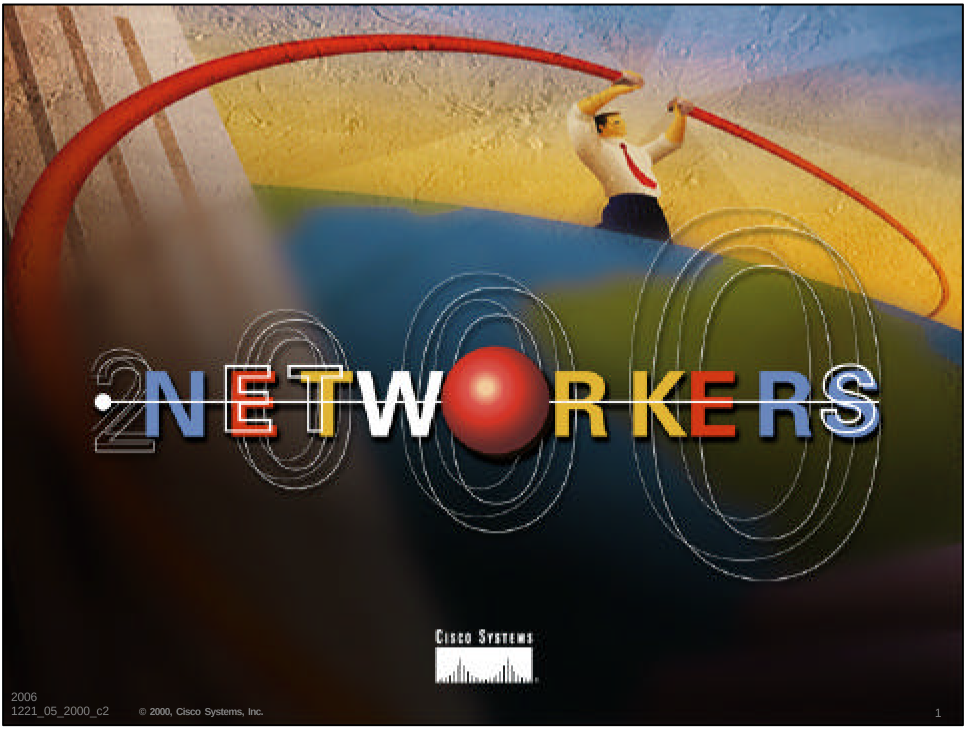
NETWORKERS
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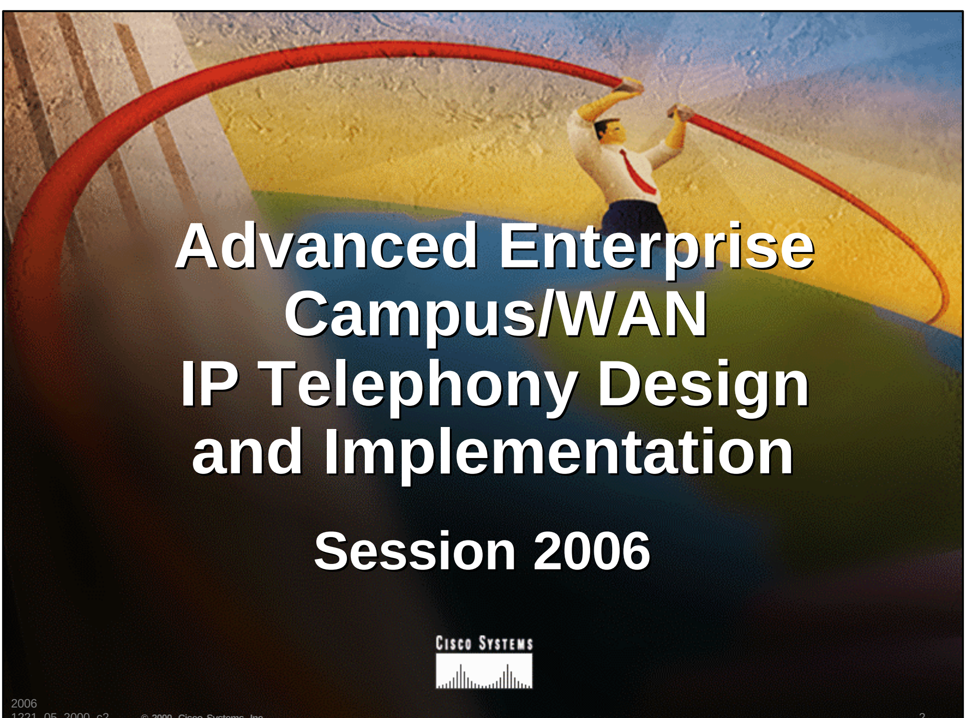
Advanced Enterprise
Campus/WAN
IP Telephony Design
and Implementation
Session 2006
Cisco Systems
2006
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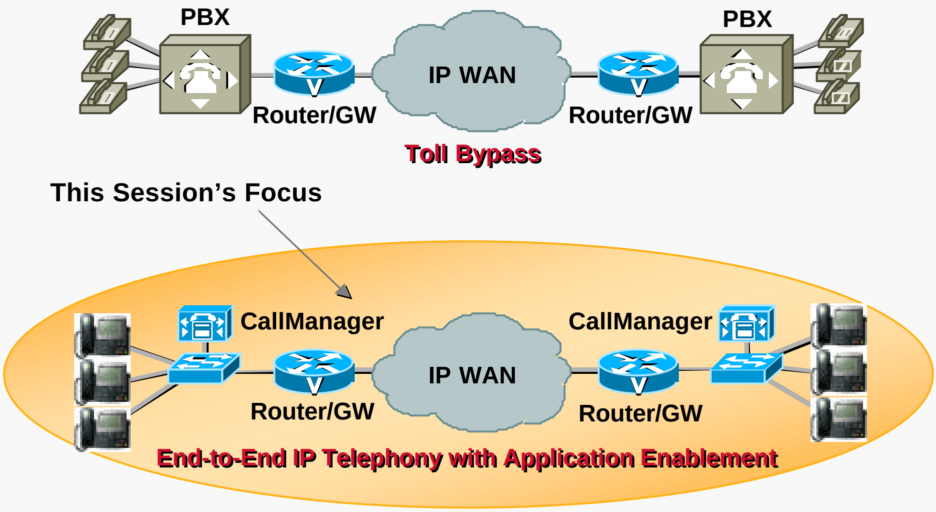
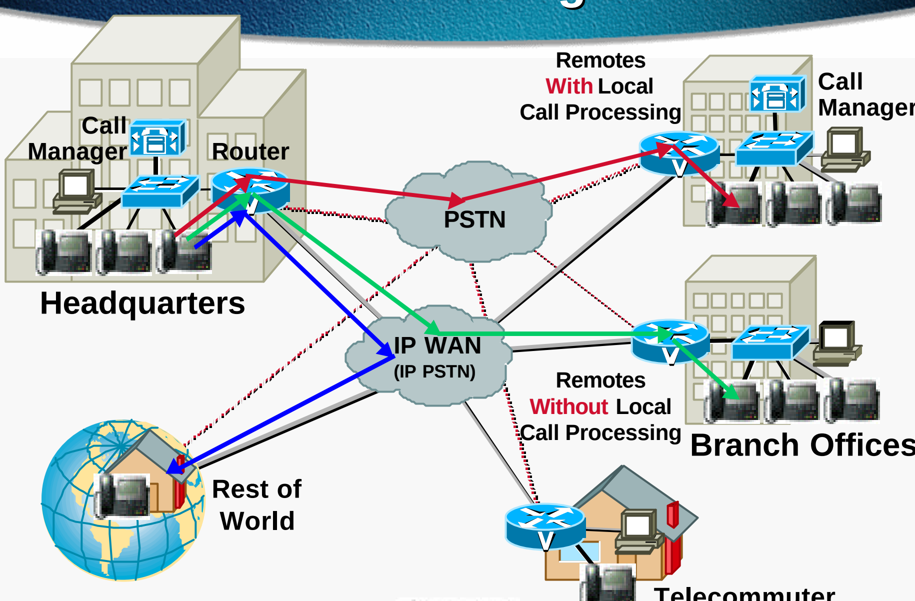
2

Solution Sets: Toll Bypass and IP Telephony



What We Are Going to Build



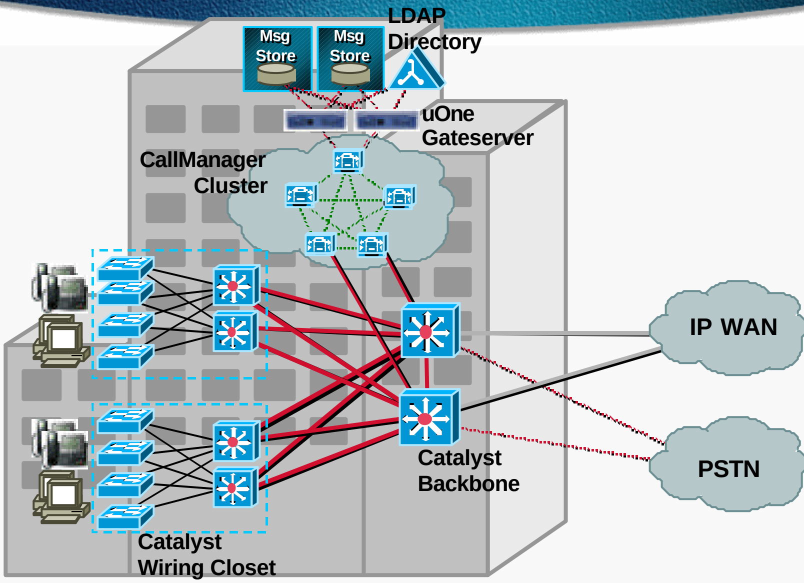
Session Objectives

- To be able to select, design and implement the appropriate IP telephony deployment model based on user requirements
- Designs based on CallManager 3.0(1)

Agenda

- General Enterprise Deployment Models
- Campus Design Considerations
- Multisite WAN Considerations
- Legacy Migration Strategies

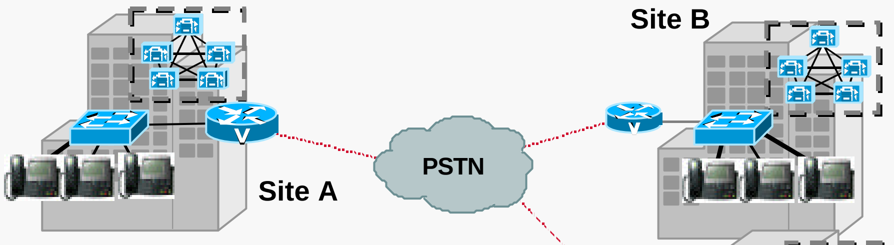
Single Site Deployments



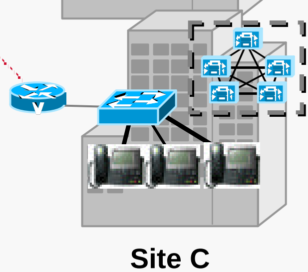
Single Site Characteristics

- Support for 10,000 users
- Robust switched network design
- CallManager cluster for redundancy and scaling
- Inline power to phone sets
- Single cable for phone and PC
- Quality of service from the desktop
- Ease of IP addressing plus adds, moves, and changes

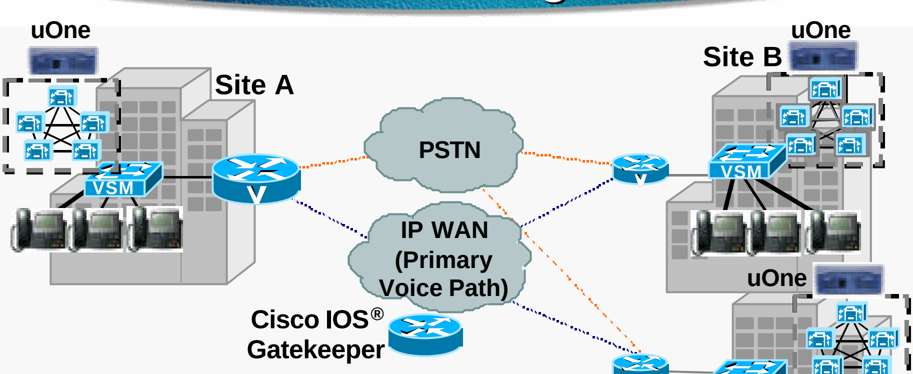
Isolated Deployments



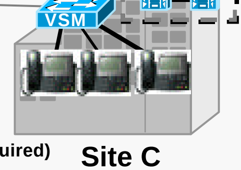
- Call processing (CallManager/cluster at each site)
- Voice message and DSP resource at each site
- 10,000 users per site
- PSTN used for all external calls
- No limit to the number of Sites
- Compressed voice not required
- Uniform dial plan



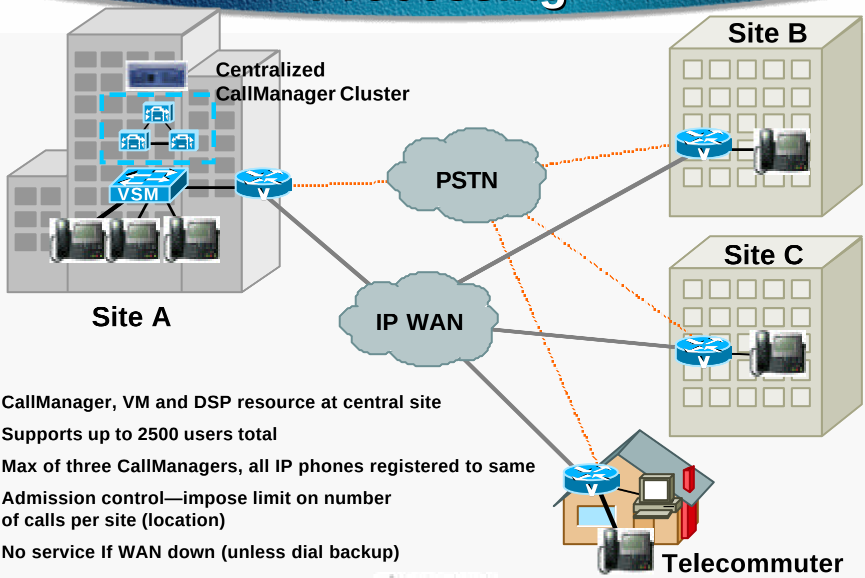
WAN—Distributed Call Processing



- CallManager, voice message and DSP resource at each site
- 10,000 users per site
- 10 sites maximum networked via IP WAN
- Compressed voice between sites
- Admission control—h.323 Gatekeeper Based (Cisco IOS Gatekeeper required)
- Transparent Use of PSTN If IP WAN unavailable



