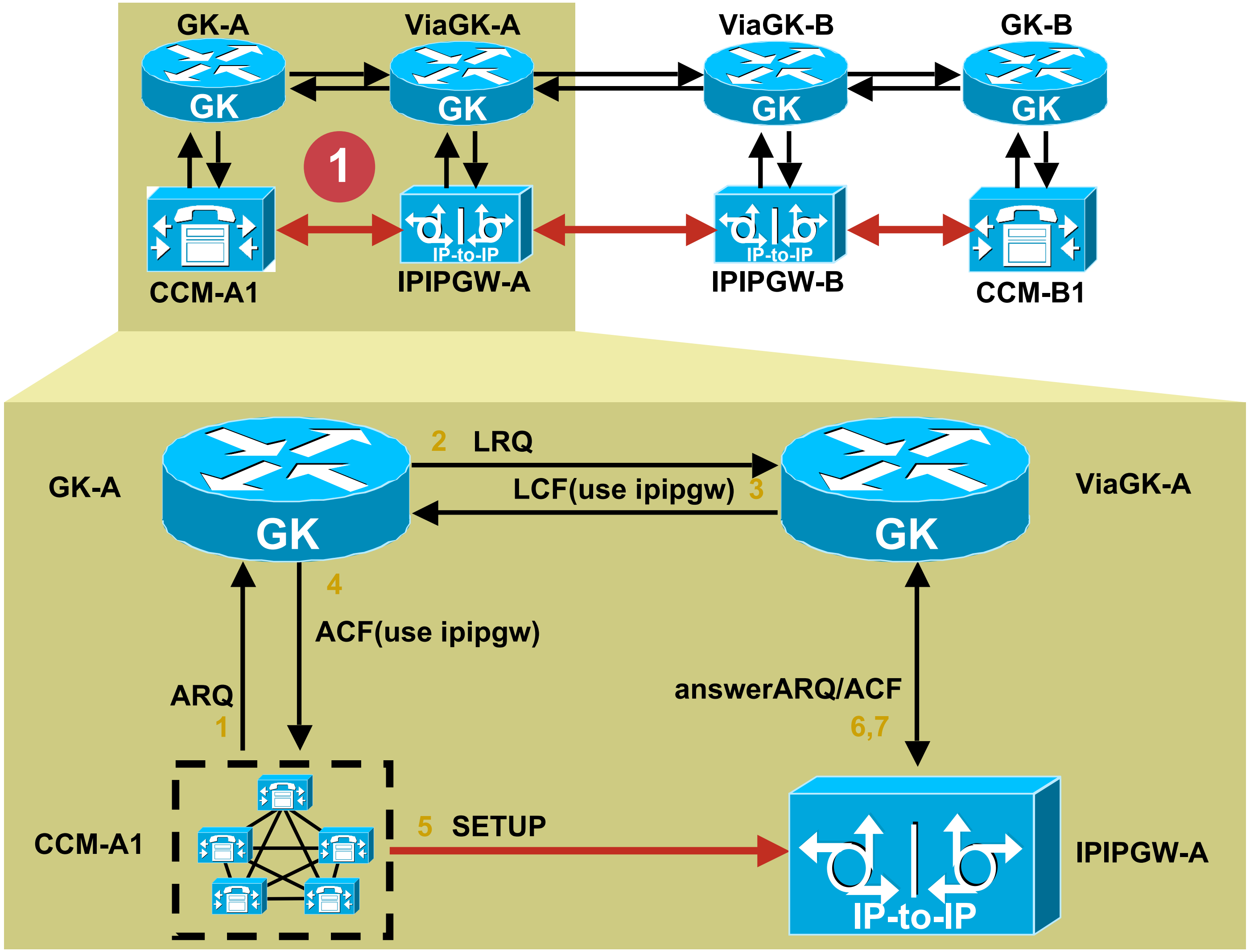
Location Request
- LocationRequest (LRQ)
- LocationConfirm (LCF)
- LocationReject (LRJ)

Disengage
- DisengageRequest (DRQ)
- DisengageConfirm (DCF)
- DisengageReject (DRJ)

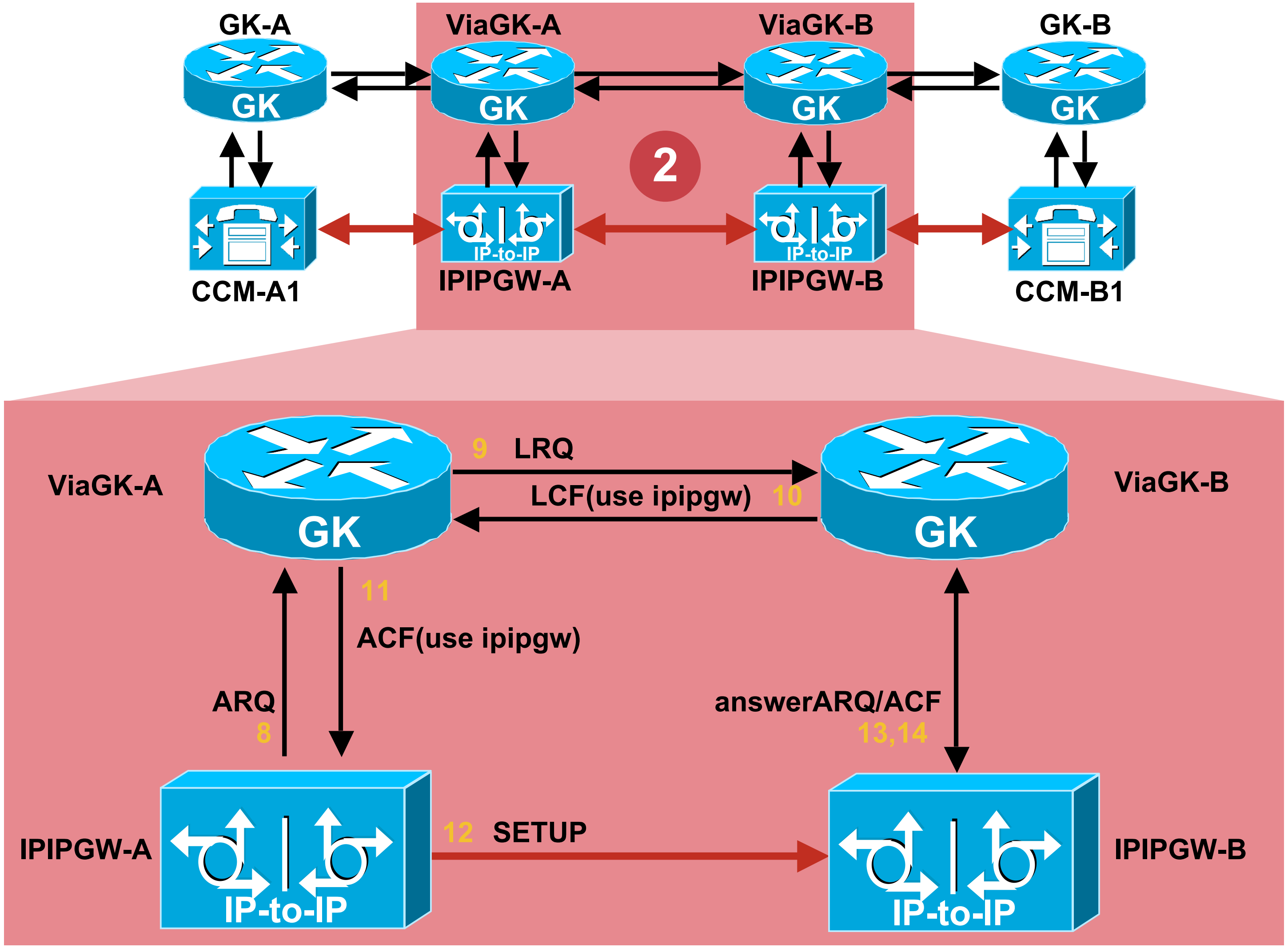
Gatekeeper and IP-IP Gateway with RSVP
Configuration Example: Scenario



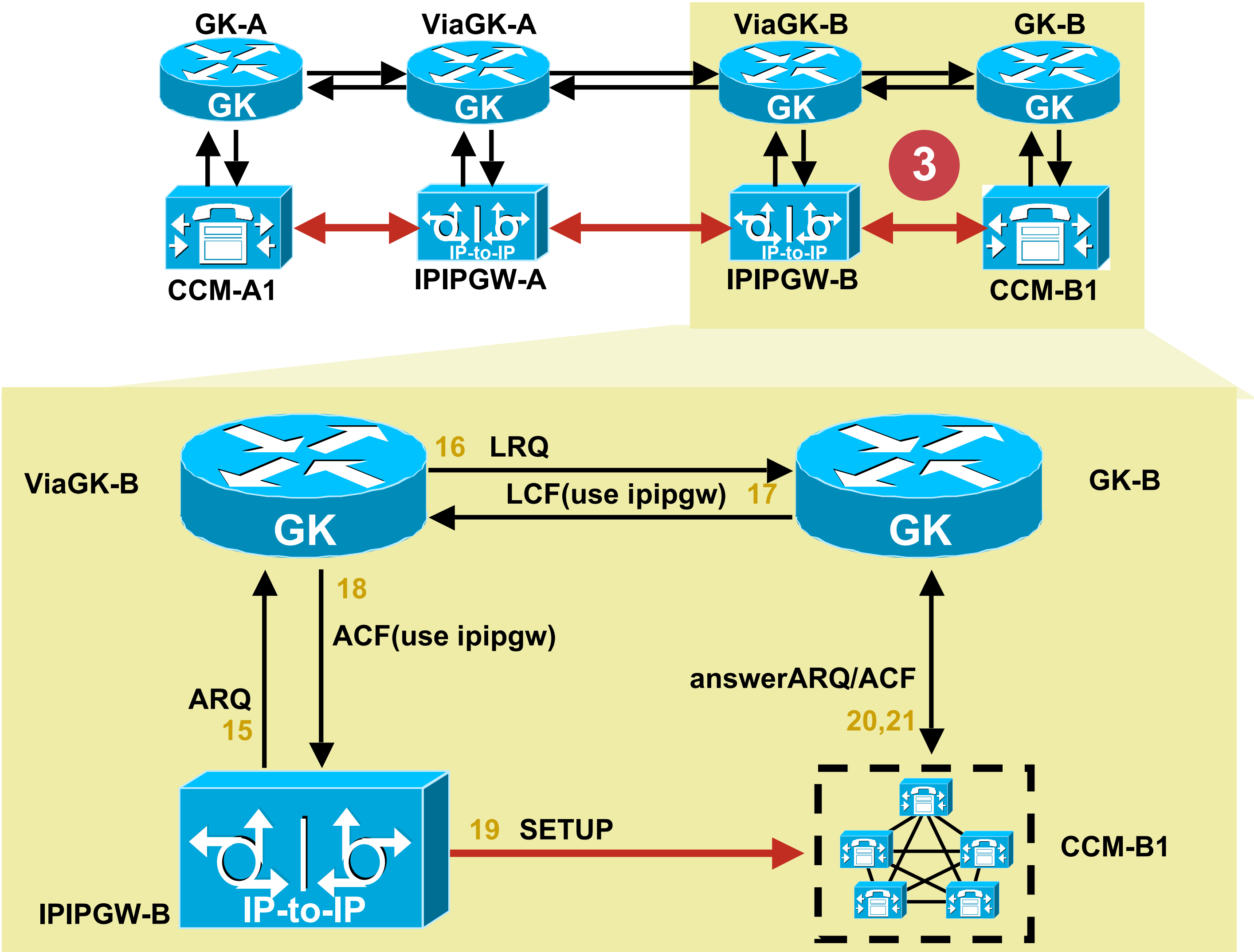
Gatekeeper and IP-IP Gateway with RSVP Configuration Example—Call Flow (1)



Gatekeeper and IP-IP Gateway with RSVP Configuration Example—Call Flow (2)



Gatekeeper and IP-IP Gateway with RSVP Configuration Example—Call Flow (3)



