Enterprise IP Telephony Design and Deployment

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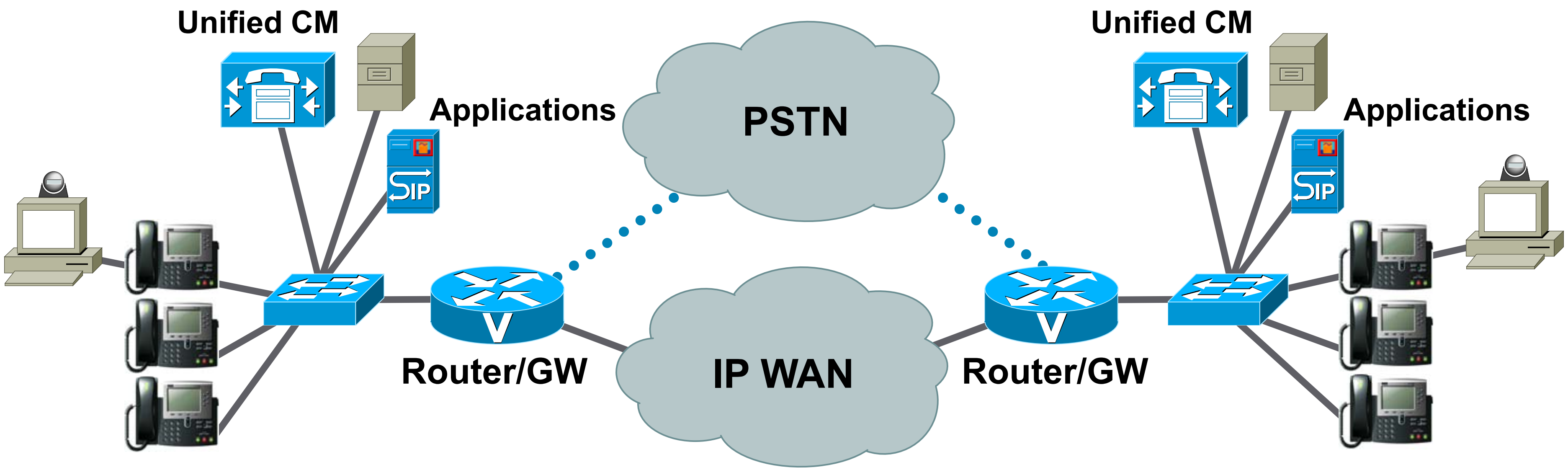
System Engineer – Unified Communication

Agenda

- Introduction
- Network Infrastructure
- Unified Communications Infrastructure
- Unified Communications Applications
- Security

Scope of This Session

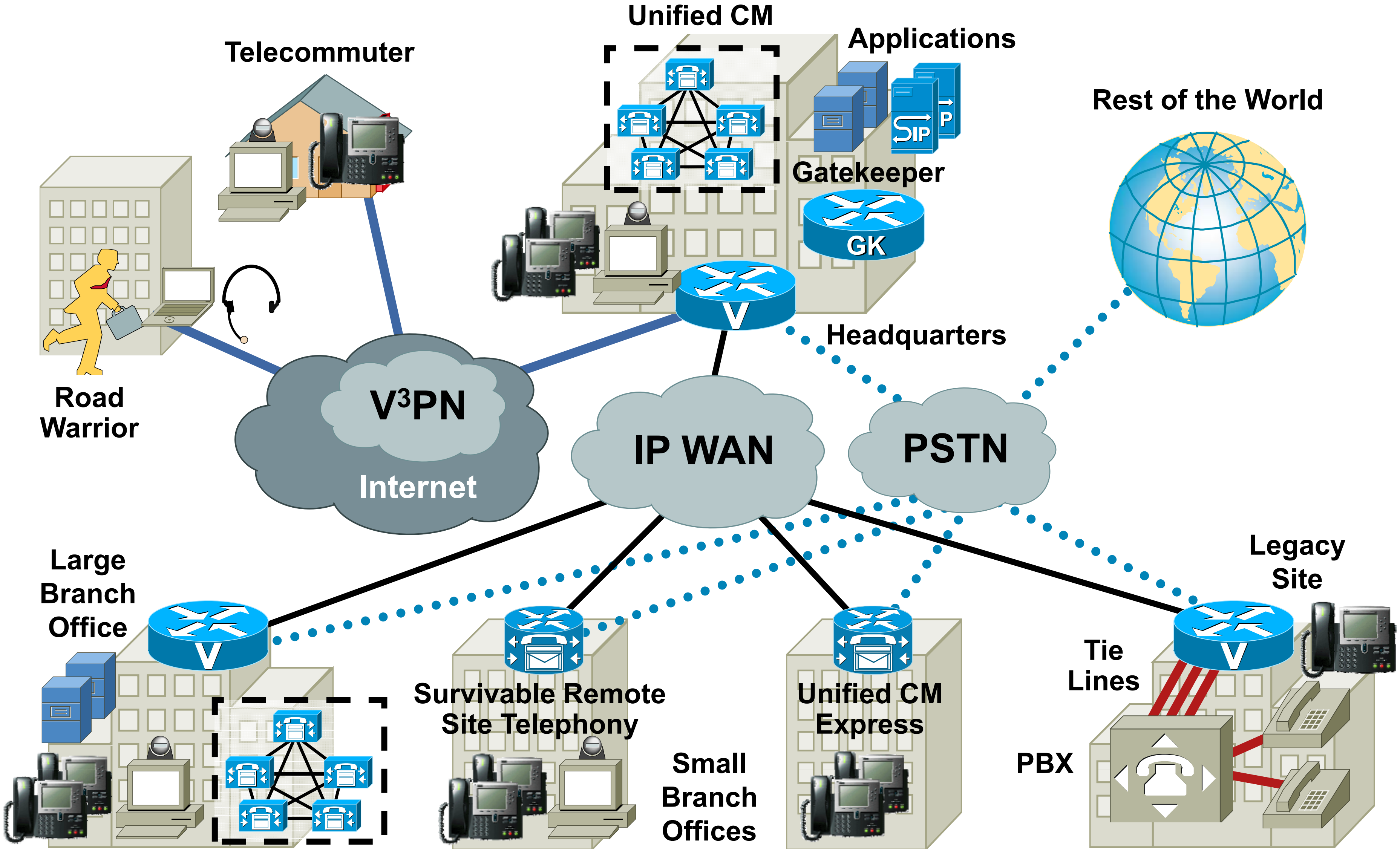


- Understanding what can be built today
- Learning how to build it
- To find out more about Unified Communications design:

  http://www.cisco.com/go/srnd/

Note : Unified CM = Cisco Unified Communications Manager

The Big Picture: End-to-End Unified Communications



The Elements of Unified Communications

UC Applications

UC Infrastructure

Network Infrastructure

Agenda

Introduction

Network Infrastructure

Unified Communications Infrastructure

Unified Communications Applications

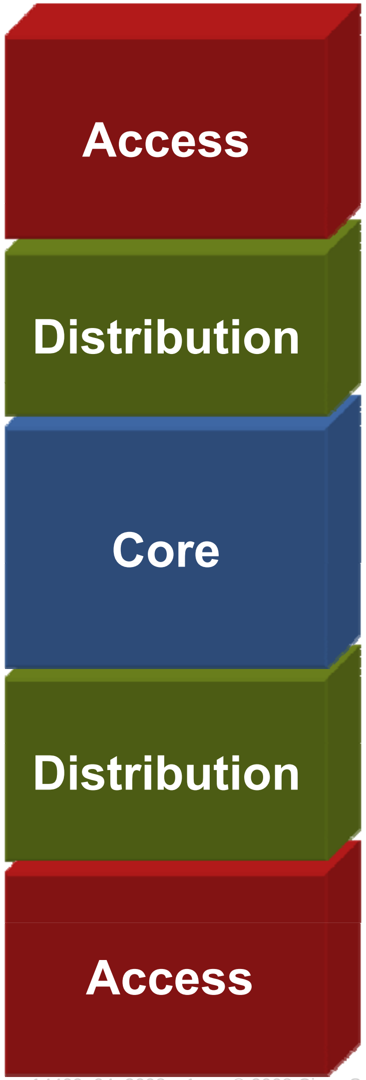
Security

Network Infrastructure Agenda

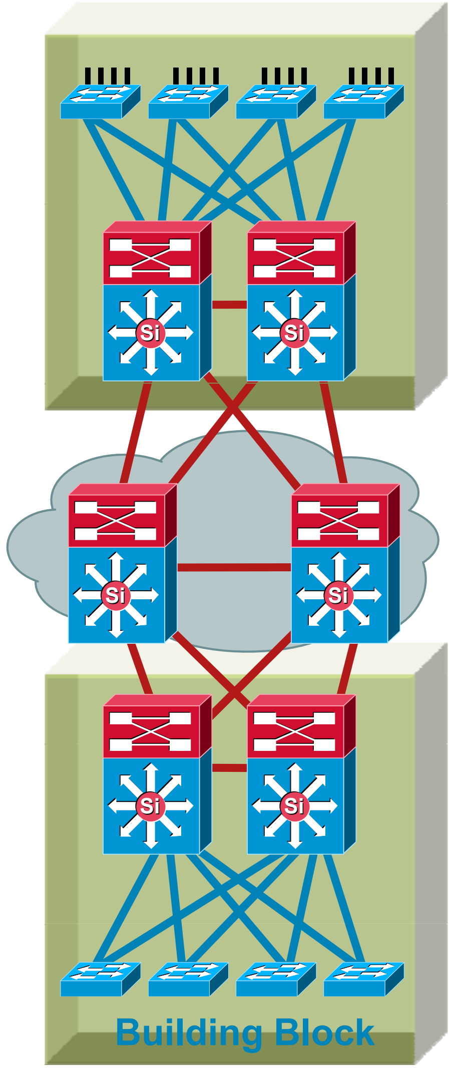
- Building a Campus Network for Unified Communications
- Enabling QoS in the Campus
- Enabling QoS in the WAN
- Overlaying Wireless LANs

Campus UC Networks: The Basics Still Apply… Hierarchical Network Design

Without a Rock Solid Foundation the Rest Doesn't Matter

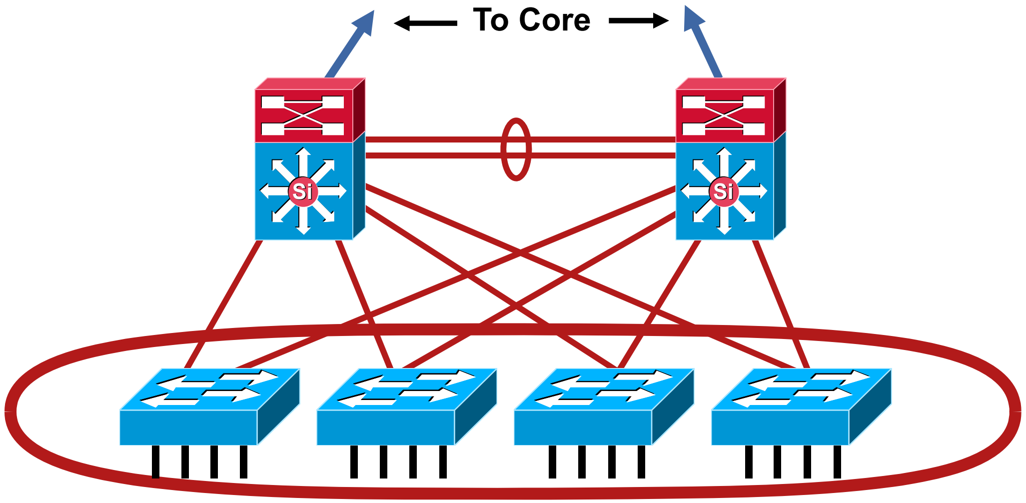


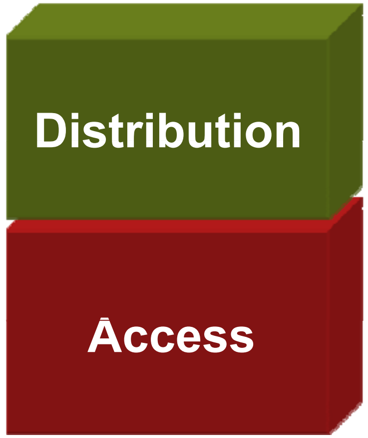
- Access/Distribution/Core hierarchy—each layer has specific role
- Modular scalable topology—building blocks
- Easy to grow, understand, and troubleshoot
- Creates small fault domains—clear demarcations and isolation
- Promotes load balancing and redundancy
- Promotes deterministic traffic patterns
- Incorporates balance of both Layer 2 and Layer 3 technology, leveraging the strength of both
- Utilizes Layer 3 Routing for load balancing, fast convergence, scalability, and control
- Sub-second convergence possible



Campus UC Networks: The Access Layer

UC Feature Rich Environment—Not Just About Connectivity





VLANS Do Not Span Access Switches

- The Access Layer provides aggregation for Voice, Video and Data endpoints
- Can provide switched or routed access—is typically feature rich…

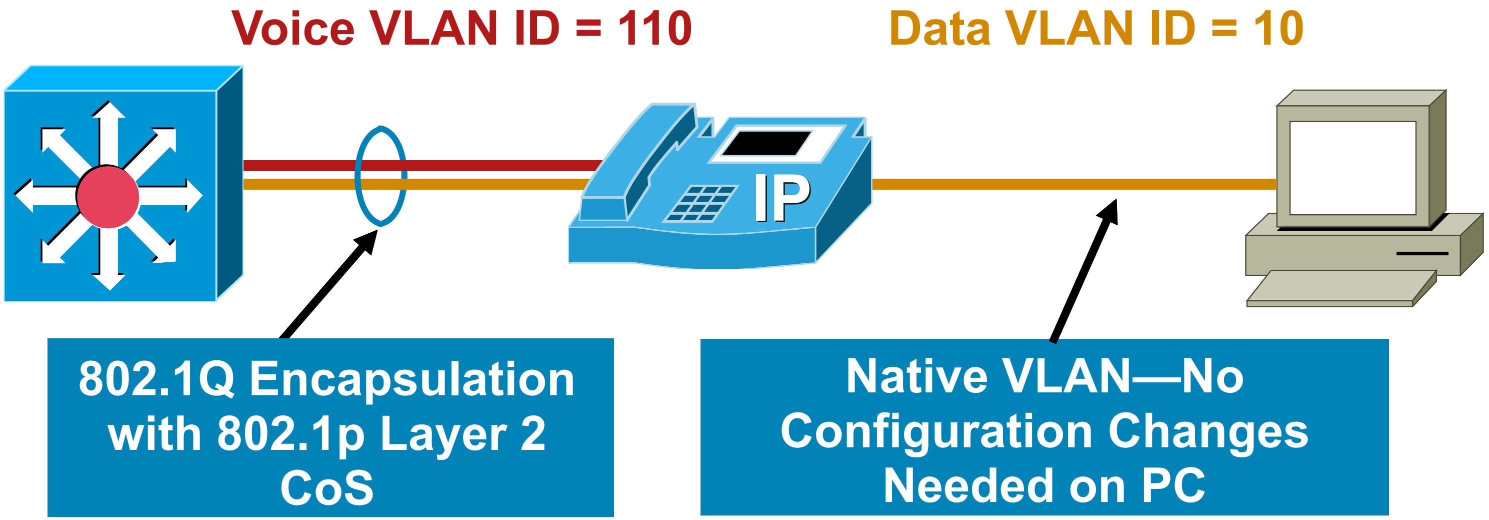
Key Features for Unified Communications

Automatic Phone Discovery
Power over Ethernet
Voice VLAN Allocation
Multiple Security Features

QoS Trust Boundaries
AutoQoS
Queuing
Network Access Control

UC Campus Networks: The Access Layer

Voice and Data VLANs

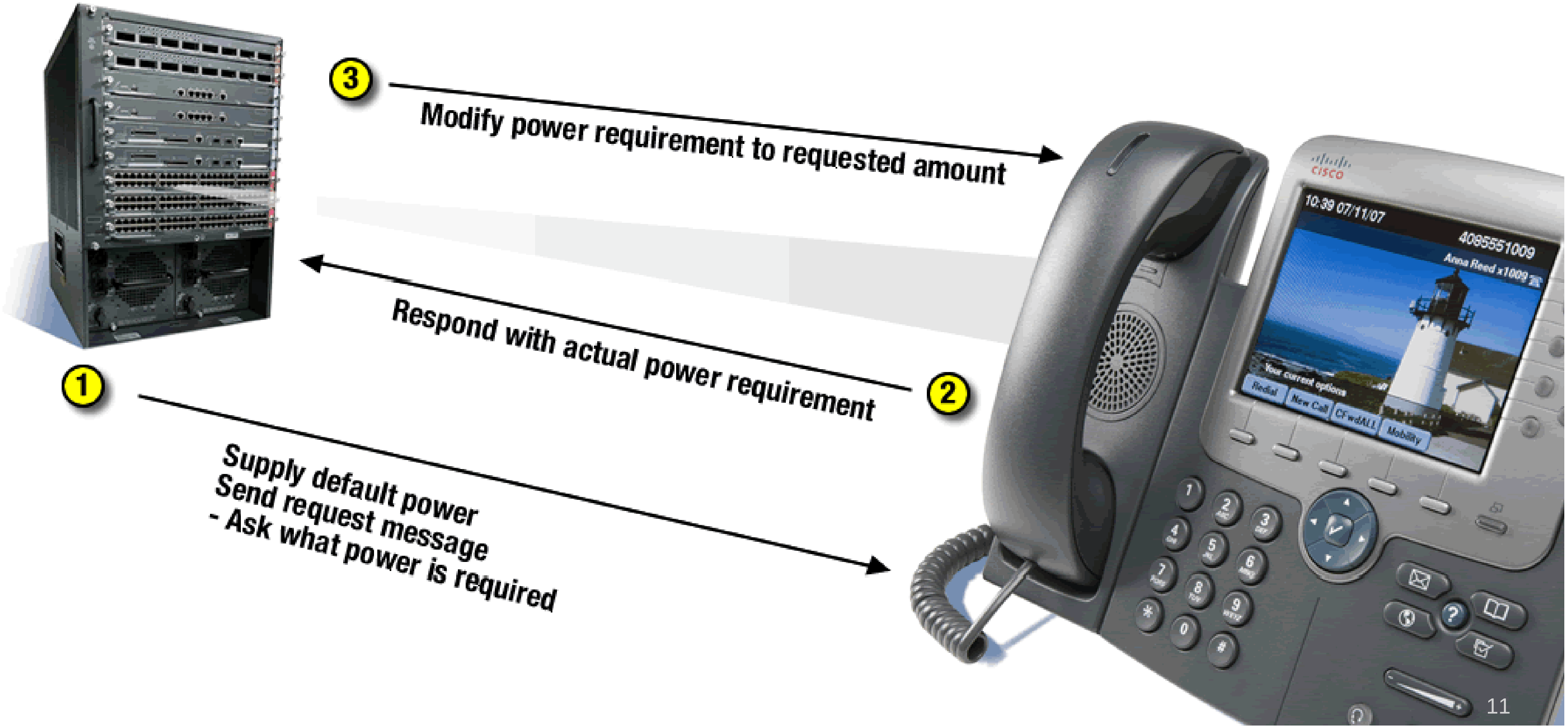


- Separate Voice and Data VLANs create partitioned broadcast domains in separate IP subnets
- Cisco Discovery Protocol (CDP) used during Phone boot up to configure Voice VLAN ID
- Phone also supplied with QoS configuration information
- For Security—different network policies can be applied for different subnets; e.g. WORM attacks can be contained to the Data VLANs

Campus UC Networks

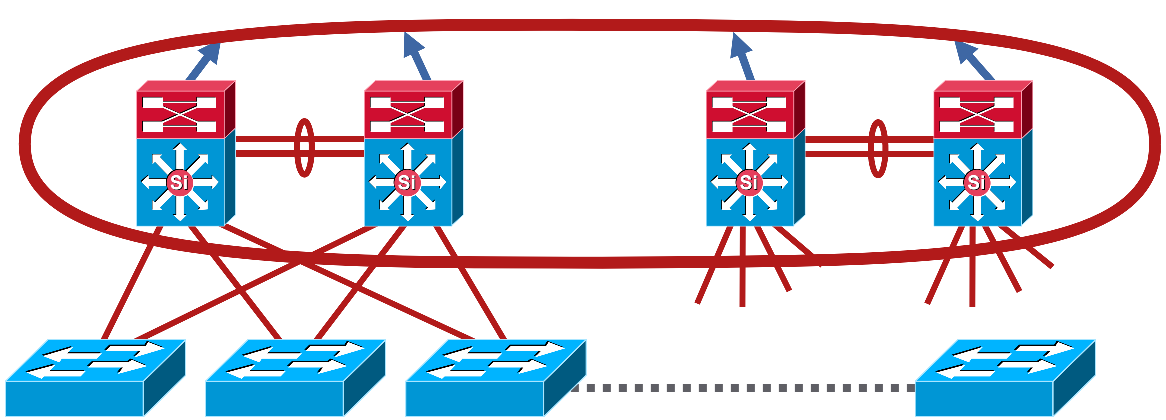
CDP and Inline Power Discovery

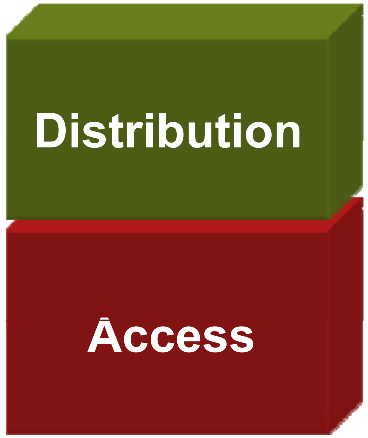
- Cisco Discovery Protocol—allows the switch to discover the attached inline powered device and negotiate the power requirements to optimize power consumption in the switch…



Campus UC Networks: The Distribution Layer

Fast Convergence, QoS, and High Availability

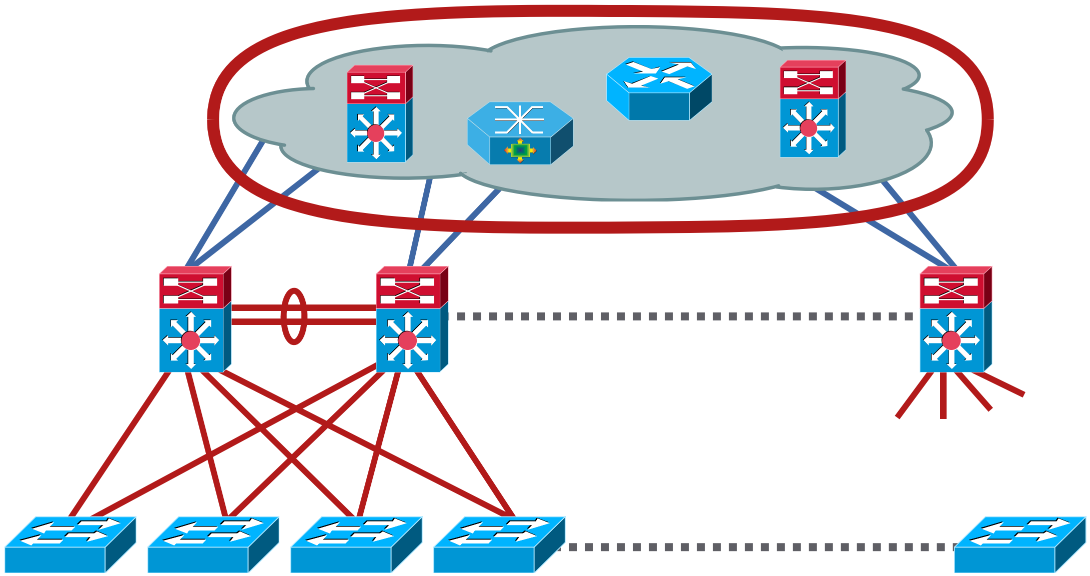


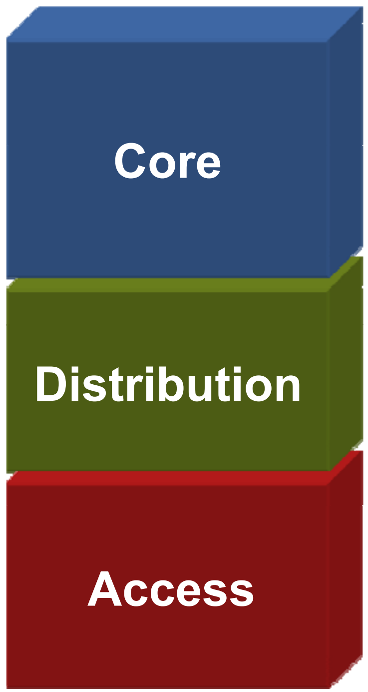


- Important considerations for Unified Communications in the Distribution Layer: Sub-Second Convergence, High Availability, Load Balancing, and QoS
- The Distribution Layer uses Layer 3 switching and aggregates wiring closet links (access layer) and uplinks to the core with route summarization
- Protects the core from high density peering and problems in the access layer
- EIGRP/OSPF—sub-second convergence possible with timer adjustment, redundant path load sharing, route summarization,
- HSRP or GLBP to provide first hop redundancy, sub-second convergence possible with timer adjustment

Campus UC Networks: The Core Layer

Scalability, High Availability, and Fast Convergence





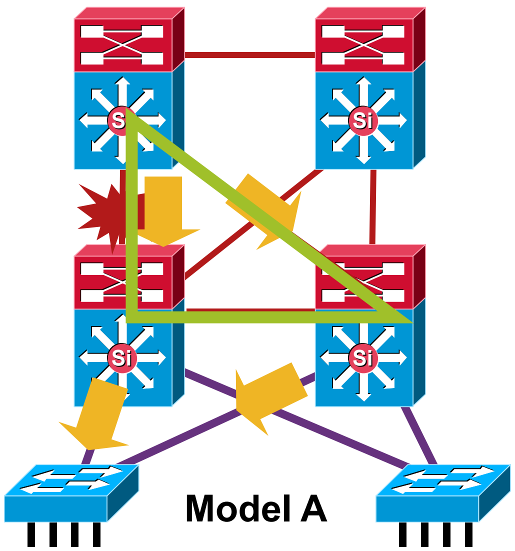
- Backbone for the network—connects network building blocks
- Performance and stability vs. complexity—less is more in the core
- Aggregation point for the distribution layer
- Tune routing protocol timers for sub second convergence
- Separate core layer helps in scalability during future growth
- Use hardware accelerated services only to maintain performance

UC Campus Network Design: Best Practice

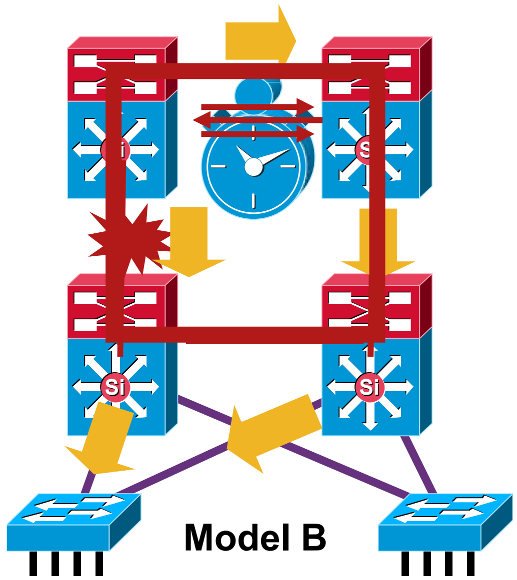
Build Triangles Not Squares

Deterministic vs. Non-Deterministic

Triangles: Link/Box Failure Does Not Require Routing Protocol Convergence



Squares: Link/Box Failure Requires Routing Protocol Convergence



- Layer 3 redundant equal cost links support fast convergence
- Hardware based—fast recovery to remaining path
- Convergence is extremely fast (dual equal-cost paths: no need for OSPF or EIGRP to recalculate a new path)

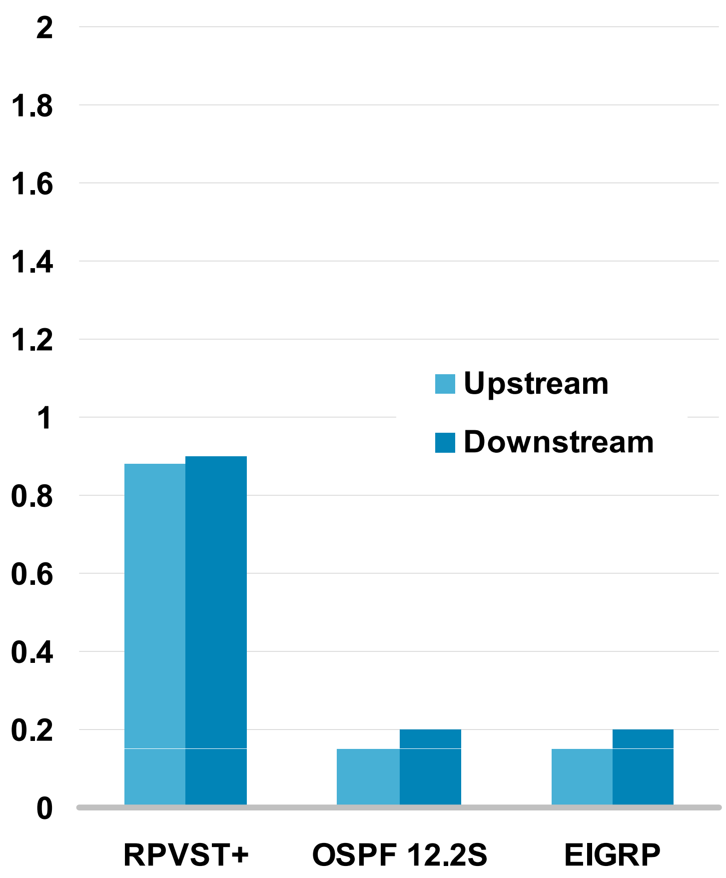
UC Campus Network Design: Routing to the Edge?

