Cisco.com

Designing an Enterprise IP Telephony Network

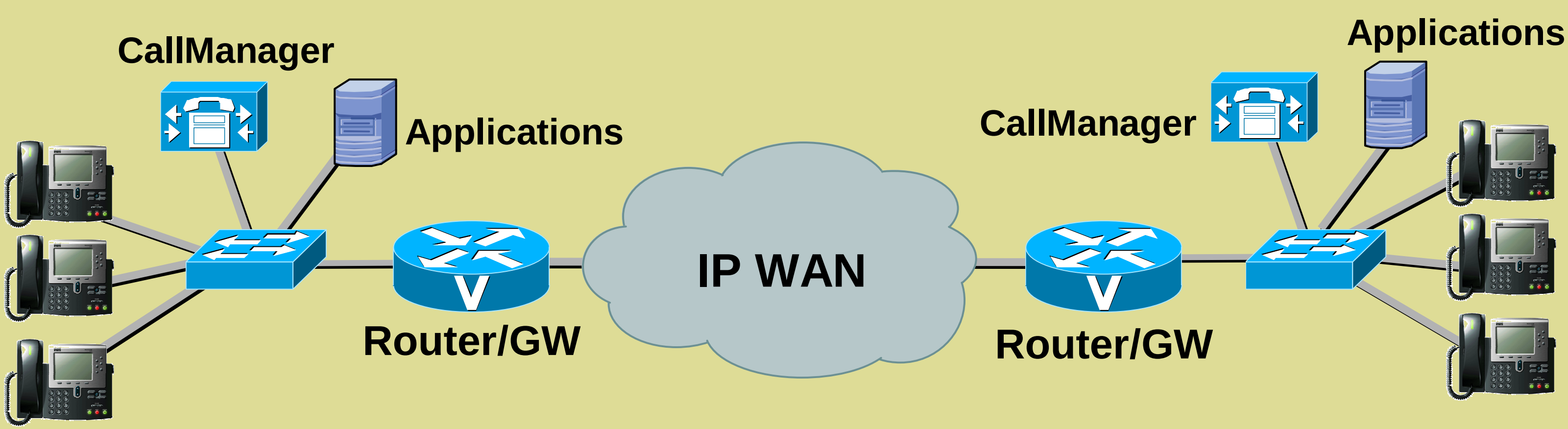
Andres Martinez

Agenda

- Introduction
- Network Infrastructure
- Telephony Infrastructure
- Applications

Focus of This Session

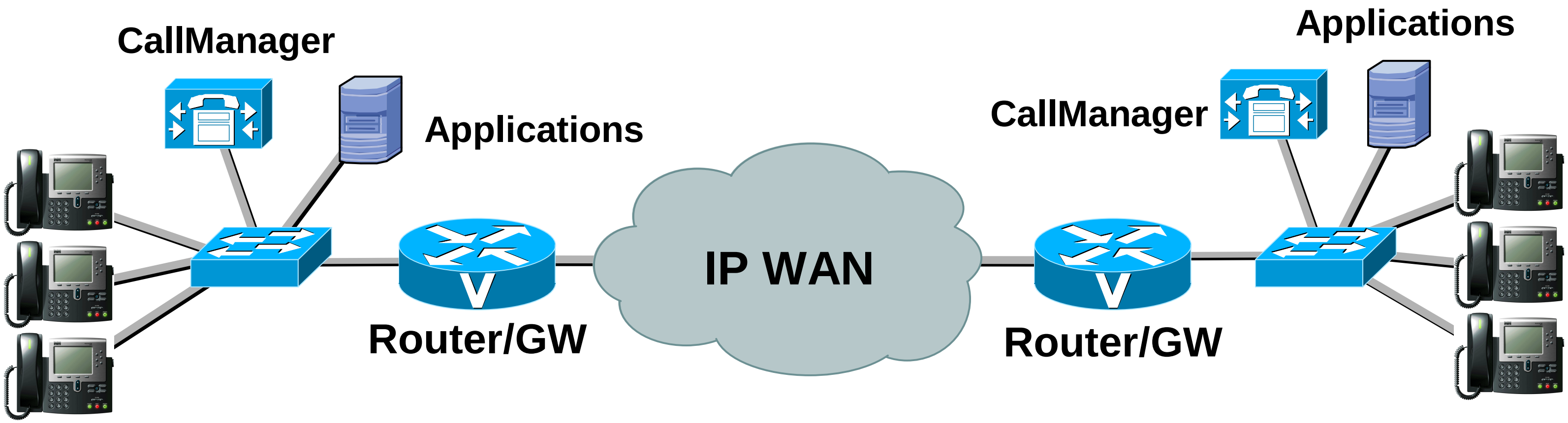
Toll Bypass



End-to-End IP Telephony with Applications

Scope of This Seminar

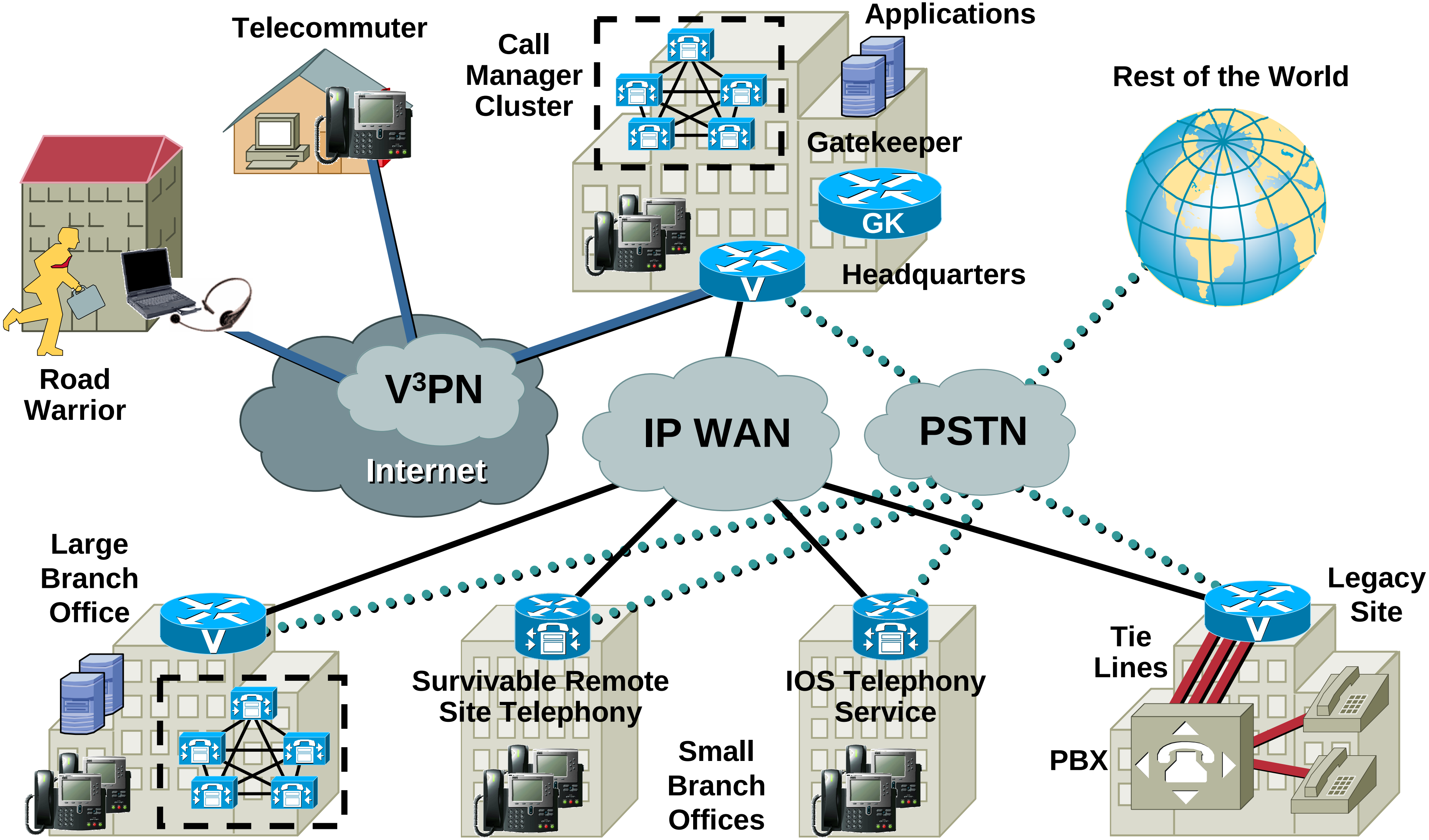
Cisco.com



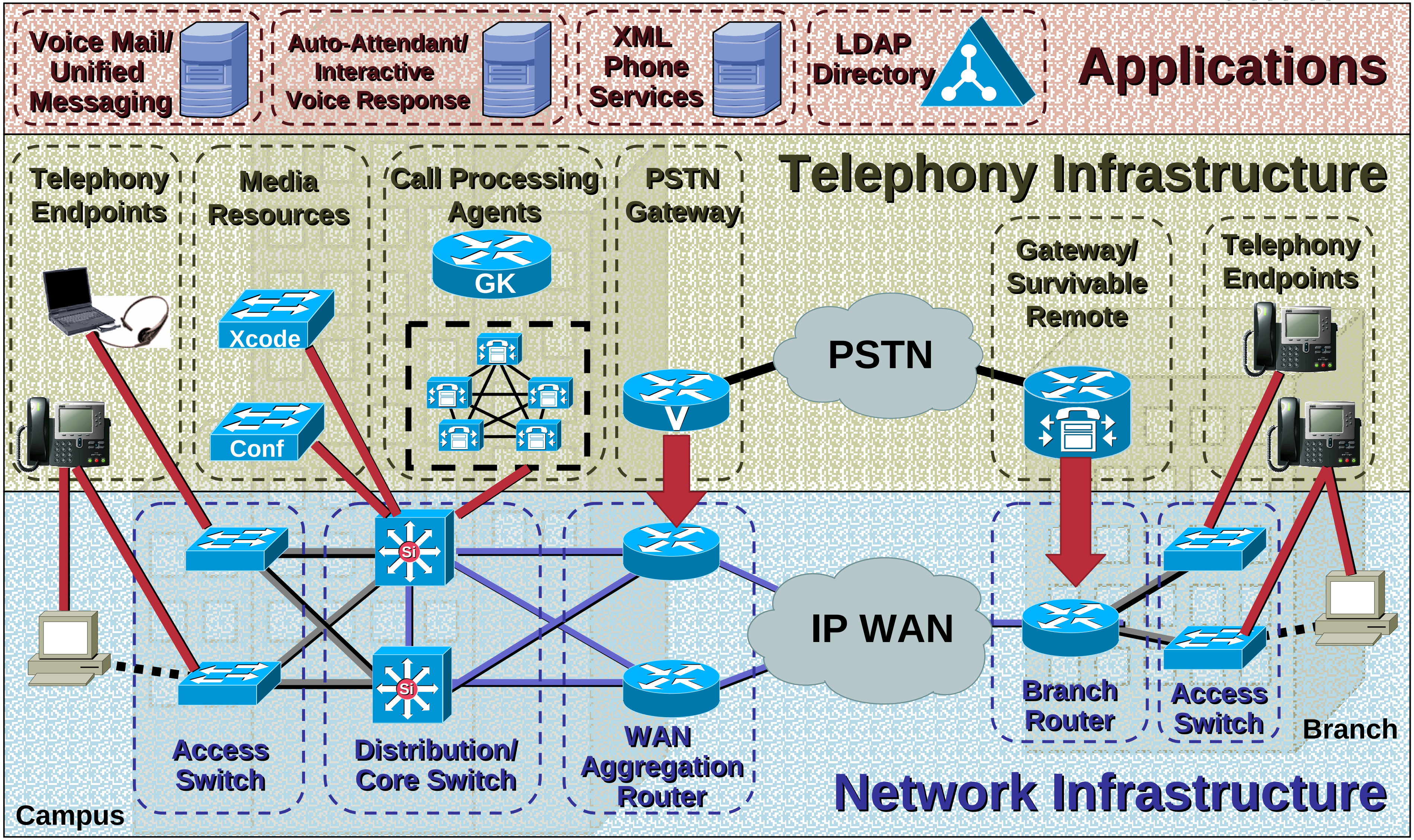
- **Understanding what can be built today**
- **Learning how to build it**
- **To find out more about IP telephony design:**

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

The Big Picture: End-to-End IP Telephony



The Elements of IP Telephony



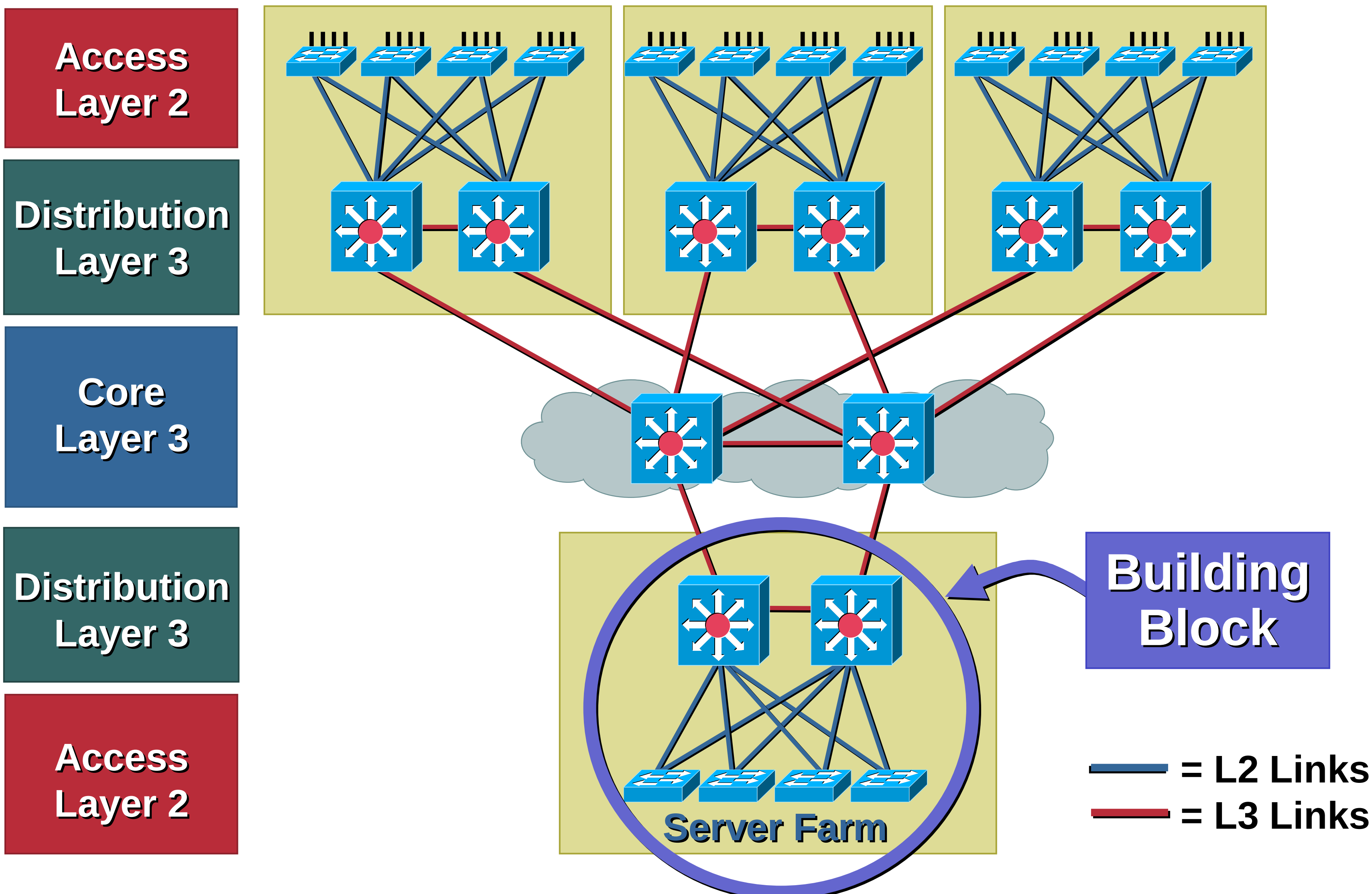
Agenda

- Introduction
- Network Infrastructure
- Telephony Infrastructure
- Applications

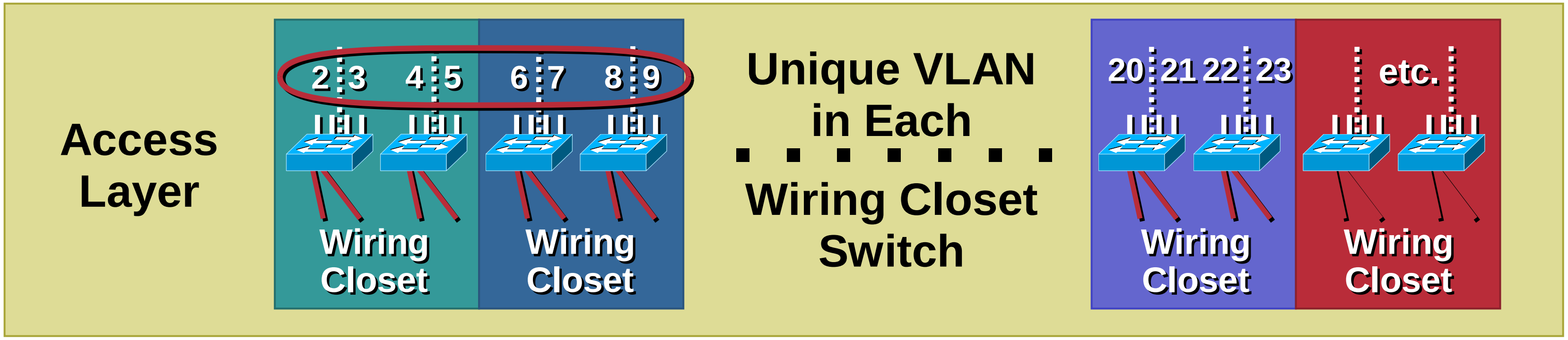
Network Infrastructure Agenda

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

Building a Campus Network
Multi-Layer Network Design



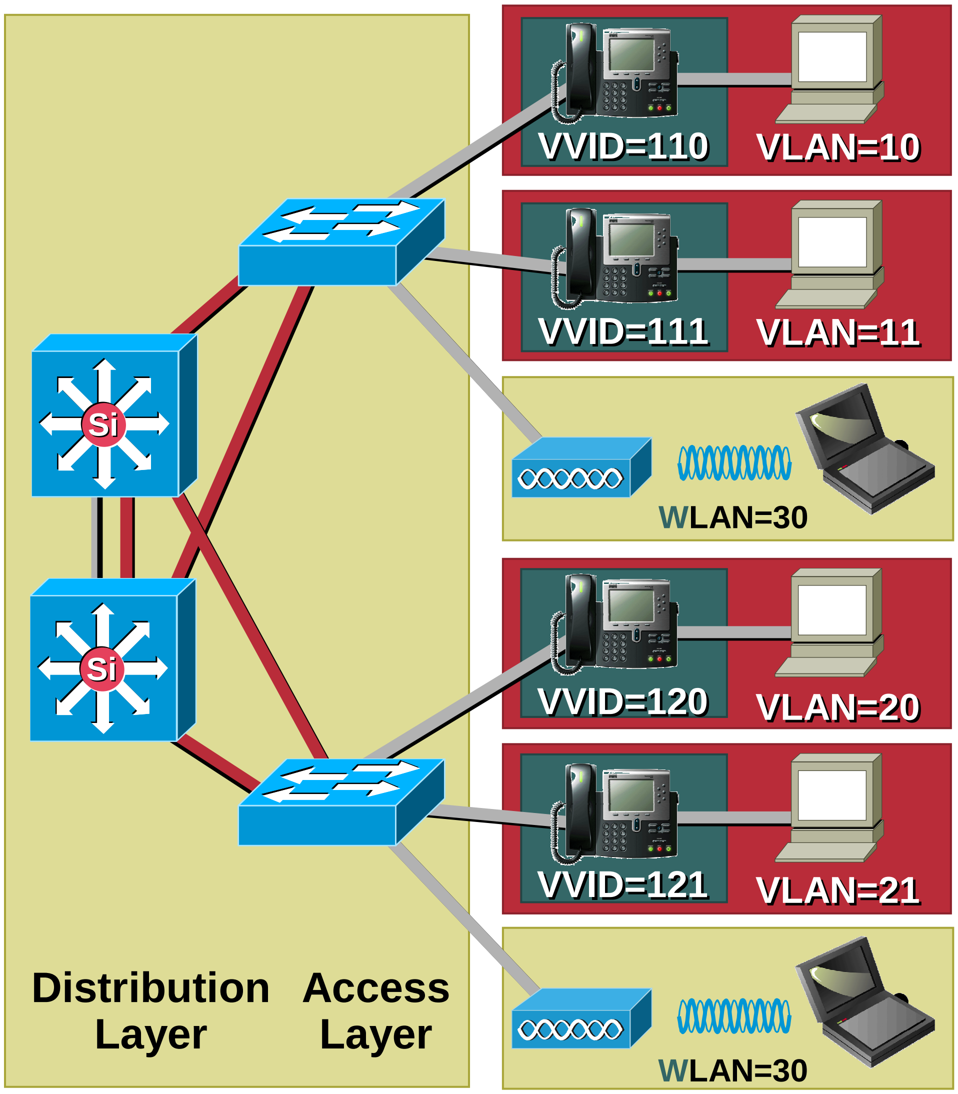
Building a Campus Network
VLAN Model



- **A VLAN = an IP subnet**
- **VLANs do <u>not</u> span different wiring closet switches with a few exceptions**
- **If 2+ VLANs per access switch, load sharing is very easy to achieve**
- **This model achieves fast convergence and high stability**

