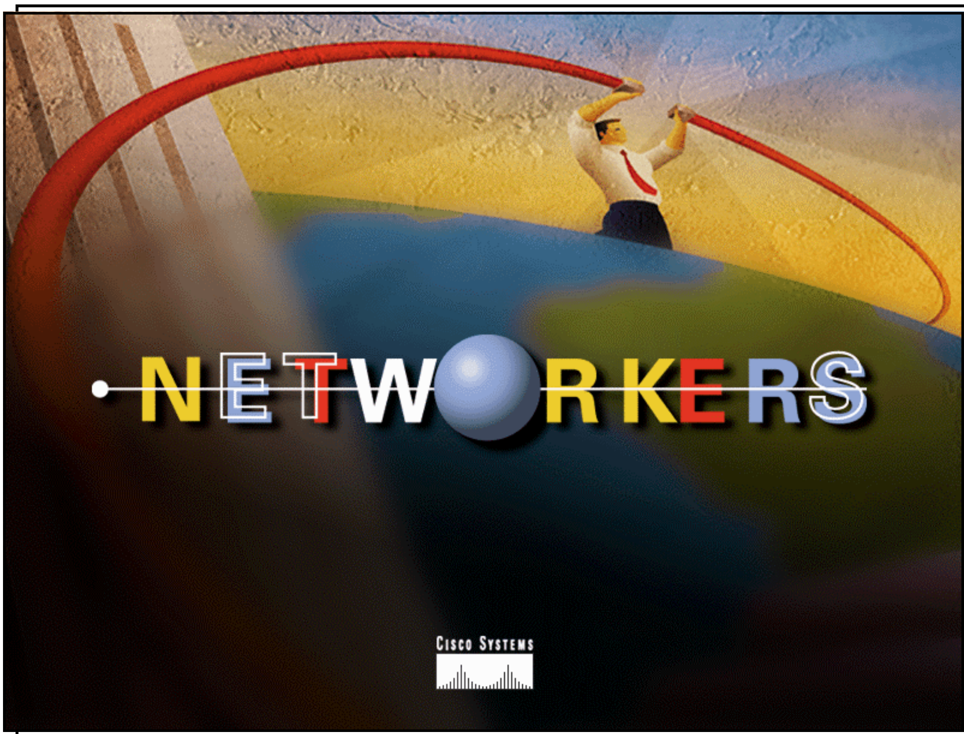
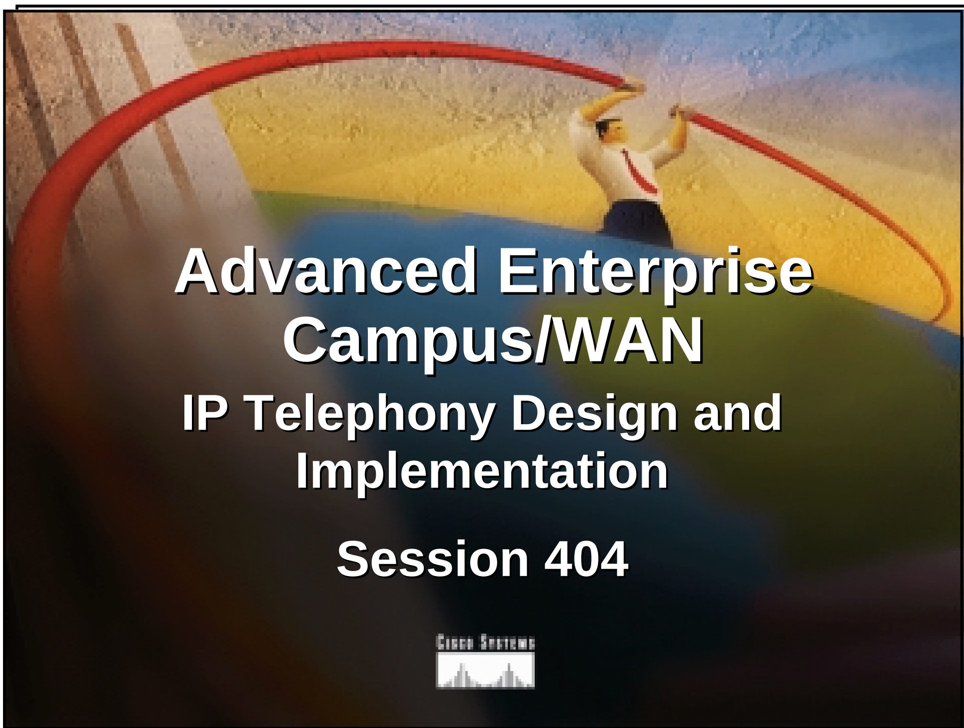
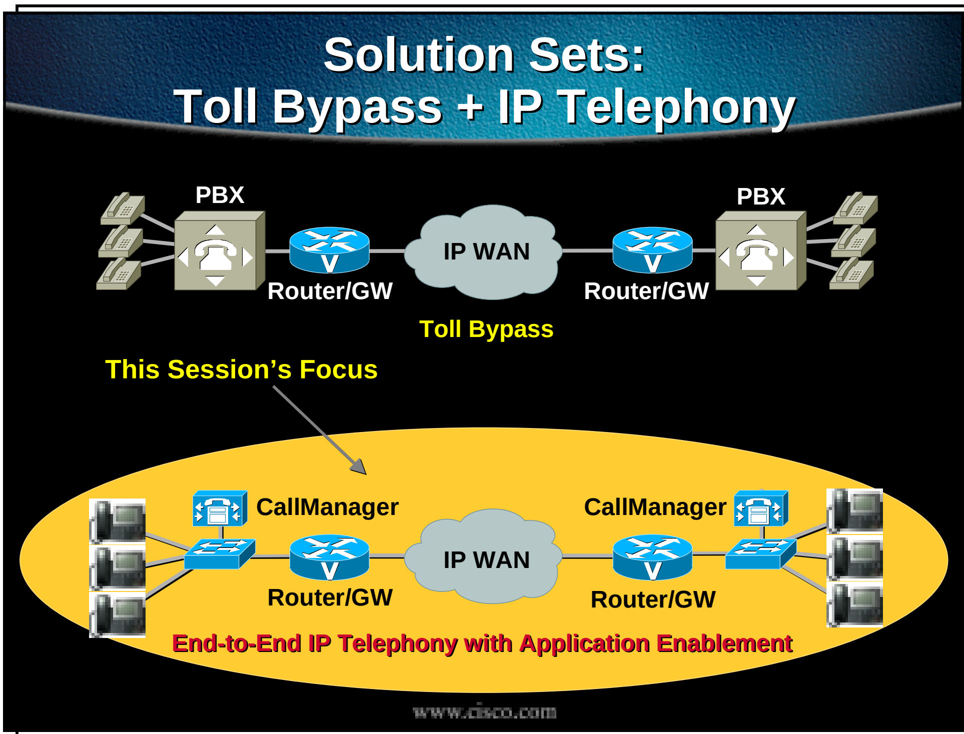
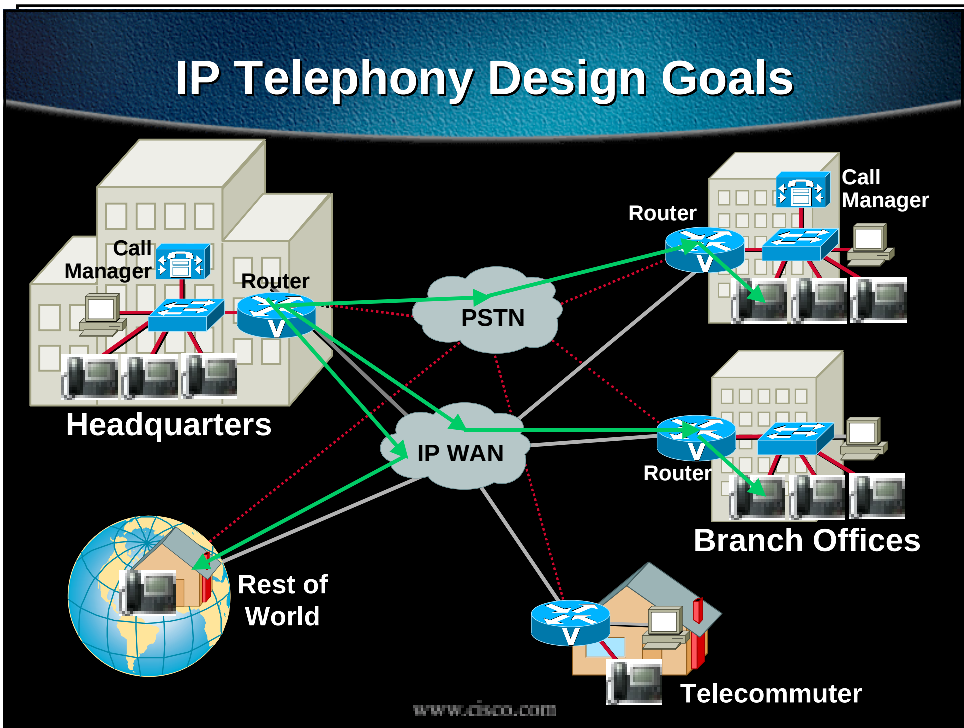

NETWORKERS
Cisco Systems

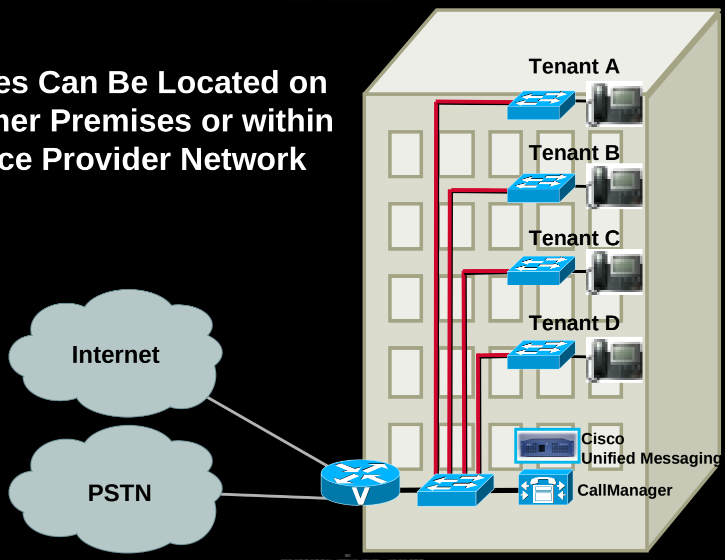

Advanced Enterprise Campus/WAN
IP Telephony Design and Implementation
Session 404
Cisco Systems


Solution Sets:
Toll Bypass + IP Telephony
PBX
Router/GW
IP WAN
Router/GW
PBX
Toll Bypass
This Session's Focus
CallManager
CallManager
IP WAN
Router/GW
Router/GW
End-to-End IP Telephony with Application Enablement
www.cisco.com


IP Telephony Design Goals
Call
Manager
Router
Headquarters
PSTN
Router
Call
Manager
IP WAN
Router
Branch Offices
Rest of
World
Telecommuter
www.cisco.com

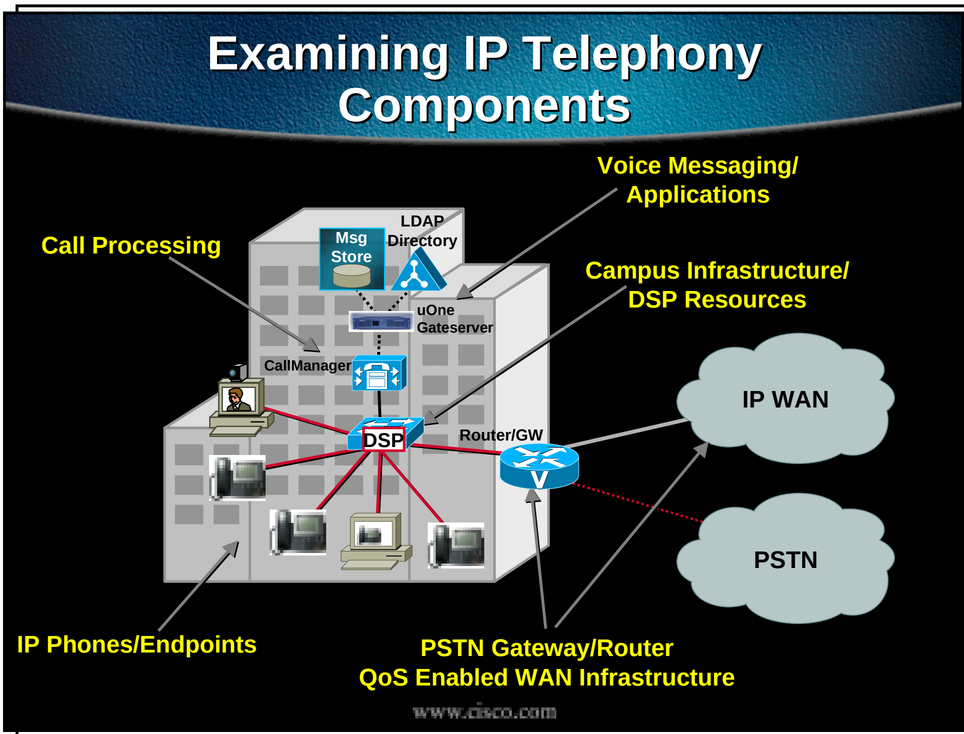
Managed Services/IP Centrex

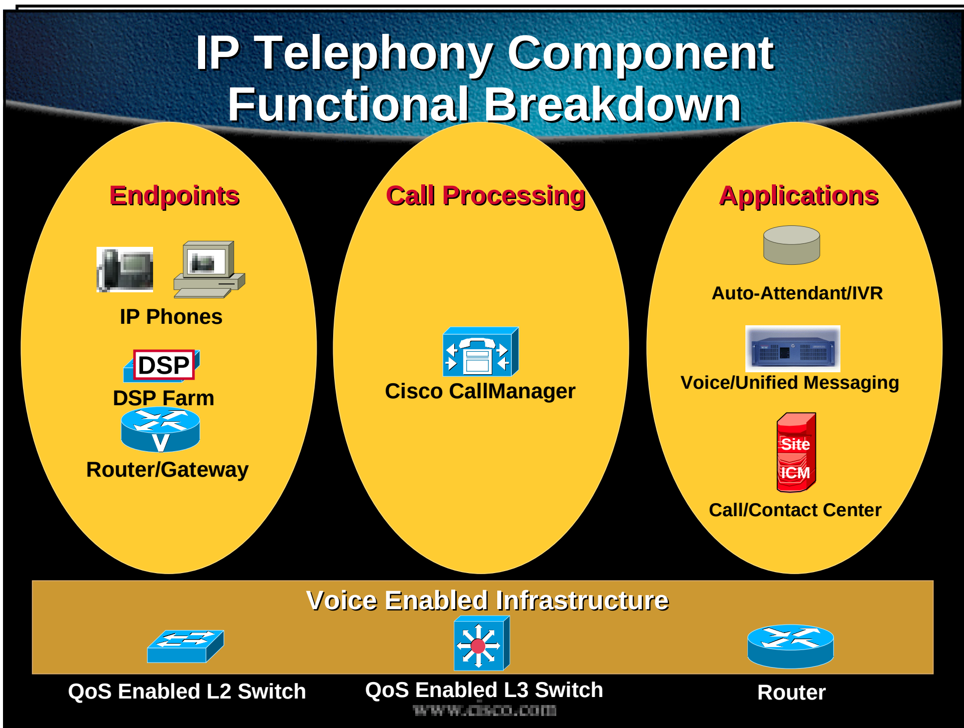
Services Can Be Located on Customer Premises or within Service Provider Network



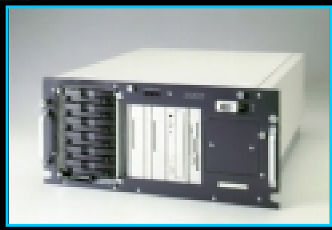
Agenda

- IP Telephony (AVVID) System Ingredients
- General Enterprise Deployment Models
- Campus Design Considerations
- Multisite WAN Considerations
- Voice Messaging Design Considerations


Examining IP Telephony Components
Voice Messaging/ Applications
Call Processing
Msg Store
LDAP Directory
Campus Infrastructure/ DSP Resources
uOne Gateserver
CallManager
IP WAN
DSP
Router/GW
PSTN
IP Phones/Endpoints
PSTN Gateway/Router
QoS Enabled WAN Infrastructure
www.cisco.com


IP Telephony Component Functional Breakdown
Endpoints
Call Processing
Applications
IP Phones
Auto-Attendant/IVR
DSP
DSP Farm
Cisco CallManager
Voice/Unified Messaging
Router/Gateway
Site
ICM
Call/Contact Center
Voice Enabled Infrastructure
QoS Enabled L2 Switch
QoS Enabled L3 Switch
Router
www.cisco.com

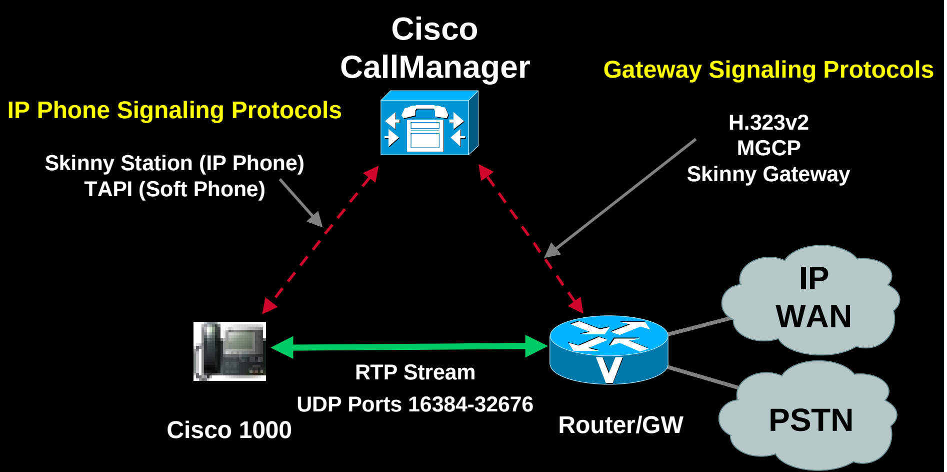
Call Processing
CallManager Primary Functions

