Advanced Enterprise Campus/WAN
IP Telephony Design and Implementation
Session 2006
Solution Sets: Toll Bypass and IP Telephony

This Session’s Focus

End-to-End IP Telephony with Application Enablement

What We Are Going to Build

Remotes With Local Call Processing
Remotes Without Local Call Processing

Rest of World
Branch Offices
Telecommuter
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

- CallManager Cluster
- LDAP Directory
- uOne Gateserver
- Catalyst Backbone
- IP WAN
- PSTN

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
### General Enterprise Deployment Models

- 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**

- **Call Processing**
- **PSTN Gateway/Router**
- **IP Phones/Endpoints**
- **Voice Messaging/Applications**
- **Campus Infrastructure/DSP Resources**

**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

Cisco CallManager

IP Phone Signaling Protocols
Skinny Station (IP Phone)
TAPI (Soft Phone)

Gateway Signaling Protocols
H.323v2
MGCP
Skinny Gateway

RTP Stream
UDP Ports 16384-32676

IP WAN
PSTN

1000
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**
  - 4 Wires
  - 48V DC Power
  - 10/100 Ethernet

- **Catalyst Power Patch Panel**
  - 8 Wires
  - Injects DC Power
  - 10/100 Ethernet

**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**
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

- QoS on Potential Congestion (Prioritization and Proper Classification)
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

Primary
Secondary
Tertiary/Last Resort

• 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

Endpoints:
- Many require “Out of Band DTMF”
- Prevents inband DTMF distortion

Voice Messaging
- RTP Voice Stream
- Skinny Station Protocol
- H.323v2, MGCP or Skinny Gateway
- Out of Band DTMF Path

Endpoints:
- Many require “Out of Band DTMF”
- Prevents inband DTMF distortion

Gateway Requirements
CallManager Redundancy

Primary CallManager for GW fails
MGCP/Skinny GWs rehome to Secondary CallManager H.323v2 fails 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

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### Dial Plan Goal: Transparent Automatic Route Selection

- **User dials 555-1212**
- **Does the WAN have sufficient resources to place call?** Yes
- **CCM strips 1st 3 digits for IP WAN**
- **“1212” sent to remote site** Proper QoS Ensured

- **User dials 555-1212**
- **IP WAN goes down or not enough resources**
- **CCM inserts “1408” for PSTN**
- **“1408-555-1212” sent to PSTN**
- **“1408-555-1212” sent to PSTN**

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### Diagram Details

- **CallManager**
- **IP WAN**
- **PSTN**
- **Router/GW**
- **User dials 555-1212**
- **IP Phones**
- **DSP**
- **CCM inserts “1408” for PSTN**
- **“Dynamic Alternate Routing”**

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*Presentation_ID.scr*
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**

- More Than One Route Pattern May Point to Same Route List
- Many Route Lists May Exist
- More Than One Route List May Point to Same Route Group

**Dial Plan Example Configuration**

"Distributed Call Processing"

- **San Jose**
  - Gatekeeper(s)
  - Secondary Voice Path: Prepend "1408" and send to PSTN
  - (408) 526-XXXX
  - 5 Digit Internal Dialing

- **Philadelphia**
  - Primary Voice Path: Strip "52" and deliver 61111 to Remote CallManager
  - (215) 555-XXXX
  - 5 Digit Internal Dialing

- **IP WAN**
- **PSTN**
- **Remote Sites**

- Users Required to Dial 7 Digits for Intersite Calls:
  - "526-1111"

- Five Digit Internal Dialing within a Site
- Seven Digit Dialing "Between" Sites
Philadelphia Route Pattern Configuration

Route Pattern Configuration

<table>
<thead>
<tr>
<th>Route Pattern Configuration</th>
<th>Route Pattern Configuration</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Access Code</strong></td>
<td><strong>Access Code</strong></td>
</tr>
<tr>
<td>52.XXXXX</td>
<td>52.XXXXX</td>
</tr>
</tbody>
</table>

