CCIE Voice

TECCCIE-3002

Tectorial Agenda

Session 1	CCIE® Program Overview
Session 2	CCIE Voice Overview
Session 3	Campus Infrastructure and Network Services
Session 4	Cisco Unified Communications Manager
Session 5	Cisco Unified Communications Manager Express
Session 6	Voice Gateways and Protocols
Session 7	Dial Plan Considerations
Session 8	High Availability
Session 9	Media Resources
Session 10	QoS
Session 11	Unified Contact Center Express and B-ACD for CUCME
Session 12	Cisco Unity Connection and Cisco Unity Express
Session 13	Cisco Unified Presence
Session 14	Preparation Tips and Test-Taking Strategies/Q&A

Disclaimer

- Not all the topics discussed today appear on every exam
- For time reasons, we're unable to discuss every feature and topic possible on every exam; rather, we will try to cover the most important ones

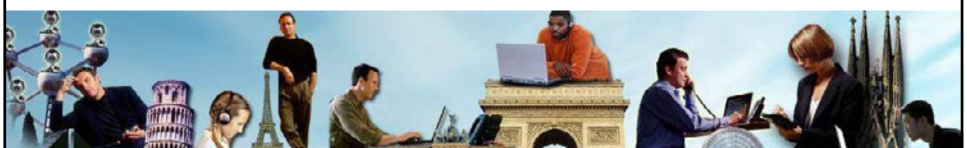
Session 1

CCIE Program Overview

CCIE Certification

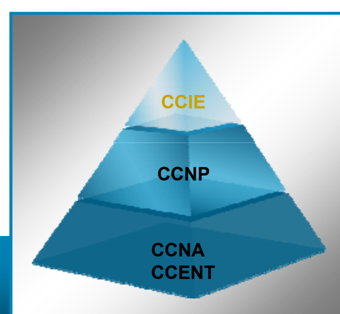
- Most highly respected IT certification for more than 15 years
- Industry standard for validating expert skills and experience
- More than 20,000 CCIEs worldwide—less than 3% of all professionals certified by Cisco
- Demonstrate strong commitment and investment to networking career, life-long learning, and dedication to remaining an active CCIE

CISCO CERTIFIED CCIE



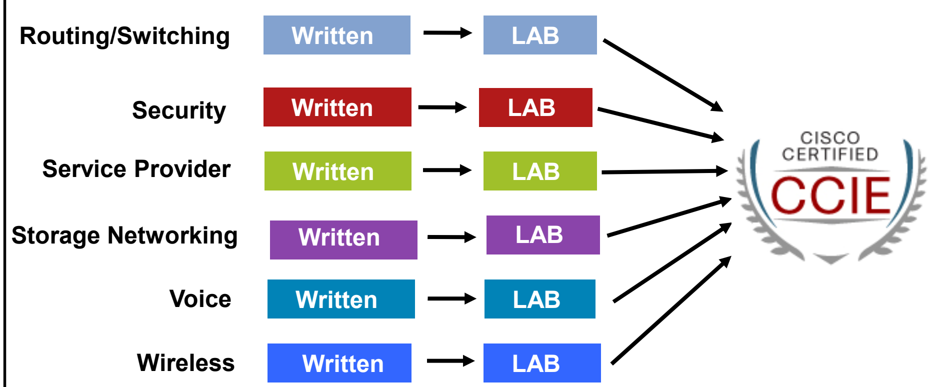
Cisco CCIE Certification

- **CCIE R&S:** Configure and troubleshoot complex converged IP networks
- **CCIE Security:** Configure complex, end-to-end secure networks, troubleshoot environments, and anticipate and respond to network attacks
- **CCIE Service Provider:** Configure and troubleshoot advanced technologies to support service provider networks
- **CCIE Storage:** Configure and troubleshoot storage area networks over a variety of interfaces
- **CCIE Voice:** Configure complex, end-to-end unified communications systems; troubleshoot and resolve VoIP-related problems
- **CCIE Wireless:** Plan, design, implement, operate, and troubleshoot wireless network and mobility infrastructure



www.cisco.com/go/learnnetspace

CCIE Tracks and Exams



CCIE Tracks Facts

Routing and Switching
- Core networking cert
- 64% of all bookings
- Labs in all regions, all worldwide locations

Security
- Introduced 2002
- 13% of bookings
- Labs in Beijing, Hong Kong, Brussels, RTP, San Jose, Sydney, Dubai, Bangalore and Tokyo

Voice
- Introduced 2003
- 16% of bookings
- Labs in Brussels, RTP, San Jose, Sydney and Tokyo
- Coming to Bangalore, Dubai and China

Storage Networking
- Introduced 2004
- 1% of bookings
- Labs in Brussels and RTP

Service Provider Networks
- Introduced 2002
- 6% of bookings
- Labs in Brussels, Beijing, Hong Kong, RTP, Sao Paulo, Sydney

Wireless
- Introduced 2009
- Labs in Brussels and San Jose

Available in Six Technical Specialties

CCIE Information Worldwide

Total of Worldwide CCIEs:	19,134*
Total of Routing and Switching CCIEs:	16,727*
Total of Security CCIEs:	2,147*
Total of Service Provider CCIEs:	1,182*
Total of Storage Networking CCIEs:	140*
Total of Voice CCIEs:	996**

*Updated 29-Feb-2009

** Updated 29-May-2009

Multiple Certifications

Many CCIEs Have Gone on to Pass the Certification Exams In Additional Tracks, Becoming a "Multiple CCIE." Below Are Selected Statistics on CCIEs Who Are Certified in More Than One Track

Total with Multiple Certifications Worldwide:	1,974
Total of Routing and Switching and Security CCIEs:	739
Total of Routing and Switching and Service Provider CCIEs:	496
Total of Routing and Switching and Storage Networking CCIEs:	35
Total of Routing and Switching and Voice CCIEs:	258
Total with 3 or More Certifications	316

http://www.cisco.com/web/learning/le3/ccie/certified_ccies/worldwide.html

Step 1: CCIE Written Exams

- Available worldwide at Prometric and VUE for ~$300 USD, adjusted for exchange rate and local taxes where applicable
- Two-hour exam with 100 multiple-choice questions
- Closed book; no outside reference materials allowed
- Pass/fail results are available immediately following the exam; the passing score is set by statistical analysis and is subject to periodic change
- Waiting period of five calendar days to retake the exam
- Candidates who pass a CCIE written exam must wait a minimum of six months before taking the same number exam
- From passing written "Must" take first lab exam attempt within 18 months
- No "skip-question" functionality

Step 2: CCIE Lab Exams

- Available in select Cisco locations for $1,400 USD, adjusted for exchange rates and local taxes where applicable, not including travel and lodging
- Eight-hour exam requires working configurations and troubleshooting to demonstrate expertise
- Cisco documentation available via Cisco Web; no personal materials allowed in lab
- Minimum score of 80% to pass
- Scores can be viewed normally online within 48 hours and failing score reports indicate areas where additional study may be useful

Session 2

CCIE Voice Overview

CCIE Voice Overview

- CCIE Voice certification recognizes experts with the highest level of technical knowledge and hands-on experience in building, configuring, and troubleshooting a Cisco Unified Communications solution
- CCIE Voice exams covers the technologies and applications that are commonly deployed in Cisco Unified Communications networks
- Introduced in September 2003
- ~1,000 in the world

CCIE Voice Written Exam Blueprint

Major Topics Covered:

- Cisco Unified Communications Manager
- Cisco Cisco Unified Communications Manager Express and Cisco Unity Express
- Telephony and VoIP Protocols (Analog and Digital, H.323, MGCP, SCCP, SIP, etc.)
- High availability considerations
- QoS
- Operations and Network Management
- Security
- Video
- Cisco Unified Contact Center Express
- Infrastructure protocols

Detailed CCIE Voice Written Blueprint Is Posted on the CCIE Webpage

http://www.cisco.com/web/learning/le3/ccie/voice/wr_exam_blueprint.html

CCIE Voice Lab Exam Overview

- An 8-hour, hands-on, 100-point lab exam; candidates must score 80 or above to pass
- Candidate builds, troubleshoots, and optimizes a voice network to supplied specifications on a provided Voice equipment rack
- Physical cabling is done. IP routing protocol (OSPF), and WAN (Frame Relay) are preconfigured
- Unified Communications applications are installed, with some pre-configuration of basic tasks, such as device registration and baseline application integrations**

** new in v3.0 lab blueprint, effective starting July 16th, 2009.

CCIE Voice Lab Blueprint v3.0 (I)

Implement and Troubleshoot:

- Campus Infrastructure and Services
- CUCM and CUCME Endpoints
- Voice Gateways
- Call Routing Policies
- High Availability Features
- Media Resources
- QoS and Call Admission Control

Effective starting July 16th, 2009

CCIE Voice Lab Blueprint v3.0 (II)

Implement and Troubleshoot:

- QoS and Call Admission Control
- Supplementary Services
- Other CUCM Voice Applications
- Cisco Unified Contact Center Express
- Voicemail Messaging
- Cisco Unified Presence

Effective starting July 16th, 2009

Detailed CCIE Voice Lab v3.0 Blueprint Is on the CLN web site:

https://cisco.hosted.jivesoftware.com/docs/DOC-3569

CCIE Voice Lab v3.0 Equipment

- Cisco MCS-7845 Media Convergence Servers
- Cisco 3825 Series Integrated Services Routers (ISR)
- Cisco 2821 Series Integrated Services Routers (ISR)
- ISR Modules and Interface Cards
  - VWIC2-1MFT-T1/E1
  - PVDM2
  - HWIC-4ESW-POE
  - NME-CUE
- Cisco Catalyst 3750 Series Switches
- IP Phones (7965) and Soft Clients

