Enterprise IP Telephony Design and Configuration

Session 2102
VoIP Solution Sets:
Toll Bypass and IP Telephony

End-to-End IP Telephony with Application Enablement

IP Telephony Design Goals

VoIP Solution Sets:
Toll Bypass and IP Telephony

End-to-End IP Telephony with Application Enablement
Session Focus...CAMPUS Deployment and Configuration

Agenda

- Infrastructure
- End-Point Connectivity Options
- Power to IP Phones
- IP Addressing (Plug and Play Implementation)
- Quality of Service (QoS)
- CallManager Operation and Configuration
- Gateway Selection
- Dial Plan 101
- Voice Mail
- Summary
Infrastructure

Multilayer Design Architecture

Start with a High-Speed Core
Core Could Be Layer 2 or 3
Core Provides Transport
Add Building Modules
Let’s Look at a Module
Start with Layer 3 Distribution
Add the Access Layer
Collapse Access to Distribution

**Join Distribution Switches?**
Depends on Workgroup Servers
Server Farm Is Another Module
Typical Enterprise Network

End-Point Connectivity Options
Connectivity Options

1. Single Cable
2. Multiple Cables
3. Soft Phone

Cisco IP Phone Port Speed and Duplex

- Cisco IP phone sets P0 and P1 to auto-negotiate hard-coded
- In the absence of auto-negotiation, set switch (or PC) to 100M FD or 100M HD or 10M HD
  - 10M full-duplex (not a standard) will result in FCS and alignment errors
Power to IP Phones

3 Ways to Power IP Phones

• Inline power
  Needs powered line cards for catalyst switches
  Uses pairs 2 and 3 (same as Ethernet) for delivering power

• External power
  Needs external power patch panel
  Patch panel delivers power over pairs 1 and 4

• Wall power
  Needs DC converter for connecting IP phone to wall outlet

Combination of Ways Can Be Used for Redundancy
### Catalyst Power Patch Panel

- **Switch Side RJ-45**
  - 1, 3, 5
  - 2, 4, 6
- **Phone Side RJ-45**
  - 1, 3, 5
  - 2, 4, 6

- 2 Pairs (4 Wires)
- 4 Pairs (8 Wires)

### Switch Configuration Example for Inline Power Mechanism

- **Configuring inline power**: (for Catalyst OS-based switches)
  
  **Command**: `set port inlinepower <mod/port> <auto|off>`
  
  **Successful output**: Inline power for port 7/1 set to auto.
  
  **Failure output**: Failed to set the inline power for port 7/1
  
  For IOS-Based Inline Power Switch: (Interface Command)
  
  ```
  power inline {auto | never}
  ```

- **Configuring default allocation**:
  
  **Command**: `set inlinepower defaultallocation <value>`
  
  **Successful output**: Default Inline Power allocation per port: 10.0 Watts (0.24 Amps @42V)
  
  **Failure output**: Default port inline power should be in the range of 2000..12500 (mW)
Verifying Inline Power Status

Command: show port inlinepower <mod>|<mod/port>
Output:

Default Inline Power allocation per port: 12.500 Watts (0.29 Amps @42V)
Total inline power drawn by module 7:  37.80 Watts (0.90 Amps @42V)
module 5:  37.80 Watts ( 0.90

<table>
<thead>
<tr>
<th>Port</th>
<th>InlinePowered</th>
<th>PowerAllocated</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Admin Oper</td>
<td>Detected mWatt</td>
</tr>
<tr>
<td>7/1</td>
<td>auto off</td>
<td>no</td>
</tr>
<tr>
<td>7/2</td>
<td>auto on yes</td>
<td>12600</td>
</tr>
<tr>
<td>7/3</td>
<td>auto faulty yes</td>
<td>12600</td>
</tr>
<tr>
<td>7/4</td>
<td>auto deny yes</td>
<td>0</td>
</tr>
<tr>
<td>7/5</td>
<td>on deny yes</td>
<td>0</td>
</tr>
<tr>
<td>7/6</td>
<td>on off no</td>
<td>0</td>
</tr>
<tr>
<td>7/7</td>
<td>off off no</td>
<td>0</td>
</tr>
</tbody>
</table>