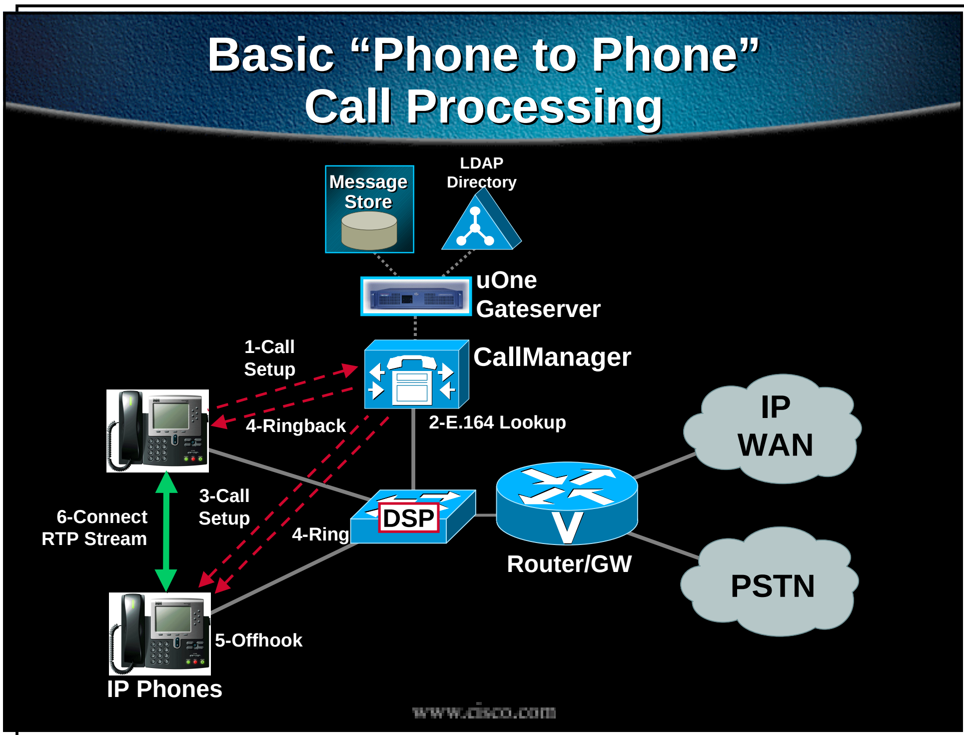


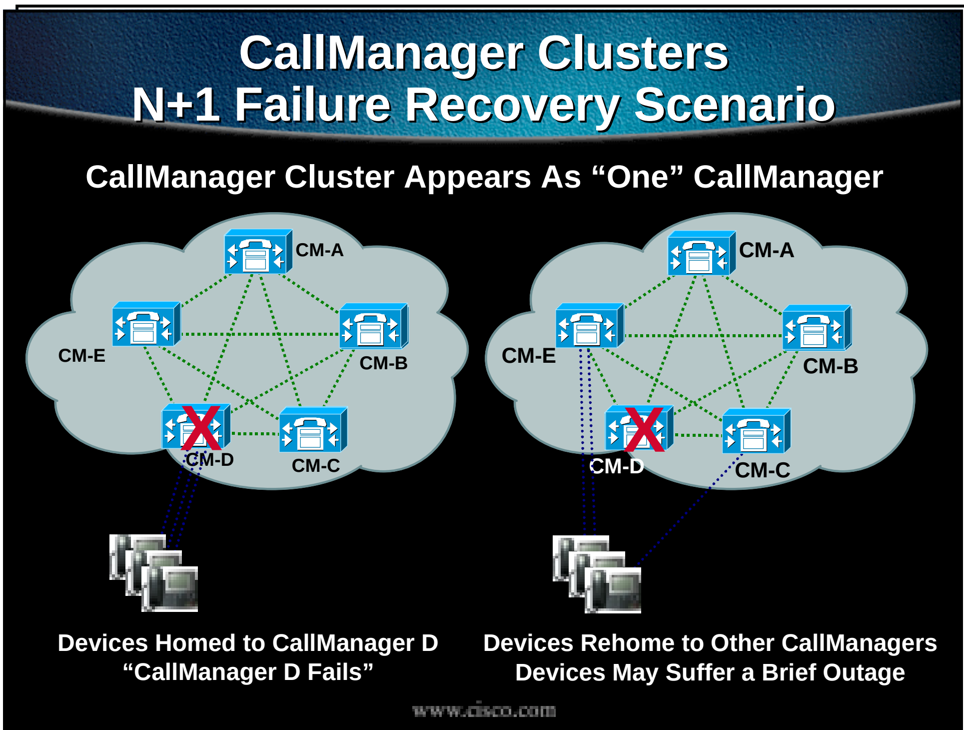
MCS-7830

- Call processing
- Signaling + device control
- Features, capabilities and dial plan
- Operations, administration, maintenance and provisioning (OAM&P)
- Programming interface to external voice processing applications

Cisco CallManager
Signaling/Call Control




Basic "Phone to Phone" Call Processing
Message Store
LDAP Directory
uOne Gateserver
CallManager
1-Call Setup
4-Ringback
2-E.164 Lookup
IP WAN
3-Call Setup
6-Connect RTP Stream
4-Ring
DSP
Router/GW
PSTN
5-Offhook
IP Phones
www.cisco.com


CallManager Clusters N+1 Failure Recovery Scenario
CallManager Cluster Appears As "One" CallManager
CM-A
CM-E
CM-B
CM-D
CM-C
CM-A
CM-E
CM-B
CM-D
CM-C
Devices Homed to CallManager D "CallManager D Fails"
Devices Rehome to Other CallManagers Devices May Suffer a Brief Outage
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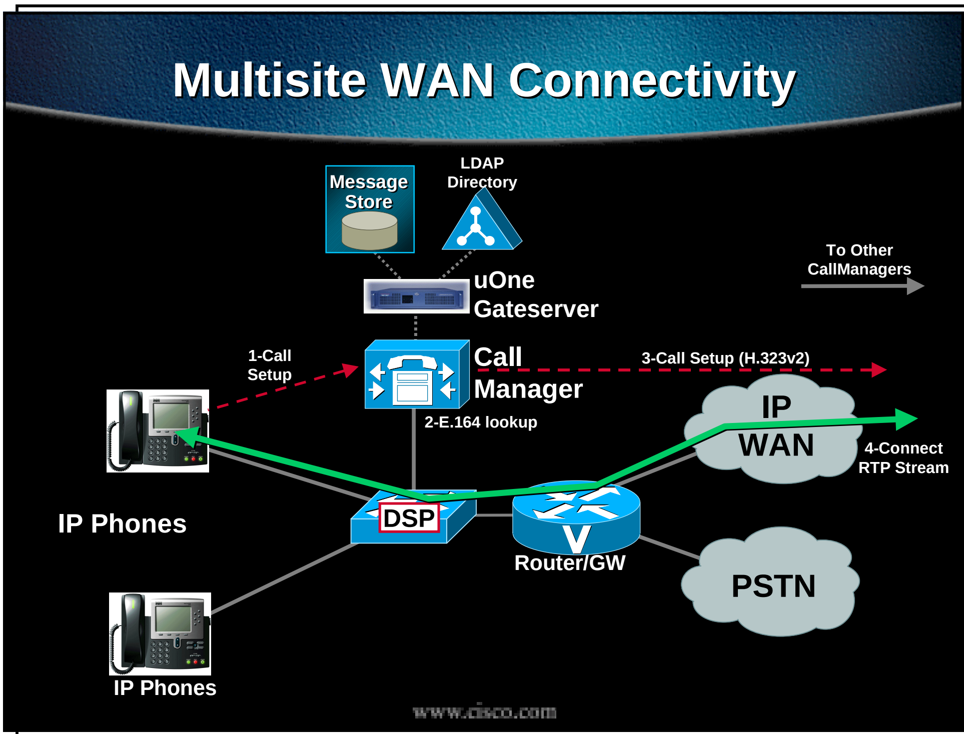
Multisite WAN Connectivity
Message Store
LDAP Directory
uOne Gateserver
To Other CallManagers
1-Call Setup
Call Manager
3-Call Setup (H.323v2)
2-E.164 lookup
IP WAN
4-Connect RTP Stream
IP Phones
DSP
Router/GW
PSTN
IP Phones
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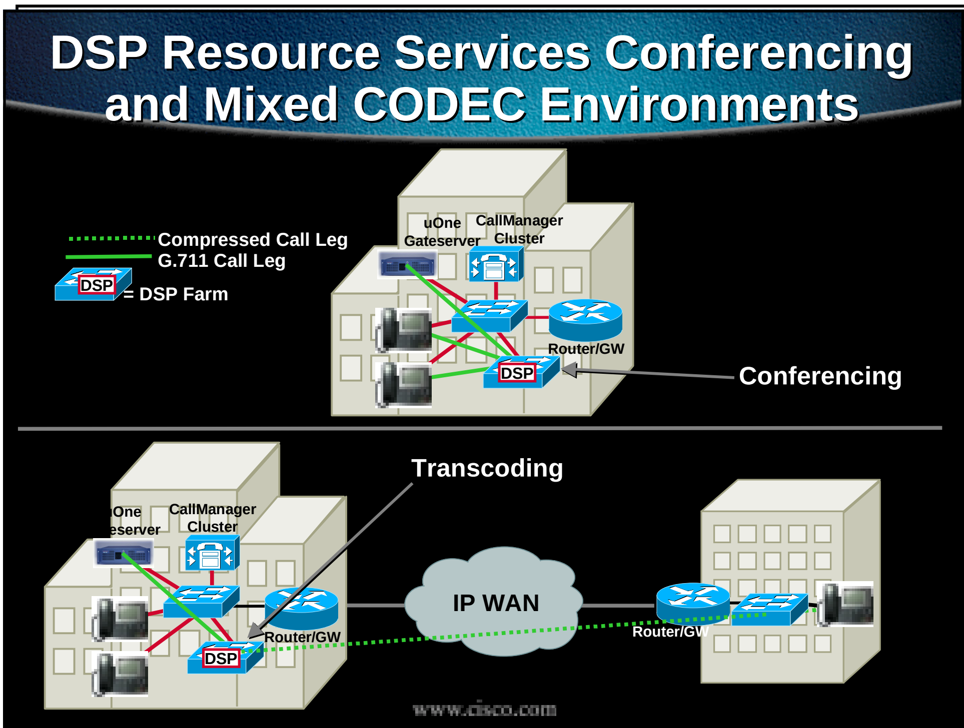
DSP Resource Services Conferencing and Mixed CODEC Environments
Compressed Call Leg
G.711 Call Leg
DSP = DSP Farm
uOne Gateserver
CallManager Cluster
Router/GW
DSP
Conferencing
Transcoding
CallManager Cluster
Router/GW
DSP
IP WAN
Router/GW
www.cisco.com

IP WAN Requirements
Call Admission Control + Dial Plan

- Determining if IP WAN can accept call
  - Call admission control
- What to do if admission control says no
  - Flexible dial plan structure

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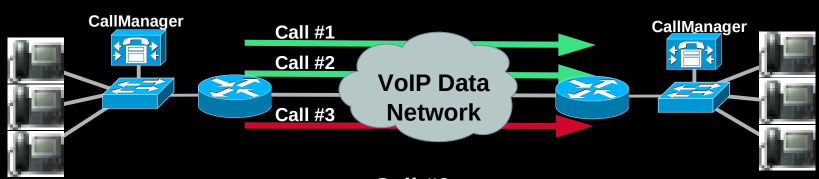
Admission Control
Why Do I Care?

Example:
WAN Bandwidth Can Only Support Two Calls
What Happens when 3rd Call Attempted?



Call #3
Causes Poor Quality for ALL Calls

Many Tools to Give Voice Priority Over Data
Call Admission Control Is About Preventing Voice Over Subscription

www.cisco.com

CallManager Admission Control Mechanisms

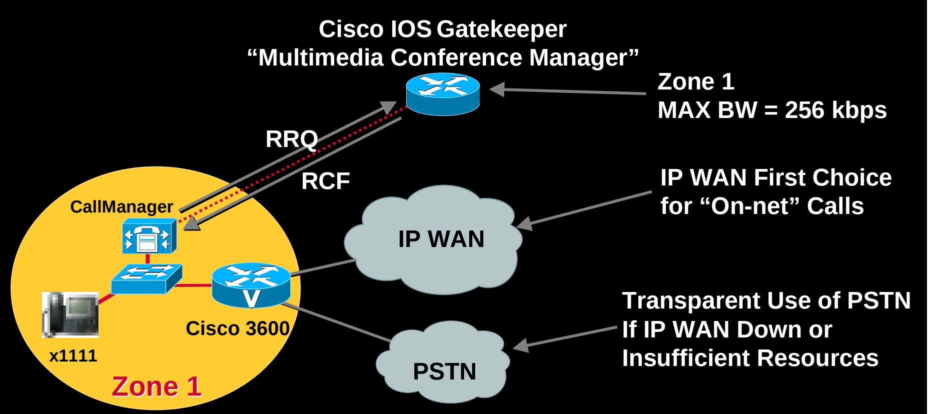
- **H.323 gatekeeper based**

  Can perform dynamic alternate routing

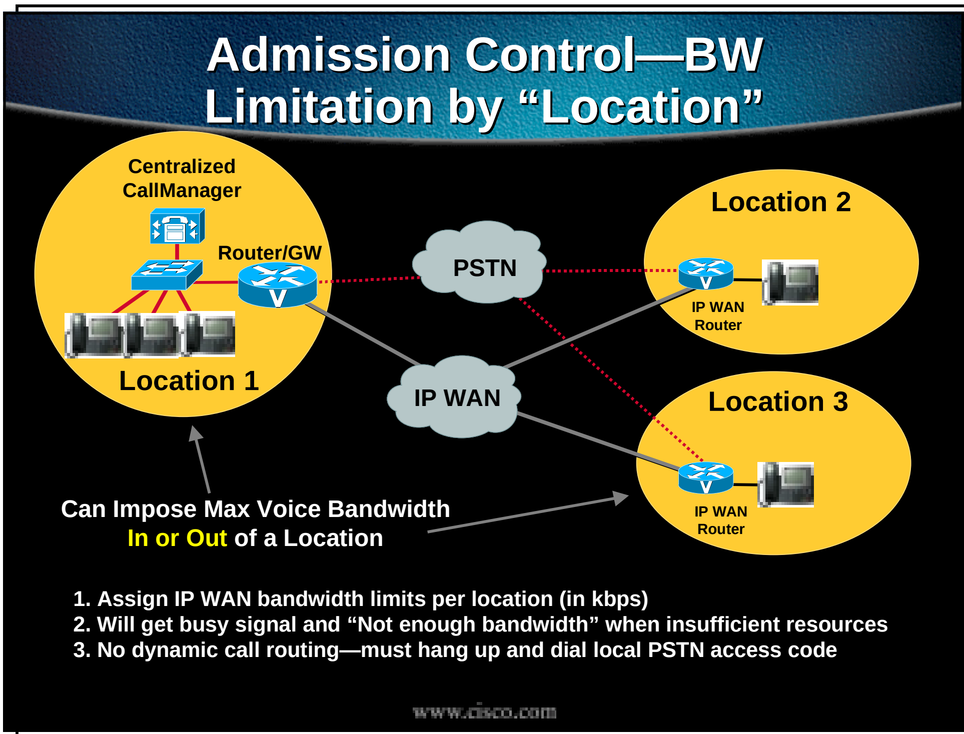
- **Bandwidth limits based on “location”**

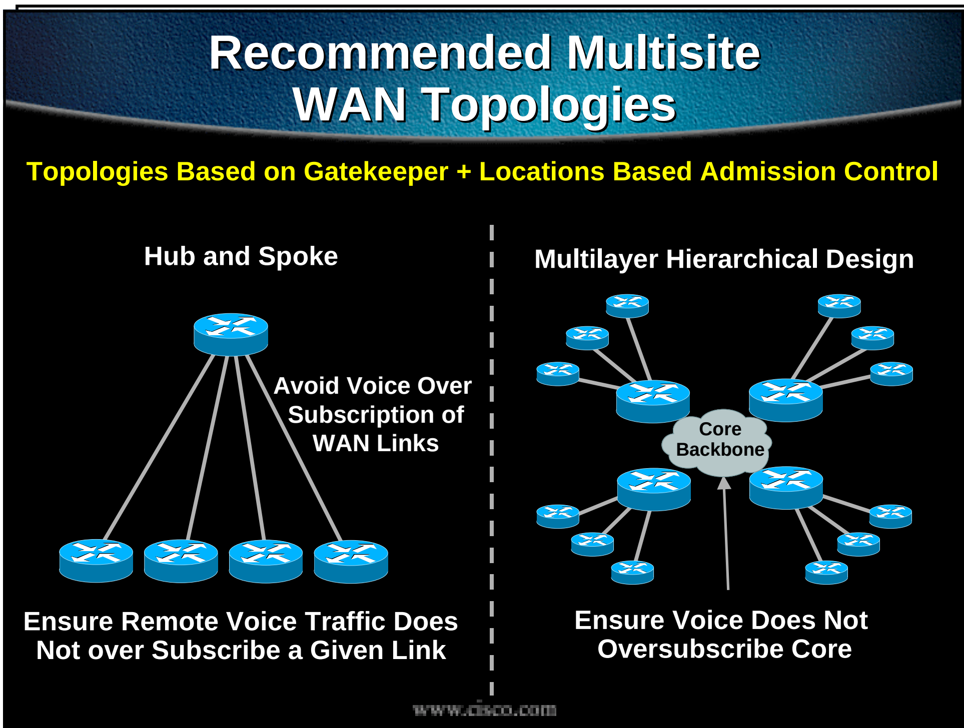
  Cannot perform dynamic alternate routing

Admission Control—H.323 RAS CallManager Registers with Cisco IOS® Gatekeeper as a VoIP Gateway



**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**


Admission Control—BW Limitation by "Location"
Centralized CallManager
Router/GW
PSTN
Location 2
IP WAN Router
Location 1
IP WAN
Location 3
IP WAN Router
Can Impose Max Voice Bandwidth
In or Out of a Location
1. Assign IP WAN bandwidth limits per location (in kbps)
2. Will get busy signal and "Not enough bandwidth" when insufficient resources
3. No dynamic call routing—must hang up and dial local PSTN access code
www.cisco.com