Gatekeeper and IP-IP Gateway with RSVP Configuration Example—Configs

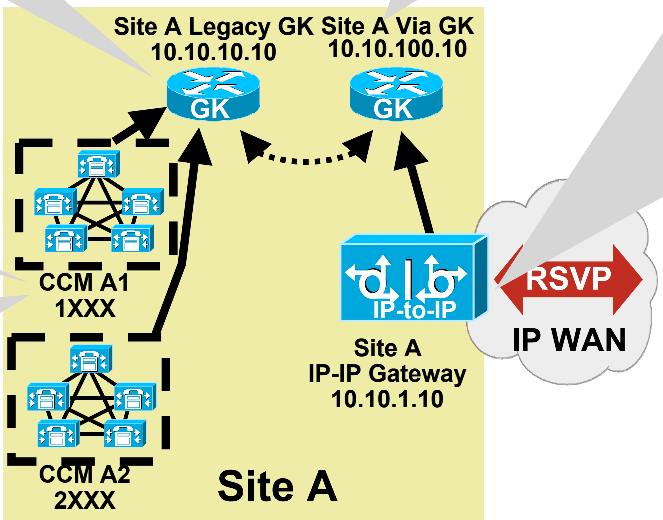
```
gatekeeper
 zone local CCM-A1 customer.com
   10.10.10.10
 zone local CCM-A2 customer.com
 zone remote A-VIAGK customer.com
   10.10.100.10
 zone prefix CCM-A1 1...
 zone prefix CCM-A2 2...
 zone prefix A-VIAGK 3...
 zone prefix A-VIAGK 4...
 gw-type-prefix 1#* default-technology
 arq reject-unknown-prefix
 no shutdown
```

```
gatekeeper
 zone local A-VIAGK customer.com
   10.10.100.10
 zone remote CCM-A1 customer.com
   10.10.10.10
 zone remote CCM-A2 customer.com
   10.10.10.10
 zone remote B-VIAGK customer.com
   10.20.200.20 invia A-VIAGK outvia A-VIAGK
 zone prefix B-VIAGK 3...
 zone prefix B-VIAGK 4...
 zone prefix CCM-A1 1...
 zone prefix CCM-A2 2...
 arq reject-unknown-prefix
 no shutdown
```

```
voice service voip
 allow-connections h323 to h323
 h323
   h225 h245-address
   ccm-compatible
   call sync-rsvp slow-start
!
gateway
!
interface FastEthernet0/1
 ip address 10.10.1.10 255.255.255.0
 h323-gateway voip interface
 h323-gateway voip id A-VIAGK ipaddr
    10.10.100.10
 h323-gateway voip h323-id A-IPIPGW
 h323-gateway voip bind srcaddr
10.10.1.10
!
dial-peer voice 10 voip
 destination-pattern [3-4]...
 session target ras
 req-qos guaranteed-delay audio
 req-qos guaranteed-delay video
 acc-qos guaranteed-delay audio
 acc-qos guaranteed-delay video
 codec transparent
!
dial-peer voice 11 voip
 destination-pattern [1-2]...
 session target ras
 codec transparent
```

Route Pattern: [34]XXX
Route List: SiteB_RL
Route Group: SiteB_RG
Trunk: Intercluster Trunk, MTP Required

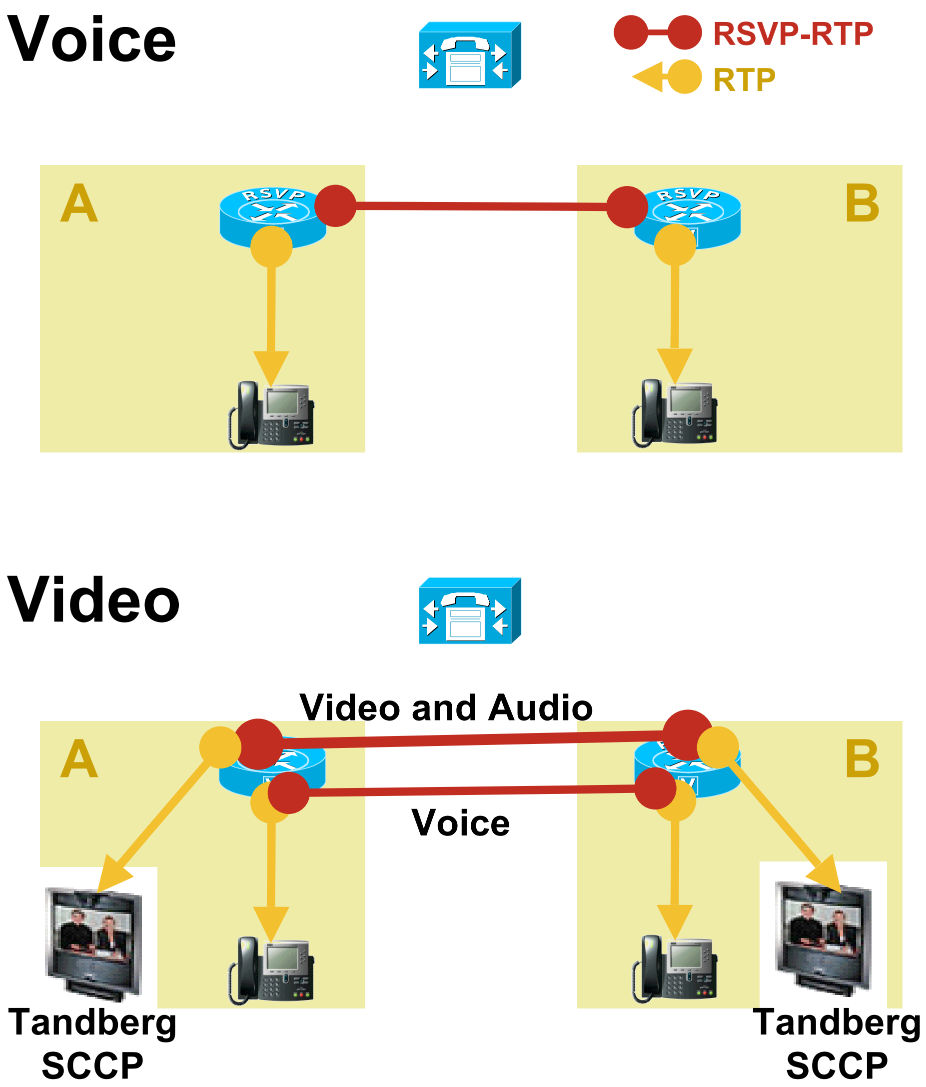
Route Pattern: 2XXX
Route List: SiteA2_RL
Route Group: SiteA2_RG
Trunk: Intercluster Trunk, No MTP Required



Appendix: Cisco CallManager with RSVP—Enabled Locations

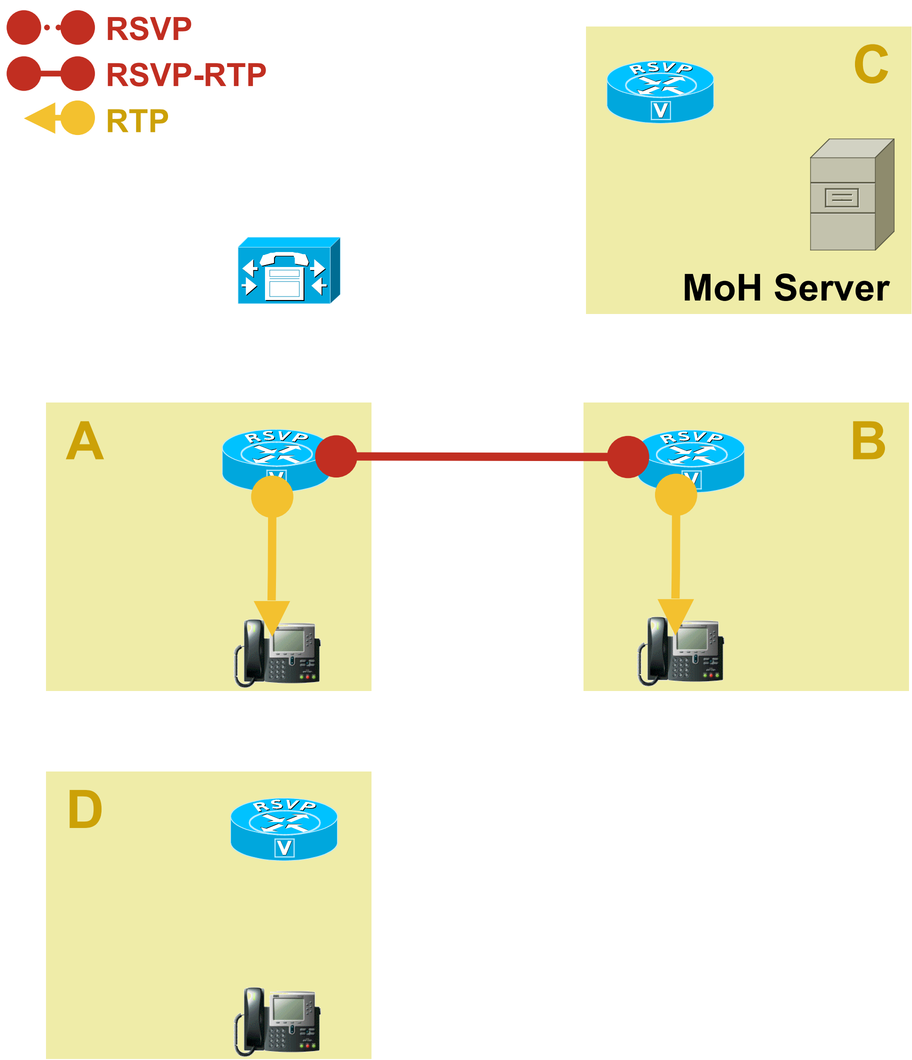
CallManager RSVP-Enabled Locations
Basic Voice and Video Calls

- RSVP between A and B established by Cisco Unified CM based on locations policy configuration
- Reservations may be used for audio and video, or for audio only (depends on configured policy)



CallManager RSVP-Enabled Locations
Transfer: Initial Call

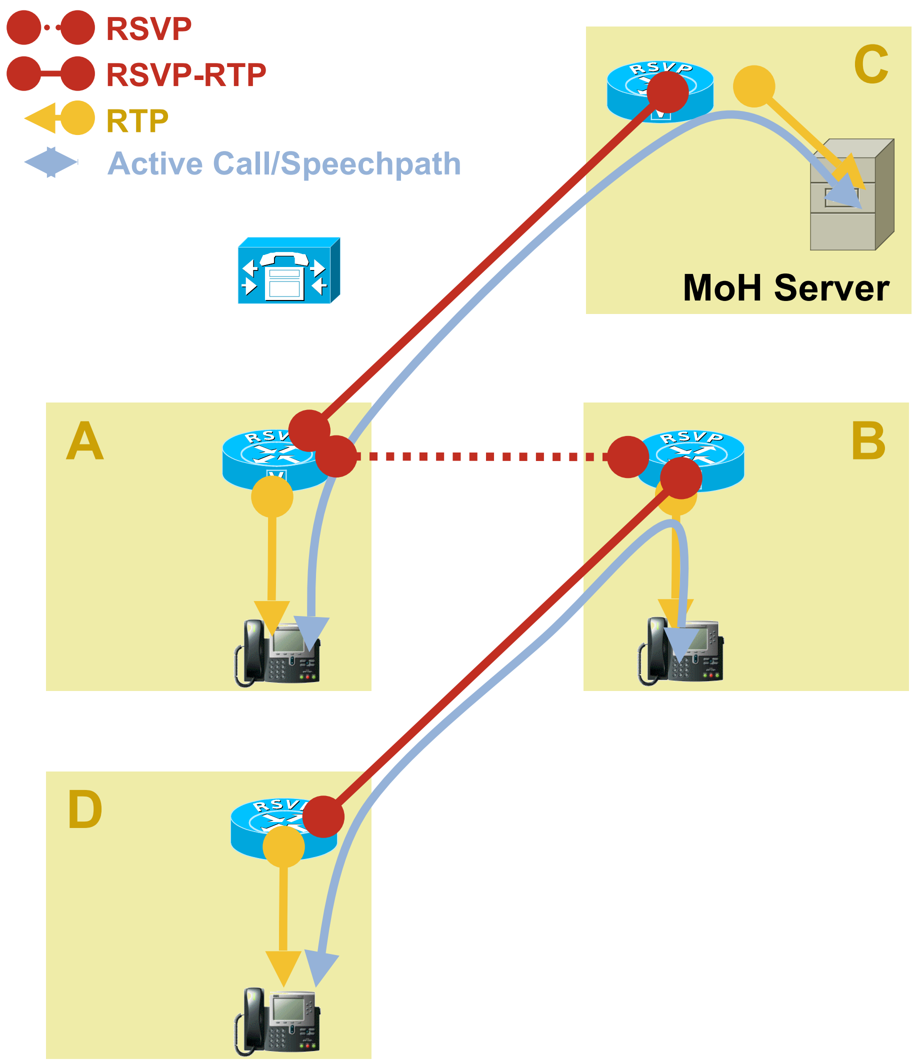
- A calls B
- B puts A on hold and dials D
- A hears MoH
- B completes the transfer
- A talks to D