**GK Controlled**: Send "XXXXX" in H.323 setup.

**Local GW receives DID and sends internal dial length to CM "XXXXX"**

**1st Choice Route Group**
- Discard Access Code "52"
- H.323 device—GK Controlled
- Route Group "SJ-IPWAN"—GK Controlled

**2nd Choice Route Group**
- Prepend "1408"
- Route Group "PHL-PSTN"

**Digit manipulation**
- **Digit manipulation**
- **Digit manipulation**

**Route List**
- SJ

**Partition**
- Who can reach "52.XXXXX"
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

Route Groups used to reach Route Pattern
(Each Route Group has unique Digit Manipulation)

1st Choice for San Jose—SJ IPWAN
2nd Choice for San Jose—PSTN
“SJ IPWAN” Route Group “Settings Override That of Route Pattern”

Route Details Configuration

Route List: SJ
Route Group: SJ IPWAN

Calling Party Transformations
- These settings will override that of Route Pattern Page.
- Use Calling Party Transformation Mask:

Discard Access Code

“PHL PSTN” Route Group

Route Details Configuration

Route List: SJ
Route Group: PHL PSTN

Calling Party Transformations
- These settings will override that of Route Pattern Page.
- Use Calling Party Transformation Mask:

Pre-pend “1408”
Route Group
“Prioritized Trunk Group”

Route Group Configuration

Device(s) that the Route Group Points to

Route Group

“Prioritized Trunk Group”

Typical Route Group Device Types

Route Group

Skinny Based
MGCP Based
H.323 Based
H.323 Based

DT-24+
Cat 6K GW’s
AT + AS GW’s

Configured as:
Device Protocol=H.225

VG200
Cisco IOS GWs
Remote CallManager

Configured as:
Device Protocol=Inter-cluster Trunk

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Minimal Gateway Configuration

For Incoming Calls
Assuming DID
(Direct Inward Dial)

Dial-Peer for all incoming calls
from PSTN to Call Manager’s IP Address
Must be G.711

Dial peer 1
voip
CODEC g711ulaw
dtmf-relay h245-alphanumeric
destination-pattern 6....
session target ipv4:10.1.10.5

Dial peer 2
pots

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

Dial Peer for 911 Services

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

Tenant A
Tenant B
Tenant C
Tenant D
CallManager

Internet
PSTN
Understanding Partitions and Calling Search Spaces

Partition/Calling Search Space
Analogous to:
Subnet/Access-List

Where You Are

Where You May Reach

Inbound Access-List (Calling Search Space)

Subnet (Partition) A
Subnet (Partition) B
Subnet (Partition) C

Configuring Calling Restrictions

San Jose

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

Access Code of “9” for local PSTN calls

Employee Phones
Lobby Phones

PSTN
Partitions Devices Placed in a Partition

Partitions with unique reachability Characteristics

<table>
<thead>
<tr>
<th>Partitions: SJ Users</th>
</tr>
</thead>
<tbody>
<tr>
<td>SJ Users</td>
</tr>
</tbody>
</table>

List of Directory Numbers / Route Patterns:

<table>
<thead>
<tr>
<th>Directory Number</th>
<th>Pattern Usage</th>
<th>Device Usage</th>
<th>Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>172</td>
<td>Device SEP005C36FD4EC</td>
<td></td>
<td></td>
</tr>
<tr>
<td>2072</td>
<td>Device SEP005C36FD4EC</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Called Search Space: “Where I Can Dial”

Calling Search Space Configuration

Calling Search Space: Unrestricted

Route Partitions

Devices assigned “Unrestricted” Calling Search Space may call devices in any Partition