Inline Power Syslog Messages

Not enough power available:
%SYS-3-PORT_NOPowerAvail:Device on port 5/12 will remain unpowered

Link did not come up after powering on the port:
%SYS-3-PORT_DEVICENOLINK:Device on port 5/26 powered but no link up

Faulty port power:
%SYS-6-PORT_INLINEPWRFLTY:Port 5/7 reporting inline power as faulty
Power Supply Characteristics

- 2500W power supply delivers 55A from a 220V source
  Needs a 20A circuit
  Power cord for 220VAC in the U.S. market is a NEMA 6-20 or NEMA L6-20 which has a twist lock
- 1300W power supply delivers 27A from a 110V source
  Needs a 20A circuit
  Power cord for 110VAC in the U.S. market is a NEMA 5-20P
- 1050W power supply delivers 20A from a 110V source
  Needs 15A circuit
  Power cord for 110VAC in the U.S. market is a NEMA 5-15P

Catalyst 6000 Power Reference Chart

<table>
<thead>
<tr>
<th>Product</th>
<th>Power Required</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sup1A + PFC + 2GBIC's</td>
<td>2.5A</td>
</tr>
<tr>
<td>48-Port 10/100 Ethernet</td>
<td>2.7A</td>
</tr>
<tr>
<td>Cisco IP Phone</td>
<td>0.15A</td>
</tr>
<tr>
<td>Fully Loaded Inline Power Module w/48 Phones</td>
<td>10A</td>
</tr>
<tr>
<td>8-Port GE (w/GBIC’s)</td>
<td>2A</td>
</tr>
<tr>
<td>16-Port GE (MT-RJ)</td>
<td>2.5A</td>
</tr>
<tr>
<td>16-Port GE (w/GBIC’s)</td>
<td>2.8A</td>
</tr>
<tr>
<td>24-Port 100FX (MT-RJ)</td>
<td>2A</td>
</tr>
<tr>
<td>24-Port Analog FXS Gateway</td>
<td>1.6A</td>
</tr>
<tr>
<td>8-Port T1/E1 Gateway OR DSP Farm</td>
<td>2A</td>
</tr>
<tr>
<td>Multi-layer Switched Feature Card (MSFC)</td>
<td>0.8A</td>
</tr>
</tbody>
</table>

6K Chassis Will Always Reserve Power for Redundant Supervisor; Unless Some other Card Is Plugged in Slot 2
### No. of Phones ↔ Power Supply

<table>
<thead>
<tr>
<th>Single PS or 2 PS’s Configured in Redundant Mode</th>
<th>2 PS’s Configured in Non-Redundant Mode</th>
<th>Total No. of Phones Supported</th>
<th>Total No. of Phones Supported</th>
</tr>
</thead>
<tbody>
<tr>
<td>2500W</td>
<td>2500W</td>
<td>240 (5 Inline Power Cards with Phones)</td>
<td>384 (8 Inline Power Cards with Phones)</td>
</tr>
<tr>
<td>1300W</td>
<td>1300W</td>
<td>96 (2 Inline Power Cards with Phones)</td>
<td>230 (4–5 Inline Power Cards with Phones)</td>
</tr>
<tr>
<td>1050W</td>
<td>1050W</td>
<td>60 (1–2 Inline Power Cards with Phones)</td>
<td>160 (3–4 Inline Power Cards with Phones)</td>
</tr>
</tbody>
</table>

Number of Phones Shown in Table above Assumes no other Linecards; if other Cards Are Added, Adjust the Numbers Based on the Power Requirement of the Card as Shown Earlier in Reference Chart.