WAN—Centralized Call Processing

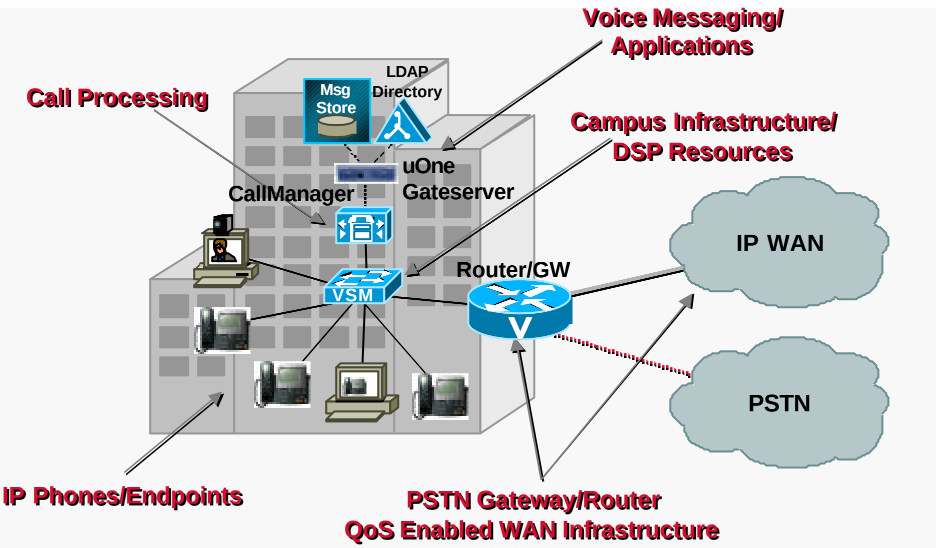


- CallManager, VM and DSP resource at central site
- Supports up to 2500 users total
- Max of three CallManagers, all IP phones registered to same
- Admission control—impose limit on number of calls per site (location)
- No service If WAN down (unless dial backup)

Agenda

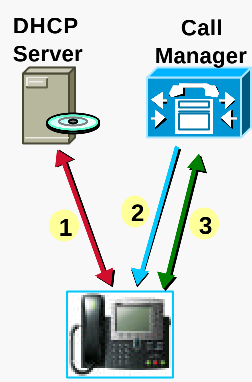
- General Enterprise Deployment Models
- Campus Design Considerations
- Multisite WAN Considerations
- Legacy Migration Strategies

IP Telephony Components



IP Phone Initialization

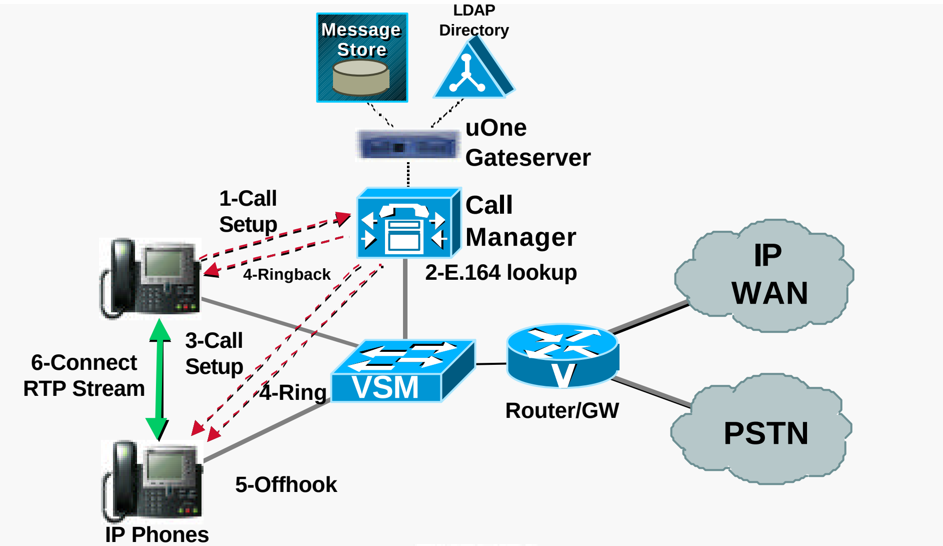
1. Phones make DHCP request to get an IP address, gateway, boot server, etc.
2. Phones make TFTP boot file request to get CM IP addresses
3. Phones register with CM and get Display Templates and ready to receive/place calls



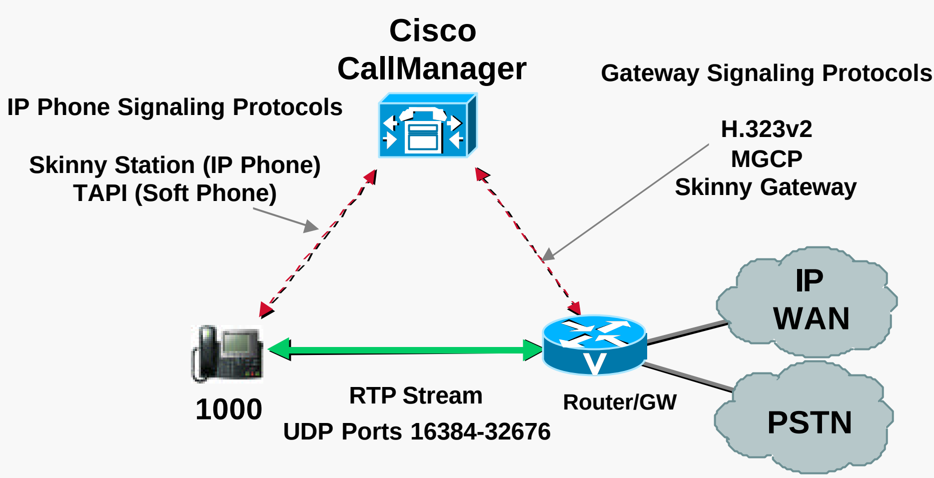
Ease of Moves, Adds, and Changes

- Add a new device
  - Plug it in out of the box
- Move a device
  - Unplug and plug in new location
- Changes
  - Simple web based interface

Basic Call Processing

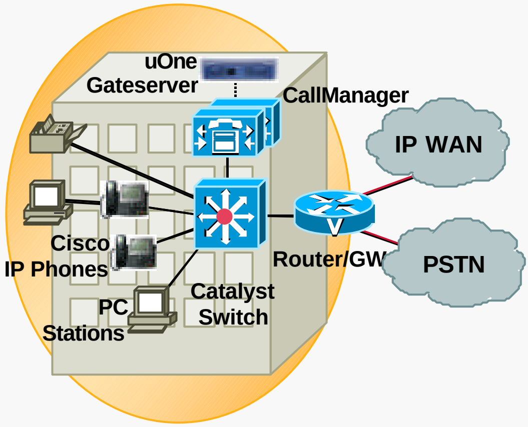


CallManager Call-Control Protocols

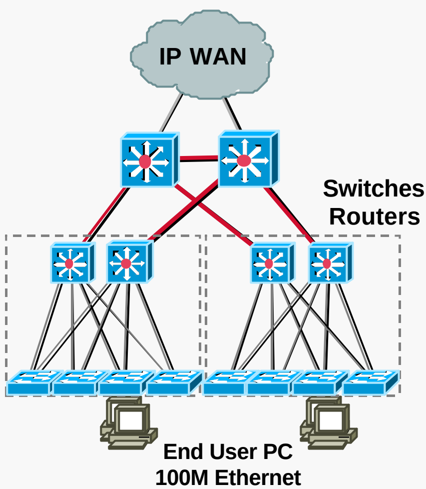


Primary Campus Site Considerations

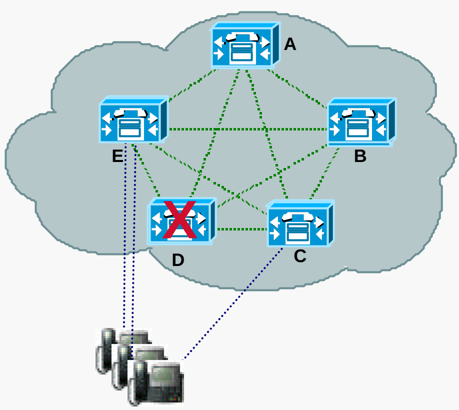
- QoS enabled infrastructure
- Call processing
- Dial plan
- Voice mail



System Redundancy



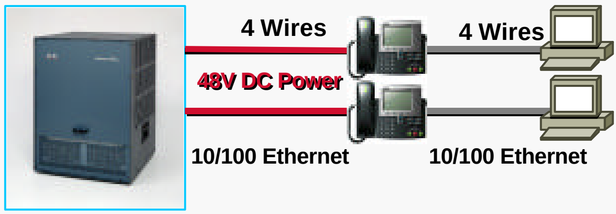
Network Design



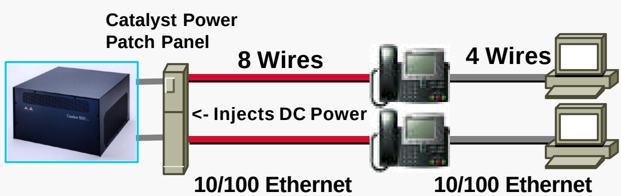
CallManager Clustering

Single Wire and Power Options

New in-line power on Catalyst® Switches



Catalyst Switch with regular 10/100 Ethernet Line Cards



IP Address Plan

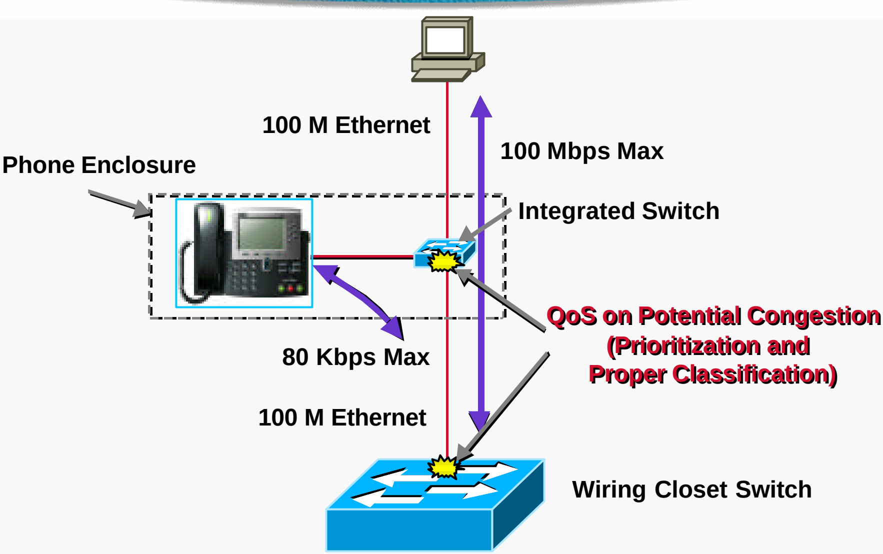
- IP phones need addresses too!
  - Configure phones statically or use DHCP
- Address space options:
  - Double current address space
  - Phones on separate subnets
  - Secondary addressing per subnet
  - Use of RFC addresses for “voice” subnet
- Phones don’t work across NAT/PAT/ firewall boundaries today

Automatic Subnet Placement

Catalyst Multiservice Port Provides Automatic Phone VLAN Configuration

- No end-user intervention required
- Provides the benefits of VLAN technology for the phone
- Preserves existing IP address structure
- Uses standards-based 802.1Q technology between switch and phone

Potential Campus Congestion

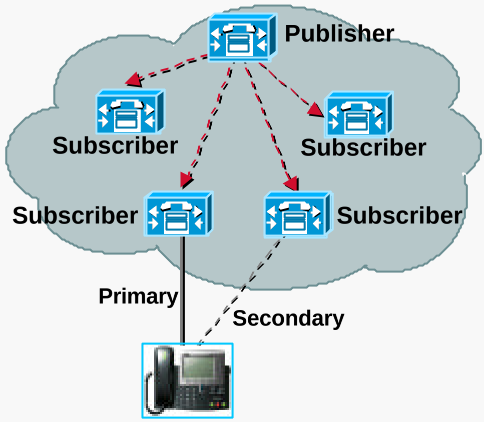


CallManager Clusters N+1 Redundancy

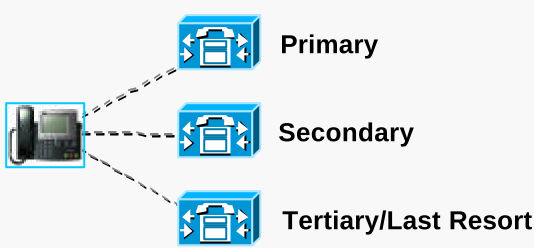
CallManager Cluster Characteristics

- Appears as one distributed CallManager
- 2500 users maximum per CallManager (even under failure conditions)
- Maximum of 10,000 users in a cluster
- Maximum of six CallManagers in a cluster
- Cluster members are confined to a campus

SQL 7.0 Database



CallManager Failover



- Each device (IP phone + skinny gateway) has a prioritized list of up to three CallManagers to which it can connect
- This is called a CallManager Group and this list is downloaded during device initialization

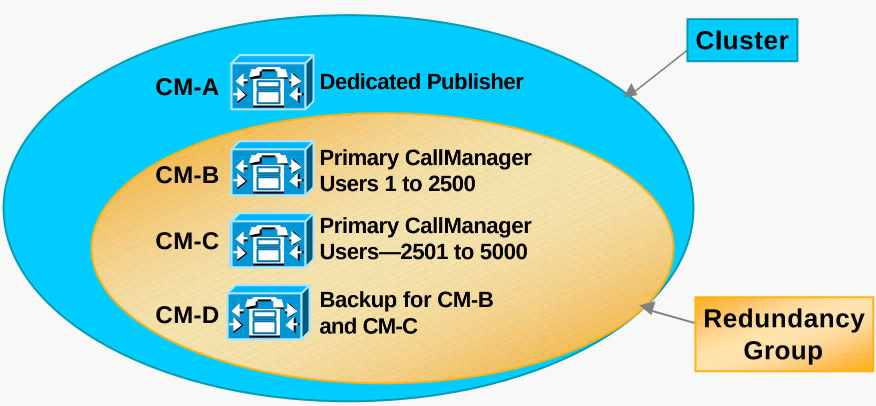
Cluster Recommendations Up to 2500 Users (with Redundancy)