Provides Real Advantages for UC Traffic

Both L2 and L3 Can Provide Sub-Second Convergence

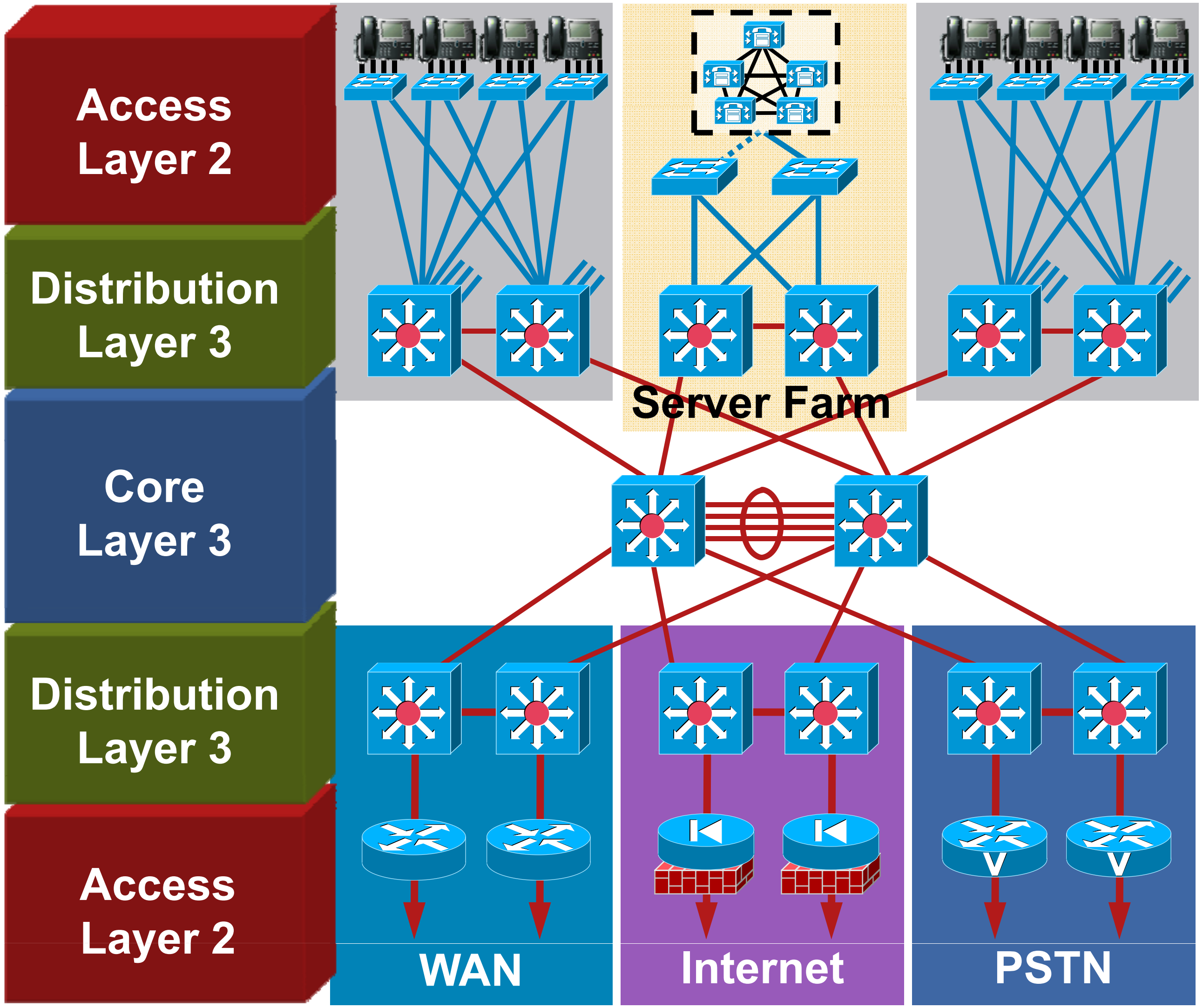
L3 Provides Sub-200 msec Convergence—Highly Desirable for UC

- Easier implement, less to get right
  - No matching of STP/HSRP/GLBP priority
  - No L2/L3 Multicast topology inconsistencies
- Well known tool set
  - traceroute, show ip route, show ip eigrp neighbor, etc.
- Most Cisco Catalyst® switches support L3 Switching
- EIGRP converges in <200 msec
- OSPF with sub-second tuning converges in <200 msec
- RPVST+ convergence times dependent on GLBP/ HSRP tuning
- New features such as VSS also offer great improvements in redundancy and convergence



Building a Campus UC Network
Summary

- Access layer
    - Automatic Phone Discovery
    - Power over Ethernet
    - Voice VLAN allocation
    - Multiple Security features
    - QoS Trust Boundaries
    - AutoQoS
    - Queuing
    - Network Access Control
    - Layer 3 to the edge?
- Distribution Layer
    - Fast Convergence
    - QoS
    - High Availability
- Core Layer
    - Fast Convergence
    - Scalability
    - High Availability



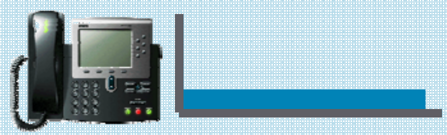
http://www.cisco.com/en/US/netsol/ns656/networking_solutions_design_guidances_list.html#anchor2

Network Infrastructure Agenda

- Building a Campus Network for Unified Communications
- Enabling QoS in the Campus
- Enabling QoS in the WAN
- Overlaying Wireless LANs

QoS in the Campus
Traffic Profiles and Requirements

Voice



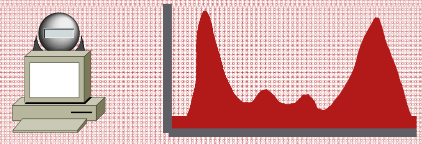
- Smooth
- Benign
- Drop sensitive
- Delay sensitive
- UDP priority

Bandwidth per Call Depends on Codec, Sampling-Rate, and Layer 2 Media

- Latency ≤ 150 ms
- Jitter ≤ 30 ms
- Loss ≤ 1%

One-Way Requirements

Video-Conf



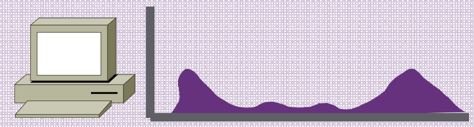
- Bursty
- Greedy
- Drop sensitive
- Delay sensitive
- UDP priority

IP/VC Has the Same Requirements as VoIP, But Has Radically Different Traffic Patterns (BW Varies Greatly)

- Latency ≤ 150 ms
- Jitter ≤ 30 ms
- Loss ≤ 1%

One-Way Requirements

Data



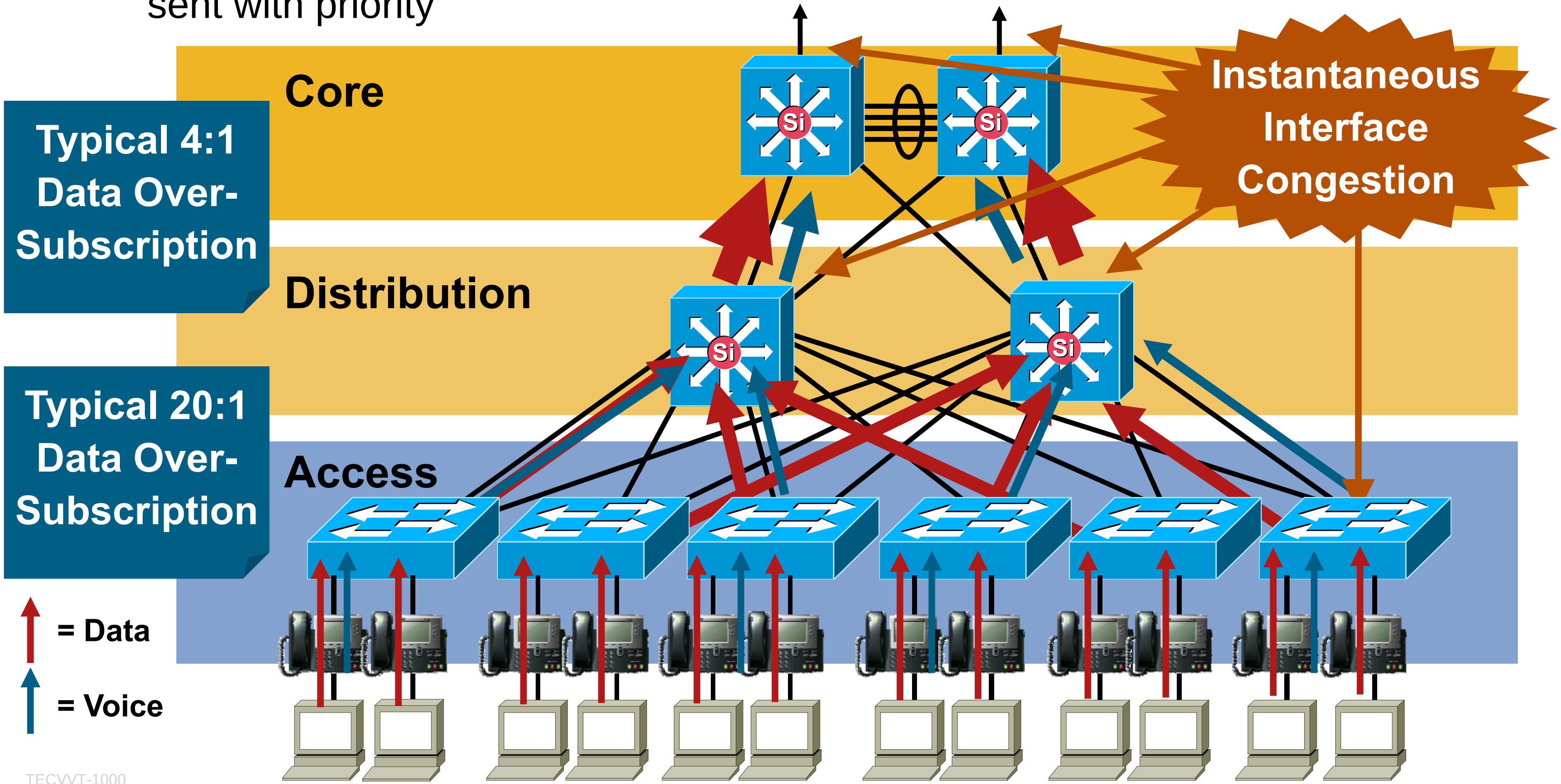
- Smooth/bursty
- Benign/greedy
- Drop insensitive
- Delay insensitive
- TCP retransmits

Traffic Patterns for Data Vary Among Applications

Data Classes:
Mission-Critical Apps
Transactional/Interactive Apps
Bulk Data Apps
Best Effort Apps (Default)

Why Enable QoS in the Campus?

- Adding more bandwidth to avoid congestion doesn't really help as the key issue is buffer size… QoS allows drop and delay sensitive traffic to be sent with priority



Enabling QoS in the Campus

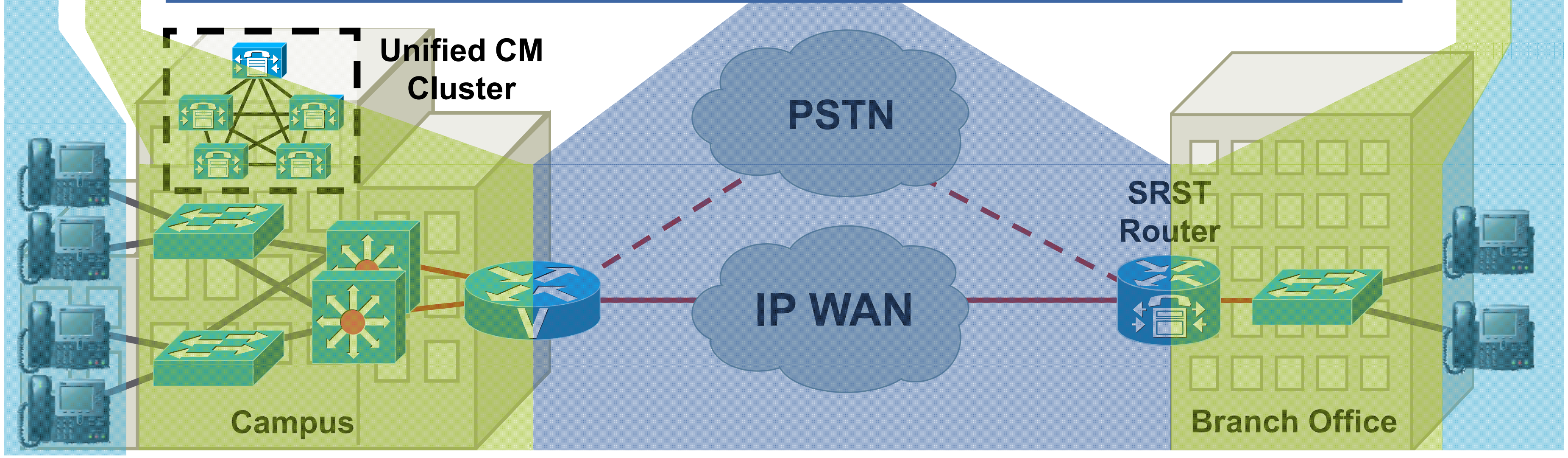
Cisco's Approach to QoS

Classification: Mark the Packets with a Specific Priority Denoting a Requirement for Class of Service from the Network

Trust Boundary: Define and Enforce a Trust Boundary at the Network Edge

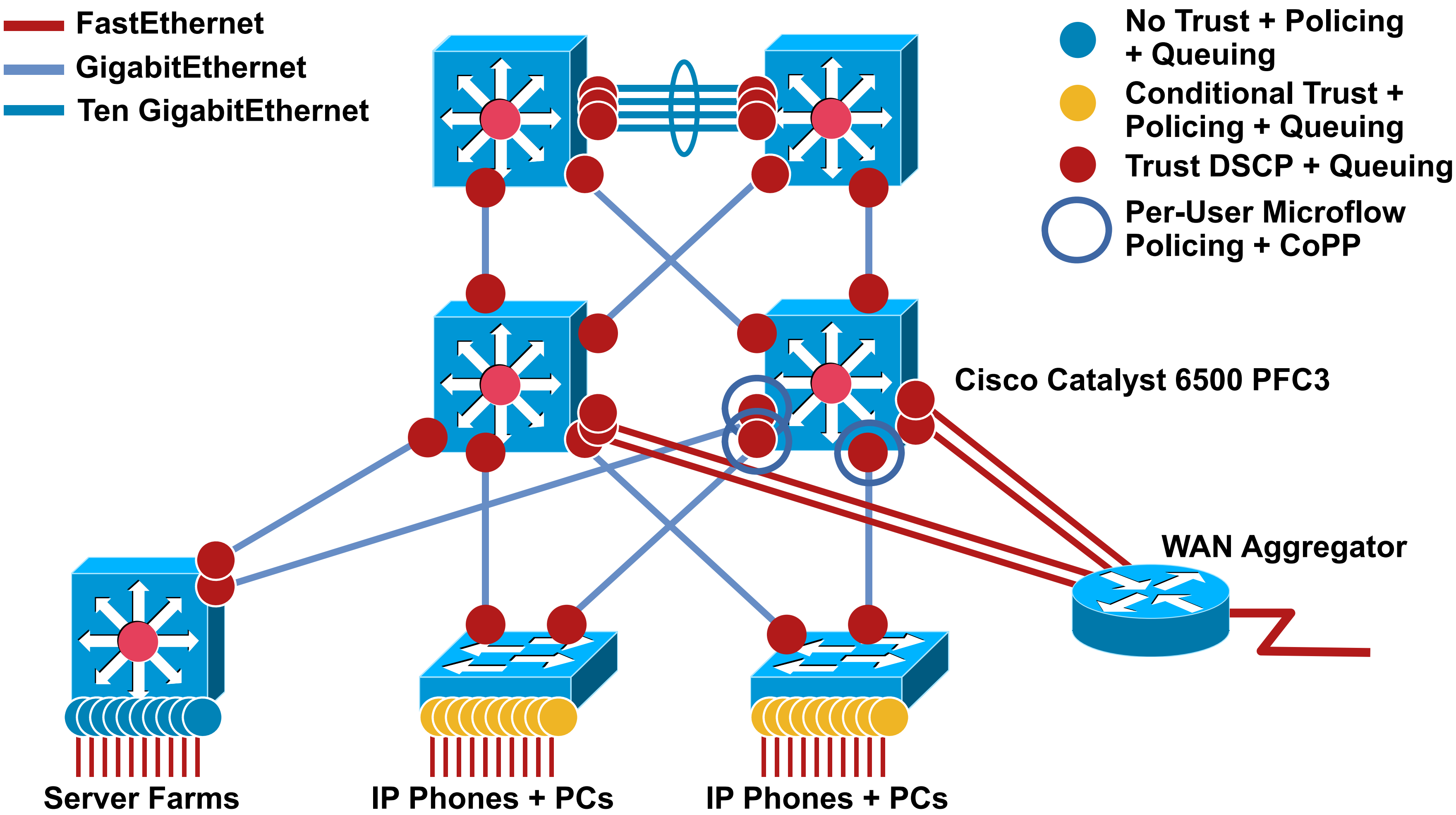
Scheduling: Assign Packets to One of Multiple Queues (Based on Classification) for Expedited Treatment through the Network

Provisioning: Accurately Calculate the Required Bandwidth for All Applications Plus Element Overhead



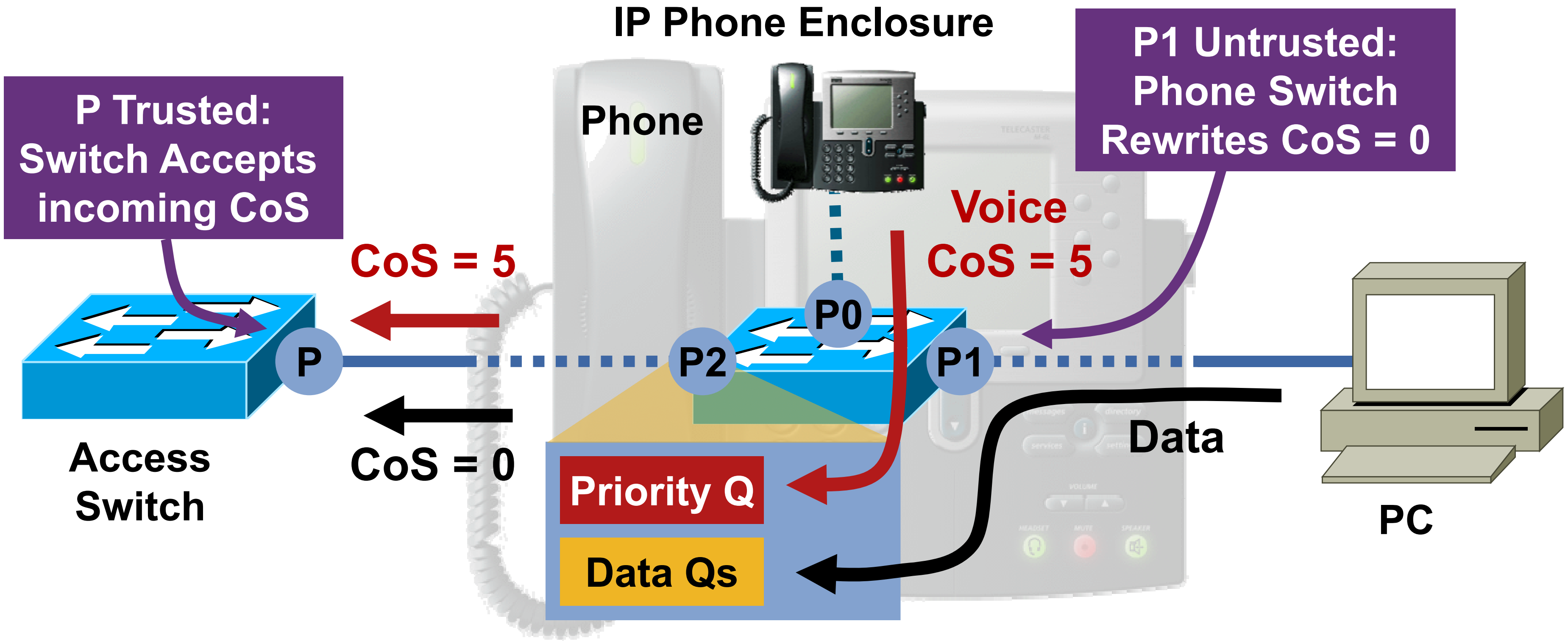
Campus QoS Considerations

Where Is QoS Required Within the Campus?



QoS in the Campus

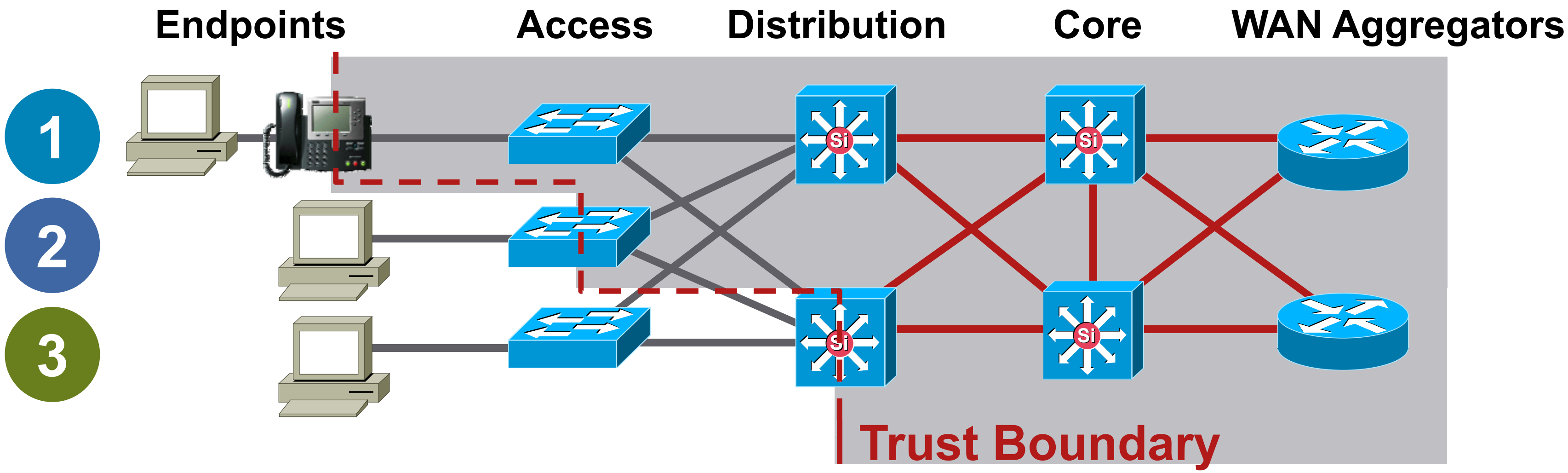
Traffic Queuing and Scheduling in IP Phones



- Voice media traffic is marked with CoS 5/DSCP EF (high priority)
- Data traffic from the PC is remarked with CoS 0 (low priority) by the IP phone switch

Campus QoS Considerations

Establishing Trust Boundaries



- For scalability, classification should be done as close to the edge as possible
- The outermost trusted devices represent the trust boundary

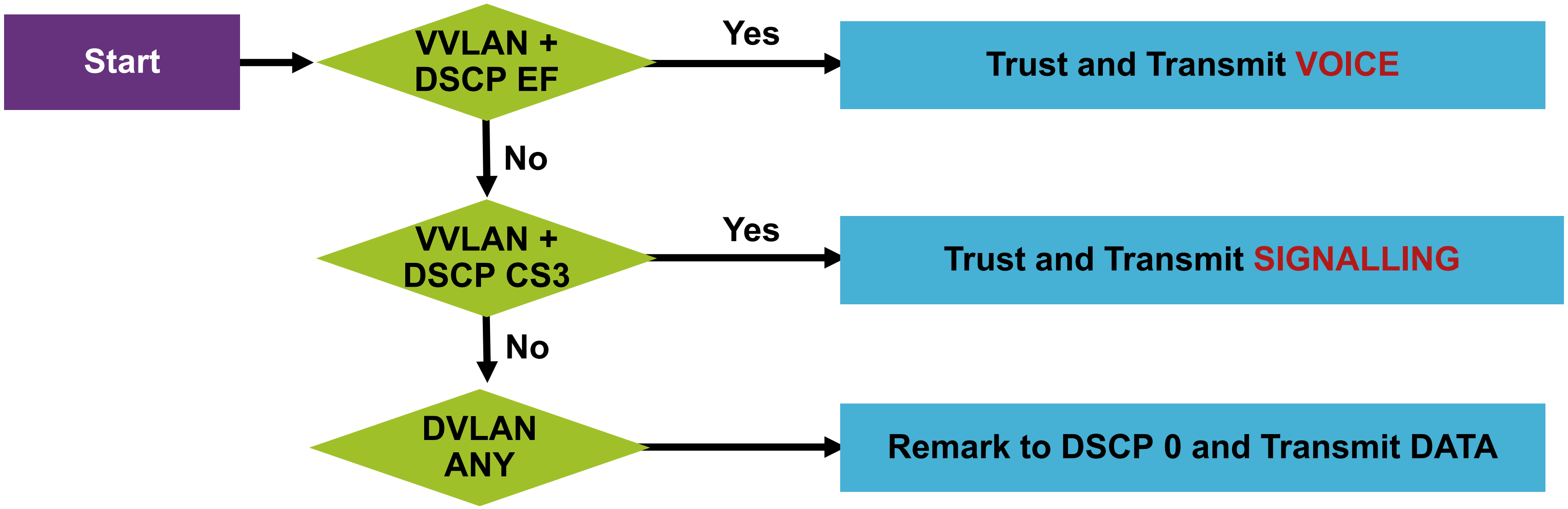
1. **Optimal Trust Boundary: Trusted Endpoint**
   A device is trusted if it correctly classifies packets
2. **Optimal Trust Boundary: Untrusted Endpoint**
3. **Suboptimal Trust Boundary**
   Only use if access switch cannot perform classification

QoS in the Campus: Switch Port AutoQoS

- AutoQoS allows the application of a pre-defined set of QoS statements to an interface with one CLI command…

AutoQoS in Access Layer Switch Ports

AutoQoS Example: VoIP Model



- Voice Traffic identified by QoS (DSCP) value, queued and transmitted
- Signalling Traffic identified by QoS (DSCP) value, queued and transmitted
- All Data Traffic has DSCP value re-marked to 0 then transmitted
- Bespoke AutoQoS Macros can be configured and applied to ports, e.g. AutoQoS for Voice, Video, Business Data and Best Effort Data

AutoQoS in the Campus

AutoQoS Macro Example

For campus Cisco Catalyst switches, AutoQoS command macro enables the following QoS features automatically:

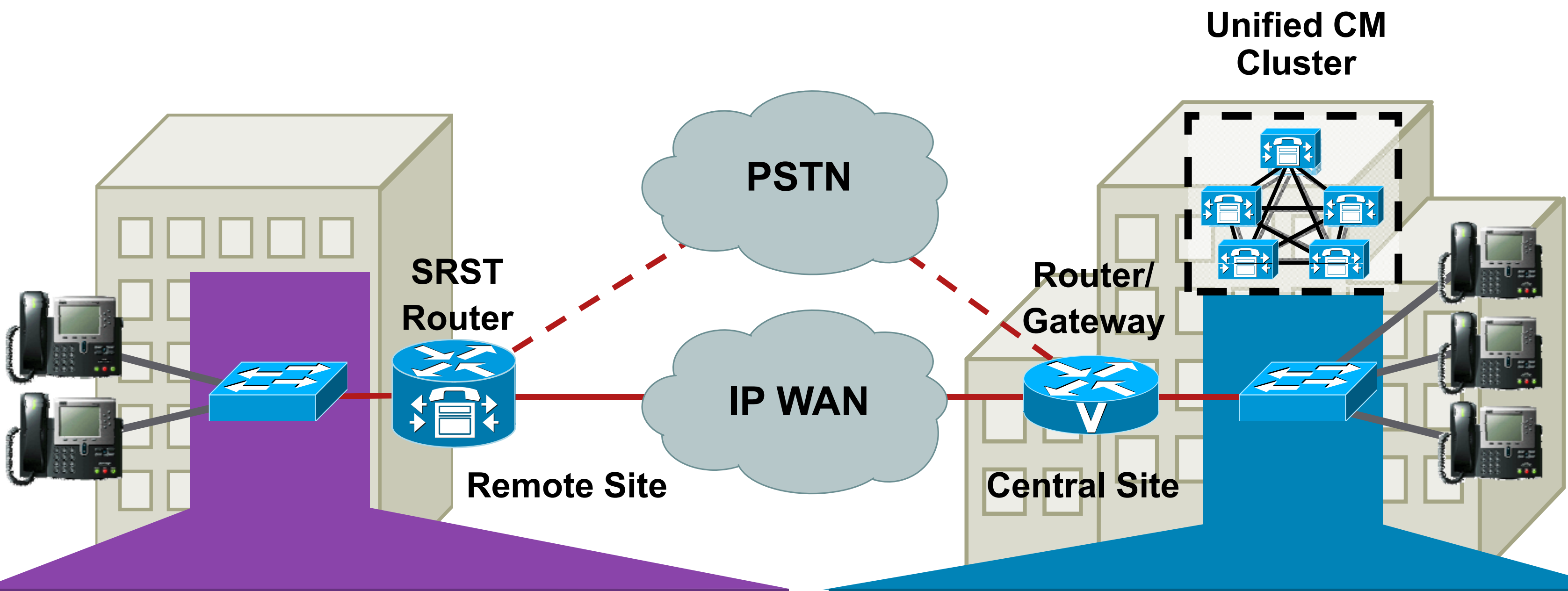
- Enforces a trust boundary at Cisco IP Phones
- Enforces a trust boundary on Cisco Catalyst switch access ports and uplinks/downlinks
- Enables Cisco Catalyst strict priority queuing for voice and weighted round robin queuing for data traffic
- Modifies queue admission criteria (i.e. CoS-to-queue mapping)
- Modifies queue sizes, as well as queue weights where required
- Modifies CoS-to-DSCP and IP precedence to-DSCP mappings

Catalyst (config-if) # auto qos voip cisco-phone

```
mls qos map cos-dscp 0 8 16 26 32 46 48 56
mls qos srr-queue output cos-map queue 1 threshold 3  5
mls qos srr-queue output cos-map queue 2 threshold 3  3 6 7
mls qos srr-queue output cos-map queue 3 threshold 3  2 4
mls qos srr-queue output cos-map queue 4 threshold 2  1
mls qos srr-queue output cos-map queue 4 threshold 3  0
mls qos srr-queue output dscp-map queue 1 threshold 3  40 41 42 43 44 45 46 47
mls qos srr-queue output dscp-map queue 2 threshold 3  24 25 26 27 28 29 30 31
mls qos srr-queue output dscp-map queue 2 threshold 3  48 49 50 51 52 53 54 55
mls qos srr-queue output dscp-map queue 2 threshold 3  56 57 58 59 60 61 62 63
mls qos srr-queue output dscp-map queue 3 threshold 3  16 17 18 19 20 21 22 23
mls qos srr-queue output dscp-map queue 3 threshold 3  32 33 34 35 36 37 38 39
mls qos srr-queue output dscp-map queue 4 threshold 1  8
mls qos srr-queue output dscp-map queue 4 threshold 2  9 10 11 12 13 14 15
mls qos srr-queue output dscp-map queue 4 threshold 3  0 1 2 3 4 5 6 7
mls qos queue-set output 1 threshold 1 138 138 92 138
mls qos queue-set output 1 threshold 2 138 138 92 400
mls qos queue-set output 1 threshold 3 36 77 100 318
mls qos queue-set output 1 threshold 4 20 50 67 400
mls qos queue-set output 2 threshold 1 149 149 100 149
mls qos queue-set output 2 threshold 2 118 118 100 235
mls qos queue-set output 2 threshold 3 41 68 100 272
mls qos queue-set output 2 threshold 4 42 72 100 242
mls qos queue-set output 1 buffers 10 10 26 54
mls qos queue-set output 2 buffers 16 6 17 61
mls qos
!
interface GigabitEthernet0/1
 srr-queue bandwidth share 10 10 60 20
 srr-queue bandwidth shape  10  0  0  0
 queue-set 2
 mls qos trust device cisco-phone
 mls qos trust cos
 auto qos voip cisco-phone
```

Enabling QoS in the Campus

Platform Recommendations



- Cisco Catalyst 4x00
- Cisco Catalyst 3560, 3750
- Cisco Catalyst 2950, 2960
- Cisco IOS® Router SW NM

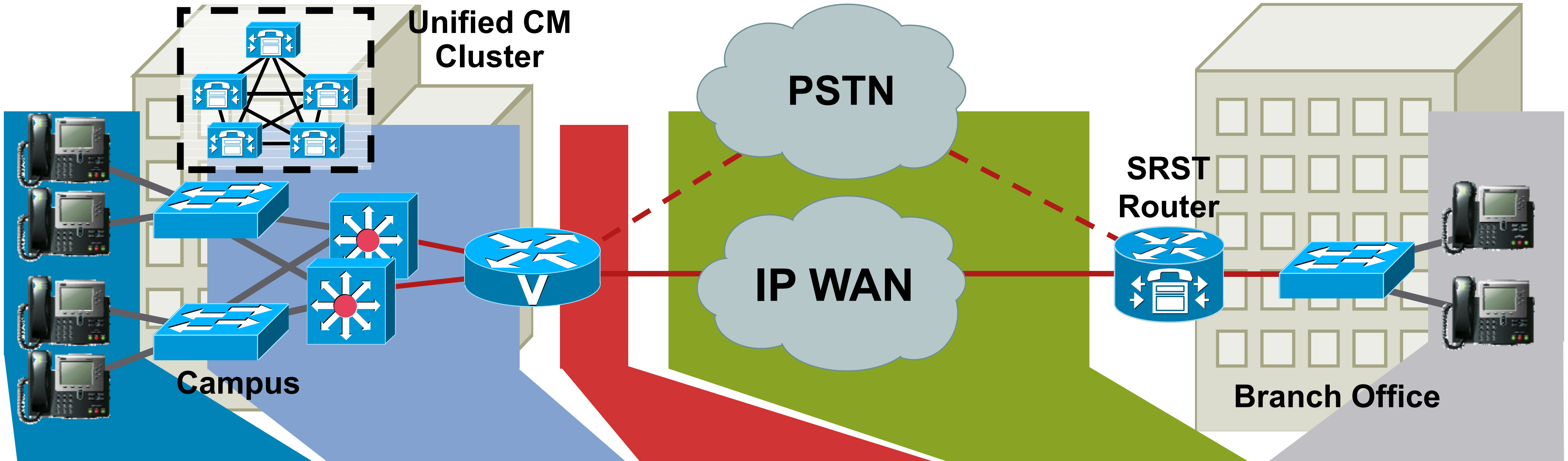
- Cisco Catalyst 6500
- Cisco Catalyst 4500
- Cisco Catalyst 4000

Network Infrastructure Agenda

- Building a Campus Network
- Enabling QoS in the Campus
- Enabling QoS in the WAN
- Overlaying Wireless LANs

Enabling QoS in the WAN

Elements That Affect End-to-End Delay



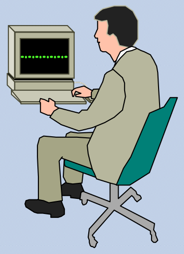
CODEC	Queuing	Serialization	Propagation and Network	Jitter Buffer
G.729A: 25 ms	Variable (Can Be Reduced Using Priority Queuing)	Variable (Can Be Reduced Using LFI)	6.3 µs/Km + Network Delay (Variable)	20–50 ms

End-to-End Delay (Aim for < 150 ms)

QoS in the WAN: QoS Considerations

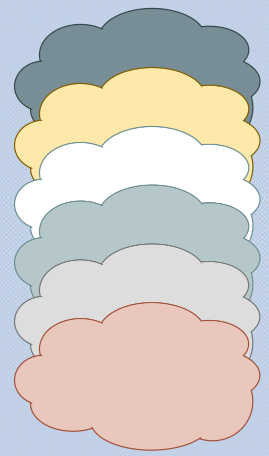
Best Effort vs. Guaranteed Quality

Guaranteed Voice Quality



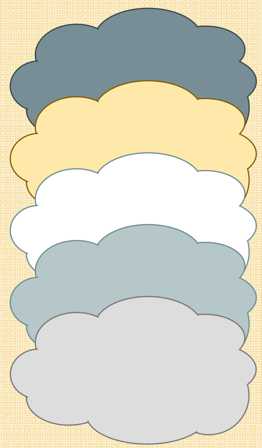
Call Agents

Business Critical Calls

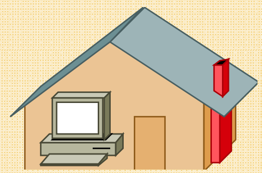


- Leased Lines
- Frame Relay
- ATM
- ATM/Frame Relay
- IP-SEC V³PN
- MPLS

Best Effort Voice Quality



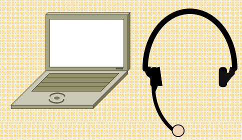
- DSL
- Cable
- Wireless
- Internet
- VPN