### IP Addressing
IP Address Plan

• Each IP phone needs an address
  Configure phones statically or use DHCP

• Addressing options:
  Double current address space
  Phones on separate subnets
  Phones in a different address space
  (Real addresses or private addresses)
**Recommendations**

- Add IP phones with DHCP as the mechanism for getting addressees
- Keep IP phones on a separate subnet; i.e. voice subnet is different from data subnet
- If subnets are available in existing address space then use them for IP phones
- If not, then use new address space (real or private address space)
- LAN and private IP WAN will carry these routes and route between both the address spaces

**Catalyst Auxiliary VLAN (Plug and Play Implementation)**

This Feature Provides Automatic Phone VLAN Configuration

- Phone VLAN = 200
- PC VLAN = 3
- IP Phone: IP Subnet B
- Desktop PC: IP Subnet A
- No end-user intervention required
- Provides the benefits of VLAN technology for the phone
- Preserves existing IP address structure
- Uses 802.1Q technology between switch and phone
A Multi-VLAN Access Port

• An access port able to handle 2 VLANs
• Native VLAN (PVID) and auxiliary VLAN (VVID)
• Hardware set to dot1q trunk

Configuring Auxiliary VLAN on Catalyst Switches with CatOS

Any Switch Running CatOS 5.5 or Higher
Catalyst 2948G, 4000, 5000 and 6000

Console> (enable) set port auxiliaryvlan help
Usage: set port auxiliaryvlan <mod/port>
   <vlan|untagged|dot1p|none>
   (vlan = 1..1000)

Console> (enable) set port auxiliaryvlan 2/1-3 222
Auxiliaryvlan 222 configuration successful.
AuxiliaryVlan AuxVlanStatus Mod/Ports
------------------- -------------------
222            active            1/2,2/1-3

For IOS-Based Switches {12.0(5) XU} (Interface Command)
Catalyst 2900XL and 3500XL
switchport voicevlan <vlan-id>
**Auxiliary VLAN Status**

```
Console> show port auxiliaryvlan 222
 AuxiliaryVlan AuxVlanStatus Mod/Ports
----------------- ------------- ----------------------
 222      active      1/2,2/1-3
```

```
Console> (enable)

Console> show port 2/1
...
Port AuxiliaryVlan AuxVlan-Status
----- ------------- --------------
 2/1    222           active
```

```
Console> (enable)
```

---

**Catalyst Switch <-> Phone Interaction**

1. Phone Discovery
2. Provide Power
3. CDP

- Unpowered phone plugs into powered linecard port
- Port senses the device using phone discovery mechanism and reports it to the supervisor
- Supervisor checks power budget, allocates default amount and informs port to apply power
- Port turns on power to the phone and reports linkup to supervisor, once the PHY on the phone is enabled
- If phone was powered by external patch panel or wall power, switch port will report linkup to supervisor
- Phone begins CDP exchange with the switch and gets its VLAN ID (VVID) as well as reports actual power needed for operation
- Phone will now send a DHCP request on that VLAN for an IP address
Phone’s Actions on Startup

1. Get IP address, mask, DNS, etc.
   - Static or DHCP
2. Get TFTP server address
   - Static address
   - Option 150 (single IP address)
   - Option 66 (first IP address or DNS name)
   - Look up CiscoCM1.your.domain
3. Get configuration from CallManager TFTP*
   - List of up to three CallManagers
   - Region info and keyboard template
   - Version of code to run
4. Get new code (one time only)
5. Register with CallManager

*Use Configuration in Flash after Timeout

Quality of Service
Domains of QoS Consideration

Session Focus

Avoiding Loss, Delay, and Delay Variation (Jitter)

WAN

Campus Classification Queuing

WAN Edge Classification Queuing

WAN Backbone

Voice-Enabled Infrastructure

Need for Campus QoS

Speed Mismatch

Many to One

Aggregation

10 Mbps

1000 Mbps

Switching Fabric

Packets that Made through; Rest Are Dropped

Link Utilization 60%

Example: 100 Mbps Link

Any of the Above Scenarios Could Result in Packet Loss and/or Delay Due to Re-Transmission