Overlaying Wireless LANs
VLAN Design

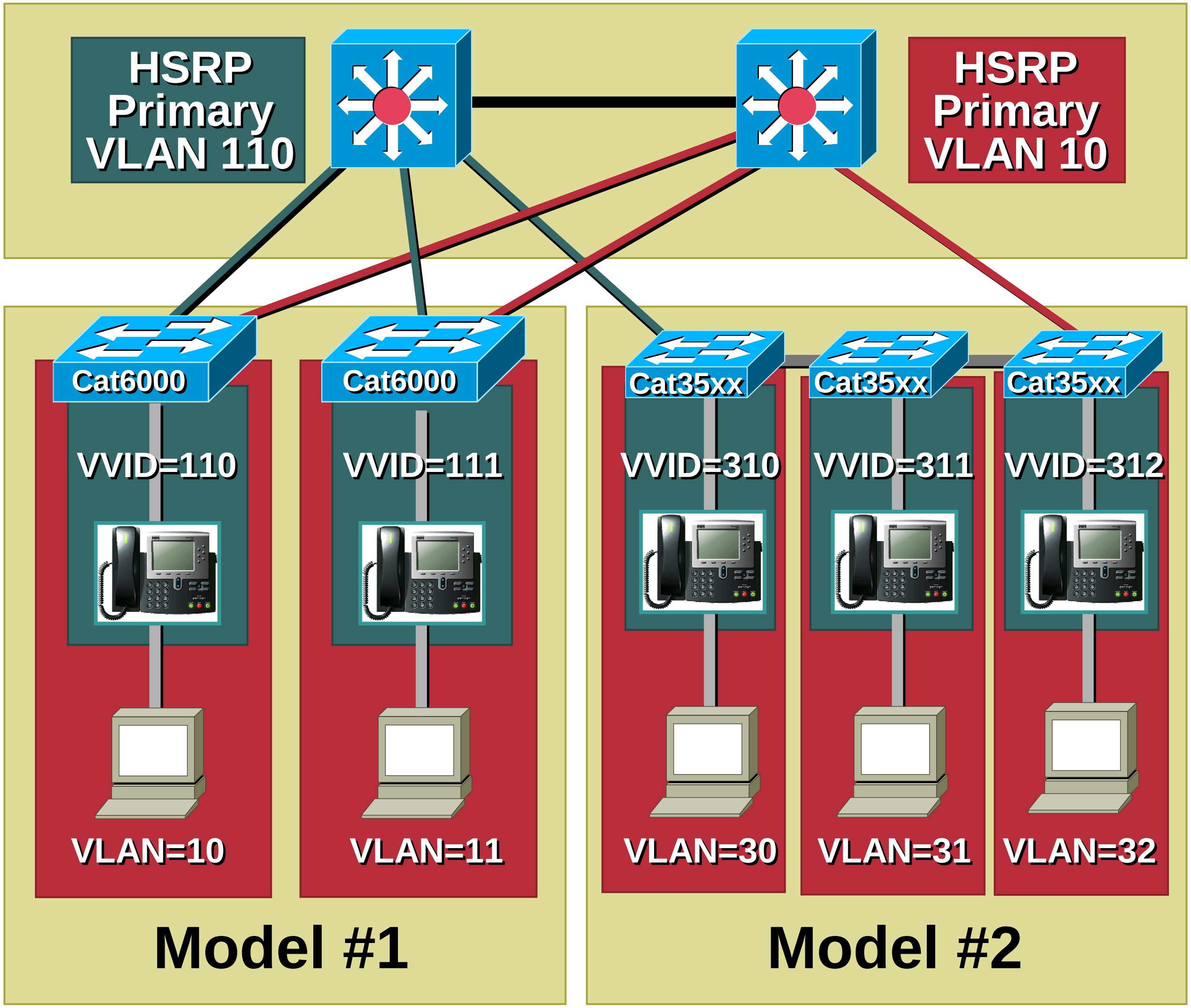
- **Create a single VLAN for the wireless LAN per campus building**
- **Need a L2 link between distribution switches to carry the wireless VLAN**
- **Spanning tree convergence only affects the WLAN**
- **Layer 2 roaming within the building (Layer 2 domain spans multiple wiring closet switches)**



Building a Campus Network
Distribution Layer

Distribution Layer Features:

- Passive interface default
- HSRP, HSRP Track/Preempt
- OSPF/EIGRP:
  - Adjust timers
  - Summary address
  - Path costs

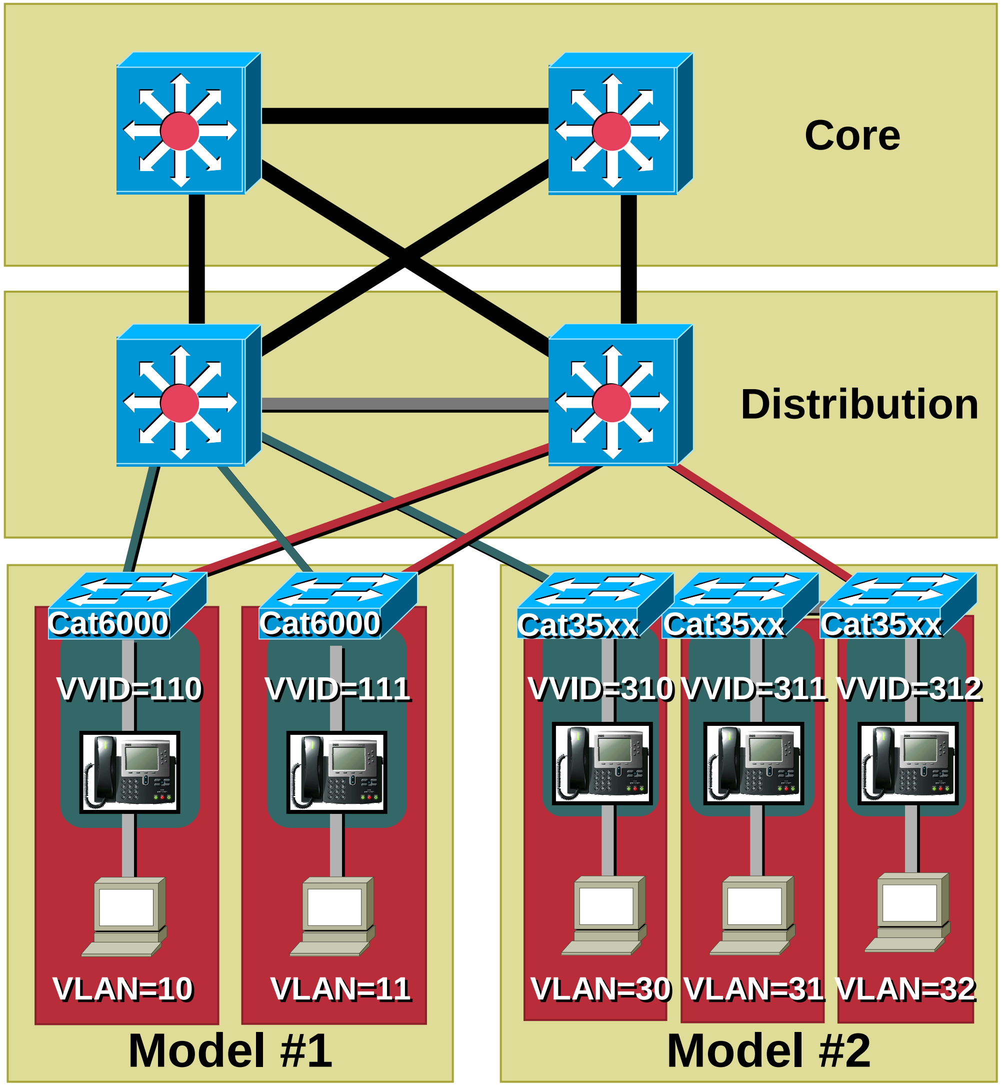


Building a Campus Network
Core Layer

Cisco.com

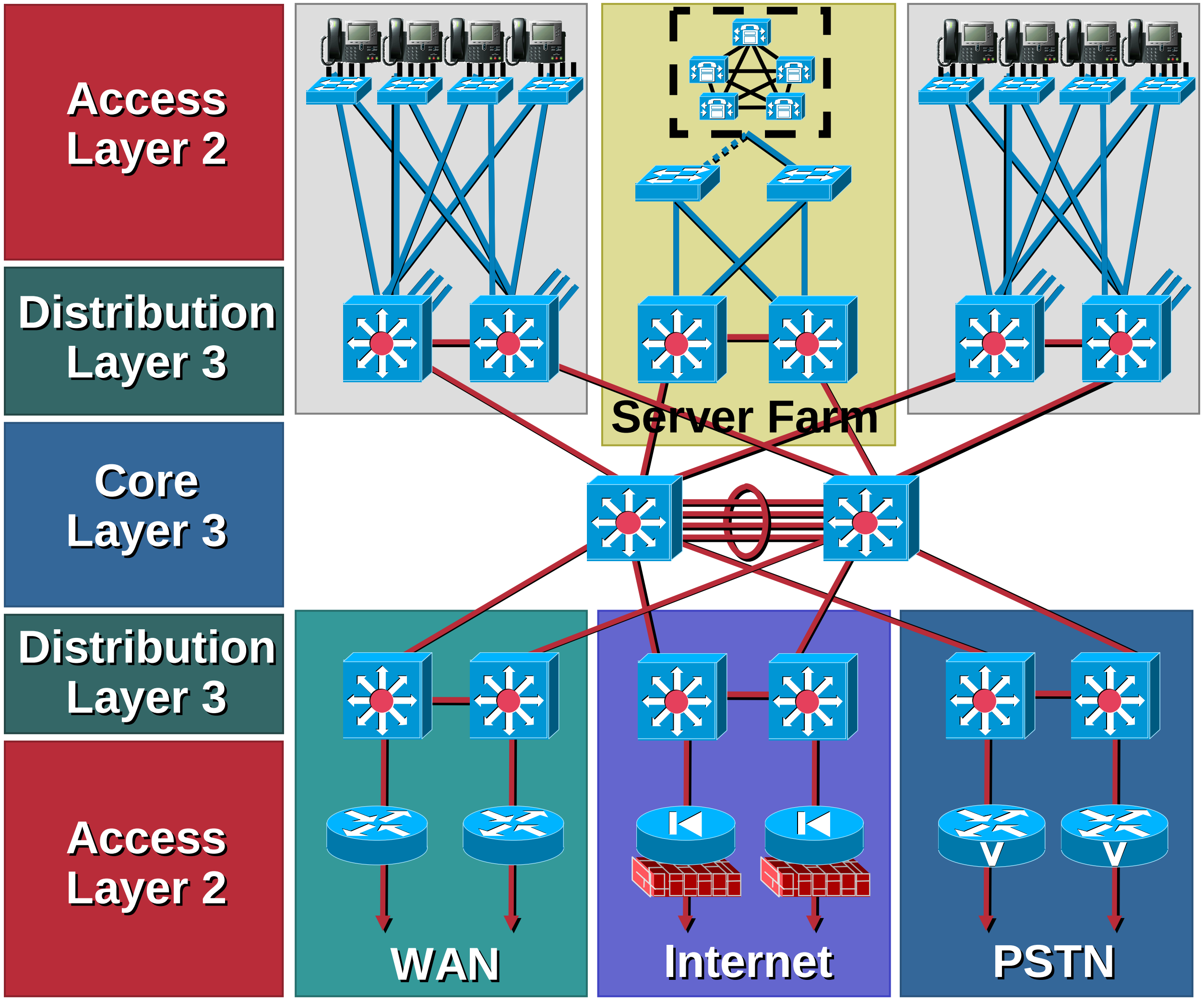
Core Layer Features:

- Each link belongs to its own /30 subnet
- No STP in the core—All routed
- Load balancing to the core/server farm by default
- Tune routing protocol timers for fast convergence

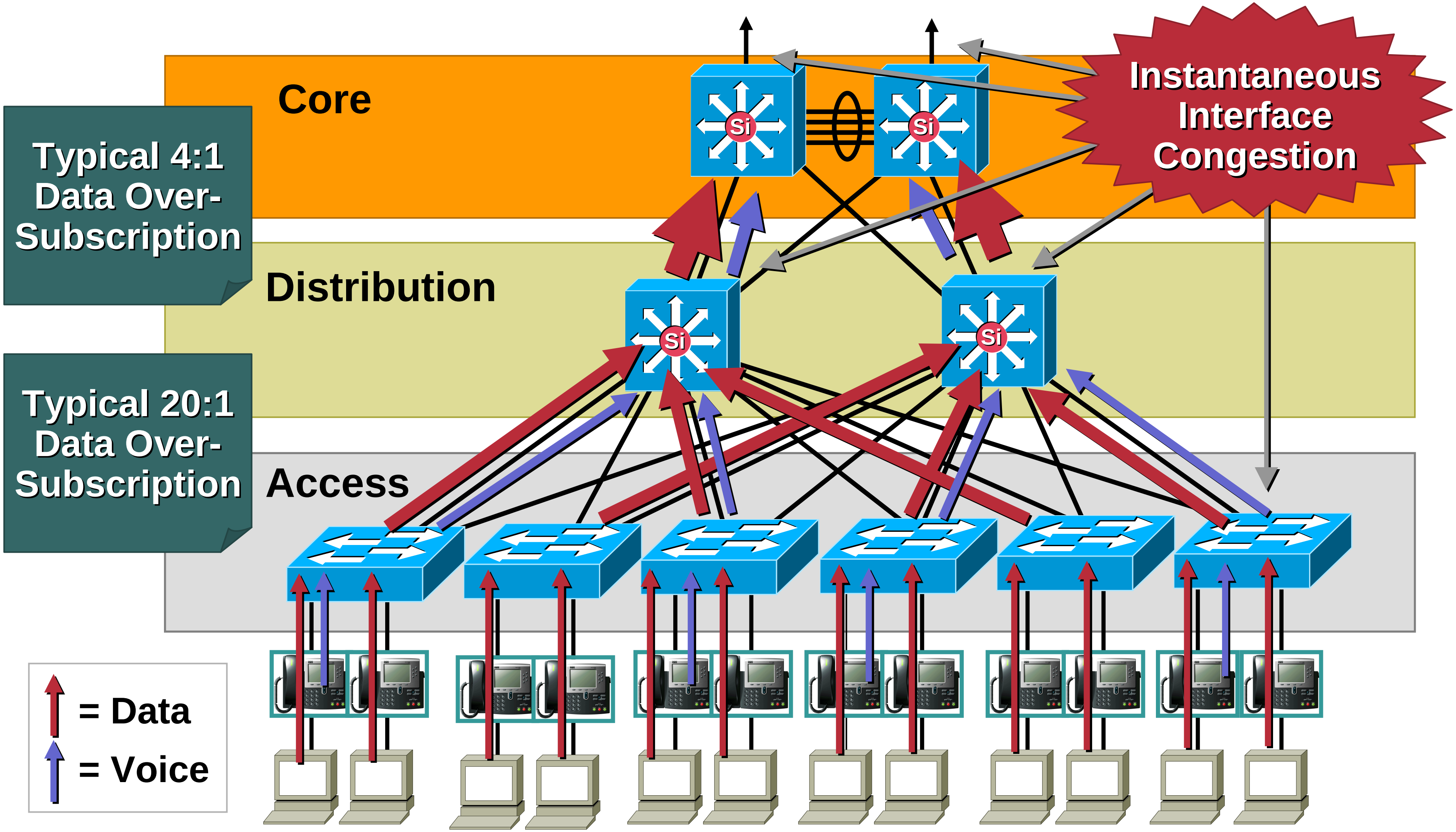


Building a Campus Network
Summary

- **Access Layer**
  - Per-VLAN spanning-tree
  - Rootguard portfast UplinkFast
- **Distribution Layer**
  - HSRP with load balancing
  - OSPF/EIGRP configured for fast convergence
- **Core**
  - OSPF/EIGRP configured for fast convergence



Enabling QoS in the Campus
Congestion Scenario: TCP Traffic Burst + VoIP



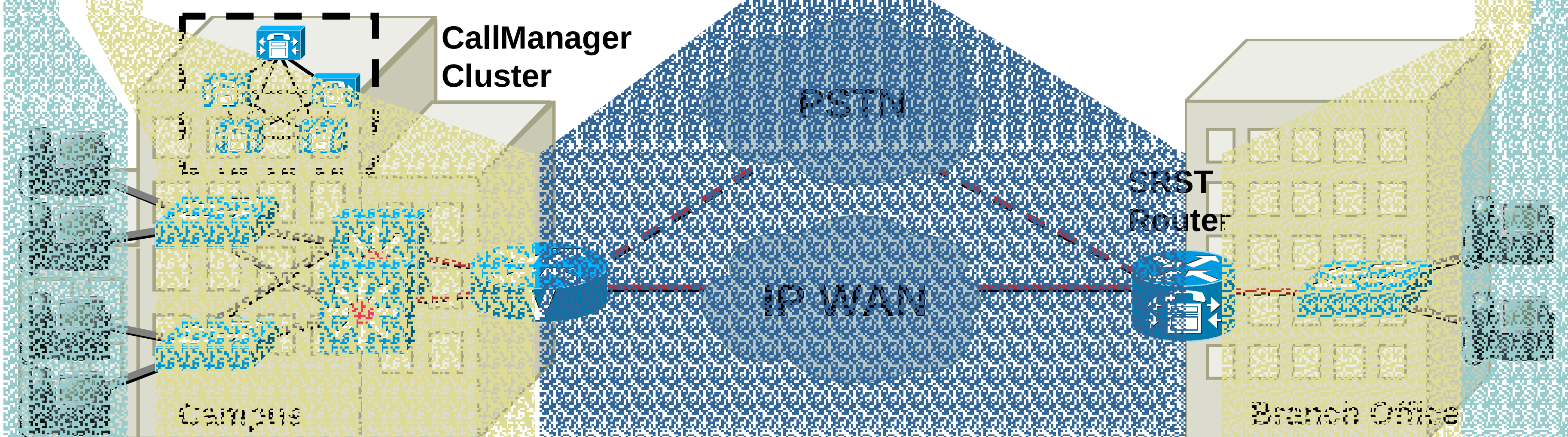
Enabling QoS in the Campus
Cisco's Approach to QoS

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

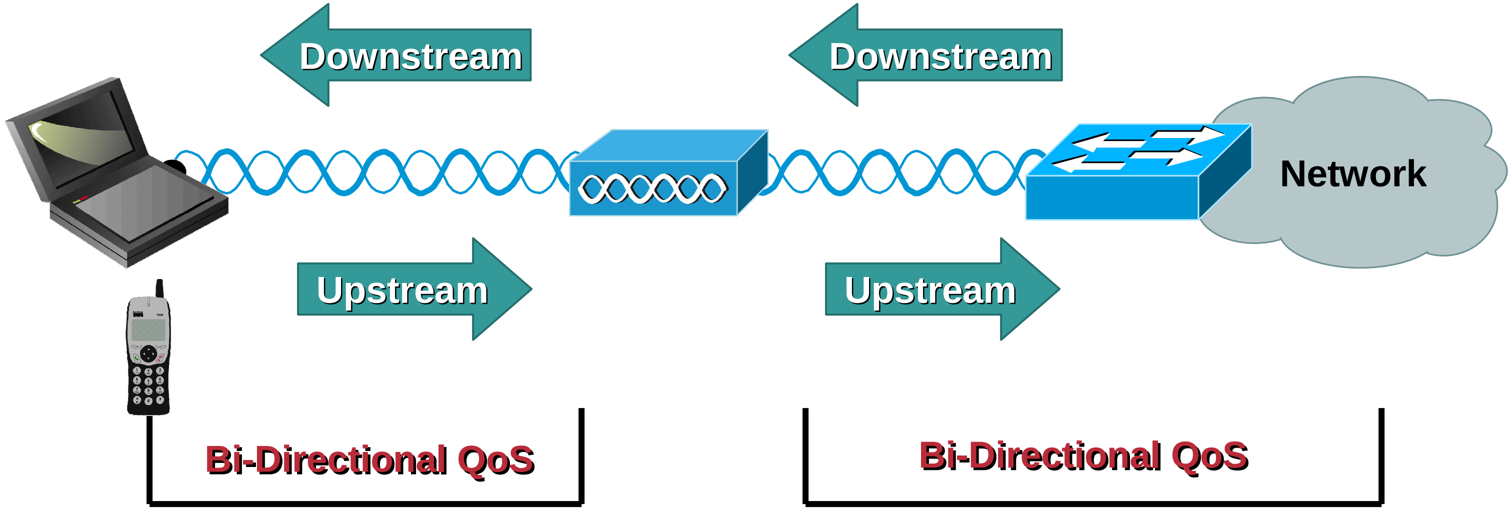
Trust Boundary: Define and Enforce a Trust Boundary at the Network Edge

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



WLAN QoS Challenges



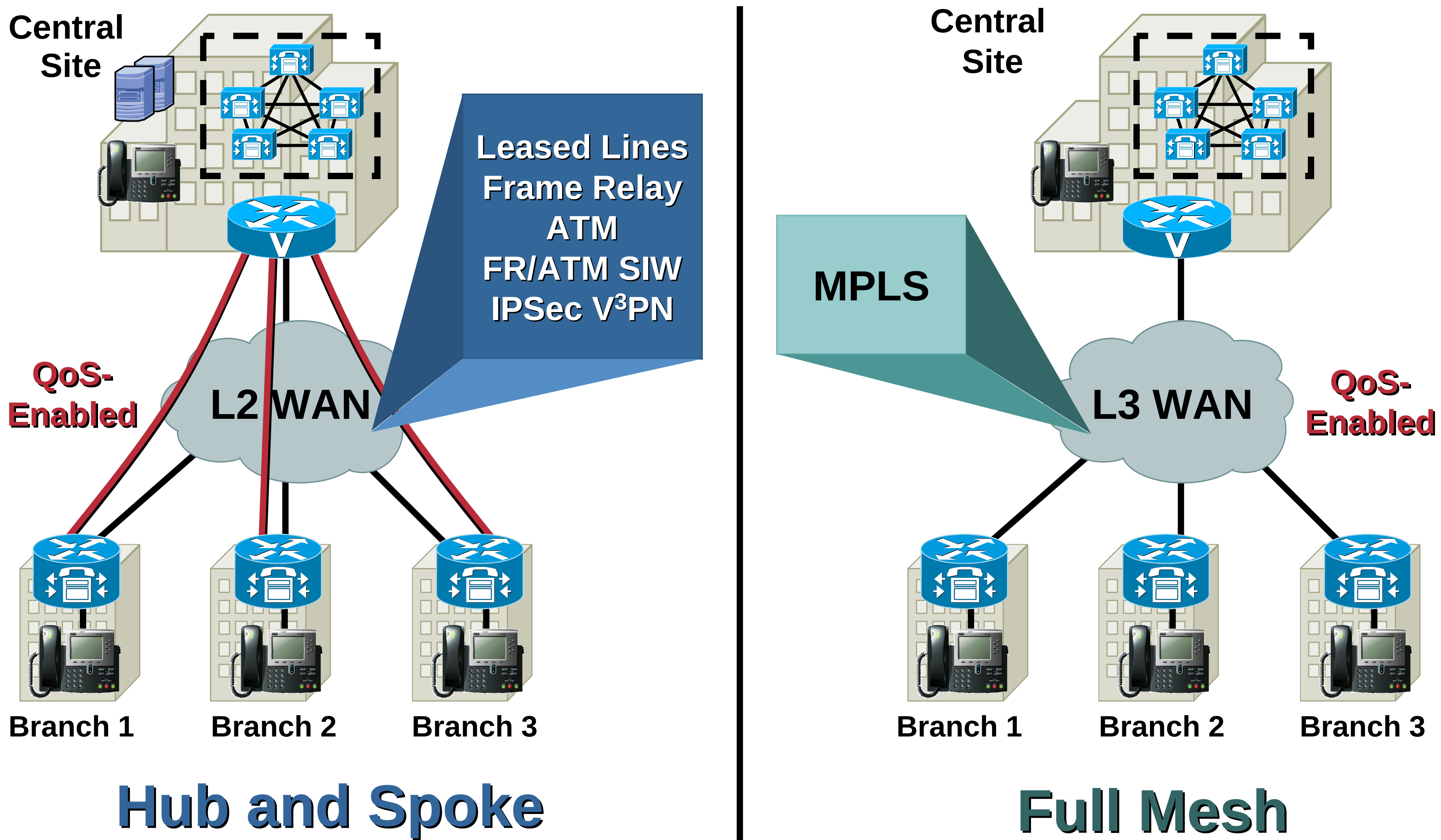
- Downstream traffic from AP receives “soft QoS” as VoIP receives preferred treatment, but not a strict priority queue
- Upstream traffic from Cisco 7920 uses enhanced access to RF medium, so it will also get “soft QoS”