Telecommuters

Road Warriors

Intra Company Calls



Enabling QoS in the WAN

General Guidelines

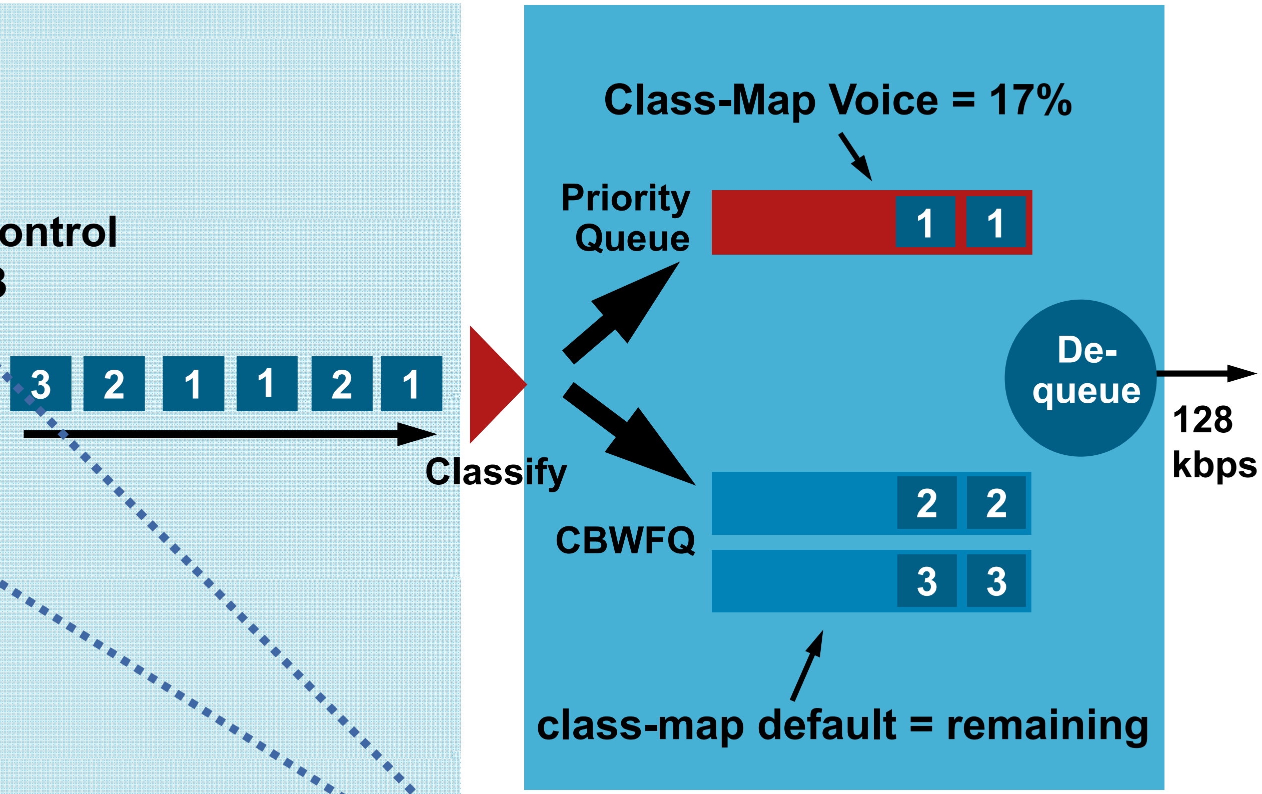
- Use LLQ anytime VoIP over the WAN is involved
- Traffic shaping is a requirement for Frame Relay/ ATM environments
- Use LFI techniques for all links below 768 Kbps
  - Don't use LFI for any video-over-IP applications
- TX-ring sizes may require modification
- Properly provision the WAN bandwidth
- Call admission control is a requirement where VoIP calls can over-subscribe the provisioned Bandwidth
- Use cRTP carefully
- Map QoS from L3 (IP precedence or DSCP) to L2 (802.1p) at remote branches if switch is L2 only

QoS in the WAN:

Prioritizing Voice Traffic

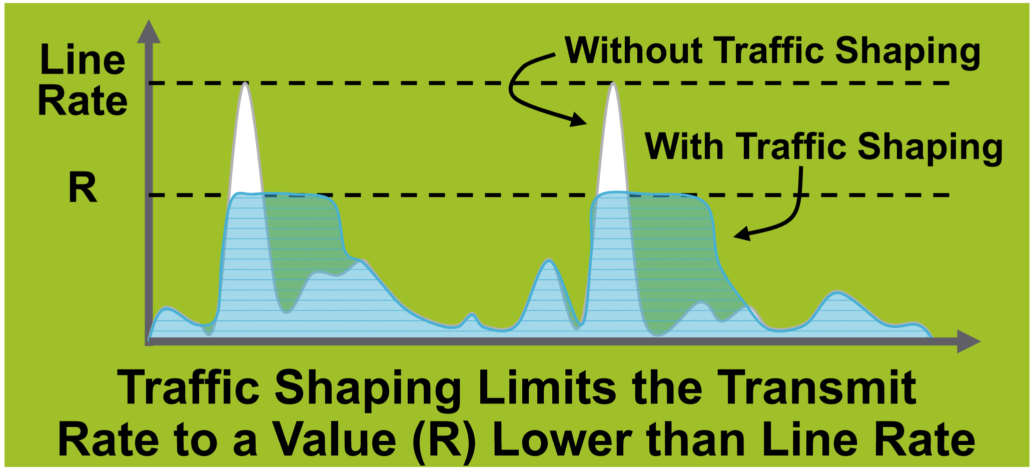
Low Latency Queuing Example

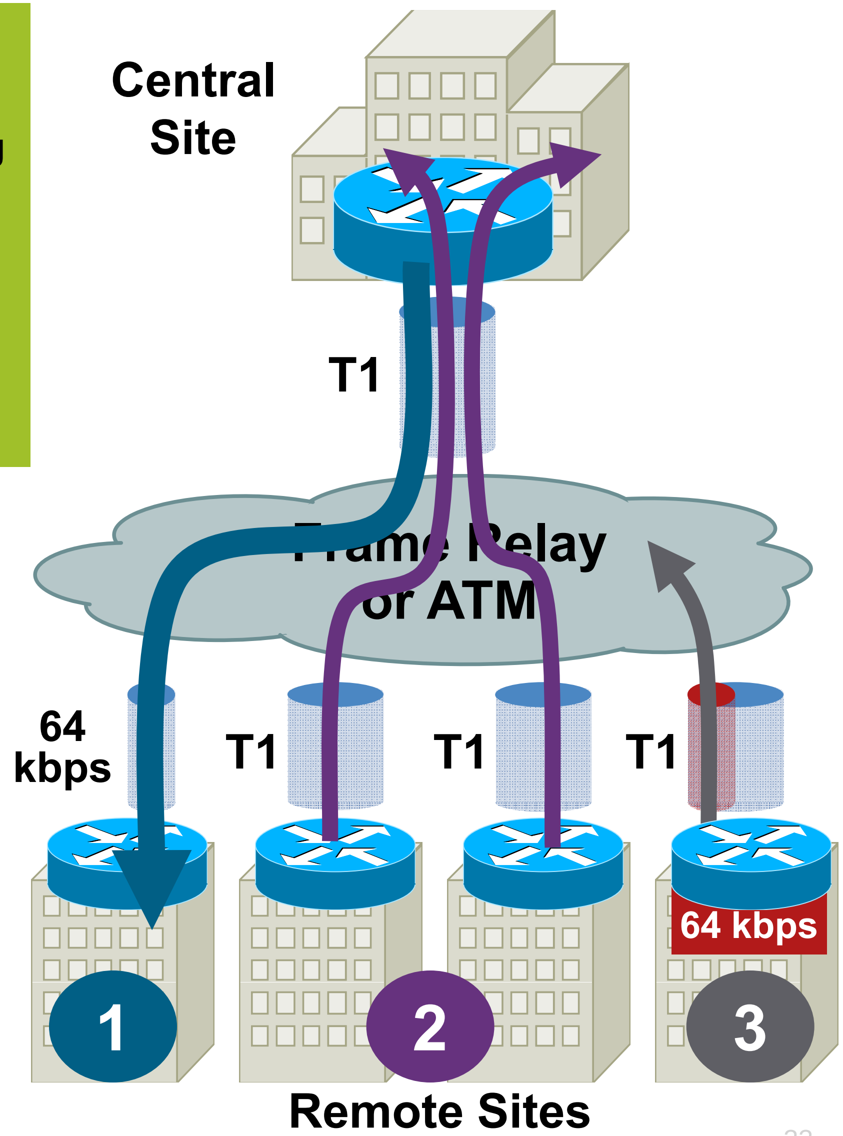
```
class-map class-default
  match any
class-map match-all voice
  match ip dscp ef
Class-map match-all voice-control
  match ip dscp af31 ; or CS3
!
policy-map WAN
 class voice
    priority percent 17
 class voice-control
    bandwidth percent 2
 class class-default
    fair-queue
!
interface Serial0/1
 ip address 10.1.6.2 255.255.255.0
 bandwidth 128
 no ip directed-broadcast
 service-policy output WAN
!
```



Any Packet with DSCP = 46 (PHB=EF) Gets Assigned to a Class that Will Get a High Priority Queue with 17% Bandwidth

QoS in the WAN: Traffic Shaping





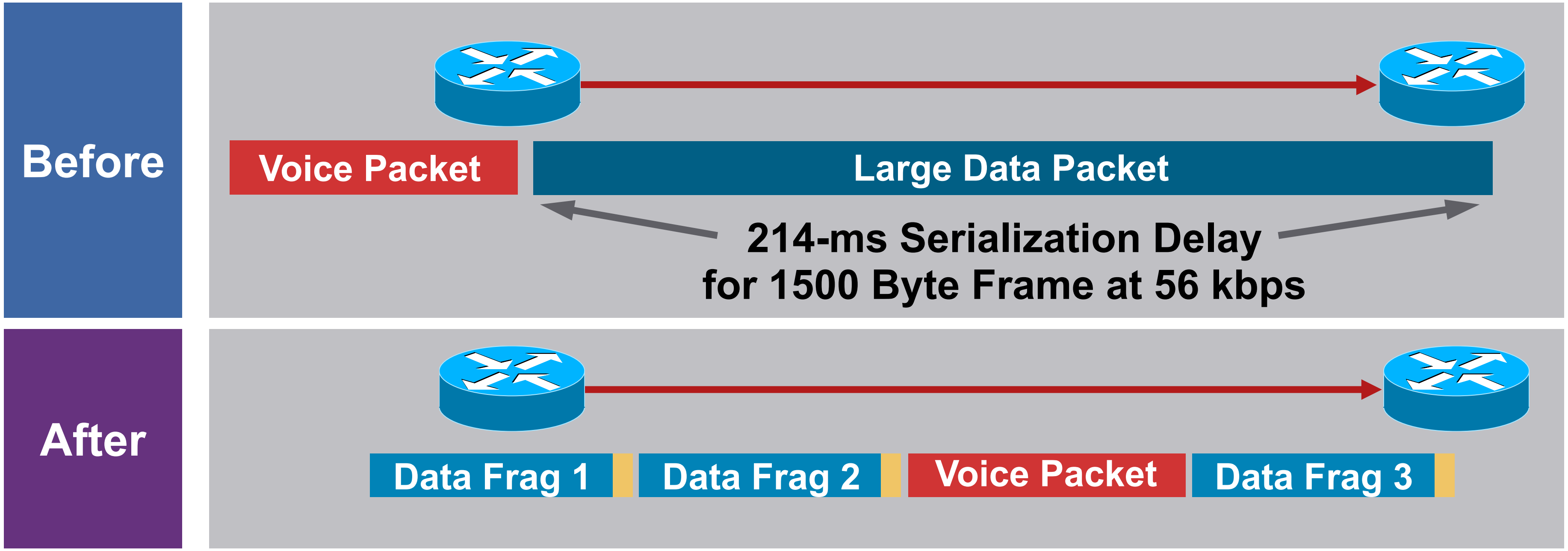
Why Is It Needed?

1. Line speed mismatch
2. Remote to central site oversubscription
3. To prevent bursting above Committed Rate (CIR)

QoS in the WAN

Reducing Serialization Delay for Voice Packets

Link Fragmentation and Interleaving (LFI)

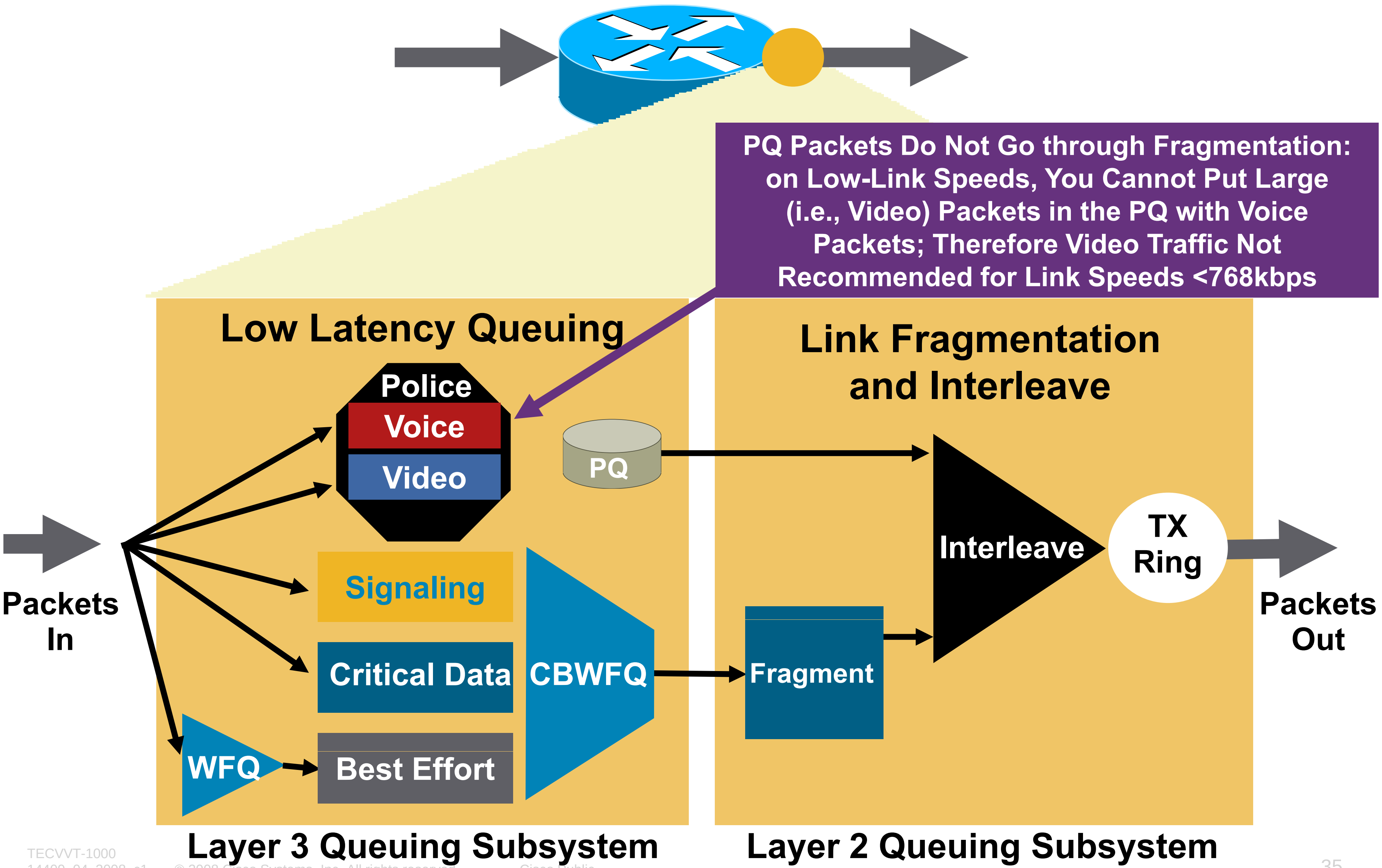


Link Type	LFI Mechanism
Pt-to-Pt Links:	MLPPP
Frame Relay:	FRF.12
ATM:	MLPPP over ATM
ATM/Frame-Relay SIW:	MLPPP over ATM and FR

Note *LFI is not required for link speeds greater than 768kbps

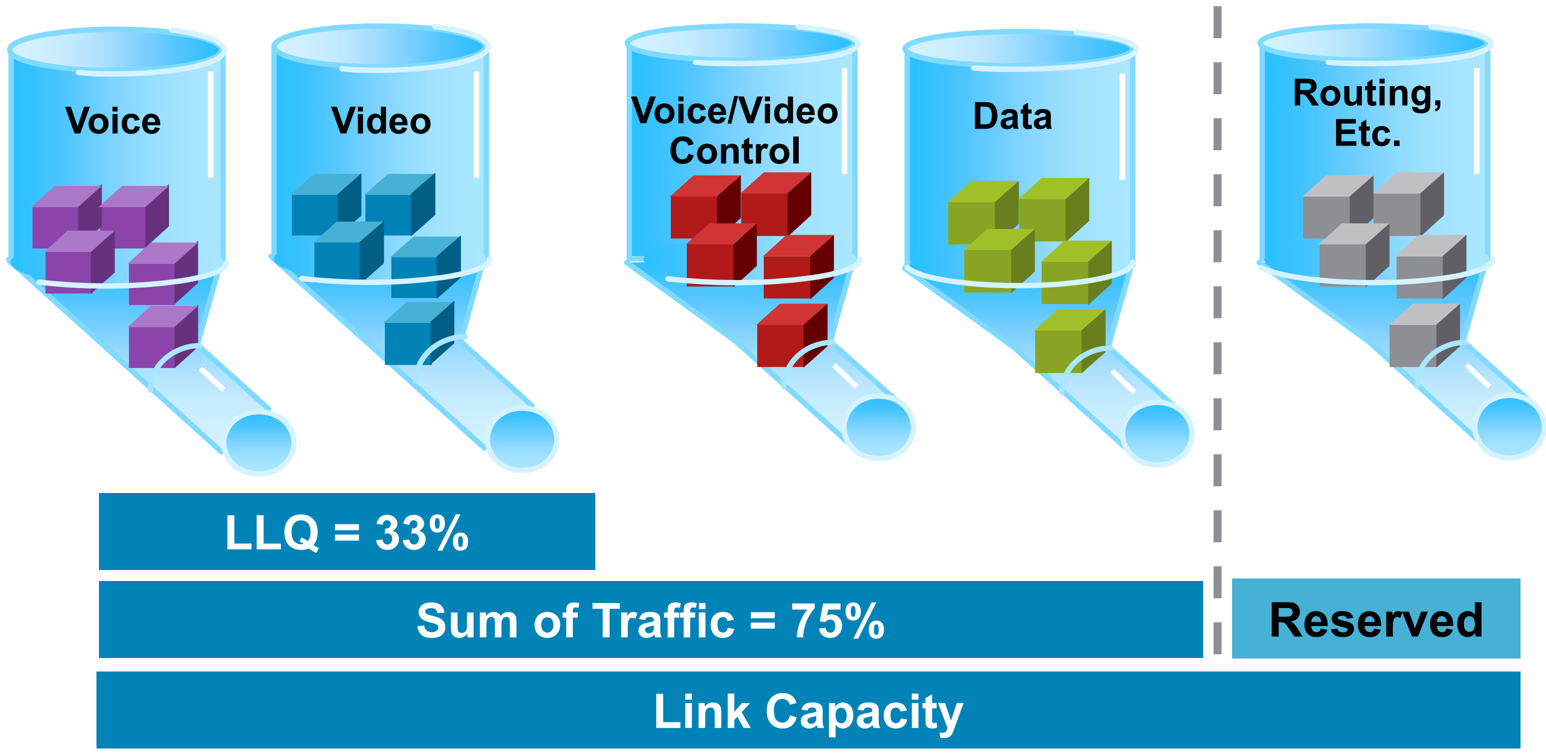
QoS in the WAN

Packet Scheduling, Fragmentation and Interleaving



QoS in the WAN
Bandwidth Provisioning

**Voice Is Not Free—Especially on Low-Speed Links—
Engineer the Network for Data, Voice, and Video**



Link Capacity = (Min BW for Voice + Min BW for Video + Min BW for Data)/0.75

QoS in the WAN

Provisioning Bandwidth for Voice-Bearer Traffic

CODEC	Sampling Rate	Voice Payload in Bytes	Packets per Second	Bandwidth per Conversion
G.711/G722-64k	20 msec	160	50	80 kbps
G.711/G722-64k	30 msec	240	33	74 kbps
G.729A	20 msec	20	50	24 kbps
G.729A	30 msec	30	33	18 kbps

A More Accurate Method for Provisioning Is to Include the Layer 2 Headers into the Bandwidth Calculations:

CODEC	Ethernet 14 Byes of Header	PPP 6 Bytes of Header	ATM 53 Bytes Cells with a 48-Byte Payload	Frame Relay 4 Bytes of Header
G.711/G722-64k at 50 pps	85.6 kbps	82.4 kbps	106 kbps	81.6 kbps
G.711/G722-64k at 33 pps	77.6 kbps	75.5 kbps	84 kbps	75 kbps
G.729A at 50 pps	29.6 kbps	26.4 kbps	42.4 kbps	25.6 kbps
G.729A at 33 pps	22.2 kbps	20 kbps	28 kbps	19.5 kbps

QoS in the WAN

Calculating Layer 2/3 Overhead for Video

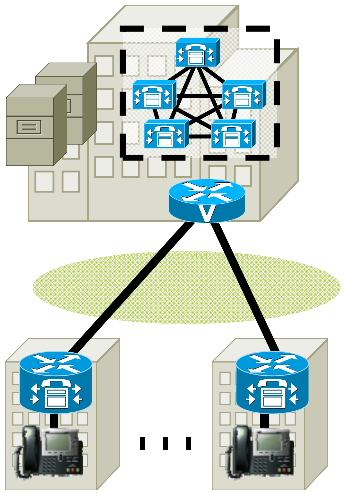
- Harder to calculate video bandwidth because payload size is variable (video is bursty!)
- General rule of thumb is to add 20% for all Layer 2/Layer 3 overhead
- Call bandwidth is typically the "maximum" transmission bandwidth of the call; average is usually much less

Video Data Rate and Bandwidth Required

128k = 153k
384k = 460k
512k = 614k
768k = 921k
1.5M = 1.8M

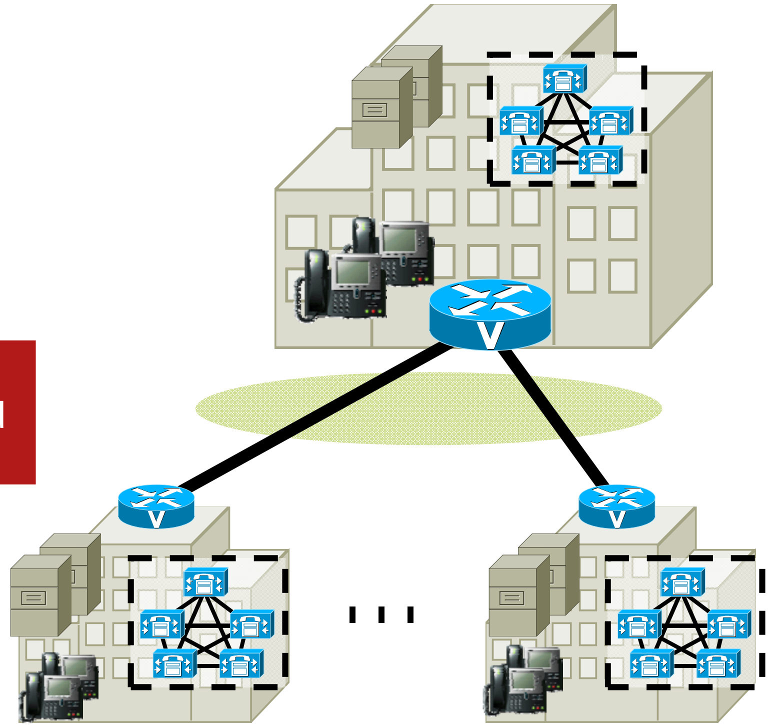
QoS in the WAN

Provisioning Bandwidth for Signaling Traffic



Please Refer to UC SRND 7.X for Encrypted Call Control Bandwidth

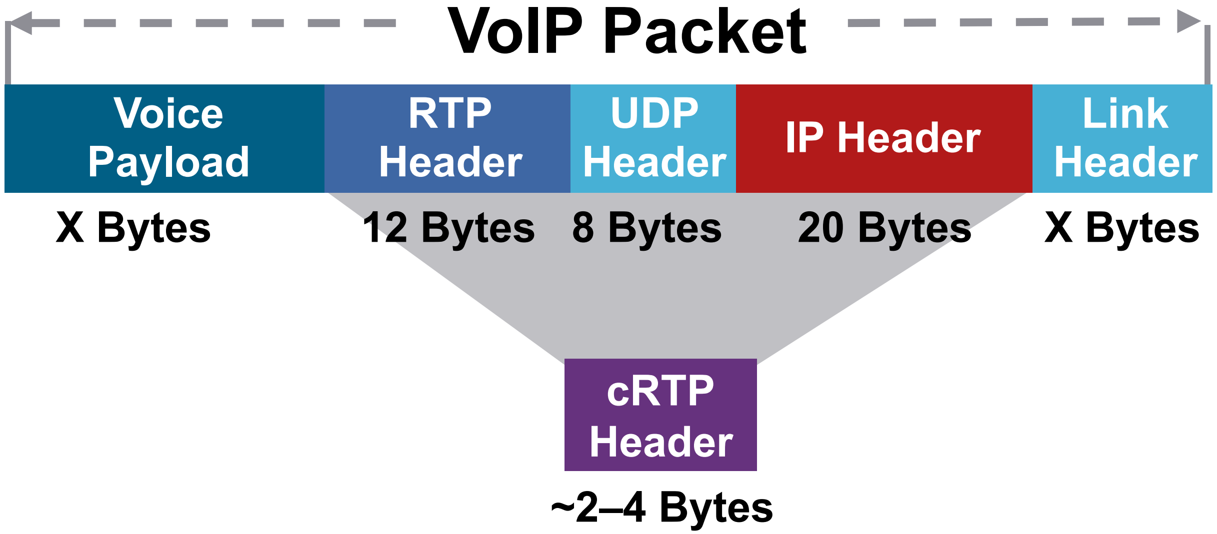
Centralized Call Processing (As per 7.X SRND, No Encryption)		
Number of IP Phones, Gateways	SCCP Control Bandwidth	SIP Control Bandwidth
1 to 30	8 kbps	8 kbps
50	14 kbps	27 kbps
100	27 kbps	54 kbps
150	40 kbps	81 kbps



Distributed Call Processing	
Number of Virtual Tie Lines	Bandwidth
1 to 70	8 kbps

QoS in the WAN

Provisioning Bandwidth with Compressed RTP (cRTP)



- Compresses RTP + UDP + IP headers (40 bytes) down to 2–4 bytes
- Enabled on point-to-point links—impacts router CPU

CODEC	PPP 6 Bytes of Header without CRTP	PPP 6 Bytes of Header with CRTP	Percent Bandwidth Reduction
G.711 at 50 pps	82.4 kbps	**68 kbps**	**17.5%**
G.711 at 33 pps	75.5 kbps	**66 kbps**	**12.5%**
G.729A at 50 pps	26.4 kbps	**12 kbps**	**54.5%**
G.729A at 33 pps	20 kbps	**10.5 kbps**	**47.5%**

QoS in the WAN
Cisco IOS AutoQoS for WAN Links

- Similar to AutoQoS in Cisco Catalyst switches
- Use AutoDiscovery to:

  Determine WAN traffic types and their offered bit rate

- Use AutoQoS to:

  Apply map classes to match on QoS values/traffic types

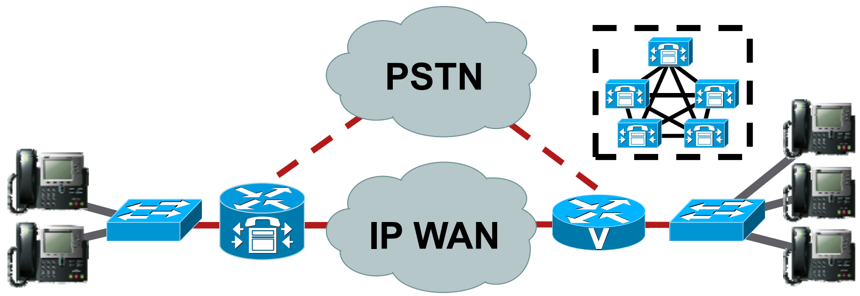
  Queue traffic types appropriately

  Assign WAN queue bandwidth based on traffic type

  Mark or Re-Mark QoS DSCP values

  Assign QoS policy to WAN interfaces

AutoDiscovery	Cisco AutoQoS Policy
Application and Protocol-Types	Cisco AutoQoS Class-Maps Match Statements
Offered Bit Rate (Average and Peak)	Minimum Bandwidth to Class Queues, Scheduling and WRED



*AutoQoS was introduced in 12.3(11)T

Enabling QoS

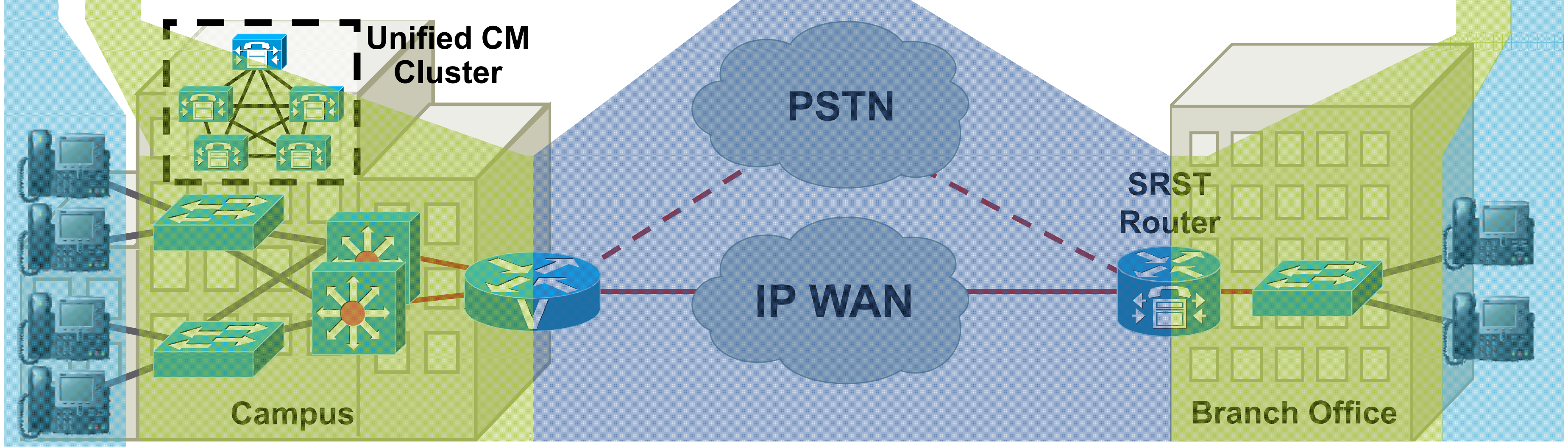
QoS Approach Summary

Classification: Mark the Packets with a Specific Priority Denoting a Requirement for Class of Service from the Network

Trust Boundary: Define and Enforce a Trust Boundary at the Network Edge

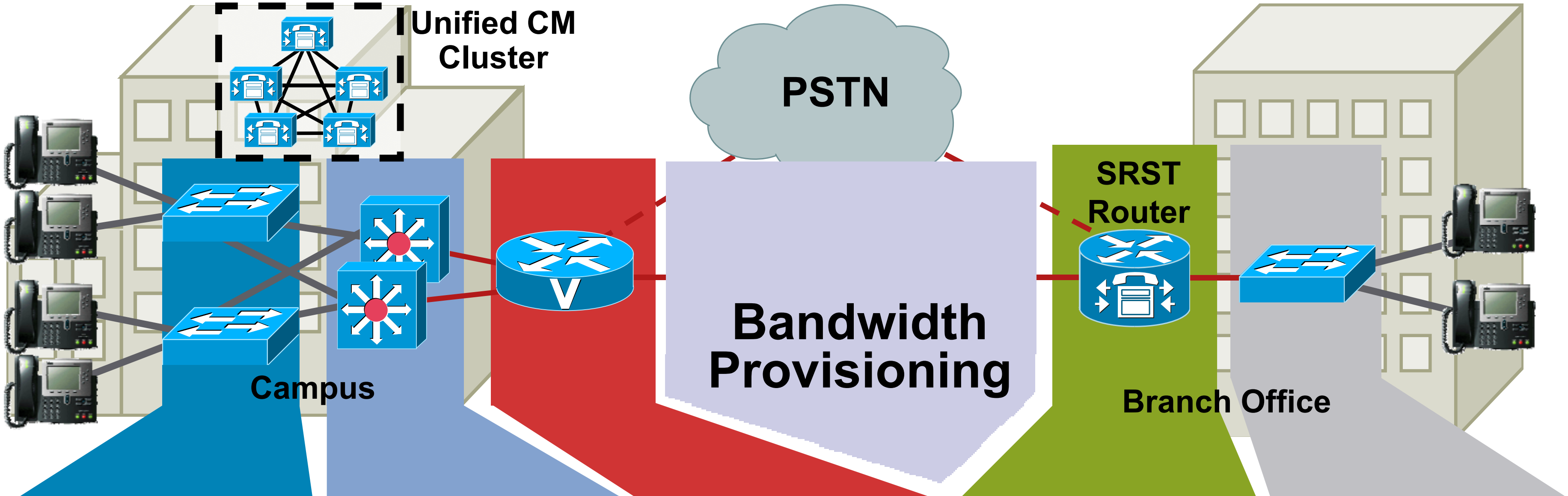
Scheduling: Assign Packets to One of Multiple Queues (Based on Classification) for Expedited Treatment through the Network

Provisioning: Accurately Calculate the Required Bandwidth for All Applications Plus Element Overhead



Enabling QoS

Overall QoS Design Summary



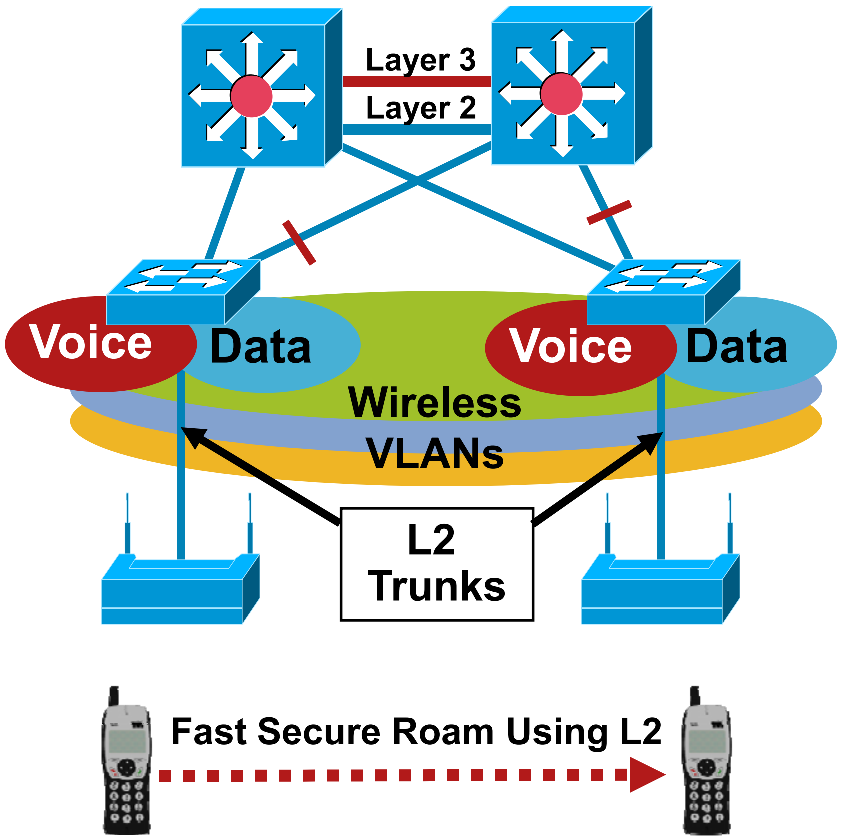
Campus Access	Campus Distribution	WAN Aggregation	Branch Router	Branch Switch
Inline power	Multiple Qs	Multiple Qs	Multiple Qs	Inline Power
Multiple Qs	802.1p/Q	802.1p/Q	802.1p/Q	Multiple Qs
802.1p/Q	Classification	Traffic Shaping	Link Efficiency (LFI, cRTP)	802.1p/Q
Fast link Convergence	Reclassification	Link Efficiency (LFI, cRTP)	Classification	
		Classification	Reclassification	
		Reclassification		