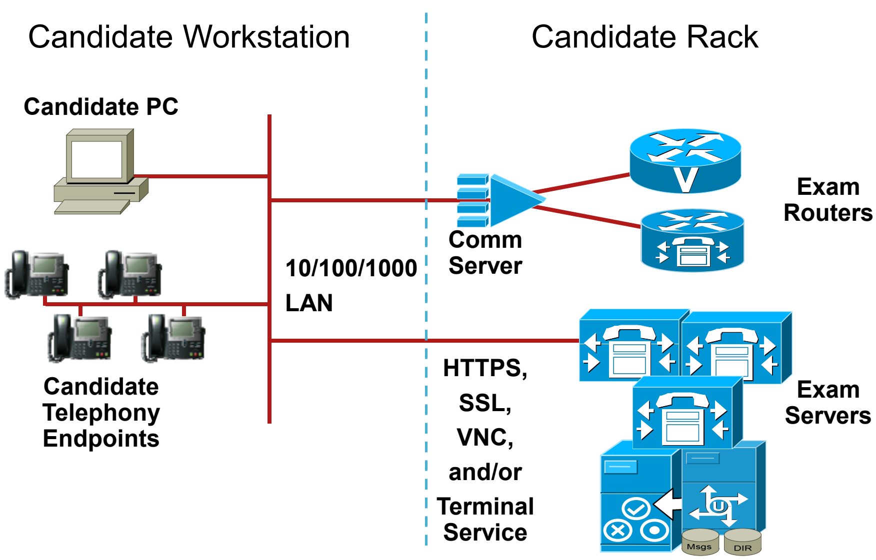
Effective starting July 16th, 2009

CCIE Voice Lab v3.0 Software

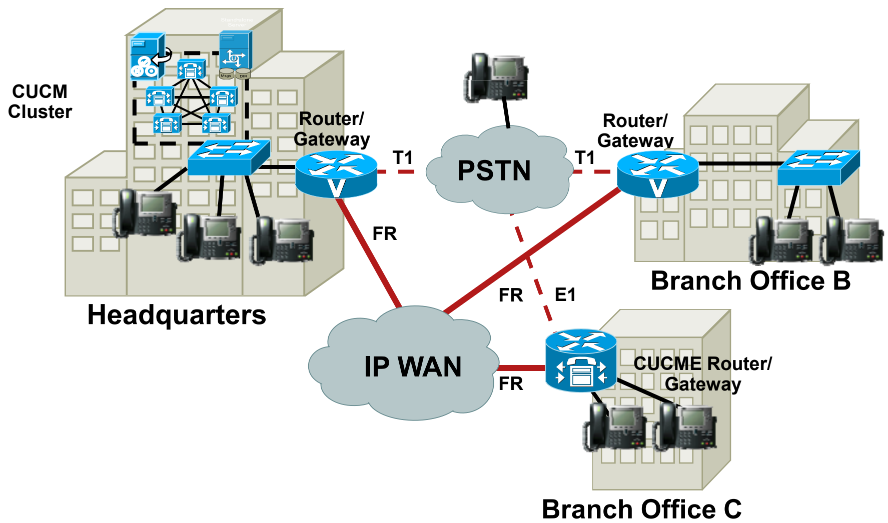
- Cisco Unified Communications Manager 7.0
- Cisco Unified Communications Manager Express 7.0
- Cisco Unified Contact Center Express 7.0
- Cisco Unified Presence 7.0
- Cisco Unity Connection 7.0
- All routers use IOS version 12.4T Train.
- Cisco Catalyst 3750 Series Switches uses 12.2 Main Train

Effective starting July 16th, 2009

CCIE Voice Lab Rack Access



CCIE Voice Lab Sample Topology



Core Knowledge Section – Overview

- Cisco CCIE team has implemented a new type of question format to the CCIE Voice Lab exams: Core Knowledge Section a.k.a. Interview Section.
- In addition to the live configuration scenarios, candidates will be asked a series of open-ended short-answer questions, covered from the lab exam blueprint.
- No new topics are being added.
- The new short-answer questions will be randomly selected for each candidate every day

Core Knowledge Section – Why

Why are you adding short-answer questions to the CCIE Lab Exam?

- One of the primary goals to introduce the new Core Knowledge Section is maintain exam security and integrity and ensure only qualified candidates achieve certification.
- The questions will be designed to validate concepts, theory, architecture and fundamental knowledge of products & protocols.

Core Knowledge Section – Format

- Candidates will be asked four open-ended questions, computer-delivered, drawn from a pool of questions based on the material covered on the lab exam blueprint.
- Core Knowledge section format will not be multiple-choice type questions.
- Candidates will be required to type out their answers, which typically require five words or less.
- No changes are being made to the lab exam blueprint or to the length of the lab exam.

Core Knowledge Section – Time

- Candidates are allowed a maximum of 30 minutes to complete the questions. The 30 minutes is inclusive in the total length of the lab exam.
- The total length of the CCIE lab exam will remain eight hours.
- Candidates cannot use Cisco Documentation.
- Well-prepared candidates should be able to answer the questions in 15 minutes or less and move immediately to the configuration section.

Core Knowledge Section – Scoring

- The Core Knowledge section is scored Pass/Fail and every candidate will be required to pass in order to achieve CCIE certification.
- A candidate must answer at least three of the four short-answer questions correctly to Pass the Core Knowledge section, which will be indicated with a 100% mark on the score report.
- If a candidate answers fewer than three correctly, the Core Knowledge section will be marked 0%, indicating a Fail. A 0% does not necessarily indicate the candidate answered all the questions incorrectly and the zero does not get averaged into the overall score.

Core Knowledge Section – Example

- Question:

  Which protocol is used to by Cisco switch to inform Cisco ip phones the appropriate VLAN ID to tag voice traffic?

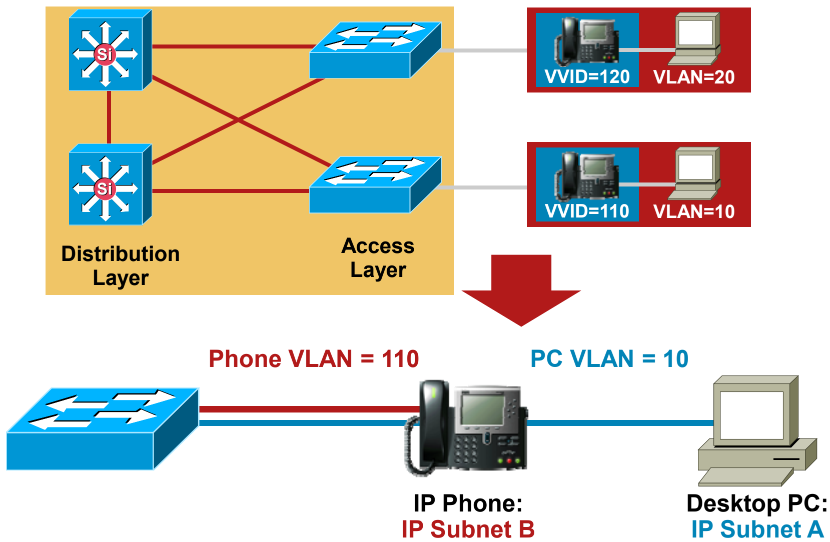
- Answer:

  CDP or Cisco Discovery Protocol

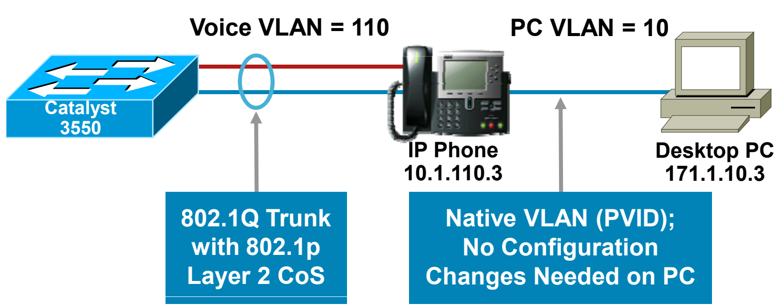
Session 3

Campus Infrastructure
and Services

Campus VLAN Design



Voice VLAN Configuration




```
interface FastEthernet0/1
 no ip address
 switchport access vlan 10
 switchport voice vlan 110
 spanning-tree portfast
```

Unified Communications Infrastructure

Network Services: IP Phone Bootup Process

1. Inline Power (ILP)

   Inline Power Initialization

2. Cisco Discovery Protocol (CDP) or Link Layer Discovery Protocol-Media Endpoint Discovery (LLDP-MED)

   ILP Negotiation, Voice VLAN ID

3. Dynamic Host Configuration Protocol (DHCP)

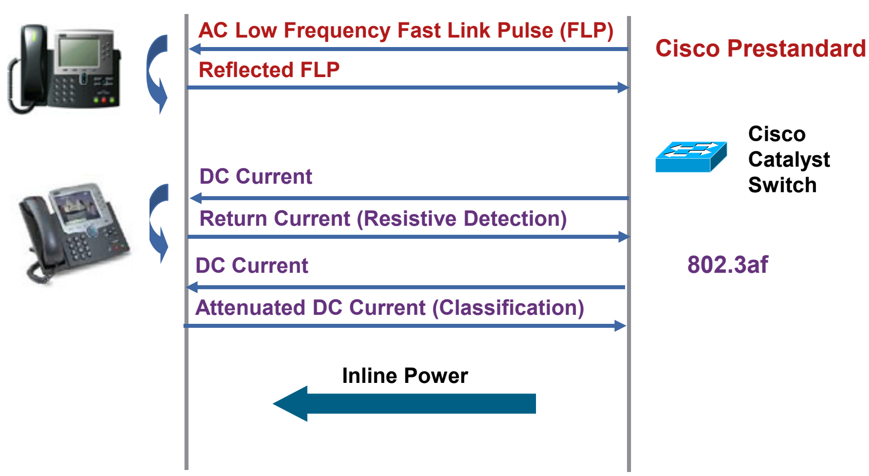
   IP Assignment, TFTP Server Allocation, DNS (optional)