Delay-Sensitive Applications Like Voice Cannot Tolerate this

Packets from Different Applications
Campus QoS

- Two imp steps to campus QoS implementation
  - **Classification**—Marking the packet with a specific priority
  - **Queuing**—Assigning packets to one of multiple queues (based on classification) for expedited treatment through the network

Solutions for Campus QoS Issues

- **Classification**
  - CoS/ToS/DSCP
- **Queuing and scheduling**
  - Priority queuing
  - WRR
- **Policing** (optional)
Classify at Layer 3 or Layer 2

- **Standard IPV4:** Three MSB Called IP Precedence (DiffServ May Use Six D.S. Bits Plus Two for Flow Control)
- **Layer 2**
  - ISL
  - 802.1Q/p
- **Layer 3**
  - IPV4

<table>
<thead>
<tr>
<th>Field</th>
<th>Length</th>
</tr>
</thead>
<tbody>
<tr>
<td>Version</td>
<td>1 Byte</td>
</tr>
<tr>
<td>ToS</td>
<td>1 Byte</td>
</tr>
<tr>
<td>Len</td>
<td>1 Byte</td>
</tr>
<tr>
<td>ID</td>
<td>1 Byte</td>
</tr>
<tr>
<td>Offset</td>
<td>1 Byte</td>
</tr>
<tr>
<td>TTL</td>
<td>1 Byte</td>
</tr>
<tr>
<td>Proto</td>
<td>1 Byte</td>
</tr>
<tr>
<td>FCS</td>
<td>4 Bytes</td>
</tr>
<tr>
<td>IP-SA</td>
<td>4 Bytes</td>
</tr>
<tr>
<td>IP-DA</td>
<td>4 Bytes</td>
</tr>
<tr>
<td>Data</td>
<td>Encapsulated Frame 1...24.5 KBytes</td>
</tr>
<tr>
<td>FCS</td>
<td>4 Bytes</td>
</tr>
</tbody>
</table>

- **Three Bits Used for CoS:**
  - Layer 2 = CoS (User Priority)
  - Layer 3 = ToS

What Happens Inside the Phone

- IP phone sends voice packets (RTP stream) marked at CoS/ToS value 5
- PC may or may not send a CoS
- Phone can manipulate PC CoS
- Capabilities of switch will determine what can be achieved
Configuring Catalyst Switch for Extended QoS

Catalyst 6000

Console> (enable) set port qos help
Usage:

set port qos <mod/port> trust-ext <untrusted/trust-cos>

set port qos <mod/port> cos-ext <cos-value>

PC Is Not Trusted
Normal Mode (Default)

e.g. set port qos 2/1 trust-ext untrusted

Phone Sets PC CoS to Zero

Untrusted Phone ASIC Will Re-Write CoS = 0
PC Is Trusted

Phone Does Not Change PC CoS

e.g. set port qos 2/1 trust-ext trust-cos

COS = 5

COS = 2

COS = 7

PC Is Not Trusted But Needs Better than CoS = 0

Phone Sets PC CoS to a Specific Value

e.g. set port qos 2/1 cos-ext 2

COS = 5

COS = 7

PC is Untrusted; Phone ASIC Will Re-Write CoS Based on Switch Configuration
e.g. Extended COS = 2
Output Scheduling

Switch Fabric

- Low
- Medium
- High
- Priority

WRR Queue Scheduler

Priority Queue
- Packets Are Always Serviced First
- Remaining Queues Use WRR

WRR Works Between Queues for Expedited Treatment of Packets Based on Servicing Ratio

Output Port

Campus QoS—An Example

Areas Where QoS May Be a Concern

Use DSCP Upstream

IP Phone
- Voice: CoS = 5, IP Prec = 5, DSCP = EF
- PC: Reclassify CoS = 0

1/ Access Layer (L2)
1) CoS used as entrance Criteria to PQ
2) Where support exists
   - Map CoS to DSCP
   - Map CoS properties to VLAN