Network Infrastructure Agenda

- Building a Campus Network
- Enabling QoS in the Campus
- Enabling QoS in the WAN
- Overlaying Wireless LANs

Overlaying Wireless LANs

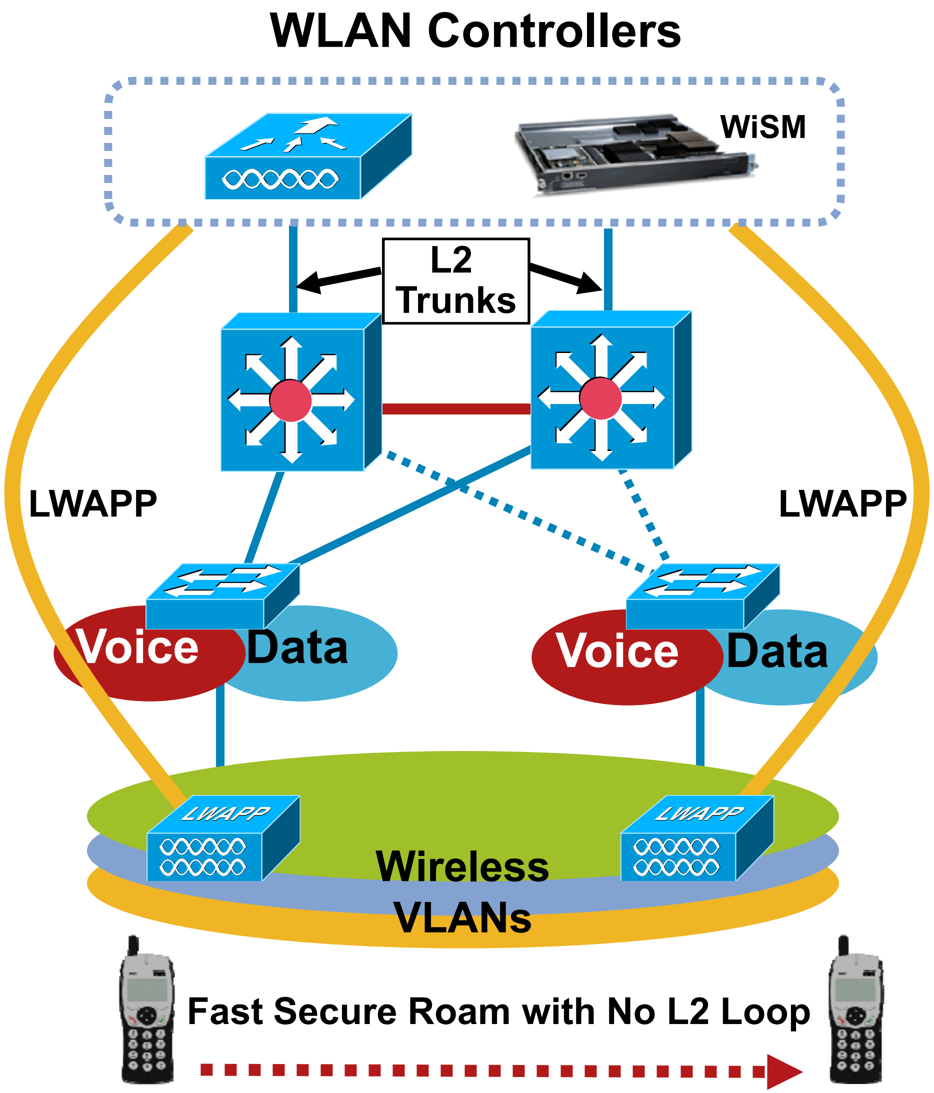
Non-Controller-Based Wireless



- Layer 2 roaming requires spanning at least two VLANs between wiring closet switches
  1. Common 'Trunk' or native VLAN for Access Points (APs) to communicate to Wireless Domain Service (WDS)
  2. The Wireless Voice VLAN
- Use an 802.1Q trunk for switch to AP connection
- Different WLAN authentication/ encryption methods require distinct VLANs

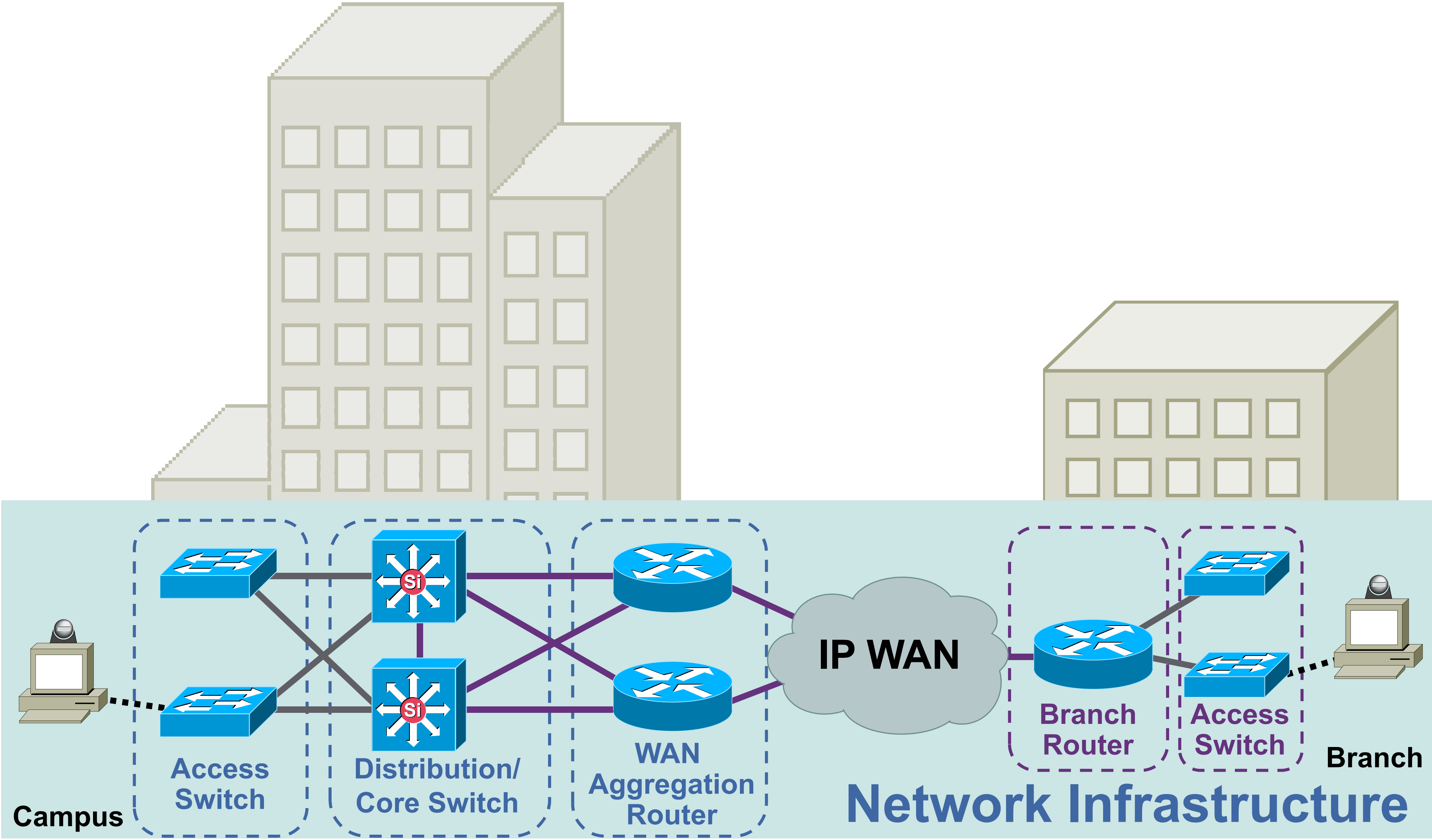
Overlaying Wireless LANs

Controller-Based WLAN: The Architectural Shift



- Cisco WLAN controller and Wireless Services Module (WiSM) provide for a centralized point to bridge all traffic into the Campus
- Control and Data traffic is tunneled to a centralized controller (via Light Weight Access Point Protocol LWAPP)
- No longer a need to span a VLAN between closets (no STP loops)
- No need for trunks between APs and access layer switches
- Details in "Enterprise Mobility 3.0 Design Guide" at www.cisco.com/go/srnd

What We Have Built So Far

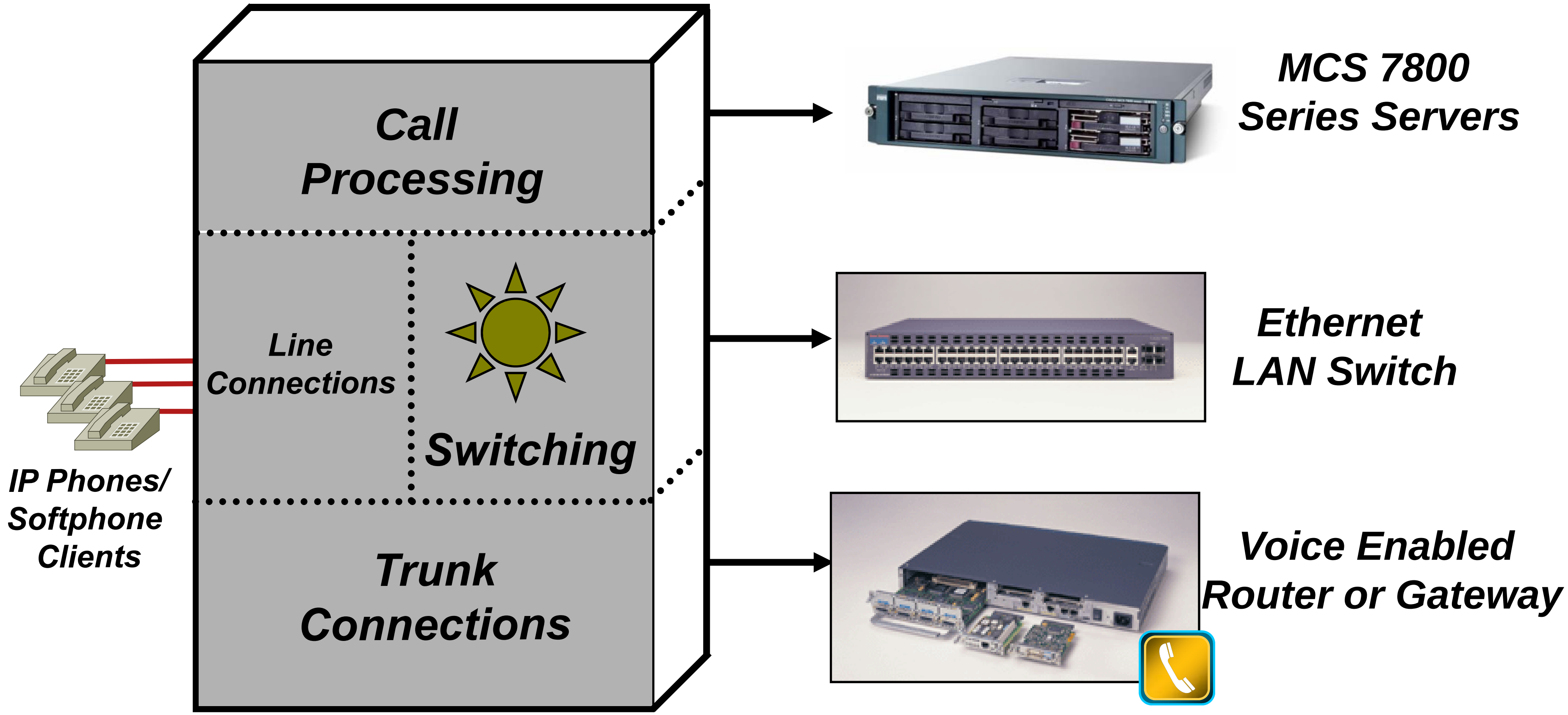


Agenda

- Introduction
- Network Infrastructure
- **Unified Communications Infrastructure**
- Unified Communications Applications
- Security

IP Telephony Architecture

IP Telephony Replaces PBX Architecture



Unified Communications Infrastructure

Deployment Models

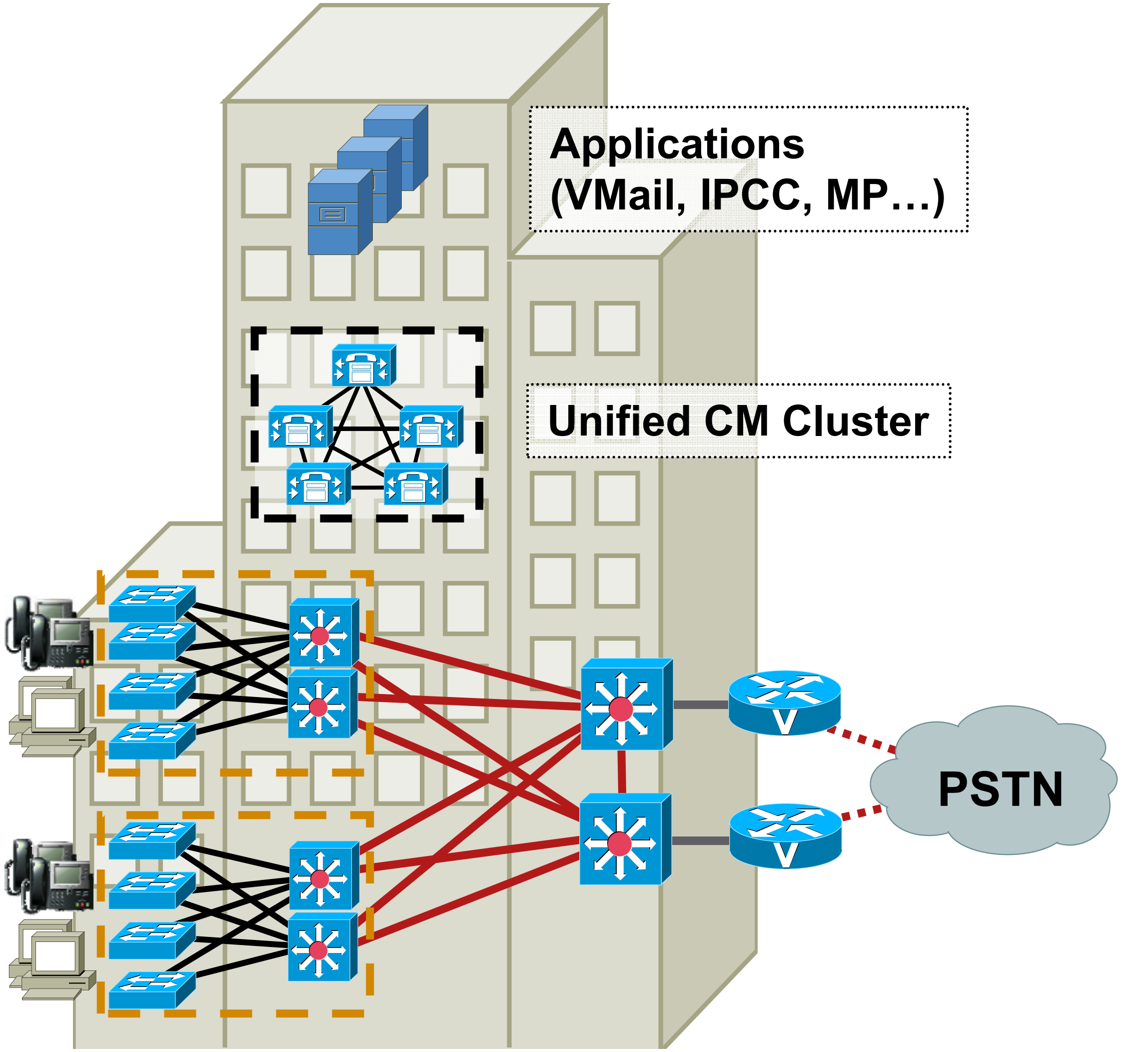
These Deployment Models Are ‘Call Processing’-Based Models Dictated By:

- Physical Location of Unified CM cluster Servers
- Physical Location of Unified CM cluster IP Phones
- Number of Unified CM clusters

Deployment Models

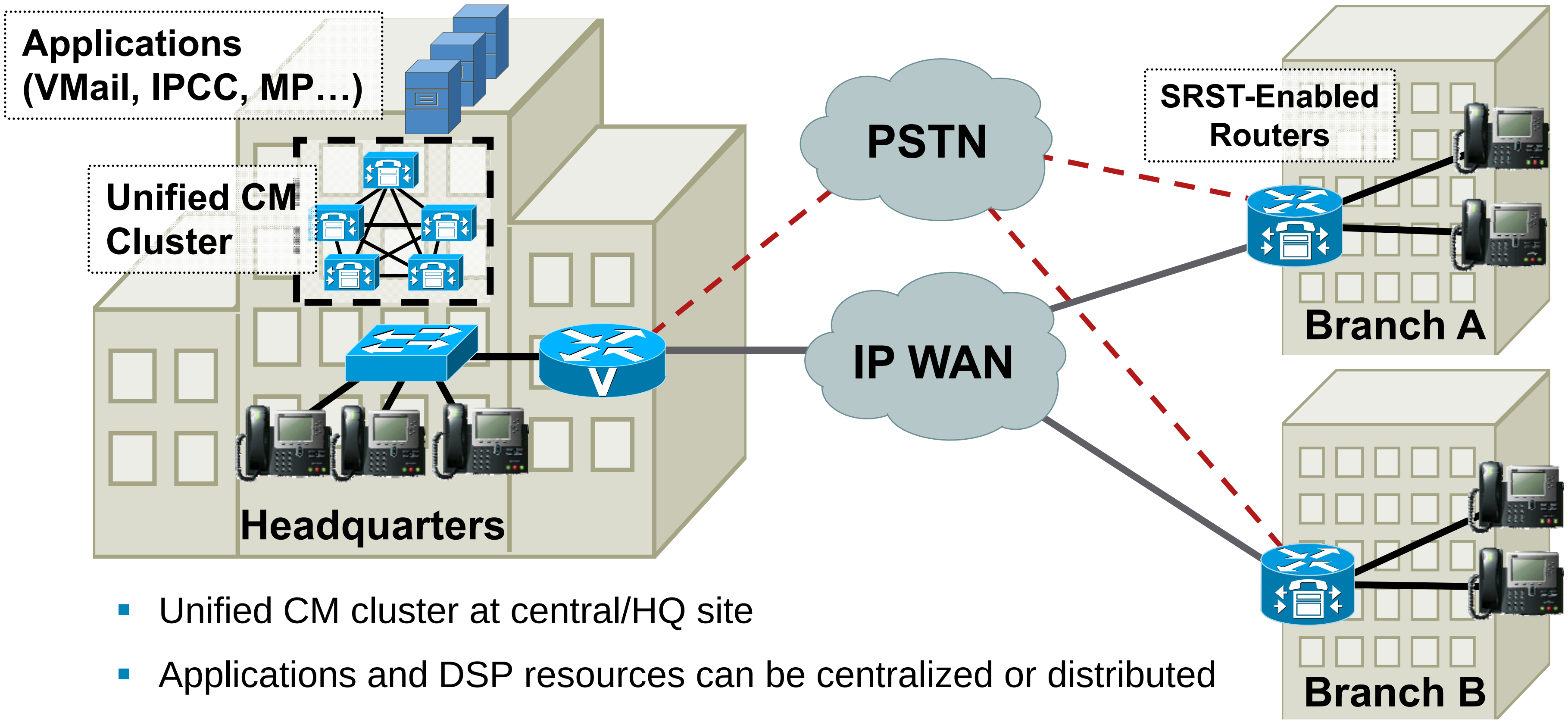
Single Site

- Unified CM, applications and DSP resources at same physical location
- Supports up to 30,000 SIP or SCCP phones per cluster
- PSTN used for all external calls



Deployment Models

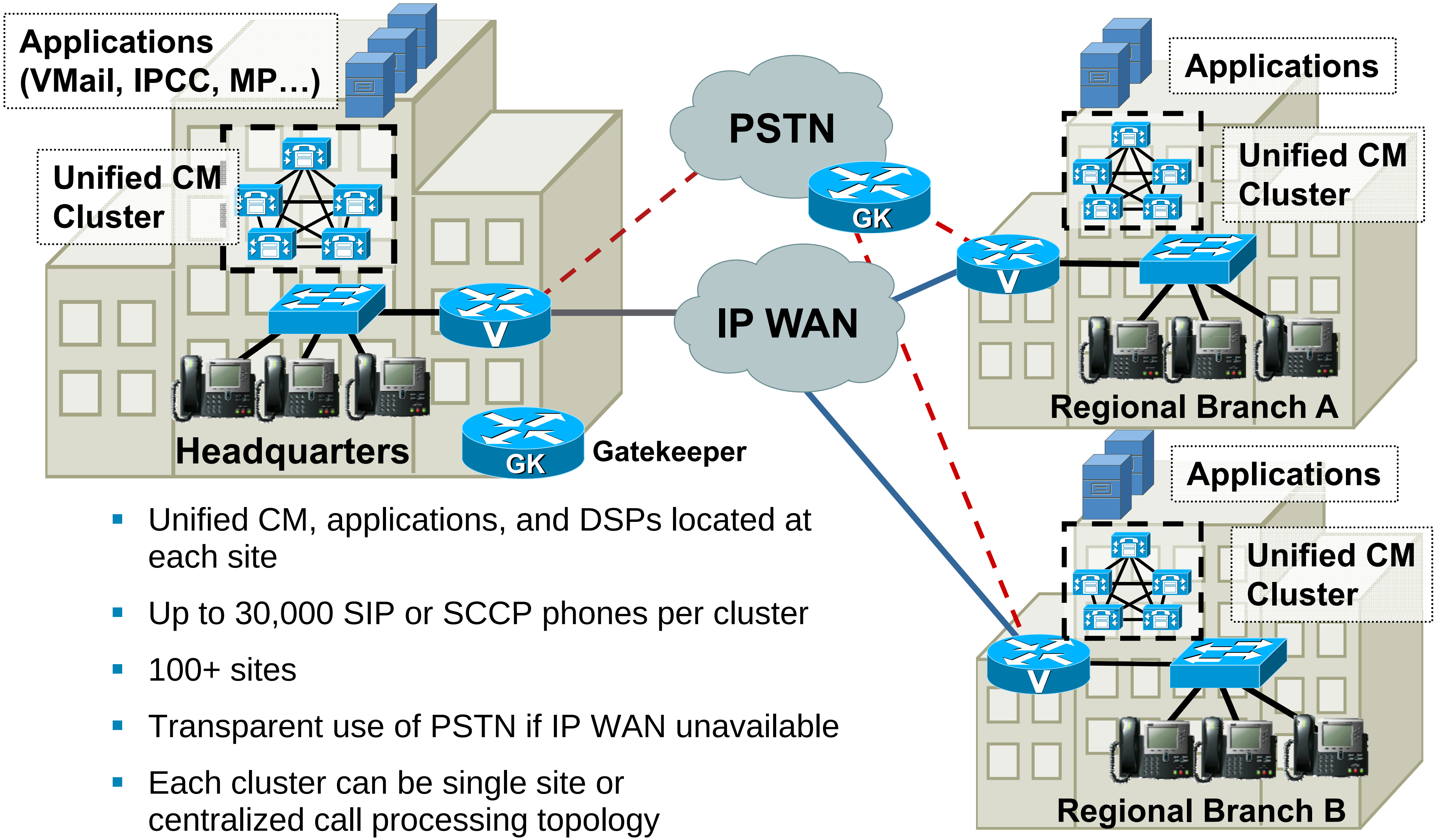
Centralized Call Processing



- Unified CM cluster at central/HQ site
- Applications and DSP resources can be centralized or distributed
- Supports up to 30,000 SIP or SCCP phones per cluster
- If WAN is “busy”, transparent use of PSTN **(Automated Alternate Routing—AAR)**
- Survivable Remote Site Telephony (SRST) for remote branches
- Maximum 1000 sites per cluster (500 branches before Unified CM 6.x)

Deployment Models

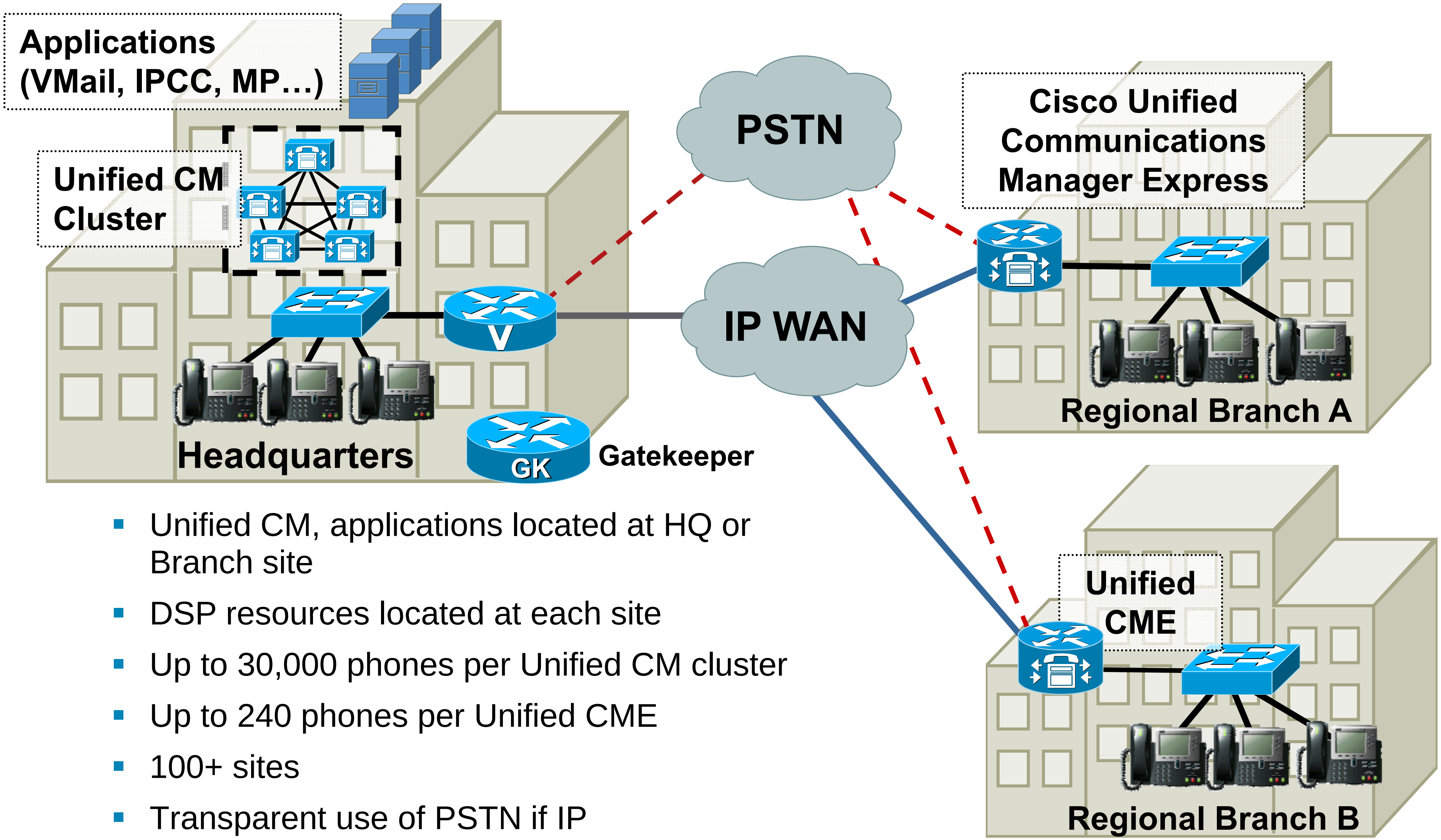
Distributed Call Processing (Unified CM-Unified CM Model)



- Unified CM, applications, and DSPs located at each site
- Up to 30,000 SIP or SCCP phones per cluster
- 100+ sites
- Transparent use of PSTN if IP WAN unavailable
- Each cluster can be single site or centralized call processing topology

Deployment Models

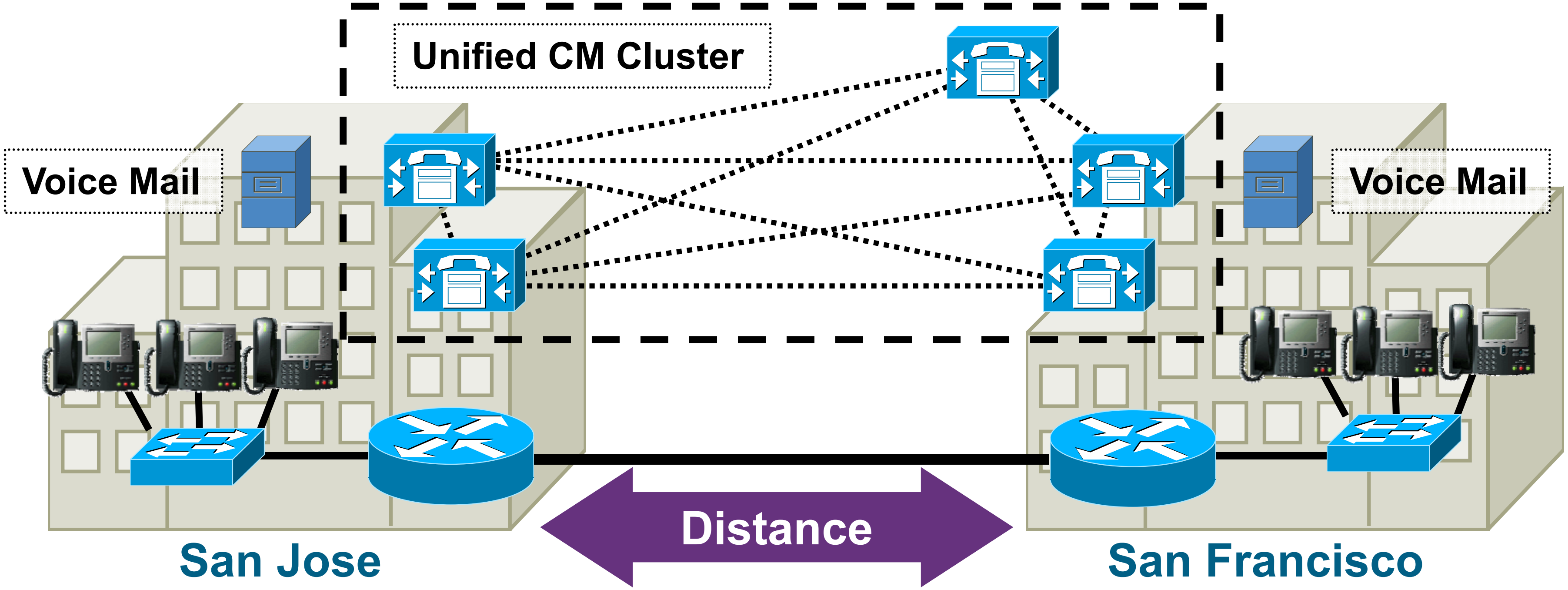
Distributed Call Processing (Unified CM-Unified CME Model)



- Unified CM, applications located at HQ or Branch site
- DSP resources located at each site
- Up to 30,000 phones per Unified CM cluster
- Up to 240 phones per Unified CME
- 100+ sites
- Transparent use of PSTN if IP WAN unavailable

Deployment Models

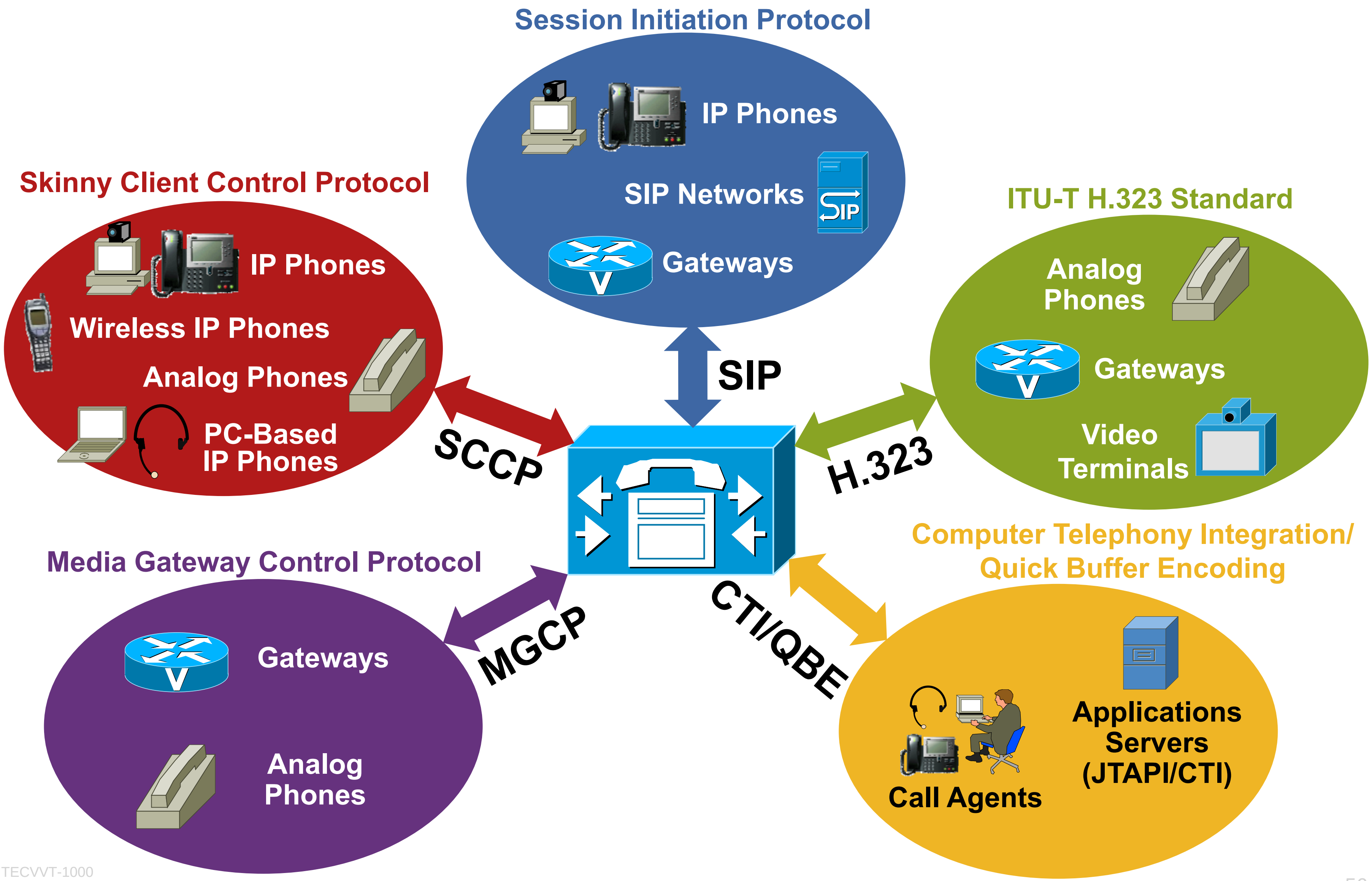
Clustering over the WAN (CoW)