4. Trivial File Transfer Protocol (TFTP)

   Configuration File, IP Phone Firmware

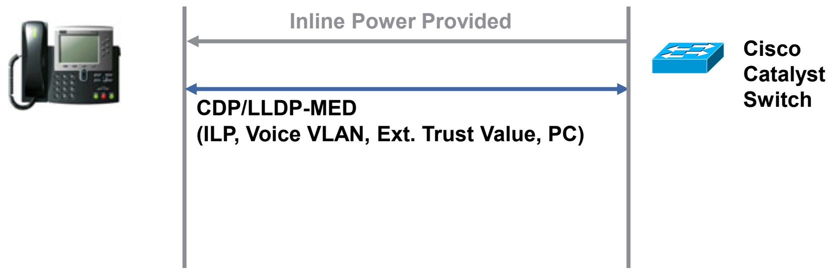
Unified Communications Infrastructure

Network Services: Inline Power



On Phone: Mute, Headset, Speaker Buttons Are Illuminated

Unified Communications Infrastructure Network Services: CDP or LLDP-MED



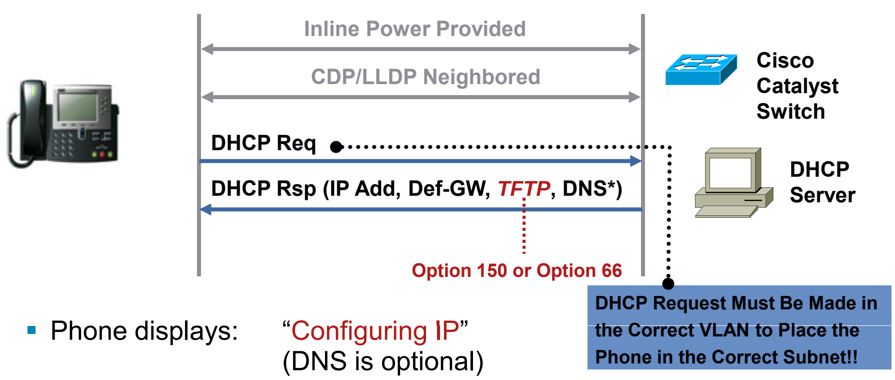
- Phone displays: “Configuring VLAN”
- Phone settings: Settings=>NetCfg=>“Operational VLAN ID”

LLDP-MED is supported as of IP Phone Firmware 8.3(3)

LLDP-MED and CDP White Paper:
http://www.cisco.com/en/US/technologies/tk652/tk701/technologies_white_paper0900aecd804cd46d.html

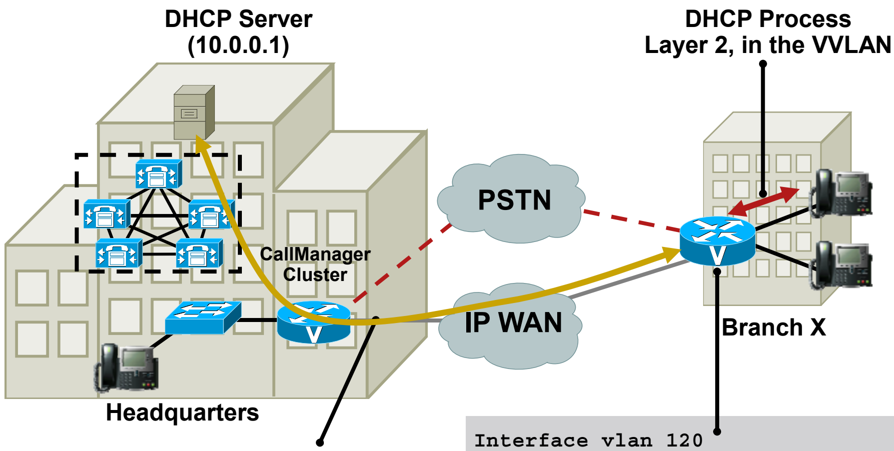
Unified Communications Infrastructure

Network Services: DHCP



- Phone displays: “Configuring IP”
  (DNS is optional)
- Phone settings: Settings=>NetCfg=>“DHCP Server”
  Settings=>NetCfg=>“IP Address”
  Settings=>NetCfg=>“TFTP Server X”

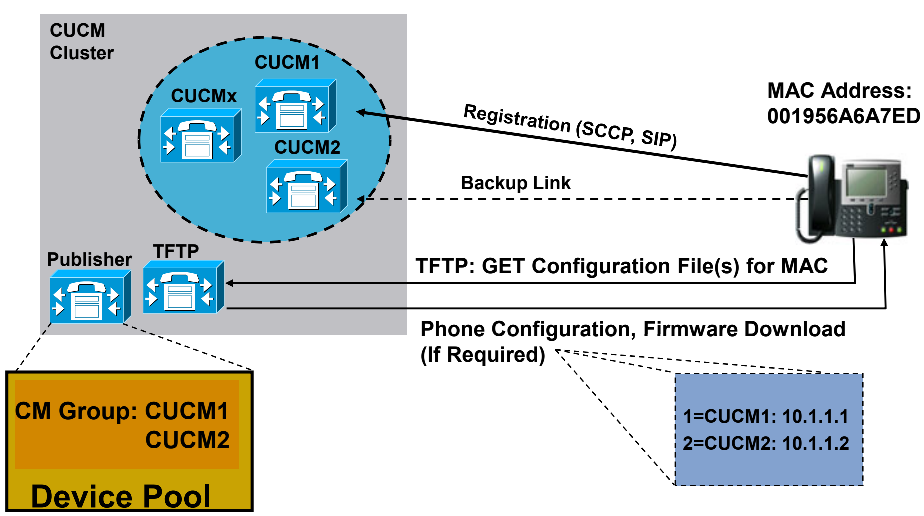
Unified Communications Infrastructure
Network Services: DHCP



- DHCP process tunneled at Layer 3
- DHCP relay agent (IP helper-address)
- Identification of scope relies on router ID (typically the default gateway's IP address)

```
Interface vlan 120
ip address 120.1.1.1 255.255.255.0
ip helper-address 10.0.0.1
```

Unified Communications Infrastructure
Network Services: TFTP



Summary:
Infrastructure and Network Services

Be Familiar with the Following:

- Voice and data vlan configuration
- CUCM DHCP server and its options
- Cisco IOS DHCP server and its options
- DHCP relay configuration on routers

Q and A

Session 4

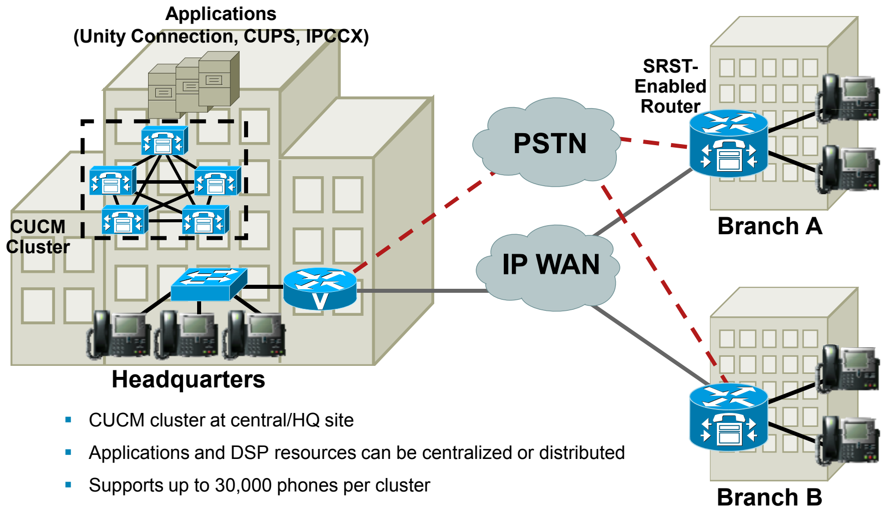
CUCM Fundamentals

Cisco Unified Communications Manager (CUCM) Fundamentals

- CUCM deployment models
  - Centralized
  - Distributed
- CUCM scalability and redundancy
  - CUCM clustering
  - Database vs. run-time call processing data
  - Clustering examples

Deployment Models

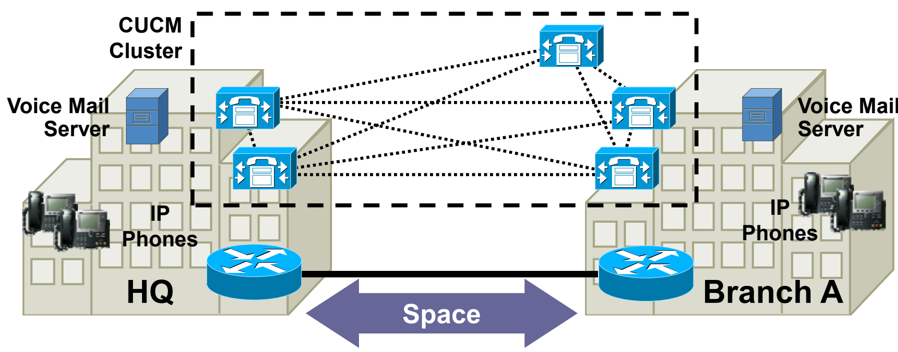
Centralized Call Processing



- CUCM cluster at central/HQ site
- Applications and DSP resources can be centralized or distributed
- Supports up to 30,000 phones per cluster
- Survivable remote site telephony for remote branches
- Maximum 1000 branches per cluster (500 branches before CUCM 6.X)

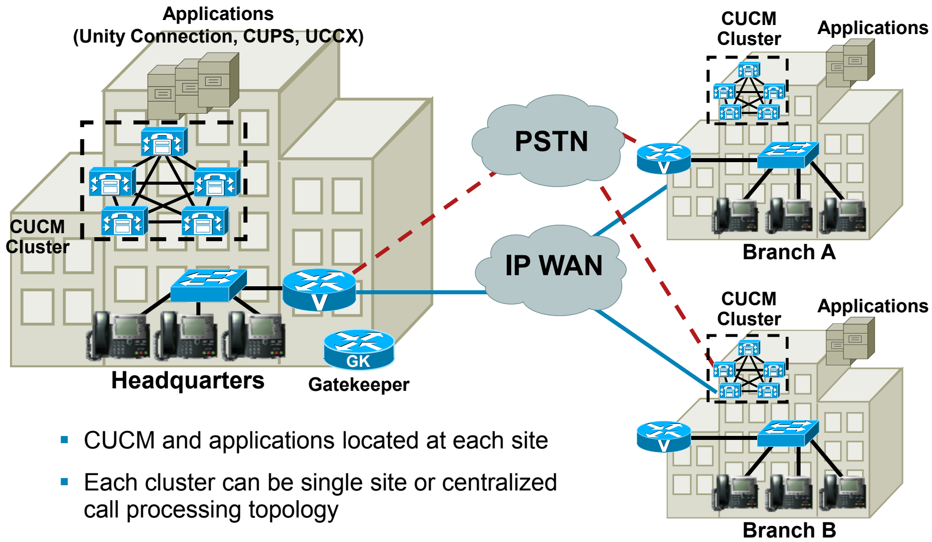
Deployment Models

Clustering Over the WAN (COW)