2/ Distribution Layer
1) Map CoS to DSCP
   - Map CoS properties to VLAN
2) Map IP Addr to DSCP
3) MAP L4 to DSCP
**Branch QoS—An Example**

Areas Where QoS May Be a Concern

1/ Access Layer (L2)
- CoS used as entrance criteria to PQ
- Where support exists
  - Map CoS to DSCP
  - Map QoS properties to VLAN

2/ WAN layer
- Map IP Addr to DSCP
- Map L4 to DSCP

Reference Commands

- Use CoS and/or ToS values to make intelligent drops via WRED
  
  E.g. set qos drop-threshold 1p1q4t rx queue 1 20 40 75 100
  
  Receive drop thresholds for queue 1 set at 20% 40% 75% 100%

- Switch can use CoS/ToS and packet characteristics (L3, L4 information) to re-classify if necessary at L3 or L2
  
  E.g. set qos acl TEST ip dscp 56 any any

- Use CoS/ToS to schedule packets in and out of the switch fabric via WRR and/or priority queue
  
  E.g. set qos wrr 2q2t 30 70

- If switch has enhanced QoS capabilities like policing, use it to limit bursty data traffic
  
  E.g. set qos policer microflow TEST rate 1000 burst 10000 policed-dscp
Recommendations for Campus QoS

- Classify and/or re-classify packets as close to the edge as possible
- Determine your trust boundary
- Make access layer as the trust boundary
- If wiring closet switch does not have appropriate QoS hooks, shrink trust boundary to distribution layer, no further
- Use appropriate QoS mechanisms available based on platform selection
- For very small offices with low user count and no QoS capable switch, WAN edge can be the trust boundary

Don’t Forget WAN QoS (Covered in Detail Session 2006)

- Prioritization
  Classification and queuing
- Slow link efficiency
  Link Fragmentation and Interleave (LFI)
  Compression (cRTP)
  Voice Activity Detection (VAD)
- Traffic shaping
  Speed mismatches
CallManager Primary Functions
(Call Processing)

Connectivity, Signaling and Device Control (IP Phones, Network Gateways, Etc.)
Operation, Administration, Maintenance and Provisioning (OAM&P)

Functions (PBX and More...)
- Call setup/teardown, supervision
- IP phone auto registration
- Dial plan implementation, routing
- Call detail logging
- Support for applications (Voice mail, IVR, Call Center...)

Protocols:
- Cisco skinny station (client/server)
- H.323 (peer-to-peer)
- PBX protocols (CAS, CCS...)
- PRI signaling
- MGCP
- Future standard protocols

Support for:
- Media Termination Protocol (MTP – add in SW for supplementary services
- Conference bridge SW
- Other applications (unified messaging, TAPI, JTAPI)
- Message waiting indicator

Communicates with:
- IP phones
- Soft phones
- Other CallManagers
- Integrated PBXs
- Gateways
- Unified messaging
- Other applications
Making a Call
IP Telephone to IP Telephone

1. Off-Hook and Digit Stimulus
2. Play Tone Commands
3. Ring Command
4. Off-Hook Stimulus
5. Setup Media Stream Command
6. Audio Stream Established

TCP Signaling (Port 2000)

CallManager Interface

Cisco CallManager 3.0 Administration

Cisco Systems, Inc.
16593 North Dallas Parkway
Suite: 200
Dallas, Texas 75247

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Configuration Example
Adding a New IP Phone

Add a New Phone Template

Phone Template: New
Status: Ready
Select a template below and click Copy to create a new template based on the selected template:
- Default 12 SP+
- Default 30 SP+
- Default 30 VIP
- Default 7600

OS:
Select a phone model below and click Continue to create a new button layout:
- 12S
- 12SP
- 12SP+ 
- 30SP+
- 30VIP
- IP Phone 7600

Continue

Configuration Example
Phone Parameters

Phone Configuration

Phone: 680000000000 (Cisco 12SP+)
Device Information
- TAC Address: 1234567
- Description: Office Phone
- Device Pool: Default
- Location: 123
- Calling Search Space: Restricted
- Button Template: Default