- Unified CM servers in a cluster separated by WAN for spatial redundancy
- Applications may be located at each site, thus separated by WAN
- Single point of administration, feature transparency (e.g. Extension Mobility), unified dial plan
- Maximum 80-ms round-trip delay between any two Unified CM across the WAN
- 1.5 Mbps bandwidth for each 10,000 BHCA between sites
- Maximum of eight active locations

Unified Communications Infrastructure

Signaling Protocols: Unified CM as “Protocol Translator”



Unified Communications Infrastructure

Network Services: IP Phone Bootup Process

1. Inline Power (ILP)

   Inline Power Initialization

2. Cisco Discovery Protocol (CDP) or Link Layer Discovery Protocol-Media Endpoint Discovery (LLDP-MED)

   ILP Negotiation, Voice VLAN ID

3. Dynamic Host Configuration Protocol (DHCP)

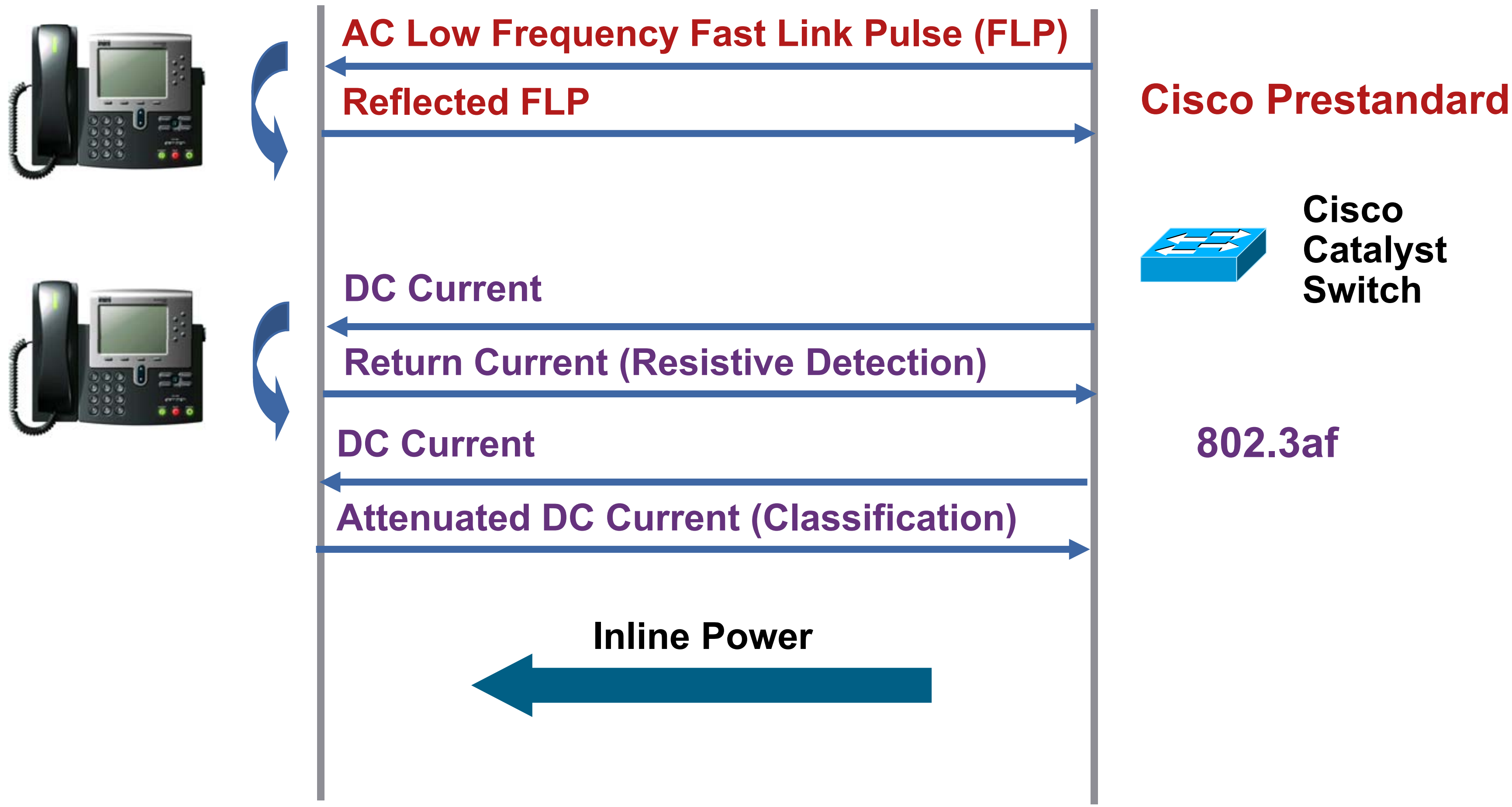
   IP Assignment, TFTP Server Allocation, DNS (optional)

4. Trivial File Transfer Protocol (TFTP)

   Configuration File, IP Phone Firmware

Unified Communications Infrastructure

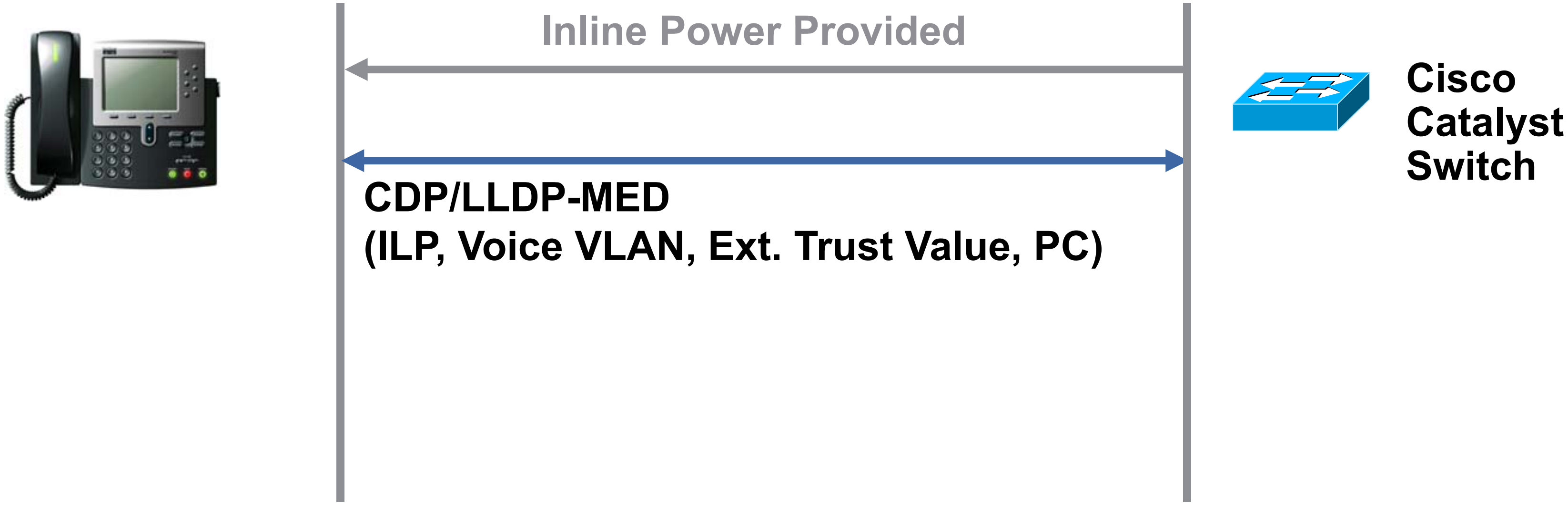
Network Services: Inline Power



On Phone: Mute, Headset, Speaker Buttons Are Illuminated

Unified Communications Infrastructure

Network Services: CDP or LLDP-MED



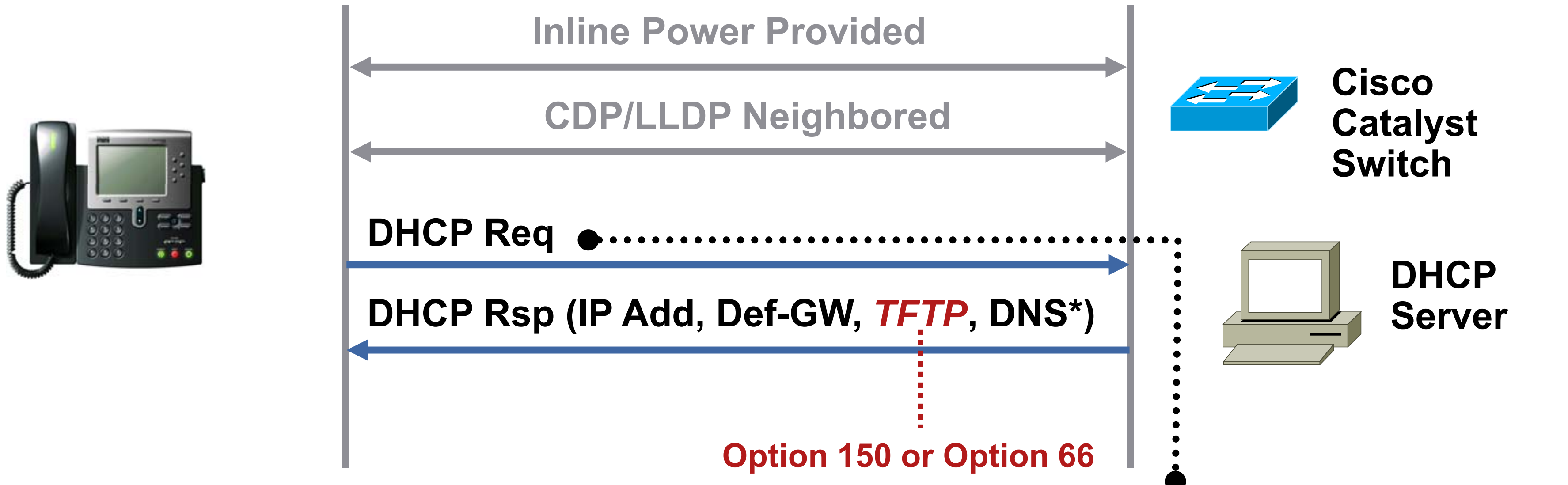
- Phone displays: "Configuring VLAN"
- Phone settings: Settings=>NetCfg=>"Operational VLAN ID"

LLDP-MED is supported as of IP Phone Firmware 8.3(3)

LLDP-MED and CDP White Paper:
http://www.cisco.com/en/US/technologies/tk652/tk701/technologies_white_paper0900aecd804cd46d.html

Unified Communications Infrastructure

Network Services: DHCP

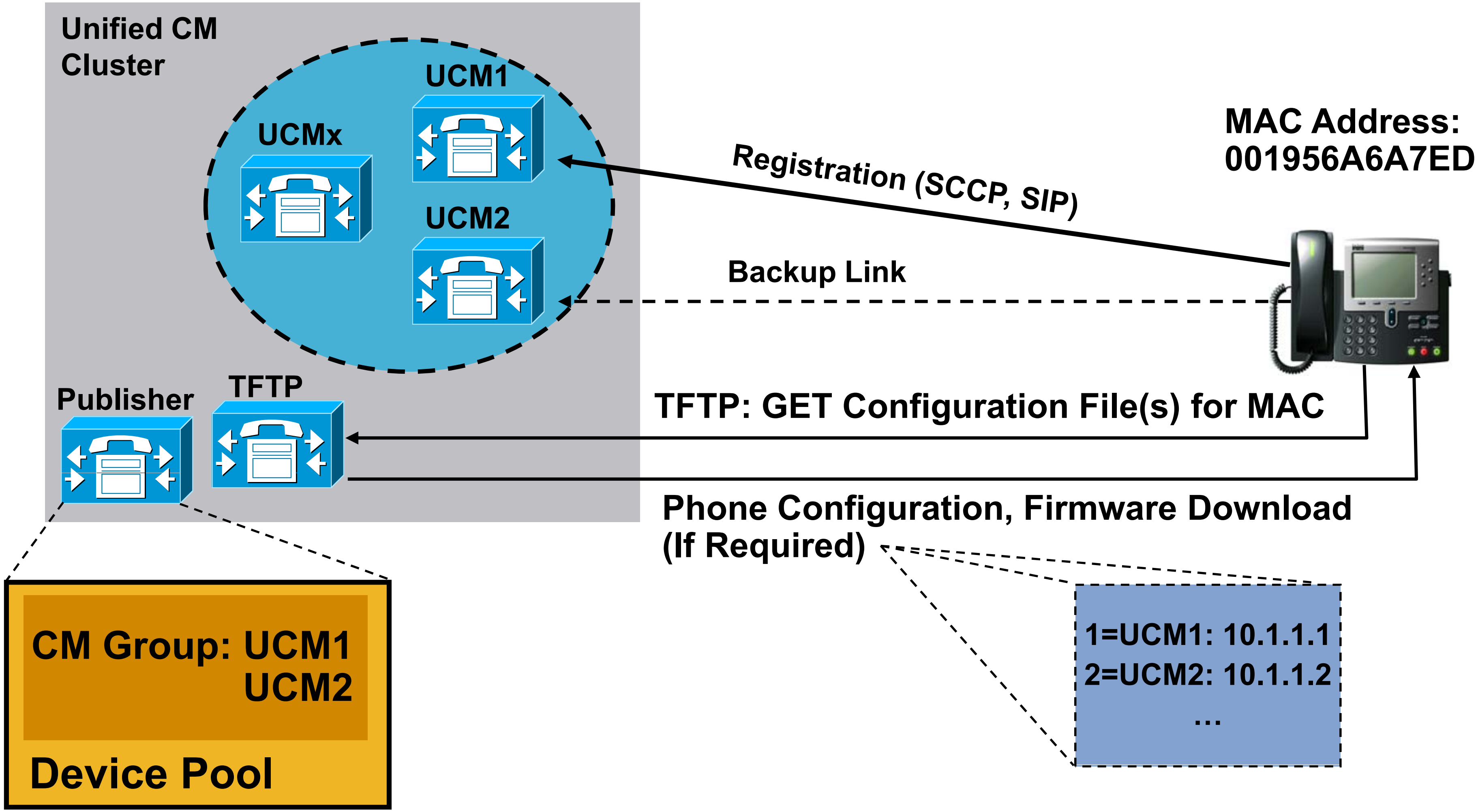


DHCP Request Must Be Made in the Correct VLAN to Place the Phone in the Correct Subnet!!

- Phone displays: "Configuring IP"
  (DNS is optional)
- Phone settings: Settings=>NetCfg=>"DHCP Server"
  Settings=>NetCfg=>"IP Address"
  Settings=>NetCfg=>"TFTP Server X"

Unified Communications Infrastructure

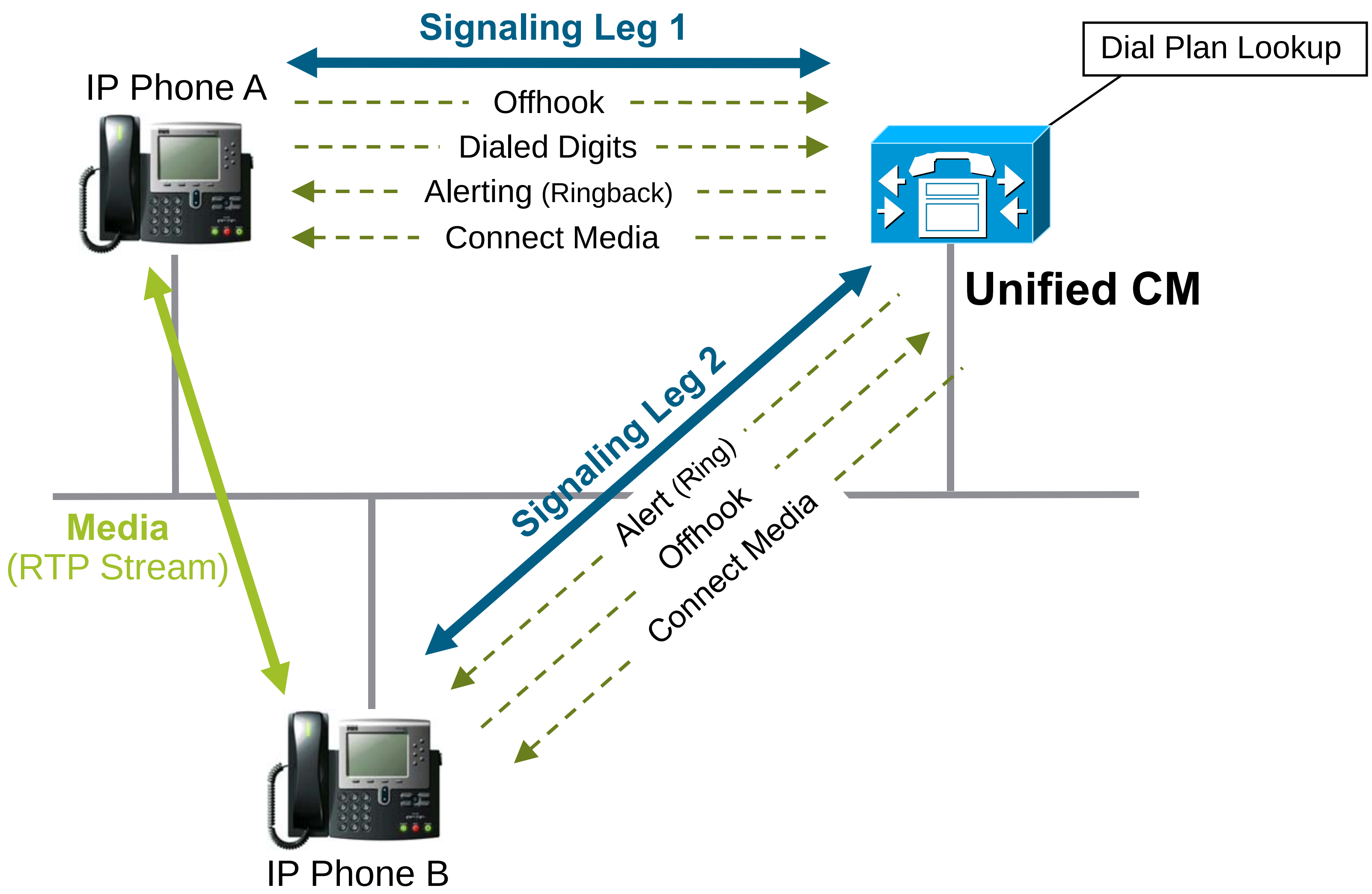
Network Services: TFTP



Unified Communications Infrastructure

Basic Call Processing: Single Site Deployment Model

IP Phone to IP Phone Example

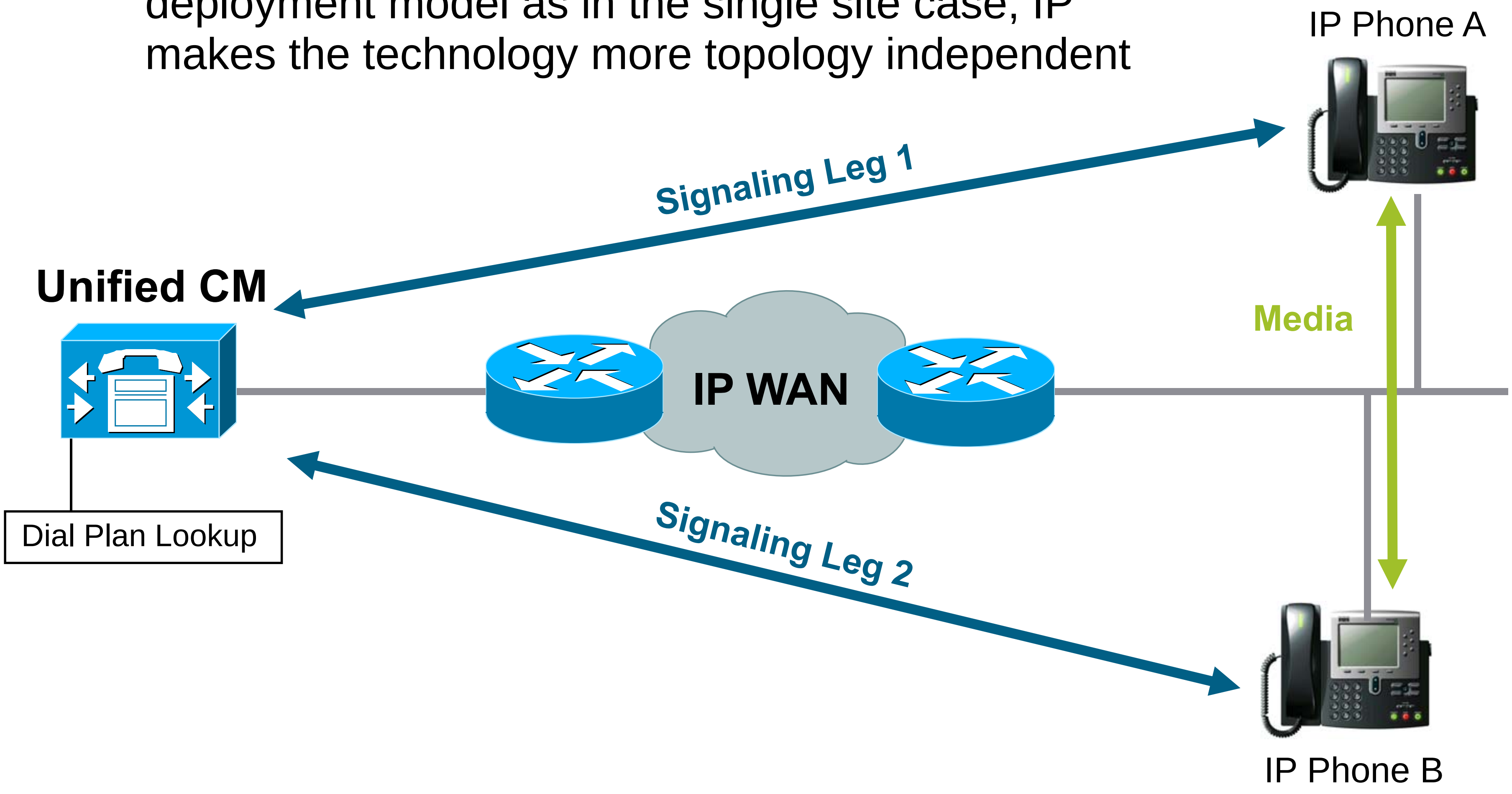


Unified Communications Infrastructure

Basic Call Processing: Centralized Deployment Model

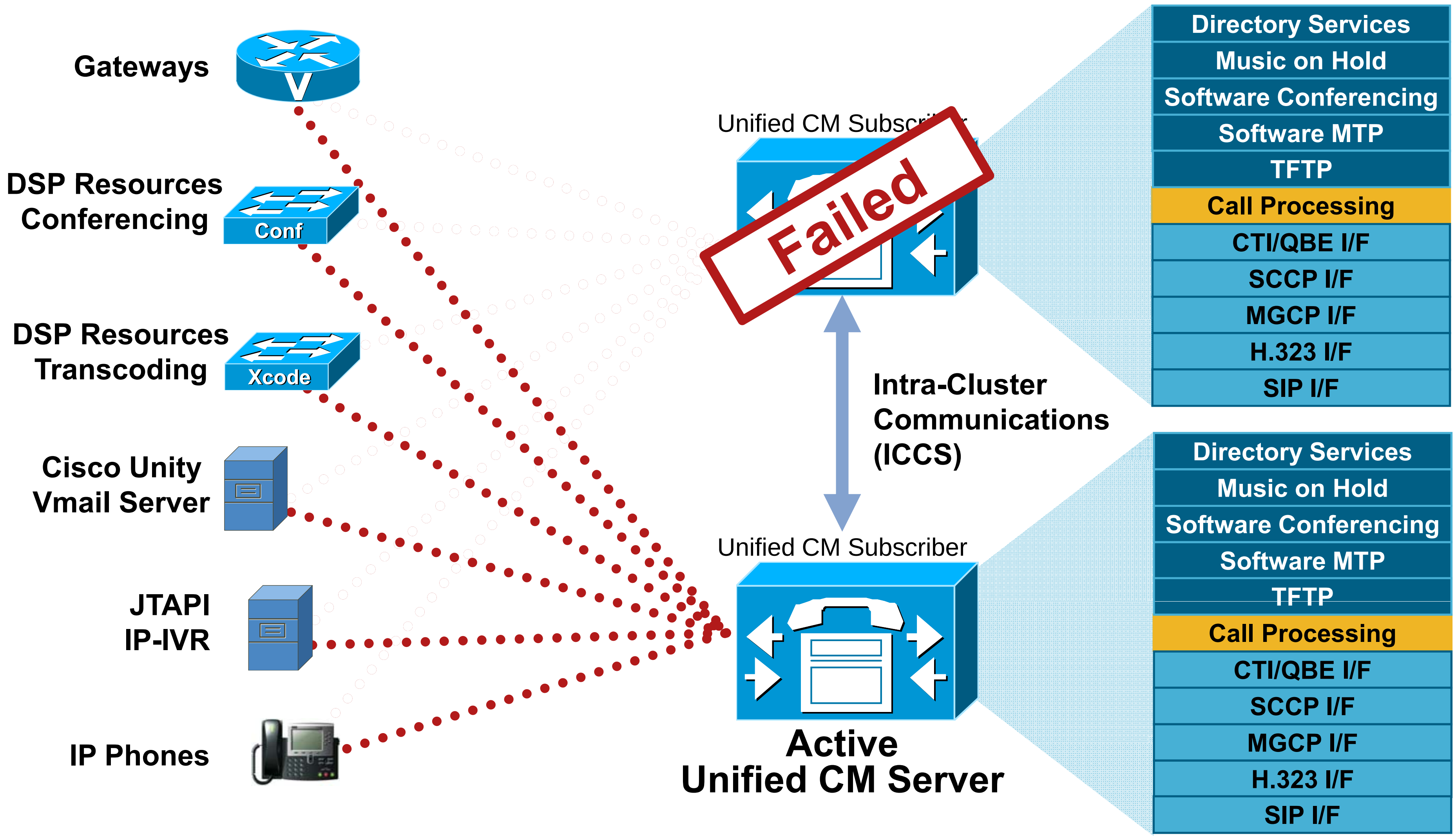
IP Phone to IP Phone Example

- Call Processing is essentially the same in this deployment model as in the single site case; IP makes the technology more topology independent



Unified Communications Infrastructure

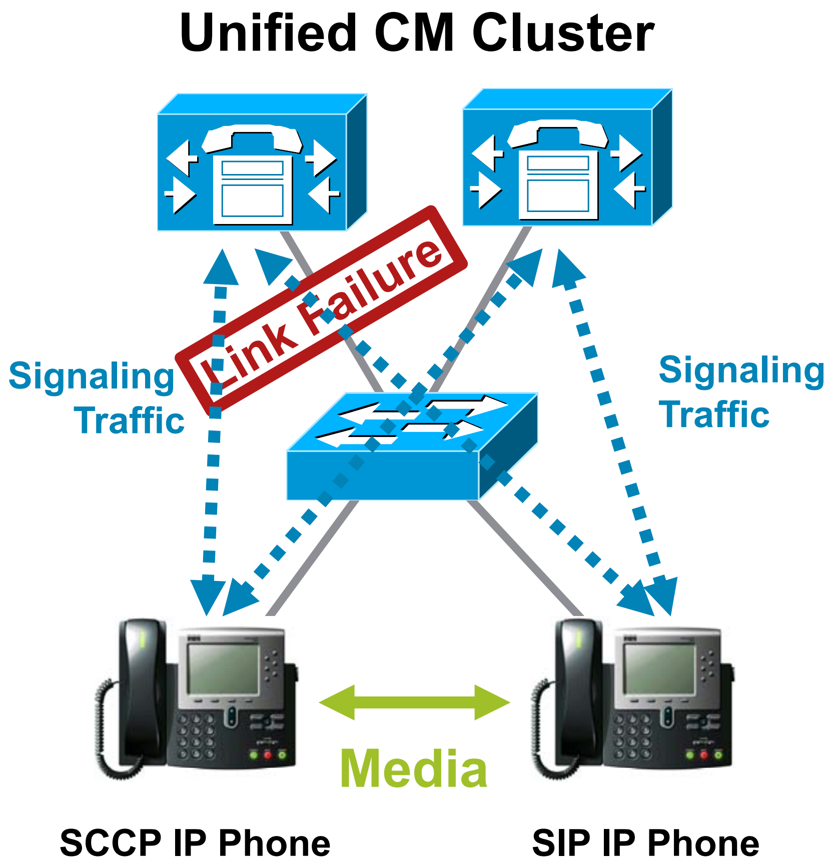
Failover and Redundancy: Server Redundancy



Unified Communications Infrastructure

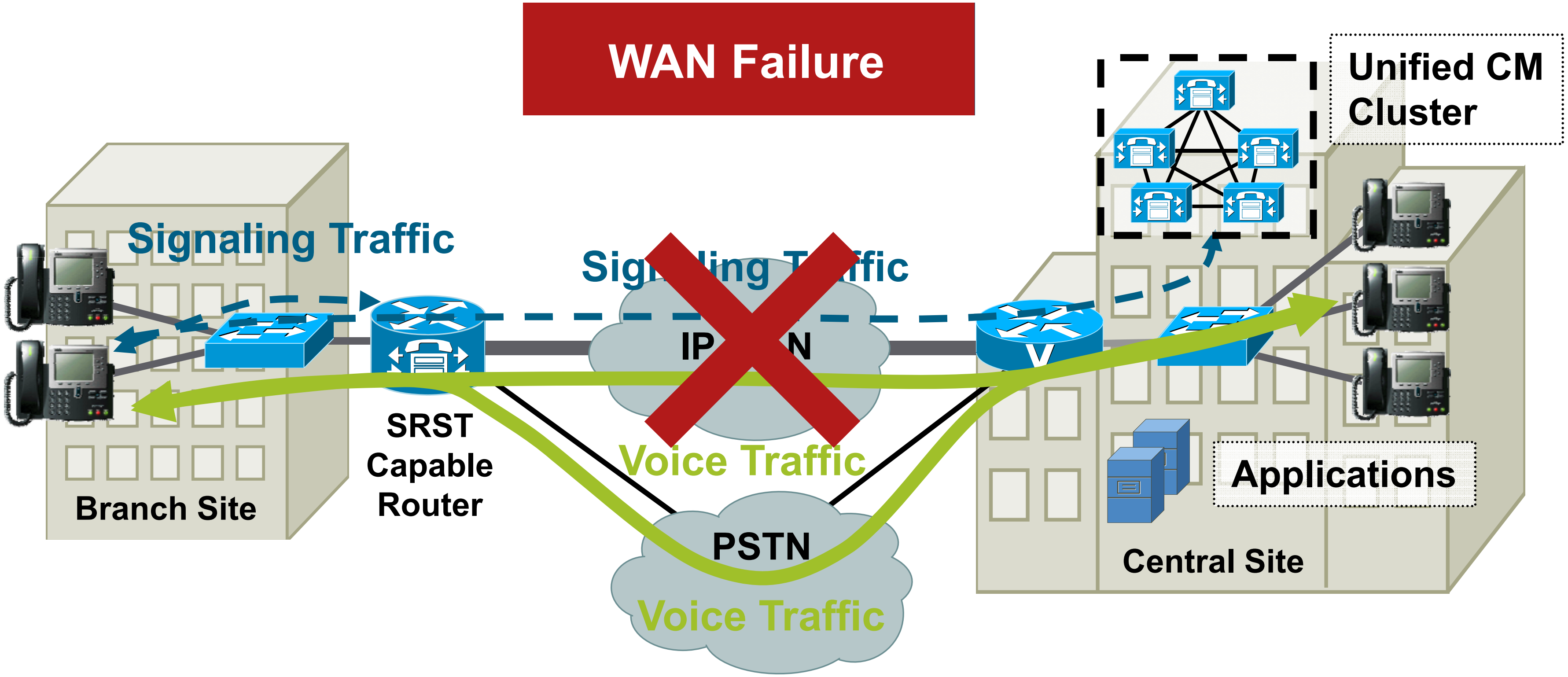
Failover and Redundancy: Media Survivability

- Media does not fail during loss of connectivity to Unified CM
- No Services (e.g. hold, transfer, etc.) when Unified CM not available
- Once the call is complete, phones re-register to backup Unified CM