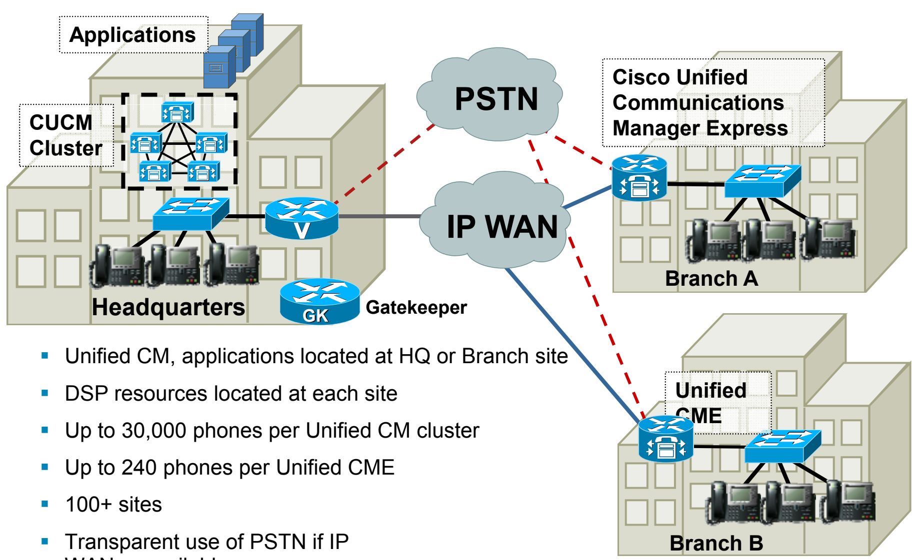
- CUCM servers in a cluster separated by WAN for spatial redundancy
- Max 40-ms round-trip delay between any two CUCM across the WAN
- The minimum recommended bandwidth between sites that are clustered over the WAN is 1.544 Mbps
- Additional bandwidth required for database repair or reset

Deployment Models
Distributed Call Processing



- CUCM and applications located at each site
- Each cluster can be single site or centralized call processing topology

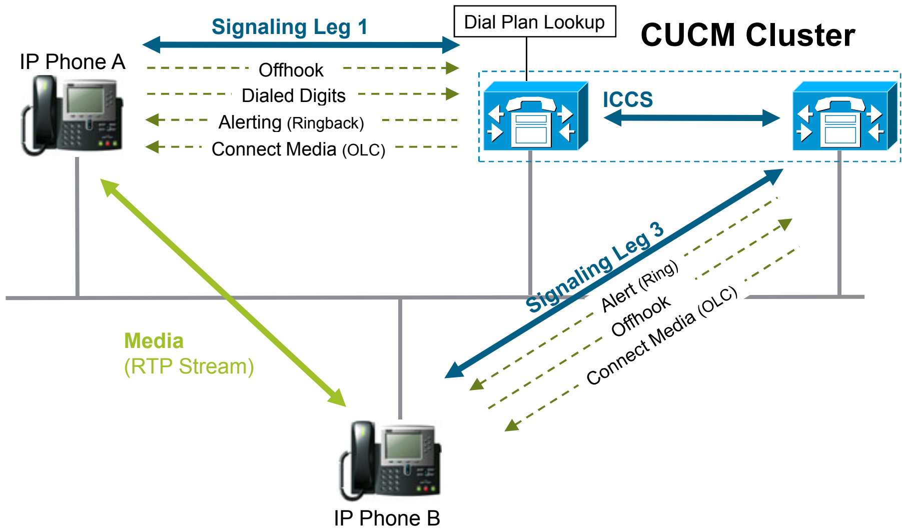
Deployment Models
Distributed Call Processing (Unified CM-Unified CME Model)



- Unified CM, applications located at HQ or Branch site
- DSP resources located at each site
- Up to 30,000 phones per Unified CM cluster
- Up to 240 phones per Unified CME
- 100+ sites
- Transparent use of PSTN if IP WAN unavailable

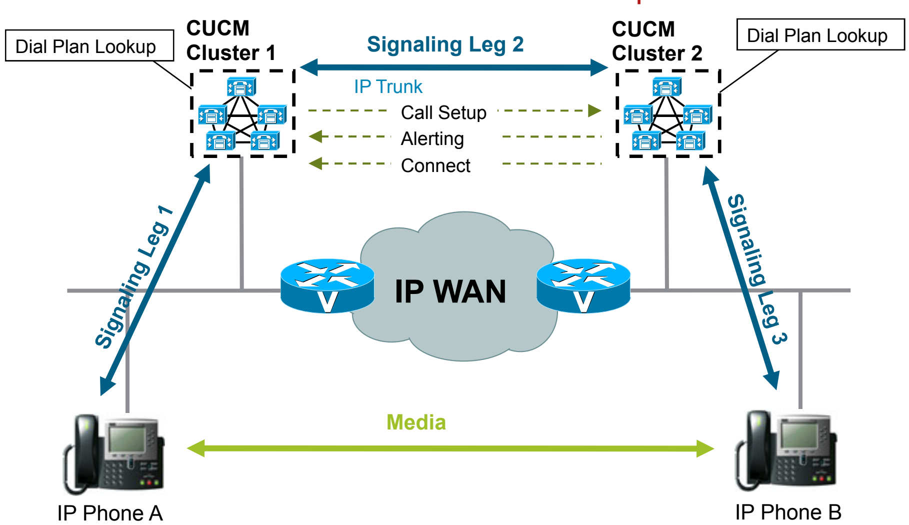
Example Call Signaling Flow (I)

Intra-Cluster IP Phone to IP Phone Example



Example Call Signaling Flow (II)

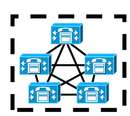
Inter-Cluster IP Phone to IP Phone Example



Cisco CallManager Scalability and Redundancy

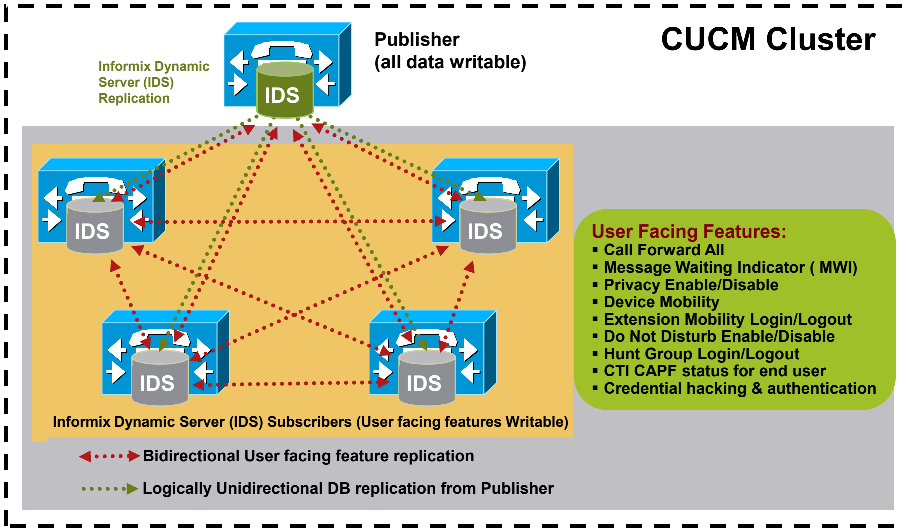
CUCM Cluster Facts

- The cluster appears as one entity, with a single point of administration (the publisher)

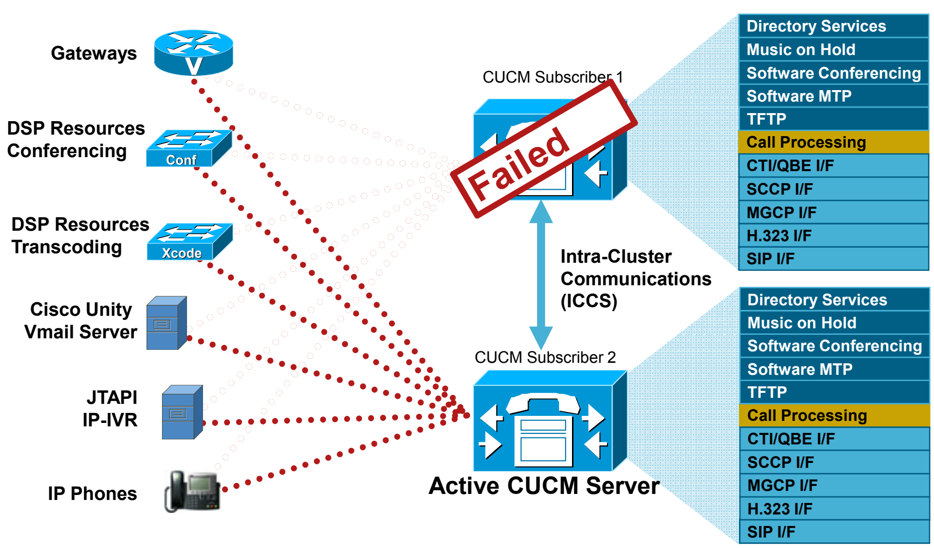


- Several functions can be collocated on the same server, depending on cluster size and server type
- Maximum of 19 subscribers per cluster (20 servers in a cluster including the publisher)
- Maximum of eight call processing servers per cluster
- Maximum of 7500 IP Phones per Cisco Unified CM server (server platform dependant)
- Maximum of 30,000 IP Phones per Cisco Unified CM cluster (server platform and configuration dependant)