Recommended Multisite WAN Topologies
Topologies Based on Gatekeeper + Locations Based Admission Control
Hub and Spoke
Multilayer Hierarchical Design
Avoid Voice Over Subscription of WAN Links
Core Backbone
Ensure Remote Voice Traffic Does Not over Subscribe a Given Link
Ensure Voice Does Not Oversubscribe Core
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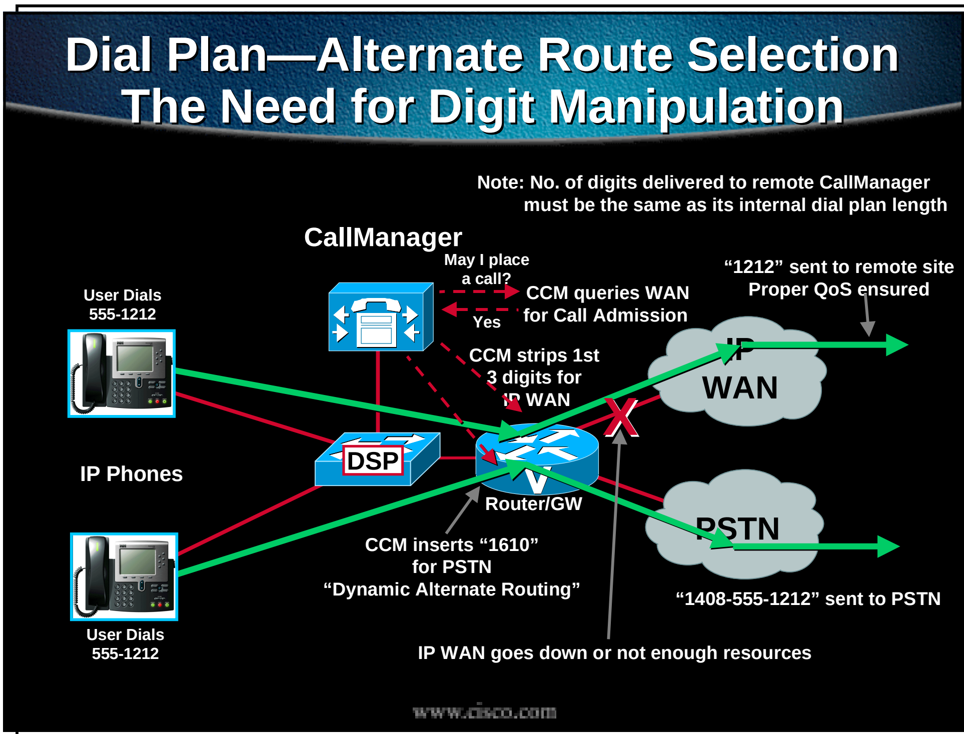
Dial Plan—Alternate Route Selection
The Need for Digit Manipulation
Note: No. of digits delivered to remote CallManager must be the same as its internal dial plan length
CallManager
May I place a call?
CCM queries WAN for Call Admission
Yes
"1212" sent to remote site
Proper QoS ensured
User Dials 555-1212
CCM strips 1st 3 digits for IP WAN
IP WAN
DSP
IP Phones
Router/GW
PSTN
CCM inserts "1610" for PSTN
"Dynamic Alternate Routing"
"1408-555-1212" sent to PSTN
User Dials 555-1212
IP WAN goes down or not enough resources
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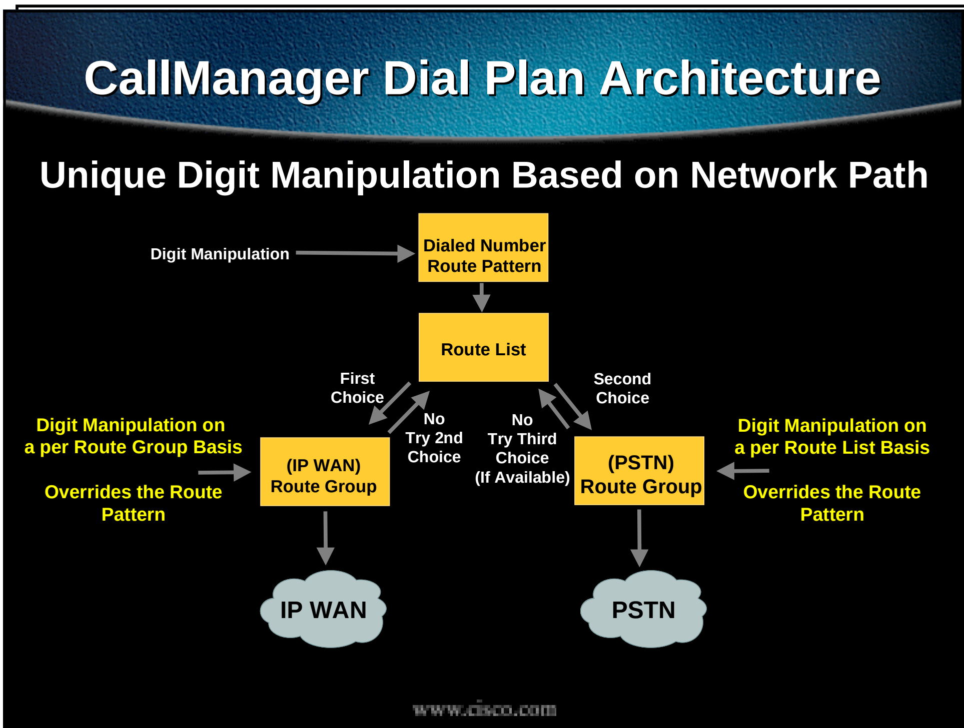
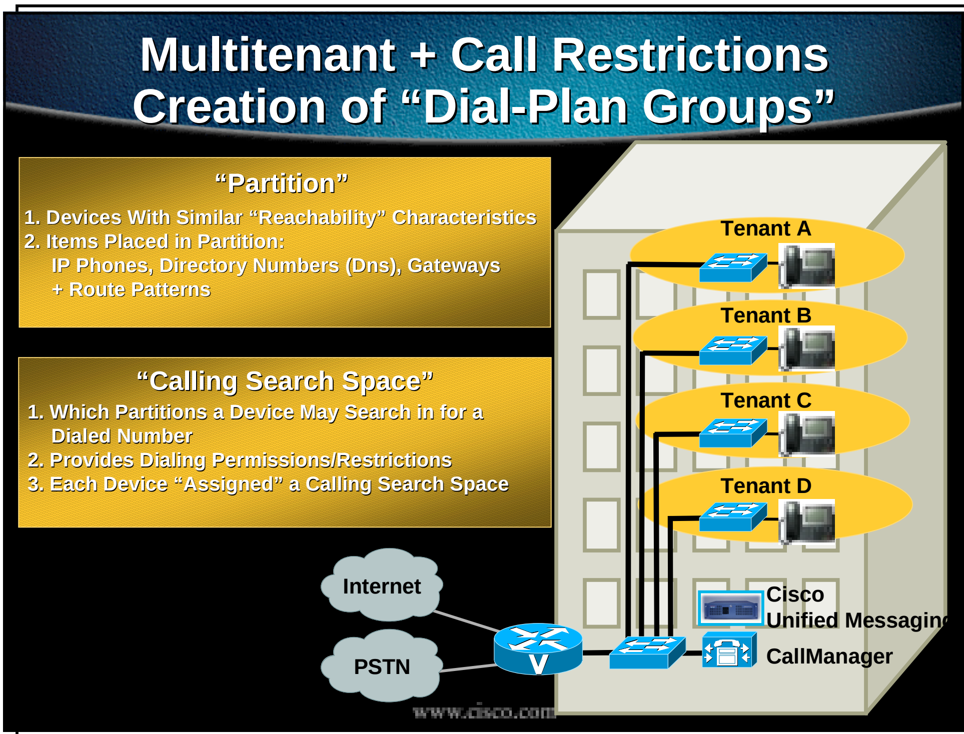
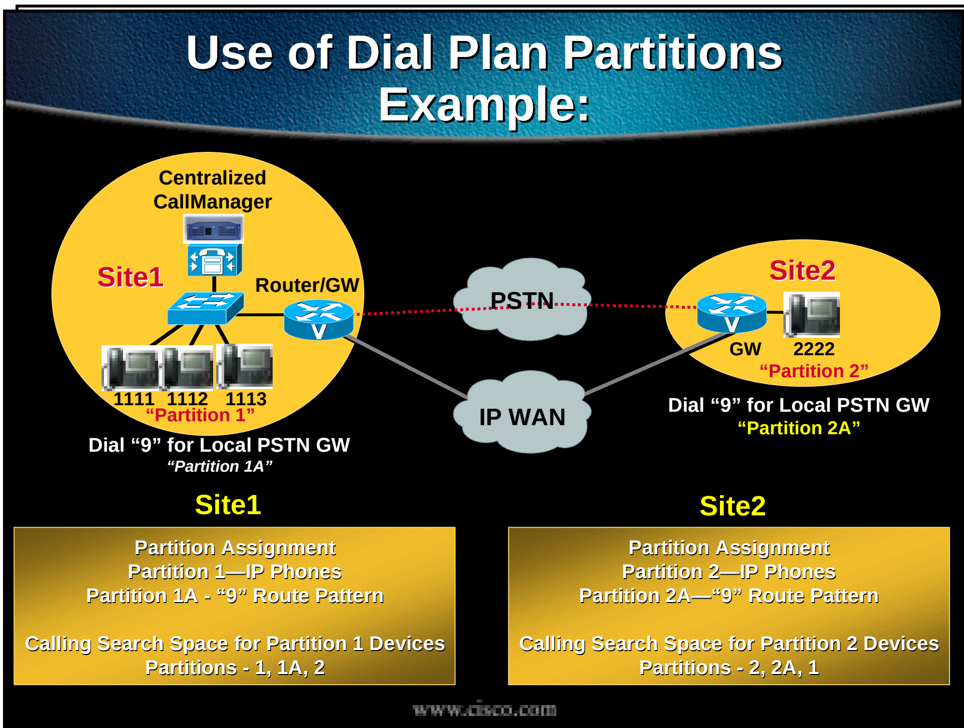

CallManager Dial Plan Architecture
Unique Digit Manipulation Based on Network Path
Digit Manipulation
Dialed Number Route Pattern
Route List
First Choice
Second Choice
No Try 2nd Choice
No Try Third Choice (If Available)
Digit Manipulation on a per Route Group Basis
Overrides the Route Pattern
(IP WAN) Route Group
(PSTN) Route Group
Digit Manipulation on a per Route List Basis
Overrides the Route Pattern
IP WAN
PSTN
www.cisco.com


Multitenant + 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. Which Partitions a Device May Search in for a Dialed Number
2. Provides Dialing Permissions/Restrictions
3. Each Device “Assigned” a Calling Search Space
Tenant A
Tenant B
Tenant C
Tenant D
Internet
PSTN
Cisco Unified Messaging
CallManager
www.cisco.com


Use of Dial Plan Partitions
Example:
Centralized CallManager
Site1
Router/GW
1111 1112 1113
“Partition 1”
Dial “9” for Local PSTN GW
“Partition 1A”
PSTN
IP WAN
Site2
GW 2222
“Partition 2”
Dial “9” for Local PSTN GW
“Partition 2A”
Site1
Partition Assignment
Partition 1—IP Phones
Partition 1A - “9” Route Pattern
Calling Search Space for Partition 1 Devices
Partitions - 1, 1A, 2
Site2
Partition Assignment
Partition 2—IP Phones
Partition 2A—“9” Route Pattern
Calling Search Space for Partition 2 Devices
Partitions - 2, 2A, 1
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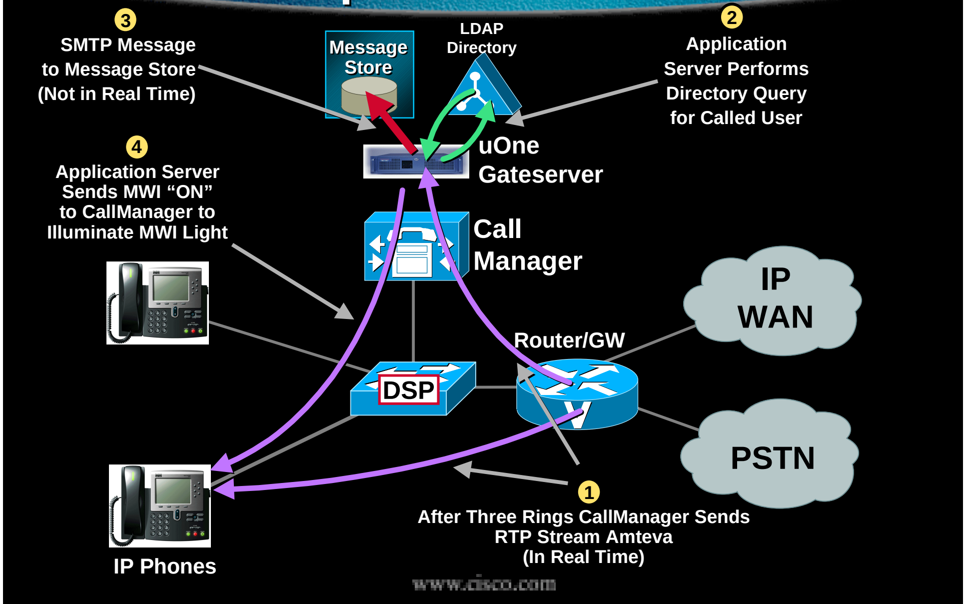
Voice/Unified Messaging Call Flow IP Phone Implemented in Gateserver



Agenda

- IP Telephony (AVVID) System Ingredients
- General Enterprise Deployment Models
- Campus Design Considerations
- Multisite WAN Considerations
- Voice Messaging Design Considerations

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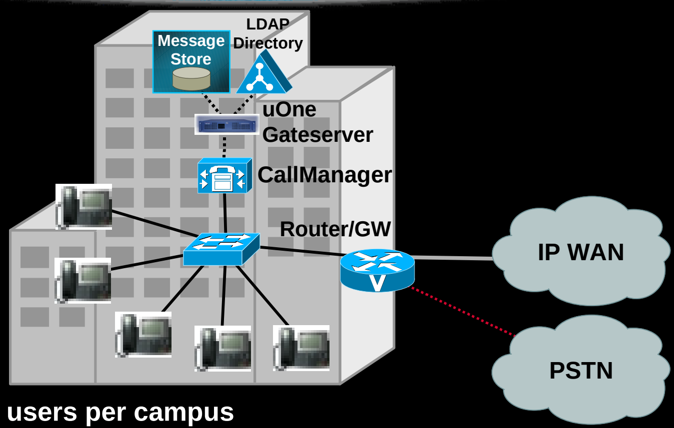
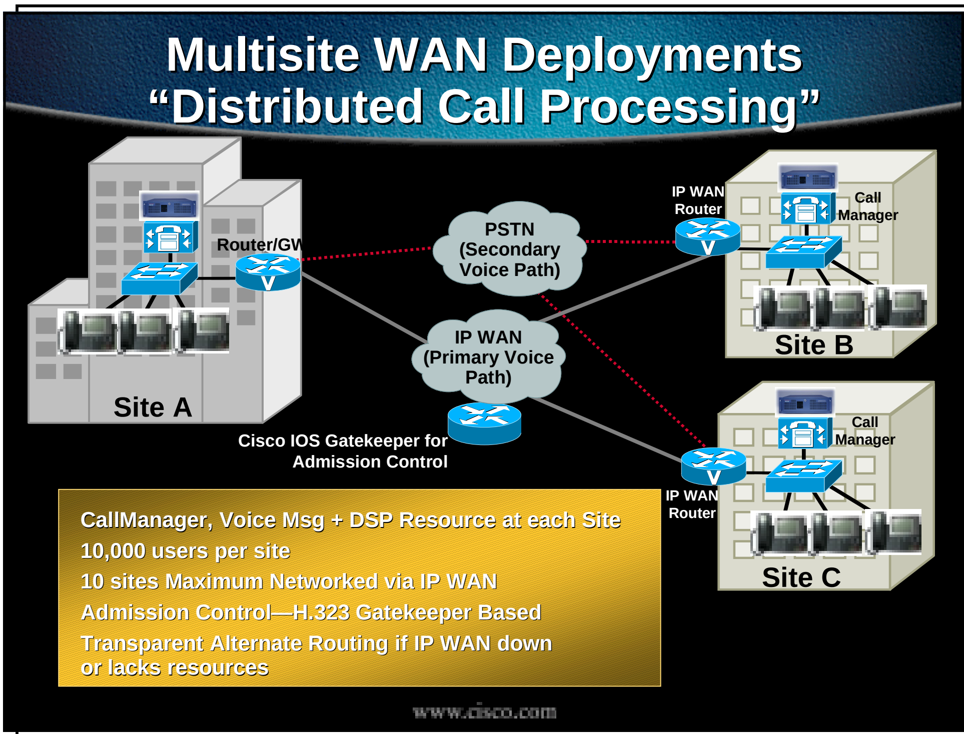
Basic Deployment Models

- Individual campus deployments
- Multisite WAN (distributed call processing)
- Multisite WAN (centralized call processing)

Individual Campus Deployments



- Up to 10,000 users per campus
- CallManager + voice messaging at each site
- Up to 5 distributed CallManagers in a cluster
- Redundancy + equipment will vary with campus size