- A cluster of two CallManagers

  Single active CallManager

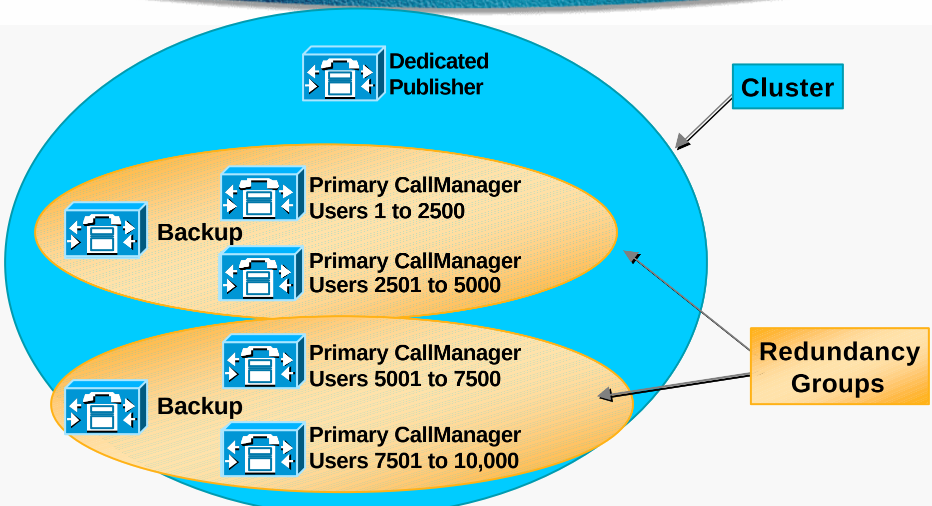
  Dedicated publisher also acts as a standby

Cluster Recommendations Up to 5000 Users (with Redundancy)



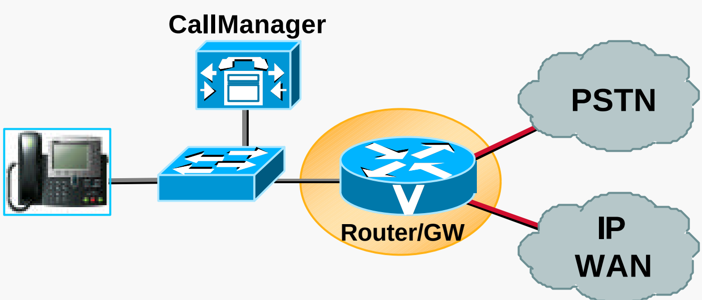
- Every IP phone would have a CallManager group consisting of two CallManagers (primary and backup)

Cluster Recommendations Up to 10,000 Users (with Redundancy)



- Every IP phone would have a CallManager group consisting of two CallManagers (primary and backup)

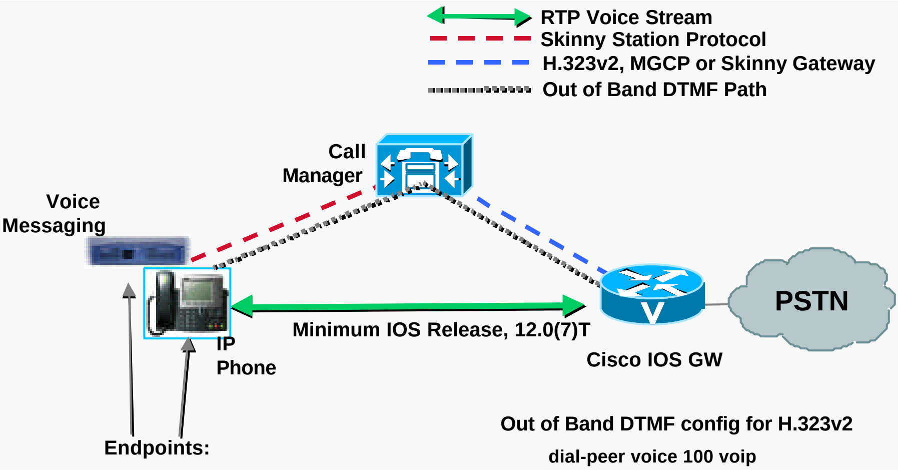
Selecting the Proper Gateway



Gateway Selection Criteria

Voice Port Density Requirements
Support for required PSTN Signaling Types
Support for Required WAN interfaces and QoS

Gateway Requirements “Out of Band” DTMF Support

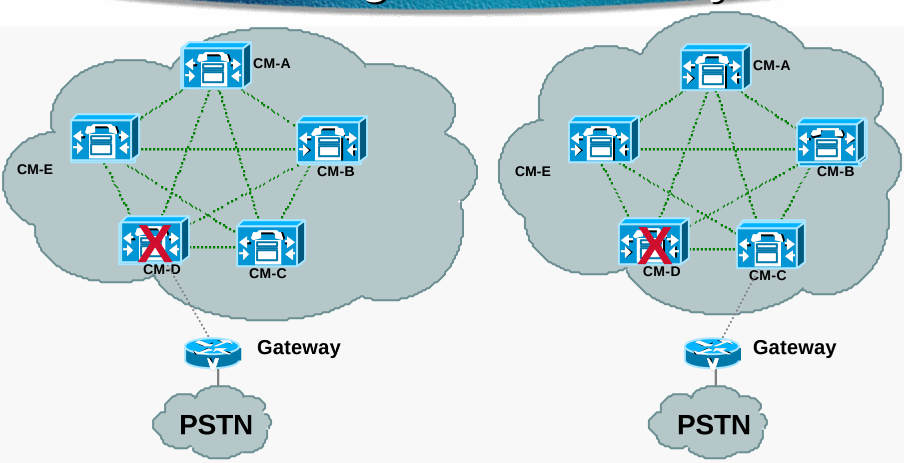


Many require “Out of Band DTMF”
Prevents inband DTMF distortion

Out of Band DTMF config for H.323v2

```
dial-peer voice 100 voip
destination-pattern 555….
session target ipv4:10.1.1.1
dtmf-relay h245-alphanumeric
```


Gateway Requirements CallManager Redundancy



Primary CallManager for GW fails

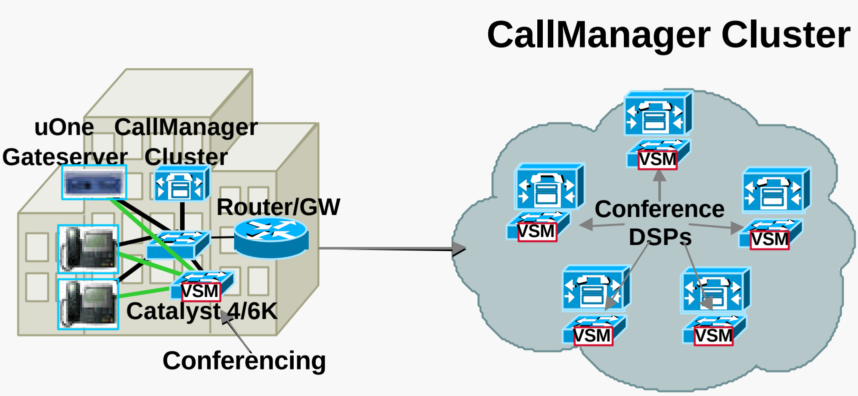
MGCP/Skinny GWs rehome to Secondary CallManager H.323v2 falls back to alternate VoIP peer

Gateway Selection Criteria PSTN/PBX Signaling Support

- T1/E1—CAS, PRI
  - Cisco 1750/2600/3600
  - DT-24/30+, VG200
  - Cisco AS5300
  - Catalyst 4xxx/6xxx
  - Cisco 7200/7500
- E1 R2
  - Cisco AS5300 only
- BRI or analog E&M
  - Cisco 1750, 2600, 3600
- Analog FXO or FXS
  - AT/AS + VG200
  - Cisco 1750, 2600, 3600

(Standalone Gateways)

DSP Resource Provisioning (Conferencing)



- VSM—Voice Services Module

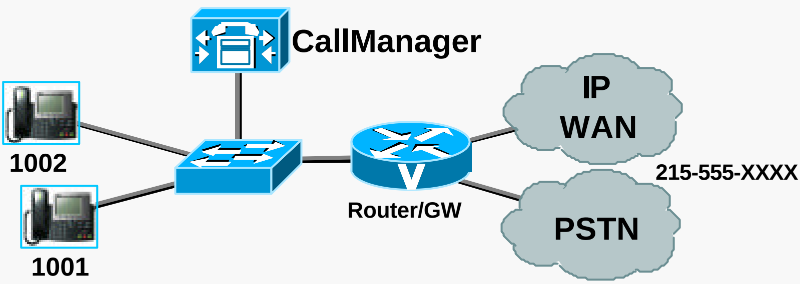
DSP Resource Required per CallManager

CallManager Dial Plan Functions

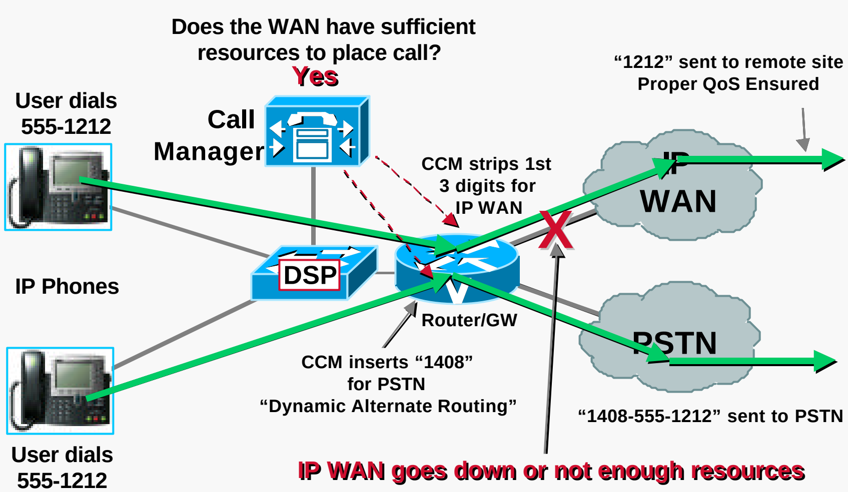
Flexible Call Routing—Multiple paths to destination

Digit Manipulation—Adding and stripping digits

Call Restrictions—Who can dial where

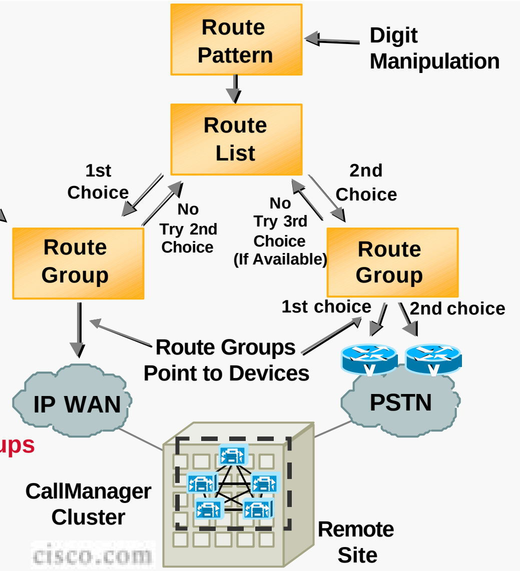


Dial Plan Goal Transparent Automatic Route Selection



CallManager Dial Plan Architecture

Digit Manipulation on
a per Route Group Basis
(Overrides the Route Pattern)



Devices assigned in Route Groups

1. Gateways
2. Remote CallManagers

Definition by Function

- **Route patterns**

  “Match” of an E.164 address range or specific address

  Points to a single “route list”