Load Information
- Language: English

DTMF Capabilities for this phone:
- Send In-band DTMF
- Send Out-of-band DTMF

* indicates a required item.
Configuration Example
Changing Line Parameters

### Line 2 for SEP0010CBOO8F2 ( Lester Bloggs)

<table>
<thead>
<tr>
<th>Status</th>
<th>Update</th>
<th>Update and Close</th>
<th>Delete</th>
<th>Restart Devices</th>
<th>Cancel</th>
</tr>
</thead>
</table>

#### Directory Number
- **Directory Number**: 4463

#### Directory Number Settings
- **Call Waiting**: Default
- **Calling Search Space**: Unrestricted

#### Call Forward and Pickup Settings
- **Forward All**: None
- **Forward Busy**: 4111
- **Forward No Answer**: 4111
- **Call Pickup Group**: None

#### Line Settings for this Phone
- **Display**: 4463
- **Label**: 4463
- **External Phone Number Mask**: Included

*indicates required item; changes to Line or Directory Number settings require restart.

---

**CallManager Groups**

- **Primary**
- **Secondary**
- **Last Resort**

Each Device (IP Phone and Skinny Gateway) Has a Prioritized List of Up to 3 Call Managers to which It Can Connect. This is Called a “CallManager Group”

This List is Downloaded During Device Initialization.
Adding New CallManager to the Group

CallManager Group Configuration

Cisco CallManager Group: Default

- Name:
- Copy:
- Update:
- Delete:
- Default/Devices:
- Cancel:

Available Cisco CallManagers
Selected Cisco CallManagers
(ordered by highest priority)

Server Information:
- Cisco CallManager Name:
- Description:
- Ethernet Phone Port:
- Digital Port:
- Analog Port:
- MGCP Listen Port:
- MGCP Keep-alive Port:

Auto-registration Information:
- Starting Directory Number:
- Ending Directory Number:
- Partition:
- External Phone Number Mask:

Notes:
- * Indicates required field.
Regions

Regions are defined for putting devices which have similar codec characteristics. This also allows for defining inter-region codec to be used.

Device Pools

Device pools are used to define 3 attributes:

- Region (a.k.a. Codec)
- Time Zone
- CallManager Redundancy Group
### CallManager Device List View

<table>
<thead>
<tr>
<th>Device Type</th>
<th>Load Information</th>
<th>Device FQDN</th>
<th>Phone Template</th>
</tr>
</thead>
<tbody>
<tr>
<td>12 S</td>
<td>Default</td>
<td>Default</td>
<td>Default 12 S+</td>
</tr>
<tr>
<td>12 SP+</td>
<td>Default</td>
<td>Default</td>
<td>Default 12 SP+</td>
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<tr>
<td>12 SP</td>
<td>Default</td>
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<td>3D SP+</td>
<td>Default</td>
<td>Default</td>
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<td>3D VIP</td>
<td>Default</td>
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### CallManager Cluster

**CallManager Cluster Sizing**

- 6 CallManagers maximum in a cluster
- Cluster(s) cannot extend across WAN
- 2500 users maximum per CallManager
- Maximum of 10,000 users in a cluster
- Provision for CallManager failure in a cluster

**CallManager Cluster IP Phone Provisioning**

*Planning Assumes for Failure of One CallManager at a Time*

<table>
<thead>
<tr>
<th>CMs in Cluster</th>
<th>Max users per cluster</th>
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<tr>
<td>1</td>
<td>2500</td>
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<tr>
<td>2</td>
<td>2500</td>
</tr>
<tr>
<td>3</td>
<td>5000</td>
</tr>
<tr>
<td>4</td>
<td>7500</td>
</tr>
<tr>
<td>5</td>
<td>10,000</td>
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CallManager Clusters
N+1 Failure Recovery Scenario

CallManager Cluster Appears as “One” CallManager

N+1 Failure Recovery Scenario

CallManager Clustering and Design Principles Are Discussed at Length in the Session:
Advanced Enterprise IP Telephony Design and Implementation #2006

Cluster Recommendations
Up to 2,500 Users

A cluster of two CallManagers
Single active CallManager
Dedicated publisher also acts as a standby
What Is a Gateway?