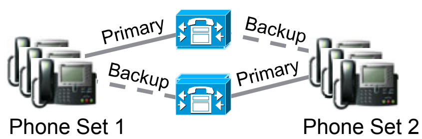
CUCM Database Resiliency:
(6.x and Onwards)



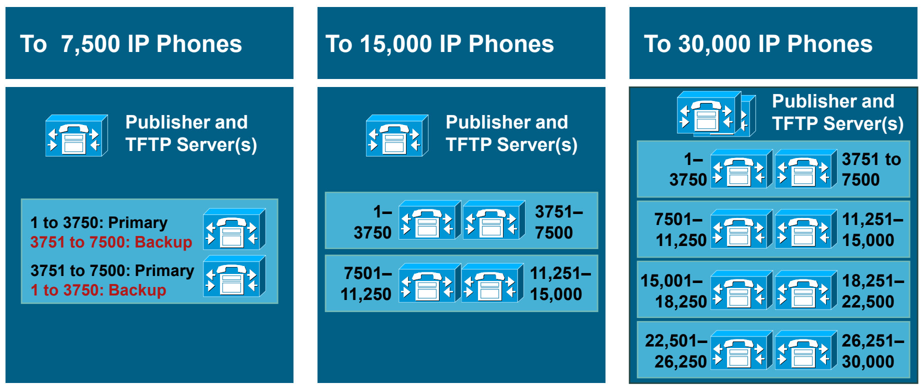
CUCM Server Failover and Redundancy



CUCM Server Failover and Redundancy



- MCS 7845 supports 7500 phones/server
- Load-share between primary and backup servers



Summary:

CUCM Fundamentals

Be Familiar with the Following

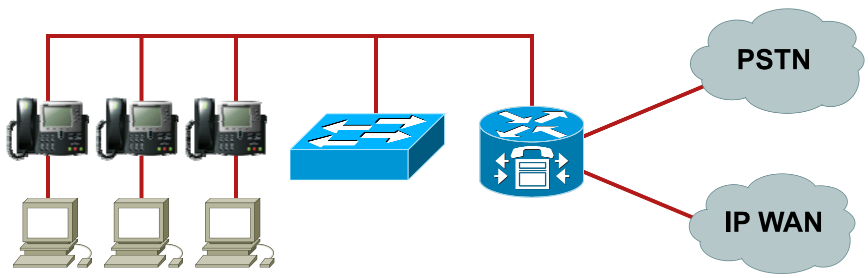
- Centralized vs. distributed call processing
- Difference between publisher/subscriber server vs. primary/secondary call processing server
- Cisco CallManager group
- Device pool
- Auto-registration
- IP phone configuration fields

Q and A

Session 5

Cisco Unified Communications Manager Express

CUCME: Cisco Unified Communications Manager Express



- Cisco Unified Communications Manager in an Cisco IOS router
- Router provides call processing to SCCP or SIP endpoints
- Same router also serves as an PSTN gateway; terminating IP packet voice to TDM voice and vice versa

Basic CUCME Configurations (SCCP)

```
telephony-service
  ip source-address 10.1.1.1 port 2000
  max-dn 48
  max-ephone 24
  create-cnf files

ephone-dn 1
  number 2001

ephone 4
  mac-address 1111.2222.3333
  button 1:1
```

Global command to enter CUCME system configuration mode

Mandatory command to enable router to receive and process SCCP messages

Mandatory commands which define the max. # of IP phones and directory numbers (DNs) supported by CUCME; Default is “0”

Mandatory command which builds the XML config files for the CUCME IP phones

Creates an instance of a phone line with a directory number of 2001

Creates an instance of an IP phone, specifying the MAC address and mapping an directory number to its first line button

Additional CUCME Concepts (SCCP)

- Call presentation
- Call distribution
- Configurable softkeys

CUCME SCCP Call Presentation

- Key switch: one call per line/button (default)
  - No call-waiting for second call on same line
- PBX style: two calls per line/button
  - Call-waiting for second incoming call
  - Place outgoing consultation call during call transfer
- Octo-line: eight call per line/button
  - Similar to CUCM IP phones
  - Up to 8 active calls (incoming + outgoing) per button
  - Octo-line DN can split its channels among the phones sharing the DN
  - Additional use-case for octo-line DNs: to facilitate 8-participants CUCME hardware conferences

CUCME SCCP Call Presentation Configuration

One Call per Line/Button

```
ephone-dn 10
   number 2001
ephone 1
   mac-address 1111.2222.3333
   button 1:10
```

Two Calls per Line/Button

```
ephone-dn 10 dual-line
   number 2001
ephone 1
   mac-address 1111.2222.3333
   button 1:10
```

Eight Calls per Line/Button

```
ephone-dn 10 octo-line
   number 2001
ephone 1
   mac-address 1111.2222.3333
   button 1:10
```

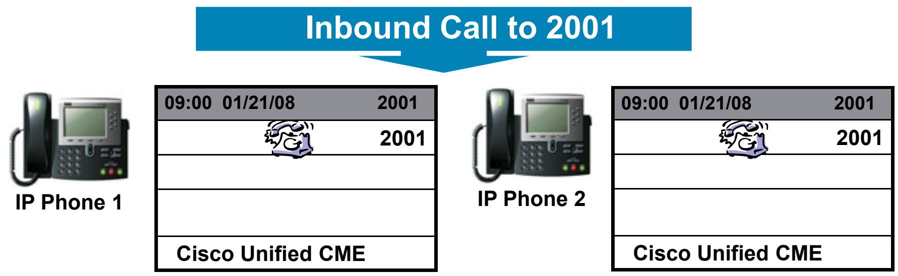
CUCME SCCP Call Distribution

Multiple Ways to Route and Hunt Calls

- Parallel call distribution using shared lines
- Sequential call distribution using call forward on busy/no-answer
- Sequential call distribution of the same DN using preference, huntstop, huntstop channel CLIs
- ephone-hunt
- Overlay option in the ephone “button” command
- Combination of the above

CUCME SCCP Call Distribution:
Shared Lines

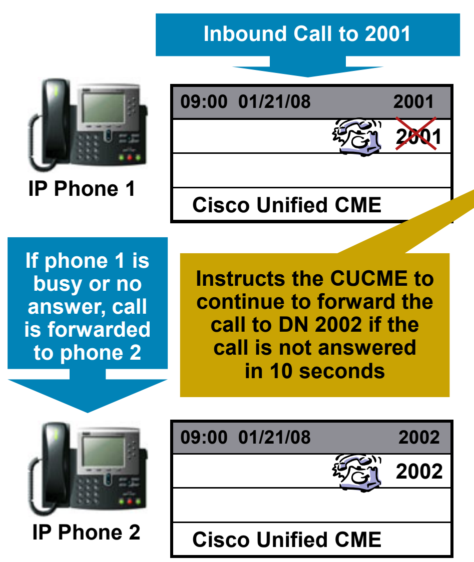
- In parallel" call distribution to multiple phones at same time




```
ephone-dn 10
   number 2001
ephone 1
  mac-address 1111.1111.1111
  button 1:10
```

```
ephone-dn 10
   number 2001
ephone 2
  mac-address 2222.2222.2222
  button 1:10
```

CUCME SCCP Call Distribution:
Sequential Different DNs Using Call Forward



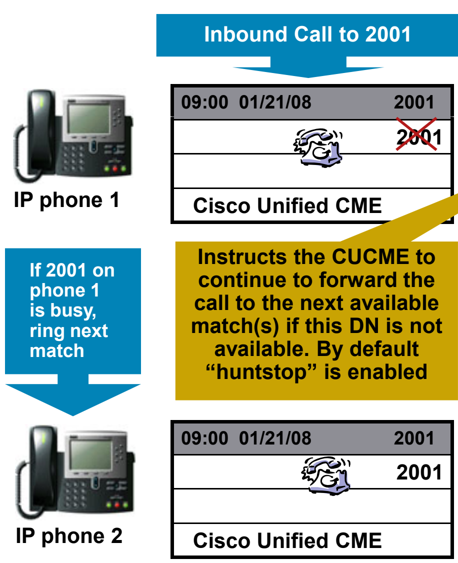

```
ephone-dn 10
   number 2001
   call-forward busy 2002
   Call-forward noans 2002
   timeout 10
ephone 1
  mac-address 1111.1111.1111
  button 1:10
```

```
ephone-dn 11
   number 2002
   call-forward busy 2003
   Call-forward noans 2003
   timeout 10
ephone 2
  mac-address 2222.2222.2222
  button 1:11
```

CUCME SCCP Call Distribution:
Sequential Same DN

- Create multiple ephone-dn entries with the same DN number and assign to different phones
- Control sequential hunt order using
  - preference
  - [no] huntstop
  - huntstop channel
- Only one phone rings at a time

CUCME SCCP Call Distribution:
Sequential Same DN



Preference 0 is the highest priority and the default value, it does not appear in configuration

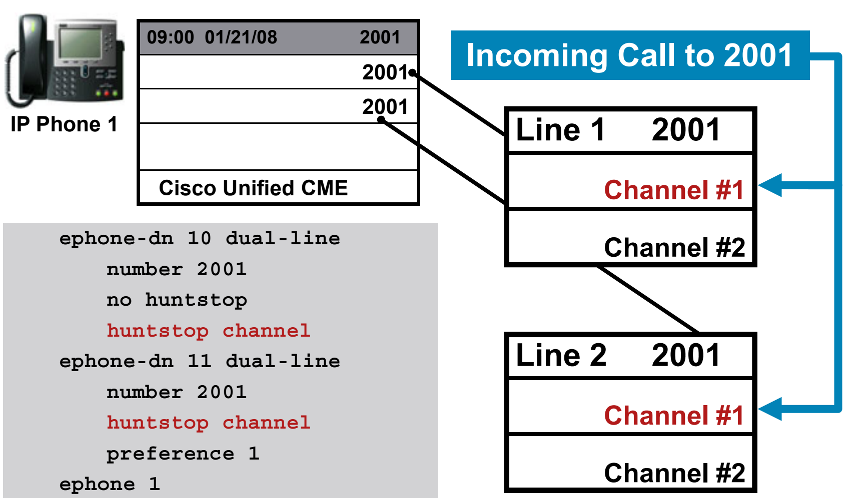
```
ephone-dn 10
    number 2001
    no huntstop
    preference 0
ephone 1
   mac-address 1111.1111.1111
   button 1:10
```

```
ephone-dn 11
    number 2001
    preference 1
ephone 2
   mac-address 2222.2222.2222
   button 1:11
```