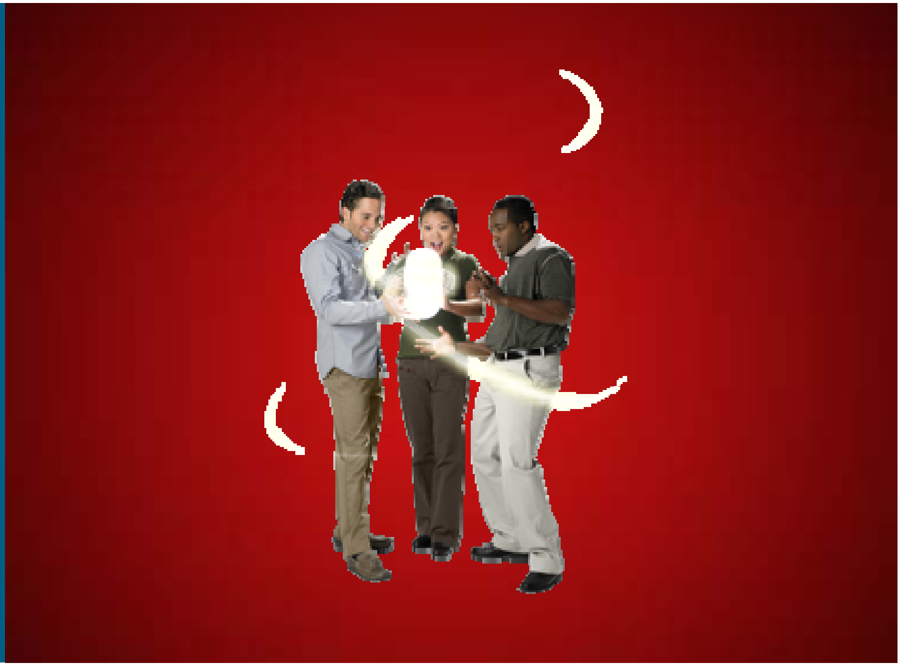
Unified Communications Infrastructure

Failover and Redundancy: **S**urvivable **R**emote **S**ite **T**elephony



- IP Phones have SRST router IP as the last option in their CM GROUP configuration
- Support for both SIP and SCCP IP Phones
- With SRST, only a **subset** of features are available to the phones (DID, DOD, call hold, transfer, speed dial, caller ID, etc.)
- H323 PSTN GW connectivity option during failure modes via VoIP/POTS dial-peers; MGCP GWs require the ‘MGCP Fallback to H323’ feature

Unified Communications Infrastructure



Part 2

Media Resources

Conferencing, Transcoding, Music on Hold

- **Conference Bridge**

  DSPs needed for multi-codec conferences

- **Media Termination Point**

  Media Termination

  DSPs **optional**

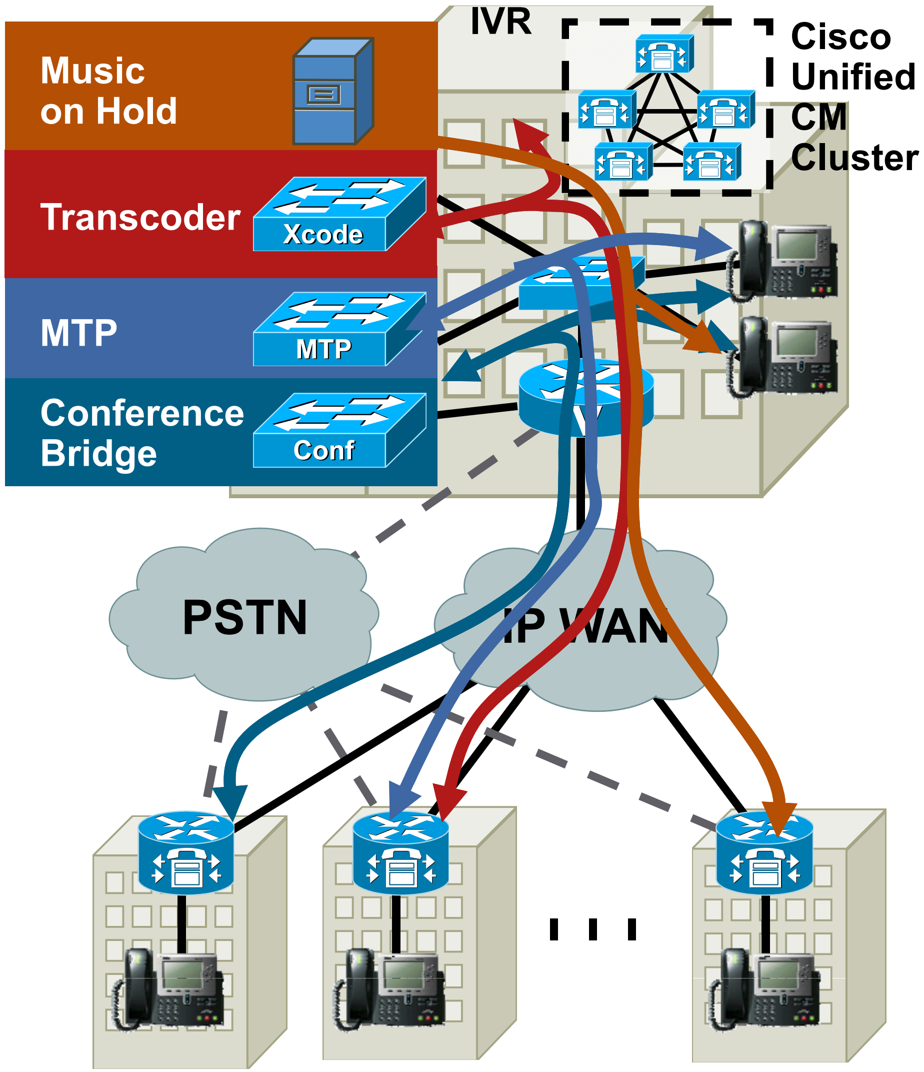
- **Transcoding**

  DSPs needed to transcode multiple CODEC types (e.g., G.711 to G.729)

  Automatic codec selection

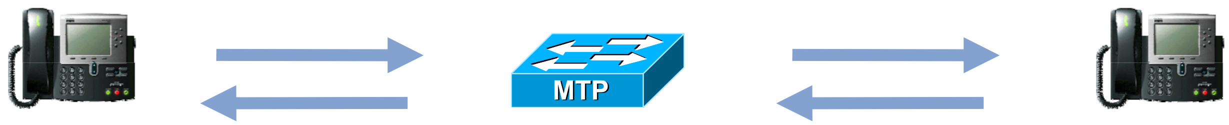
- **Music on Hold**

  Multiple source types possible (centralized or branch-based)



Media Resources

Media Termination Point and Transcoder

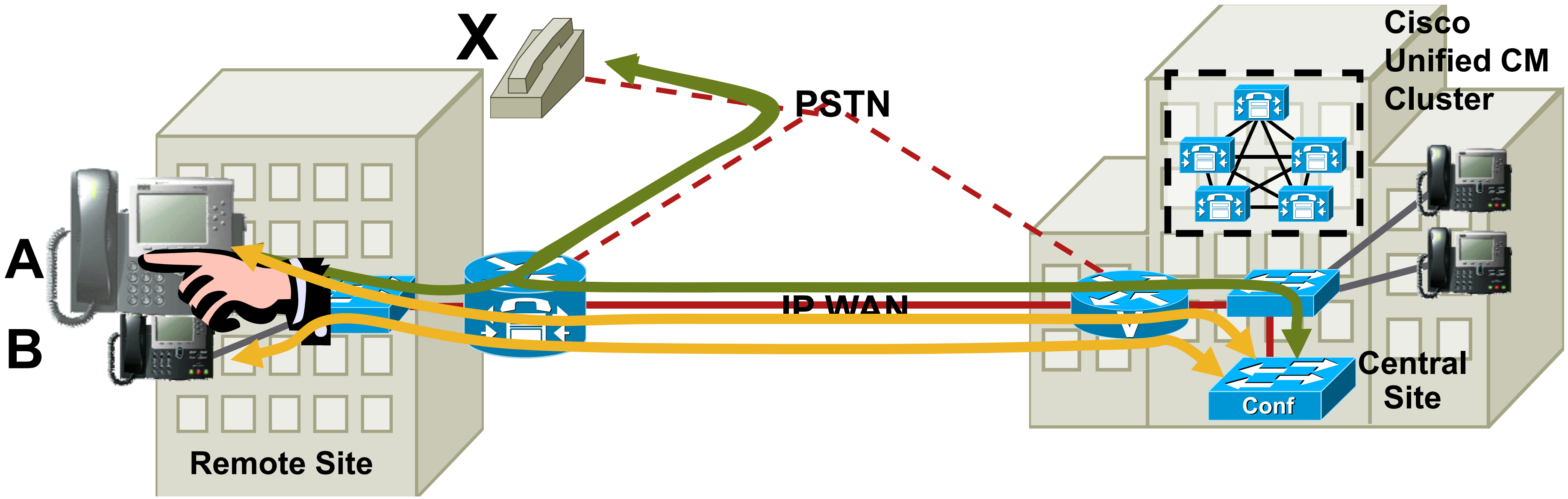


- Terminates media streams (same codec type)
- Transrating of media streams (20ms → 30ms)
- H.323 Outbound FastStart (vs. slow start)
- SIP outbound early-offer (vs. delayed-offer)
- DTMF—Relay

- Enhanced version of MTP resource
- Transcoder = converts from one codec to another

Media Resources

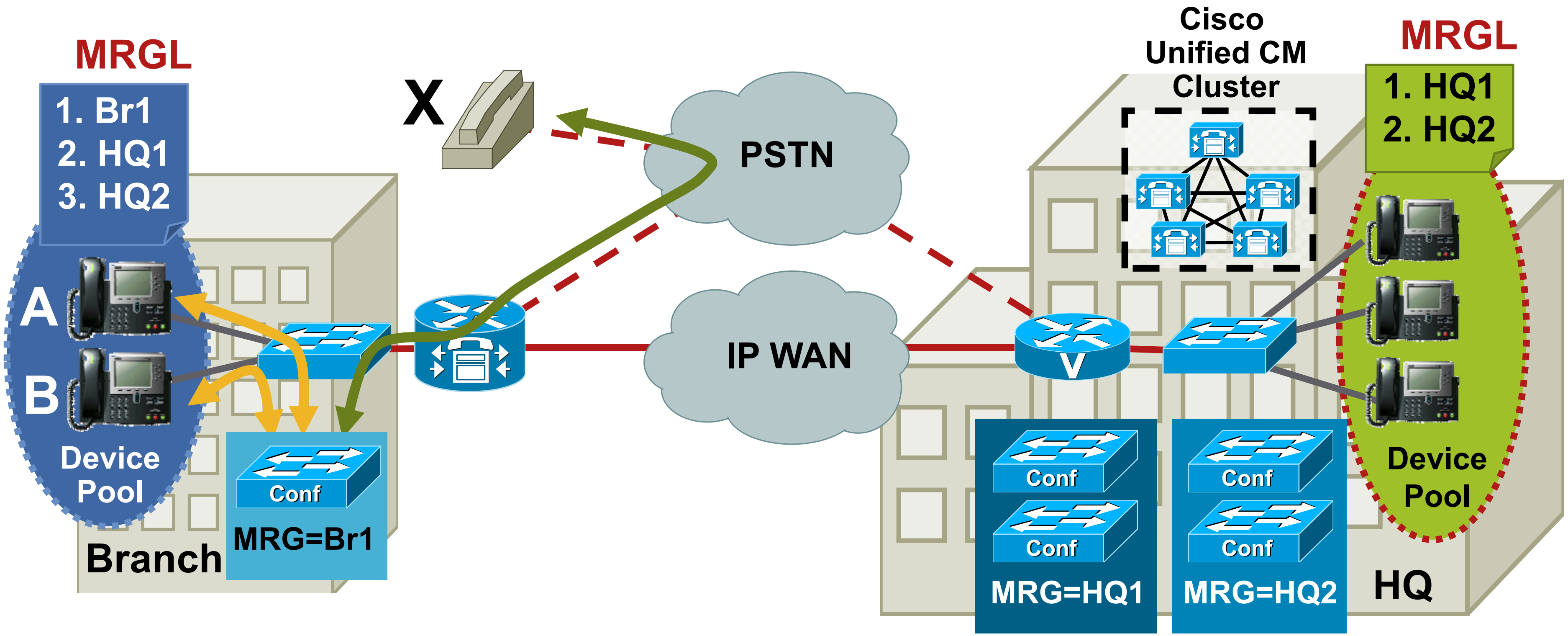
Centralized Conferencing Resources



- External caller X calls A—no voice across WAN
- A conferences B in
- Three voice streams across WAN
- Maximum 3-party conferencing in Unified SRST mode

Media Resources

Distributed Conferencing Resources

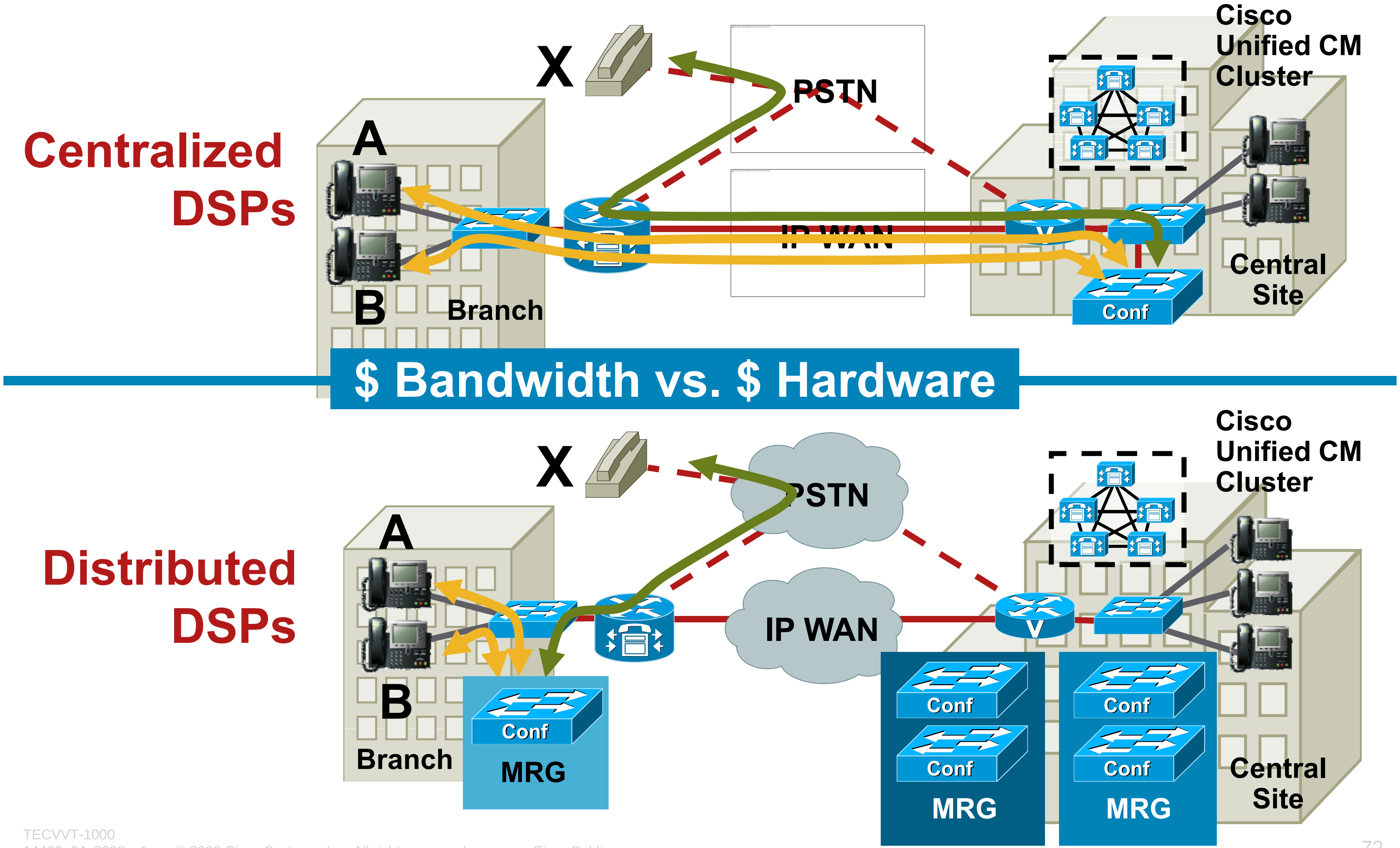


- Conference between A, B, and X—No voice across WAN
- Requires extra hardware at branch
- Maximum three-party conferencing in Unified SRST mode

MRG = Media Resource Group
MRGL = Media Resource Group List

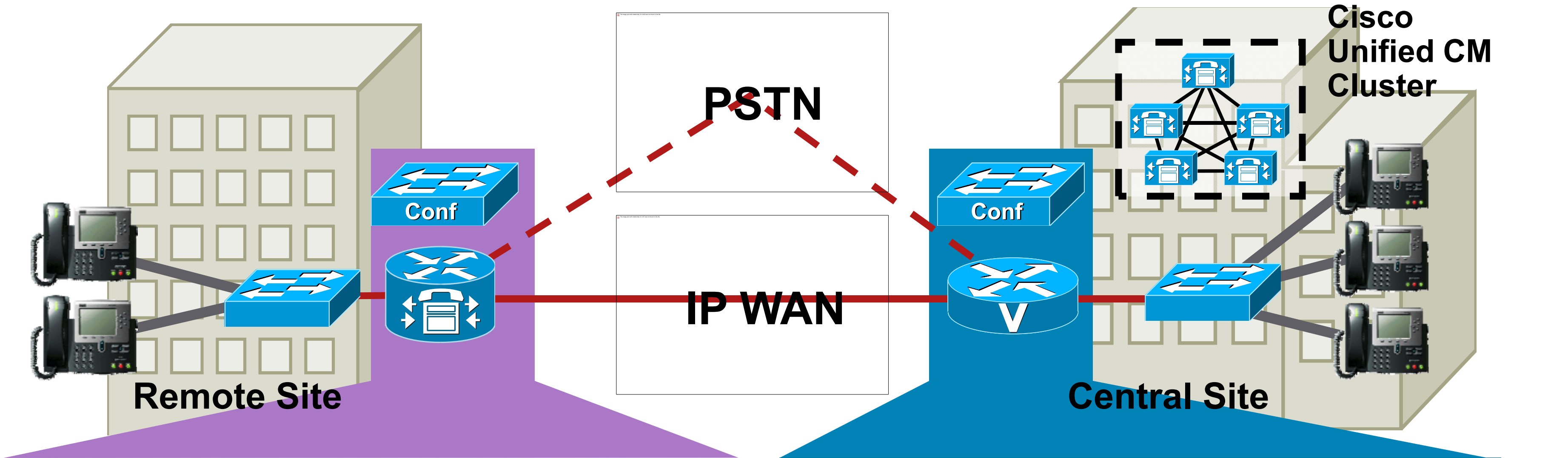
Media Resources

Centralized vs. Distributed DSPs



Media Resources

DSP Platform Recommendations

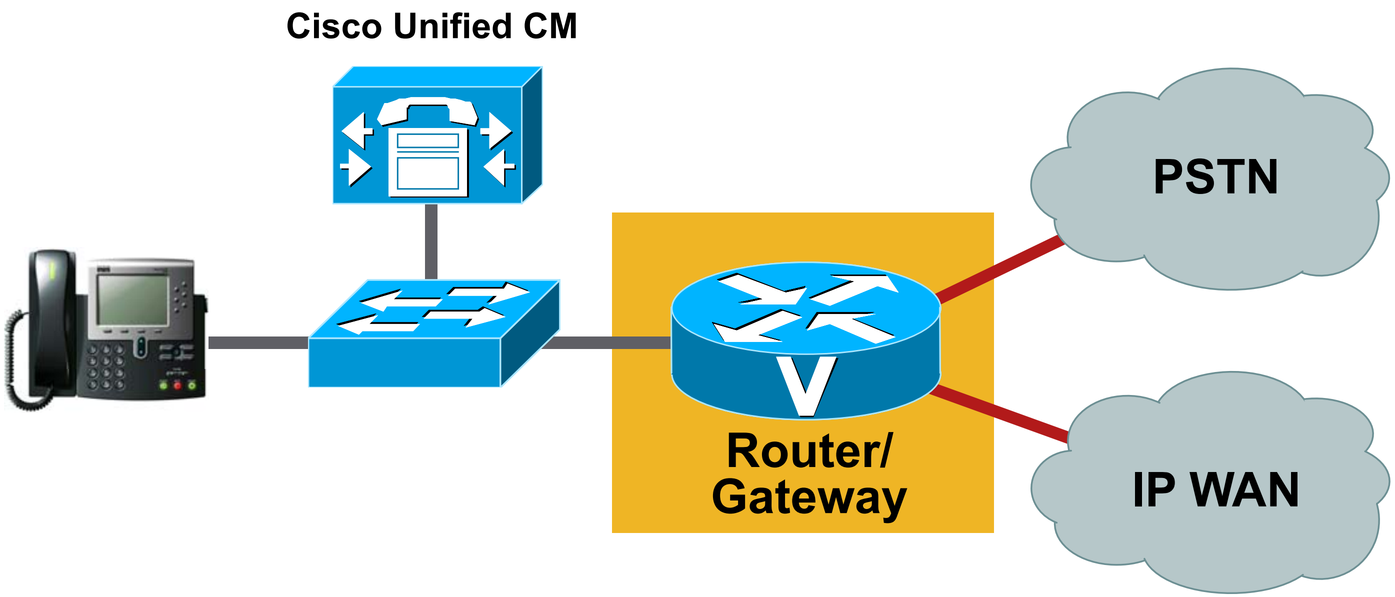


- NM-HDV
- NM-HDV2
- ISR (onboard)

- WS 6608 (32 parties)
- CMM (128 parties)
- NM-HDV2 (eight parties)
- Hardware CFB
- Software CFB

Gateways

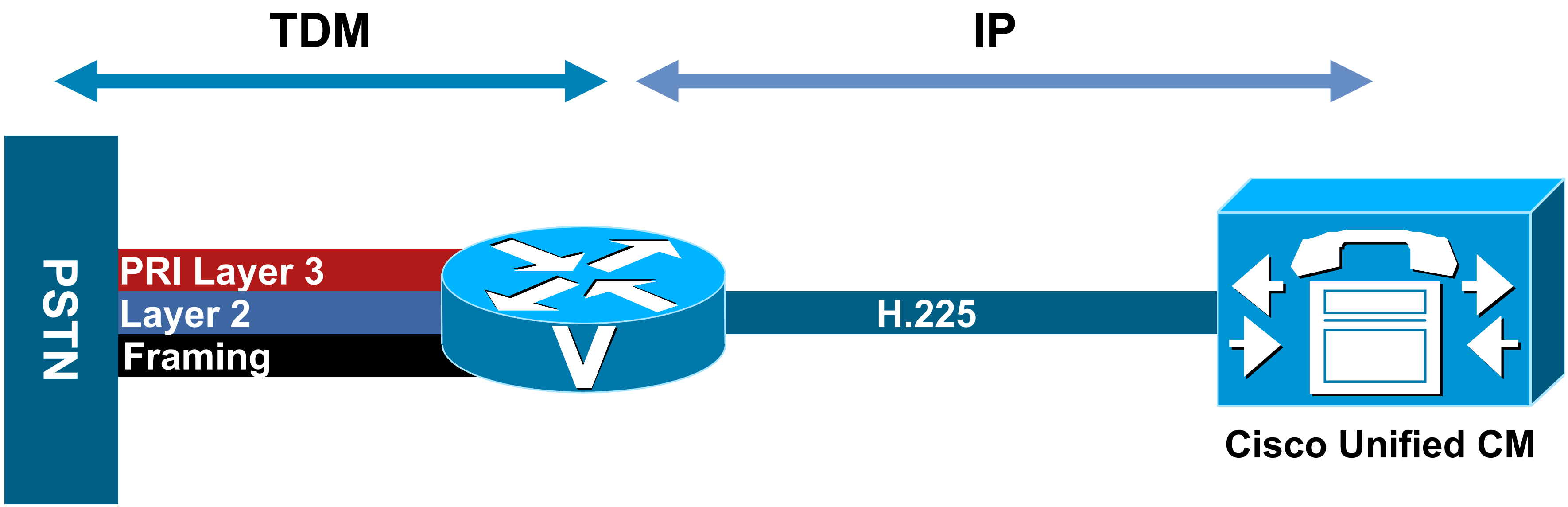
Gateway Selection Criteria



- Voice port density requirements
- Signaling protocol (H.323, MGCP, SIP, etc.)
- Support for required PSTN signaling types
- Support for required WAN interfaces and QoS

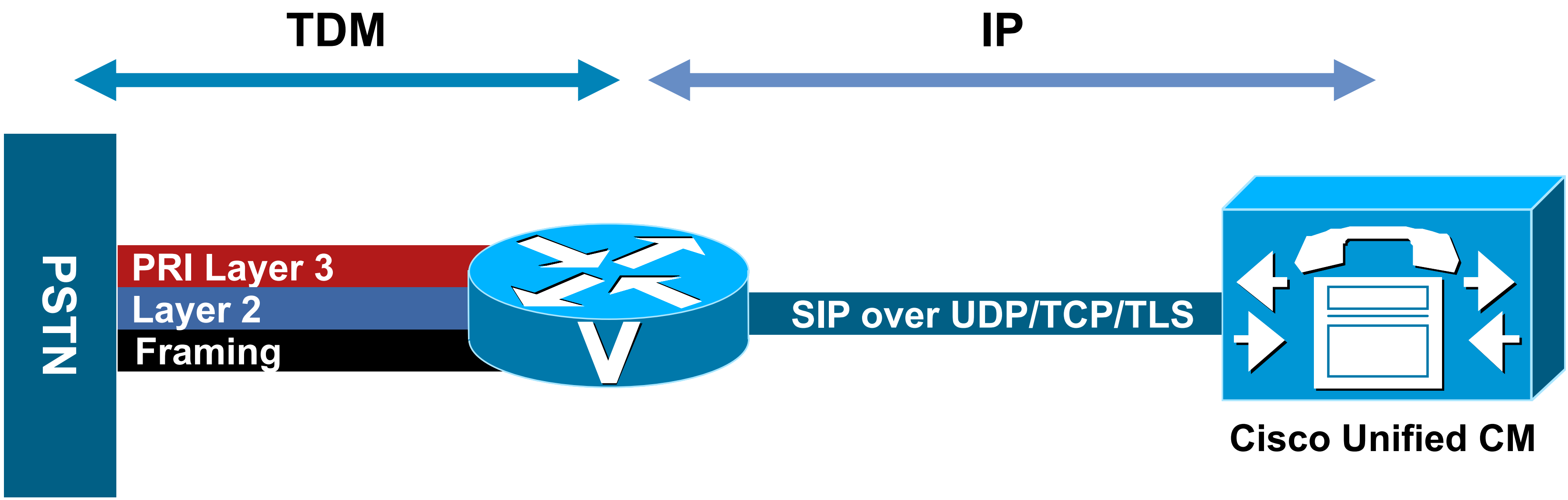
Gateways

H.323



- All PSTN signaling terminates on gateway
- H.225 communication between gateway and Cisco Unified CM
- H.323 is a “peer-to-peer” protocol: each side can make decisions

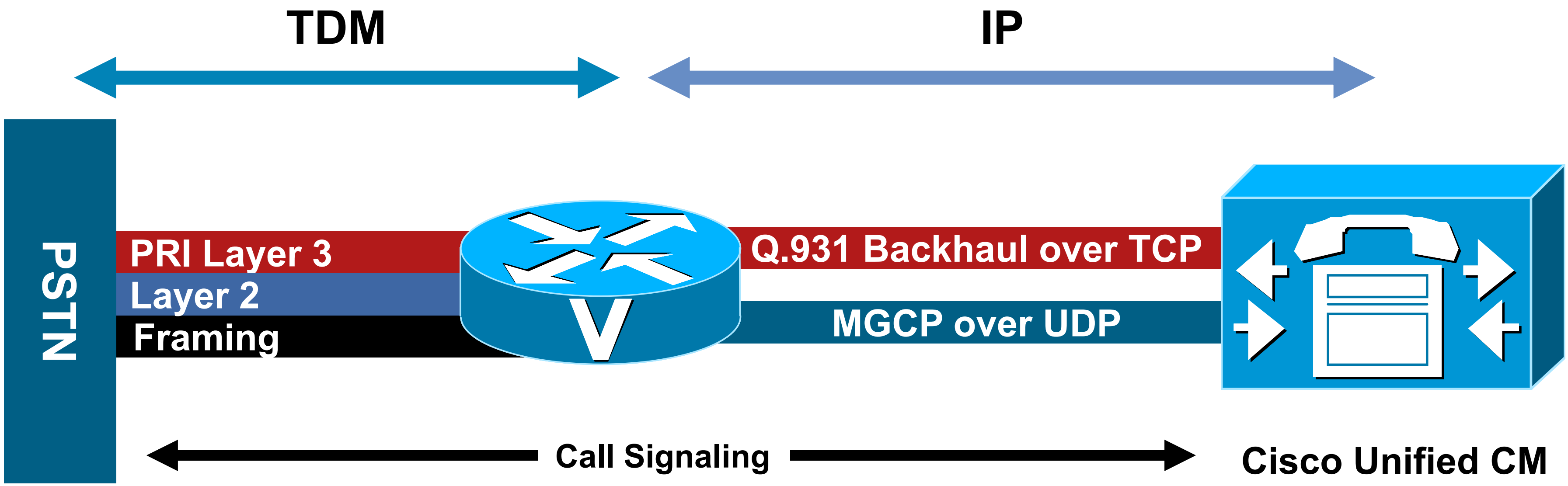
Gateways
SIP



- All PSTN signaling terminates on gateway
- SIP communication between gateway and Cisco Unified CM
- SIP is a "peer-to-peer" protocol: each side can make decisions

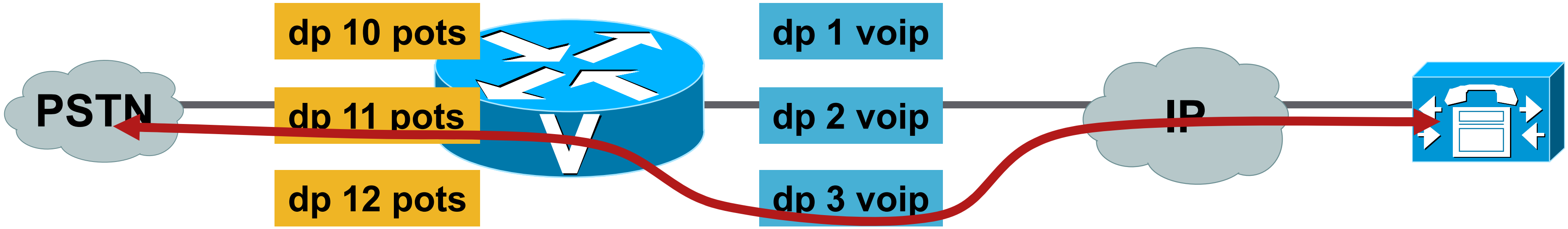
Gateways

MGCP: Q.931 Backhaul



- Framing and layer 2 signaling terminates at the gateway
- Layer 3 signaling is backhauled to the Cisco Unified CM
- MGCP is a “client-server” protocol: all call-related decision making is done by the server
- MGCP 0.1 with Cisco Unified CM only

The Power of Cisco IOS Dial-Peers: H.323 and SIP



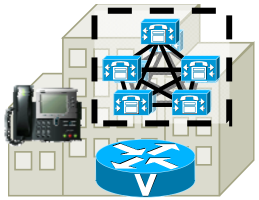
Dial-Peers Allow You to:

- Switch calls intelligently if required (interpret the dial plan)
- Digit manipulation (called, calling and numbering plan)
- Failover (preferences) to alternate destinations
- Load balancing
- Video ISDN switching
- Insert applications into the call path: TCL/VXML
  - Build support for signaling variations (e.g. CLID on T1 CAS)
  - Hookflash trunk release on FXO
  - VXML call control for call centers
  - Redistribute calls-in-q for CVP
  - AA in the GW

These Capabilities Do Not Exist for MGCP-Controlled GWs

Protocol Deployment Considerations

Large/Campus Sites



- Characteristics of larger site(s)—often best served by MGCP
- High-density GWs to PSTN, often PRI
- Dedicated GW platforms
- Caller ID/name delivery required
- Digital TDM protocol (often PRI)
- QSIG connectivity (with supplementary services) to legacy PBXs required
- Other considerations

  NFAS is H.323/SIP only

  Very high density GWs such as T3 (5x00) are H.323/SIP only

Small/Branch Sites

- Characteristics of branch site(s)—often best served by H.323/SIP
- Low-density GW to PSTN, often analog
- GW and router features used on same platform (integrated access)
- Caller ID on analog FXO required
- Mixes of PSTN TDM protocols required (FXO, A-DID, BRI, Frac-PRI)
- CVP/VXML application control
- Other considerations

  Can mix H.323 and MGCP on the same GW (not on same voice port)

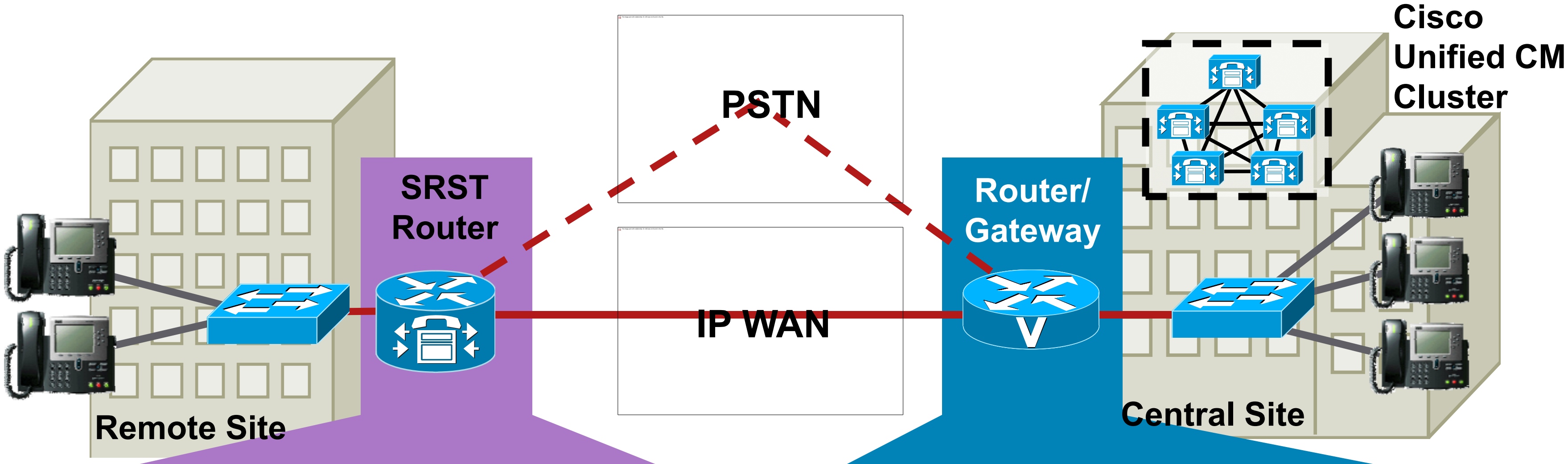
  H.323 dial-peers are needed anyway for MGCP GW Fallback