WLAN Sizing—
Number of Concurrent CALLS per AP

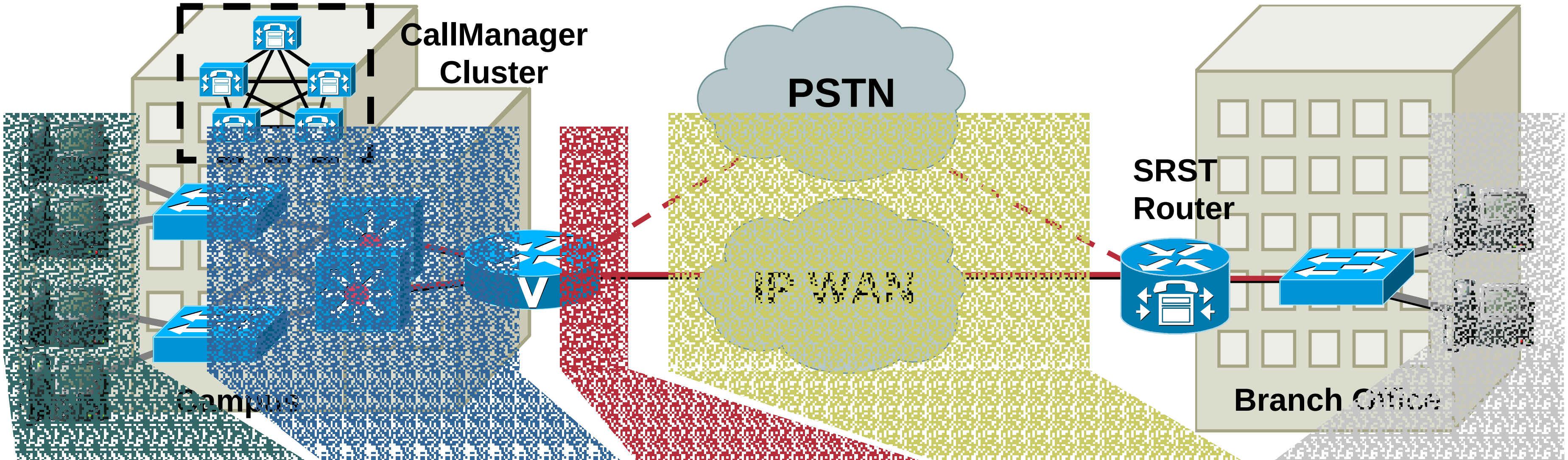
- **G.711—7 concurrent calls**
- **G.729—8 concurrent calls**
- **Call numbers are with Voice Activity Detection (VAD) disabled (VAD is a global parameter on CM)**
- **20ms sampling rates on phones 100pps (full-duplex) of Real Time Protocol (RTP)**

QoS and WAN Considerations
WAN Topologies and Technologies

Cisco.com



Enabling QoS in the WAN
Elements that Affect End-to-End Delay

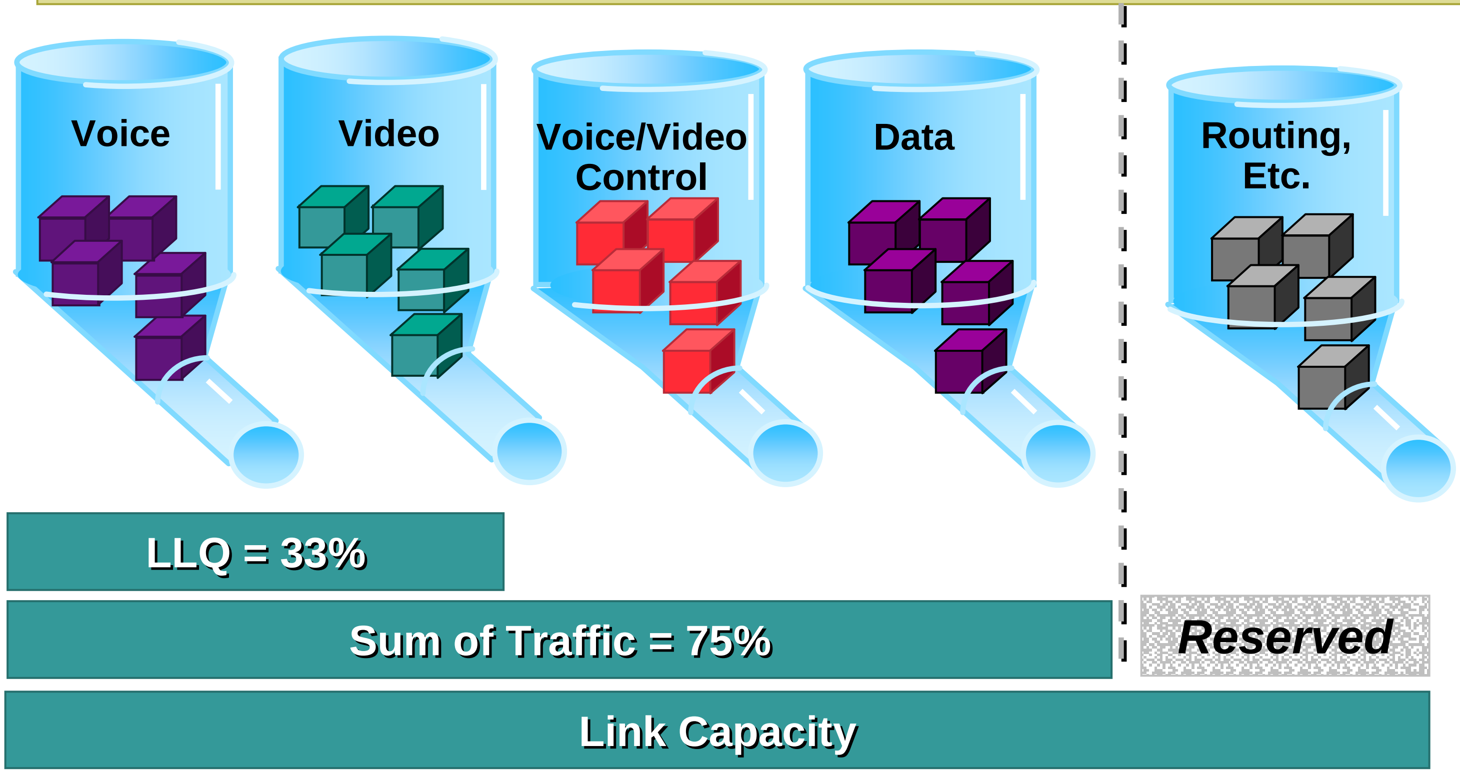


CODEC	Queuing	Serialization	Propagation and Network	Jitter Buffer
G.729A: 25 ms	Variable *(Can Be Reduced Using LLQ)*	Variable *(Can Be Reduced Using LFI)*	6.3 µs/Km + Network Delay (Variable)	20-50 ms

End-to-End Delay (Should be < 150 ms)

Enabling QoS in the WAN
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

Enabling QoS in the WAN
Provisioning Tables for Voice Bearer Traffic

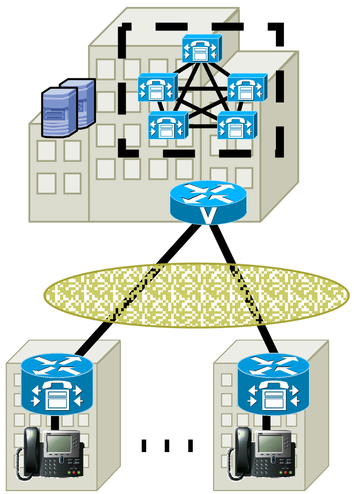
Cisco.com

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

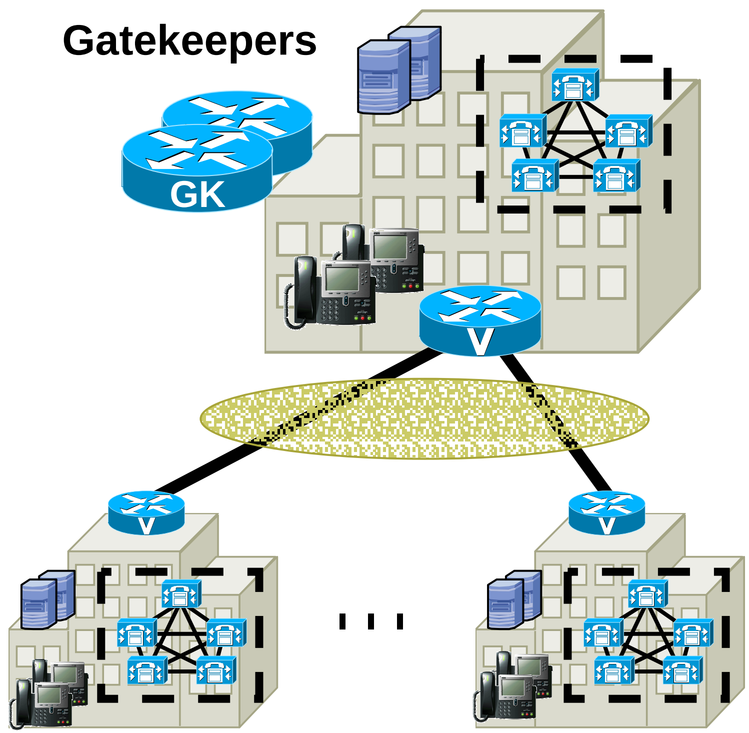
A More Accurate Method for Provisioning Is to Include the Layer 2 Headers into the Bandwidth Calculations:

CODEC	Ethernet 14 Bytes of Header	PPP 6 Bytes of Header	ATM 53 Bytes Cells with a 48 Byte Payload	Frame Relay 4 Bytes of Header
G.711 at 50 pps	85.6 kbps	82.4 kbps	106 kbps	81.6 kbps
G.711 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

Enabling QoS in the WAN
Provisioning Tables for Signaling Traffic

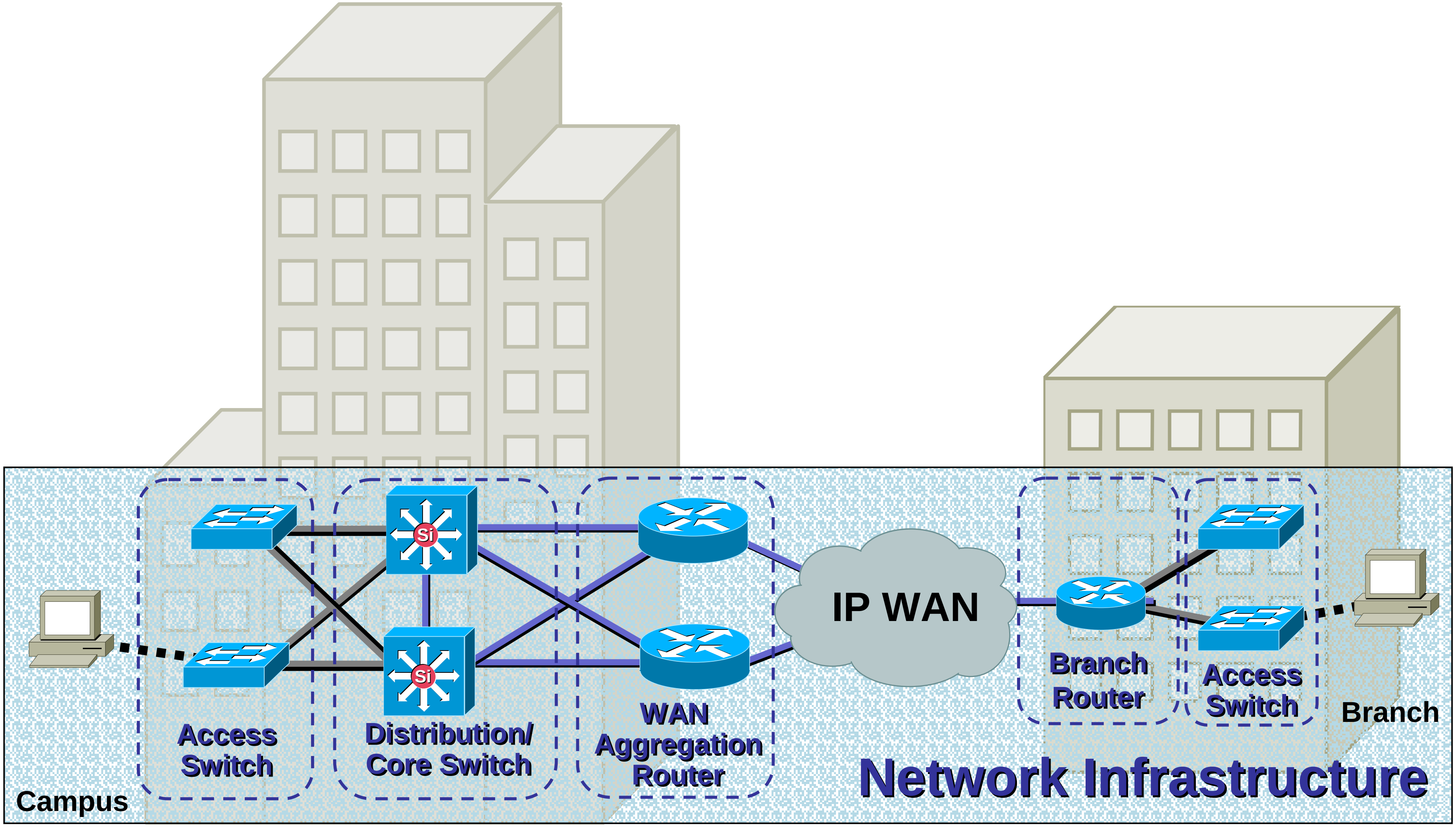


Centralized Call Processing	
# of IP Phones, Gateways	Bandwidth
1 to 30	8 kbps
50	11 kbps
100	23 kbps
150	34 kbps



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

What We Have Built so Far



Agenda

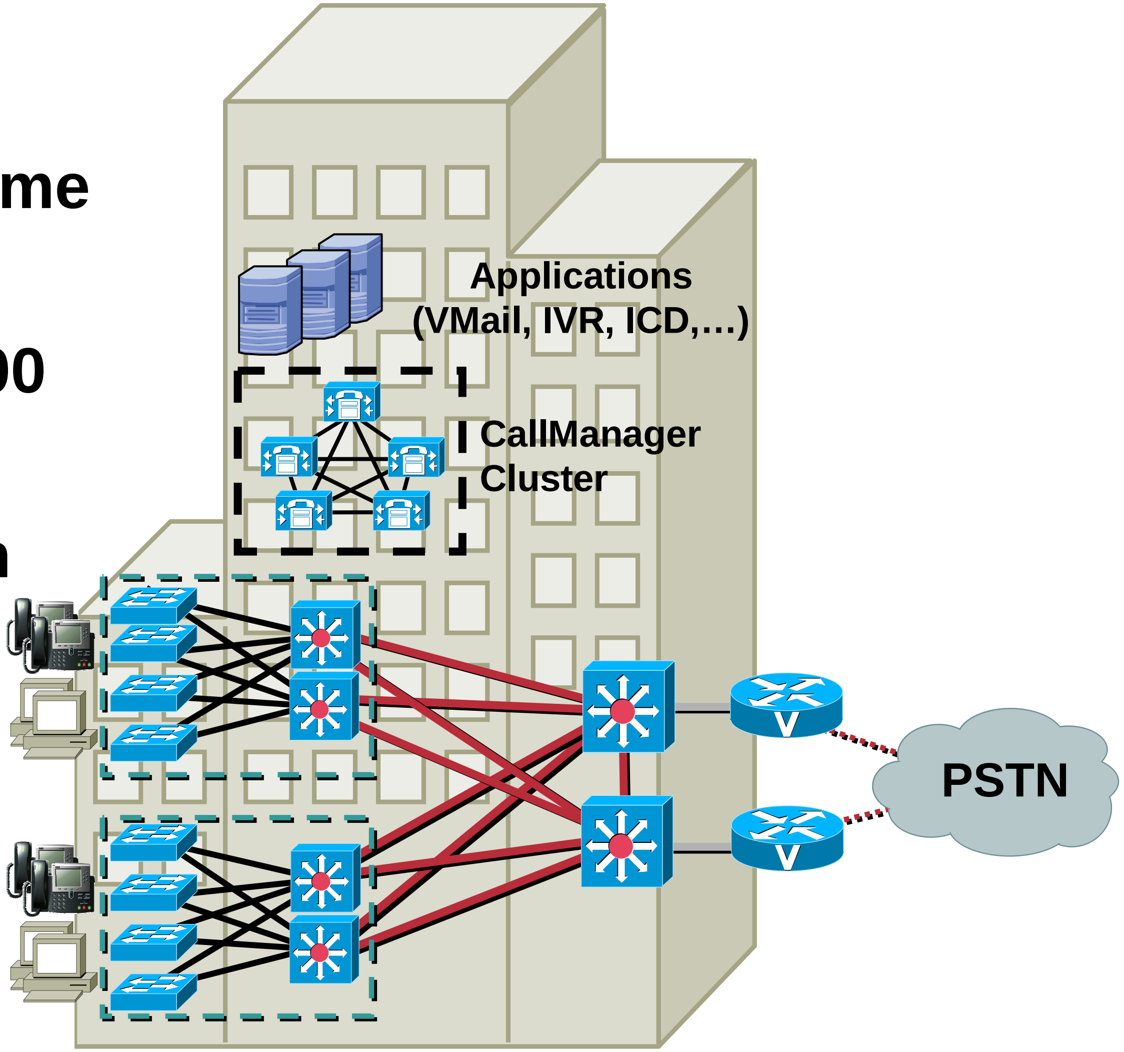
- Introduction
- Network Infrastructure
- Telephony Infrastructure
- Applications

Telephony Infrastructure Agenda (1/2)

- Deployment Models
- Signaling Protocols
- Gateways
- Media Resources
- Call Processing/provisioning

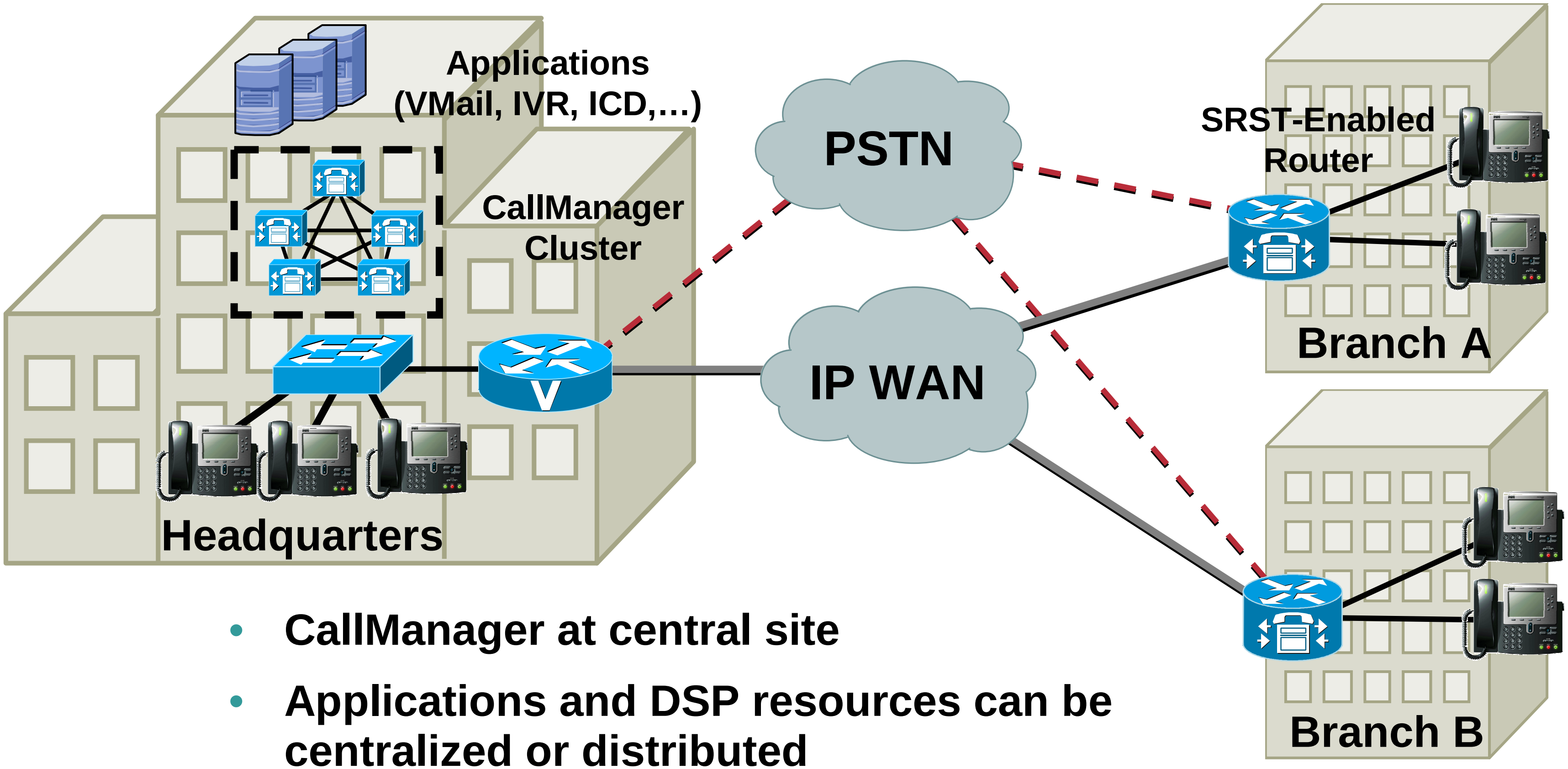
Deployment Models
Single Site

- **Cisco CallManager, applications and DSP resources at same physical location**
- **Supports up to 30,000 lines per cluster**
- **Multiple clusters can be interconnected via inter-cluster trunks**
- **PSTN used for all external calls**