- **Route lists**

  How to “reach” a destination via prioritized route groups

  Multiple route patterns may point to single route list

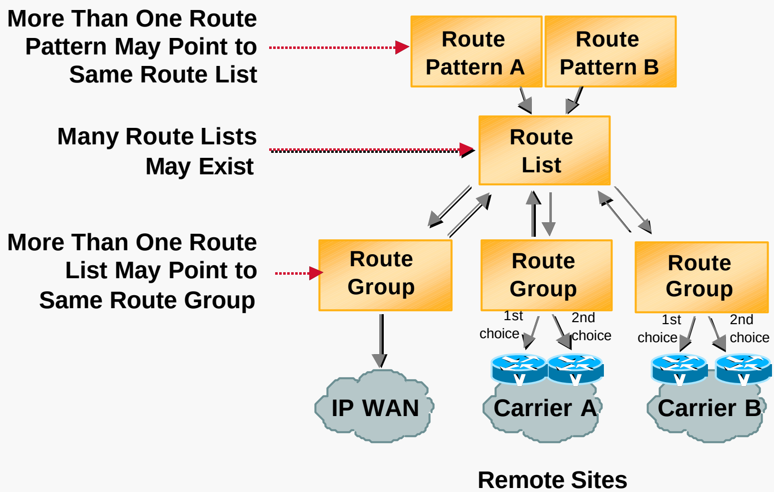
- **Route groups**

  Forms a prioritized “trunk group” by pointing to devices

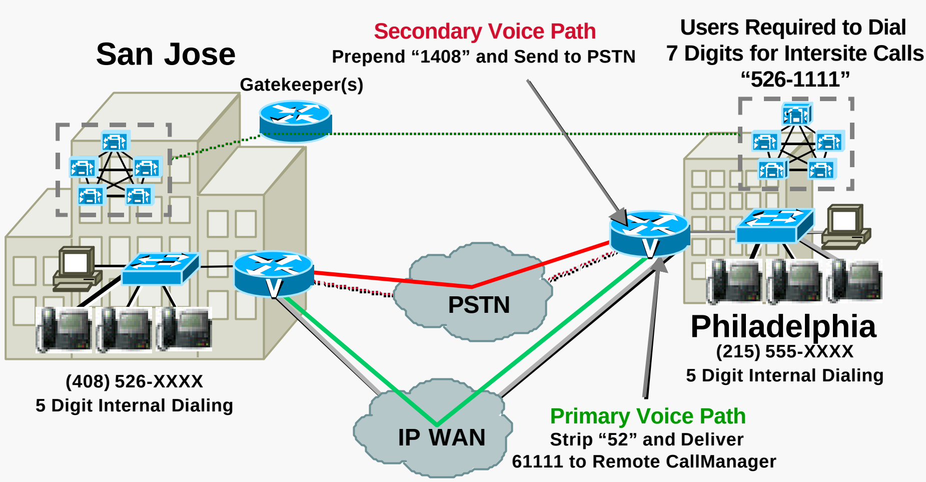
- **Devices**

  Gateways or remote CallManagers

Dial Plan Characteristics

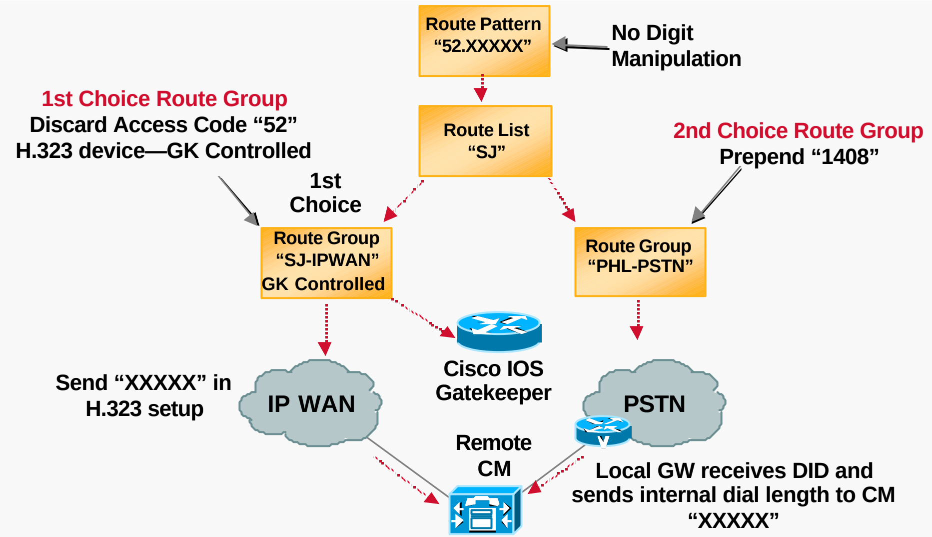


Dial Plan Example Configuration "Distributed Call Processing"

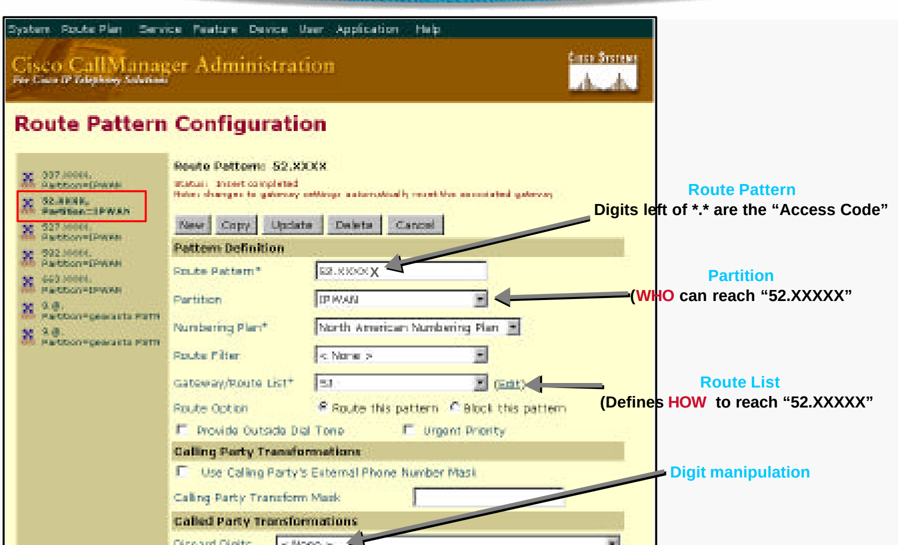


Five Digit Internal Dialing within a Site
Seven Digit Dialing "Between" Sites

Philadelphia Route Pattern Configuration



Route Pattern Configuration

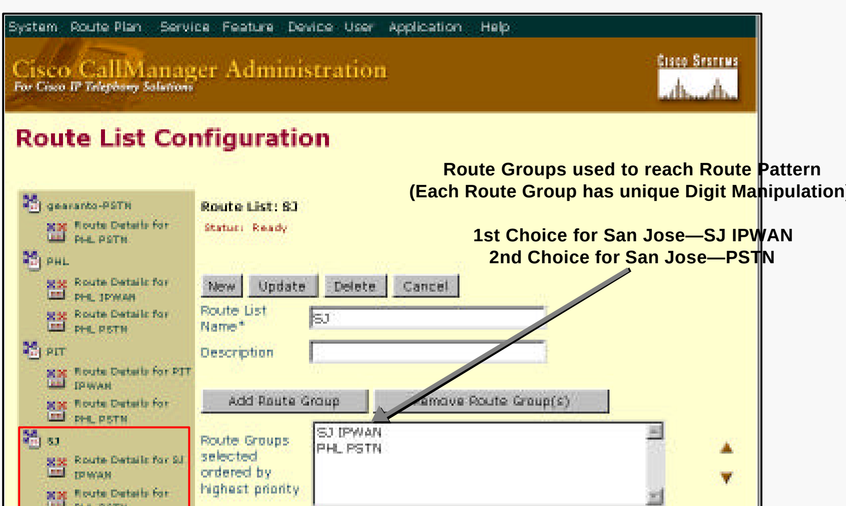


Route Pattern Notes

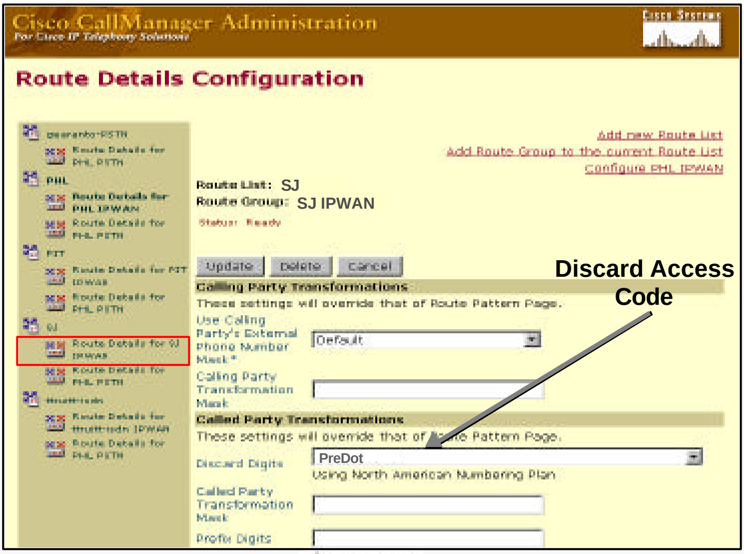
- **CallManager matches most specific pattern**
  - **IP phone 1111 will match before a route pattern of 11XX**
- **Wildcards**

X	**Single digit (0-9)**
@	**North American numbering plan**
!	**One or more digits (0-9)**
.	**Terminates access code**

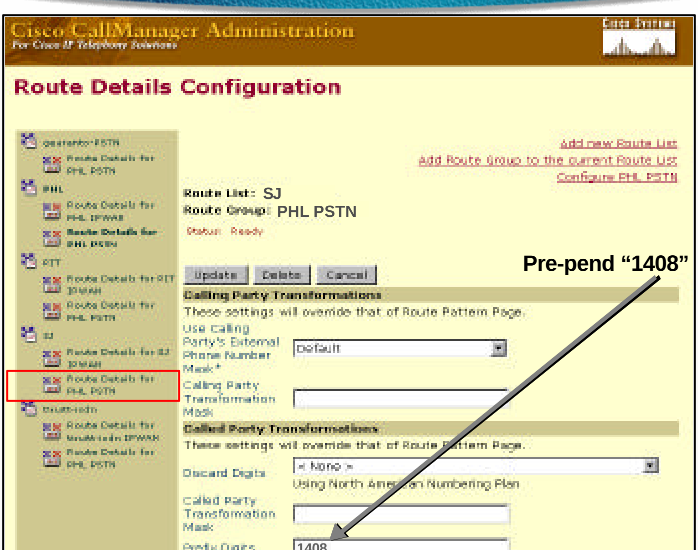
Route List Configuration



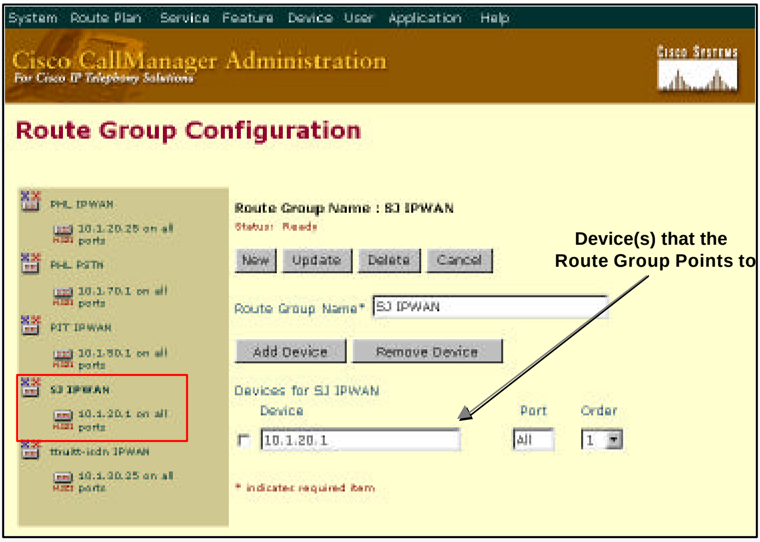
“SJ IPWAN” Route Group “Settings Override That of Route Pattern”



“PHL PSTN” Route Group

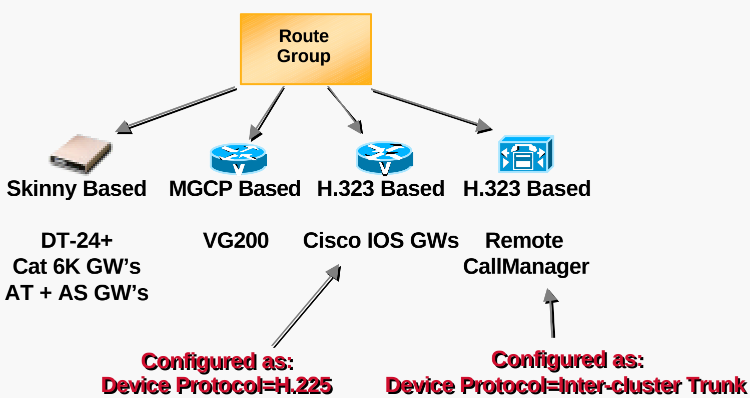


Route Group "Prioritized Trunk Group"



  cisco.com

Typical Route Group Device Types



  cisco.com

Minimal Gateway Configuration

For Incoming Calls Assuming DID (Direct Inward Dial)

```
dial-peer voice 1 voip
 CODEC g711ulaw
 dtmf-relay h245-alphanumeric
 destination-pattern 6....
 session target  ipv4:10.1.10.5
!
dial-peer voice 2 pots
 destination-pattern .......
 port 1/0:1
!
dial-peer voice 3 pots
 destination-pattern 1..........
 prefix 1
 port 1/0:1
!
dial-peer voice 4 pots
 destination-pattern 911
 prefix 911
 port 1/0:1
```

- Dial-Peer for all incoming calls from PSTN to Call Manager's IP Address Must be G.711
- CallManager's IP Address
- Dial Peer for all 7 digit outgoing PSTN Numbers
- Dial Peer for all 10 digit outgoing PSTN Numbers
- Dial Peer for 911 Services

For Outgoing Calls

Note - For "NON DID" Incoming Calls (Analog trunks)

1. Dedicate each incoming trunk to user with "Connection PLAR"

or

2. Send all calls to an Attendant/Auto Attendant

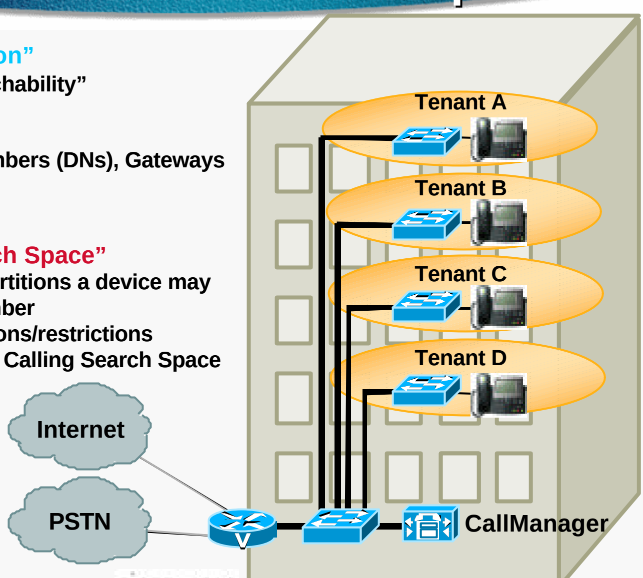
Multitenant and Call Restrictions Creation of "Dial Plan Groups"

"Partition"

1. Devices with similar "reachability" characteristics
2. Items placed in Partition:
   IP Phones, Directory Numbers (DNs), Gateways + Route Patterns

"Calling Search Space"

1. Set of "Rules"—Which Partitions a device may search in for a dialed number
2. Provides dialing permissions/restrictions
3. Each device "assigned" a Calling Search Space