Multisite WAN Deployments
“Distributed Call Processing”
Router/GW
Site A
PSTN
(Secondary
Voice Path)
IP WAN
(Primary Voice
Path)
Cisco IOS Gatekeeper for
Admission Control
IP WAN
Router
Call
Manager
Site B
Call
Manager
IP WAN
Router
Site C
CallManager, Voice Msg + DSP Resource at each Site
10,000 users per site
10 sites Maximum Networked via IP WAN
Admission Control—H.323 Gatekeeper Based
Transparent Alternate Routing if IP WAN down
or lacks resources
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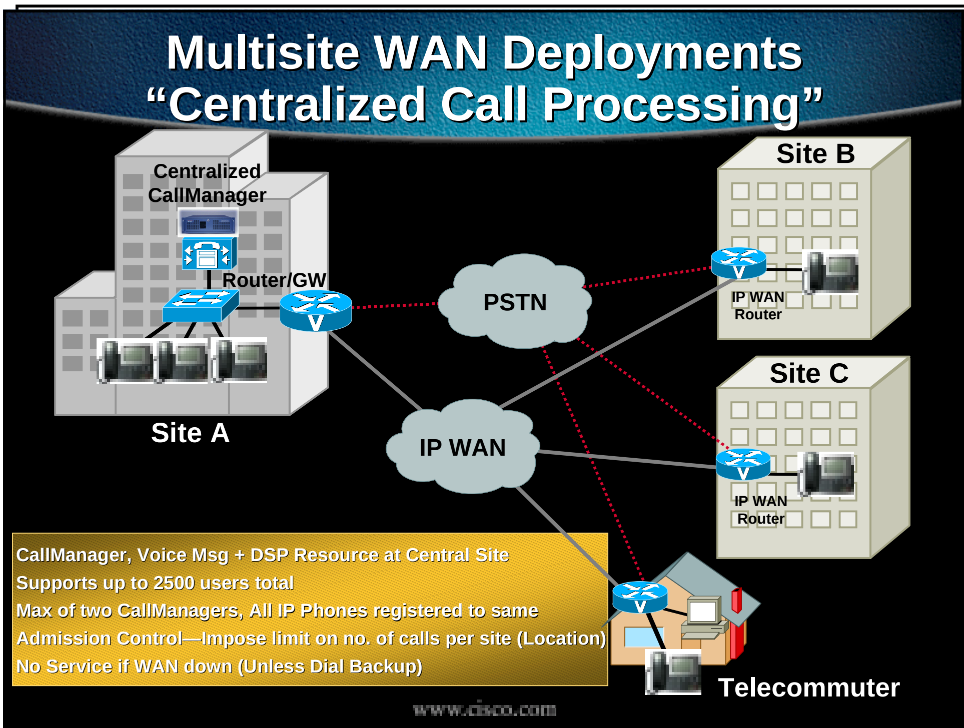
Multisite WAN Deployments
“Centralized Call Processing”
Centralized
CallManager
Router/GW
Site A
PSTN
IP WAN
Site B
IP WAN
Router
Site C
IP WAN
Router
CallManager, Voice Msg + DSP Resource at Central Site
Supports up to 2500 users total
Max of two CallManagers, All IP Phones registered to same
Admission Control—Impose limit on no. of calls per site (Location)
No Service if WAN down (Unless Dial Backup)
Telecommuter
www.cisco.com


Key Design Considerations
Yank Out My PBX,
This IP Phone System Is
Practically FREE
• Nothing comes for free, still need solid planning and design
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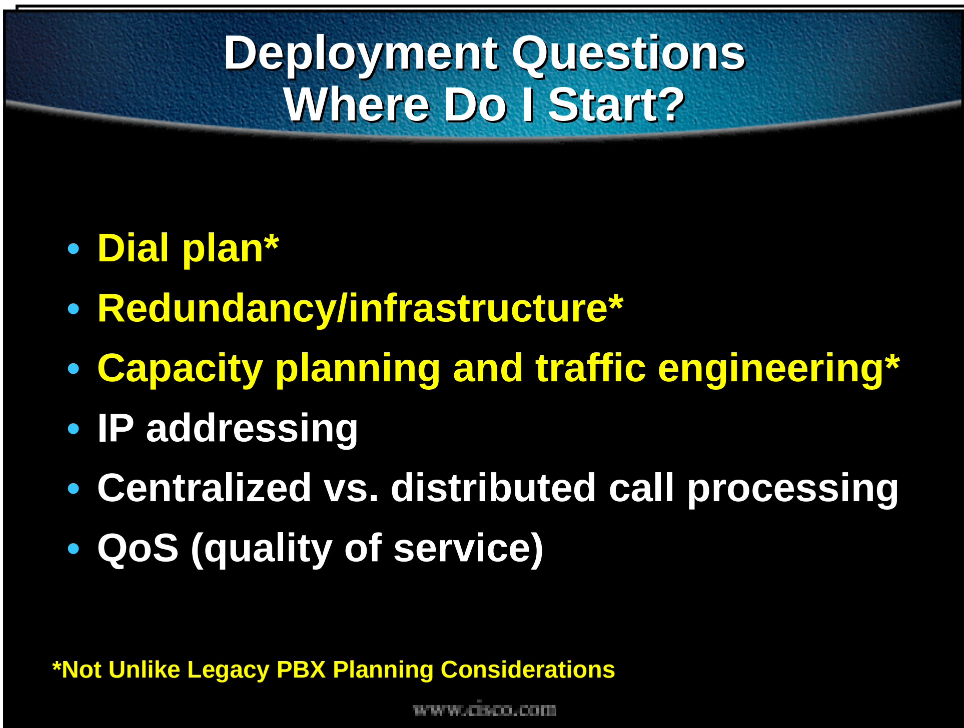
Deployment Questions
Where Do I Start?
• Dial plan*
• Redundancy/infrastructure*
• Capacity planning and traffic engineering*
• IP addressing
• Centralized vs. distributed call processing
• QoS (quality of service)
*Not Unlike Legacy PBX Planning Considerations
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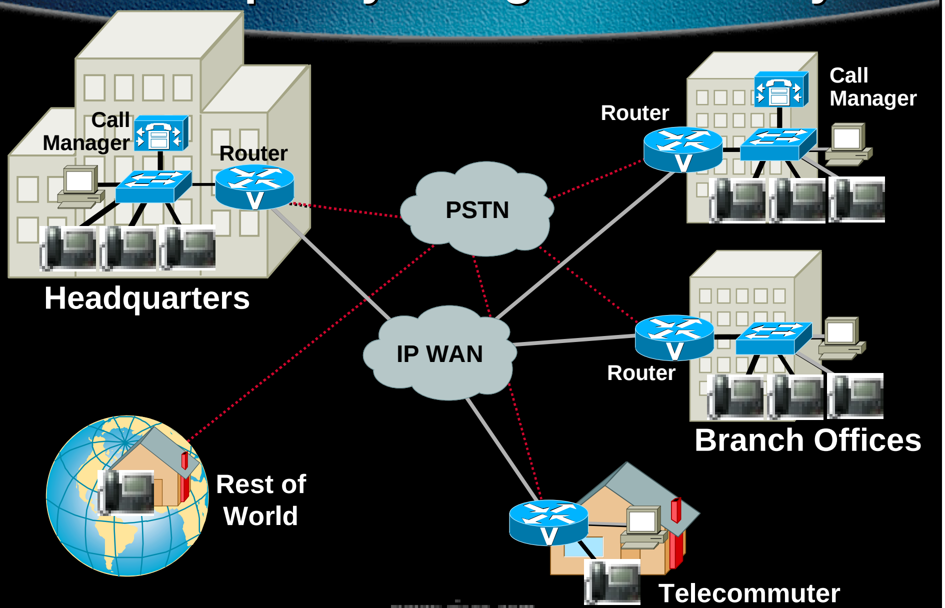
Logical Design Flow

- Campus/site
  - Campus infrastructure
  - CallManager cluster design
  - Gateway selection
  - Dial plan
- Multisite WAN
  - Multisite WAN call admission control
  - QoS (quality of service)
- Voice messaging

IP Telephony Design Case Study



Agenda

- IP Telephony (AVVID) System Ingredients
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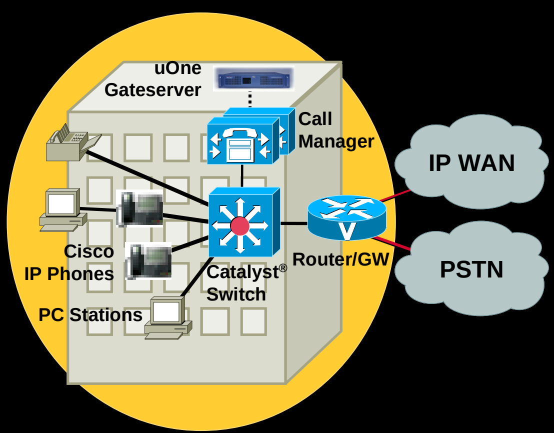
Primary Campus Site Considerations

- Redundancy
- Dial plan
- Power
- Scalability
- IP addressing
- QoS



Based on Facility Requirements

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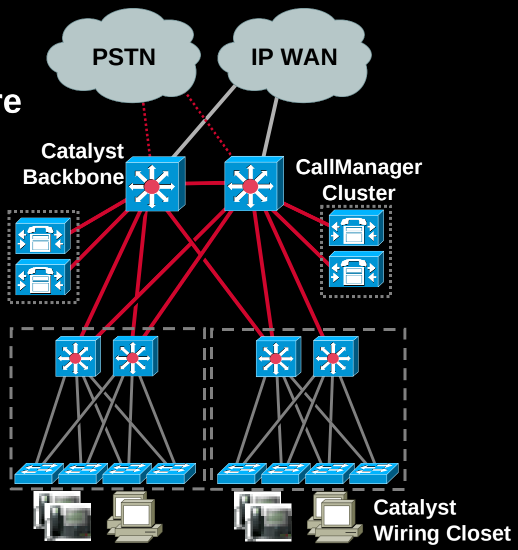
Redundant Infrastructure

- Switched infrastructure
- Redundant-switch uplinks
- Redundant core at Layer 2 and 3
- Redundant gateways
- Redundant WAN links
- Reliable power distribution



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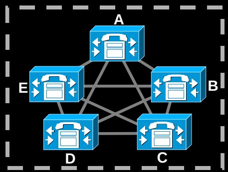
CallManager Cluster Scalability

CallManager Cluster



CallManager Cluster Sizing

1. 5 CallManagers Max in a Cluster (Cluster cannot extend across the WAN)
2. 2500 users Max per CallManager (Even under failure conditions)
3. Maximum of 10,000 users in a Cluster
4. Provision for CallManager Failure

CallManager Cluster IP Phone Provisioning

"Planning assumes for failure of one CallManager at a time"

CM's in Cluster	Max users per cluster
1	2500
2	2500
3	5000
4	7500
5	10,000

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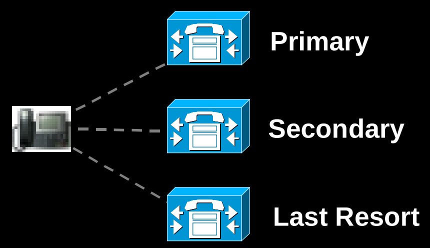
CallManager Groups



Each Device (IP Phone + Skinny Gateway) Has a Prioritized List of up to 3 Callmanagers to Which It Can Connect

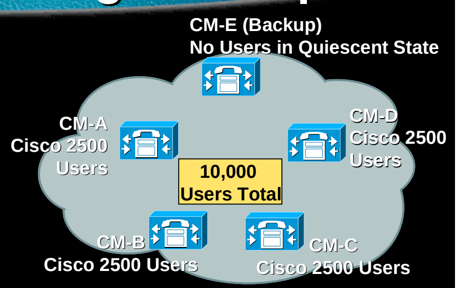
This Is Called a “CallManager Group”

This List Is Downloaded During Device Initialization

Best Practices Configuring CallManager Groups

For simple configuration dedicate a CM as a “backup” for others

Reduces number of CallManager groups needed to four

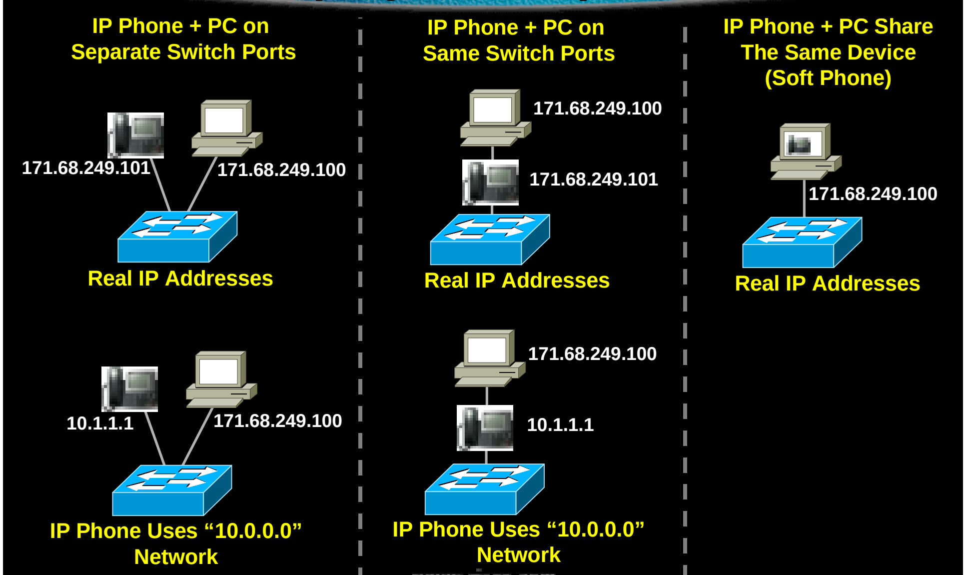


CallManager Group	Primary CallManager	Secondary CallManager
GroupA	CM-A	CM-E
GroupB	CM-B	CM-E
GroupC	CM-C	CM-E
GroupD	CM-D	CM-E

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

IP Addressing Deployment Options



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

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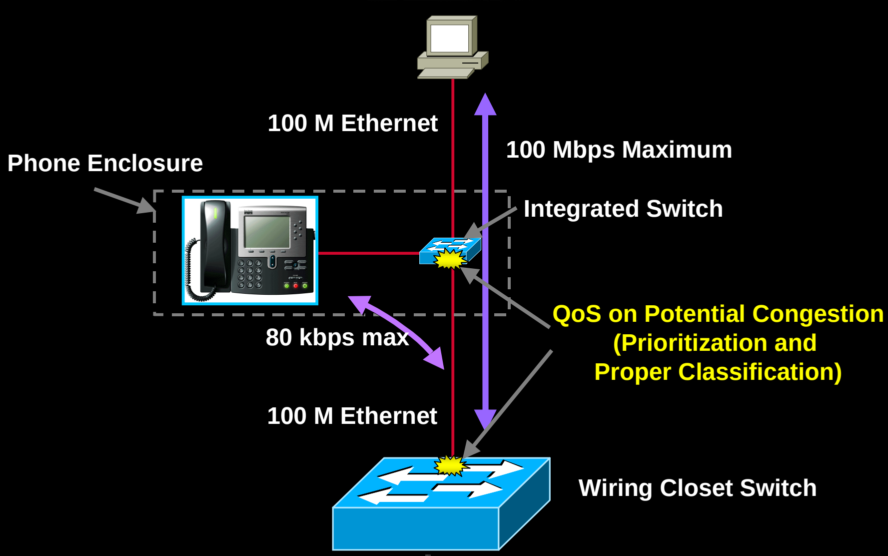
Potential Campus Congestion



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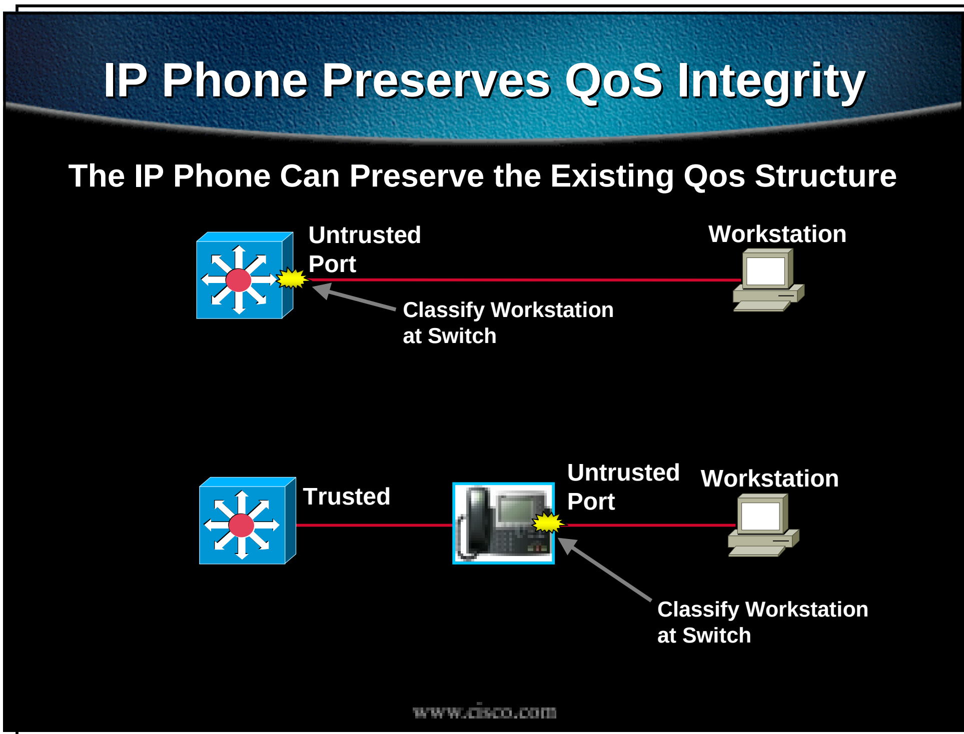
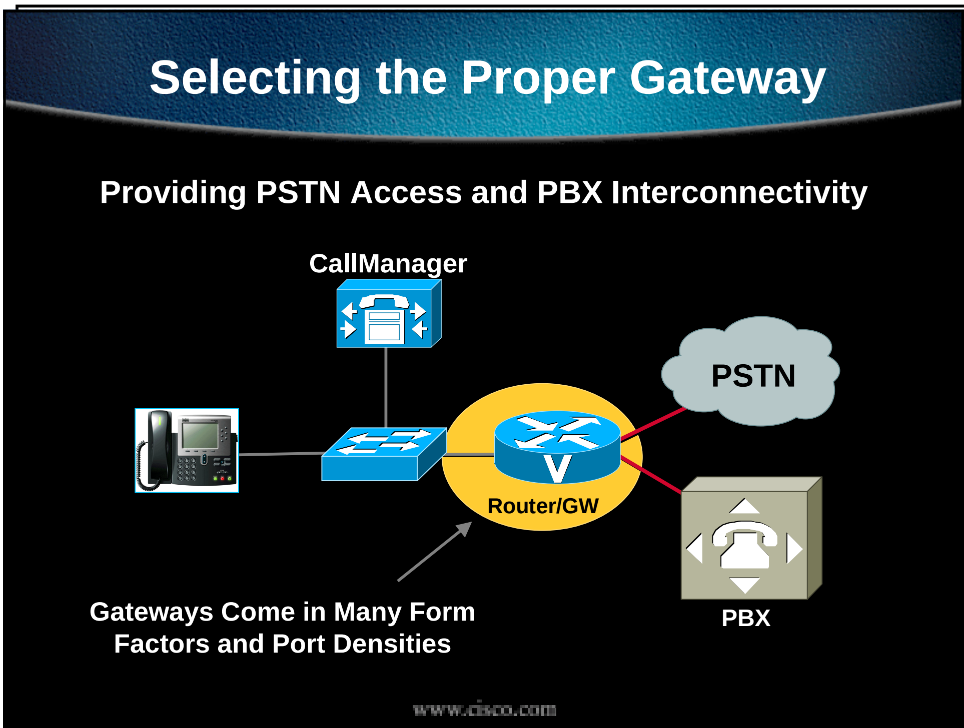