<table>
<thead>
<tr>
<th>Route Partitions</th>
<th>Selected by Default or Highest Priority</th>
</tr>
</thead>
<tbody>
<tr>
<td>SJ Users</td>
<td></td>
</tr>
<tr>
<td>SJ PSTN</td>
<td></td>
</tr>
<tr>
<td>DIVAN</td>
<td></td>
</tr>
<tr>
<td>VTL</td>
<td></td>
</tr>
</tbody>
</table>
### Assigning Partitions and Calling Search Spaces “Individual Line/DN Level”

**Individual Line Configurations Override Main Configuration**

#### Directory Number Configuration
- **Directory Number**: 572
- **Partition**: 53 Users

#### Calling Search Space Configuration
- **Calling Search Space**: Restricted
- **Call Waiting**: Default

#### Call Forward and Pickup Settings
- **Destination**: Calling Search Space
- **Forward All**: < None >
- **Forward Busy**: 1172 < None >
- **Forward No Answer**: 1111 < None >
- **Call Pickup Group**: < None >

### Configuring Digit Translation

**Translation Pattern Configuration**

- **Translation Pattern**: 1XXX
- **Pattern Definition**
  - **Partition**: Default-internal users
  - **Numbering Plan**: North American Numbering Plan
  - **Route File**: < None >
  - **Calling Search Space**: Restricted
  - **Route Type**: < Route this pattern >
  - **Use Calling Party’s External Phone Number Mask**: < None >
- **Called Party Transformations**
  - **Called Party Transform Mask**: 4XXX
  - **Called Party Transformations**

Translates anything dialed with “1XXX” to “4XXX.”
**Dial 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

**San Jose**

- Employee Phones
- Attendant (4111)
- PSTN
- PSTN DID Range 408-555-1XXX

**PSTN DID Range does not match internal Range**

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

**Voice/Unified Messaging**

1. After 3 Rings Call forwarded to Voice Mail (uOne)
2. Gateserver performs Directory Lookup
3. Light MWI on Phone
4. Voice Mail to inbox via SMTP
5. uOne Gateserver (Registers as IP Phone with CallManager)

**Common Uses**

- PSTN
- Router/GW
- CallManager
- VSM
- IP
- WAN
- LDAP
- Directory
- Message Store
- IP Phones

**uOne Gateserver**

- Indicates one of the servers responsible for managing the messaging system.
### 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

All VM Components on MCS-7835

Message Server
Directory Server
CallManager 3.0

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)

- **Campus** Classification Queuing
  - "Identifying Voice"
- **WAN Edge** Classification Queuing
  - "Prioritizing Voice"
- **WAN Backbone** Provisioning
  - "Capacity Planning"

Need for Admission Control

"Protecting Voice from Voice"

**Example:**

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

- Call No. 1
- Call No. 2
- Call No. 3

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**

- **Call Flow**
- **H.323 RAS Signaling**

- **Cisco IOS Gatekeeper**
  - May I Make a Call? (ARQ)
  - I Am Registering As A VoIP GW (RRQ)

- **Yes You Can (ACF)**
- **No You Can’t (ARJ)**
  - “Call Placed”

- **PSTN “Voice Overflow”**
- **Cisco 3600**
  - Zone 1
  - San Jose
  - Zone 2
  - Philadelphia

- **Basic CallManager Gatekeeper Interaction**
  - Can Dynamically Send Calls Across PSTN If IP WAN Unavailable

**Enabling Gatekeeper Use Before Sending Call Across IP WAN**

- **Gateway Configuration**
  - Defining H.323 GW As an “Inter-cluster Trunk” (Remote CallManager)
  - Remote H.323 Device (CallManager IP Address)
  - Device Pool to Define CODEC Used for Calls to This Device
  - Enabling Device As Gatekeeper Controlled (Will Register IP SA/DA With gatekeeper)
  - Gatekeeper IP Address
Gatekeeper Configuration

```
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
```

“Cisco IOS Gatekeeper”

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

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

Region CODEC Matrix

- “PHL” to “PHL” = G.711
- “PHL” to “IPWAN-G729” = G.729

Gatekeeper(s)