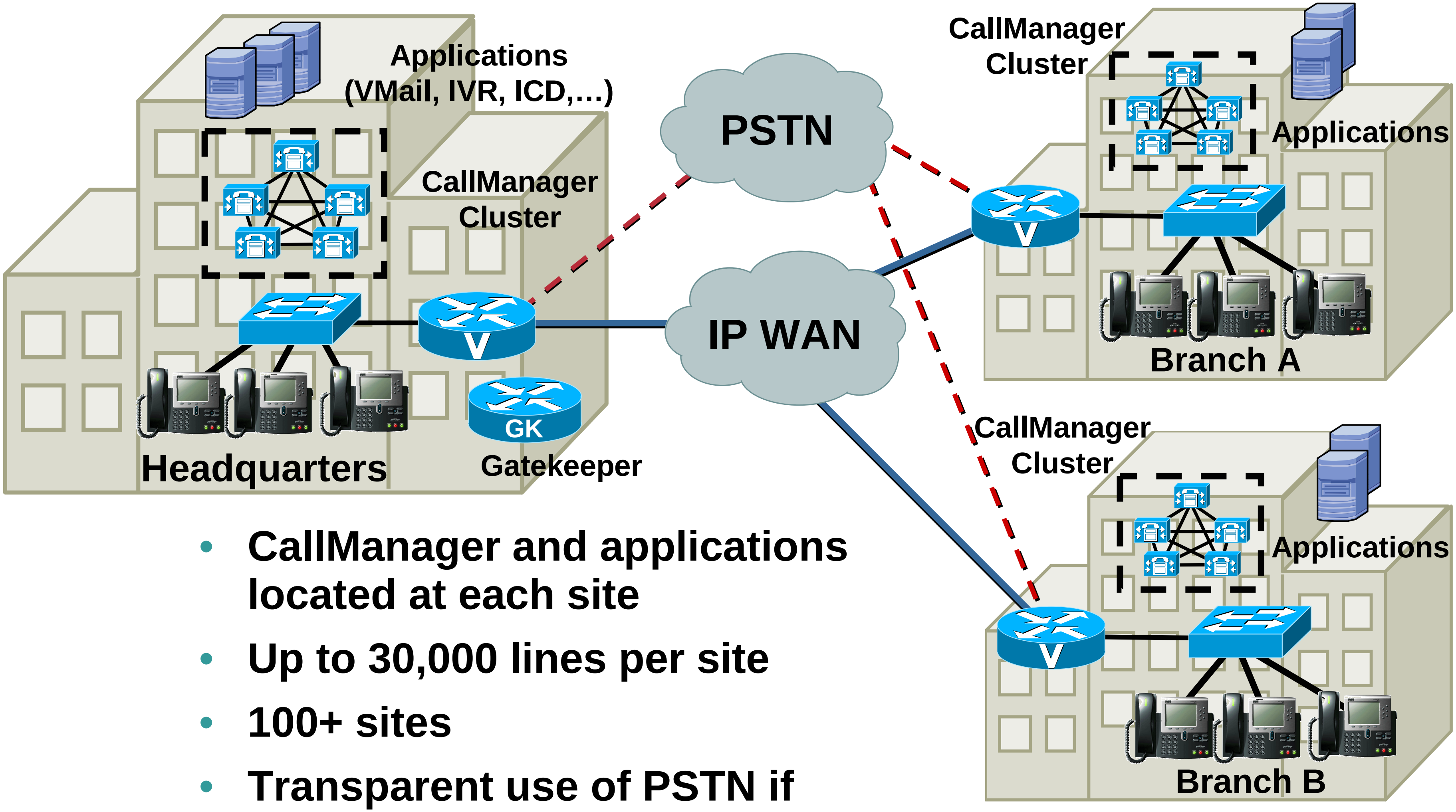
Deployment Models
Centralized Call Processing with Branches

Cisco.com



- CallManager at central site
- Applications and DSP resources can be centralized or distributed
- Supports up to 30,000 lines per cluster
- Call admission control (limit number of calls per site)
- Survivable remote site telephony for remote branches

Deployment Models
Distributed Call Processing

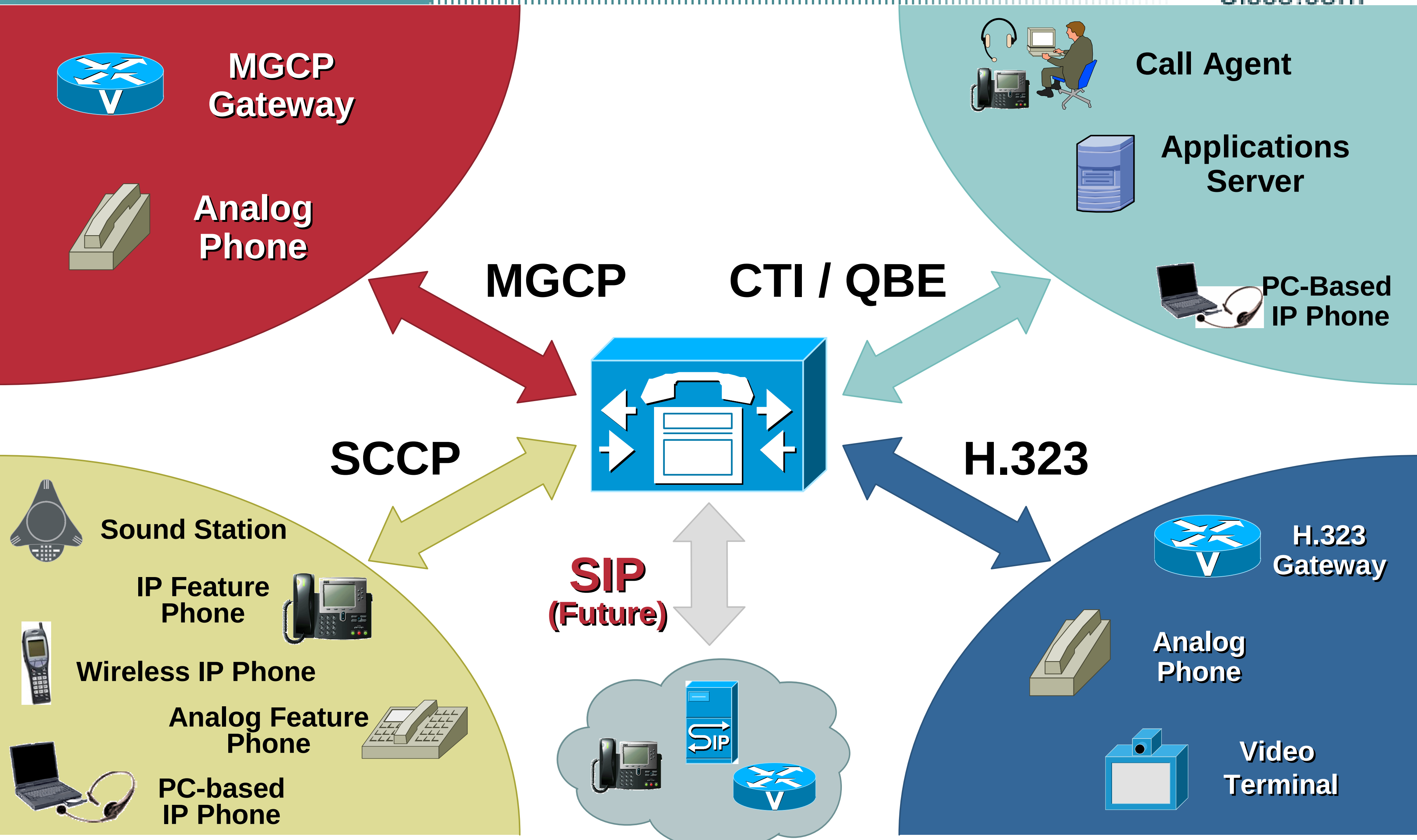


- CallManager and applications located at each site
- Up to 30,000 lines per site
- 100+ sites
- Transparent use of PSTN if IP WAN unavailable

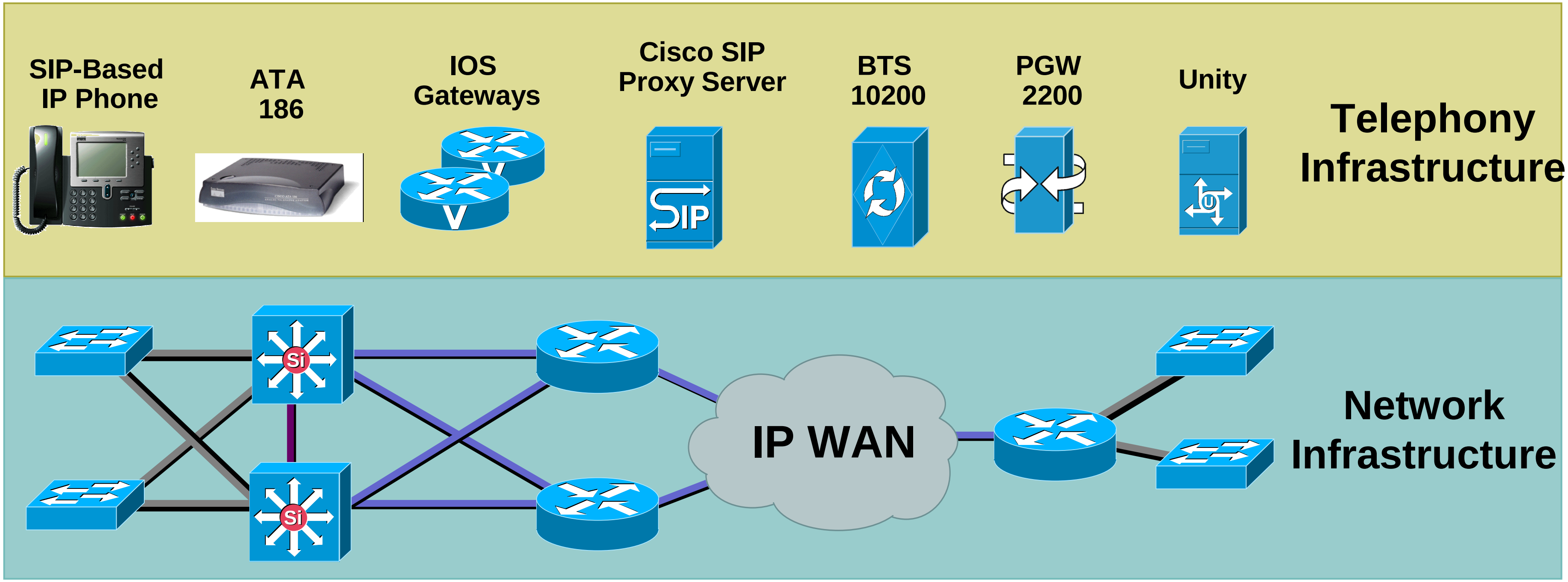
Telephony Infrastructure Agenda (1/2)

- Deployment Models
- Signaling Protocols
- Gateways
- Media Resources
- Call Processing/Provisioning

Signaling Protocols
CallManager as a “Protocol Translator”



Signaling Protocols
More about SIP



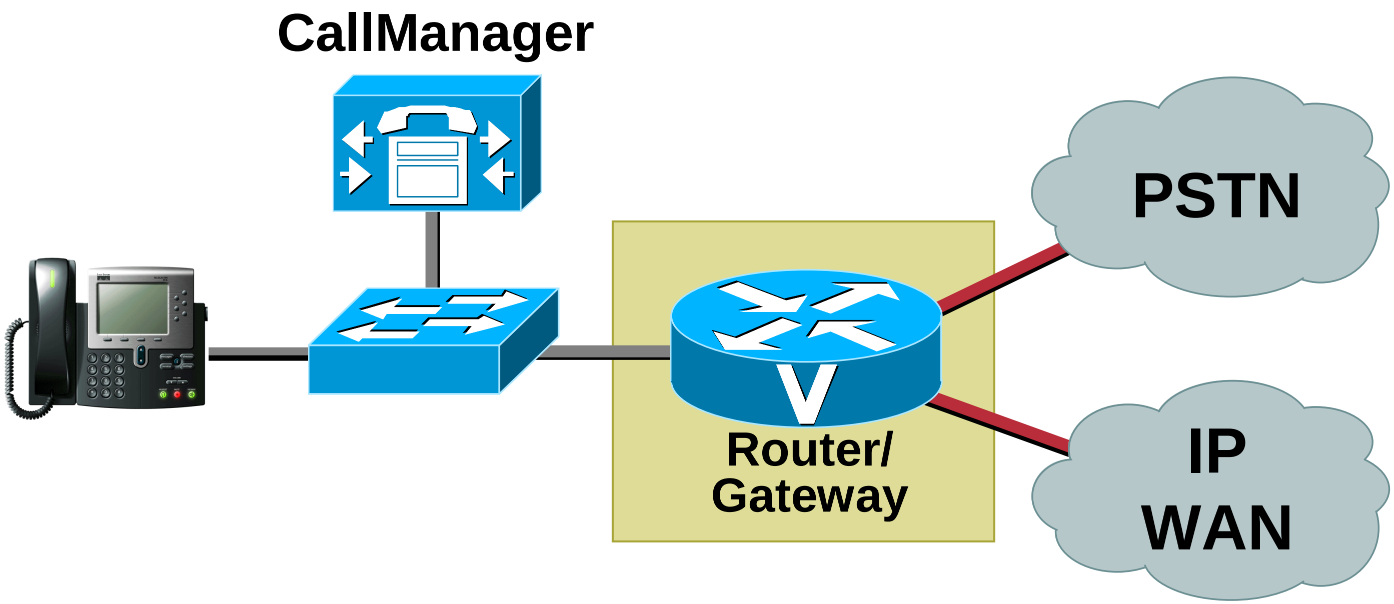
- Several Cisco voice products already support SIP
- The network infrastructure is independent of the signaling protocol
- Many PBX features cannot be delivered natively using SIP today
- Easy migration of current solution when SIP catches on in the enterprise and rolls into CallManager

Telephony Infrastructure Agenda (1/2)

- Deployment Models
- Signaling Protocols
- Gateways
- Media Resources
- Call Processing/Provisioning

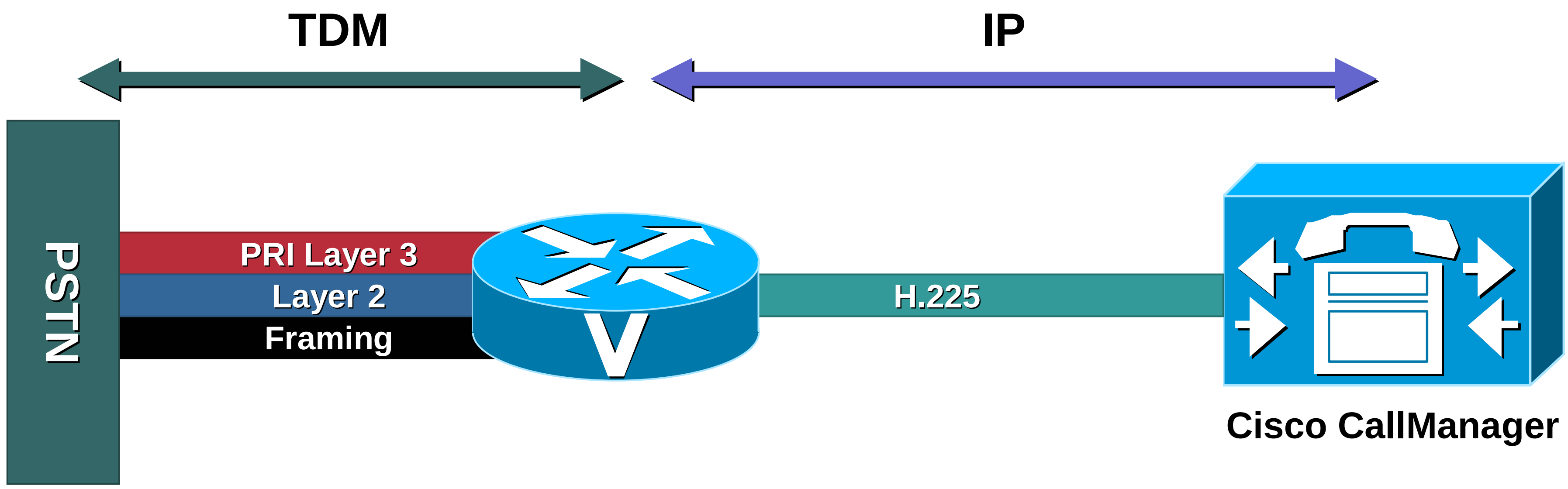
Gateways
Gateway Selection Criteria

Cisco.com



- Voice port density requirements
- Signaling protocol (H.323, MGCP, 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 CallManager
- H.323 is a “peer-to-peer” protocol

Gateways
H.323—Pros and Cons

Pros

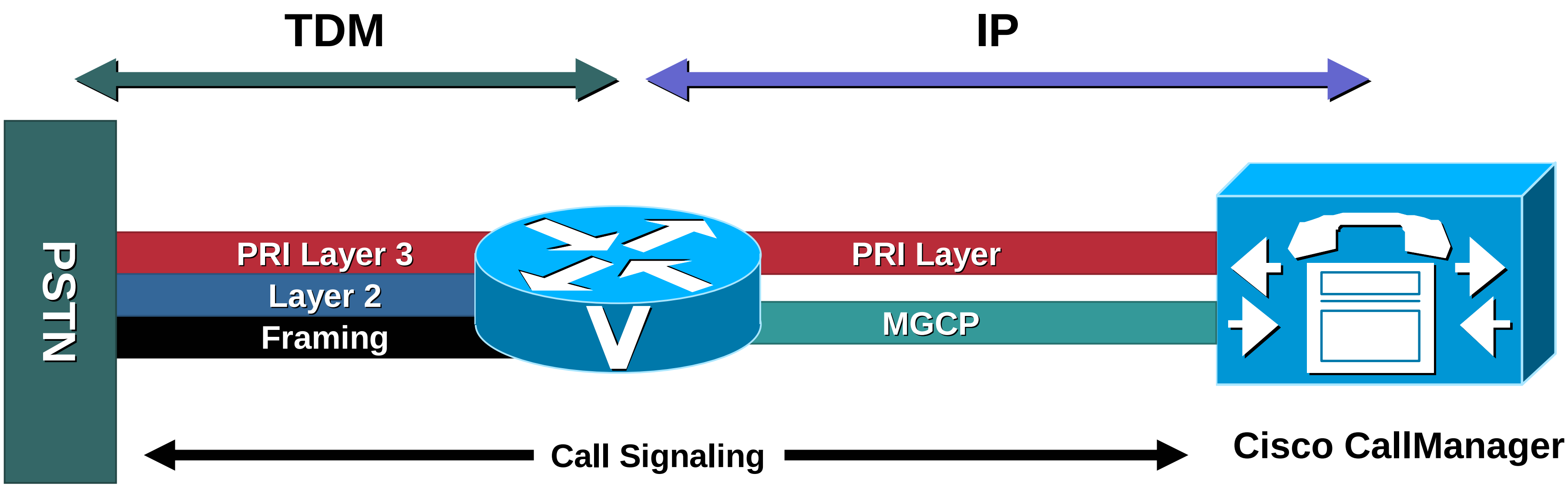
- Interoperability
- Breadth of product and interface choice
- Support for survivable remote site telephony
- Gateway intelligence

Cons

- Higher administration required
- No call preservation

Gateways
MGCP—PRI Backhaul

Cisco.com



- **Framing and layer 2 signaling terminates at the gateway**
- **Layer 3 signaling is backhauled to the CallManager**
- **MGCP is a “client-server” protocol**

Gateways
MGCP—Pros and Cons

Cisco.com

Pros

- Ease of dial plan administration
- Call preservation
- Port-level control (Required for voice mail integration)

Cons

- Dependency on connectivity to call agent

Telephony Infrastructure Agenda (1/2)

- Deployment Models
- Signaling Protocols
- Gateways
- Media Resources
- Call Processing/Provisioning

Media Resources
Conferencing, Transcoding, Music on Hold

Cisco.com

- **Conferencing**

  DSPs needed for multi-party conferences

- **Transcoding**

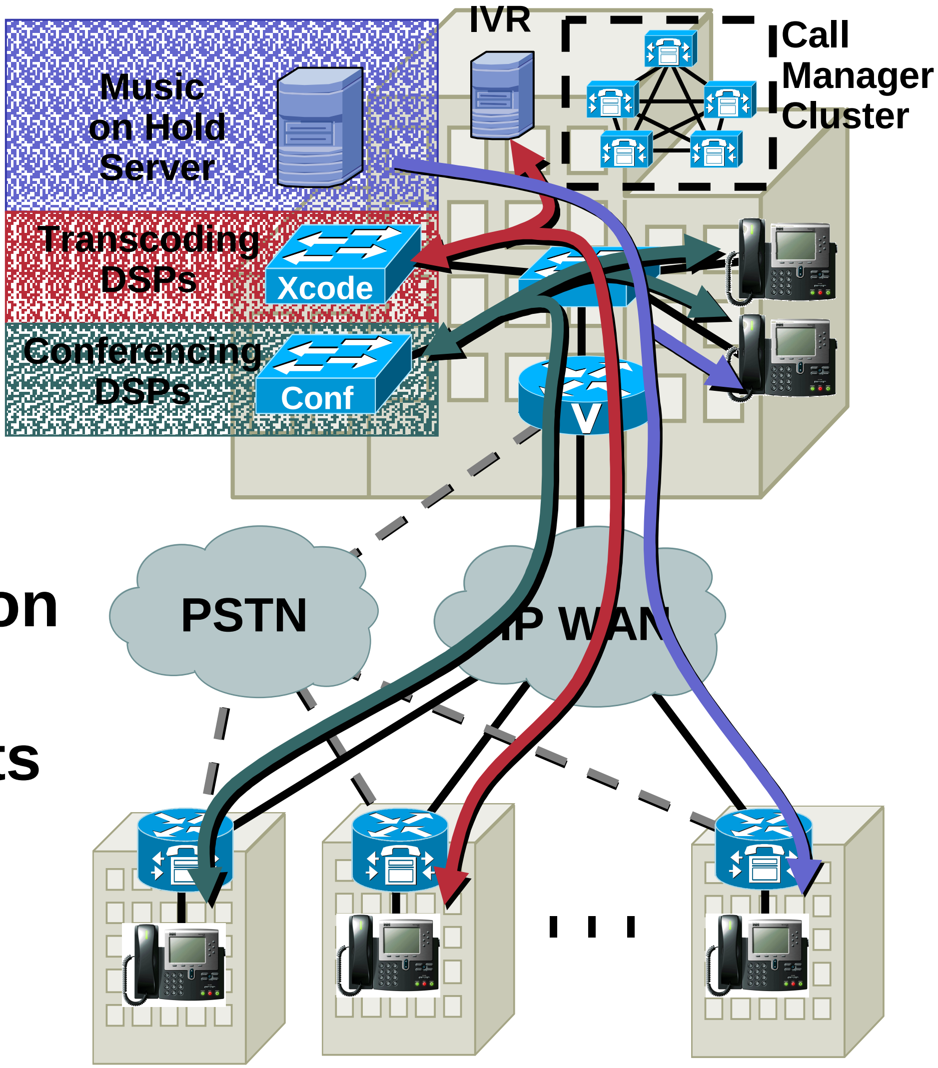
  Multiple CODEC support (e.g., G.711 to G.729)