CUCME SCCP Dual-line Huntstop Channel

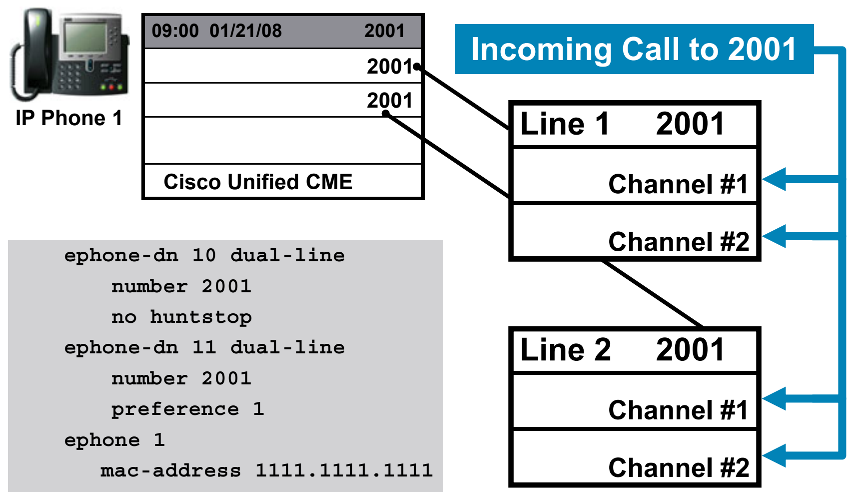
- Prevents incoming calls from hunting into the second channel of a dual-line DN
- Effectively disables call-waiting on a dual-line DN
- Reserves the second channel of a line for outgoing calls such as transfer and conference

CUCME SCCP Dual-line with Huntstop Channel




```
ephone-dn 10 dual-line
    number 2001
    no huntstop
    huntstop channel
ephone-dn 11 dual-line
    number 2001
    huntstop channel
    preference 1
ephone 1
   mac-address 1111.1111.1111
   button 1:10 2:11
```

CUCME SCCP Dual-Line Without Huntstop Channel




```
ephone-dn 10 dual-line
    number 2001
    no huntstop
ephone-dn 11 dual-line
    number 2001
    preference 1
ephone 1
   mac-address 1111.1111.1111
   button 1:10 2:11
```

CUCME SCCP Octo-line Hunting CLI

- huntstop channel: (configured under ephone-dn)

```
CUCME(config)#ephone-dn 12 ?
  dual-line  dual-line DN (2 calls per line/button)
  octo-line  octo-line DN (8 calls per line/button)
  <cr>

CUCME(config-ephone-dn)#huntstop channel ?
  <1-8>  Channel number of an octo-line dn call hunting stops at
```

- busy-trigger-per-button: (configured under ephone or ephone-template)

```
CUCME(config-ephone)#busy-trigger-per-button ?
  <1-8>  The number of calls that triggers call forward busy per octo-
line button
```

- max-calls-per-button: (configured under ephone or ephone-template)

```
CUCME(config-ephone)#max-calls-per-button ?
  <1-8>  Maximum number of calls supported per octo-line button
```

SCCP Octo-line Busy-trigger-per-button CLI

- Limits number of INCOMING calls on a phone
- Sets the maximum number of calls allowed on this phone's octo-line directory numbers before triggering Call Forward Busy or a busy tone.
- Configurable under ephone or ephone-template

```
ephone-dn 1 octo-line
 number 2001
  huntstop channel 4
!
ephone 1
 busy-trigger-per-button 2
 button 1:1
```

Octo-line channel hunting stops at channel 4: maximum 4 inbound calls for this octo-line ephone-dn

If ephone 1 has two existing calls on button 1, additional incoming calls will receive a user busy or be forwarded to CFB destination, if configured

Question: Under this configuration, can ephone 1 make any outbound calls by putting the existing calls on hold?

SCCP Max-calls-per-button CLI

- Sets the maximum number of calls, incoming and outgoing, allowed on an octo-line directory number on this phone.
- Configurable under ephone or ephone-template
- Max-calls-per-button **> = busy-trigger-per-button**

```
ephone-dn 1 octo-line
 number 2001
 huntstop channel 5
!
ephone 1
 max-calls-per-button 3
 busy-trigger-per-button 3
 button 1:1
!
ephone 2
 max-calls-per-button 4
 busy-trigger-per-button 3
 button 1:1
```

Ephone 1 can make maximum of 3 calls, inbound and outbound, on button 1

Ephone 2 can make maximum of 4 calls, inbound and outbound, on button 1

Question#1: With this configuration, what is the maximum number of concurrent inbound calls to 2001 before user busy tone is returned?

Question#2: With this configuration, what is the maximum number of concurrent outbound calls between ephone 1 and ephone 2?

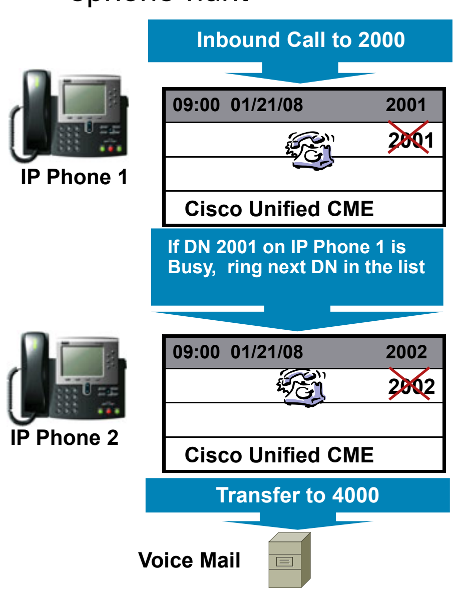
CUCME SCCP ephone-hunt

ephone-hunt Allows CUCME Administrators To:

- Define a pilot number for a hunt group
- Ring next DN in the hunt group if a DN did not answer or was busy
- Define a final destination to forward the call to if the call is not answered or all members are busy

CUCME SCCP Call Distribution:

ephone-hunt



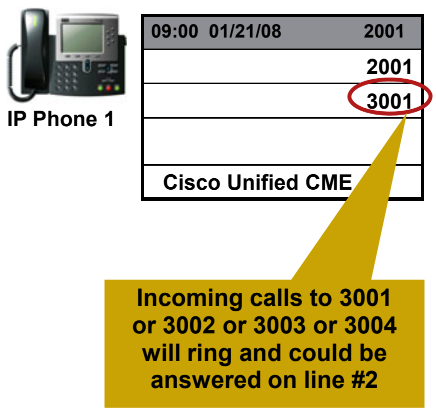

```
ephone-dn 10  dual-line
  number 2001
  huntstop  channel
ephone-dn 11  dual-line
  number 2002
  huntstop  channel
ephone 1
   mac-address 1111.1111.1111
   button 1:10
ephone 2
   mac-address 2222.2222.2222
   button 1:11
ephone-hunt 1 sequential
  pilot 2000
  list 2001, 2002
  final 4000
  timeout 10
```

Three types of Call Distribution:

- sequential
- peer
- longest-idle

CUCME SCCP Call Distribution:
Overlay

- Assign multiple ephone-dn to a single phone button




```
ephone-dn 1
  number 2001

ephone-dn 10
  number 3001

ephone-dn 11
  number 3002

ephone-dn 12
  number 3003

ephone-dn 13
  number 3004

ephone 1
  mac-address 1111.1111.1111
  button 1:1 2o10,11,12,13
```

- Octo-line DN cannot be overlaid

CUCME SCCP Configurable Softkeys

- Customizable softkey orders for various call states

```
CUCME(config)#ephone-template <tag>

CUCME(config-ephone-template)#softkey ?
  alerting       Softkey order for alerting (ring out) state
  connected      Softkey order for connected state
  hold           Softkey order for HOLD state
  idle           Softkey order for IDLE state
  remote-in-use  Softkey order for REMOTE-IN-USE state
  ringing        Softkey order for ringing state
  seized         Softkey order for seized state

CUCME(config-ephone-template)#softkeys ringing ?
  Answer   Answer
  Dnd      Do not Disturb
  HLog     HLog
```

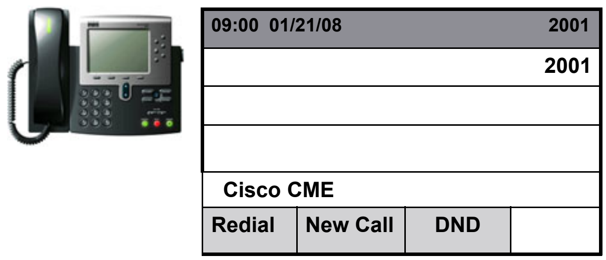
- Customized softkey templates are then applied to ephonesaa

```
ephone <tag>
  ephone-template <tag>
```

CUCME SCCP Configurable Softkeys

Example

```
CUCME#

ephone-template 1
  softkeys idle Redial Newcall Dnd

ephone 1
  ephone-template 1
  mac-address 1111.1111.1111
```



CUCME SCCP Verification CLI

```
CUCME#show ephone ?
  !!!!!!!!!!!!!!!!multiple ip phone models omitted!!!!!!!!!!!!!!!!!!!!
  H.H.H                       mac address
  anl                         SCCP Gateway (AN)
  ata                         ATA  phone emulation for analog phone
  attempted-registrations     Attempted ephone list
  bri                         SCCP Gateway (BR)
  cfa                         registered ephones with call-forward-all set
  dn                          Dn with tag assigned
  dnd                         registered ephones with do-not-disturb set
  login                       phone login status
  offhook                     Offhook phone status
  overlay                     registered ephones with overlay DNs
  phone-load                  Ephone phoneload information
  registered                  Registered ephone status
  remote                      non-local phones (with no arp entry)
  ringing                     Ringing phone status
  sockets                     Active ephone sockets
  summary                     Summary of all ephone
  tapiclients                 Ephone status of tapi client
  telephone-number            Telephone number assigned
  unregistered                Unregistered ephone status
  |                           Output modifiers
  <cr>
```