IP Phone Preserves QoS Integrity
The IP Phone Can Preserve the Existing Qos Structure
Untrusted Port
Workstation
Classify Workstation at Switch
Trusted
Untrusted Port
Workstation
Classify Workstation at Switch
www.cisco.com


Selecting the Proper Gateway
Providing PSTN Access and PBX Interconnectivity
CallManager
PSTN
Router/GW
PBX
Gateways Come in Many Form Factors and Port Densities
www.cisco.com

Gateway Selection Criteria

- Standalone vs integrated router/gateway

  Cost vs flexibility, functionality, and manageability

- Required voice port density
- Support for required PSTN signaling types
- Support for required WAN interface + QoS

Existing Sites Likely to Add Voice Ports to Existing Voice Enabled Router

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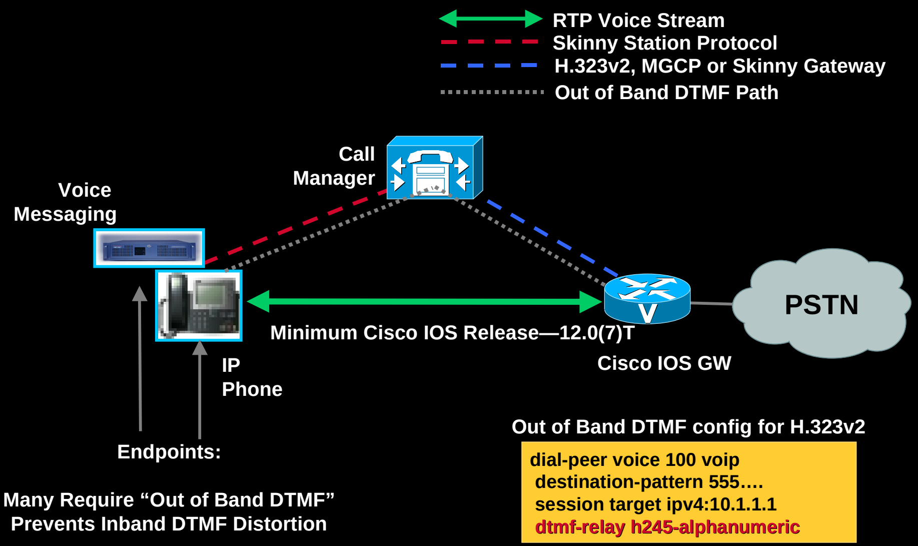
Gateway DTMF Support Out of Band




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

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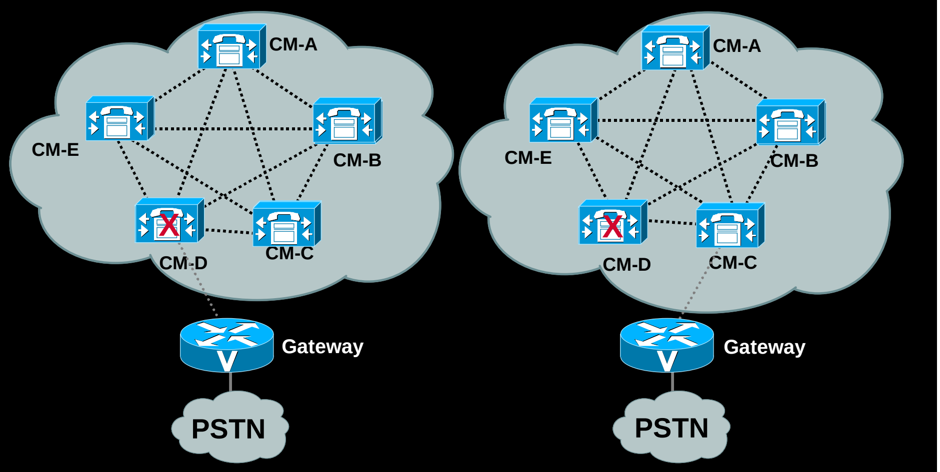
CallManager Redundancy Fail over for Gateways



Primary CCM for GW Fails

Mgcp/skinny GW Rehome to Sec CM
H.323v2 Falls Back to Alternate VoIP Peer

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PSTN/PBX Signaling Support

- T1/E1—CAS, PRI or Qsig
  - Cisco 1750/2600/3600
  - DT-24/30+
  - Cisco AS5300
  - Catalyst 4000/6000
  - 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)

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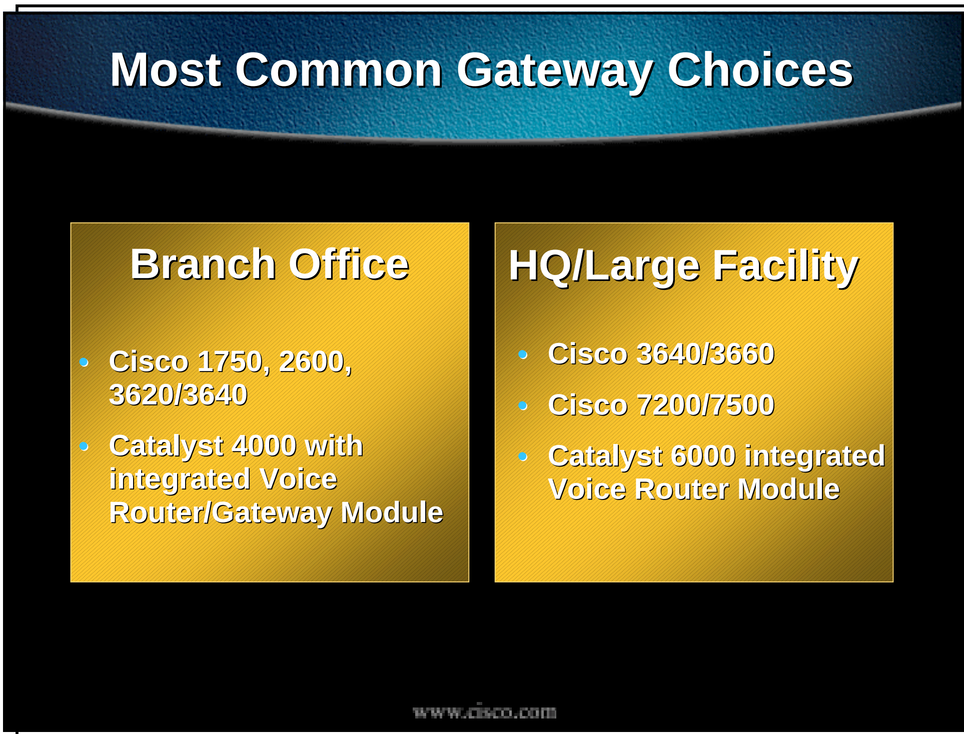
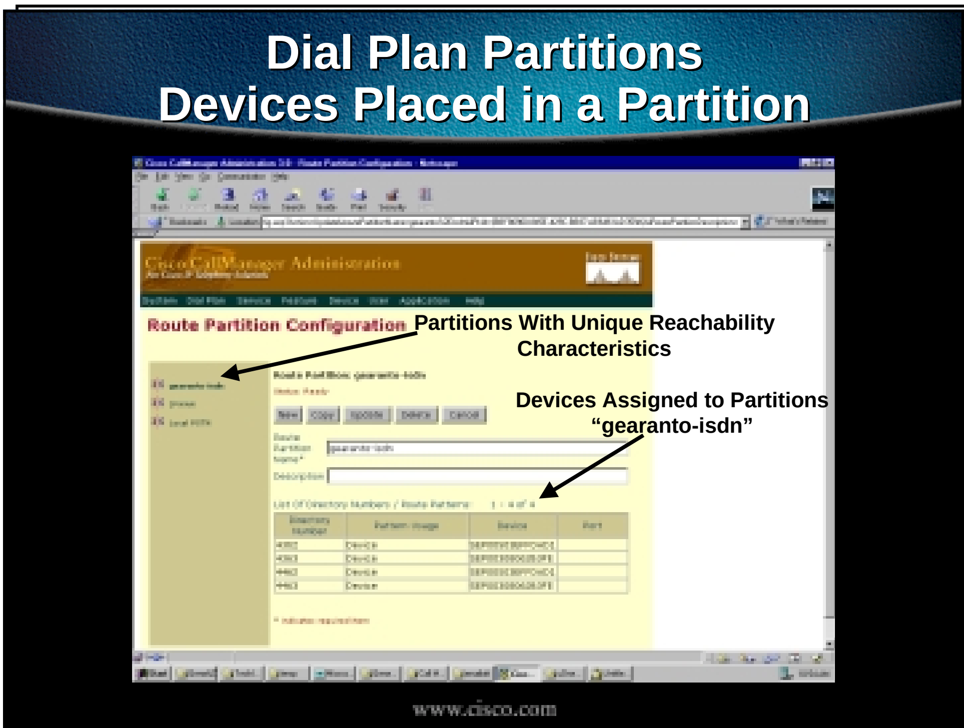
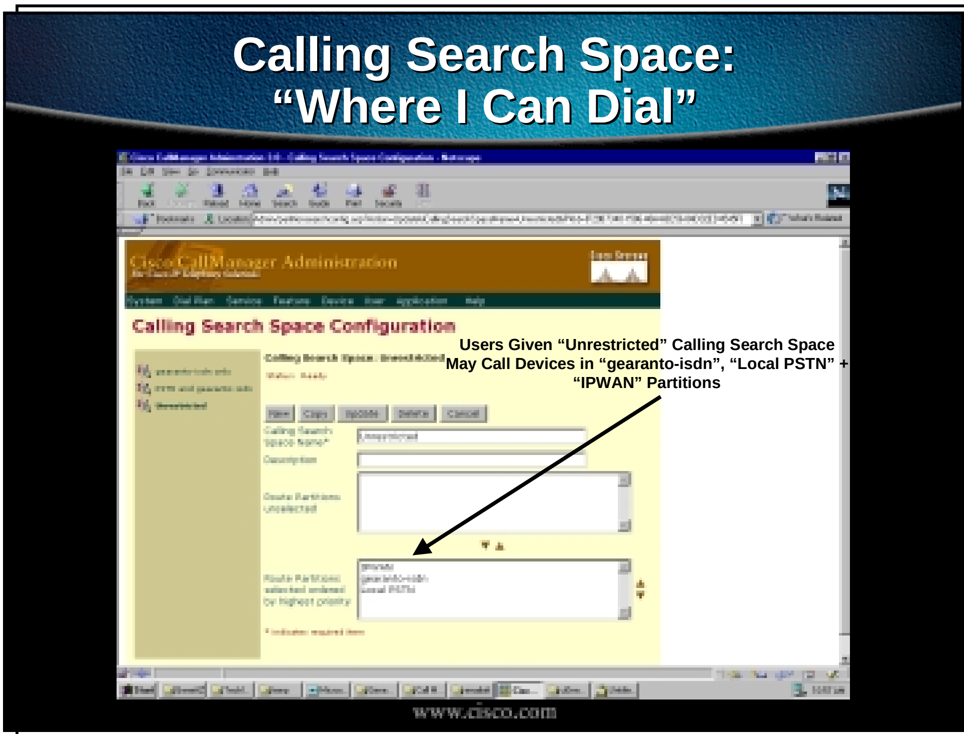
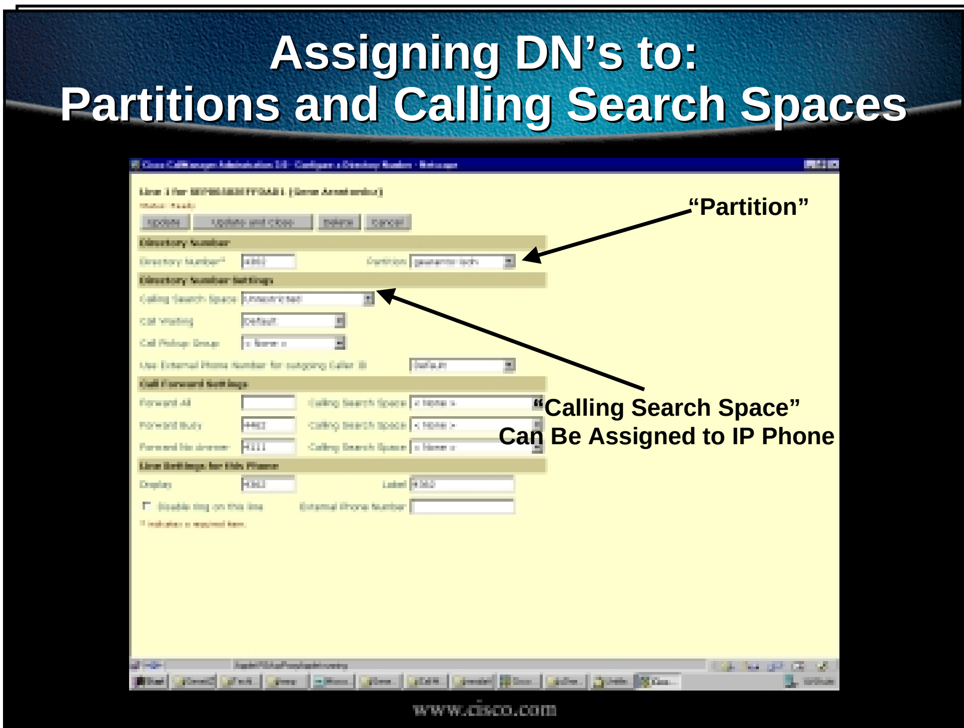
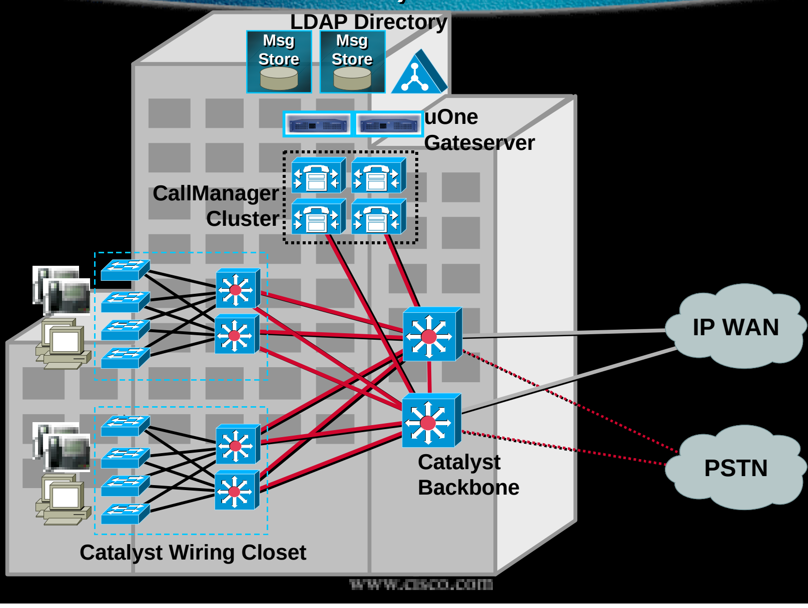

Most Common Gateway Choices
Branch Office
Cisco 1750, 2600, 3620/3640
Catalyst 4000 with integrated Voice Router/Gateway Module
HQ/Large Facility
Cisco 3640/3660
Cisco 7200/7500
Catalyst 6000 integrated Voice Router Module
www.cisco.com


Dial Plan Partitions
Devices Placed in a Partition
Cisco CallManager Administration
Route Partition Configuration
Partitions With Unique Reachability Characteristics
Devices Assigned to Partitions “gearanto-isdn”
www.cisco.com


Calling Search Space:
“Where I Can Dial”
Cisco CallManager Administration
Calling Search Space Configuration
Users Given “Unrestricted” Calling Search Space
May Call Devices in “gearanto-isdn”, “Local PSTN” +
“IPWAN” Partitions
www.cisco.com


Assigning DN’s to:
Partitions and Calling Search Spaces
“Partition”
“Calling Search Space”
Can Be Assigned to IP Phone
www.cisco.com

Large Campus Deployments 500—10,000 users



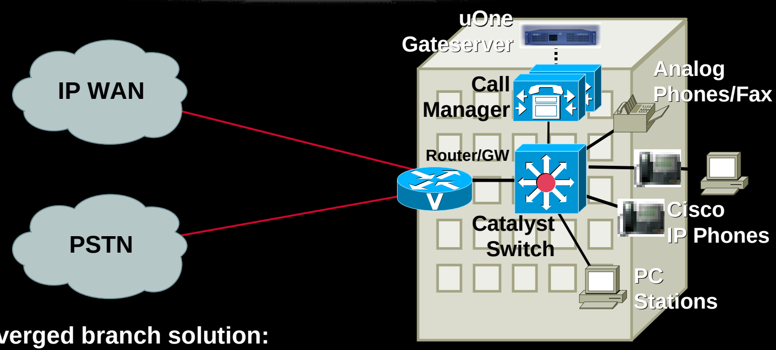
www.cisco.com

Large Campus Characteristics

- Up to 10,000 users
- Highest level of network availability
- Highly scalable
- Advanced QoS/management
- For IP WAN connectivity uses gatekeeper call admission control

www.cisco.com

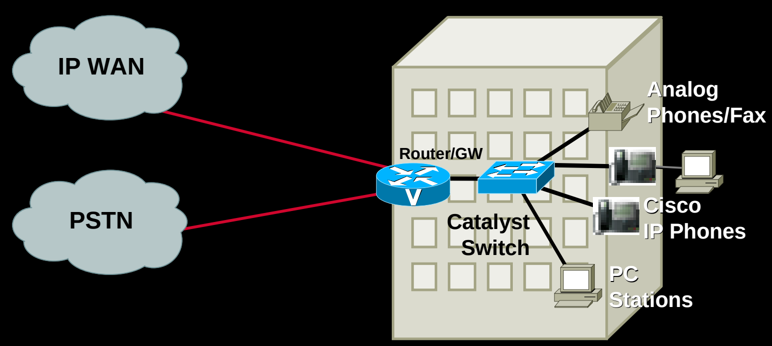
Branch Office Deployments
Less Than 500 Users (with Call Processing)