  Automatic CODEC selection

  DSPs needed in presence of single-CODEC endpoints

- **Music on Hold**

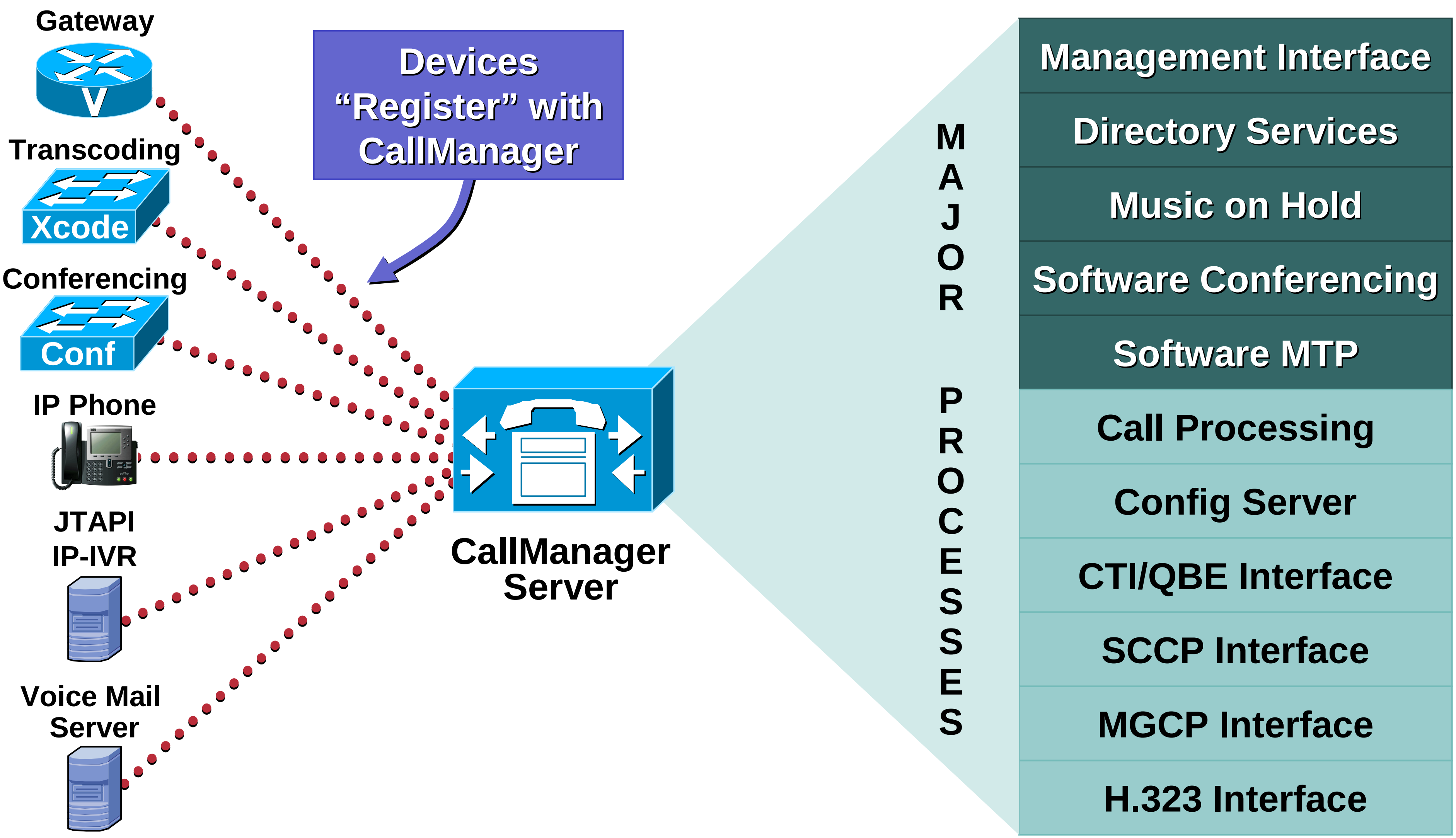
  Centralized server sends streams across the WAN



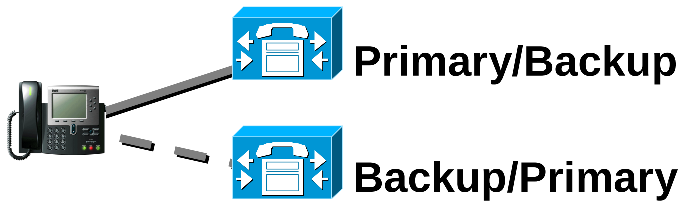
Telephony Infrastructure Agenda (1/2)

- Deployment Models
- Basic Call Processing
- Signaling Protocols
- Gateways
- Media Resources
- Call Processing/Provisioning

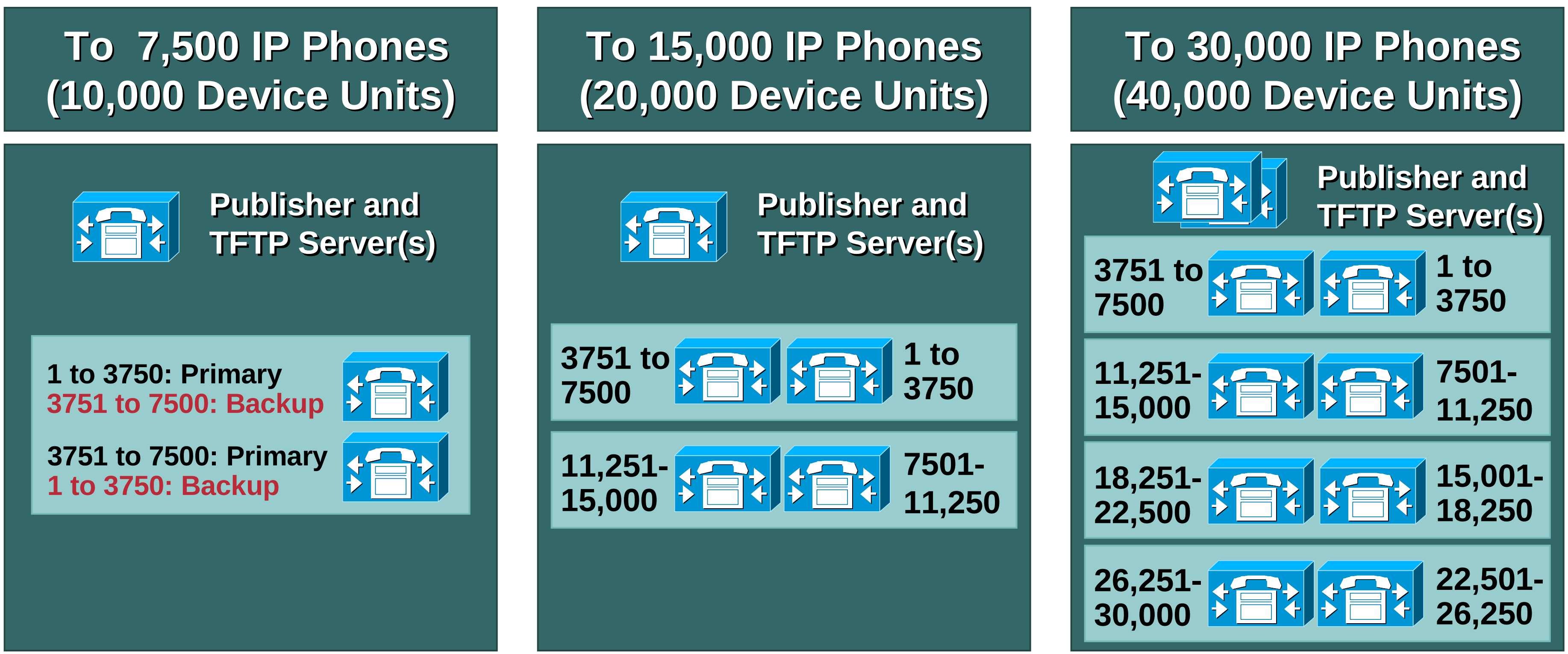
CallManager Redundancy and Scalability
What Is a Server? What Does It Do?



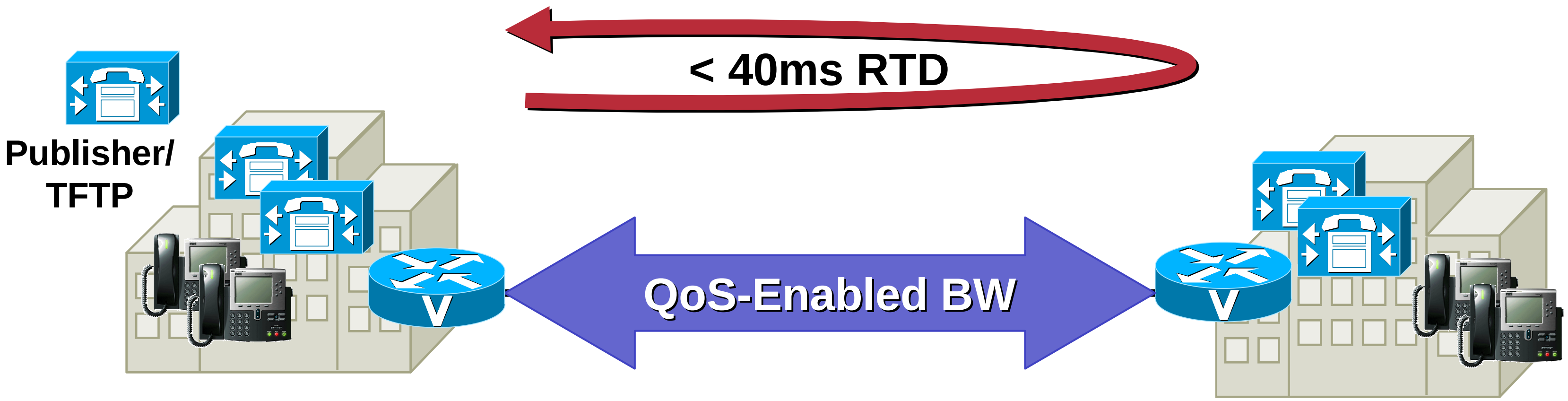
CallManager Redundancy and Scalability
Clustering—1:1 Redundancy (CM 3.3 and MCS 7845)



- **Distribute IP phones based on DN**
- **Load-share between primary and backup servers**



CallManager Redundancy and Scalability
Clustering over the WAN—Guidelines

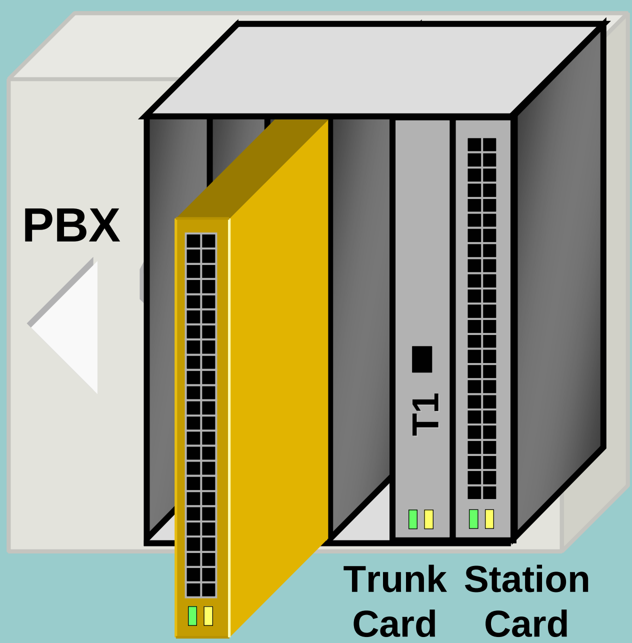


- **Max 40ms round-trip delay between ANY two CallManagers**
- **900 kbps for each 10,000 BHCA within the cluster**
- **Four active locations maximum (4 active CMs)**
- **Failover across the WAN supported (Additional BW)**

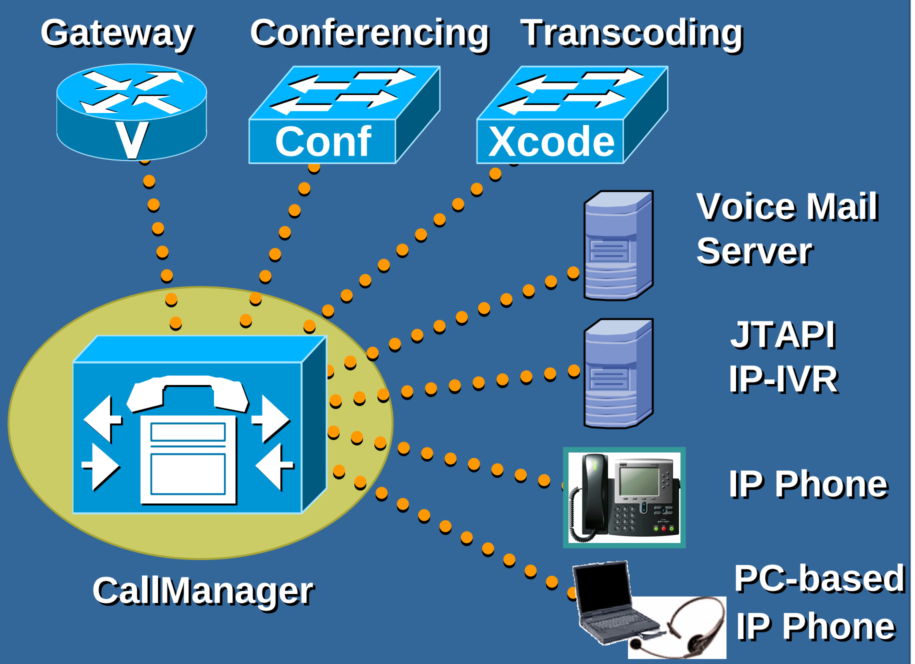
Check Out the IP Telephony Design Guide for CallManager 3.1 and 3.2 for Full Details

CallManager Provisioning
Analogy PBX—IP Telephony

Cisco.com



- “Max. 500 lines”
- *N* slots available
- Actual number of phones depends on mix of station/trunk cards



- “Max. 2,500 IP phones”
- 5,000 device units available
- Actual number of IP phones depends on mix of registered devices

CallManager Provisioning
CallManager Device Weights Table

	Weight BHCA < 6	Weight BHCA < 12	Weight BHCA < 18	Weight BHCA < 24
CTI Server Port	2	4	6	8
CTI Client Port	2	4	6	8
CTI 3rd Party	3	6	9	12
CTI Agent	6	12	18	24
CTI Route Point	2	4	6	8
SCCP Client	1	2	3	4
H.323 Client	3	6	9	12
H.323 Gateway	3	N/A	N/A	N/A
Transcoder MTP	3	N/A	N/A	N/A
MGCP GW	3	N/A	N/A	N/A
Conference	3	N/A	N/A	N/A

Note: GW's, xcoder/mtp and Conference Bridge Are Per DSO or Session

BHCA = Busy Hour Call Attempts

CallManager Provisioning
CallManager Server Platforms

Cisco.com

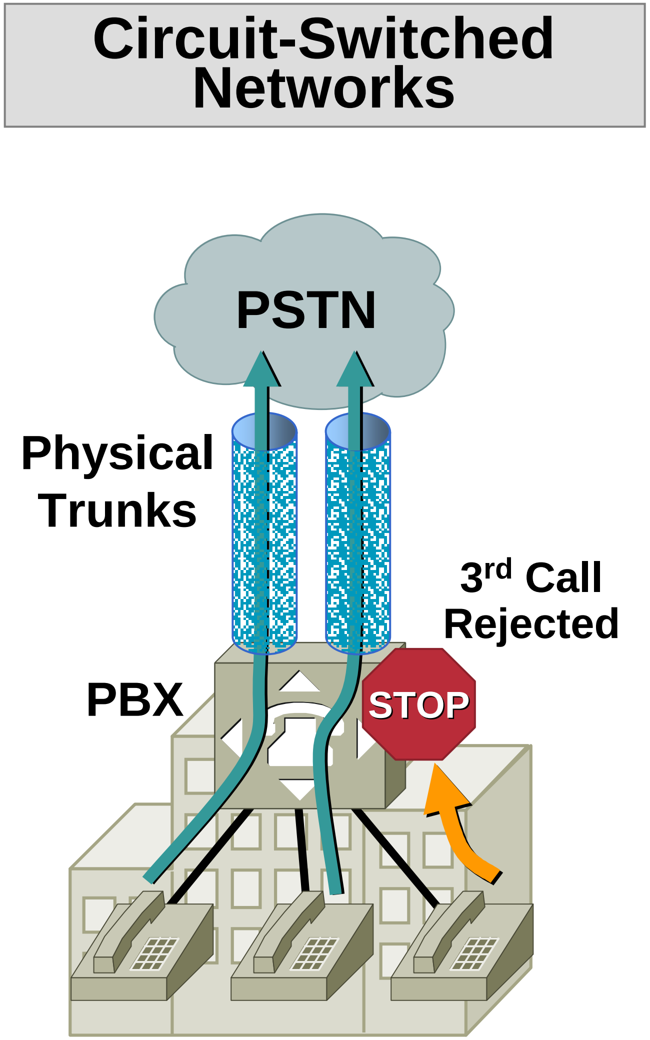
Platform	Device Units Per Server	Maximum IP Phones Per Server
MCS 7845 (Dual)	10000	7500
MCS 7835 All Models	5000	2500
Compaq DL380	5000	2500
IBM xSeries 342	5000	2500
MCS 7825	2000*	1000*
SPE310 (ICS 7750)	2000*	1000*
Compaq DL 320	2000*	1000*
IBM xSeries 330	2000*	1000*
MCS 7815-1000	400	200

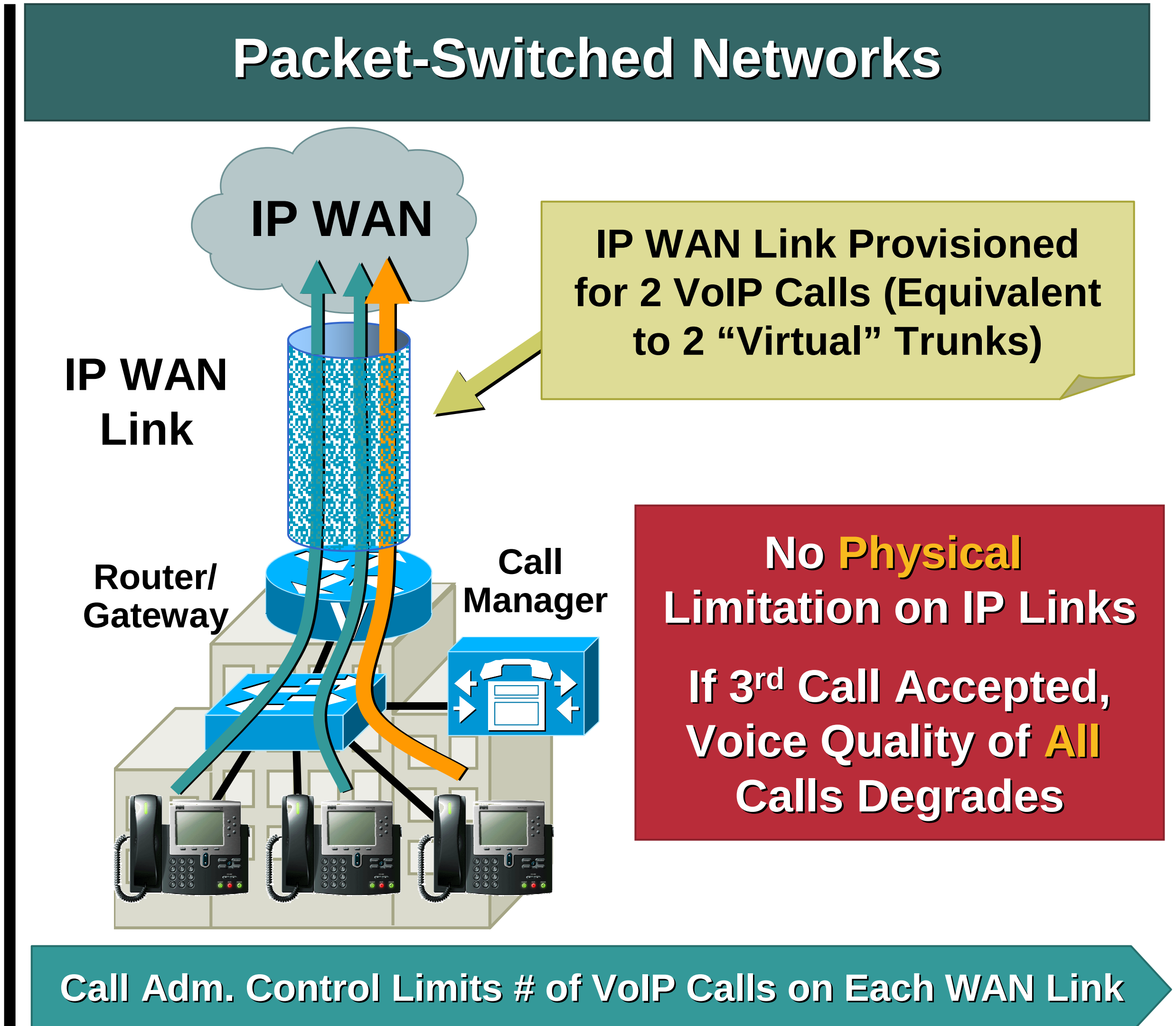
* The Maximum Number of IP Phones Supported on a Single Non-HA Platform Is 500; with a Redundant Server Configuration this Caveat Is Eliminated

Telephony Infrastructure Agenda (2/2)

- Call Admission Control
- Survivable Remote Site Telephony
- Dial Plan
- Security
- Management

Call Admission Control
Why Is It Needed?