Protocol and Platform Summary

Gateway Platform	Line Side	Trunk Side		
	SCCP (FXS)	H.323	SIP	MGCP (CUCM)
VG224	Yes	Yes	Yes	Yes
VG248	Yes	No	No	No
1751/60	No	Yes	Yes	Yes
1800	Yes*	Yes	Yes	Yes*
2600XM, 2691	No	Yes	Yes	Yes
2800	Yes	Yes	Yes	Yes
3700	No	Yes	Yes	Yes
3800	Yes	Yes	Yes	Yes
5x00	No	Yes	Yes	No
7x00	No	Yes	Yes	No
Cisco Catalyst 6K CMM	No	Yes	Yes	Yes

Gateways

Protocol and Platform Recommendations



- **H.323, SIP, MGCP fallback to H.323**
- **Standalone, Router-integrated**
- **Platforms:**
  - 17XX, 18XX
  - 26xx, 28XX
  - 37xx, 38xx

- **MGCP, SIP, H.323**
- **Standalone, Router-integrated**
- **Platforms:**
  - WS-X6608, CMM
  - 26XX, 28XX
  - 37XX, 38XX

TECVVT-1000
14400_04_2008_c1

Cisco Unified Border Element

(Formerly Cisco Multi-Service IP-to-IP Gateway)

- A Border Element is an essential component that allows the network to provide services for interconnecting IP based communications
- Examples: SIP Trunk interconnects; business-to-business CTS
- Co-existence with other features such as MTP, Unified SRST, TDM GW

Demarcation

Fault Isolation
Call Accounting
Topology Hiding

Session Management

Call Admission Control
IP QoS/SLA

Cisco Unified Border Element

H.323 to SIP
SIP to SIP
SIP Profiles and Variants

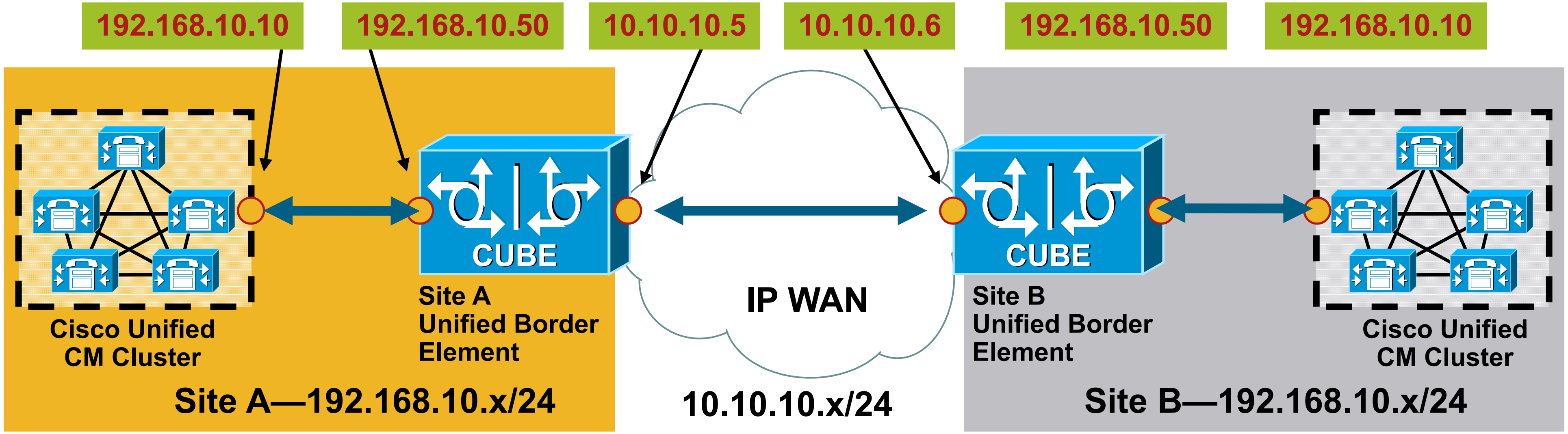
Inter-working

Cisco IOS Firewall Integration
RTP Media Validation
Signaling Protection

Security

Cisco Unified Border Element

Address Hiding

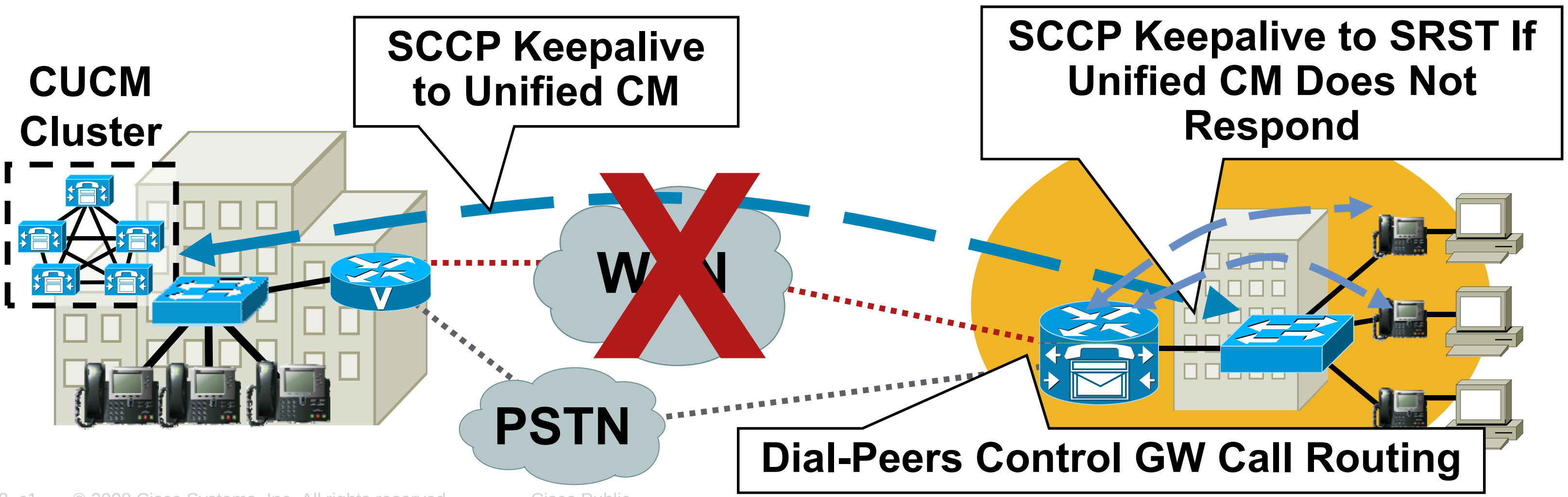


- Within the same company—between departments having overlapping addresses
- Integrating new acquisition into the existing voice network

IP Phone Failover

Unified Survivable Remote Site Telephony (SRST)

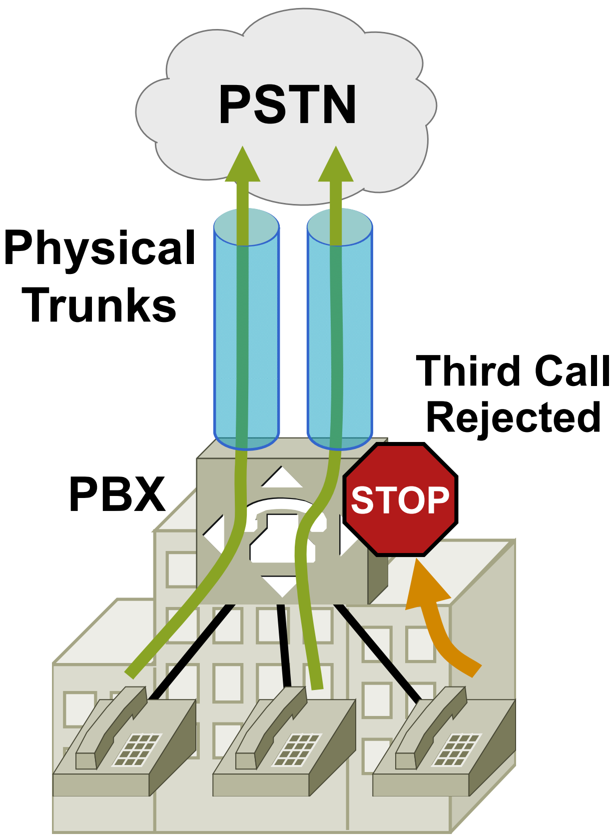
- Phones have list of backup Cisco Unified CMs to fail over to in case of no response to keepalives
  - SRST is the “Cisco Unified CM of last resort” in the phone list
- SRST only controls IP phone connectivity—it does not provide or control GW connectivity or availability
- PSTN GW connectivity during failure modes:
  - POTS/VoIP dial-peers
  - MGCP GWs requires the MGCP GW failover feature
  - Calls from IP phones (under SRST) access the dial-peers to route calls



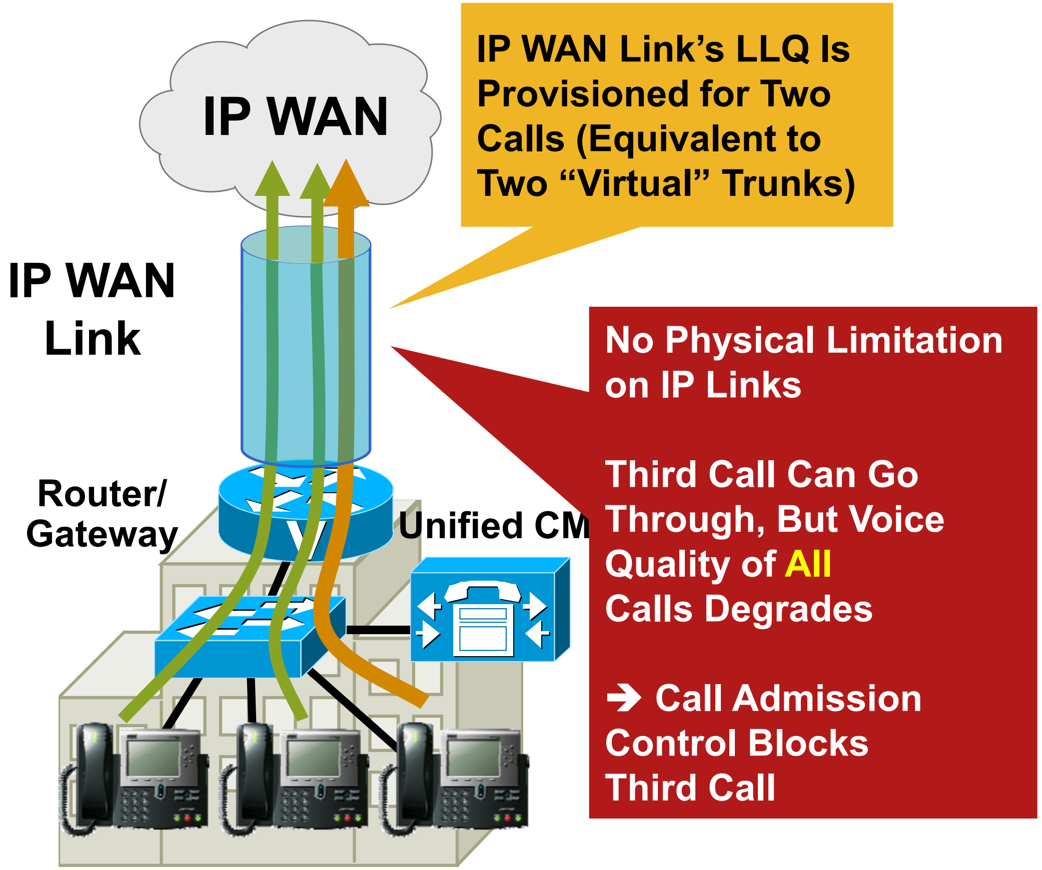
Introduction

Why Is Call Admission Control (CAC) Needed?

Circuit-Switched Networks



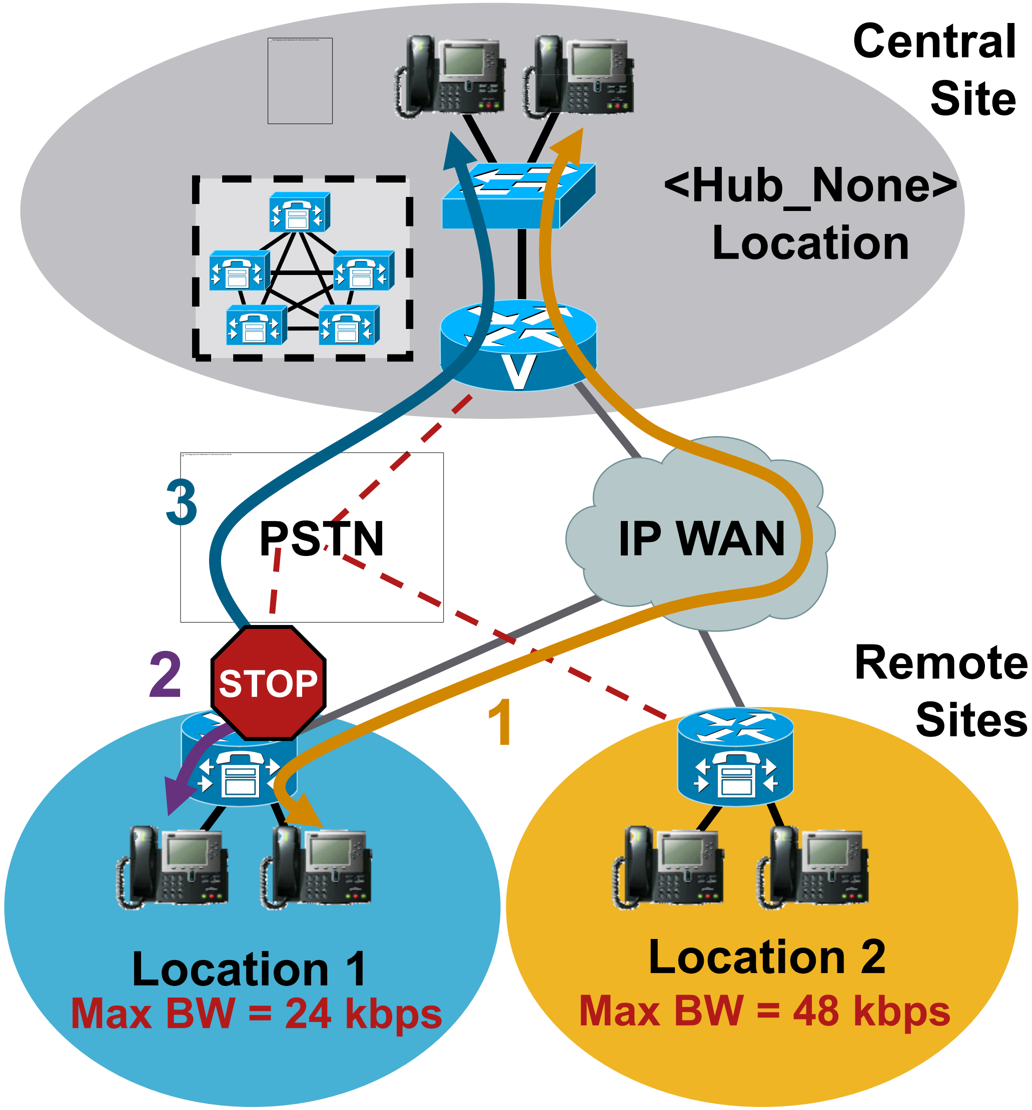
Packet-Switched Networks



Cisco Unified CM Static Locations

Concept

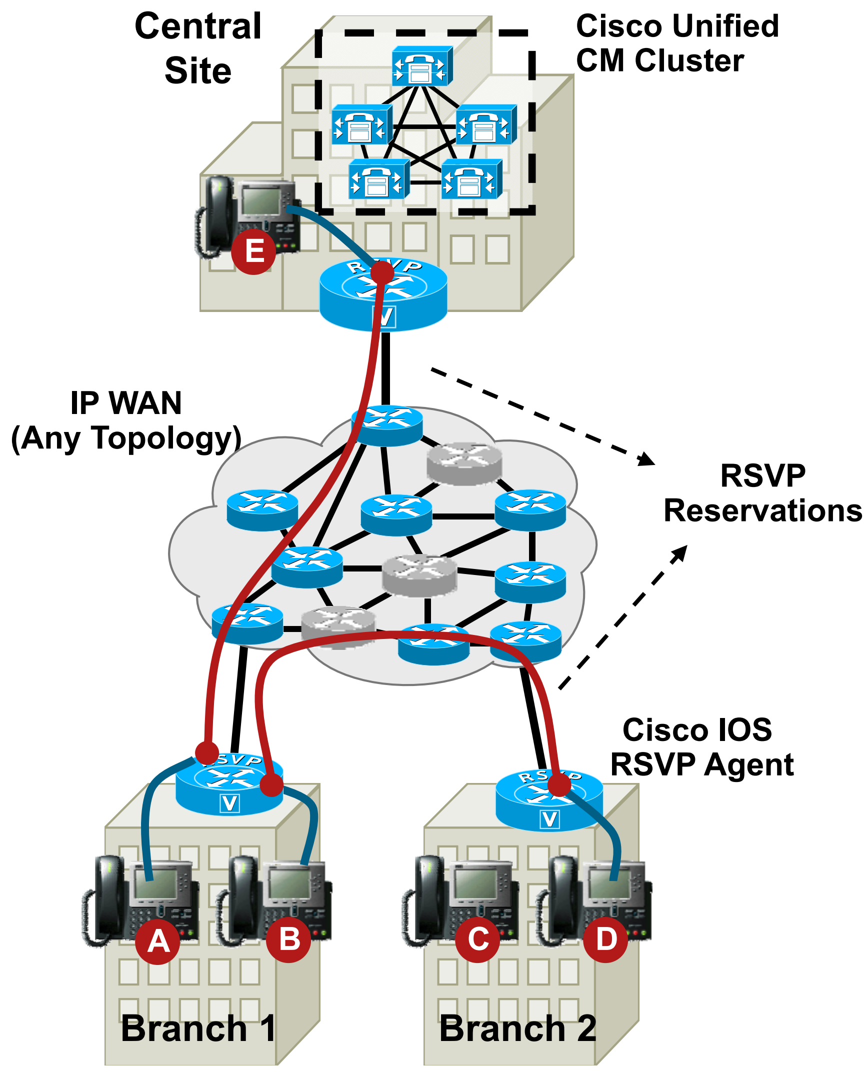
- Prevent WAN link over-subscription by limiting voice bandwidth
- Assign bandwidth limit for voice per location
- When resources are insufficient, phone gets fast-busy tone and a message is displayed
- If Automated Alternate Routing (AAR) is enabled, the call is automatically rerouted across the PSTN



Cisco Unified CM RSVP-Enabled Locations

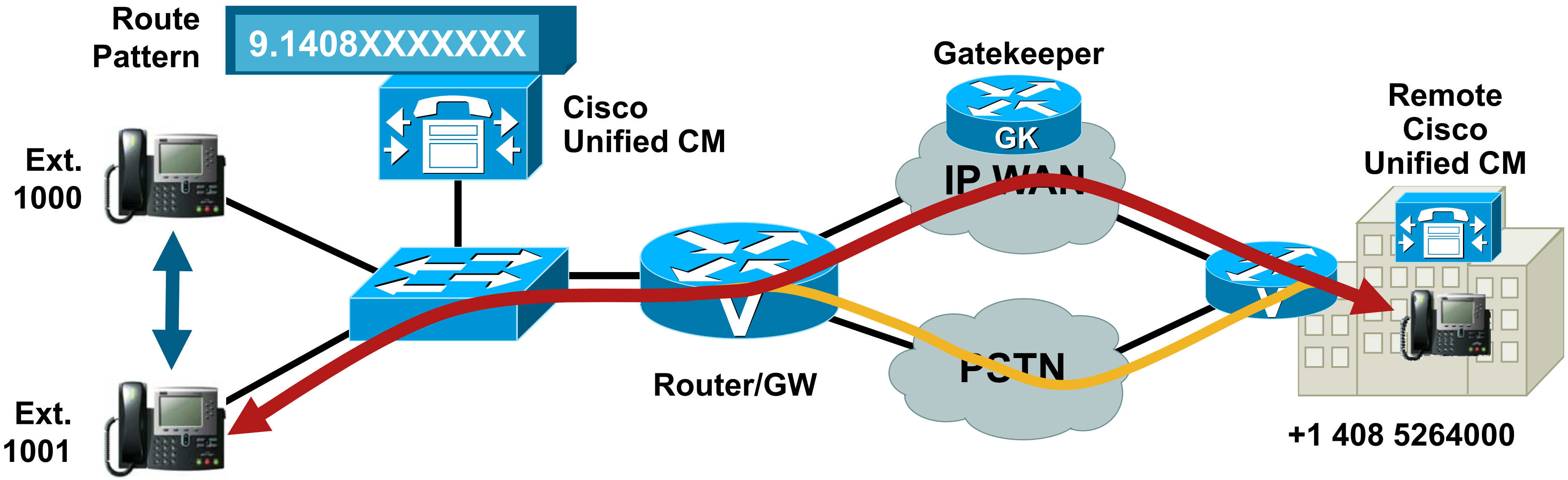
The RSVP Agent Concept

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



Dial Plan

The “IP Routing” of IP Telephony



Cisco Unified CM Routes Two Basic Call Types:

- **On-Cluster Calls:** Destination Directory Number (DN) is registered with Cisco Unified CM; DNs are considered “**internal**” routes
- **Off-Cluster Calls:** Destination Number is **not** registered with Cisco Unified CM; Route Patterns are configured to allow for “**external**” routes
- **Alternate Routes:** Allow On-Cluster and Off-Cluster calls to attempt alternate paths to destination (e.g. IP WAN not available, go through PSTN)

Cisco Unified CM Call Routing Logic

Commonly Used Wildcards

Delimiter (Does Not Match Any Digits)—Used for Discarding

Range of Digits (Between 2 and 9)

Single Digit Between 0 and 9

9 . [2-9] XXXXXX

One or More Occurrences of Digits Between 0 and 9

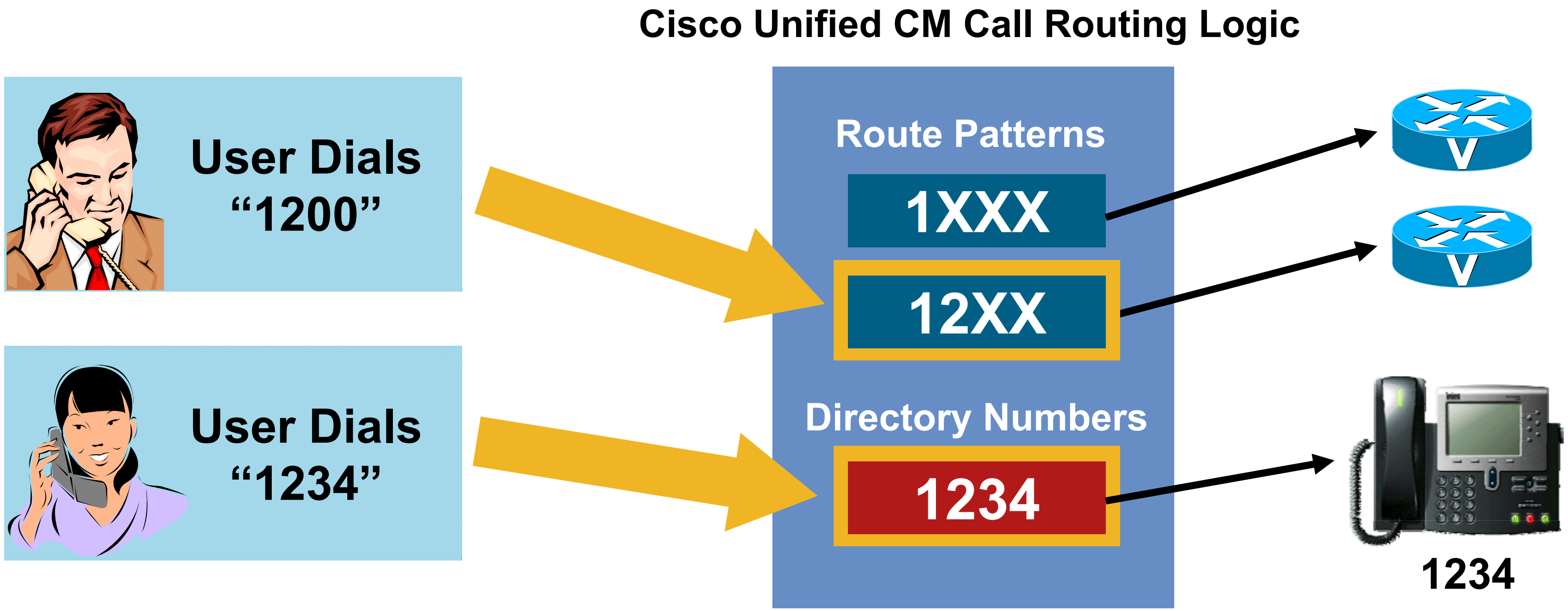
The "#" Digit—Used to Avoid InterDigit Timeout

9.011! #

9. @

A Macro That Enters the Whole North American Numbering Plan into Cisco Unified CM
(or a Different Country's Numbering Plan If Using the International Dial Plan Tool)

Cisco Unified CM Call Routing Logic
Basic Principle

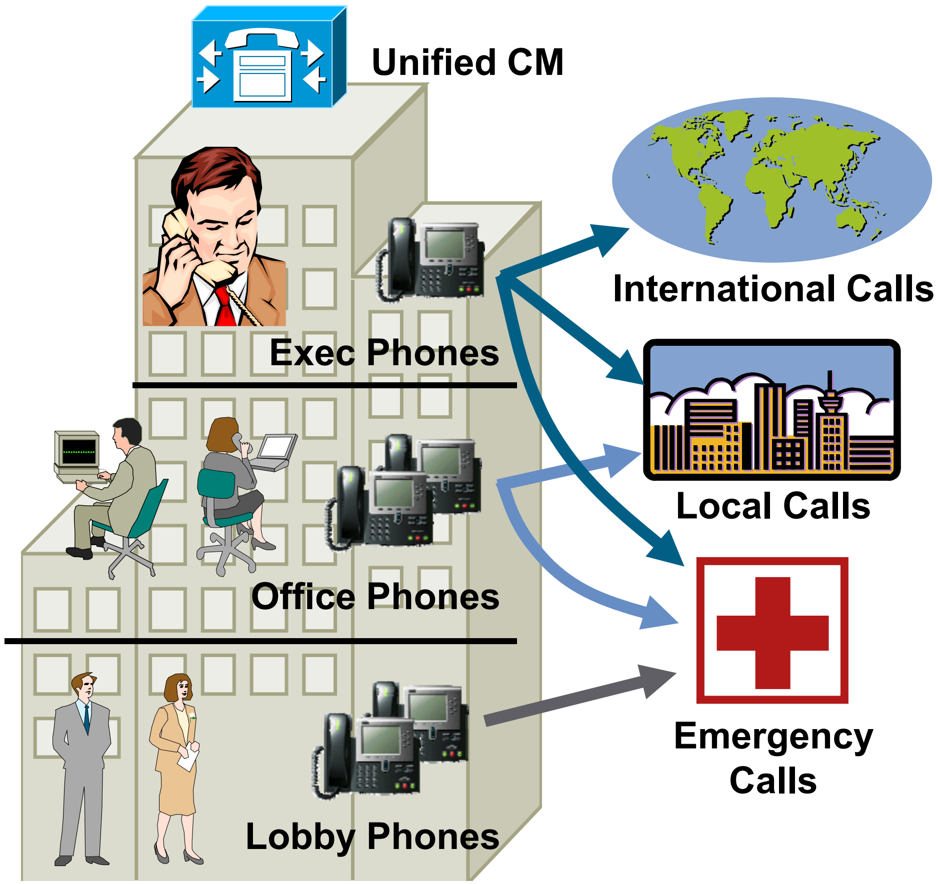


- Cisco Unified CM matches the most specific pattern (longest-match logic)
- For call routing, an IP phone directory number acts as a 'route pattern' that matches a single number

Building Classes of Service

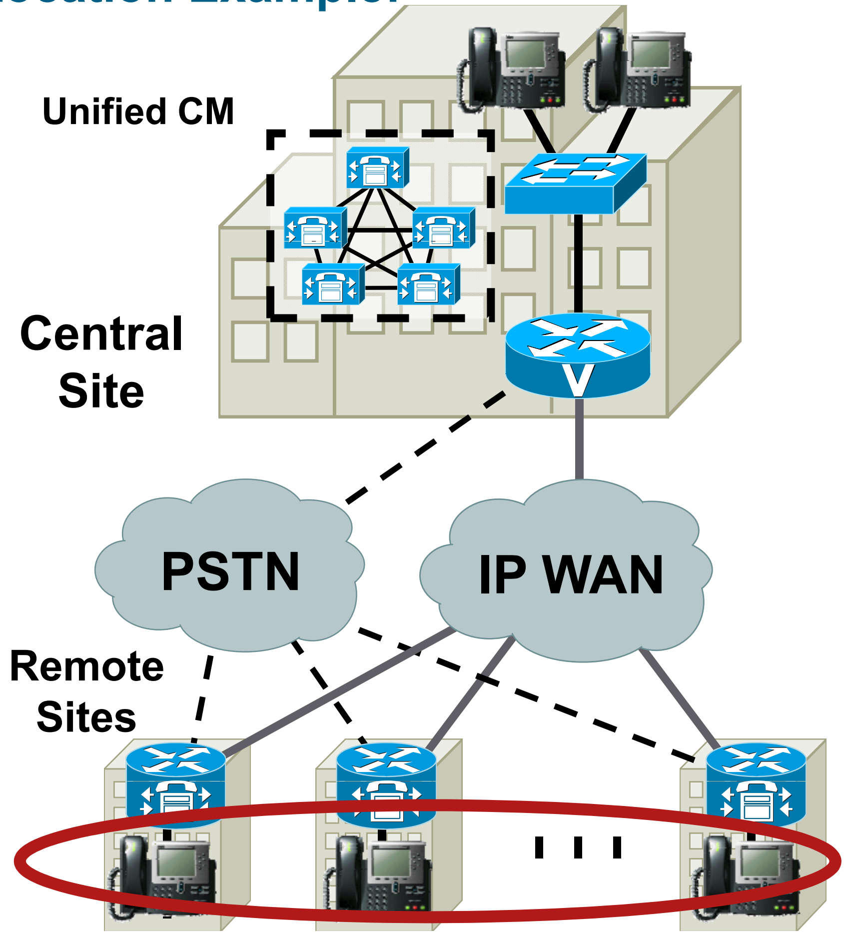
Routing by User Class or Location

User Class Example:



Define Calling Capabilities Based on Role of Directory Number

Location Example:



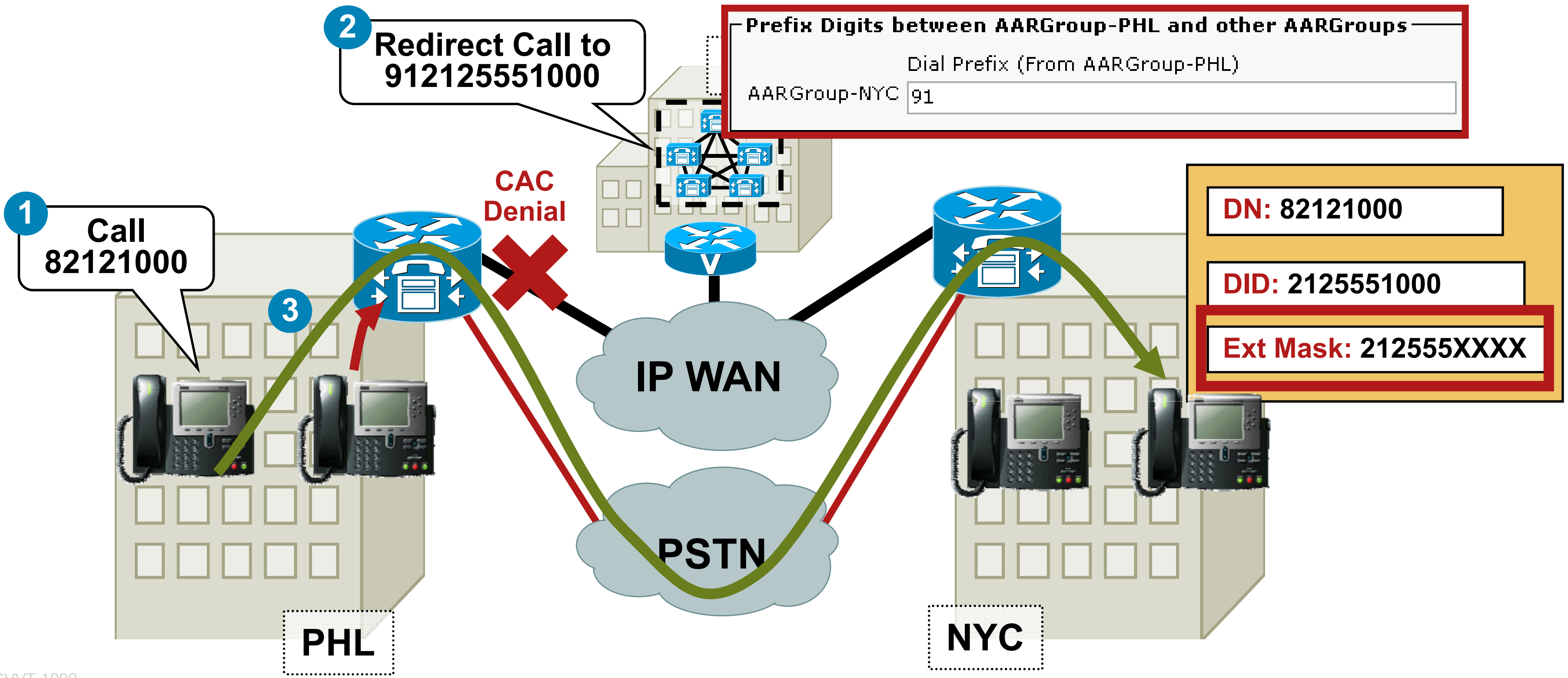
Instruct These Phones to Use Their Local Gateway for PSTN Access