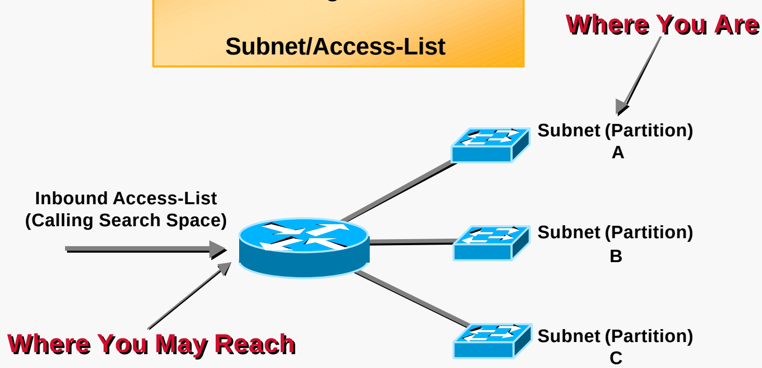


Understanding Partitions and Calling Search Spaces

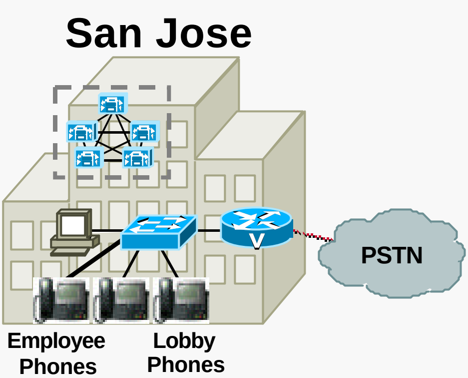
Partition/Calling Search Space

Analogous to:

Subnet/Access-List



Configuring Calling Restrictions



Access Code of “9” for local PSTN calls

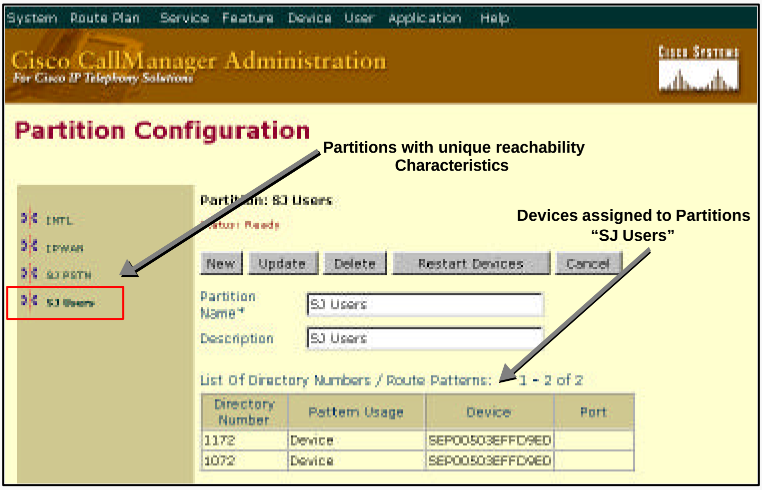
Partition Assignment
“SJ-Users” = All SJ IP Phones
“SJ-PSTN” = “9” Route PatterN

Calling Search Space
“Unrestricted” = SJ-Users, SJ-PSTN
“SJ-Only” = SJ-Users

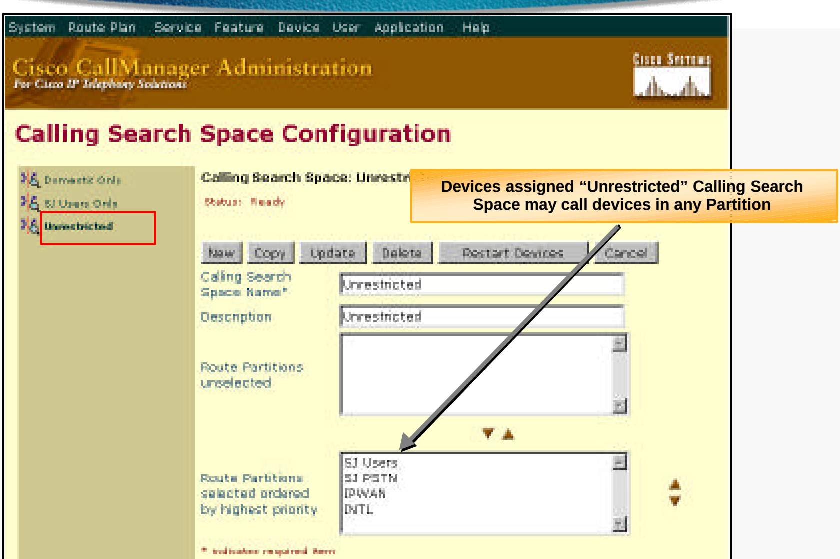
IP Phone Calling Search Space Assignment
Staff IP Phones = “Unrestricted”
Lobby IP Phones = “SJ-Only”

Employees—May dial anywhere
Lobby Phones—Only can dial internal to SJ

Partitions Devices Placed in a Partition

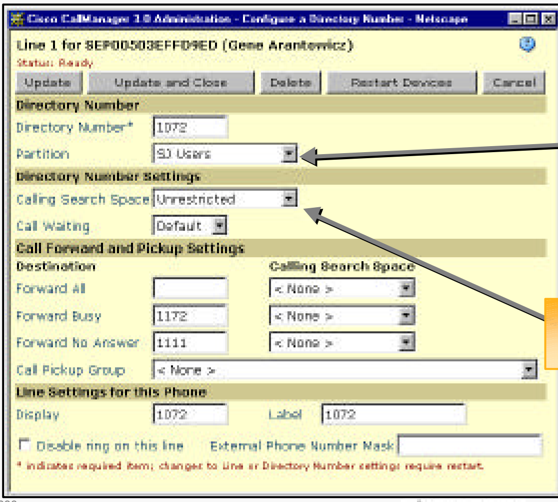


Calling Search Space: "Where I Can Dial"



Assigning Partitions and Calling Search Spaces "Individual Line/DN Level"

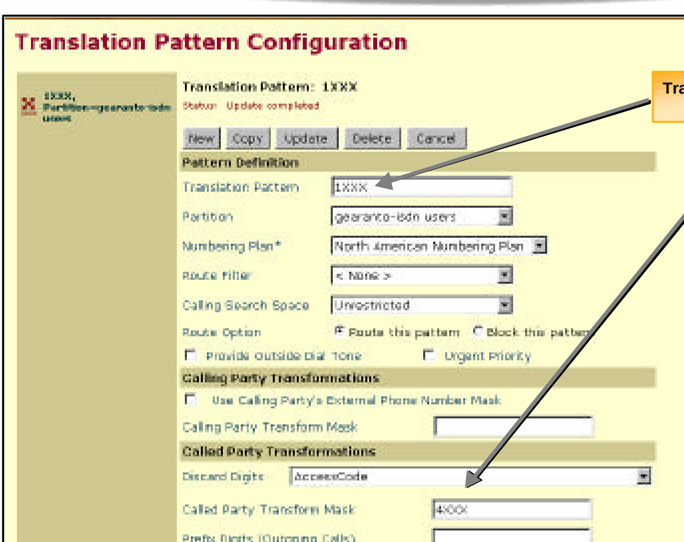
Individual Line Configurations Override Main Configuration



"Partition"
Assigned at the Individual DN configuration level

"Calling Search Space"
Can be assigned to Individual DN
(Overrides Main Phone Configuration settings)

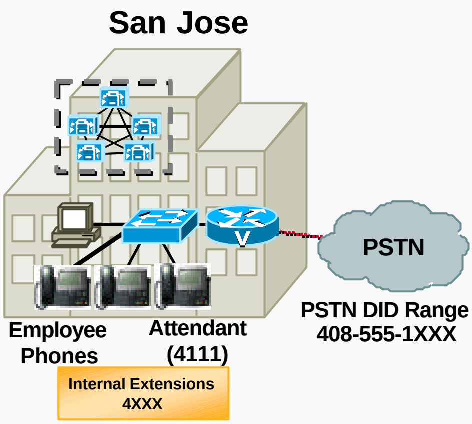
Configuring Digit Translation



Translates anything dialed with "1XXX" to 4XXX

Dialed Digit Translation

Common Uses



Can be used to send calls to unassigned DID numbers to Attendant/Recording

Translation Pattern = XXXX
Called Party Transform Mask = 4111

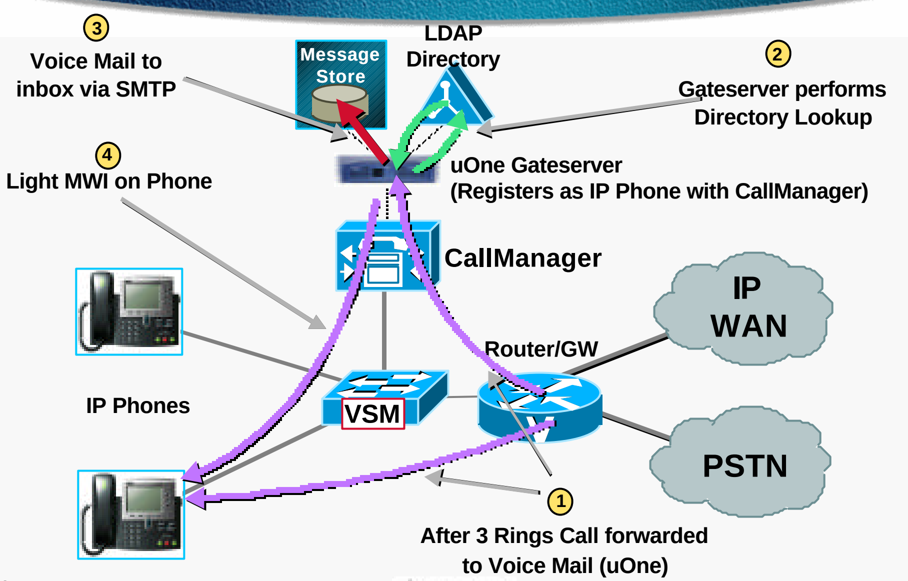
CallManager uses Longest match so "XXXX" will match any non-configured number and get sent to 4111 (Attendant)

OR

PSTN DID Range does not match internal Range

Translation Pattern = 1XXX
Called Party Transform Mask = 4XXX

Voice/Unified Messaging



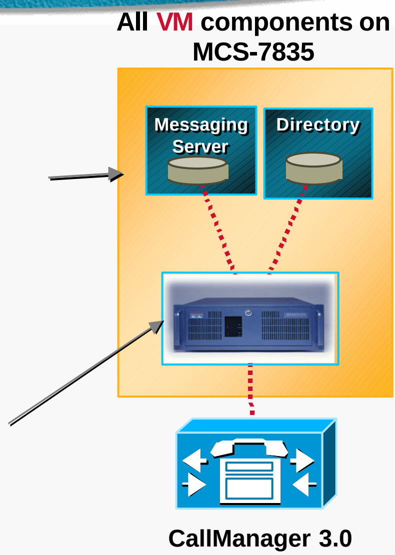
CallManager 3.0 and uOne 4.1E Corporate Edition

MCS-7835

- NT 4.0
- uOne GateServer 4.1E
- Embedded Message Server
- Embedded Directory Server
- 2x18 GB drives

uOne 4.1E—Corporate

- Gateserver
- 500 users voice mail only
- No VM networking
- 20 simultaneous VM Ports
- 30 min. of VM storage per user



uOne 4.1E Deployment Considerations

- 500 mailboxes per deployment (no VM networking between systems)
- uOne Gateserver + CallManager must be colocated
- Support for 4, 5, 7 + 10 digit dial plans
- G.711 only + no CallManager failover
- No legacy VM interoperability (AMIS-A or VPIM)

uOne 5.0E—Corporate Edition

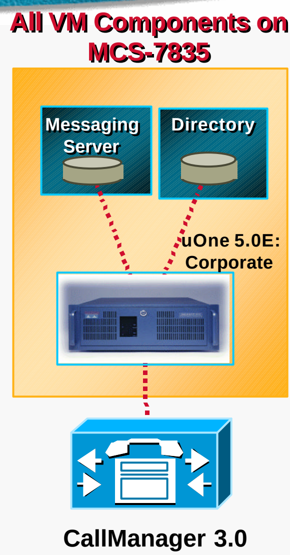
- Voice mail only
- Voice mail networking supported
- CallManager failover supported
- G.711 or G.729
- AMIS-A supported
- Included on the Cisco MCS-7835

  Windows NT 4.0

  500 user mailboxes

  20 simultaneous sessions

  uOne 5.0E

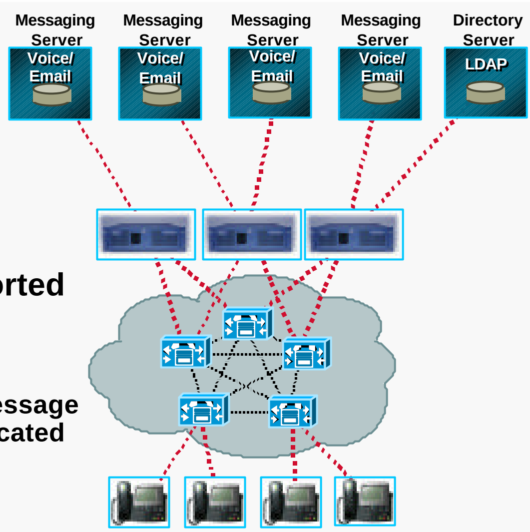


uOne 5.0E Deployment Considerations

- 10,000 user campus deployments
- VM networking supported

Note:

A User's Gateservers and Message Store Must Be Always Colocated

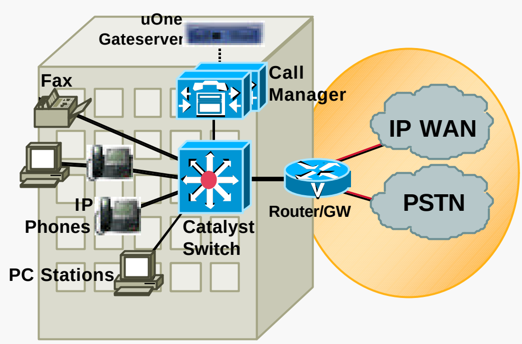


Agenda

- General Enterprise Deployment Models
- Campus Design Considerations
- Multisite WAN Considerations
- Legacy Migration Strategies