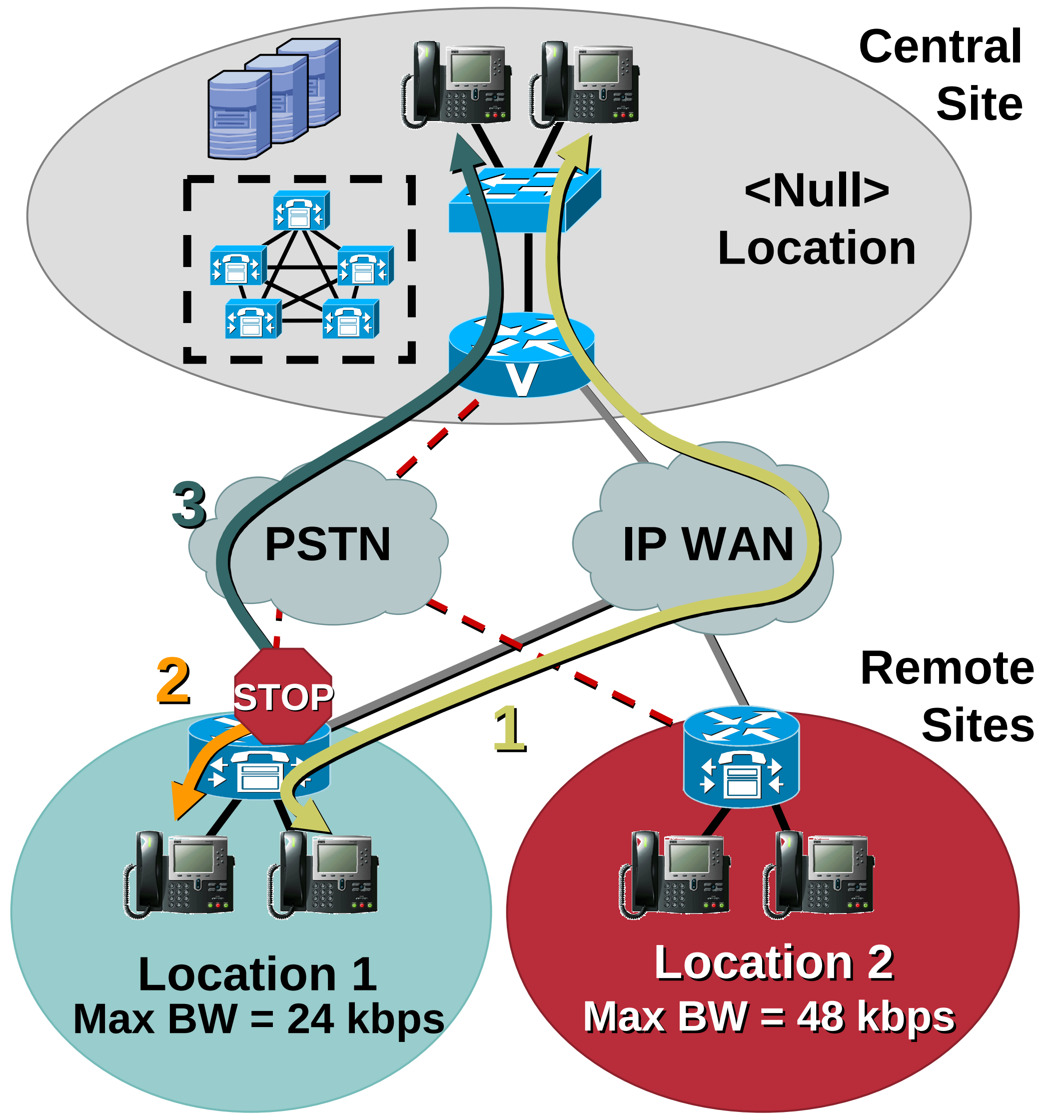




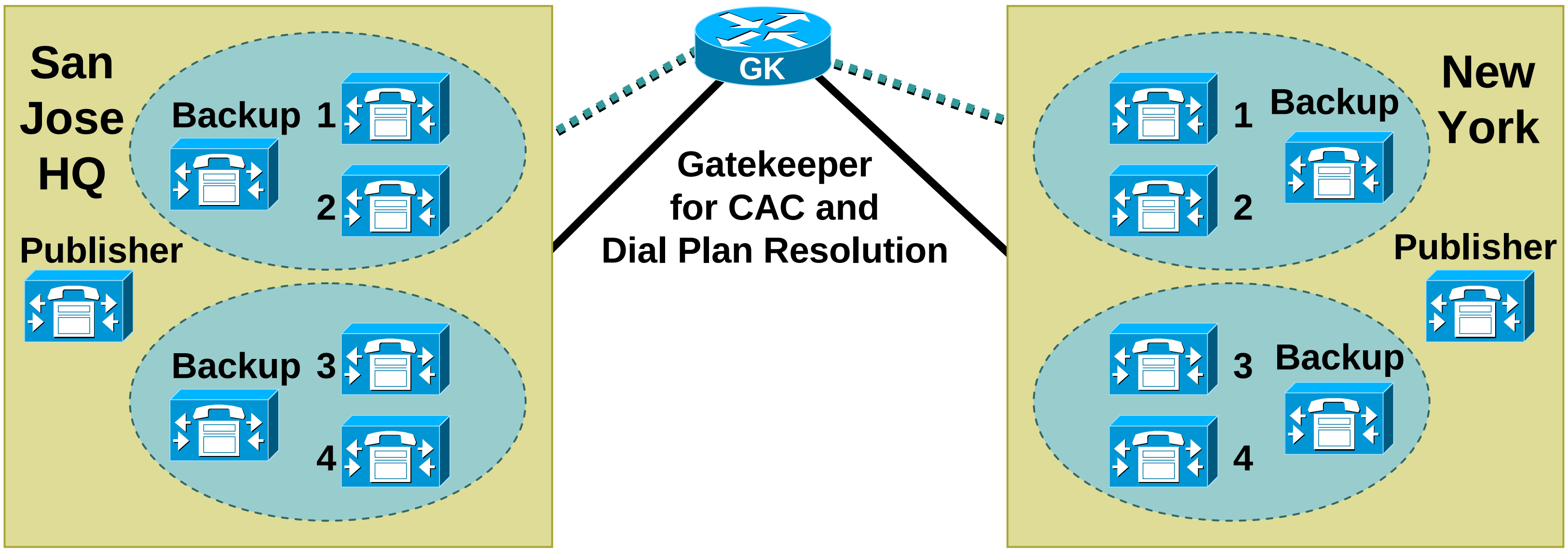
Call Admission Control
CallManager "Locations"

Cisco.com

- **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 AAR is enabled, the call is automatically rerouted across the PSTN (requires CM 3.3)**



Call Admission Control
Gatekeeper

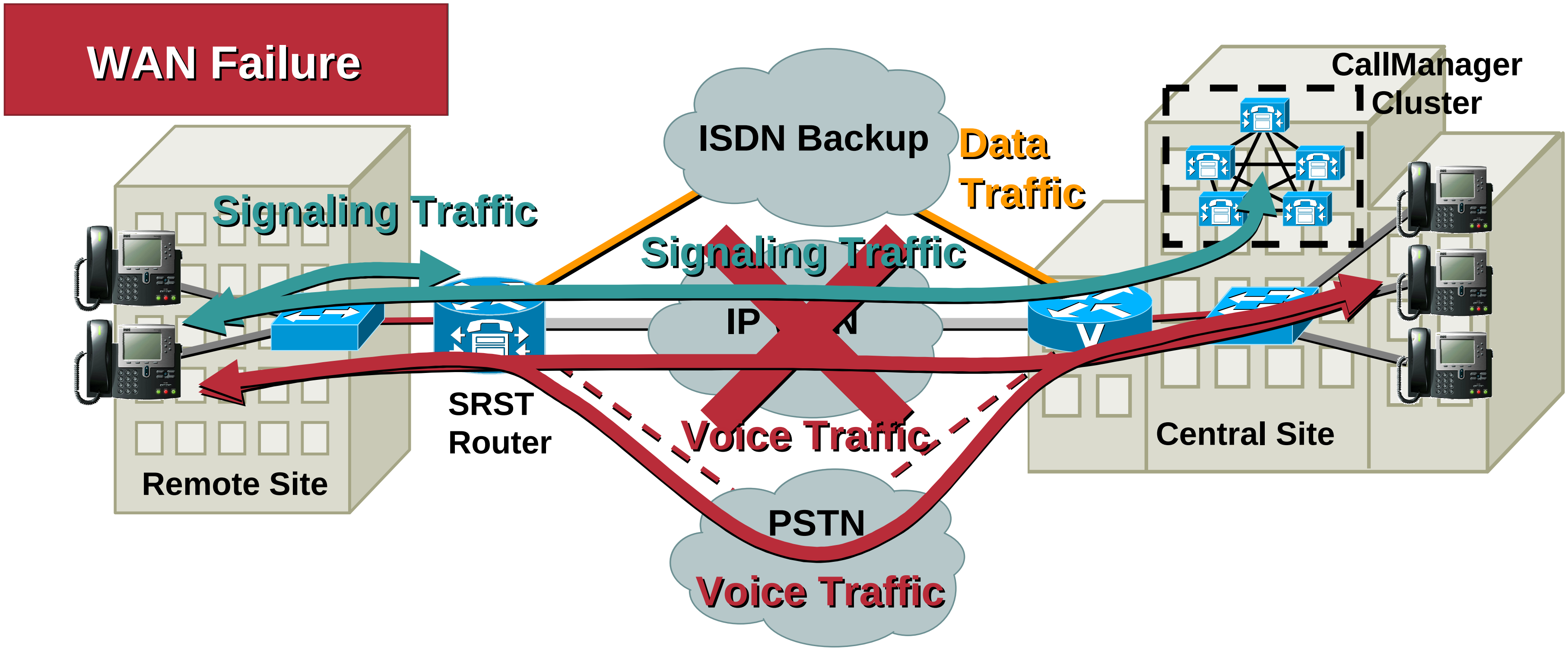


- Gatekeeper provides call admission control in presence of multiple CallManager clusters (distributed call processing deployments)
- Configure CallManager with "anonymous device" (CM 3.2) or "GK-controlled inter-cluster trunk" (CM 3.3) to use gatekeeper also to resolve E.164 addresses

Telephony Infrastructure Agenda (2/2)

- Call Admission Control
- Survivable Remote Site Telephony
- Dial Plan
- Security
- Management

Survivable Remote Site Telephony (SRST) Mode of Operation

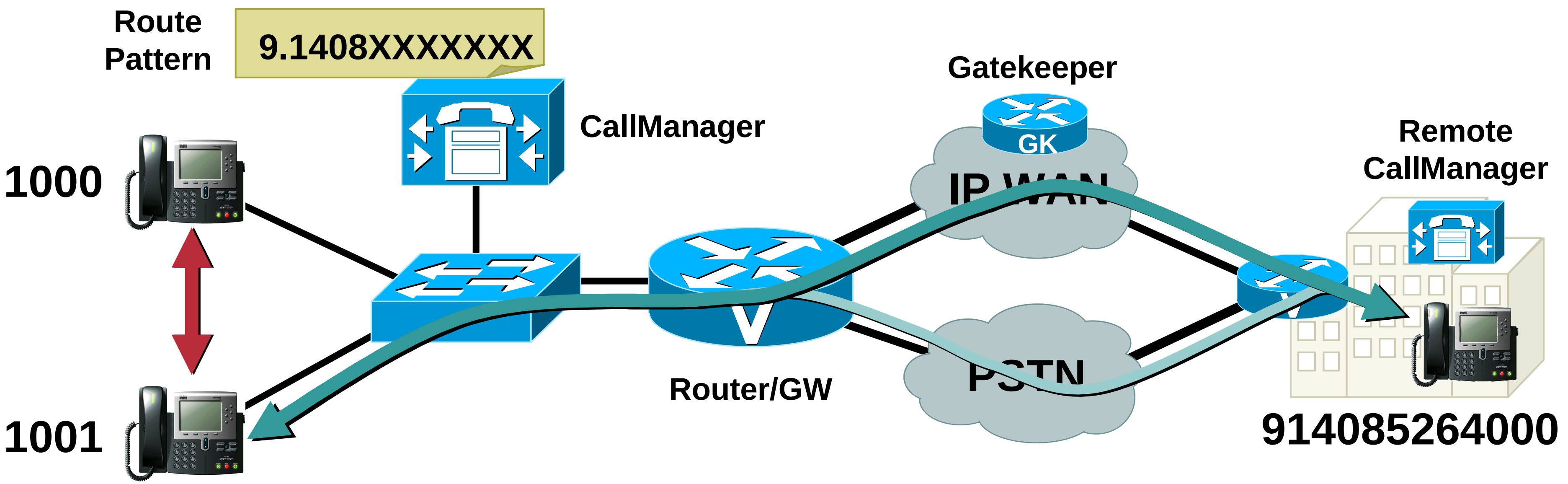


- SRST router needs minimal configuration
- Subset of features available to the phones (DID, DOD, Call Hold, Transfer, Speed Dial, Caller ID)
- ISDN backup may be used for data traffic— ACL needed on branch router to block signaling traffic

Telephony Infrastructure Agenda (2/2)

- Call Admission Control
- Survivable Remote Site Telephony
- Dial Plan
- Security
- Management

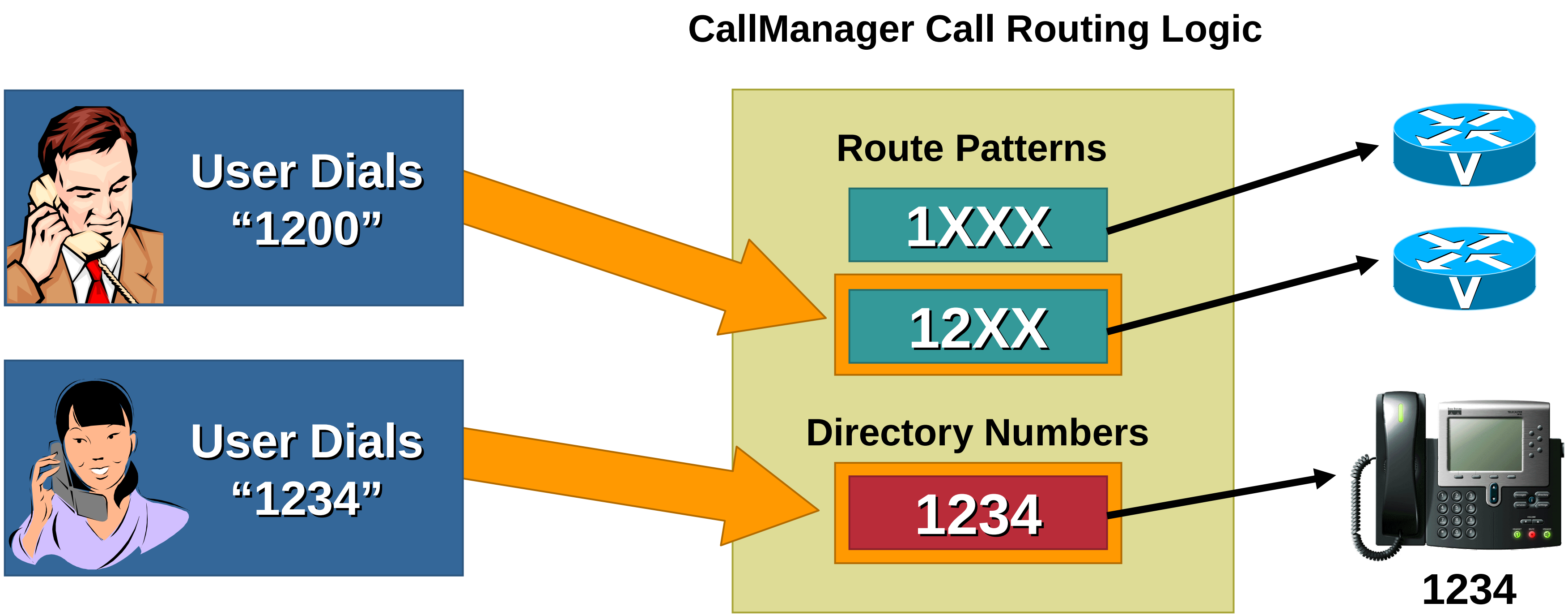
Dial Plan
The "IP Routing" of IP Telephony



CallManager Routes Two Basic Call Types:

- **On-Cluster Calls**—Destination Directory Number (DN) is registered with CallManager
- **Off-Cluster Calls**—External route patterns must be configured on CallManager

Dial Plan
CallManager Call Routing Logic



- CallManager matches the most specific pattern (longest-match logic)
- An IP phone directory number is a special case of route pattern that matches a single number

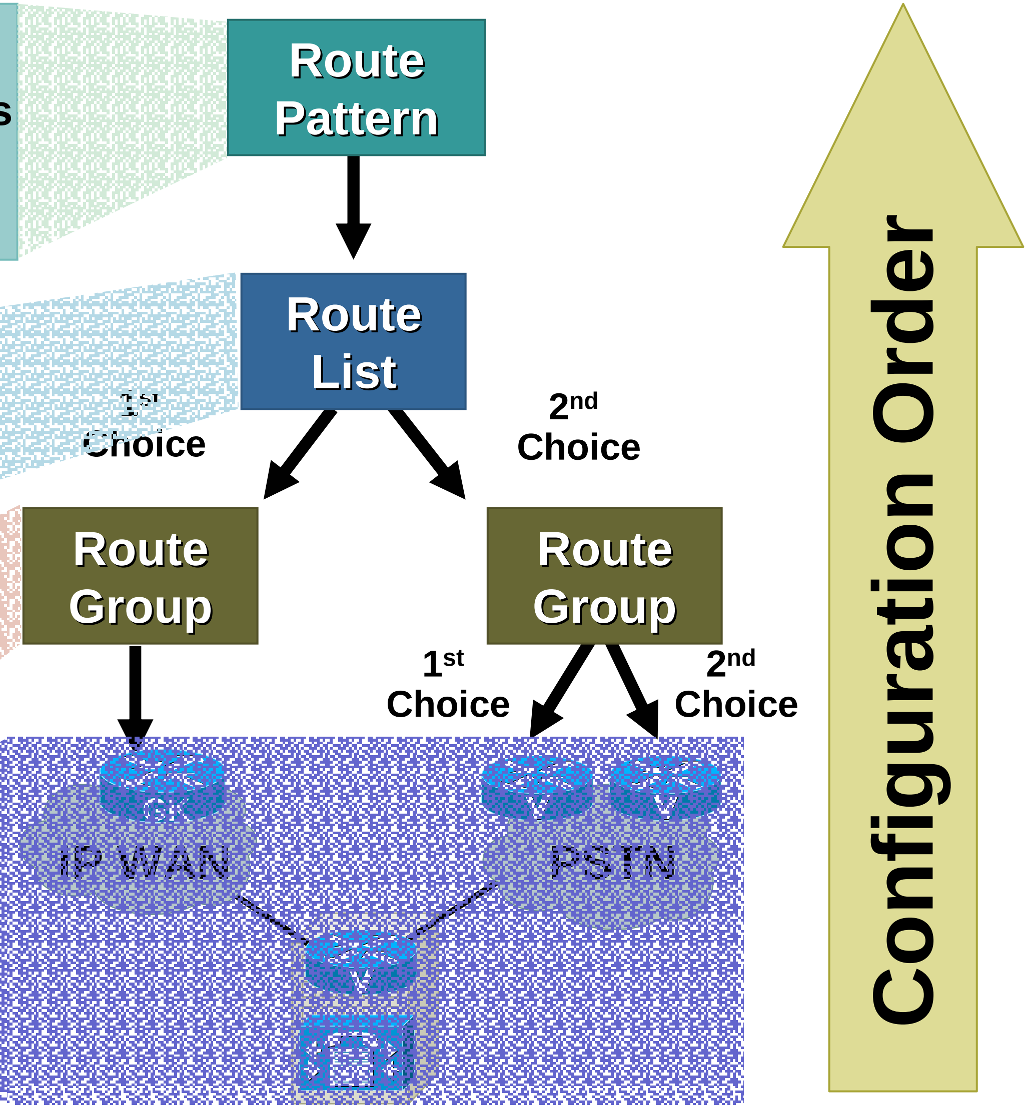
Defining External Routes
External Route Elements in CallManager

Route Pattern
- Matches dialed number for external calls
- Performs digit manipulation (optional)
- Points to a route list for routing

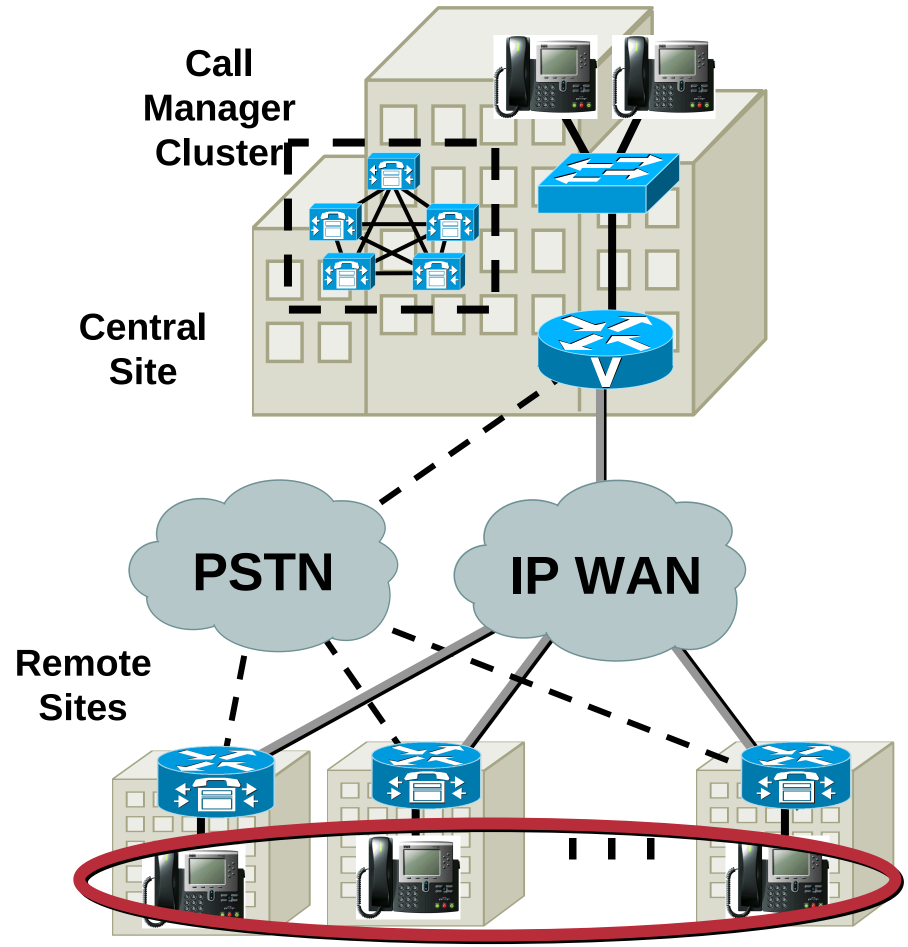
Route List
- Chooses path for call routing
- Points to prioritized route groups