CallManager RSVP-Enabled Locations Transfer: MOH and Xfer Consult Call

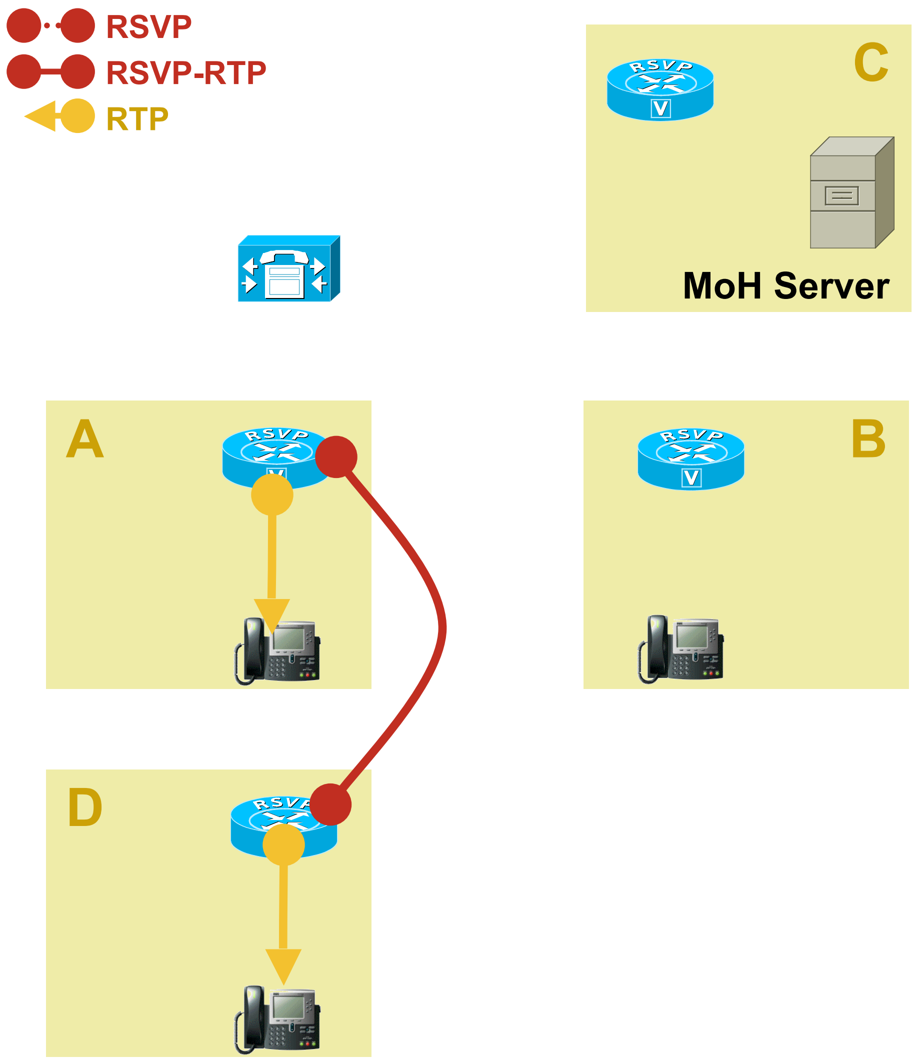
- A calls B
- B puts A on hold and dials D
- A hears MoH
- B completes the transfer
- A talks to D

- A new RSVP reservation is set up from A to C for MoH—same variations as the preceding MOH scenarios
- A new RSVP reservation is set up from B to D for the transfer consult call
- Existing A-B reservation is preserved
- There are now three RSVP reservations; two of which carry RTP packets



CallManager RSVP-Enabled Locations
Transfer: Complete Xfer

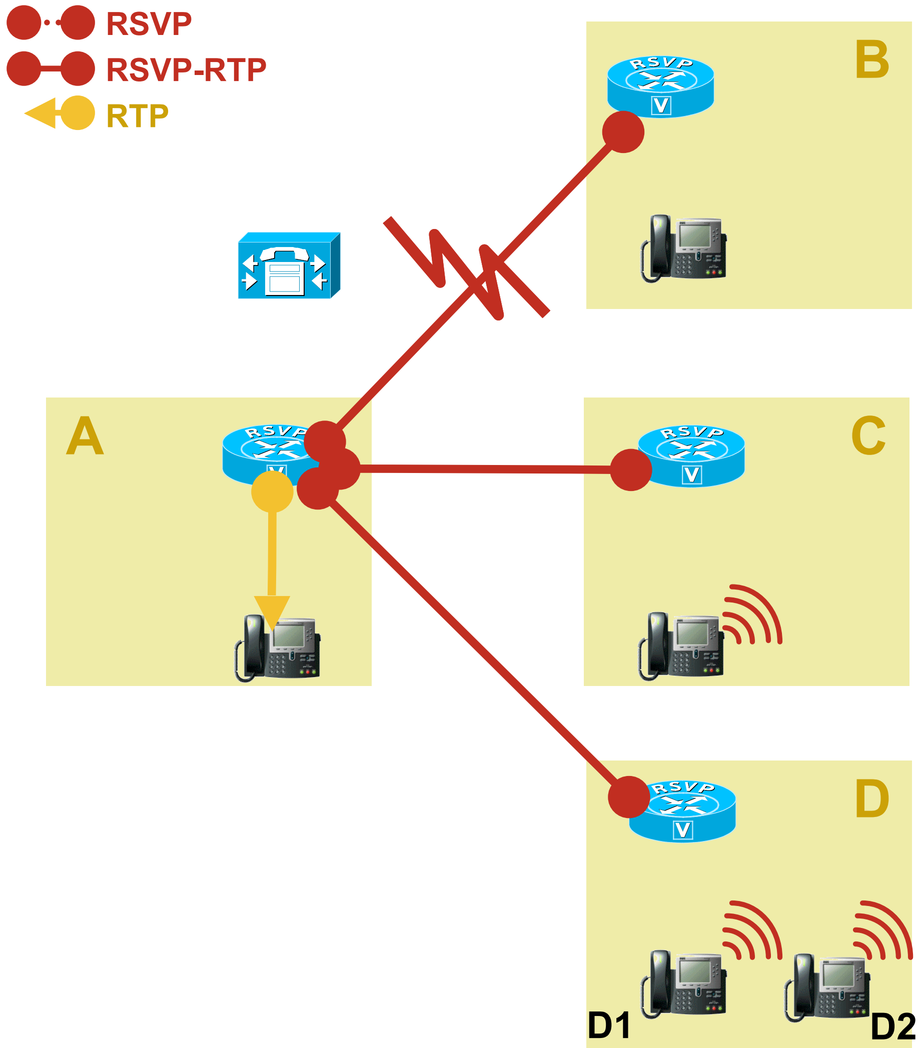
- A calls B
- B puts A on hold and dials D
- A hears MoH
- B completes the transfer
- A talks to D



- The MoH RSVP reservation from A to C is torn down
- The consult call RSVP reservation from B to D is torn down
- A new RSVP reservation is set up from A to D
- If D was collocated with either C (MoH source) or B (original called party), the existing RSVP A-C, or A-B reservation would have been reused for the final call path

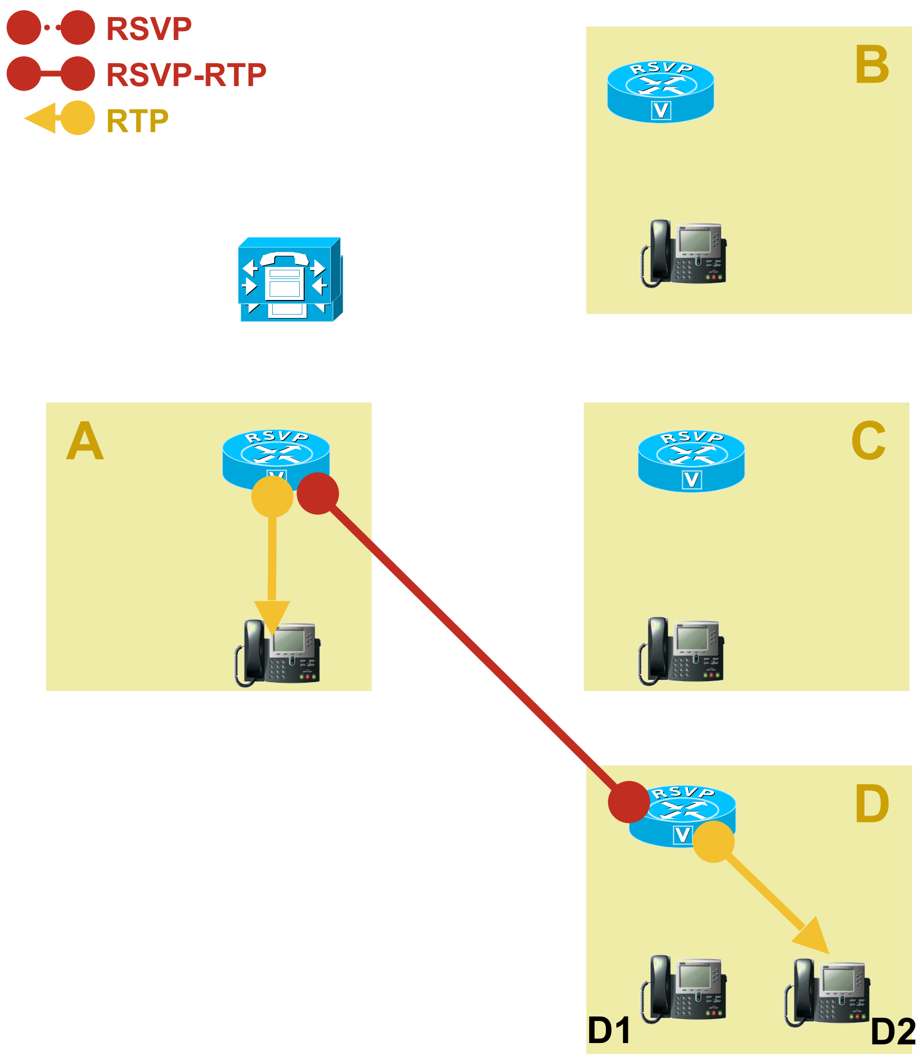
CallManager RSVP-Enabled Locations
Shared Line: Ringing

- A calls DN 2000, which appears on phones in Site B, C and D (both D1 and D2)
- RSVP reservations are set up from A to all sites (B, C and D) that have appearances of the shared line
- If the RSVP reservation fails (e.g. B)—for a “mandatory” policy site, that phone does not ring
- A single RSVP reservation (D) is set up to sites that have multiple phones with the shared line
- RSVP agent at A has one port with multiple destinations