WAN Design Considerations

- QoS
- Admission control
- Dial plan

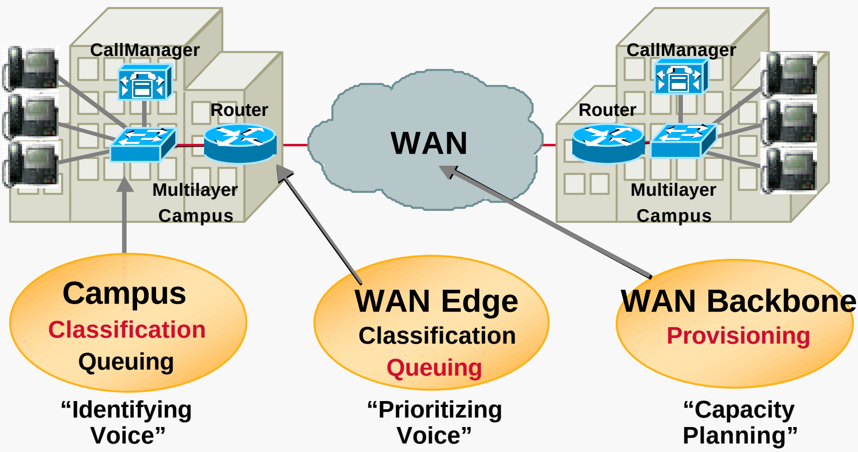


Wide Area Network Deployment Models

Distributed Call Processing
Centralized Call Processing

Domains of QoS Consideration

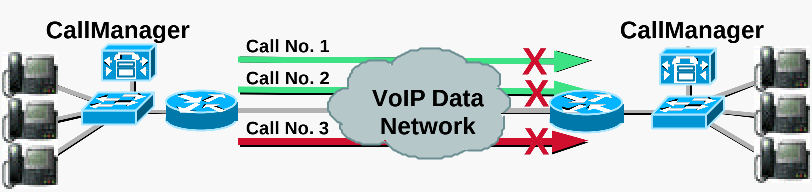
Avoiding Loss, Delay, and Delay Variation (Jitter)



Need for Admission Control "Protecting Voice from Voice"

Example:

WAN Bandwidth Can Only Support Two Calls
What Happens When Third Call Attempted?



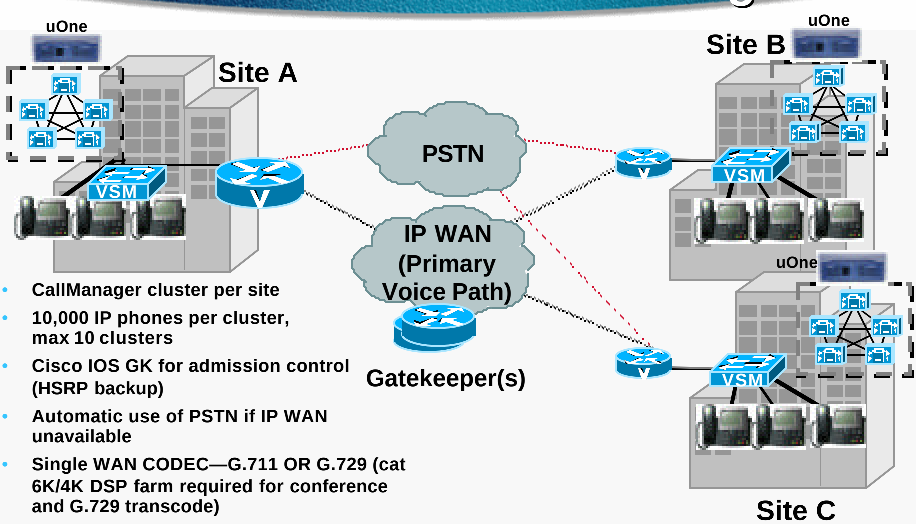
Call Number Three
Causes Poor Quality for ALL Calls

Need—to Prevent Third Call From Traversing IP WAN

Networked WAN Deployment Models

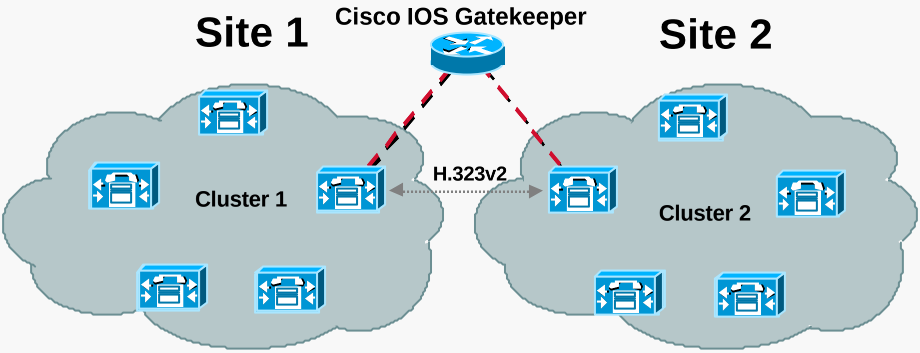
- **Distributed call processing**
  - **CallManager/cluster at each site**
- **Centralized call processing**
  - **CallManager/cluster only at central site**

WAN Deployment Models Distributed Call Processing



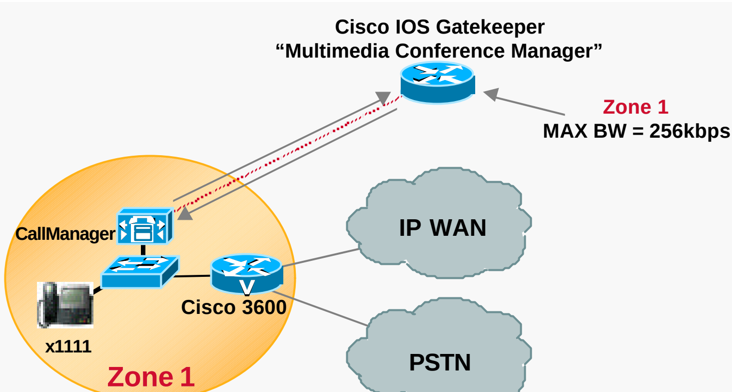
- CallManager cluster per site
- 10,000 IP phones per cluster, max 10 clusters
- Cisco IOS GK for admission control (HSRP backup)
- Automatic use of PSTN if IP WAN unavailable
- Single WAN CODEC—G.711 OR G.729 (cat 6K/4K DSP farm required for conference and G.729 transcode)
- If voice mail required then local uOne gateserver needed

Intersite WAN Communication



- H.323v2 between CallManagers (clusters)
- Standard h.323 gatekeeper for admission control
- 10 clusters networked across WAN (five if redundant)

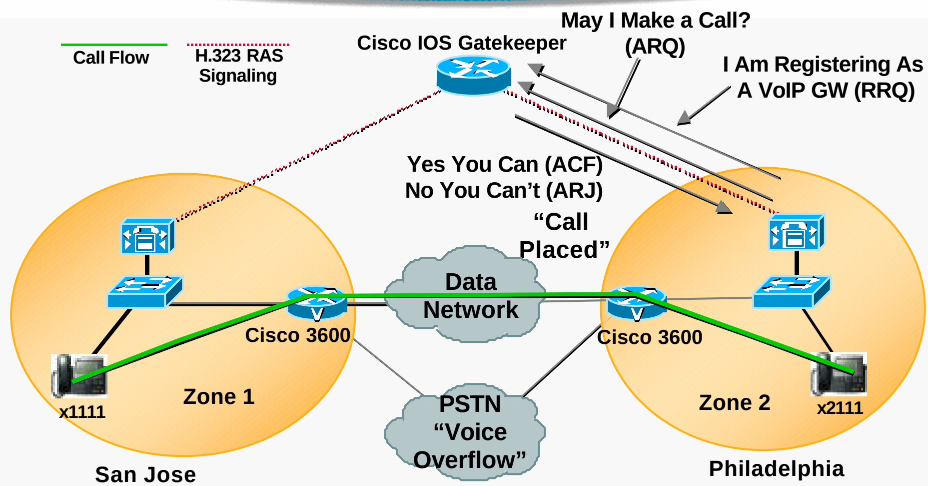
CallManager Registers as a VoIP GW



Gatekeeper associates each CallManager with a Zone

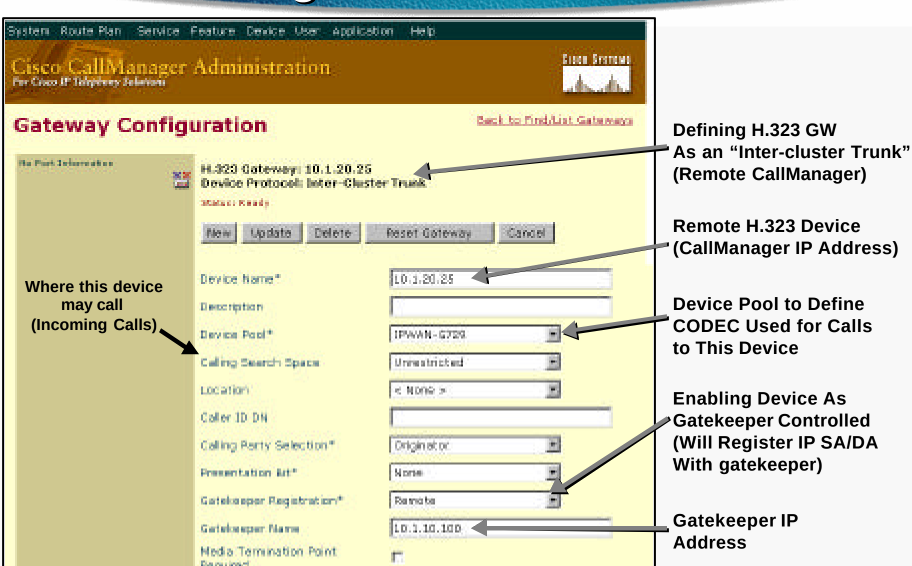
BW limits may be imposed on Zones in Cisco IOS Gatekeeper such that IP WAN voice BW in or out of a given zone will not exceed configured value

Basic CallManager Gatekeeper Interaction



Can Dynamically Send Calls Across PSTN If IP WAN Unavailable

Enabling Gatekeeper Use Before Sending Call Across IP WAN



Gatekeeper Configuration

"Cisco IOS Gatekeeper"

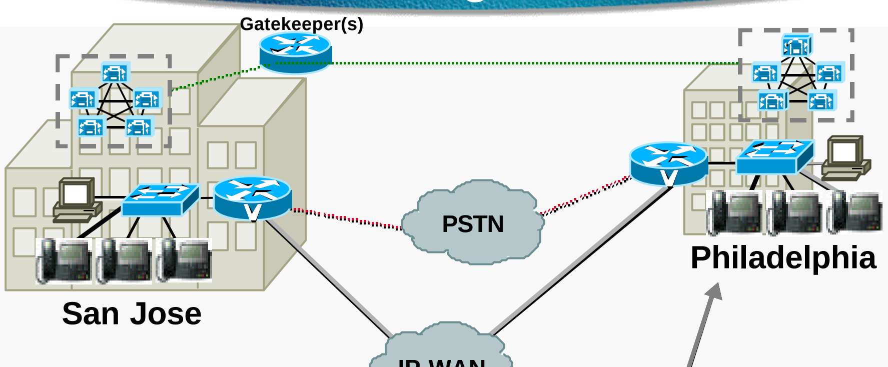
```
gatekeeper
zone local zone1 cisco.com
zone local zone2 cisco.com
zone subnet zone1 10.1.10.5/32 enable
no zone subnet zone1 0.0.0.0/0 enable
zone subnet zone2 10.1.20.25/32 enable
no zone subnet zone2 0.0.0.0/0 enable
zone bw zone1 128
zone bw zone2 128
no shutdown
```

Assigning Gatekeeper Zone Name

Assigning CallManager to Zone based on source subnet

Assigning Maximum Bandwidth in or out of a region

CODEC Selection Based on Regions

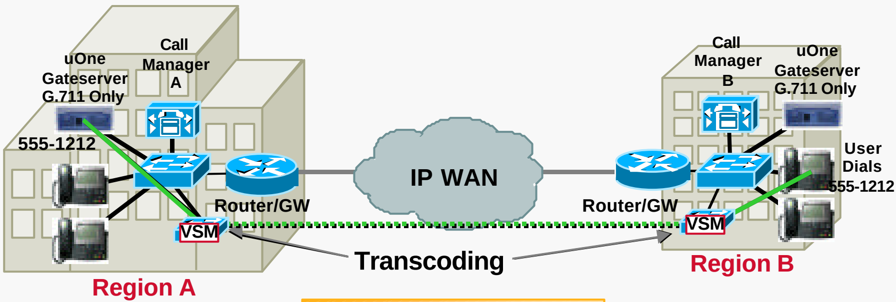


Devices	Region
IP Phones at PHL	"PHL"
SJ CallManager	"IPWAN-G729"