Route Group
- Performs digit manipulation
- Points to the actual devices

Devices
- Gateways (H.323, MGCP)
- Gatekeeper
- Inter-Cluster Trunk (remote CM)



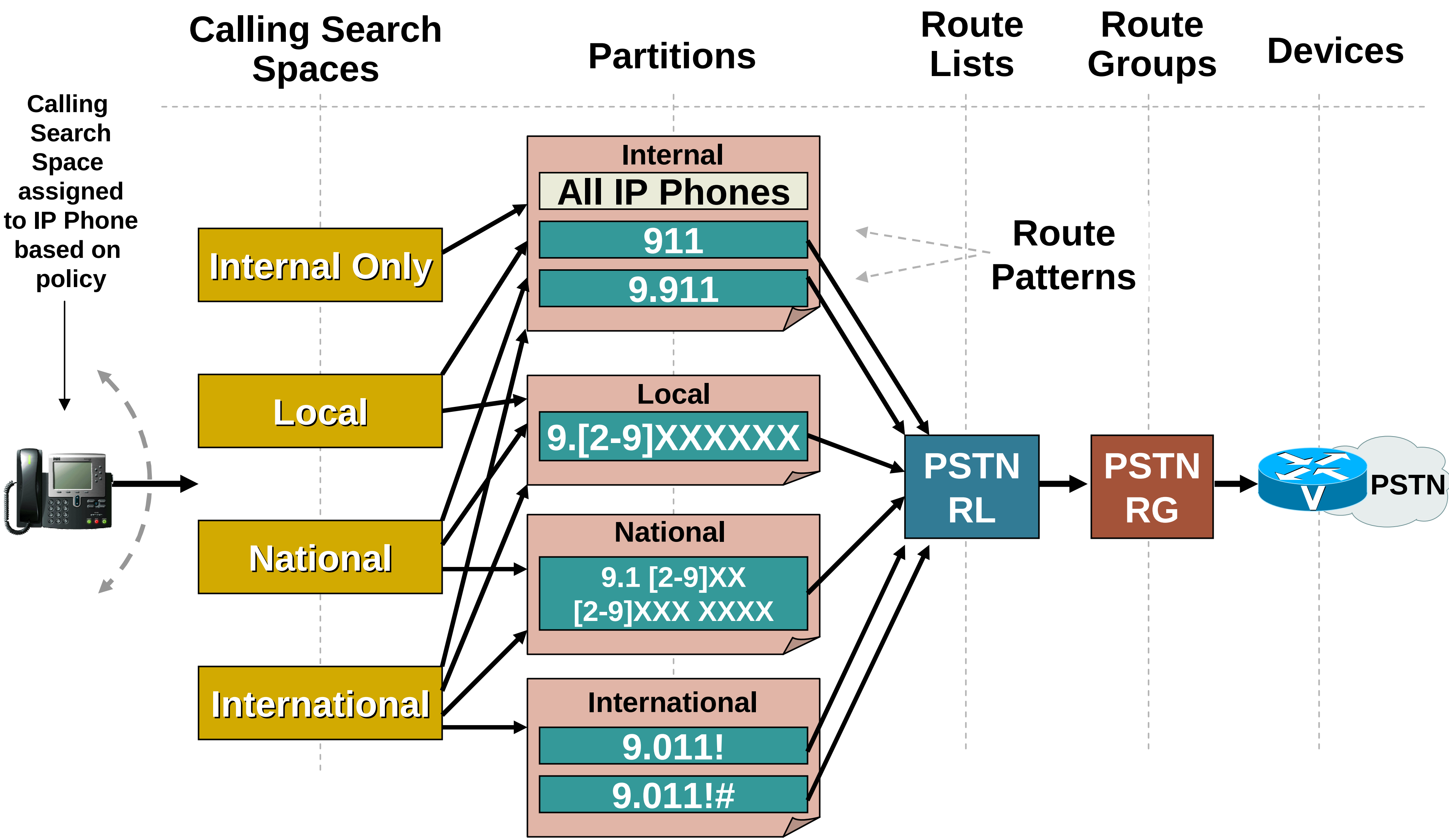
Building Classes of Service
Routing by User Class or Location



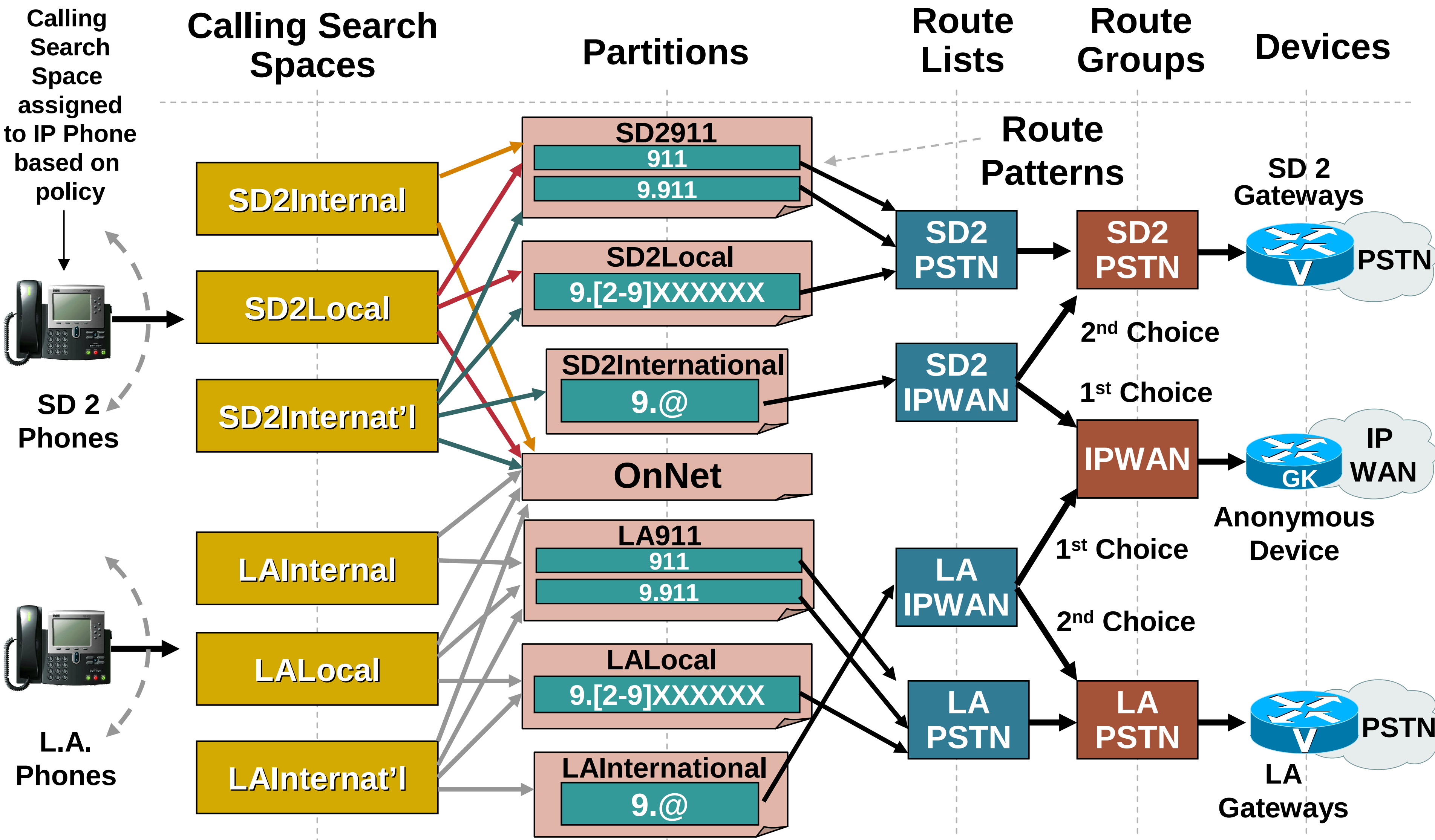
Create "Dial Plan Policy Groups" to Define Calling Restrictions

Instruct these Phones to Use Their Local Gateway for PSTN Access

Example of a Dial Plan



Another Example of a Dial Plan



Telephony Infrastructure Agenda (2/2)

- Call Admission Control
- Survivable Remote Site Telephony
- Dial Plan
- Security
- Management

Security
Areas of Focus

Server and Apps

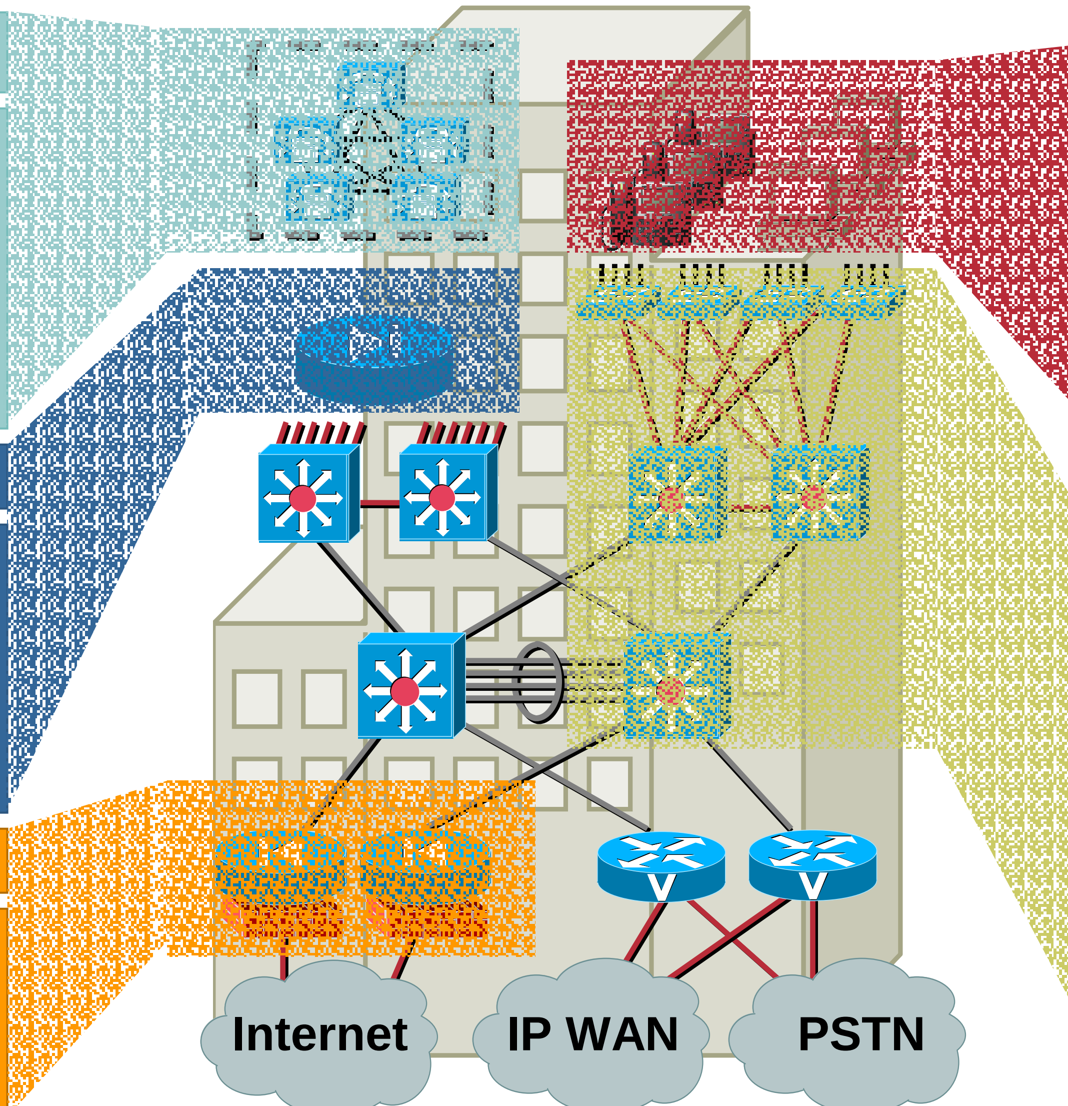
- OS updates
- Disable unused applications and services
- Toll fraud prevention measures

Firewalls

- Allow only required applications
- Control source addresses

Perimeter

- No NAT across Internet
- IOS DoS tools
- Use sensors



Endpoints

- Use separate addressing for voice and data
- RFC1918 is preferred

Network

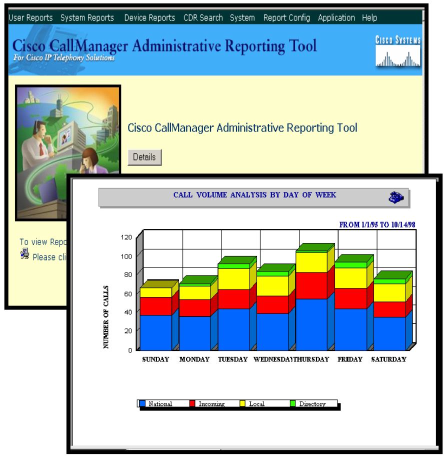
- Secure access (TACACS+, SSH, RADIUS)
- Use VLANs
- Use IP filters between voice and data network

Telephony Infrastructure Agenda (2/2)

- Call Admission Control
- Survivable Remote Site Telephony
- Dial Plan
- Security
- Management

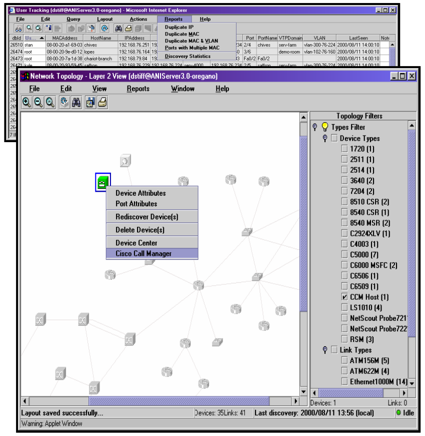
Management

Provisioning and Reporting



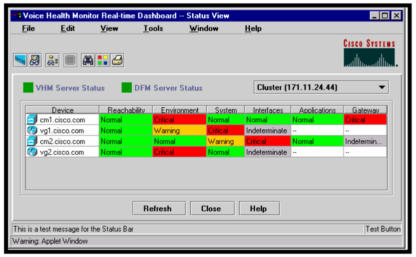
- **Provisioning:** BAT (IP phones), CVM (network), QPM PRO (QoS)
- **Reporting:** CAR

Element Management



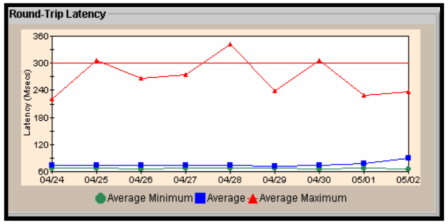
- CW2000: RME 3.2, Campus Mgr 3.1
- In-line powered switches, CCM
- Handset tracking
- CCM topology display

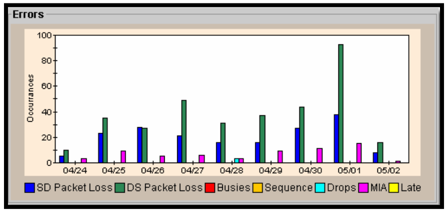
Fault Detection



- IP Telephony Manager (ITEM)
- Pro-active fault detection
- Real-time status reports on CCM, GWs, Apps

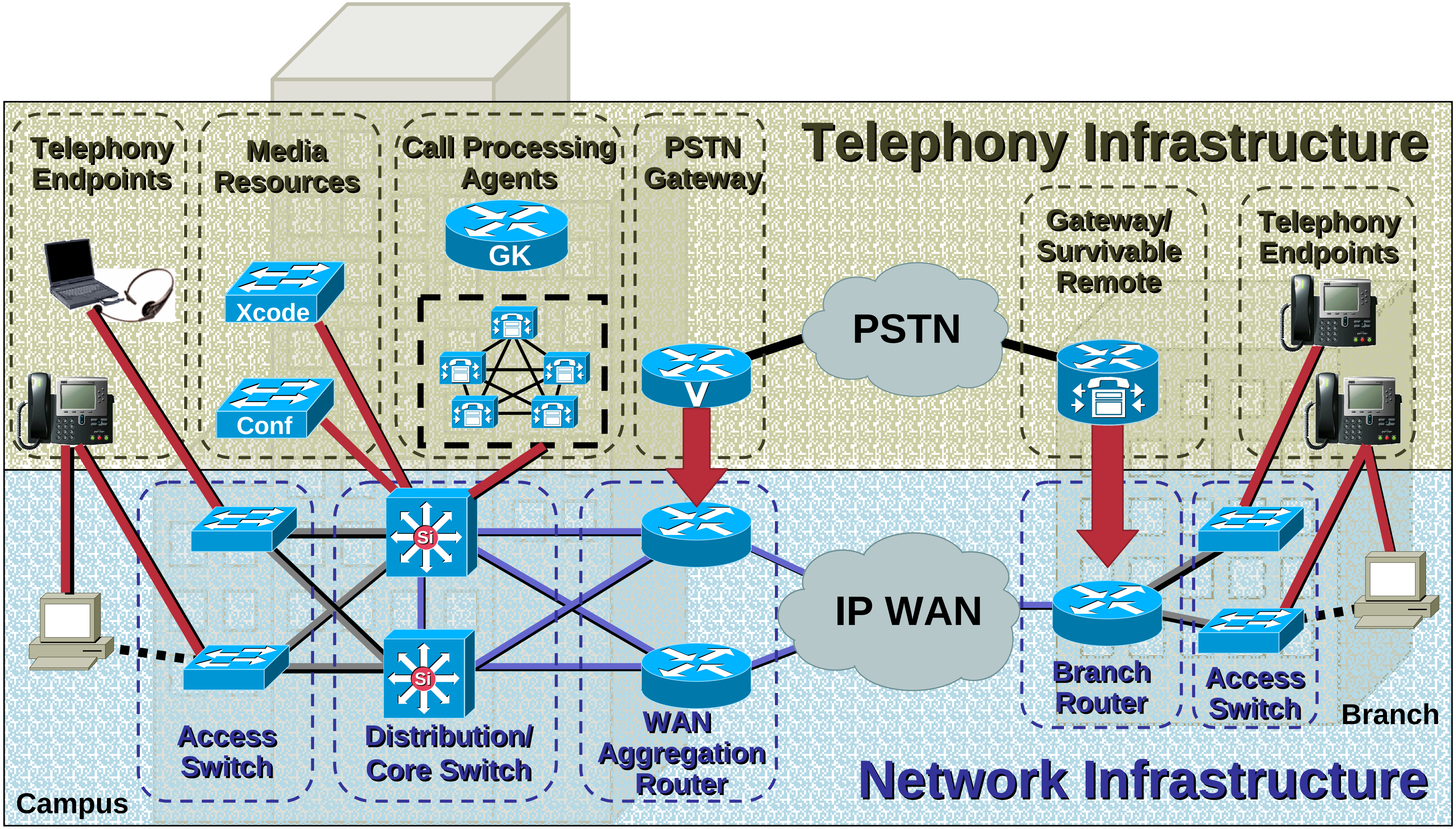
Performance Analysis





- IPM 2.2
- Real-time data on delay, jitter,…
- Generate alarms based on perf. thresholds

What We Have Built so Far



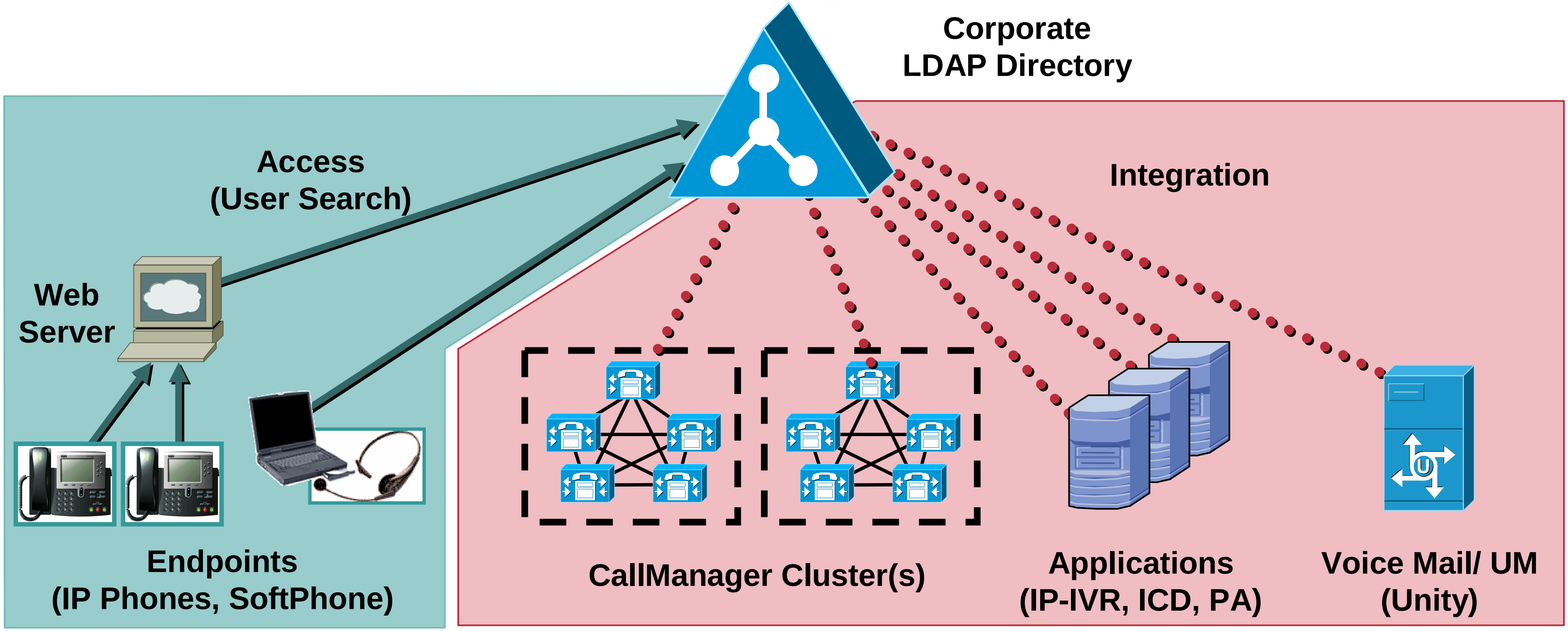
Agenda

- Introduction
- Network Infrastructure
- Telephony Infrastructure
- Applications