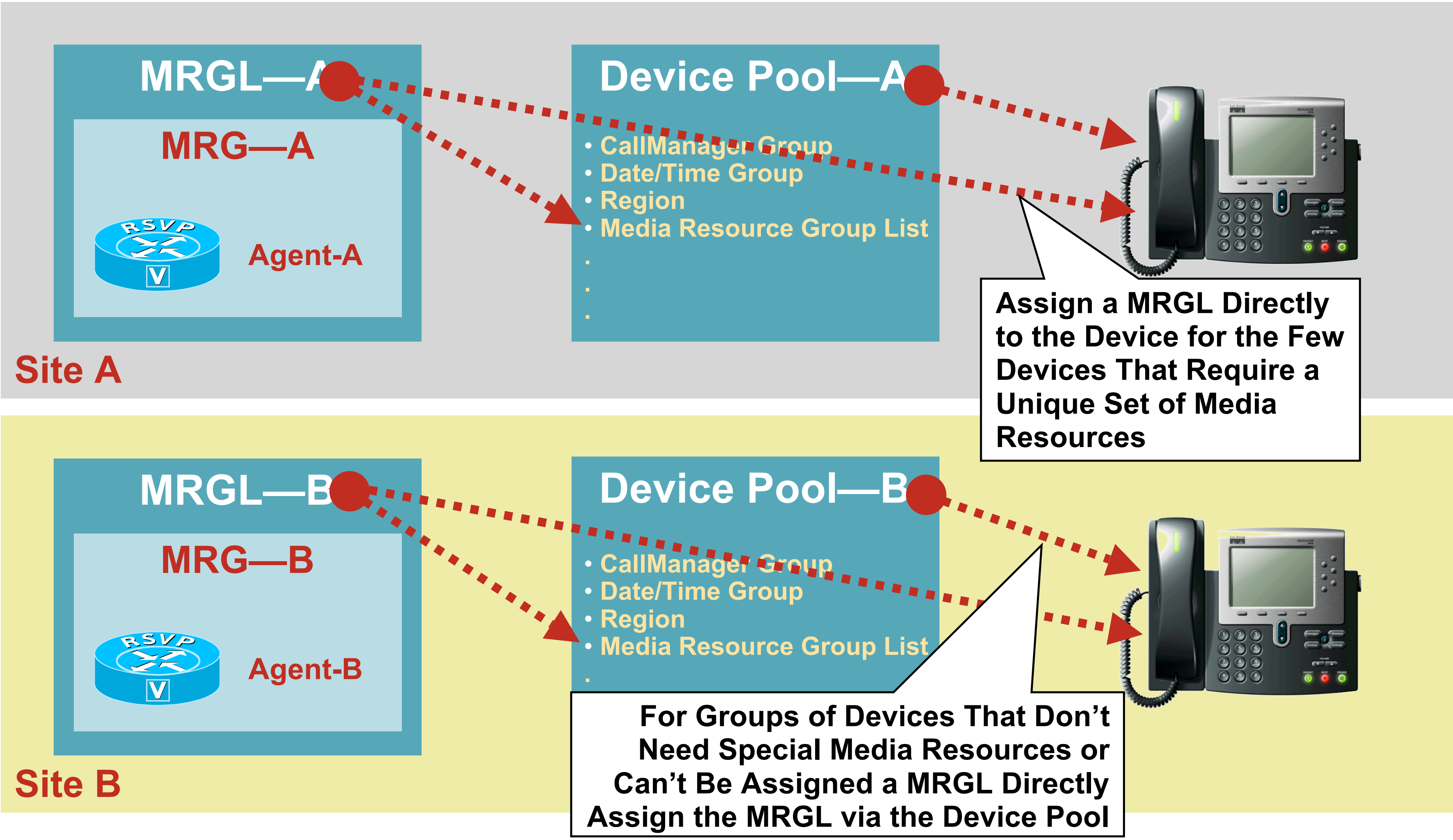


CallManager RSVP-Enabled Locations
Shared Line: Answer

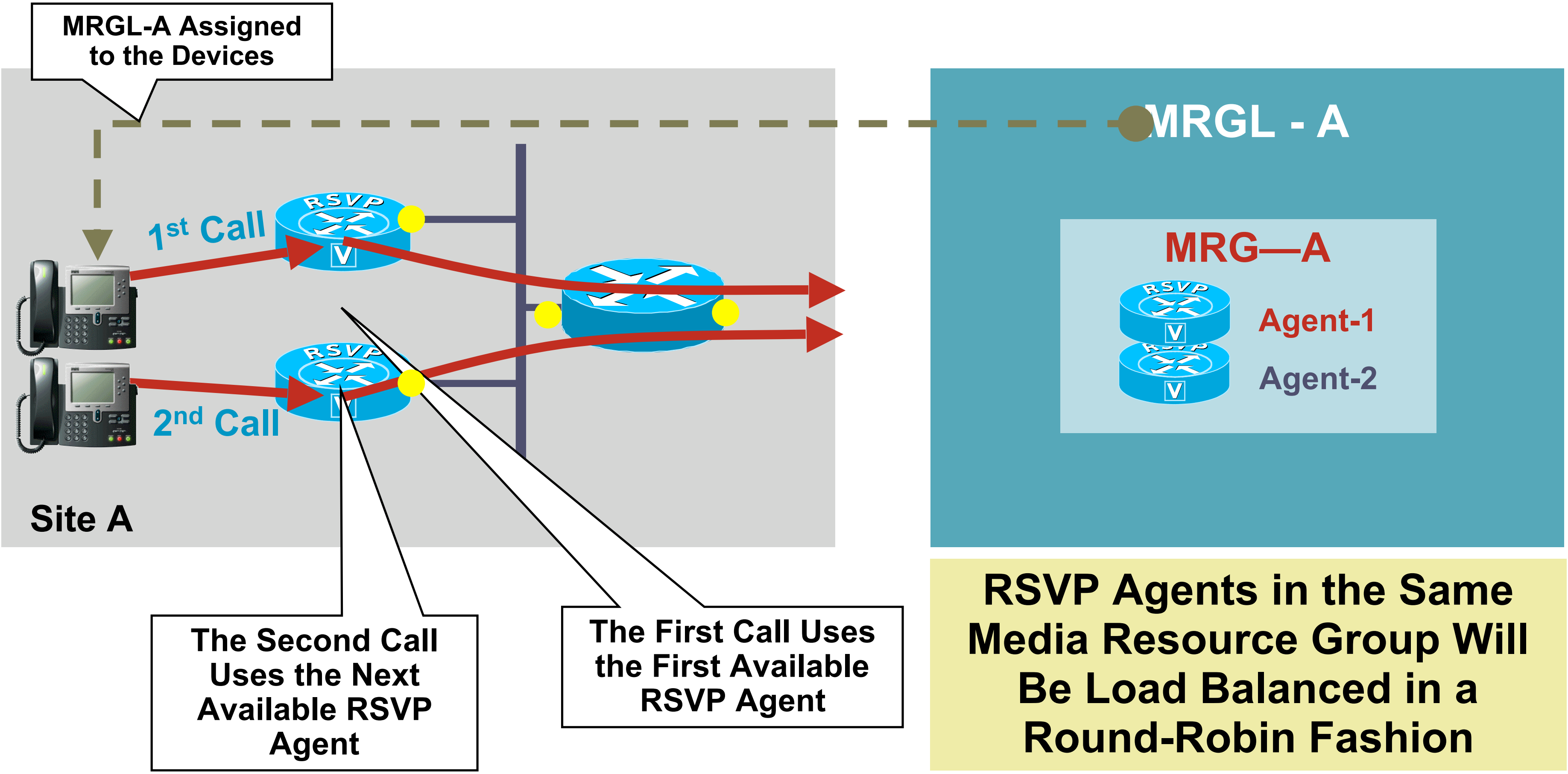
- When phone D2 answers, the RSVP reservation from A-D already in place is used
- All other phones stop ringing
- All other RSVP reservations are torn down



CallManager RSVP-Enabled Locations
How an RSVP Agent Is Selected



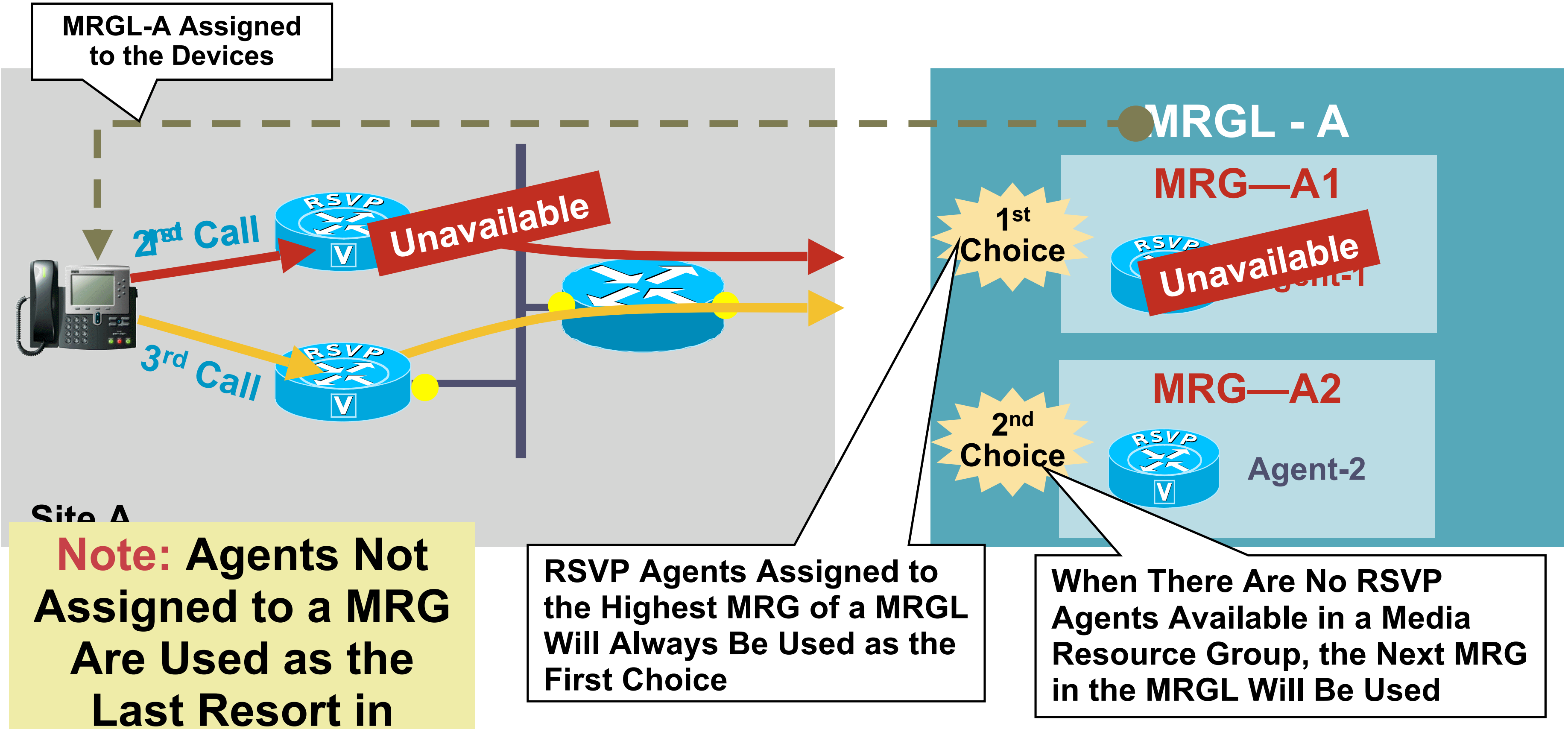
CallManager RSVP-Enabled Locations
Load Balancing



RSVP Agents in the Same Media Resource Group Will Be Load Balanced in a Round-Robin Fashion

Assign RSVP Agents to the Same Media Resource Group for Load-Balancing

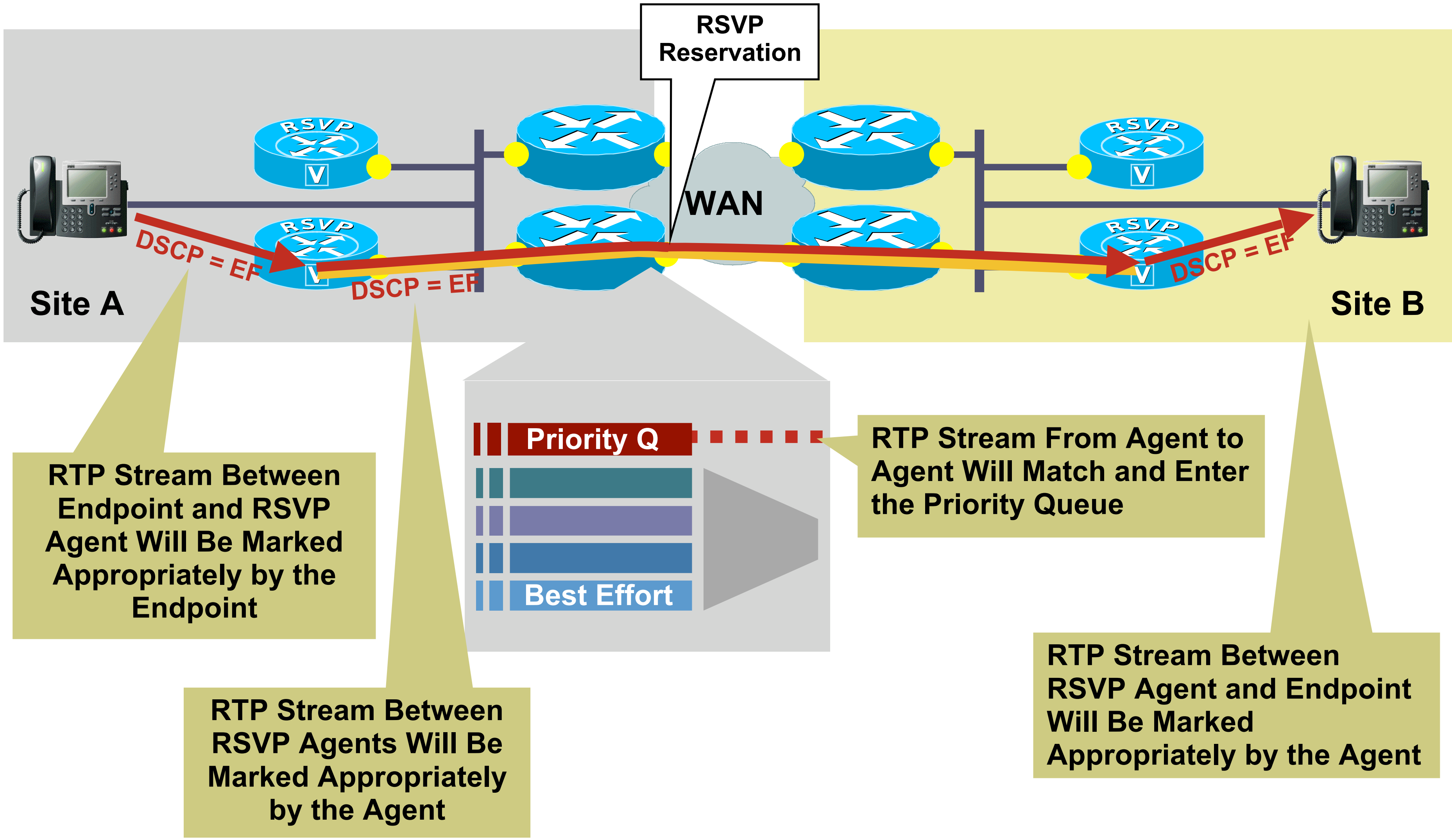
CallManager RSVP-Enabled Locations Redundancy