Gateways Come in Many Form Factors and Port Densities

Gateways Provide PSTN Access and PBX Interconnectivity

Making a Call
IP Phone to H.323 Gateway

1. Off-Hook and Digit Stimulus
2. Play Tone Commands
3. H.323 Setup
4. H.323 Connect
5. Setup Media Stream Command
6. Audio Stream Established

Off-Hook and Digit Stimulus
Play Tone Commands
H.323 Setup
H.323 Connect
Setup Media Stream Command
Audio Stream Established
Cisco Voice Gateways

Integrated Digital and Analog Gateways for Catalyst 6000 Family Switches

- Standalone Digital Gateways DT-24+, DE-30+
- Standalone Analog Gateways AS and AT
- Cisco 1750
- Cisco 2600
- Cisco 3600
- Cisco 3800
- Cisco 4010
- Cisco AS5800
- Cisco AS5300
- Cisco MC3810
- Cisco 7200
- VG-200 Standalone Analog and Digital Gateway (Modular, IOS-Based)

Gateway Selection Criteria

- Standalone vs. integrated router/gateway
  Cost vs. flexibility, functionality and manageability
- Required voice port density
- Support for required PSTN signaling types
- Gateway protocol
  - SCCP (skinny gateway) DT24+, DT 30+, Catalyst 6000 blades
  - H.323 (IOS-based gateways)
  - 1700/2600/3600/3800/AS5X00/7200
  - MGCP-based gateway VG-200 (it can be used as H.323 gateway also)
- Support for required WAN interface(s) and QoS
- Remote Sites Likely to Add Voice Ports to Existing Voice Enabled Router
PSTN/PBX Signaling Support

**T1/E1–CAS, PRI**
- Cisco 1750/2600/3600
- DT-24/30+ (Standalone)
- Cisco AS5300
- Catalyst 6000 (Integrated)
- Cisco 7200/7500

**E1 R2**
- Cisco AS5300 Only

**Analog FXO or FXS**
- Cisco 1750, 2600, 3600
- AT/AS, VG200 (Standalone)
- Catalyst 6000 (FXS Only)

**BRI or Analog E&M**
- Cisco 1750, 2600, 2600
- VG200 (Standalone)

Most Common Gateway Choices

**Branch Office**
- Cisco 1750
- Cisco 2600
- Cisco 3620/3640

**HQ/Large Facility**
- Cisco 3640/3660
- Cisco 7200/7500
- Catalyst 6000
- Integrated Gateway
Gateway Configuration Example

Click Device on the Main Configuration Screen
Select Gateway Under Device Menu

Configuration Example Gateways

Existing Gateways
Add New Gateway

Find and List Gateways
6 matching record(s) for Device Name begins with **

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<tr>
<th>Device Name</th>
<th>Description</th>
<th>Device Pool</th>
<th>Delete Reset</th>
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Configuration Example for MGCP Gateway (VG-200)

MGCP Configuration for VG-200 Gateway

MGCP Configuration

MGCP Member Configuration

Configuration Example for VG-200 Gateway
VG-200 Configuration
(Cisco IOS Portion)

mgcp
mgcp call-agent 172.20.71.30
mgcp sdp simple

ccm-manager switchback immediate
ccm-manager redundant-host 172.20.71.26
172.20.71.47
ccm-manager mgcp

interface FastEthernet0/0
ip address 172.20.71.33 255.255.255.0
no ip directed-broadcast
duplex auto
speed auto

ip default-gateway 172.20.71.1
ip classless
ip route 0.0.0.0 0.0.0.0 FastEthernet0/0
no ip http server

voice-port 1/1/0

voice-port 1/1/1

dial-peer voice 1 pots
destination-pattern 30301
port 1/0/0

dial-peer voice 2 pots
destination-pattern 30301
port 1/0/1

dial-peer voice 4 pots
destination-pattern 30301
port 1/1/1

dial-peer voice 3 pots
application MGCPAPP

voice-port 1/1/0

! voice-port 1/1/1

Dial Plan 101
What Is a Dial Plan?