Calls within PHL = G.711
Calls from PHL to SJ = G.729

Region CODEC Matrix	CODEC
"PHL" to "PHL"	G.711
"PHL to "IPWAN-G729"	G.729

DSP Resources for Transcoding



Region BW Matrix

A to A = G.711
A to B = G.729

Compressed Call Leg
G.711 Call Leg
DSP = DSP Farm

Region BW Matrix

B to B = G.711
B to A = G.729

Networked WAN Deployment Models

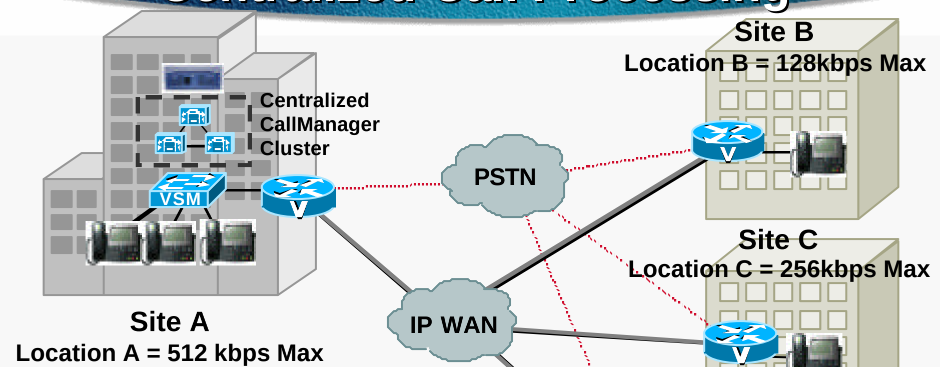
- **Distributed call processing**

  CallManager/cluster at each site

- **Centralized call processing**

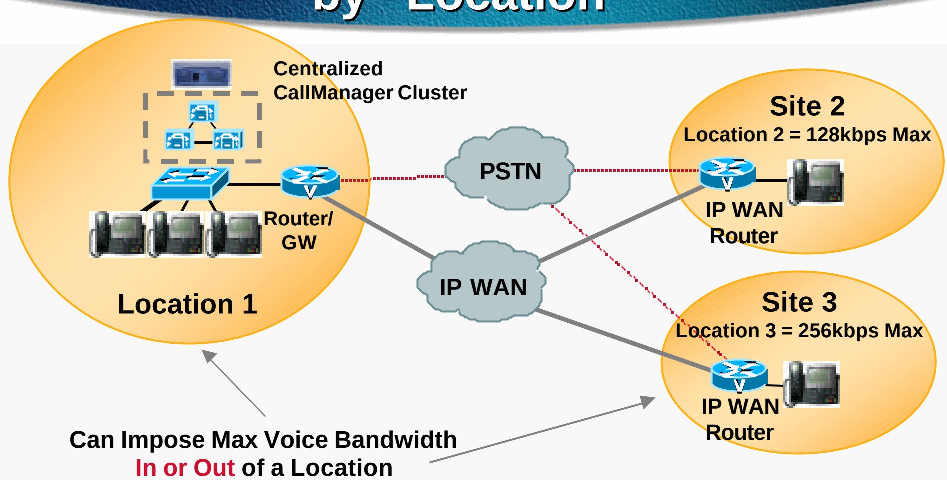
  CallManager only at central site

Multisite WAN Deployments “Centralized Call Processing”



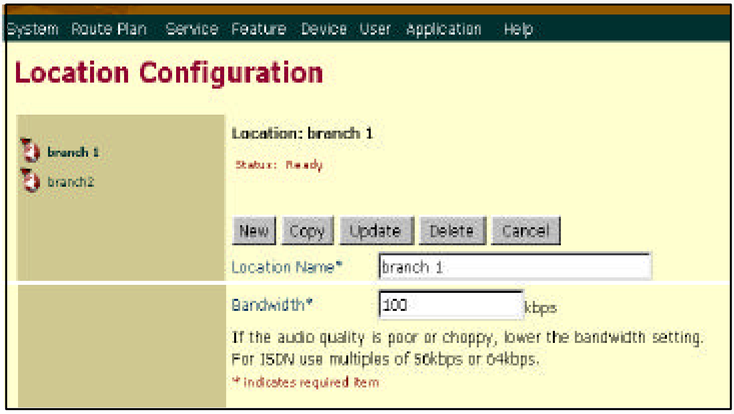
- CallManager, VM and DSP Resource at Central Site
- Supports up to 2500 users total
- Max of three CallManagers, All IP Phones registered to same CM
- One “active” CallManager
- Admission Control—limit number of calls per site
- No service if WAN down (unless dial backup)

Admission Control—BW Limitation by “Location”

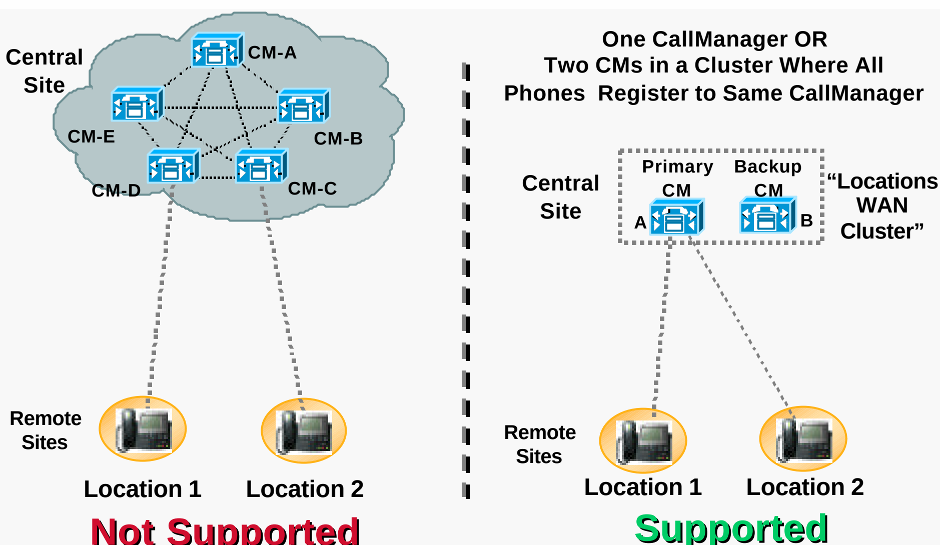


- Assign IP WAN bandwidth limits per Location (in kbps)
- Will get busy signal and indication like “All Trunks Busy” when insufficient resources
- No Automatic Route Selection—Must hangup and dial unique local PSTN access code

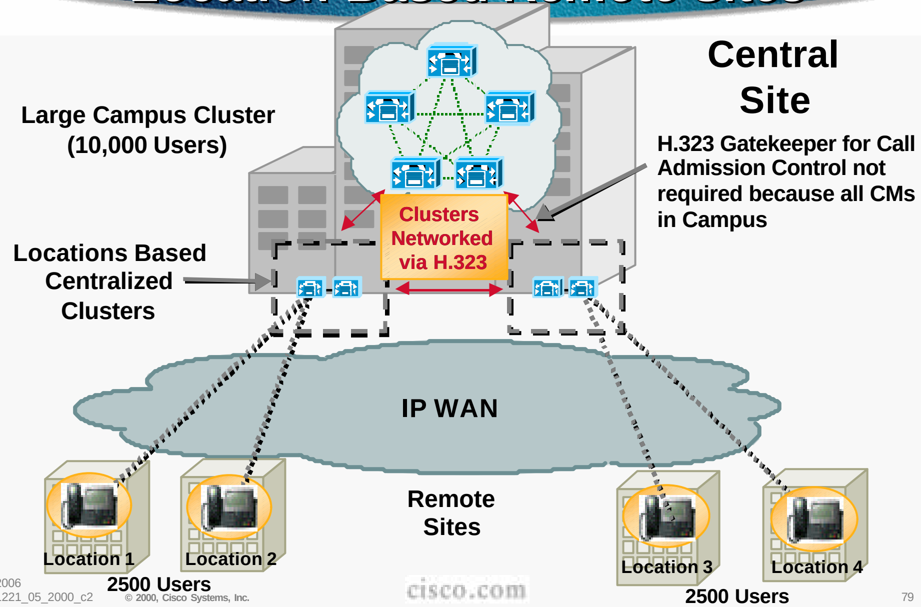
Configuring Locations for CM 3.0



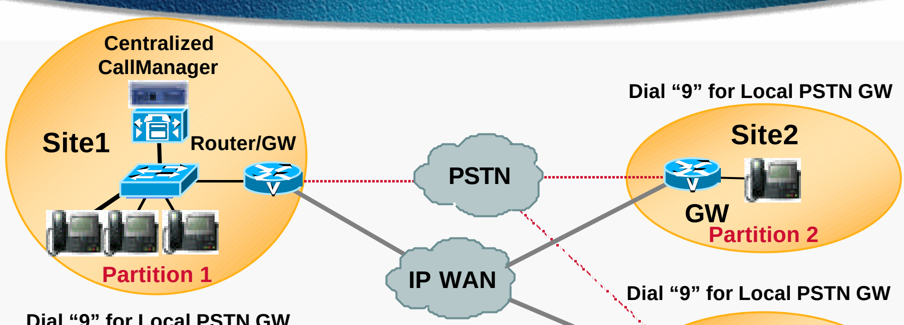
Location Based Admission Control and Cluster Interaction



Large Campus Interaction with Location-Based Remote Sites



Partitions/Multitenant Ability

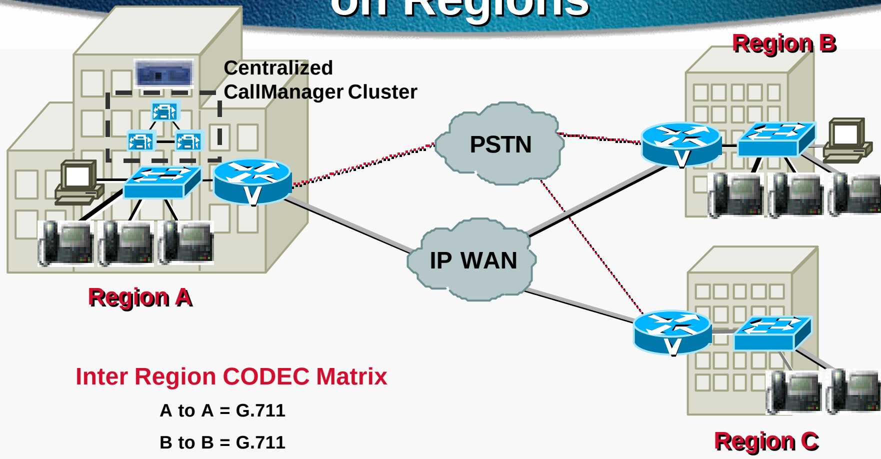


Partition = (Dial Plan Group)

1. Can provide dialing restrictions among users

2. Each Partition can be configured such that each site can dial “9” for their own local PSTN Gateway

CODEC Selection Based on Regions



Inter Region CODEC Matrix

A to A = G.711

B to B = G.711

C to C = G.711

A to C = G.729

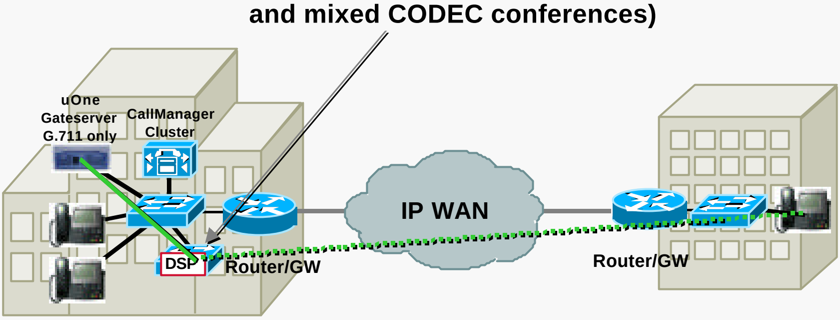
A to B = G.729

B to C = G.729

DSP Resource Use in Centralized Call Processing Model

Transcoding

(Typically for devices that only support G.711 and mixed CODEC conferences)



Compressed Call Leg

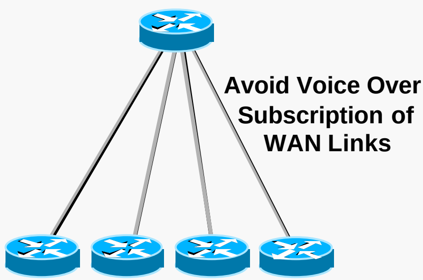
G.711 Call Leg

DSP = DSP Farm

Recommended Multisite WAN Topologies

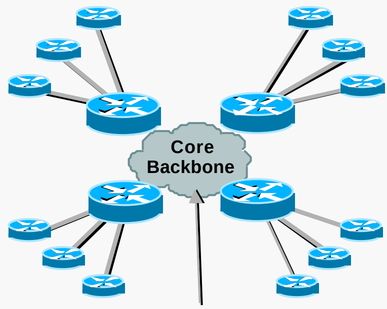
Topologies Based on Gatekeeper and Locations Based Admission Control

Hub and Spoke



Ensure Remote Voice Traffic Does not over subscribe a given link

Multilayer Hierarchical Design



Ensure Voice Does Not Oversubscribe Core

Agenda

- General Enterprise Deployment Models
- Campus Design Considerations
- Multisite WAN Considerations
- Legacy Migration Strategies