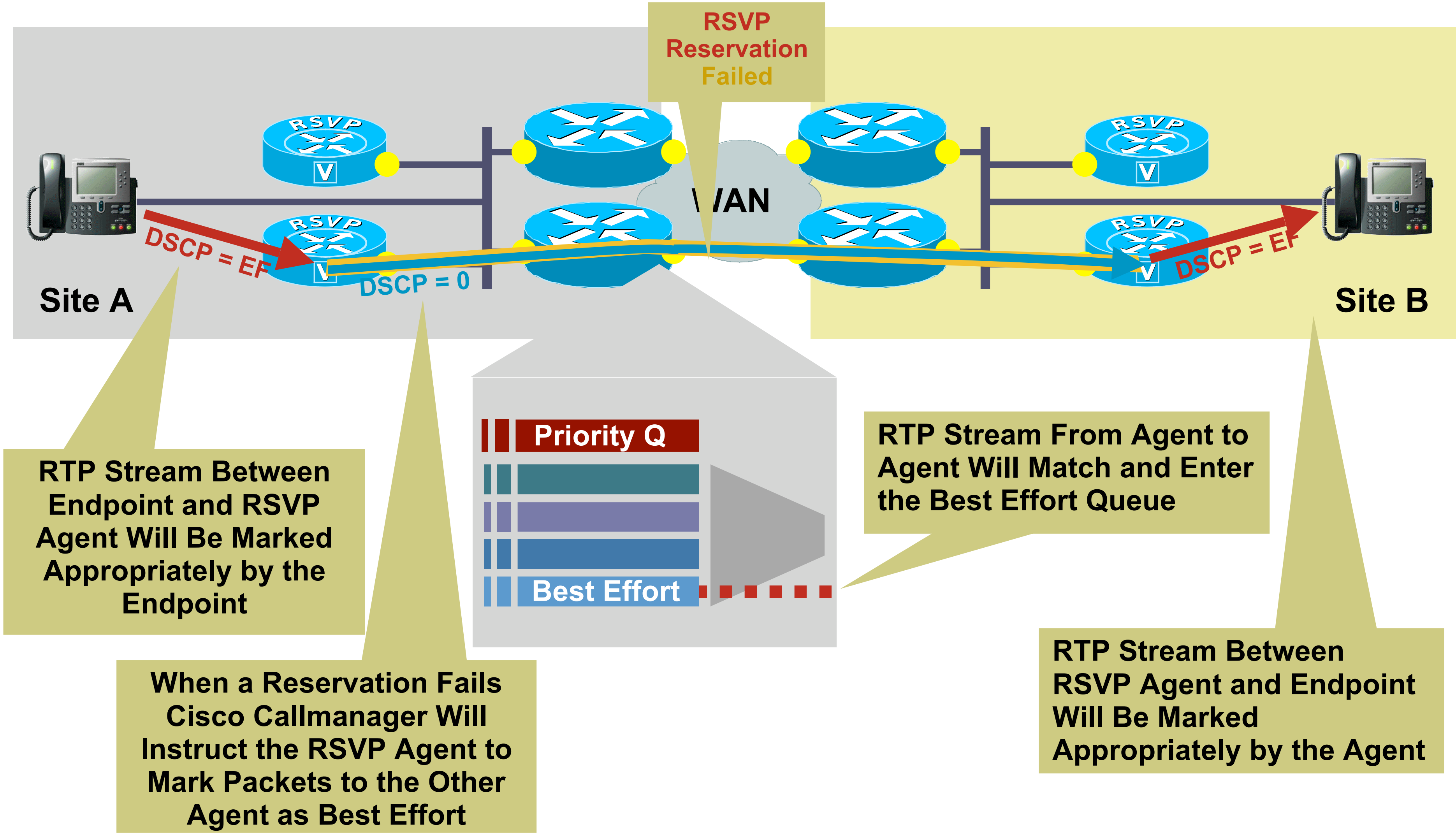
Note: Agents Not Assigned to a MRG Are Used as the Last Resort in All Mrgls

Assign RSVP Agents to Separate Media Resource Groups but the Same Resource List for Redundancy

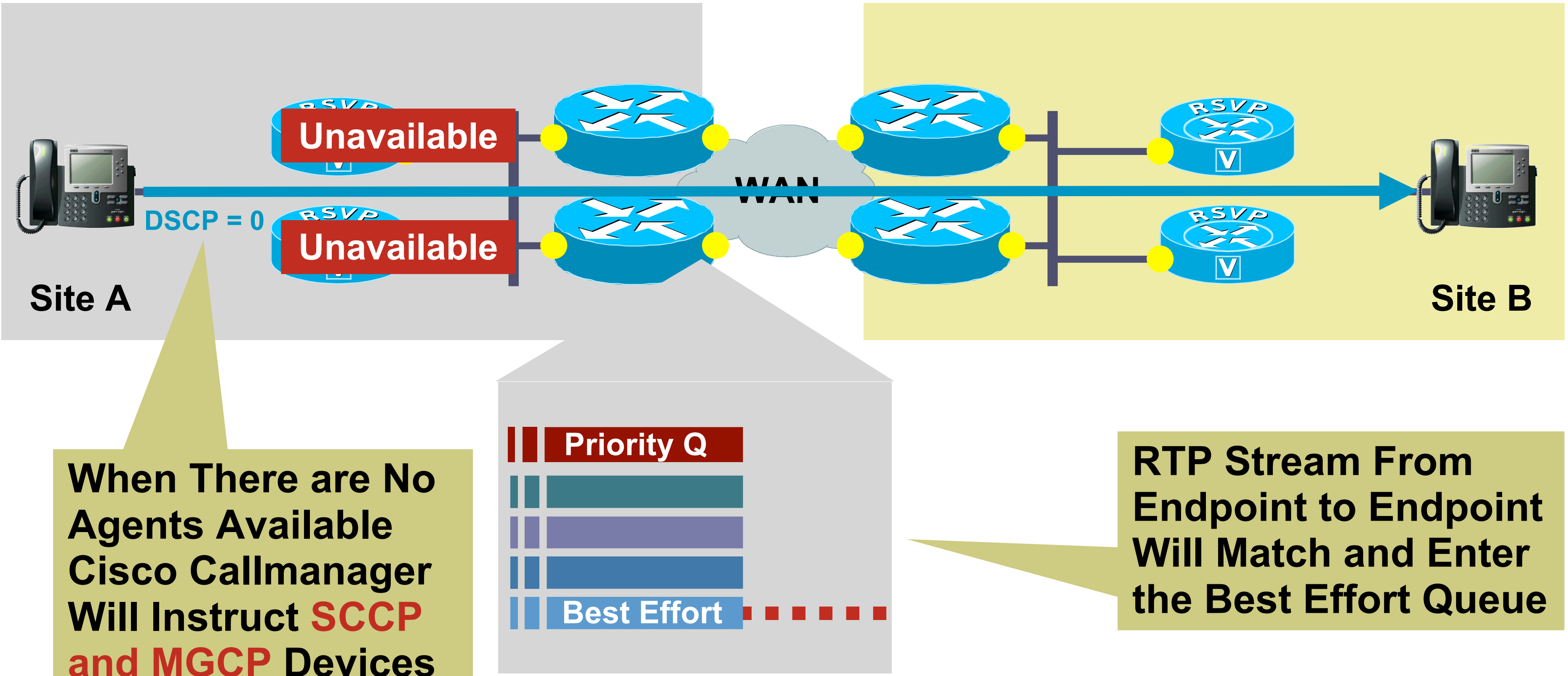
CallManager RSVP-Enabled Locations DSCP Marking—Normal Operation



CallManager RSVP-Enabled Locations
Optional RSVP Policy: Reservation Failure



CallManager RSVP-Enabled Locations
Optional RSVP Policy: Agent Unavailable

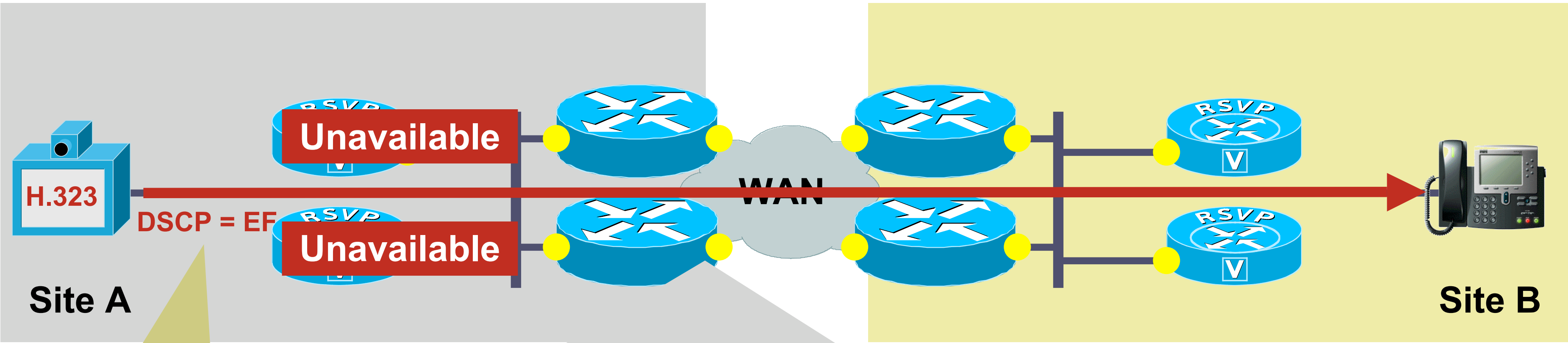


When There are No Agents Available Cisco Callmanager Will Instruct SCCP and MGCP Devices to Mark Packets to the Other Endpoint as Best Effort

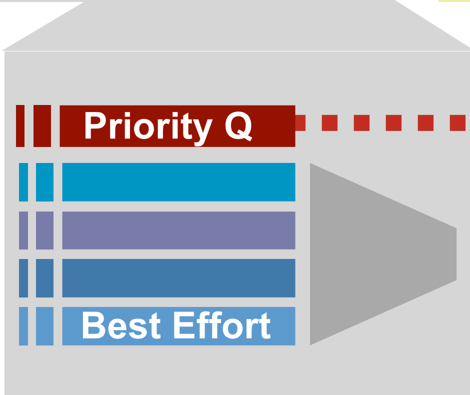
RTP Stream From Endpoint to Endpoint Will Match and Enter the Best Effort Queue

Video Calls Will Always Become "Audio-Only" After a Reservation Failure

CallManager RSVP-Enabled Locations
Optional RSVP Policy: Agent Unavailable (2)



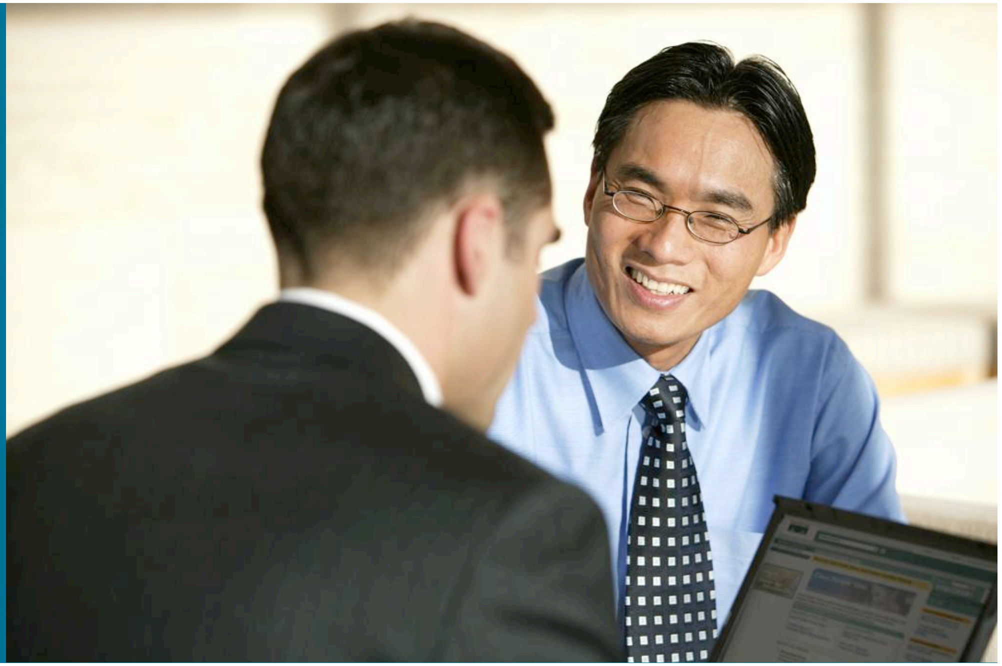
Cisco Callmanager Can Not Instruct H.323 or SIP Devices to Adjust Their QoS Markings



RTP Stream From Endpoint to Endpoint Will Match and Enter the Priority Queue

To Ensure That the Priority Queue Is Not Overrun, Set up an ACL in the WAN Router to Only Permit Packets Marked EF and AF41 If They Originate from RSVP Agent