- Converged branch solution:
- Single infrastructure
- Minimal redundancy required (backup CallManager)
- Leverage existing router as PSTN gateway
- Can use integrated router/PSTN gateway module in catalyst switch
- Uses distributed call processing model (gatekeeper for admission control)

www.cisco.com

Branch Office Deployments
1-20 Users (Without Call Processing)



Converged Branch Solution:

- Minimal cost and resiliency
- Leverage existing router as PSTN gateway
- Uses centralized call processing model (location based admission control)

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Agenda

- IP Telephony (AVVID) System Ingredients
- General Enterprise Deployment Models
- Campus Design Considerations
- **Multisite WAN Considerations**
- Voice Messaging Design Considerations

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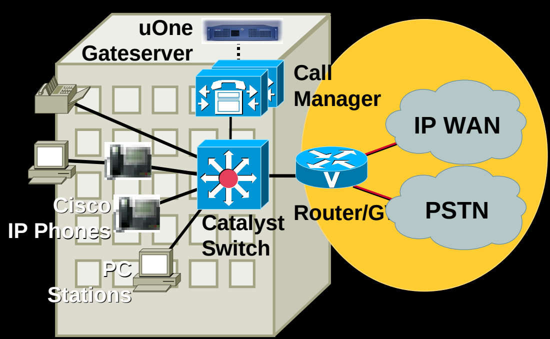
Primary Multisite WAN Considerations

- Call Admission Control
- Dial Plan
- QoS



Design depends on WAN Deployment Model

Distributed Call Processing
Centralized Call Processing

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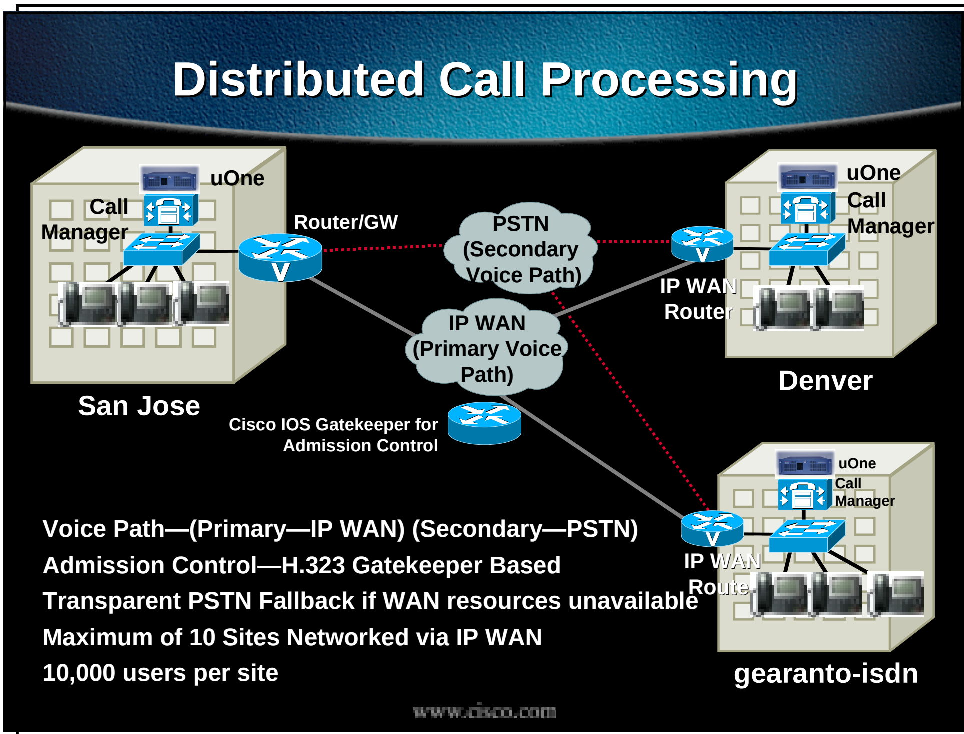
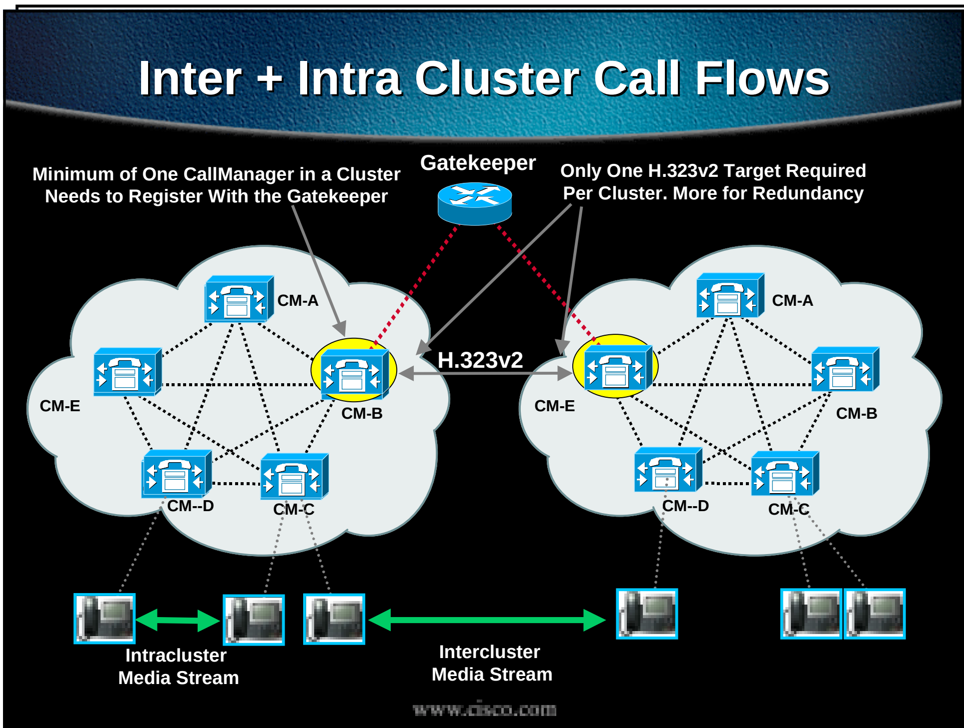
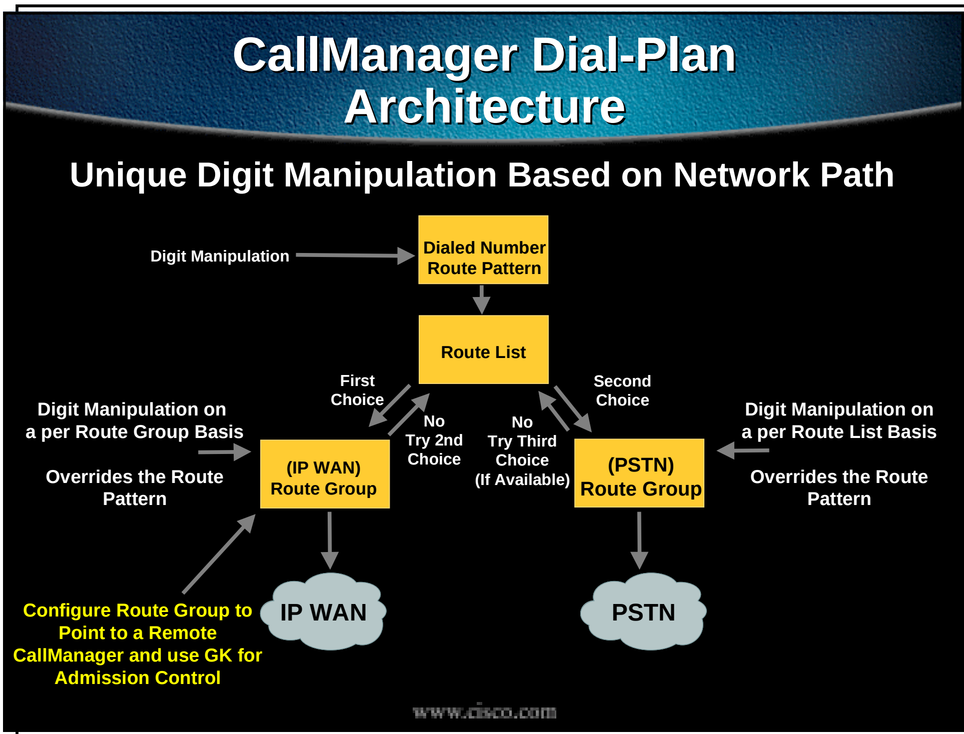
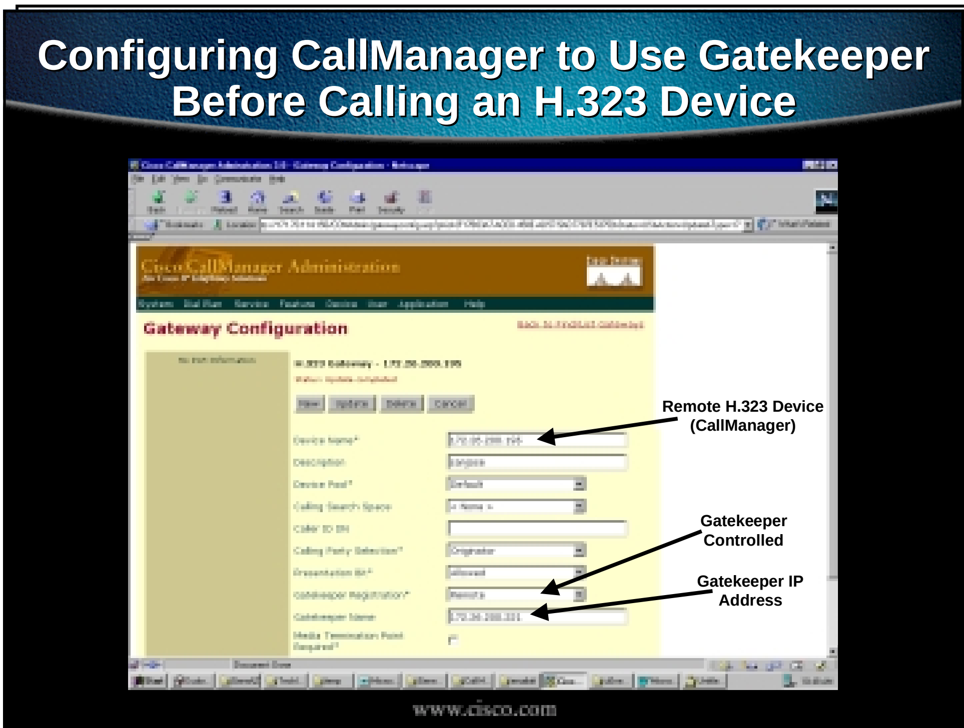

Distributed Call Processing
uOne Call Manager
Router/GW
PSTN (Secondary Voice Path)
IP WAN (Primary Voice Path)
IP WAN Router
uOne Call Manager
Denver
San Jose
Cisco IOS Gatekeeper for Admission Control
uOne Call Manager
IP WAN Router
Voice Path—(Primary—IP WAN) (Secondary—PSTN)
Admission Control—H.323 Gatekeeper Based
Transparent PSTN Fallback if WAN resources unavailable
Maximum of 10 Sites Networked via IP WAN
10,000 users per site
gearanto-isdn
www.cisco.com


Inter + Intra Cluster Call Flows
Minimum of One CallManager in a Cluster Needs to Register With the Gatekeeper
Gatekeeper
Only One H.323v2 Target Required Per Cluster. More for Redundancy
CM-A
CM-E
CM-B
CM--D
CM-C
H.323v2
CM-A
CM-E
CM-B
CM--D
CM-C
Intracluster Media Stream
Intercluster Media Stream
www.cisco.com


CallManager Dial-Plan Architecture
Unique Digit Manipulation Based on Network Path
Digit Manipulation
Dialed Number Route Pattern
Route List
First Choice
Second Choice
No Try 2nd Choice
No Try Third Choice (If Available)
Digit Manipulation on a per Route Group Basis
Overrides the Route Pattern
(IP WAN) Route Group
(PSTN) Route Group
Digit Manipulation on a per Route List Basis
Overrides the Route Pattern
IP WAN
PSTN
Configure Route Group to Point to a Remote CallManager and use GK for Admission Control
www.cisco.com


Configuring CallManager to Use Gatekeeper Before Calling an H.323 Device
Cisco CallManager Administration
Gateway Configuration
Remote H.323 Device (CallManager)
Gatekeeper Controlled
Gatekeeper IP Address
www.cisco.com

Gatekeeper Configuration

"Gatekeeper"

Assigning Gatekeeper Zone Name

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

Assigning CallManager to Zone based on source subnet

Assigning Maximum Bandwidth in or out of a region

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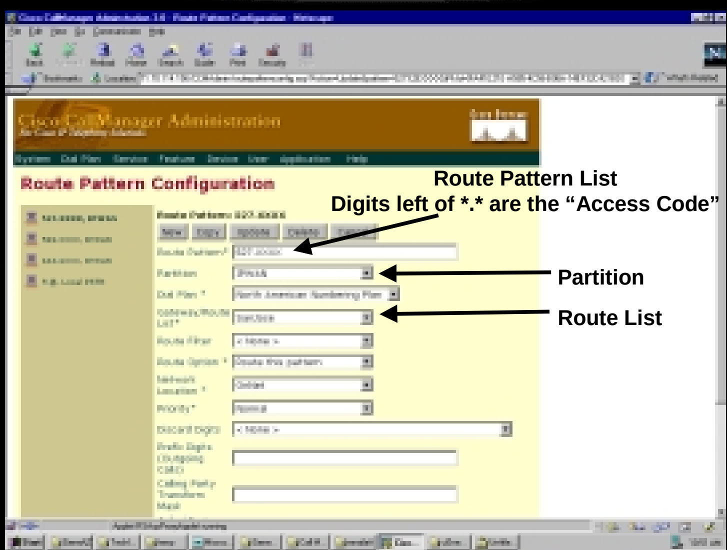
Route Pattern Configuration



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Route Pattern Notes

- CallManager matches most specific pattern
- Wildcards

X	Single digit (0-9)
N	Single digit (2-9)
@	North American Numbering Plan
!	One or more digits (0-9)
.	Terminates access code

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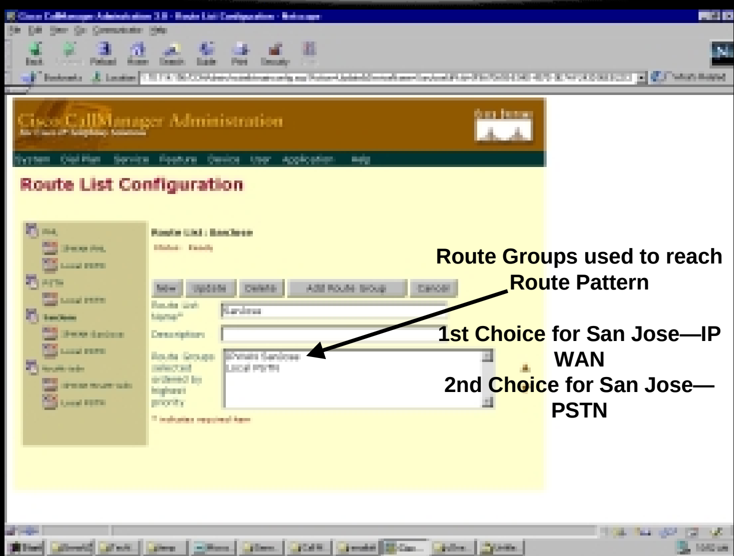
Route List Configuration



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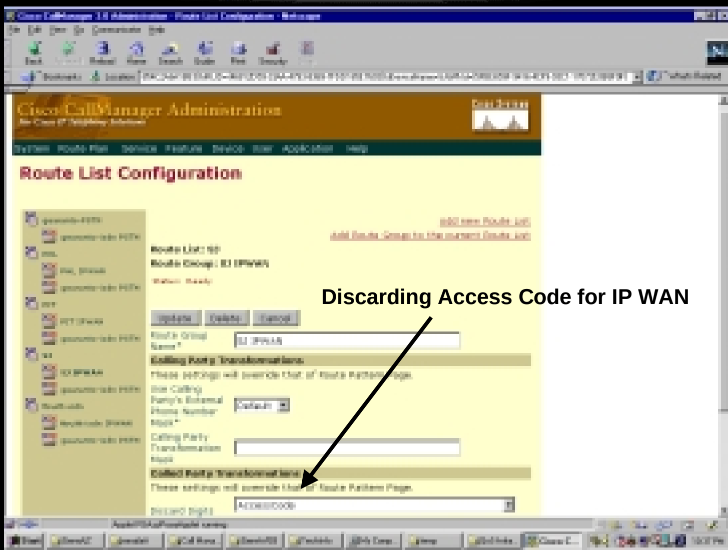
Route Group Configuration



Discarding Access Code for IP WAN

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Local Site IP WAN/PSTN Gateway Router Configuration

San Jose

Pointing all incoming calls from PSTN to Call Manager's IP Address

Dial Peer for all 10 digit PSTN Numbers

Voice T1 Configuration

```
dial-peer voice 1 voip
 codec g711ulaw
 dtmf-relay h245-alphanumeric
 destination-pattern 1...
 session target  ipv4:10.1.10.5
!
dial-peer voice 3 pots
 destination-pattern 1..........
 prefix 1
 port 1/0:1
!
controller T1 1/0
 framing esf
 linecode b8zs
 clock source line
 ds0-group 1 timeslots 1-24 type e&m-wink
!
 interface ethernet 0/0
 ip address 10.1.10.2 255.255.255.0
```

gearnto-isdn

Pointing all incoming calls from PSTN to Call Manager's IP Address

Dial Peer for all 10 digit PSTN Numbers

Voice T1 Configuration

```
dial-peer voice 1 voip
 codec g711ulaw
 dtmf-relay h245-alphanumeric
 destination-pattern 2...
 session-target ipv4:10.1.20.25
 !
dial-peer voice 3 pots
 destination-pattern 1..........
 prefix 1
 port 1/0:1
!
controller T1 1/0
 framing esf
 linecode b8zs
 clock source internal
 ds0-group 1 timeslot 1-24 type e&m-wink
!
interface ethernet 0/0
 ip address 10.1.20.1 255.255.255.0
```