Alternate Routing

Internal Routes: Automated Alternate Routing (AAR)

Concept

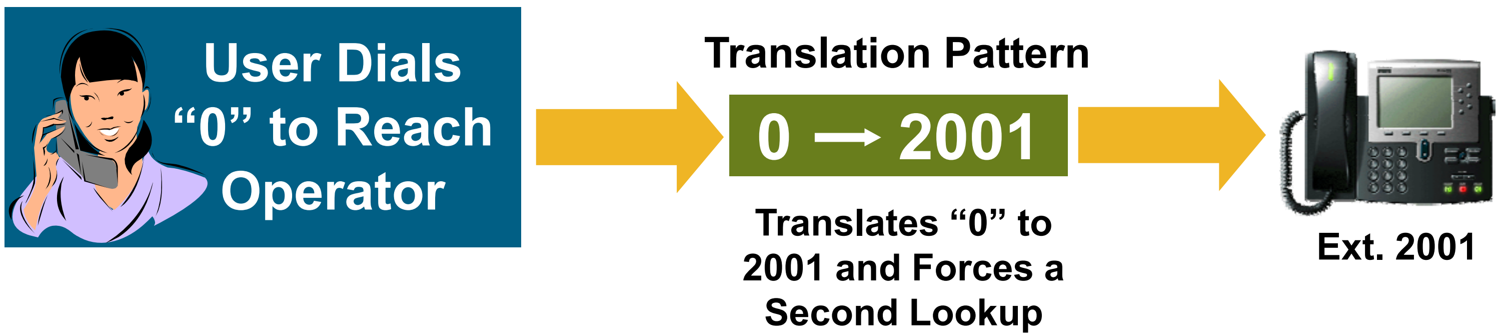
- Reroutes calls to registered DNs when call is rejected due to Call Admission Control
- Unified CM reroutes the call using number specified in “AAR Destination Mask” or “External Phone Number Mask” of the **called** party
- Prefixes “AAR/External Phone Number Mask” with digits in the AAR Group



Translation Patterns

Key Concept

- Looks like a route pattern, allows digit manipulation
- Instead of sending calls outside via a route list, forces second lookup in Cisco Unified CM, using a (possibly different) calling search space



Dial Plan

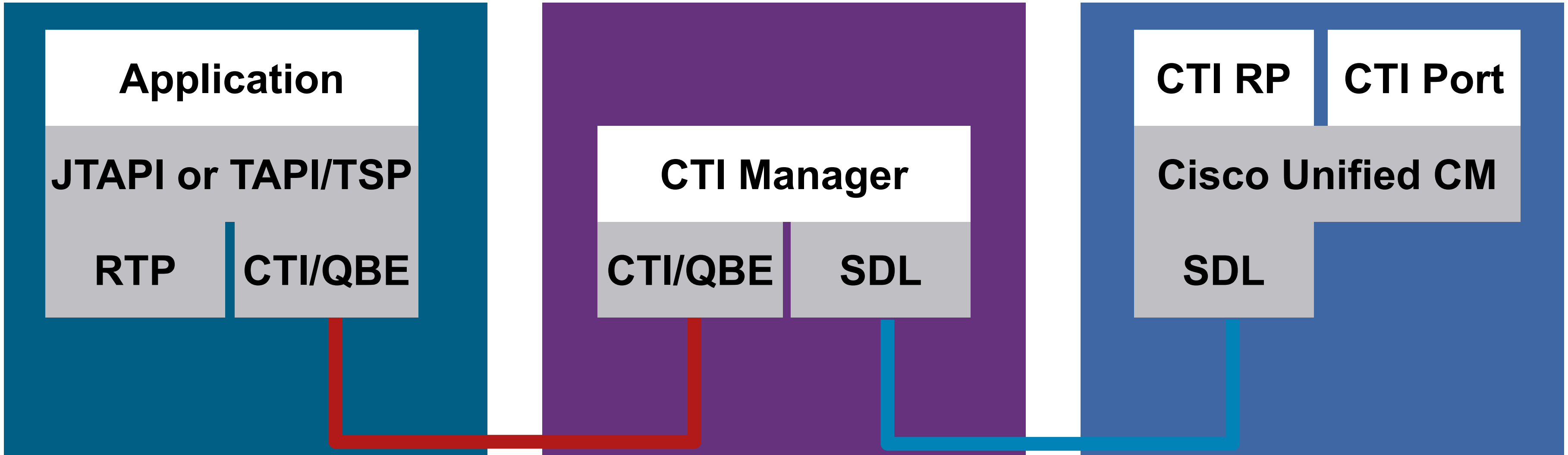
General Recommendations

- Keep it simple
- Standard naming conventions
- Plan for future growth

Agenda

- Introduction
- Network Infrastructure
- Unified Communications Infrastructure
- **Unified Communications Applications**
- Security

(J)TAPI and CTI Concepts
Functional Blocks



Application

Uses the TAPI TSP or JTAPI Plug-in to Communicate with the CTI Manager Using CTI/QBE over TCP/IP

CTI Manager

Acts as a Broker Between the Application and the Cisco Unified CM Cluster

Cisco Unified CM

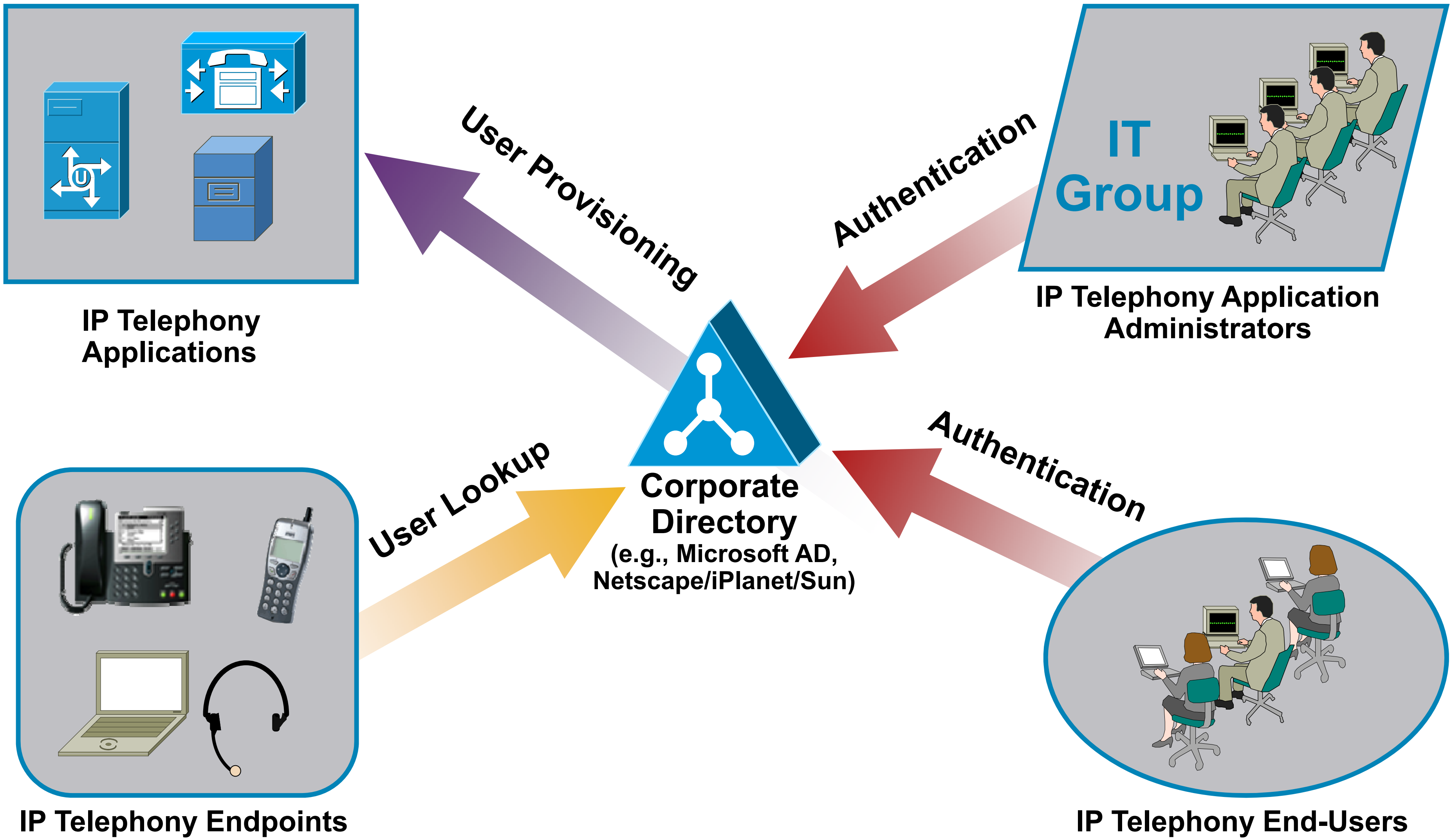
Hosts the CTI Devices and Provides Device Redundancy and Call Control

TAPI = Telephony Application Programming Interface
JTAPI = Java Telephony Application Programming Interface
TSP = Telephony Service Provider
CTI = Computer Telephony Integration

QBE = Quick Buffer Encoding
SDL = Specification and Description Language
RP = Route Point

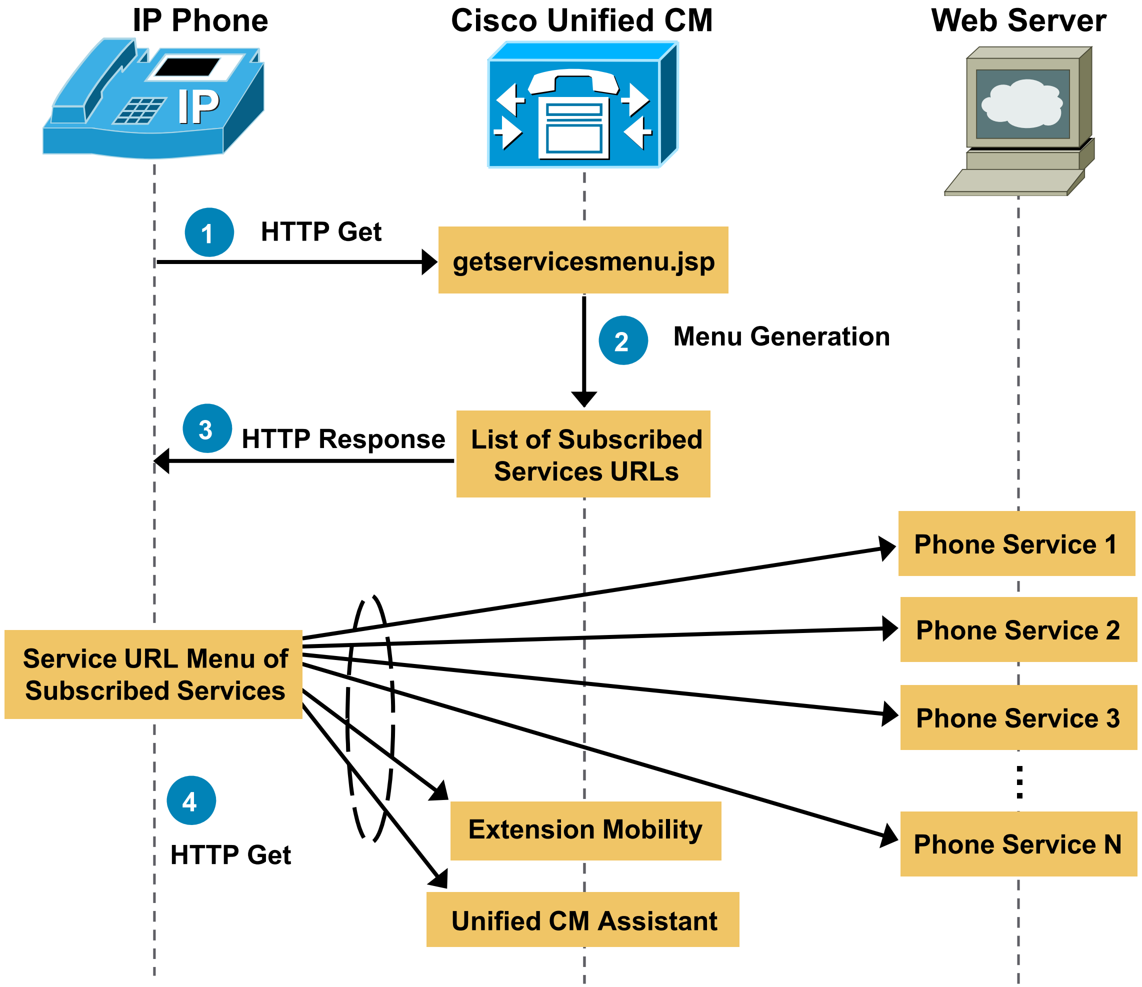
LDAP Directories

What Does “Directory Integration” Mean to You?



IP Phone Services

Architecture—User Initiated



1. User pushes "Services" button which generates HTTP Get
2. Getservicesmenu.jsp script generates menu of phone's subscribe service URLs
3. IP Phone Services returns list of subscribed services via HTTP Response
4. User selects IP Phone Service from menu which generates HTTP Get to web server

Presence

What Is Presence?

Information About a Person's Availability and Willingness to Communicate

- Examples of presence in action today

    IM "Buddy List" status indication (Available, Idle, Away)

    "Busy" tone on traditional phone

    Contact Center Agent status

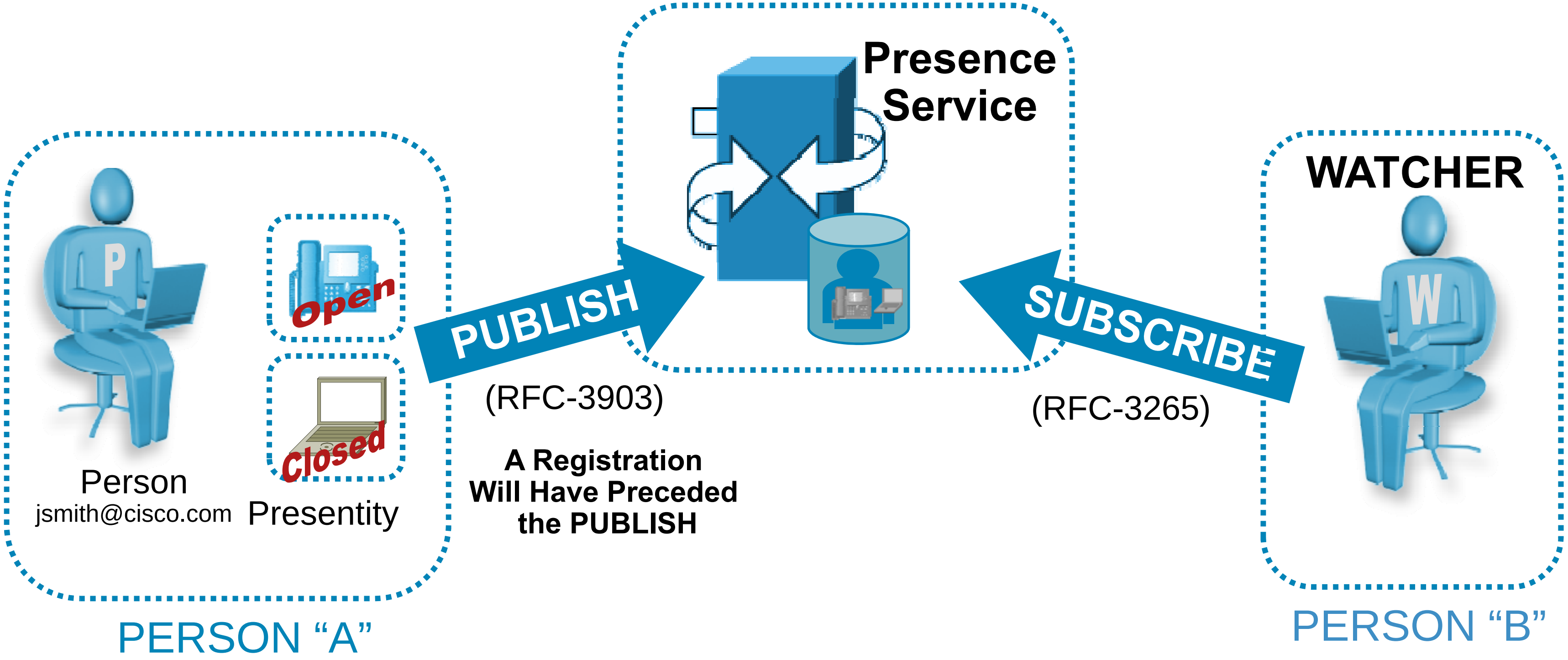
- Publish/Subscribe/Notify

    A Person can publish presence information to other users via a Presence Service

    Users of the Presence Service can subscribe to receive Notification of Status Change of a Person

Presence

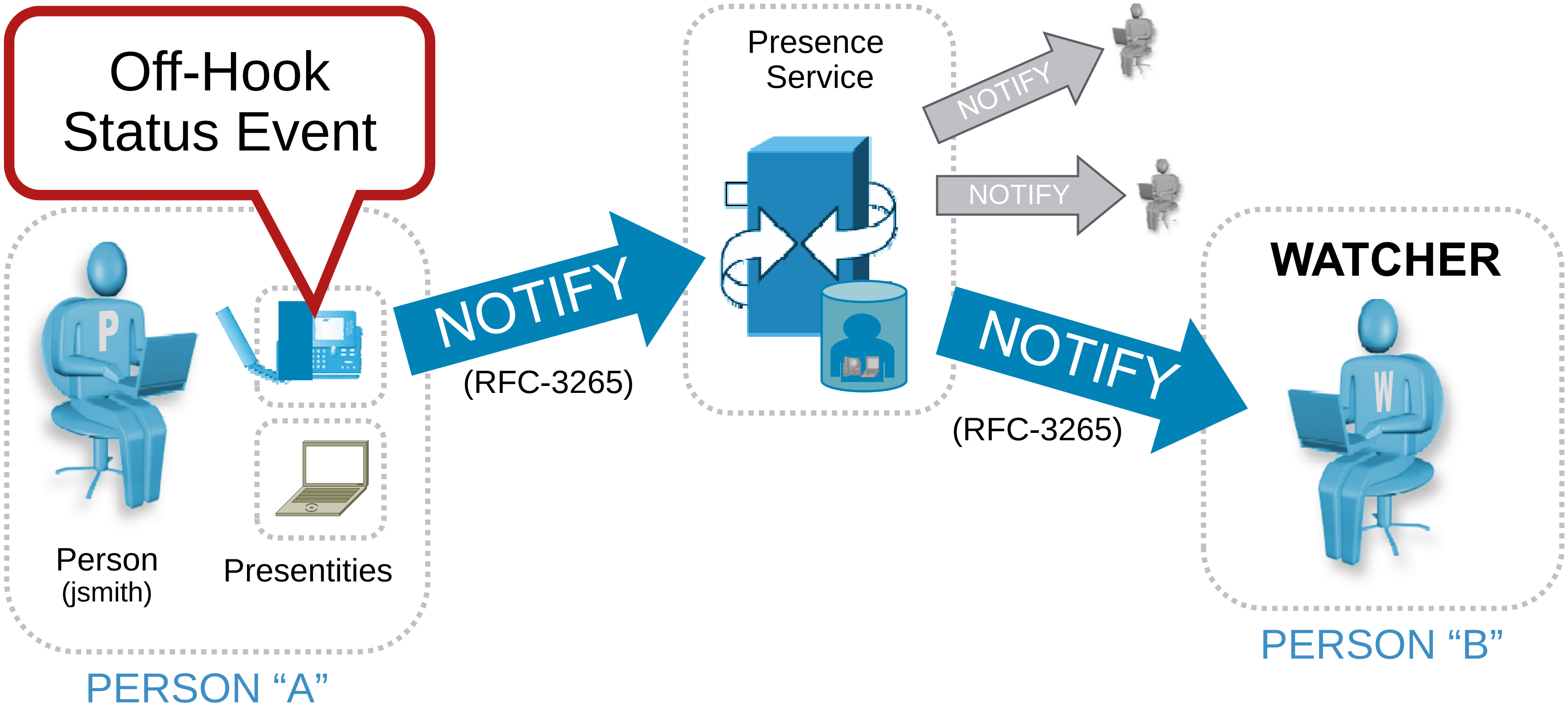
Presentity and Watcher—Publish and Subscribe



- A Person will PUBLISH the status of communication Services/Devices to the PRESENCE SERVICE using their PRESENTITY
- A WATCHER can SUBSCRIBE (for a period of time) to receive updates on status changes for the PRESENTITY
- A WATCHER can (and most likely will) also have a Presentity

Presence

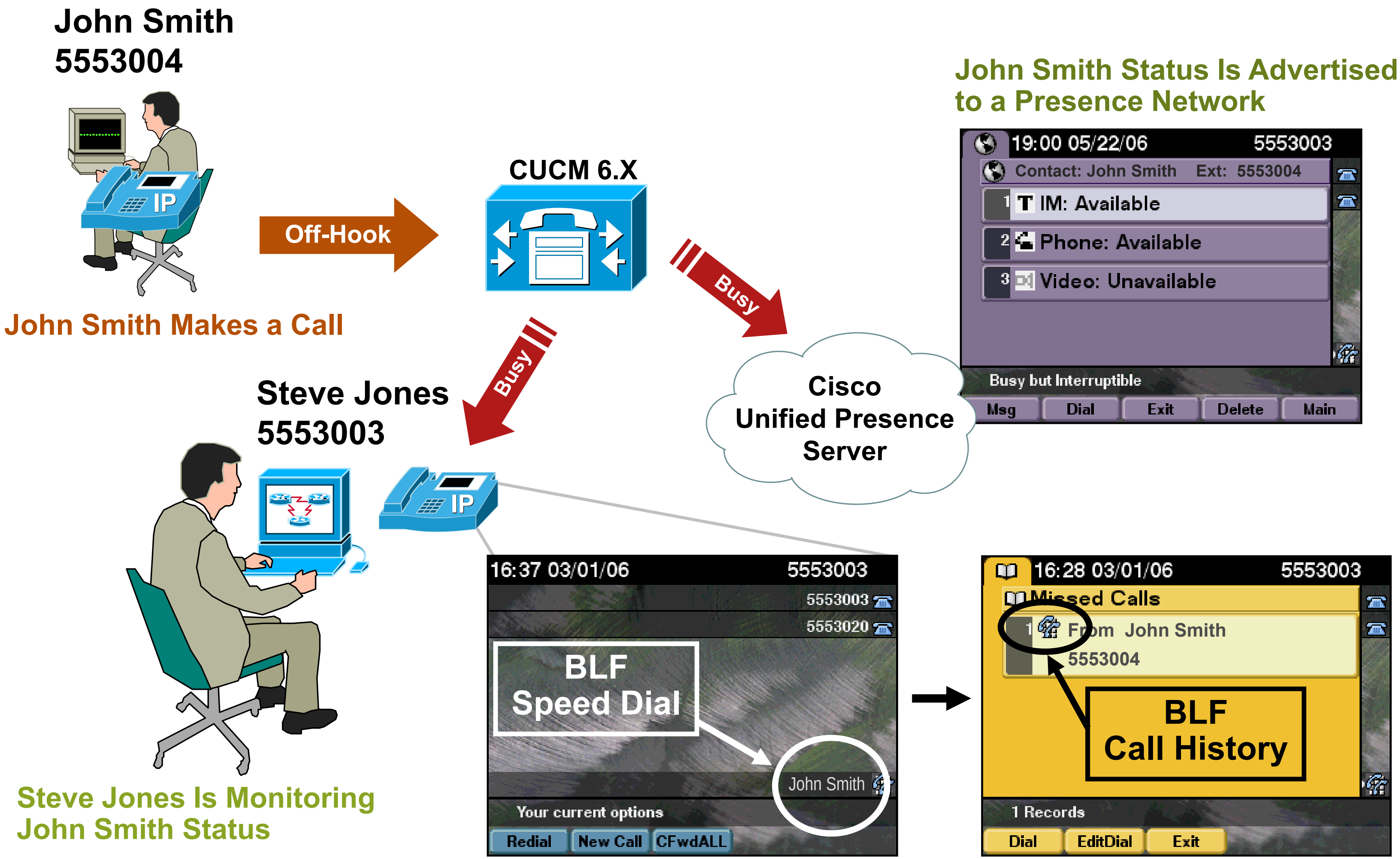
Presentity and Watcher—Notify



- On a Change of status the PRESENTITY is updated on the Presence Service
- The Presence Service will NOTIFY all the subscribers of the PRESENTITY

Presence

“In Action” with CUCM—BLF Speed Dial and Call History



Agenda

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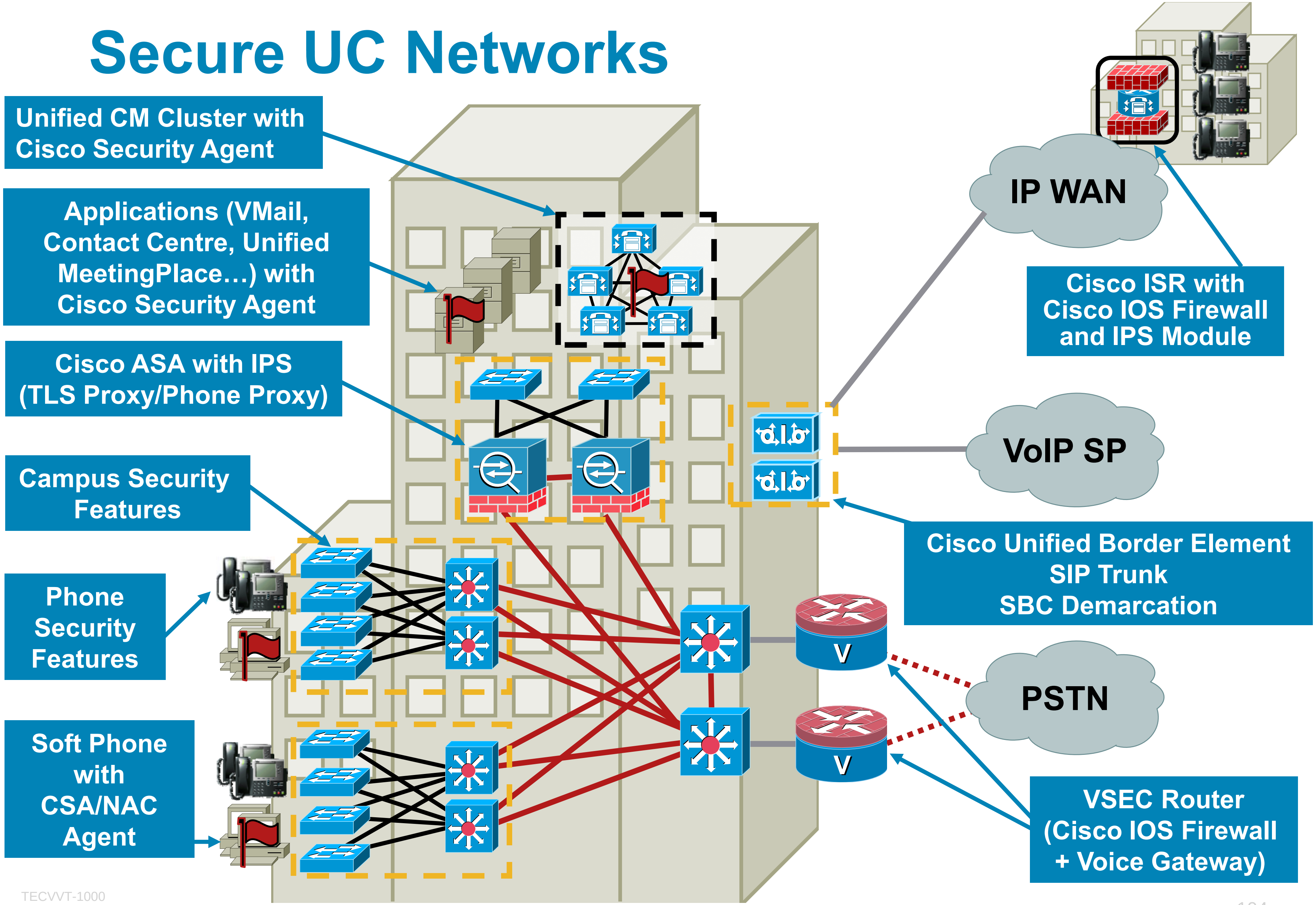
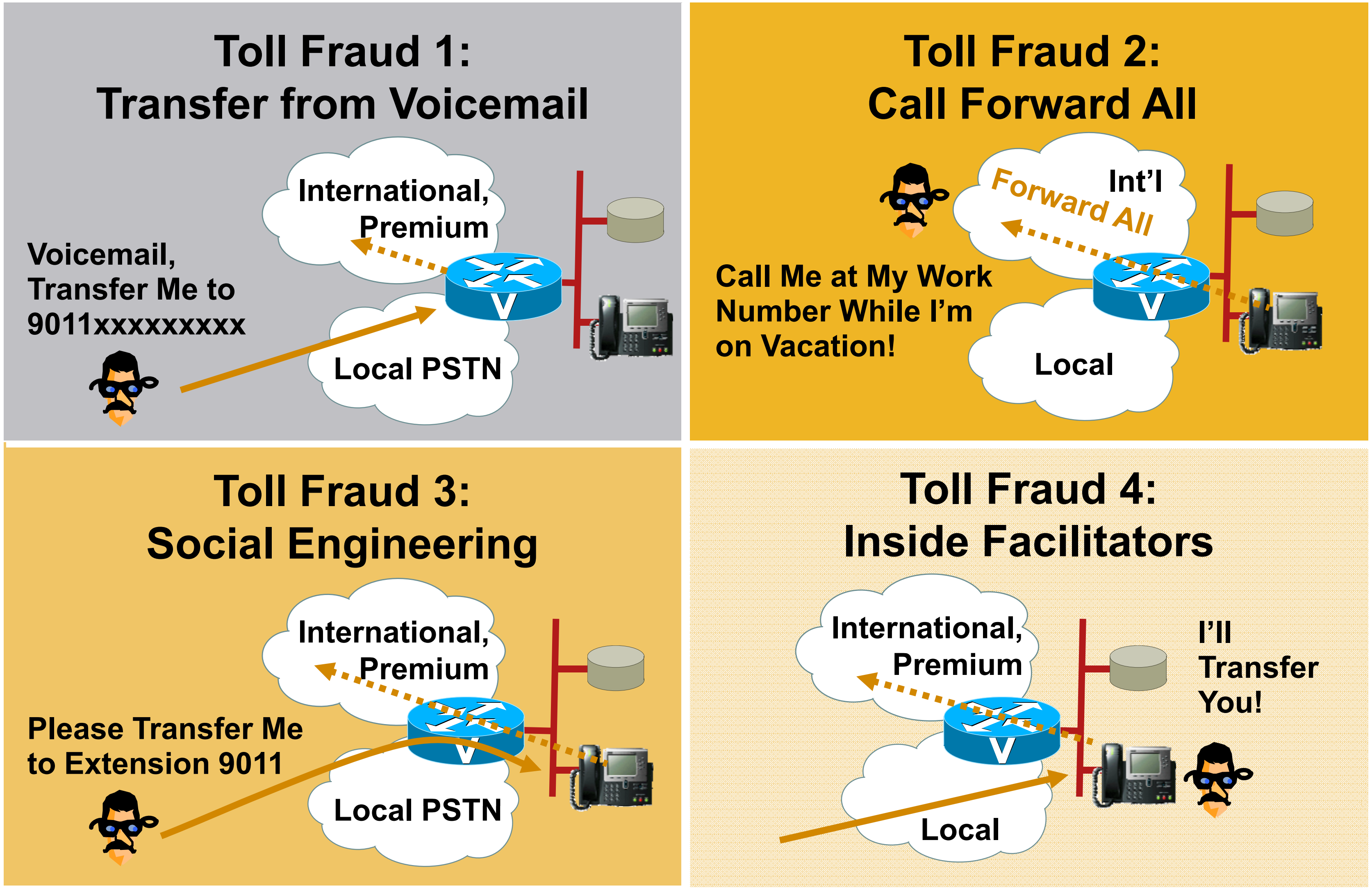
- **Security**

Secure UC Networks



Exploits of Toll Fraud



Toll Fraud Protection

- Class of Service (CoS)

  Calling Search Spaces and Partitions are the mechanism to implement COR in Cisco Unified CM thus restricting certain users/devices to disallow transferring to Long distance or international destinations

- Trunk-to-Trunk Transfer Restrictions

  Prevents users from transferring calls from one external device to another external device; service parameter within Unified CM

- Drop Conference Call Control

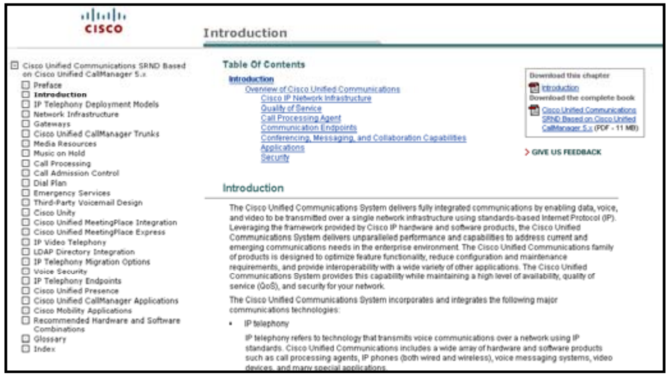
  Conference calls can be dropped when initiator hangs up; mitigates the ability for insiders to circumvent LD restrictions for external callers; service parameter within Unified CM

Q and A

Recommended Reading

- Unified Communications SRND

  http://www.cisco.com/go/srnd



- Cisco Press Book

  Cisco IP Telephony: Planning, Design, Implementation, Operation, and Optimization