Appendix: RSVP MPLS Concerns

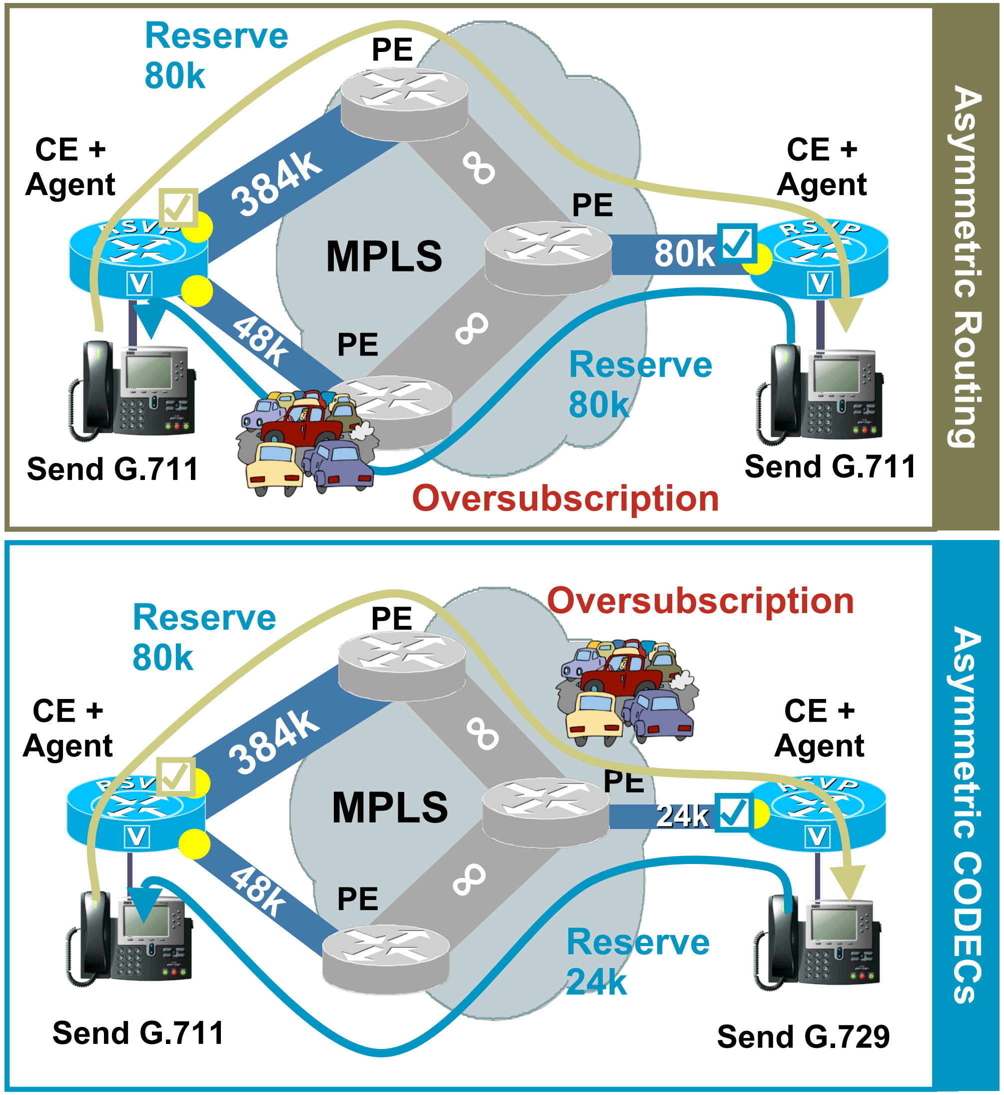


MPLS Considerations
SP-Owned MPLS-Based IP WAN

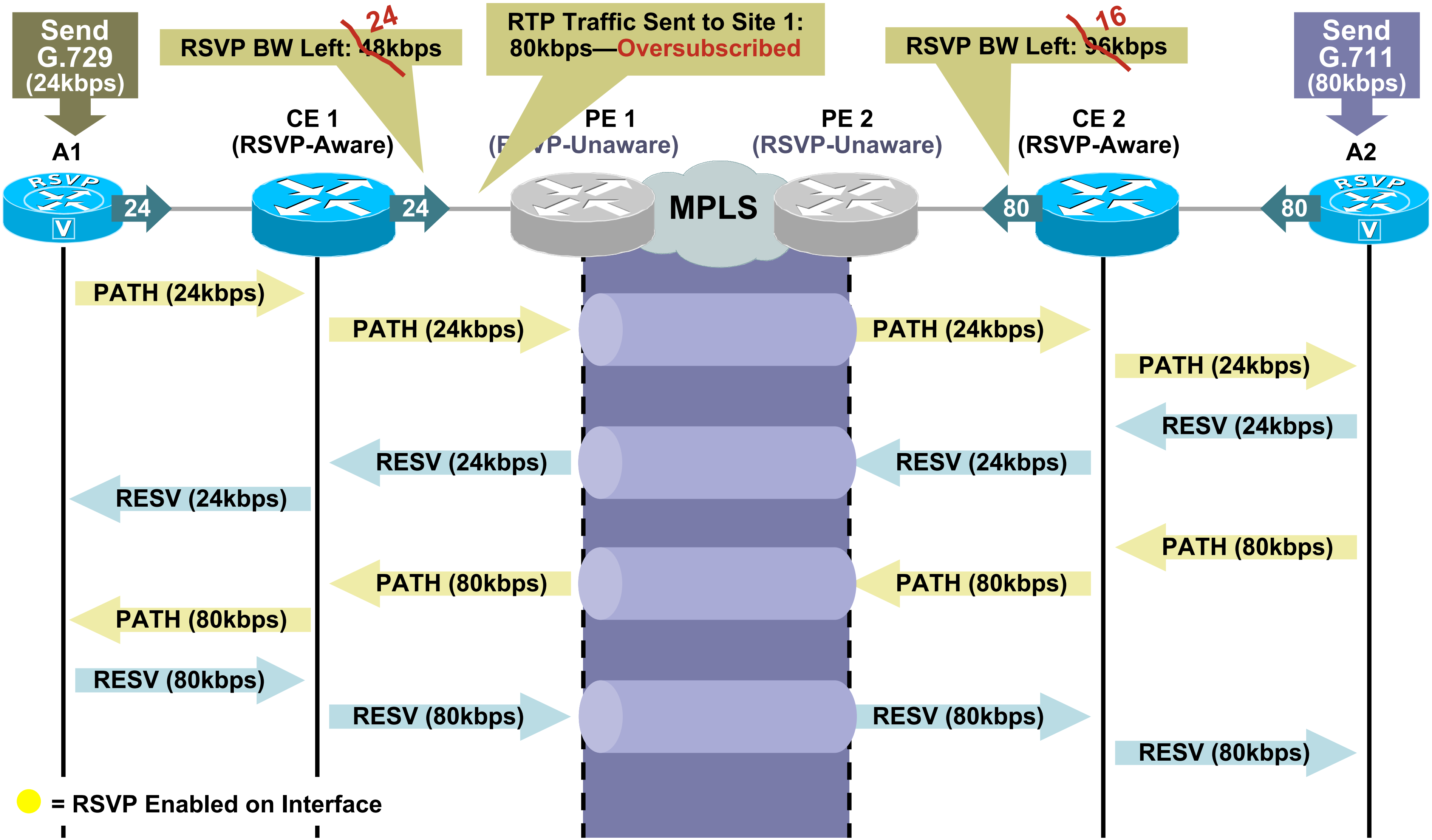
- Enterprise controls CE routers (not always)
- RSVP can't be enabled on PE routers (RSVP is not VRF-aware)
- Half-duplex RSVP reservations can only be made on CE's outgoing interfaces
- While not common, any asymmetry (routing, CODEC) presents a problem for RSVP CAC

Legend:

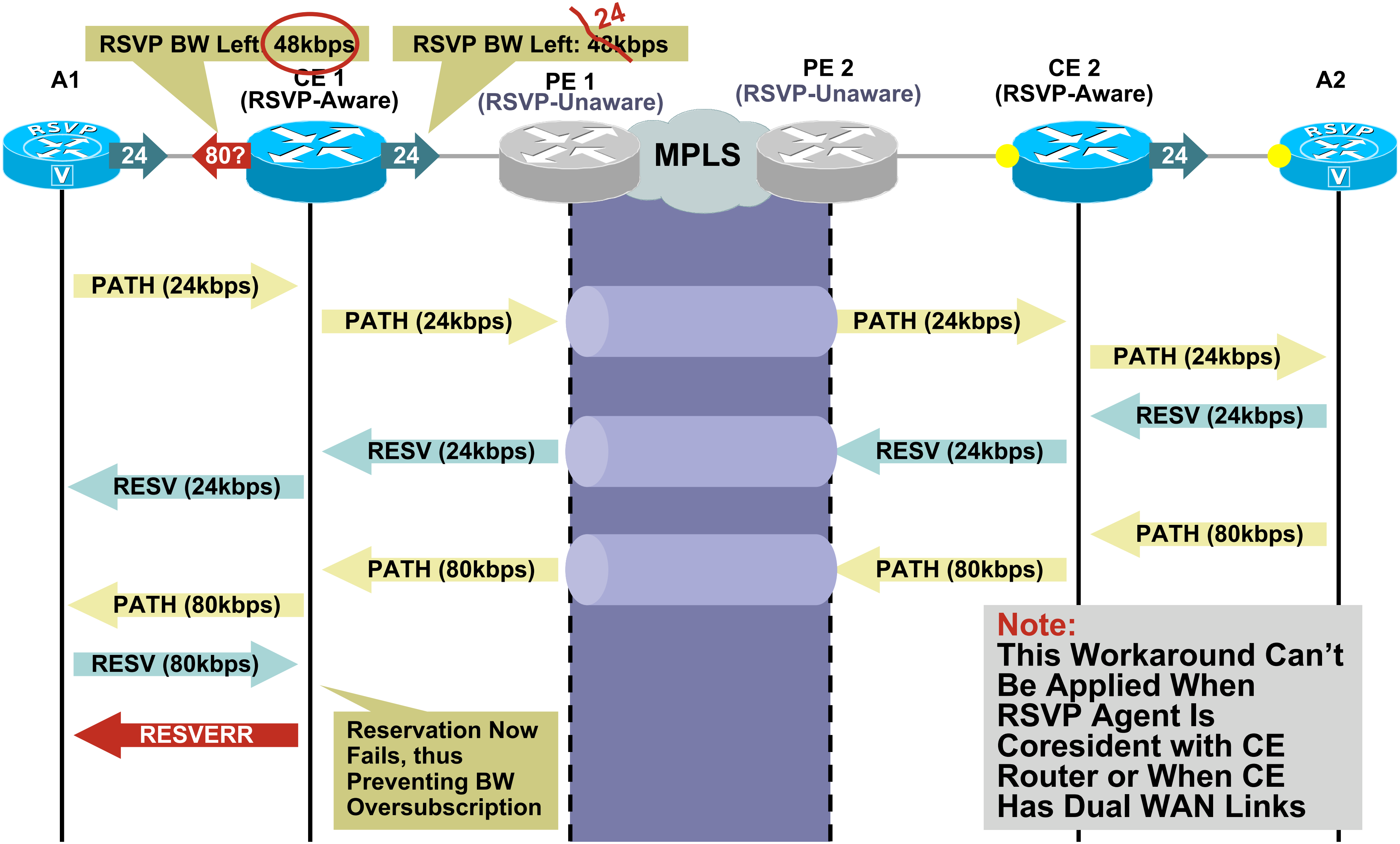
= RSVP Enabled on Router Interface



MPLS Considerations
Problem: Asymmetric CODECs



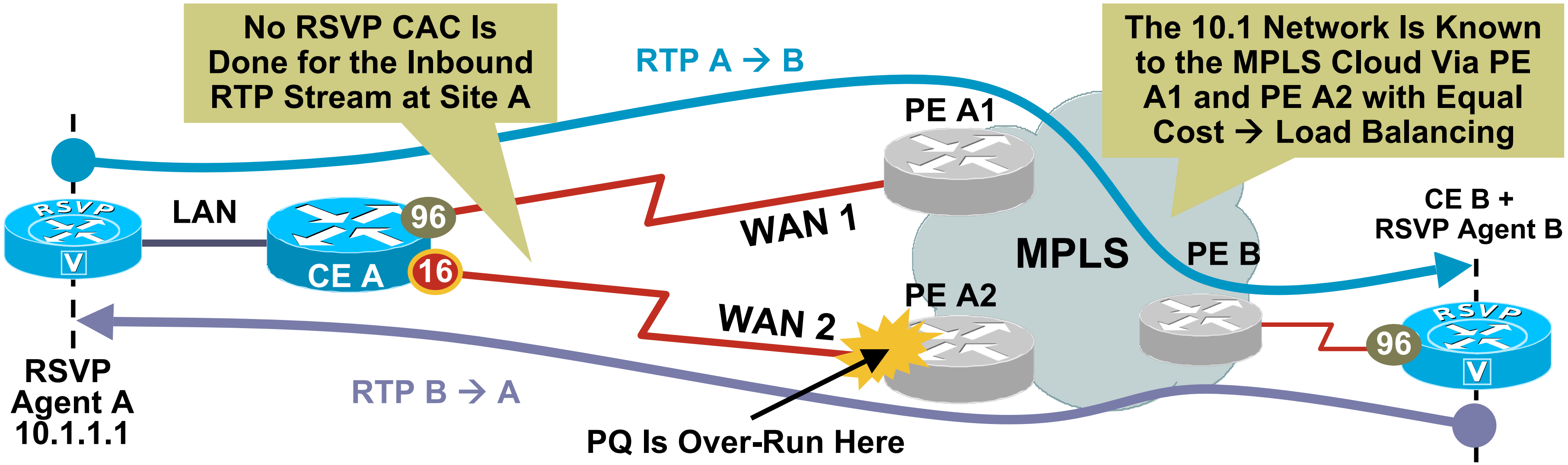
MPLS Considerations
Workaround: RSVP BW Config on CE's LAN i/f



Note:
This Workaround Can't Be Applied When RSVP Agent Is Coresident with CE Router or When CE Has Dual WAN Links