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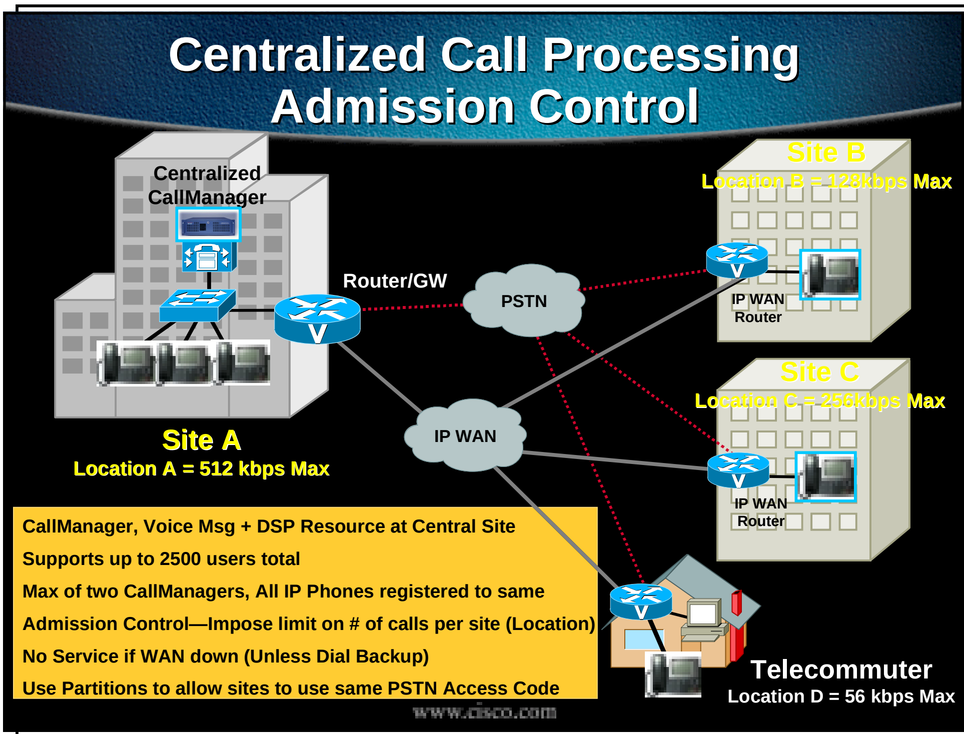
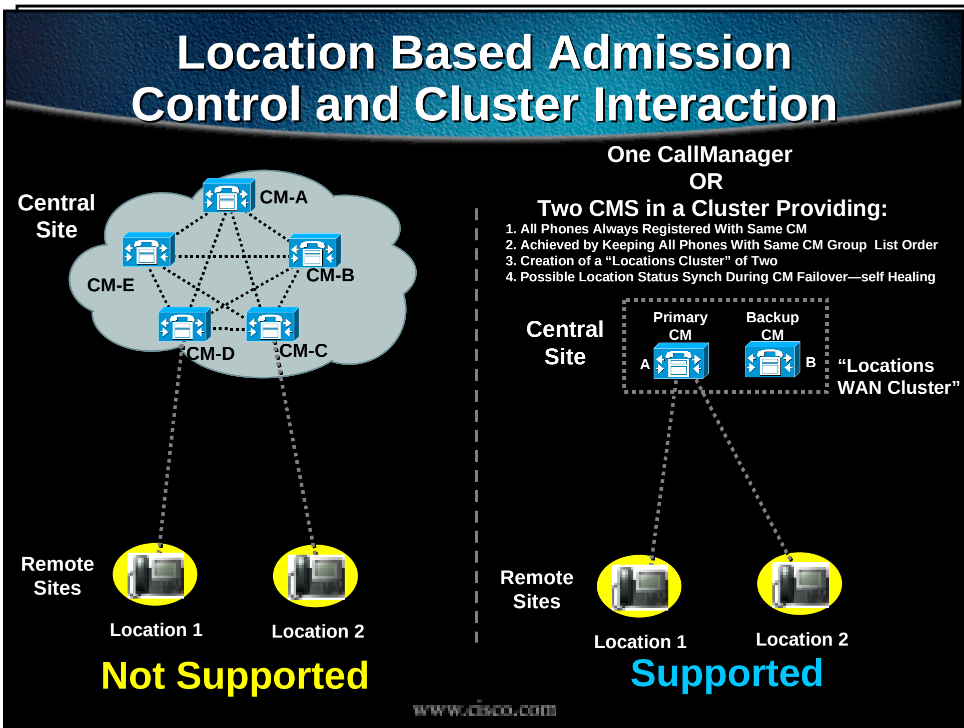
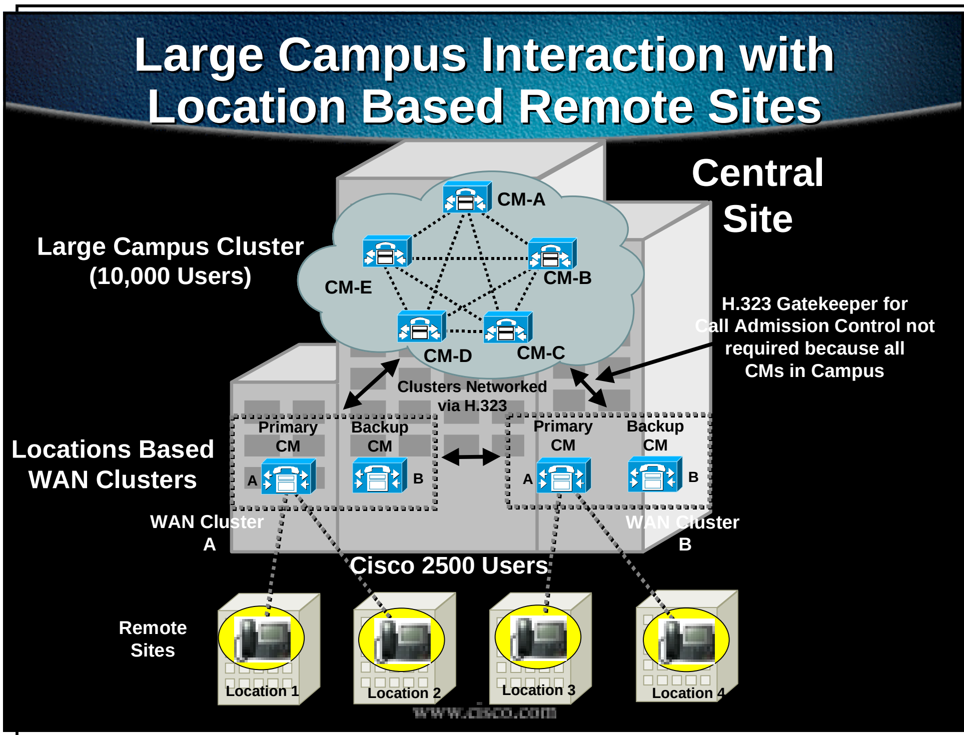
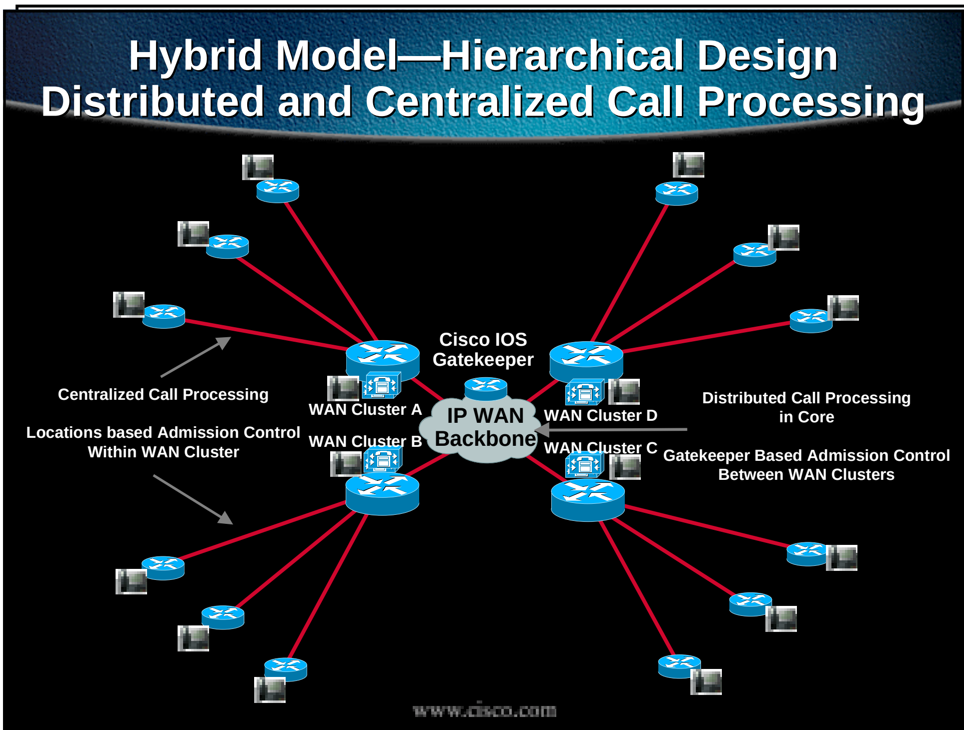


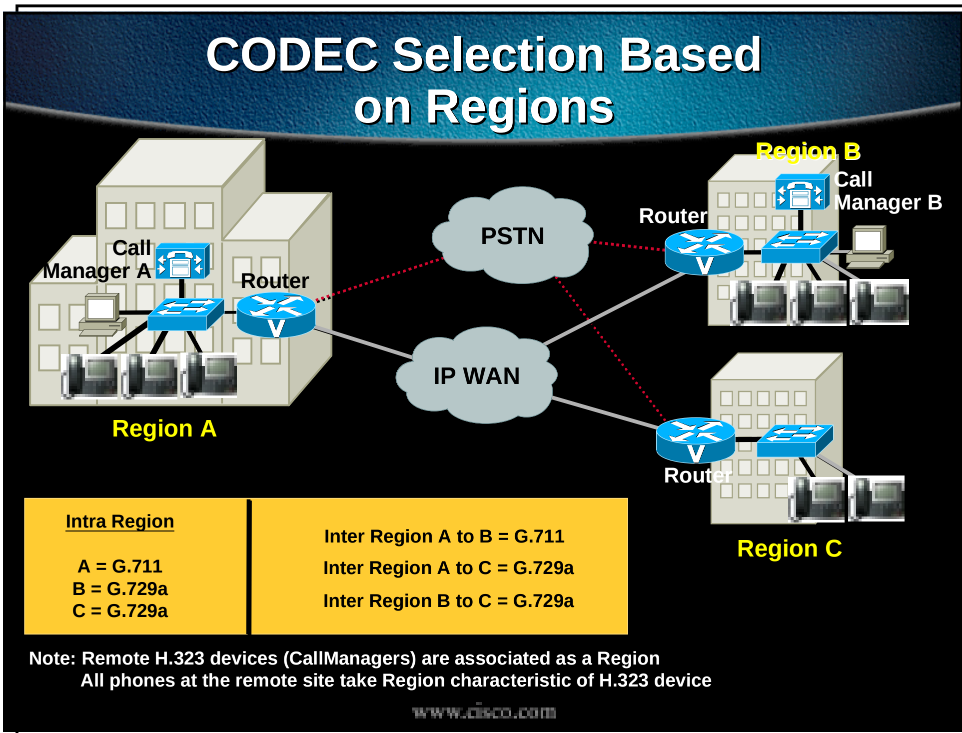
Centralized Call Processing
Admission Control
Centralized
CallManager
Router/GW
PSTN
Site B
Location B = 128kbps Max
IP WAN
Router
Site C
Location C = 256kbps Max
IP WAN
Router
IP WAN
Site A
Location A = 512 kbps Max
CallManager, Voice Msg + DSP Resource at Central Site
Supports up to 2500 users total
Max of two CallManagers, All IP Phones registered to same
Admission Control—Impose limit on # of calls per site (Location)
No Service if WAN down (Unless Dial Backup)
Use Partitions to allow sites to use same PSTN Access Code
Telecommuter
Location D = 56 kbps Max
www.cisco.com



Location Based Admission
Control and Cluster Interaction
Central
Site
CM-A
CM-E
CM-B
CM-D
CM-C
Remote
Sites
Location 1
Location 2
Not Supported
One CallManager
OR
Two CMS in a Cluster Providing:
1. All Phones Always Registered With Same CM
2. Achieved by Keeping All Phones With Same CM Group List Order
3. Creation of a "Locations Cluster" of Two
4. Possible Location Status Synch During CM Failover—self Healing
Central
Site
Primary
CM
Backup
CM
A
B
"Locations
WAN Cluster"
Remote
Sites
Location 1
Location 2
Supported
www.cisco.com


Large Campus Interaction with Location Based Remote Sites
Central Site
Large Campus Cluster (10,000 Users)
CM-A
CM-B
CM-E
CM-D
CM-C
H.323 Gatekeeper for Call Admission Control not required because all CMs in Campus
Clusters Networked via H.323
Locations Based WAN Clusters
Primary CM
Backup CM
A
B
WAN Cluster A
WAN Cluster B
Cisco 2500 Users
Remote Sites
Location 1
Location 2
Location 3
Location 4
www.cisco.com


Hybrid Model—Hierarchical Design Distributed and Centralized Call Processing
Cisco IOS Gatekeeper
Centralized Call Processing
Locations based Admission Control Within WAN Cluster
WAN Cluster A
WAN Cluster B
IP WAN Backbone
WAN Cluster D
WAN Cluster C
Distributed Call Processing in Core
Gatekeeper Based Admission Control Between WAN Clusters
www.cisco.com