Simply Stated, Dial Plan is a Numbering Scheme Created for Reachability between Devices

Example: User A (1111) dials 2222 to reach User B
CallManager should know where to send this call

Example: Same User dials 408-5551212
CallManager should know that this call has to go over PSTN

Example: Same user dials 5333 to reach User C in a remote office
Again CallManager should know that this call should be attempted over the IP WAN

This is a very basic scenario. Dial plans are a very important part of the overall design. They can get very complex in a multi-site scenario.
Example Route Pattern

Route Pattern Points to Route Lists
Voice Mail

Voice Messaging Call Flow

1. After 3 Rings, CallManager sends RTP stream to VM (in real time).
2. VM system performs directory query for called user.
3. SMTP message to Message Store (not in real time).
4. VM system sends message waiting indicator (MWI) “ON” to CallManager to illuminate MWI Light.

Incoming Call
IP Phones
Call Manager
Router/GW
PSTN
IP WAN
LDAP Directory
Message Store
Voice Mail

Cisco.com
Typical Deployment Scalable to Thousands of Users

- Message Stores: Scale as Needed (Approx. 750KB/min of VM)
- VM Servers: Scale as Needed (500 Users Per VM Server)
- CallManager Cluster: Scale as Needed (Up to 5 CMs and 10,000 Users)
- IP Phones

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

- Digital PIC Integration (Most Common Type of Octel to PBX Integration)
- Verify with Vendor for SMDI Support:
  - Octel 250/350/200/300 Certified with CallManager
  - VM System
  - SMDI
  - Analog Trunks
  - Gateway

- CallManager
- CallManager-Based Network

In Many Cases Must Convert Octel to SMDI with Analog Trunks
“Shrink + Grow” to CallManager
Keep Legacy VM

Prerequisites:
- VM System Must Be Able to Talk to Two Systems Simultaneously Via SMDI
- Must Be “Supported” by Legacy VM Vendor
- May Require PBX/VM Conversion to SMDI

More Information on PBX VM ↔ IP Telephony Is Available in Session #2100

Deployment Models
Campus Site Considerations

- Redundancy *
- Scalability
- Power
- IP Addressing
- QoS
- Gateway Selection
- Dial Plan

* Based on Facility Requirements

Deployment Examples

Individual Locations Still Have Advantages of IP Telephony
Converged Network Application Enablement
Lower Cost of Ownership
What Have We Discussed so Far… (Session Focus—IP Telephony)

- How to build an infrastructure that will support voice and data
- How to add devices to the infrastructure (PCs, IP phones, etc.)
- How to power the IP phones
- How to provide IP addresses for phones
- Why is QoS needed in campus
- How do we QoS enable the network
- What is a CallManager and how to configure it
- How does a basic call flow take place
- What is a gateway and how to configure it
- What is a dial plan
- How to implement voice messaging solutions
- Deployment examples
Recommendations

• Configure LAN switch with redundant sup and PS
• Choose appropriate connectivity option
  E.g. daisy-chained PC
• Select powering option
  Inline power, external power
• Check switch port for auto-negotiate
  10/100 ports; NA for 10M switch ports
• Check PC NIC for auto-negotiate
  10/100 NICs
• Configure VVID on the switch
  Set auxiliary vlan

Recommendations (Cont.)

• Plan for IP addressing
  Keep separate subnet for phones
  Implement DHCP
• Set appropriate QoS settings
  Classification and queuing
• Turn Portfast ON, EtherChannel OFF, DTP OFF
• Enable QoS
• Implement power back-up strategy
  UPS, generator, etc.
Enterprise IP Telephony Design and Configuration

Session 2102

Please Complete Your Evaluation Form

Session 2102
Empowering the Internet Generation℠