MPLS Considerations

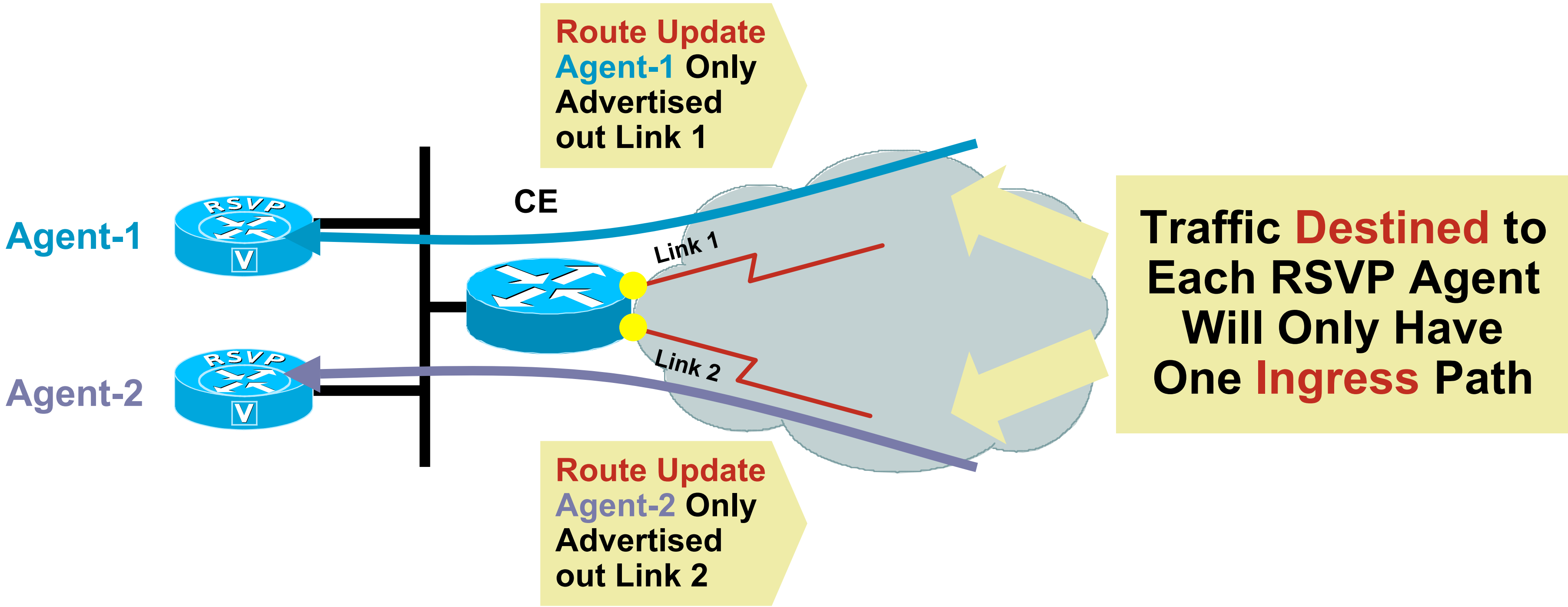
Problem: Dual-Attached CE with Load Balancing



- With load balancing, inbound traffic for RSVP agent not guaranteed to use same WAN link as outbound traffic
- This can over-run PQ, as reservations are only made on outgoing CE interfaces (in the direction of RTP stream)
- Problem does not apply to MPLS scenarios with primary/backup WAN links

Legend:
16 = Kbps Remaining in RSVP Bandwidth Pool

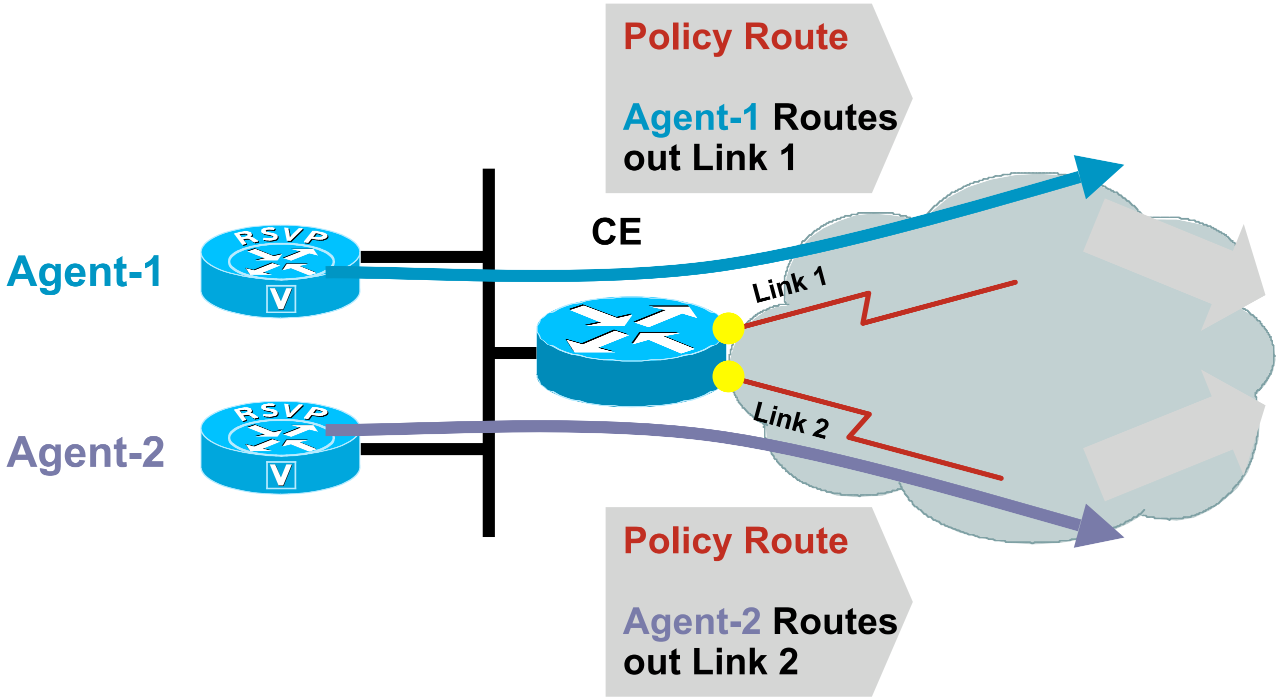
MPLS Considerations
Load-Balancing Workaround—Inbound Traffic



To "Associate" a RSVP Agent to a Specific WAN Link:

1. **Allow the Route to Reach the Specified Agent to Be Advertised out That Specific WAN Link Only**

MPLS Considerations
Load-Balancing Workaround—Outbound Traffic



Traffic Sourced From Each RSVP Agent Will Only Have One Egress Path

To "Associate" a RSVP Agent to a Specific WAN Link:

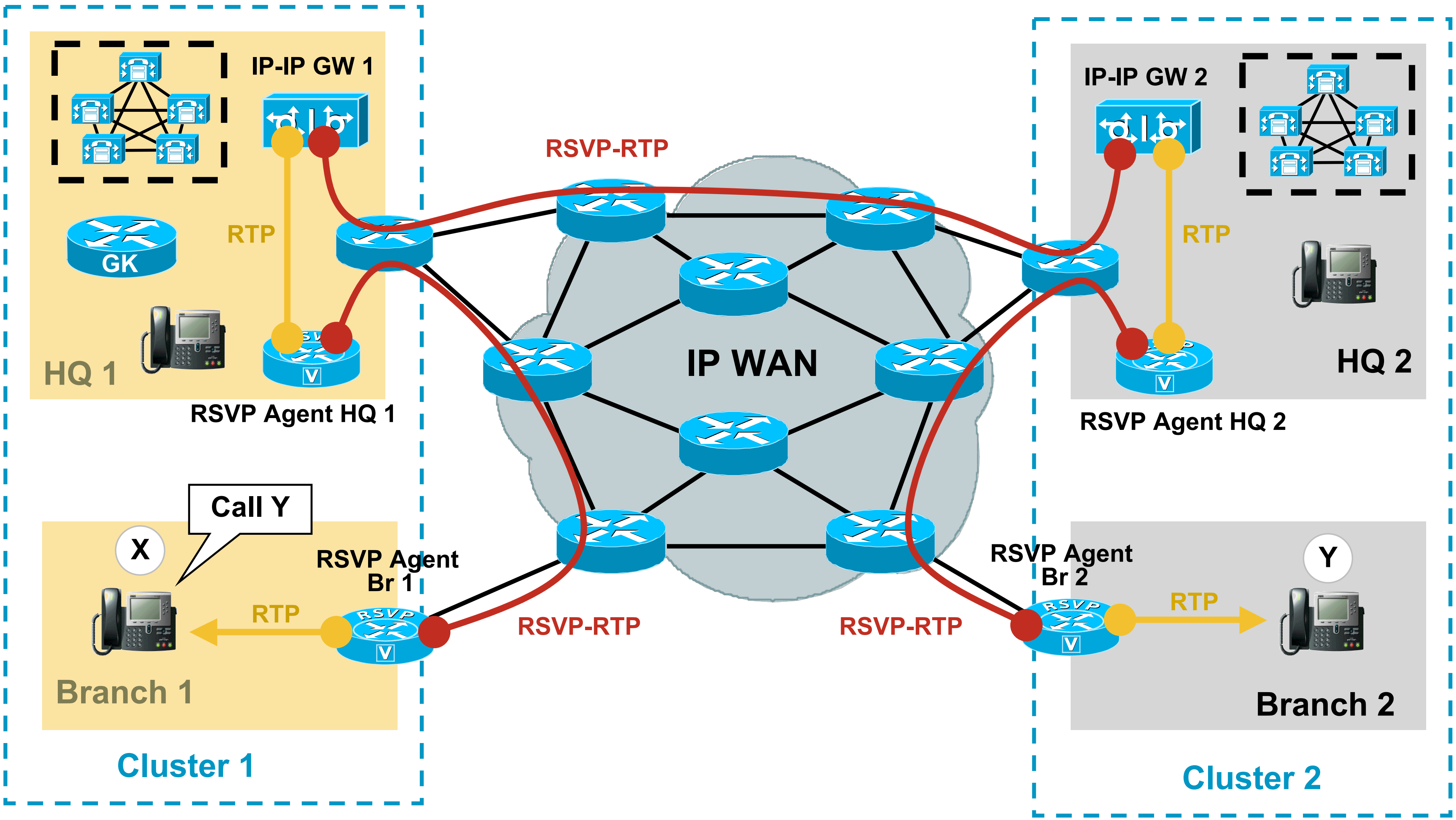
2 Policy Route Packets Sourced from the Specified Agent out That Specific WAN Link Only

MPLS Considerations
Summary of Issues

- Asymmetric CODECs when RSVP agent is co-resident with CE router
  - This should be relatively unlikely, except for cases like VTA “receive-only” mode
  - No workaround available—need to prevent it
- Dual-attached CE router with load balancing
  - This is quite common in medium-to-large sites
  - Workaround requires PBR + route filtering on CE router
  - Asymmetric CODEC issue also applies to this scenario
- All other scenarios should not present problems (including dual WAN links in primary/ backup config)
- Contact Cisco account team for RSVP MPLS issue

Appendix: Generic Topologies Distributed Deployments: IP-IP Gateway Approach

Generic Topologies
Distr. Deployments—IP-IP Gateway Approach



Generic Topologies
Distr. Deployments—IP-IP Gateway Approach

- Configure ICT with GK control or H.225 trunks
- No MTP required for supplementary services with 12.4(3rd)T or later in IP-IP GW
- MTP is required only for H.323 devices that don't support ECS or when H.323 FastStart is needed

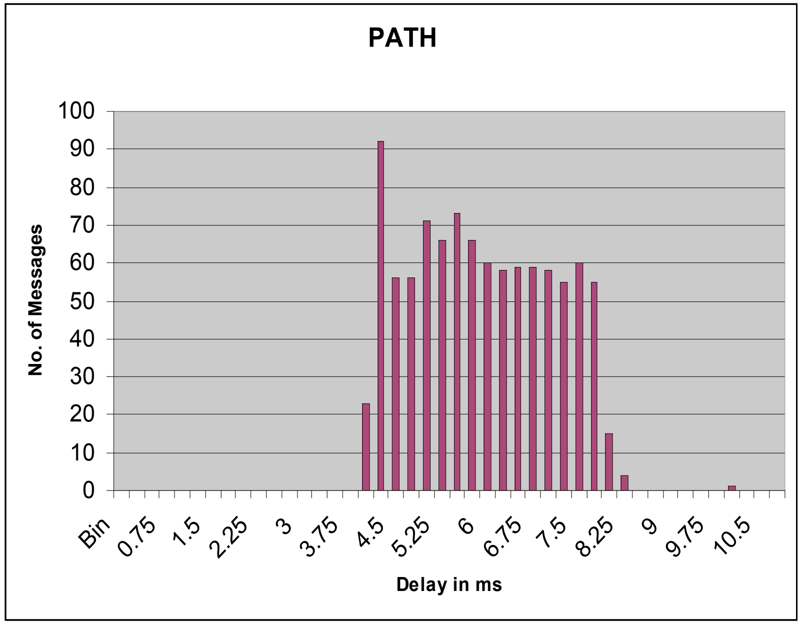
  Note: the "MTP Required" setting is incompatible with the pass-through CODEC

- MTP required on trunks has to be unchecked if video or sRTP calls are needed (they require the pass-through CODEC)

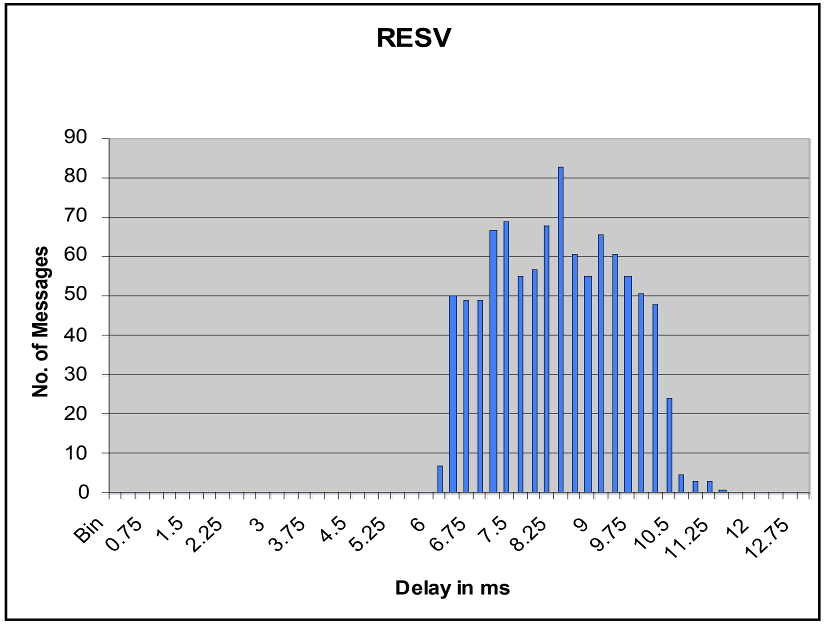
Appendix:
"I heard that RSVP does not scale—"