Common Migration Options

- Greenfields
- Flash-cut PBX/VM
- “Shrink and grow” migration

Legacy Voice Mail Integration Characteristics

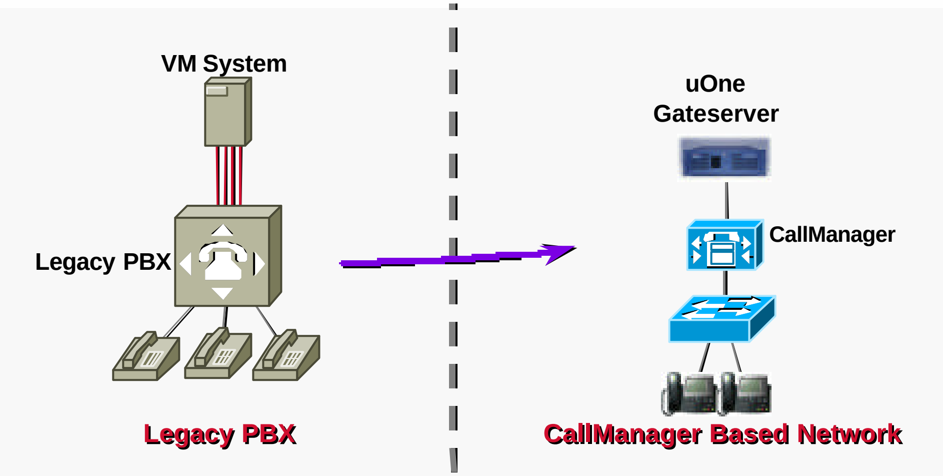
- Proprietary integration

  Proprietary software and hardware

  (Such as, Octel, Lucent, Nortel etc)
- SMDI only standard integration type

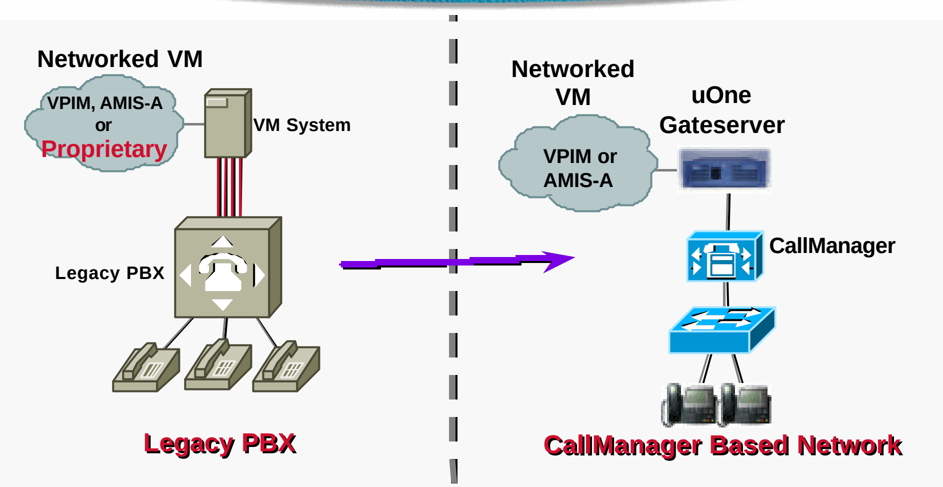
SMDI—Simple Message Desk Interface

Greenfields/Flash Cut to CallManager/uOne No VM Networking required



Relatively Issue Free

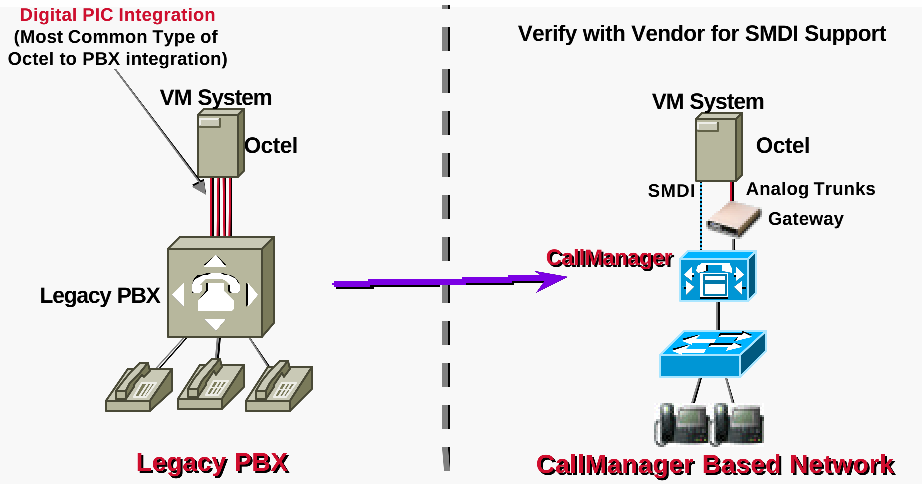
Greenfields/Flash Cut to CallManager/uOne VM Networking "Required"



Must use a Standard Voice Mail Networking Protocol

**VPIM on uOne Available Q4 '00

Flash Cut PBX—Keep Legacy VM Example—Cisco New York Office

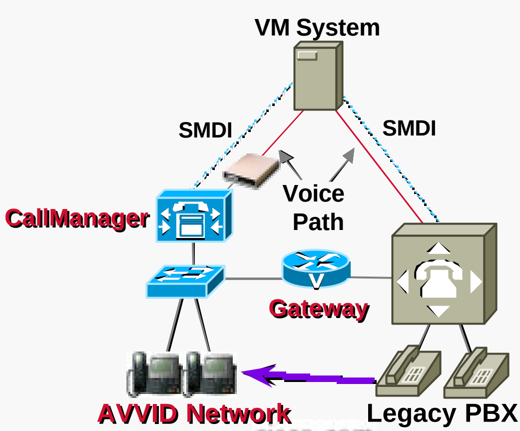


In some cases must convert VM to SMDI with Analog Trunks

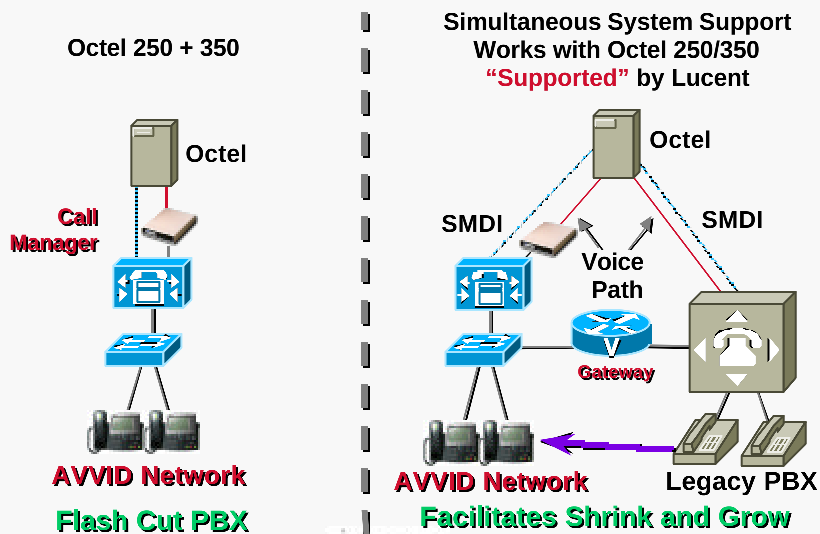
"Shrink and Grow" to CallManager Keep Legacy VM

Prerequisites:

VM System must be able to talk to two system Simultaneously
Verify for individual Vender support
May require PBX/VM conversion to SMDI

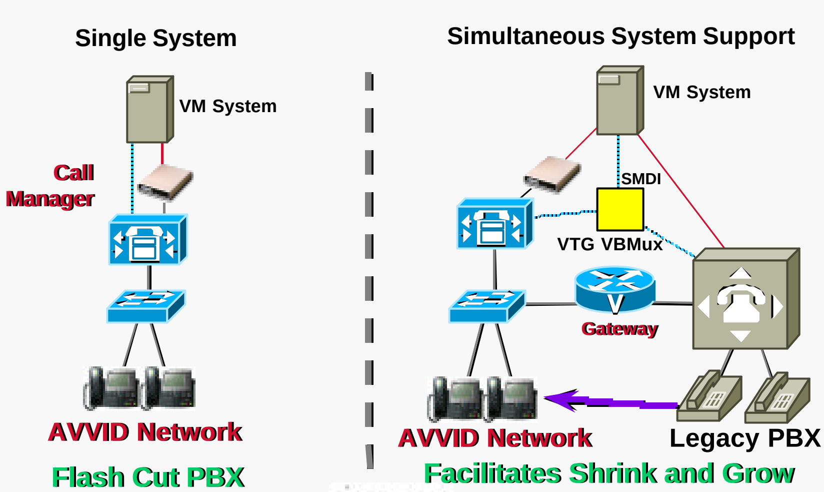


Legacy VM Migration Support Options—Dual SMDI Support

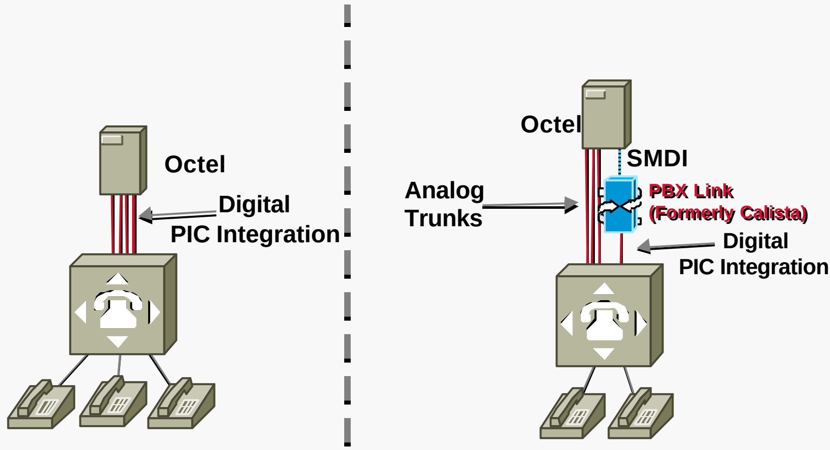


Legacy VM Migration Support Options—Single SMDI Interface

Lucent (Octel) 200/250/300/350
Lucent Intuity, Siemens Phones Mail



Converting PBX to Support SMDI



Digital Phone Interface Most Common From Octel VM Systems to Legacy PBXs

PBX Link Combined With Analog Trunks "Enables" SMDI on PBX

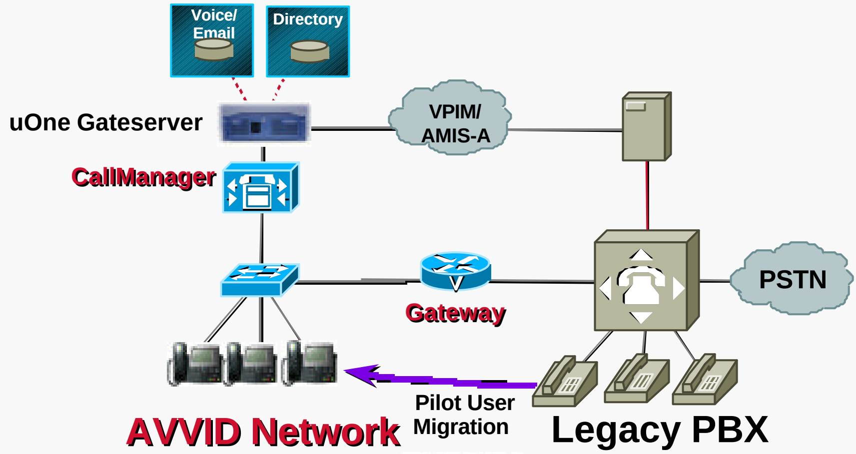
Support for Multiple SMDI Integrations

- Lucent (Octel) 250/350 "can" talk to two systems simultaneously as long as integration type is SMDI
- Lucent (Octel) 200/300 "can" talk to two systems simultaneously via VTG VB MUX
- Supported by Lucent
- Need to verify ability and support on vendor-by-vendor basis
- PBX conversion to SMDI may likely be required

“Shrink and Grow” to CallManager and uOne

Prerequisites:

uOne and Legacy VM need common VM Networking Protocol
AMIS-A (5.0E) or VPIM (Q4 00)



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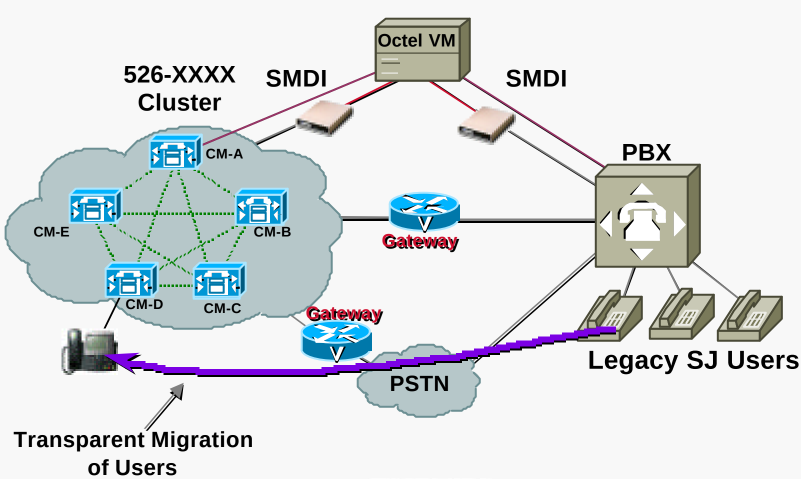
Example—Cisco San Jose Campus Migration Option

Allows for Transparent Migration from Lucent PBX to CallManager Cluster



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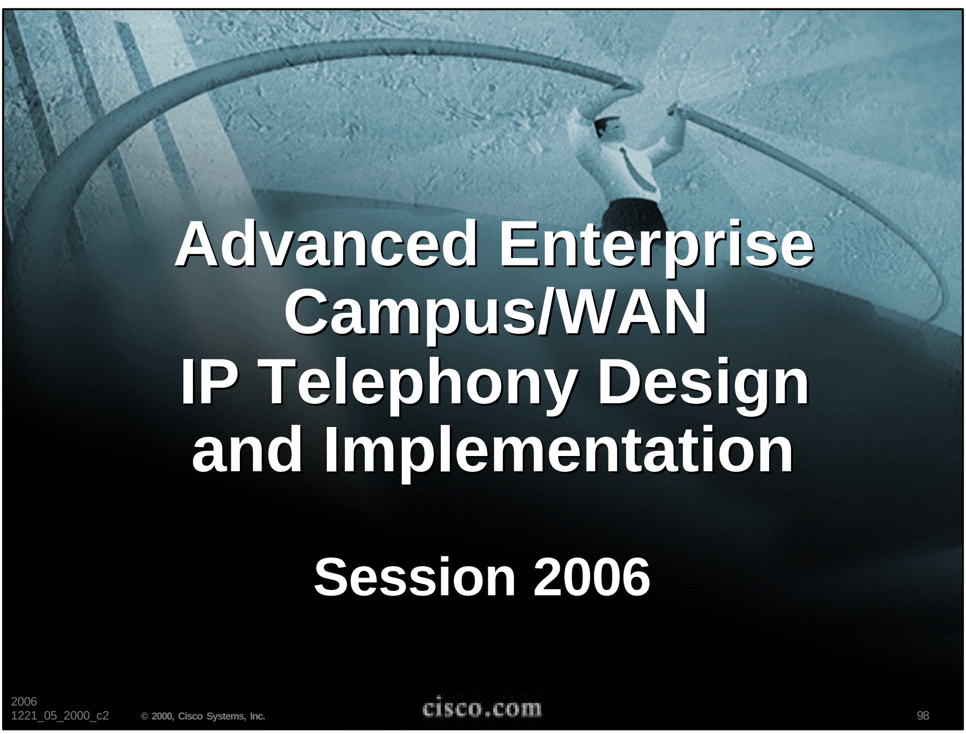
Summary—So What Did We Cover?

- Key ingredients/requirements of IP telephony networks
- How to build it
  - Campus/site deployments
  - Networked WAN deployments
  - Legacy migration options
- Know you environment and stay within design guidelines



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