CUCME SCCP Debug/Troubleshooting CLI

```
CUCME#debug ephone ?
  after-hours          Enable ephone after-hours debugging
  alarm                Enable ephone alarm message debugging
  blf                  Enable ephone BLF debugging
  ccm-compatibility    Enable ephone ccm-compatibility debugging
  detail               Enable ephone detail debugging
  error                Enable ephone error debugging
  extension-assigner   Enable ephone extension assigner debugging
  hunt-stat            Enable hunt group statistics debugging
  hw-conference        Enable hardware conference debugging
  keepalive            Enable ephone keepalive debugging
  loopback             Enable ephone loopback debugging
  message              Enable ephone skinny message debugging
  moh                  Enable ephone music-on-hold debugging
  mtp                  Enable mtp debugging
  mwi                  Enable ephone mwi debugging
  pak                  Enable ephone packet debugging
  qov                  Enable ephone voice quality debugging
  raw                  Enable ephone raw protocol debugging
  register             Enable ephone registration debugging
  sccp-state           Enable trace of SCCP call state messages
  snmp                 Enable ephone snmp debugging
  socket               Enable ephone socket I/O debugging
  srtp                 Enable ephone srtp debugging
  state                Enable ephone state debugging
  statistics           Enable ephone statistics debugging
  video                Enable ephone video debugging
  vm-integration       Enable ephone vm-integration debugging
```

CUCME SIP Lineside Configuration

```
voice service voip
 allow-connections sip to sip
 sip
  registrar server expires max 1200 min 300
!
voice register global
 mode cme
 source-address 10.1.1.1 port 5060
 max-dn 20
 max-pool 2
 tftp-path flash:
 create profile
!
voice register dn  1
 number 4001
!
voice register dn  2
 number 4002
!
voice register pool  1
 id mac 1111.2222.3333
 type 7961
 number 1 dn 1
!
voice register pool  2
 id mac 2222.3333.4444
 type 7961
 number 1 dn 2
```

Global command to enter VoIP services configuration mode

Allow connection between SIP endpoints on the router

Mandatory command to enable router to receive and process incoming SIP registrar messages

Define global parameters for Cisco SIP phones. Equivalent to the "telephony-services" command for SCCP phones.

Creates instances of SIP Direcotry numbers on CUCME. Equivalent to the "ephone-dn" command for SCCP phones

Creates instancse of SIP IP phones, specifying the MAC addresses and mapping directory numbesr to each phone's first line button

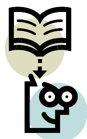
CUCME SIP Verification CLI

```
CUCME#sh voice register ?
  all              Show all pool details
  credential       Show voice register credential
  dial-peers       Show dial-peers created dynamically via REGISTERs
  dialplan         Show given dialplan details
  dn               Show given dn details
  global           Show voice register global
  pool             Show given pool details
  profile          Show text profile for ATA/7905/7912
  session-server   Show registered session servers
  statistics       Show voice register statistics
  template         Show given template details
  tftp-bind        Show voice register tftp-bind

CUCME#sh voice register dial-peers
 dial-peer voice 40001 voip
 destination-pattern 4001
 session target ipv4:10.1.1.1:5060
 session protocol sipv2
 digit collect kpml
 after-hours-exempt   FALSE
```

CUCME SIP Debug CLI

```
CUCME#debug voice register ?
  errors           voice-register errors
  events           voice-register events
  session-server   session-server debug

CUCME#debug ccsip ?
  all         Enable all SIP debugging traces
  calls       Enable CCSIP SPI calls debugging trace
  error       Enable SIP error debugging trace
  events      Enable SIP events debugging trace
  info        Enable SIP info debugging trace
  media       Enable SIP media debugging trace
  messages    Enable CCSIP SPI messages debugging trace
  preauth     Enable SIP preauth debugging traces
  states      Enable CCSIP SPI states debugging trace
  transport   Enable SIP transport debugging traces
```

Proctor Case Studies IV: CUCME #1

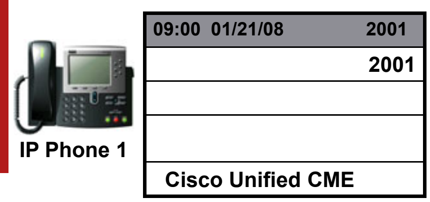


Lab Sample Question

Configure the CUCME router so that IP phone#1 will register with a Directory Number of 2001 on Line #1. Furthermore, call waiting should be enabled on this line.

Candidate's Problem Statement

"When I left the lab my CUCME phone #1 was up and I configured its first line to be 2001. I even verified that I could place call and receive calls, why did I receive no points in this section per the score report?"



Proctor Case Studies IV: CCME #1 (Cont)

Candidate's "sh ephone register" Output

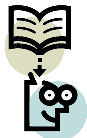
```
CCME#sh ephone register

ephone-1 Mac:000B.FDB8.21C2 TCP socket:[2] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:120.100.1.10 53102 Telecaster 7960  keepalive 7073 max_line 6
button 1: dn 10 number 2001 CH1 IDLE
```

Missing Channel 2 status: candidate did not configure a dual-line phone which enables call-waiting

It Should Have Looked Like This

```
CCME#sh ephone register

ephone-1 Mac:000B.FDB8.21C2 TCP socket:[2] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:120.100.1.10 53102 Telecaster 7960  keepalive 7073 max_line 6
button 1: dn 10 number 2001 CH1 IDLE  CH2 IDLE
```

Proctor Case Studies V: CCME #2



Lab Sample Question

Configure the CCME router so that IP phone#1, when idle, will possess the following phone appearance:

09:00 01/21/08			2001
			2001
Your current options			
Redial	New Call	DND	

Candidate's Problem Statement

"I configured the softkey templates AND the system message, but still lost the points… Did you penalize me for not capitalizing every word in the system message?"

Proctor Case Studies V: CCME #2 (Cont)

Candidate's phone#1 looked like this:

09:00 01/21/08			2001
			2001
Your current options			
Redial	New Call	DND	more

Candidate's configuration:

```
ephone-template 1
  softkeys idle Redial Newcall Dnd Cfwdall Pickup

ephone 1
  ephone-template 1
  mac-address 1111.1111.1111
```

Excessive configuration

Summary: CUCME

Be Familiar with the Following About CUCME

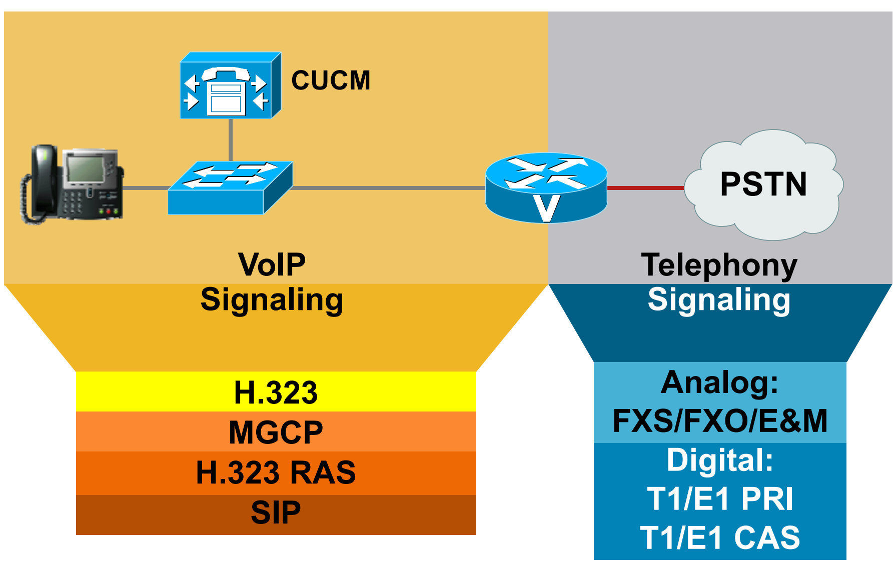
- Mandatory CUCME SCCP and SIP commands
- Configuration options to distribute calls
- Configuration options to allow/restrict calls
- Configuration options to customize phones
- Know CUCME show commands and debug commands well

Q and A

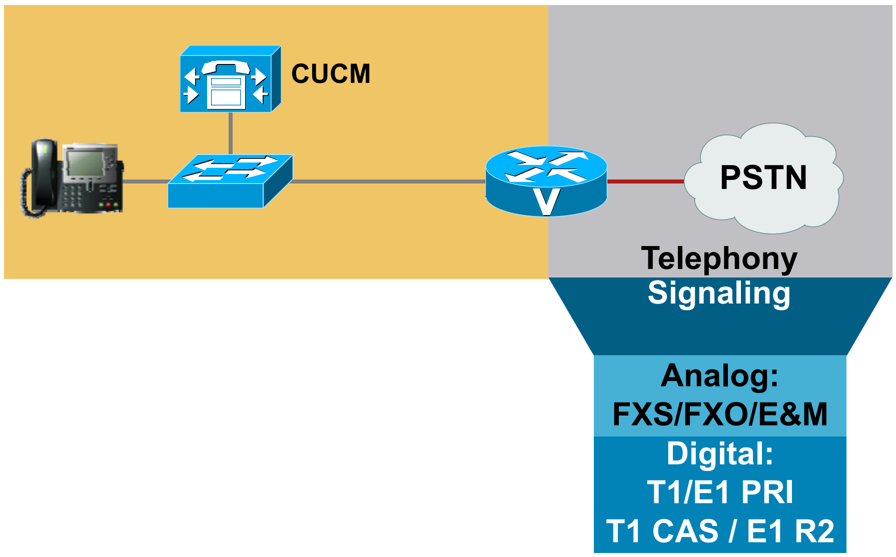
Session 6

Voice Gateways and Protocols

Voice Gateway Protocols



Voice Telephony Signaling Protocols



Digital Voice Telephony Signaling Types

- Common Channel Signaling (CCS)

  Signaling information being carried out-of-channel, separate from the voice traffic

  Most well-known CCS signaling type is ISDN-PRI

  Both with a dedicated D channel for signaling, T1-PRI has 23 bearer channels for voice and E1-PRI has 30 B channels