Networking Mythology 101

RSVP
Latency Test Results



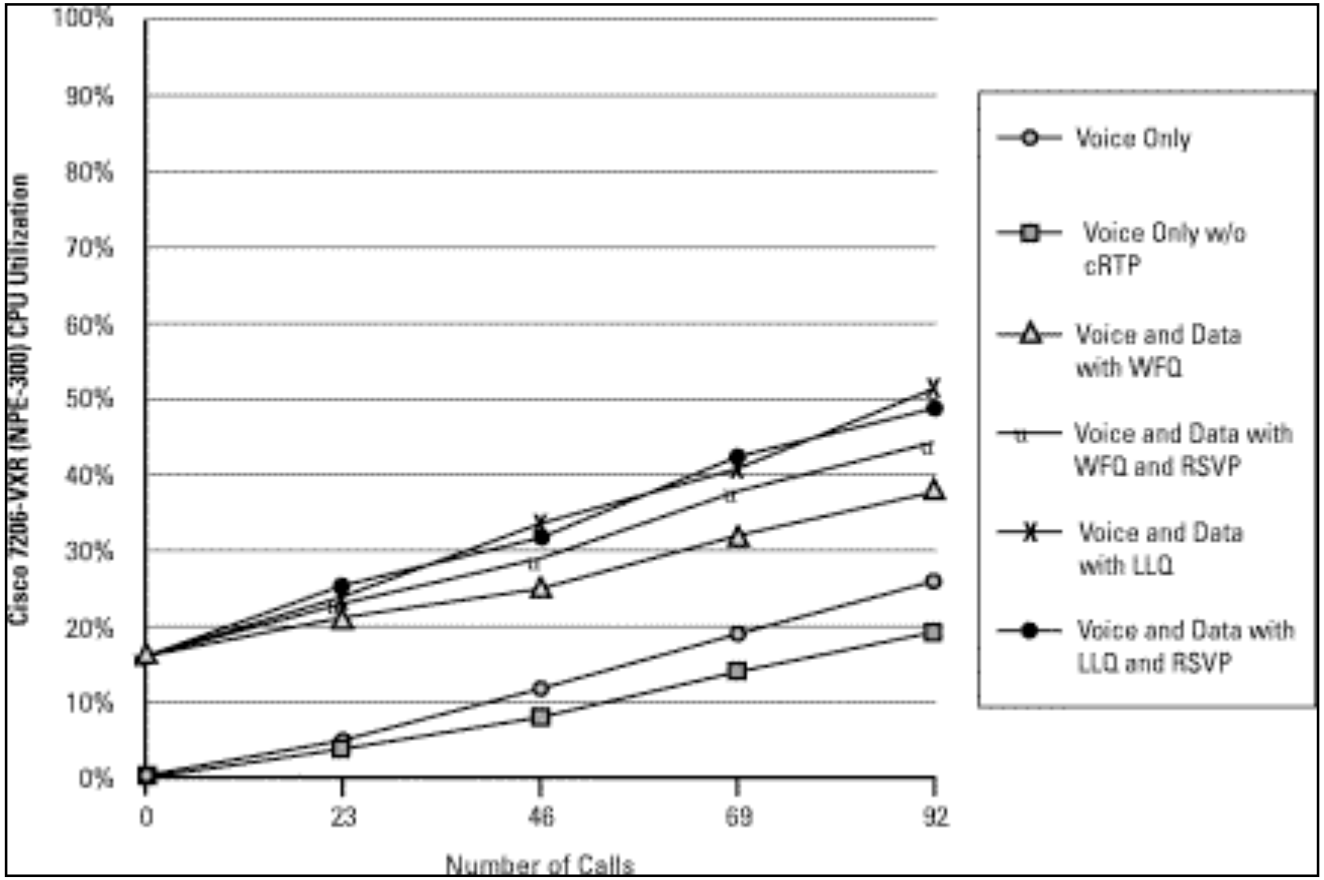
Mean	6.0395403
Mode	5.175388
Standard Deviation	1.1490104
Confidence	0.093922



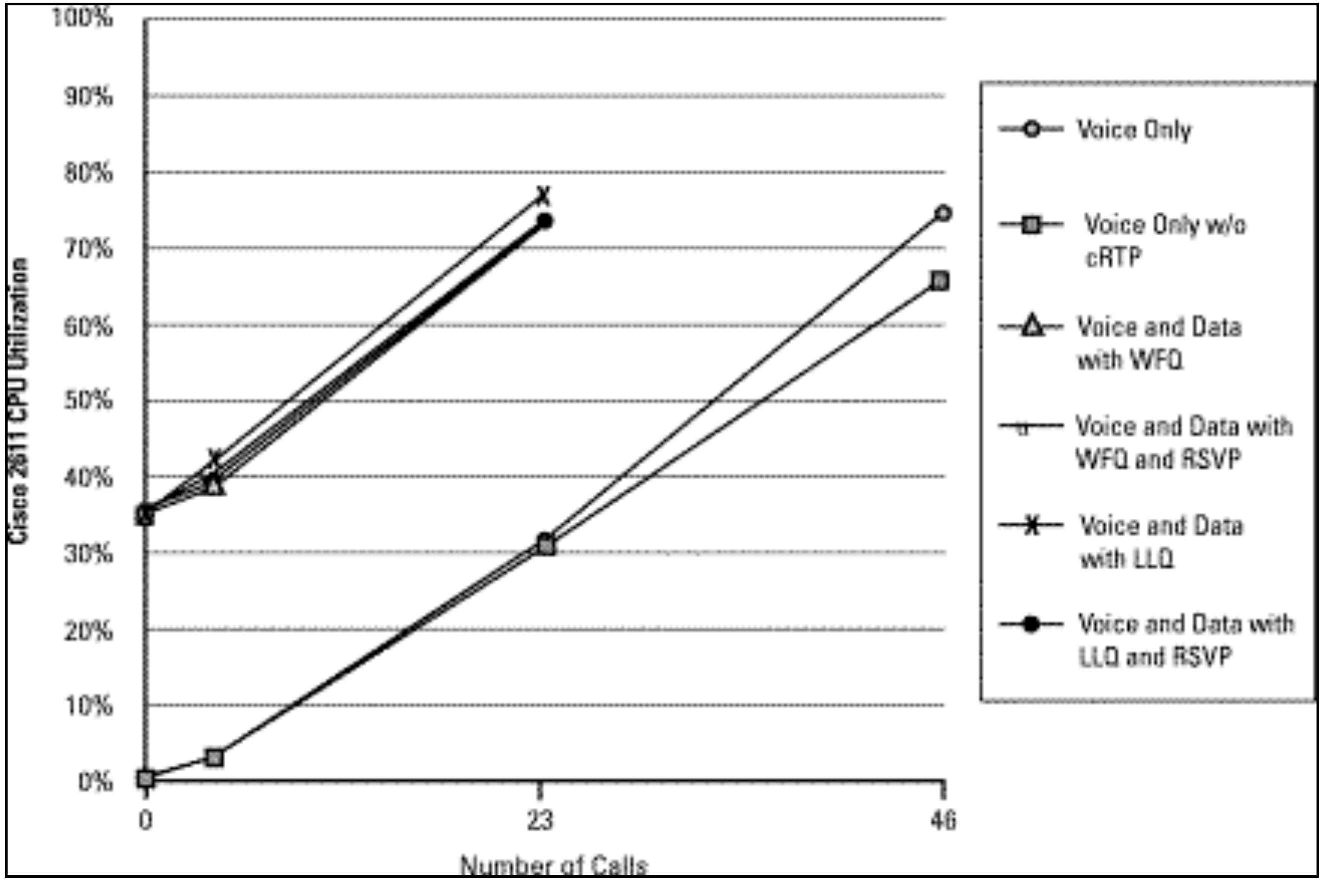
Mean	8.323788
Mode	8.901273
Standard Deviation	1.1616864
Confidence	0.0949581

100 Mbps Background Traffic, RSVP Reservation Installation with Queuing

RSVP
Classification Test Results (7200)



RSVP
Classification Test Results (2600)



RSVP
Latency and Scalability Test Results

- RSVP current performance levels often widely under-estimated
- 1,000 simultaneous RSVP reservations is fine on mid-range router platforms
- Latency very low:

  7200:

    6ms for PATH

    8ms for RESV

  3600:

    ~ 11ms for PATH

    ~ 13ms for RESV

RSVP
Other Scalability Considerations

- Protocol evolution (aggregate flows, refresh reduction, reliable messaging)
- Cisco IOS scalability enhancements (decoupled scheduling)
- New deployment options (IntServ/DiffServ)
- Selective use of RSVP throughout the network (enable only where needed)

www.cisco.com/en/US/partner/products/sw/iosswrel/ps1839/products_feature_guide09186a0080087b49.html

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- Peter Matthews
  Technical Marketing Engineer

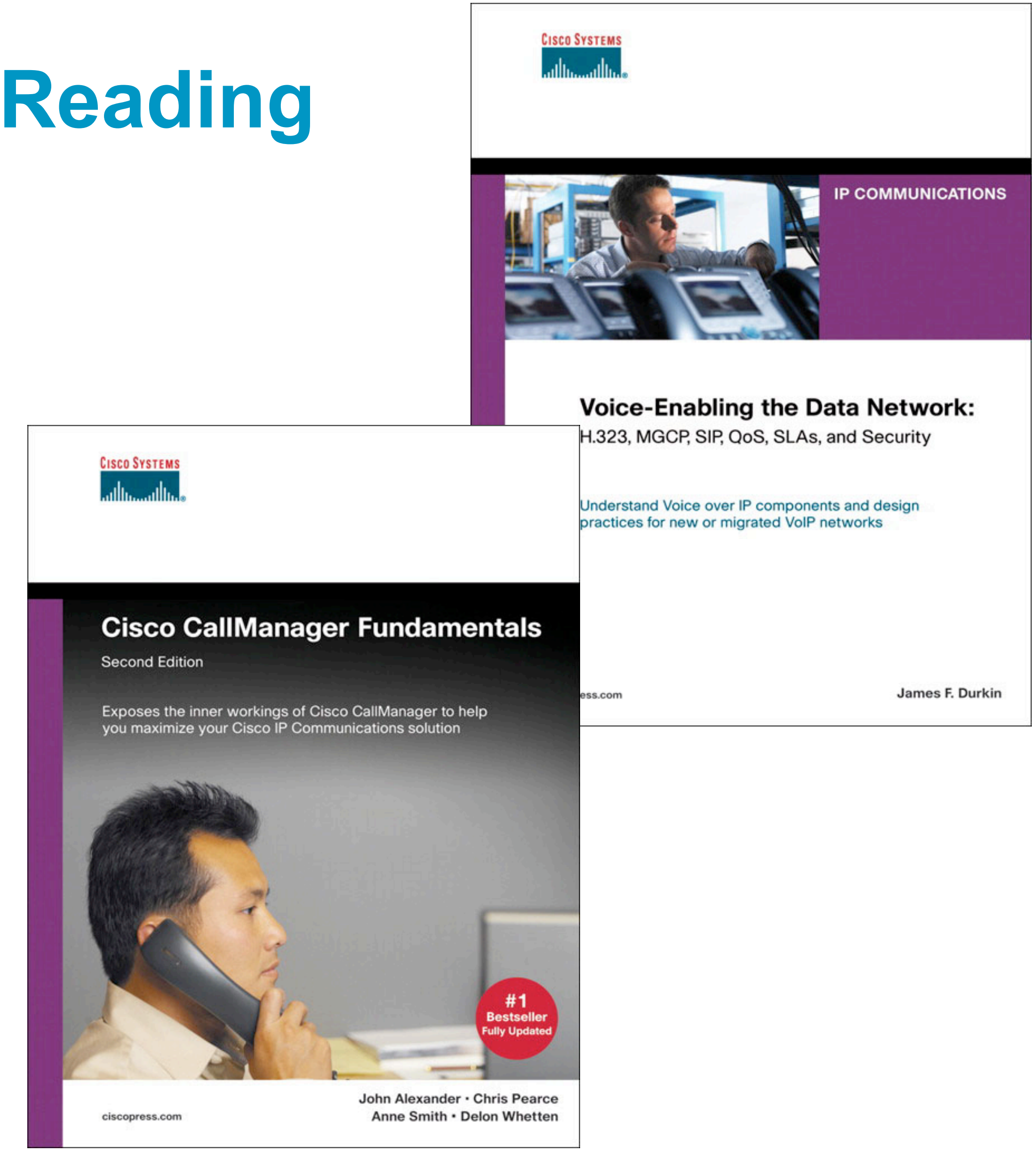
- Scott Wainner
  Distinguished System Engineer

- Steinthor Bjarnason
  Consulting Engineer

Recommended Reading

BRKUCT - 2010

- Cisco CallManager Fundamentals
- Voice-Enabling the Data Network: H.323, MGCP, SIP, QoS, SLAs, and Security



Available in the Cisco Company Store