San Jose

Philadelphia

IP WAN

PSTN

Devices

IP Phones at PHL
SJ CallManager

Region “PHL”
“IPWAN-G729”

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DSP Resources for Transcoding

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”

Site A
Location A = 512 kbps Max

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)

Site B
Location B = 128kbps Max

Site C
Location C = 256kbps Max

Telecommuter
Location D = 56 kbps Max

Admission Control—BW Limitation by “Location”

Location 1
Centralized CallManager Cluster

Location 2
Location 2 = 128kbps Max

Site 2
IP WAN Router

Site 3
Location 3 = 256kbps Max

Site 3
IP WAN Router

Can Impose Max Voice Bandwidth
In or Out of a 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 Configuration

- Location: branch 1
- Status: Ready
- Location Name: branch 1
- Bandwidth: 100 kbps

If the audio quality is poor or choppy, lower the bandwidth setting. For 1500 use multiples of 56kbps or 64kbps.

* indicates required item

Location Based Admission Control and Cluster Interaction

- One CallManager OR Two CMs in a Cluster Where All Phones Register to Same CallManager
- Central Site
- Remote Sites
- Location 1: Not Supported
- Location 2: Supported

Remote Sites

Central Site

CM-A

CM-B

CM-D

CM-E
Large Campus Interaction with Location-Based Remote Sites

Central Site

H.323 Gatekeeper for Call Admission Control not required because all CMs in Campus

Clusters Networked via H.323

IP WAN

Large Campus Cluster (10,000 Users)

Locations Based Centralized Clusters

Remote Sites

Location 1

Location 2

Location 3

Location 4

2500 Users

2500 Users

Large Campus Interaction with Location-Based Remote Sites

Partitions/Multitenant Ability

Centralized CallManager

Partition 1

Router/GW

Dial “9” for Local PSTN GW

Site 1

Site 2

Site 3

Partition 1

Partition 2

Partition 3

PSTN

IP WAN

Dial “9” for Local PSTN GW

Dial “9” for Local PSTN GW

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

Partition 1

Partition 2

Partition 3

GW

GW

GW

Dial “9” for Local PSTN GW

Dial “9” for Local PSTN GW

Dial “9” for Local PSTN GW
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)
Recommended Multisite WAN Topologies

Topologies Based on Gatekeeper and Locations Based Admission Control

- **Hub and Spoke**
  - Avoid Voice Over Subscription of WAN Links
  - 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**

1. **Digital PIC Integration**
   - (Most Common Type of Octel to PBX integration)
   - **VM System**
   - **Octel**
   - **Legacy PBX**

2. **Verify with Vendor for SMDI Support**
   - **VM System**
   - **Octel**
   - **SMDI**
   - **Analog Trunks**
   - **Gateway**
   - **CallManager**

   **CallManager Based Network**

   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 Vendor support
- May require PBX/VM conversion to SMDI

1. **VM System**
2. **SMDI**
3. **CallManager**
4. **Voice Path**
5. **Gateway**
6. **AVVID Network**
7. **Legacy PBX**
Simultaneous System Support

Octel 250 + 350

Supported

"Supported" by Lucent

Call Manager

AVVID Network

Flash Cut PBX

Octel

Simultaneous System Support Works with Octel 250/350

Legacy PBX

SMDI

Gateway

Voice Path

SMDI

AVVID Network

Facilitates Shrink and Grow

Legacy (Octel) 200/250/300/350

Lucent Intuity, Siemens Phones Mail

Single System

Call Manager

AVVID Network

Flash Cut PBX

VM System

Simultaneous System Support

VTG VBMux

Gateway

SMDI

VM System

Legacy PBX

AVVID Network

Facilitates Shrink and Grow
### Converting PBX to Support SMDI

![Diagram showing conversion process]

**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)

Example—Cisco San Jose Campus Migration Option

Allows for Transparent Migration from Lucent PBX to CallManager Cluster
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 your environment and stay within design guidelines
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