CODEC Selection Based on Regions
Region B
Call Manager B
Router
PSTN
Call Manager A
Router
IP WAN
Region A
Router
Region C
Intra Region
A = G.711
B = G.729a
C = G.729a
Inter Region A to B = G.711
Inter Region A to C = G.729a
Inter Region B to C = G.729a
Note: Remote H.323 devices (CallManagers) are associated as a Region
All phones at the remote site take Region characteristic of H.323 device
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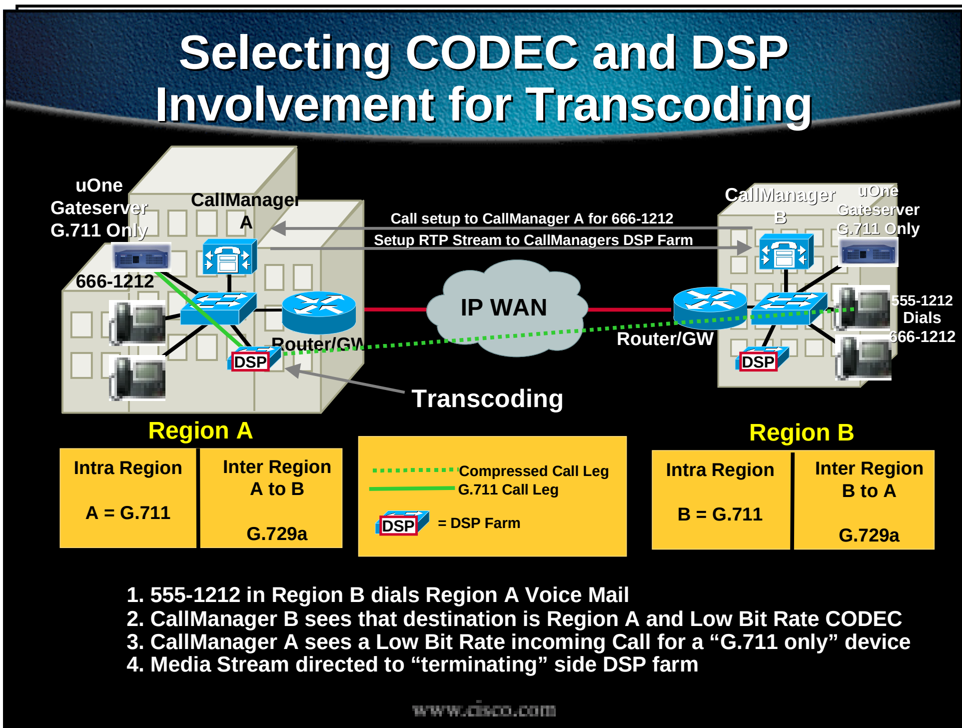
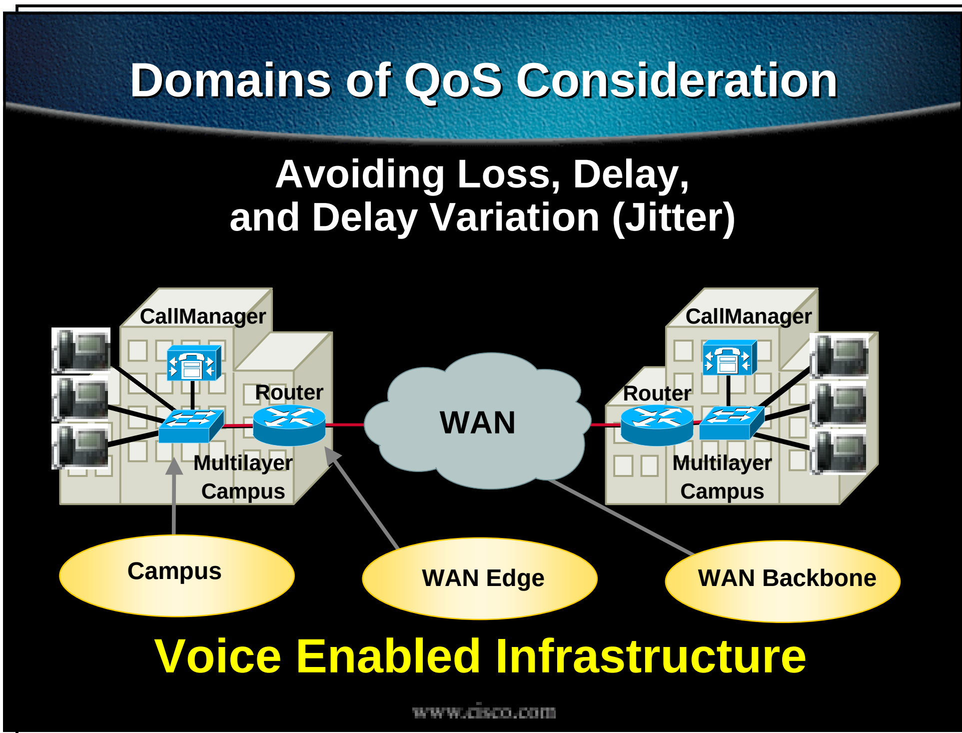
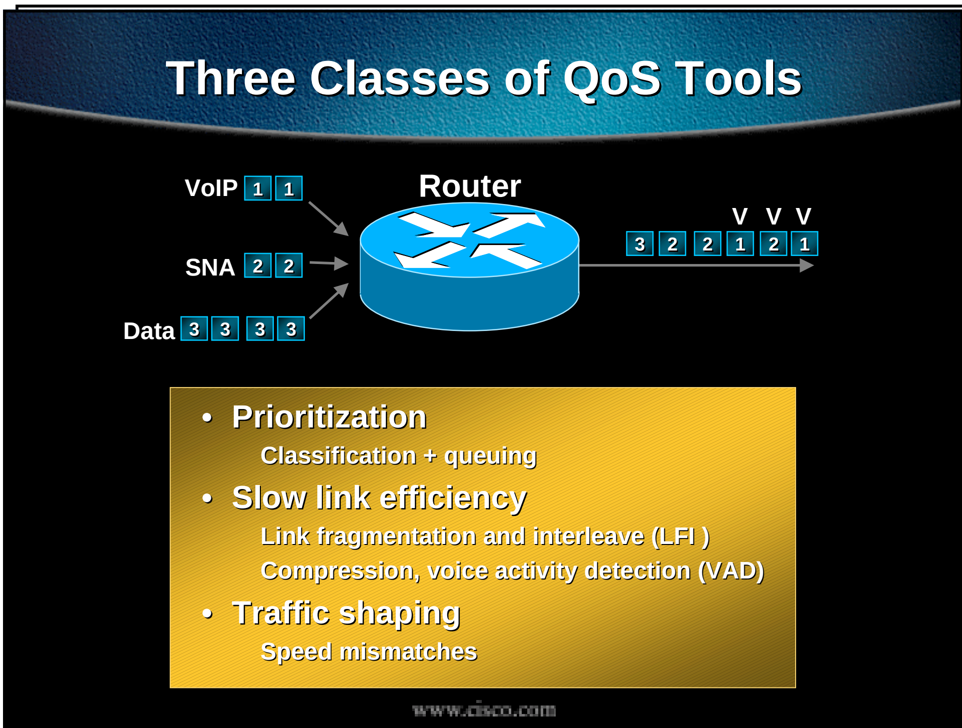
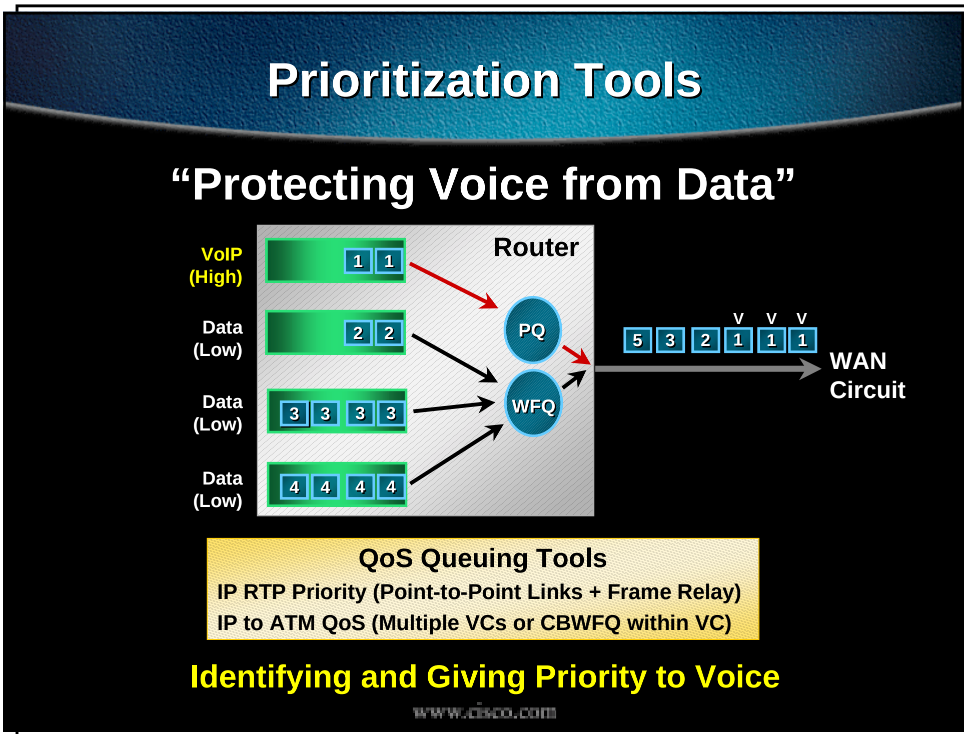
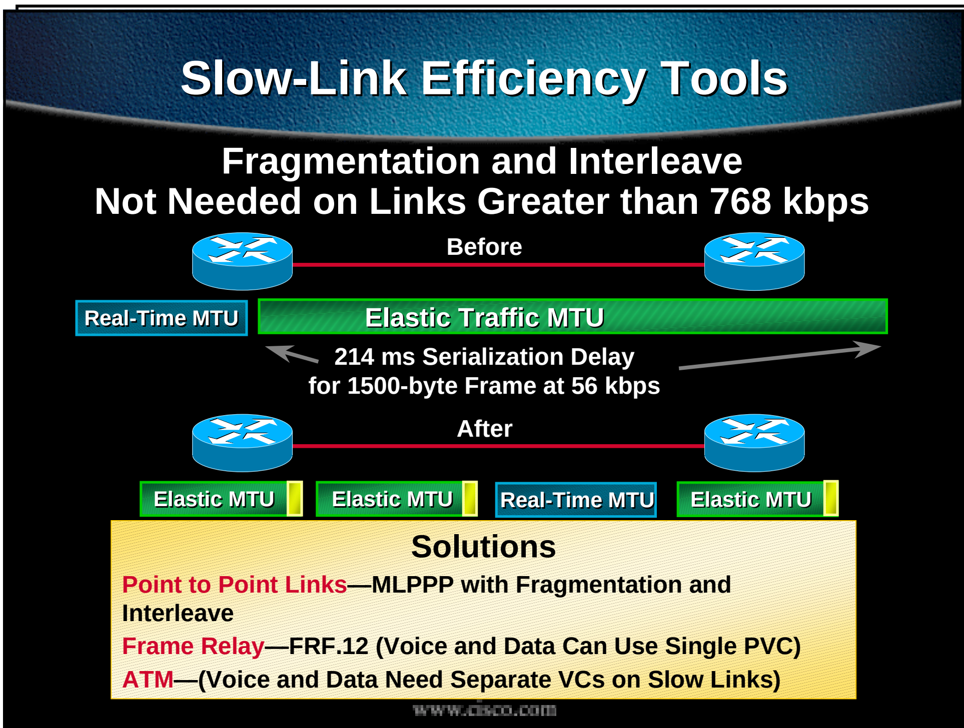

Selecting CODEC and DSP Involvement for Transcoding
uOne Gateserver G.711 Only
CallManager A
Call setup to CallManager A for 666-1212
Setup RTP Stream to CallManagers DSP Farm
CallManager B
uOne Gateserver G.711 Only
666-1212
IP WAN
555-1212 Dials 666-1212
Router/GW
Router/GW
DSP
DSP
Transcoding
Region A
Region B
Intra Region
A = G.711
Inter Region A to B
G.729a
Compressed Call Leg
G.711 Call Leg
DSP = DSP Farm
Intra Region
B = G.711
Inter Region B to A
G.729a
1. 555-1212 in Region B dials Region A Voice Mail
2. CallManager B sees that destination is Region A and Low Bit Rate CODEC
3. CallManager A sees a Low Bit Rate incoming Call for a "G.711 only" device
4. Media Stream directed to "terminating" side DSP farm
www.cisco.com


Domains of QoS Consideration
Avoiding Loss, Delay,
and Delay Variation (Jitter)
CallManager
Router
Multilayer
Campus
WAN
CallManager
Router
Multilayer
Campus
Campus
WAN Edge
WAN Backbone
Voice Enabled Infrastructure
www.cisco.com


Three Classes of QoS Tools
VoIP 1 1
SNA 2 2
Data 3 3 3 3
Router
V V V
3 2 2 1 2 1
• Prioritization
Classification + queuing
• Slow link efficiency
Link fragmentation and interleave (LFI)
Compression, voice activity detection (VAD)
• Traffic shaping
Speed mismatches
www.cisco.com


Prioritization Tools
"Protecting Voice from Data"
VoIP (High)
Data (Low)
Data (Low)
Data (Low)
Router
PQ
WFQ
WAN Circuit
QoS Queuing Tools
IP RTP Priority (Point-to-Point Links + Frame Relay)
IP to ATM QoS (Multiple VCs or CBWFQ within VC)
Identifying and Giving Priority to Voice
www.cisco.com


Slow-Link Efficiency Tools
Fragmentation and Interleave
Not Needed on Links Greater than 768 kbps
Before
Real-Time MTU
Elastic Traffic MTU
214 ms Serialization Delay
for 1500-byte Frame at 56 kbps
After
Elastic MTU
Elastic MTU
Real-Time MTU
Elastic MTU
Solutions
Point to Point Links—MLPPP with Fragmentation and Interleave
Frame Relay—FRF.12 (Voice and Data Can Use Single PVC)
ATM—(Voice and Data Need Separate VCs on Slow Links)
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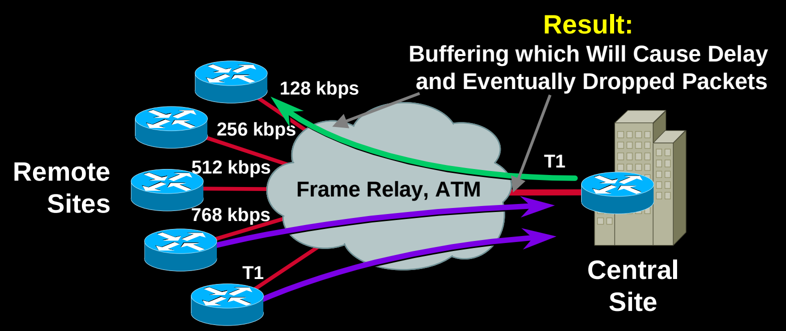
Traffic Shaping—When and Why?



1. Central to Remote-Site Speed Mismatch
2. To Avoid Remote to Central Site Over-Subscription
3. To Prohibit Bursting above Committed Rate

What Are You Guaranteed Above Your Committed Rate?

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How Much Bandwidth?

VoIP Packet

Link Header	IP Header	UDP Header	RTP Header	Voice Payload
X Bytes	20 Bytes	8 Bytes	12 Bytes	X Bytes

- Payload Size, PPS and BPS Vendor Implementation Specific
- For Example:

G.711 = 160 Byte Voice Payload at 50pps (80 kbps)
cRTP = 66 kbps

G.729a = 20 Byte Payload at 50pps (24 kbps)
cRTP = 10 kbps

Note—Link Layer Sizes Vary per Media

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Agenda

- IP Telephony (AVVID) System Ingredients
- General Enterprise Deployment Models
- Campus Design Considerations
- Multi-site WAN Considerations
- Voice Messaging Design Considerations

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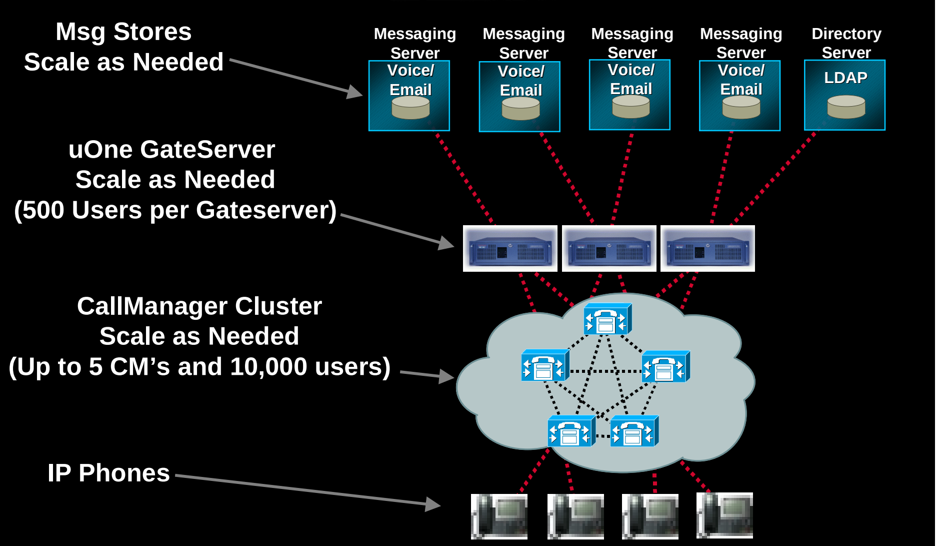
Scalable Growth from 1 to 10,000's of Users



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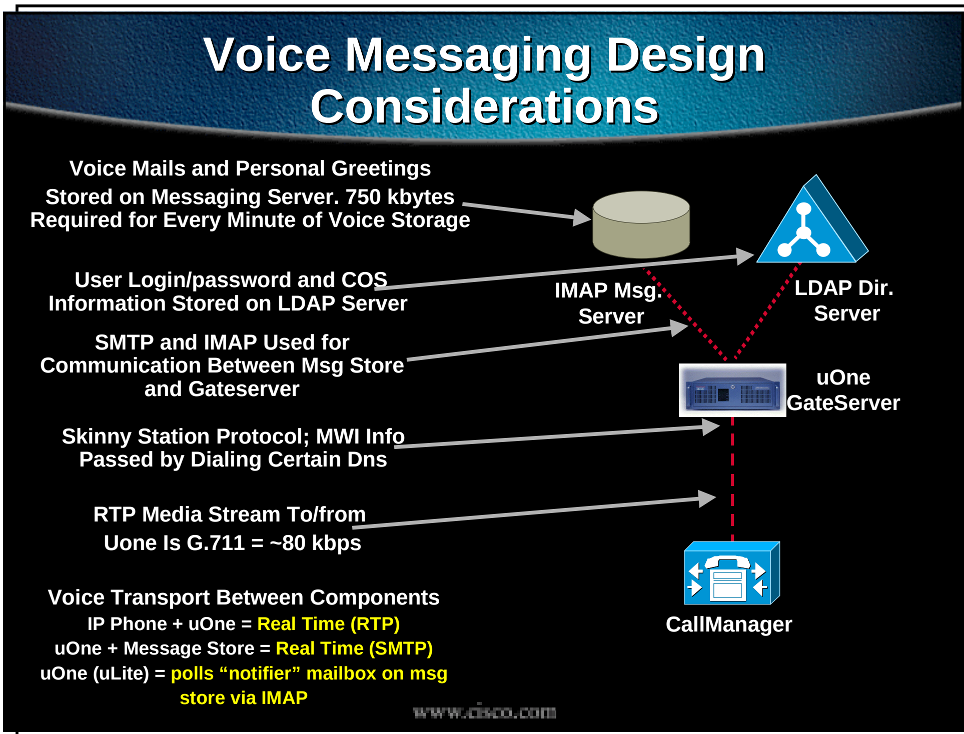
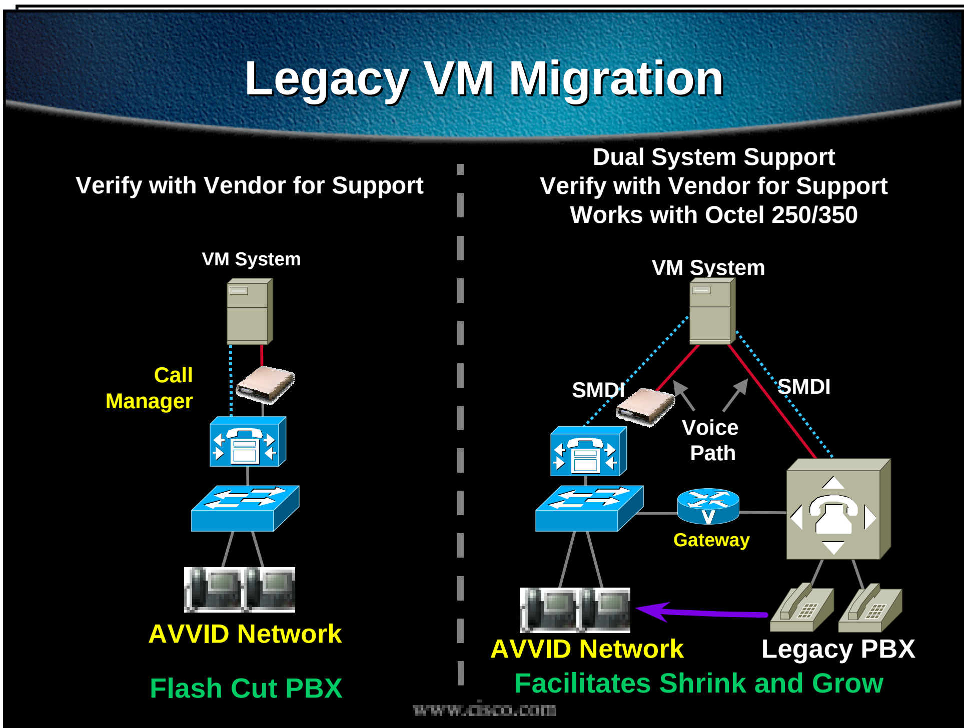

Voice Messaging Design Considerations
Voice Mails and Personal Greetings Stored on Messaging Server. 750 kbytes Required for Every Minute of Voice Storage
User Login/password and COS Information Stored on LDAP Server
SMTP and IMAP Used for Communication Between Msg Store and Gateserver
Skinny Station Protocol; MWI Info Passed by Dialing Certain Dns
RTP Media Stream To/from Uone Is G.711 = ~80 kbps
Voice Transport Between Components
IP Phone + uOne = Real Time (RTP)
uOne + Message Store = Real Time (SMTP)
uOne (uLite) = polls "notifier" mailbox on msg store via IMAP
IMAP Msg. Server
LDAP Dir. Server
uOne GateServer
CallManager
www.cisco.com


Legacy VM Migration
Verify with Vendor for Support
VM System
Call Manager
AVVID Network
Flash Cut PBX
Dual System Support
Verify with Vendor for Support
Works with Octel 250/350
VM System
SMDI
SMDI
Voice Path
Gateway
AVVID Network
Legacy PBX
Facilitates Shrink and Grow
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Please Complete Your Evaluation Form
Session 404
www.cisco.com


Cisco Systems
Empowering the Internet Generation SM
www.cisco.com