Application's Layer

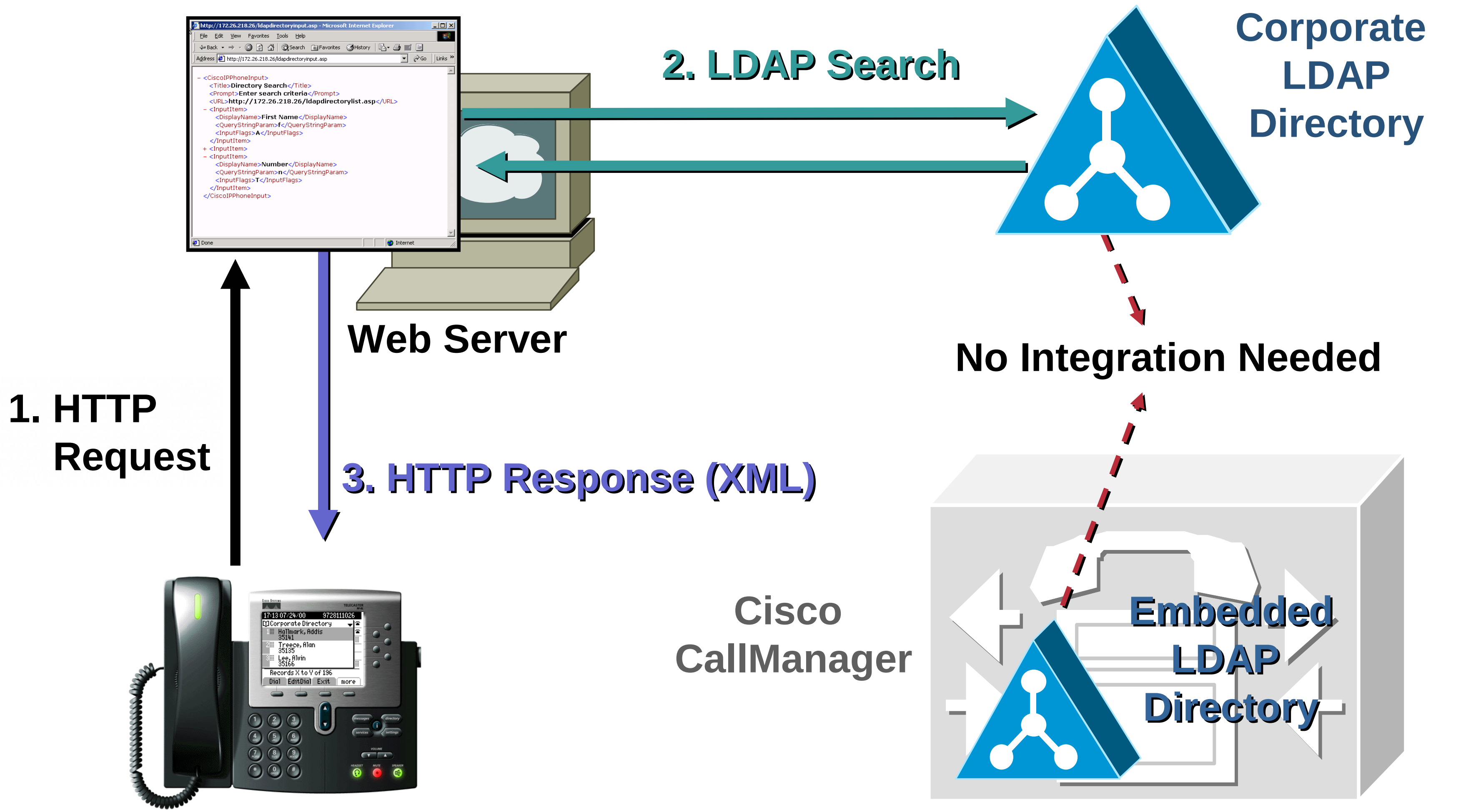
- LDAP/Directory Integration
- TAPI/JTAPI
- XML/Phone Services
- SCCP Based Applications

LDAP Directories
Directory Access and Integration

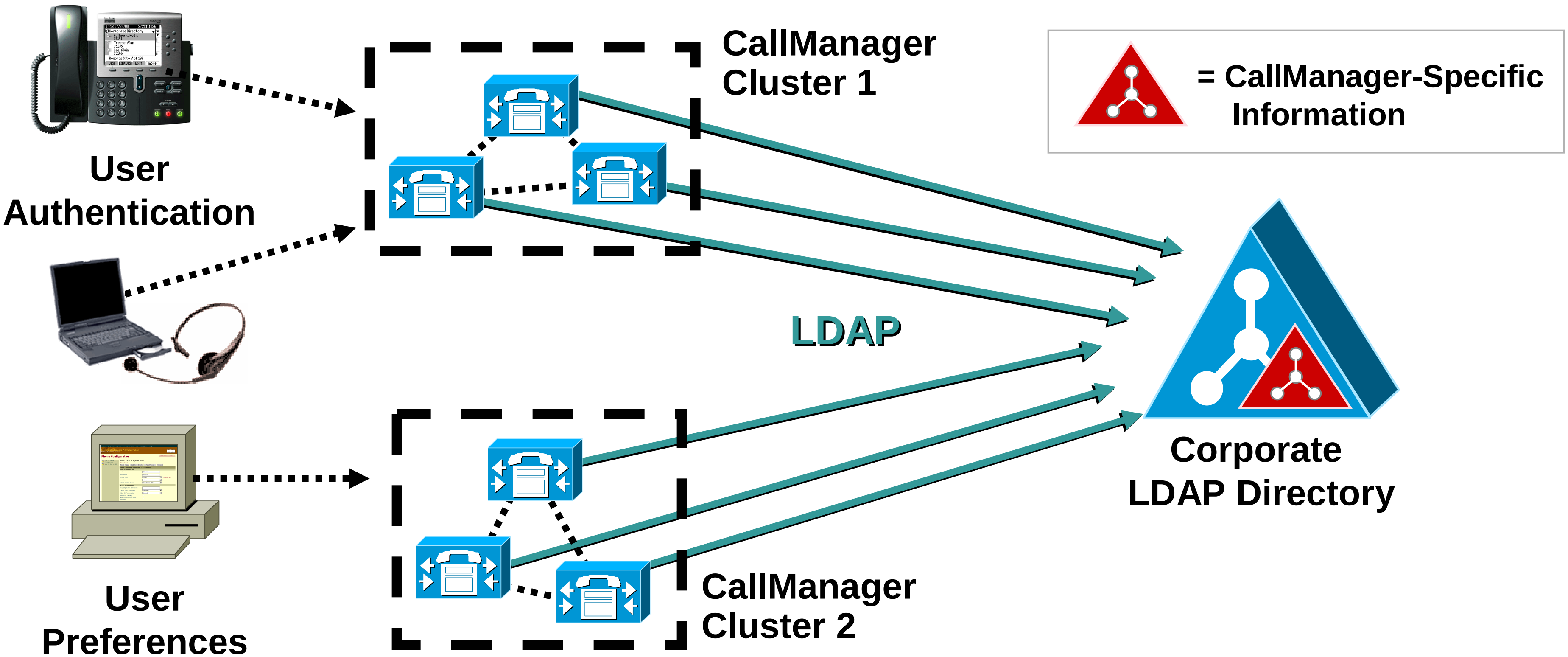


- **Directory access**: Endpoints enabled to search corporate directory
- **Directory integration**: User profile stored in a single repository—Single point of user authentication

LDAP Directories
Directory Access for IP Phones

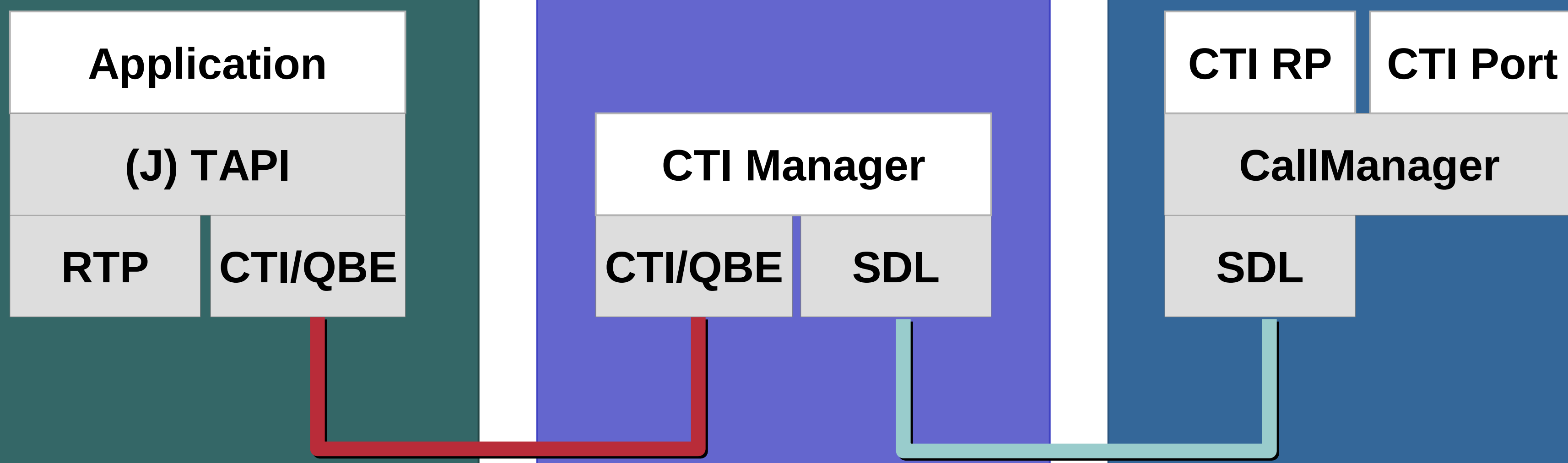


LDAP Directories
Directory Integration



- **Use the corporate LDAP directory instead of the embedded directory to store application-specific user information**
- **Supported directories: Microsoft active directory, netscape directory server**

(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 CallManager Cluster

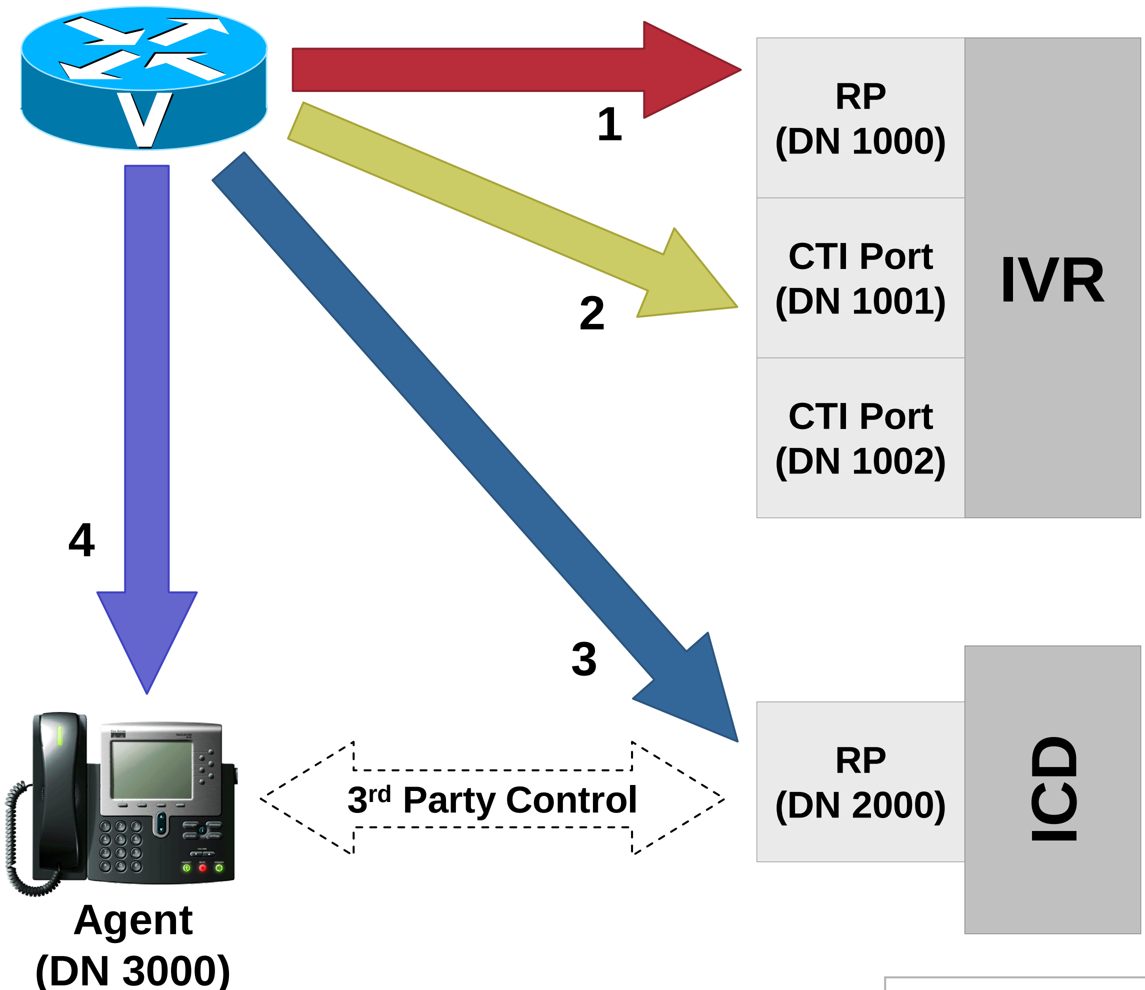
CallManager

Controls the CTI Devices and Provides Device Redundancy and Call Control

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

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

(J)TAPI and CTI Concepts
Joining All the Elements



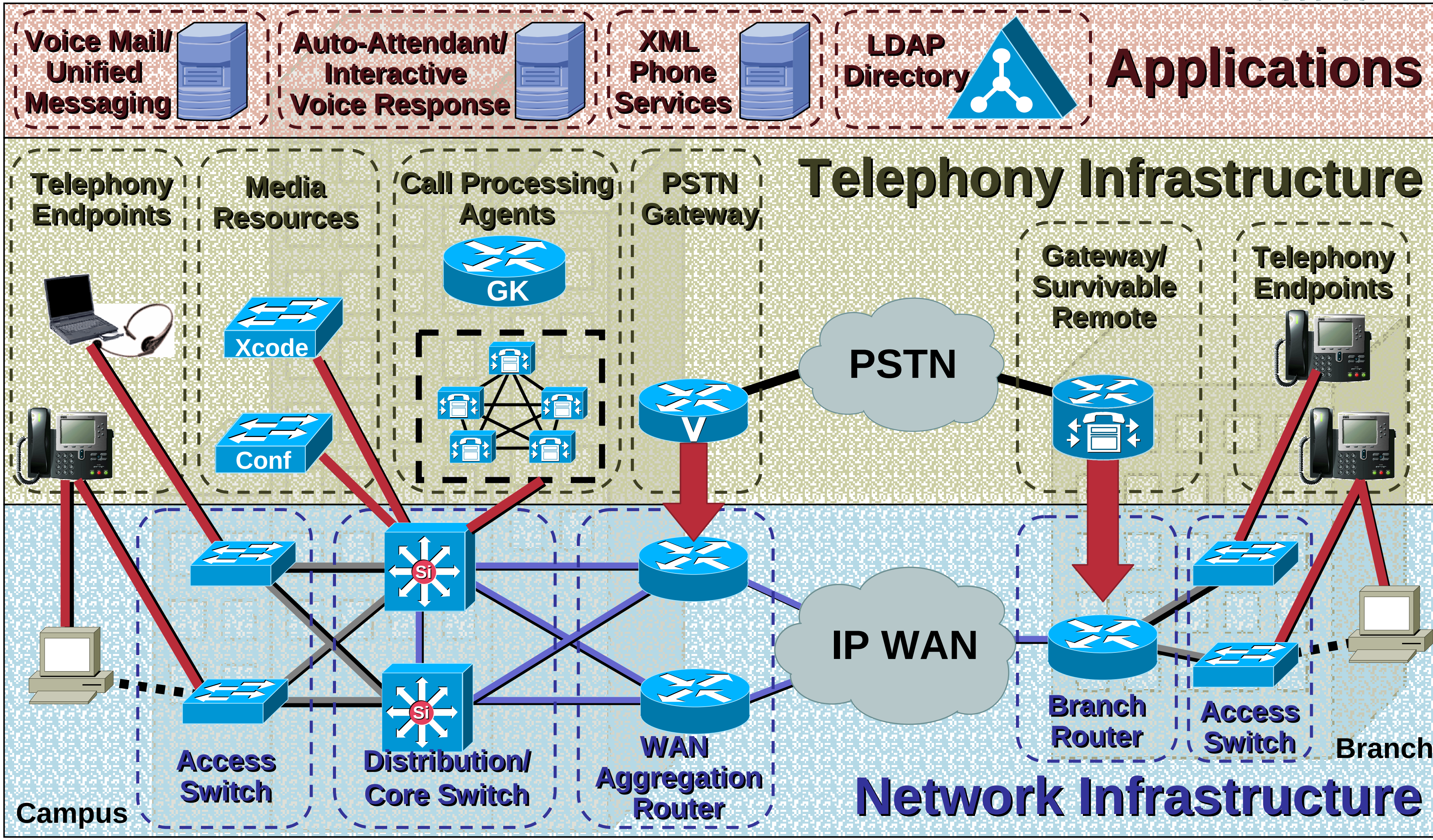
1. Incoming Call from GW to IVR's Route Point (Pilot Number 1000) IVR Application Redirect to a Free IVR CTI Port (1001)

2. IVR Answers Call on CTI Port; Plays Messages and Collects Information, Blind Transfers Caller to ICD Route Point (2000)

3. ICD Receives Incoming Call from IVR, Checks to Agent Is On-Hook via 3rd Party Control; Redirects Calls to Agent (3000)

IVR = Interactive Voice Response
ICD = Intelligent Call Distribution

What We Have Built so Far



Summary Conclusions

- What are the key components and requirements of an IP telephony solution
- How to build it:
  - Network infrastructure
  - Telephony infrastructure
  - Applications
- What are the design guidelines and recommendations

- IP Telephony is mainstream technology
- Key advantages are cost, flexibility and applications
- To learn more about IP Telephony design:

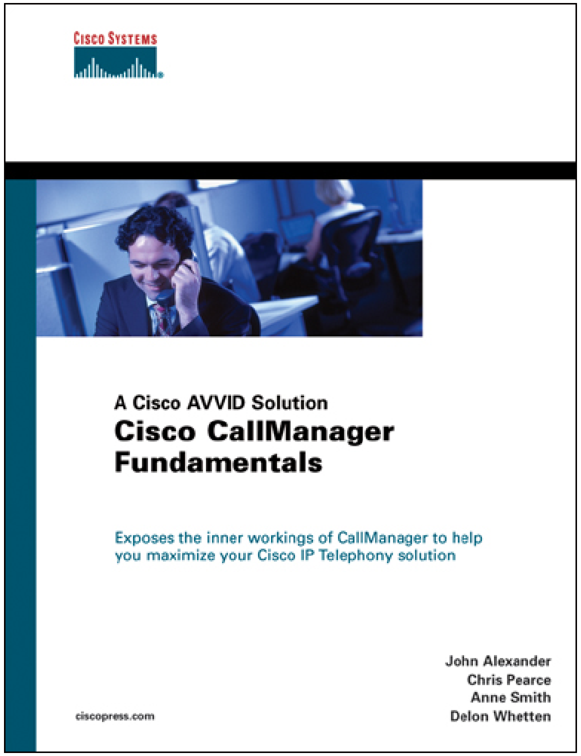
http://www.cisco.com/go/srnd/

Recommended Reading

Cisco CallManager Fundamentals
ISBN: 1587050080

Voice-Enabling the Data Network: H.323, MGCP, SIP, QoS, SLAs, and Security
ISBN: 1587050145

Voice over IP Fundamentals
ISBN: 1578701686



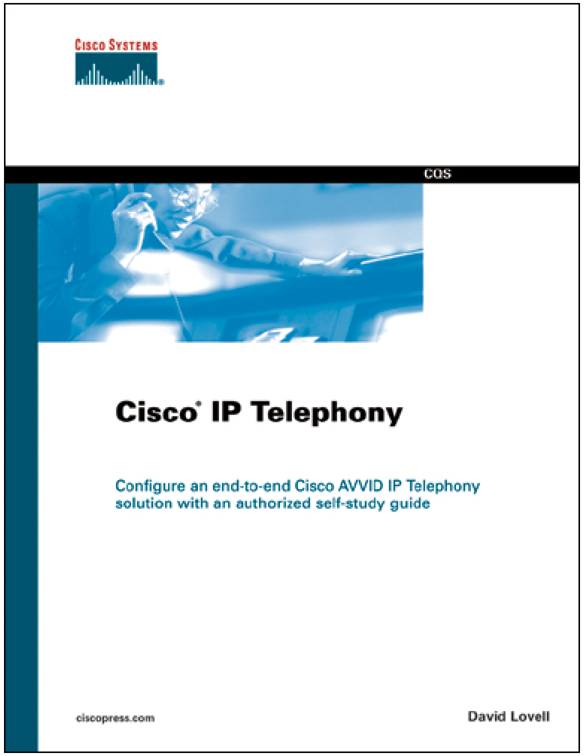
Available on-site at the Cisco Company Store

Recommended Reading

Cisco IP Telephony
ISBN: 1587050501

Cisco Voice over Frame Relay, ATM, and IP
ISBN: 1578702275

IP Telephony Unveiled
ISBN: 1587200759



Available on-site at the Cisco Company Store