- Channel Associated Signaling (CAS)

  Signaling information being carried in-channel, interleaved with voice traffic

  Common types are T1-CAS E&M emulation

Digital Voice Signaling: ISDN-PRI


```
card type t1 0 0
!
isdn switch-type primary-ni
!
controller T1 0/0/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
!
int s0/0/0:23
 isdn incoming-voice voice
 isdn switch-type primary-ni
!
voice-port 0/0/0:23
!
dial-peer voice 1 pots
 destination-pattern 3…
 direction-inward-dial
 port 0/0/0:23
!
```

- Globally defines ISDN switch type
- Defines T1-PRI under the T1 controller
- D-channel (int s0/0/0:23) and voice-port will be automatically created once pri-group is defined on the T1 controller; D-channel carries the call information such as DNIS (called number) and ANI (calling number)
- Create pots dial-peer which defines voice call routing rules

Digital Voice Signaling: T1-CAS E&M

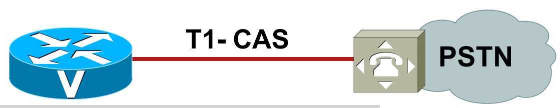
E&M Feature Group D: Double wink with the second wink to acknowledge reception of DNIS; FGD supports collection of ANI

```
Gateway(config-controller)#ds0-group 1 time 1-24 type ?
  e&m-delay-dial         E & M Delay Dial
  e&m-fgd                E & M Type II FGD
  e&m-immediate-start    E & M Immediate Start
  e&m-wink-start         E & M Wink Start
  ext-sig                External Signaling
  fgd-eana               FGD-EANA BOC side
  fgd-os                 FGD-OS BOC side
  fxo-ground-start       FXO Ground Start
  fxo-loop-start         FXO Loop Start
  fxs-ground-start       FXS Ground Start
  fxs-loop-start         FXS Loop Start
  none                   Null Signalling for External Call Control
```

Single wink is sent to the remote end to signal readiness to receive DNIS; A.K.A Feature Group B

FGD Equal Access North America; A variant of FGD which supports sending of ANI

T1-CAS E&M Configuration to Support ANI




```
controller T1 0/0
 framing esf
 linecode b8zs
 ds0-group 1 timeslots 1-12 type e&m-fgd
 ds0-group 2 timeslots 13-24 type fgd-eana
!
voice-port 0/0:1
!
voice-port 0/0:2
!
dial-peer voice 1 pots
 incoming called-number .
 direct-inward-dial
 port 0/0:1
!
dial-peer voice 2 pots
 incoming called-number .
 destination-pattern 9T
 direct-inward-dial
 port 0/0:2
```

Use first 12 channels and e&m-fgd to receive inbound calls and receive ANI information

Use last 12 channels and fgd-eana to send outbound calls and send ANI

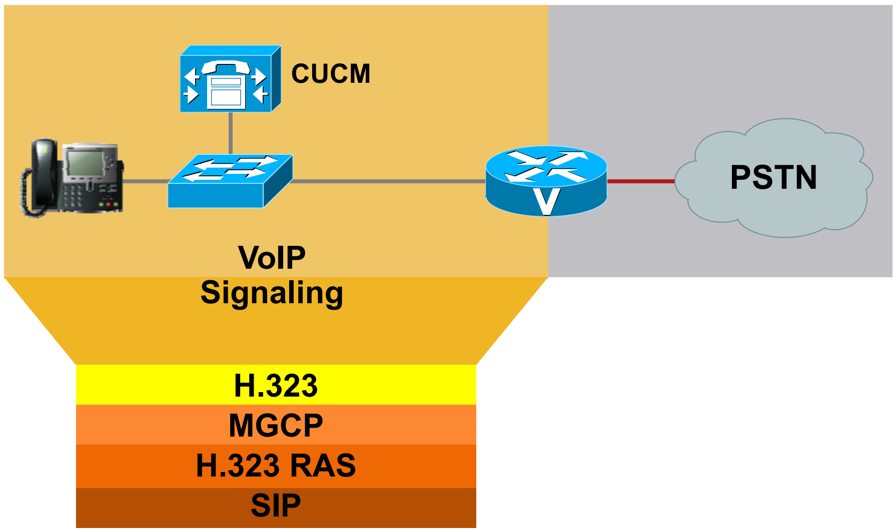
Direct-inward-dial used to prevent the gateway from generating a second dial-tone on inbound calls

Useful Cisco IOS Debug Commands: T1-PRI/CAS

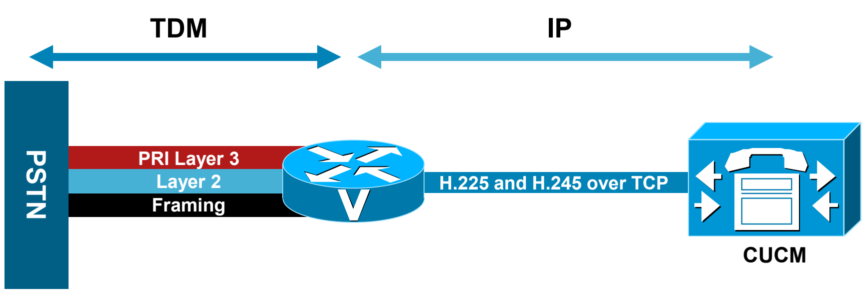
```
PRI-Gateway#debug isdn ?
  all        ISDN debug messages
  api        ISDN Application Program Interface(s)
  cc         ISDN Call Control
  error      ISDN error messages
  events     ISDN events
  mgmnt      ISDN management
  q921       ISDN Q921 frames
  q931       ISDN Q931 packets
  standard   Standard ISDN debugging messages
  tgrm       ISDN TGRM events
```

```
CAS-Gateway#debug vpm ?
  all        Enable All VPM debugging
  dsp        Enable dsp message trace (Warning: driver level trace)
  error      Enable dsp error trace
  overlay    Enable DSPware overlay debugging
  port       Debug only on port specified
  signal     Debug signaling services
  spi        Enable session debugging trace
  tgrm       Enable tgrm debugging
  trunk-sc   trunk conditioning
  voaal2     Debug Voice over AAL2
```

VoIP Signaling Protocols

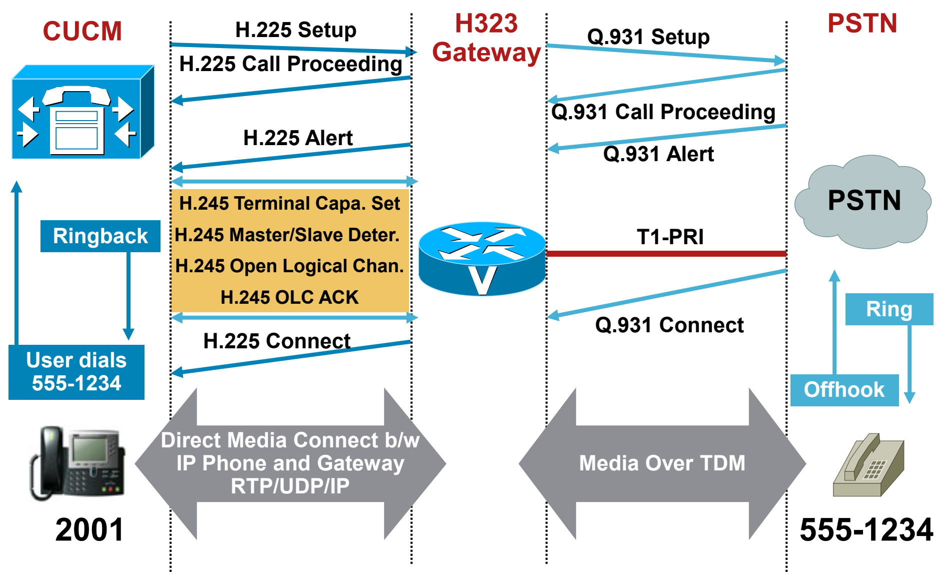


H.323



- H.323 is a "peer-to-peer" protocol
- All PSTN signaling terminates on gateway
- H.225 and H.245 signaling communications over TCP between gateways and CUCM
- Media over UDP directly between gateways and IP phones; CUCM responsible for call setup/tear-down and capability negotiation only

H.323 Call Illustration



Basic H.323 Cisco IOS Configuration

```
card type t1 0 0
!
controller T1 0/0/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
!
interface Serial0/0/0:23
 isdn switch-type primary-ni
 isdn incoming-voice voice
!
dial-peer voice 1 voip
 destination-pattern 2...
 session target ipv4:20.1.1.1
 codec g711ulaw
 dtmf-relay h245-alphanumeric
!
dial-peer voice 9 pots
 destination-pattern 9T
 direct-inward-dial
 port 0/0/0:23
```

- Defines T1-PRI as PSTN signaling
- D-channel and its configurations
- VoIP dial-peer, define H.323 call properties here
- Destination-pattern for digit matching
- Session target pointing to IP address of remote H.323 peer: i.e. CUCM's IP addr.
- Use g711u codec; default is g729
- Enables DTMF relay using H245-alpha; default is disabled
- Pots dial-peer pointing to the PRI with destination-pattern, pots peers strips explicitly matched digit(s) in destination-pattern

Additional H.323 Cisco IOS Configuration Options

```
interface loopback 0
 ip address 10.1.1.1 255.255.255.0
 h323-gateway voip interface
 h323-gateway voip bind srcaddr 10.1.1.1
!
voice class h323 1
 h225 timeout setup 5
!
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711ulaw
!
dial-peer voice 1 voip
 destination-pattern 2...
 session target ipv4:20.1.1.1
 voice-class h323 1
 voice-class codec 1
!
dial-peer voice 2 voip
 destination-pattern 2...
 session target ipv4:20.1.1.2
 voice-class h323 1
 voice-class codec 1
 preference 1
```

Forces this gateway to use the loopback interface for all H.323 signal and RTP traffic

H.225 setup redundancy: try a second VOIP dial-peer if the remote H.323 peer does not response in 5 seconds

H.245 codec negotiation flexibility: negotiate to g729 if possible; otherwise g711ulaw is okay too

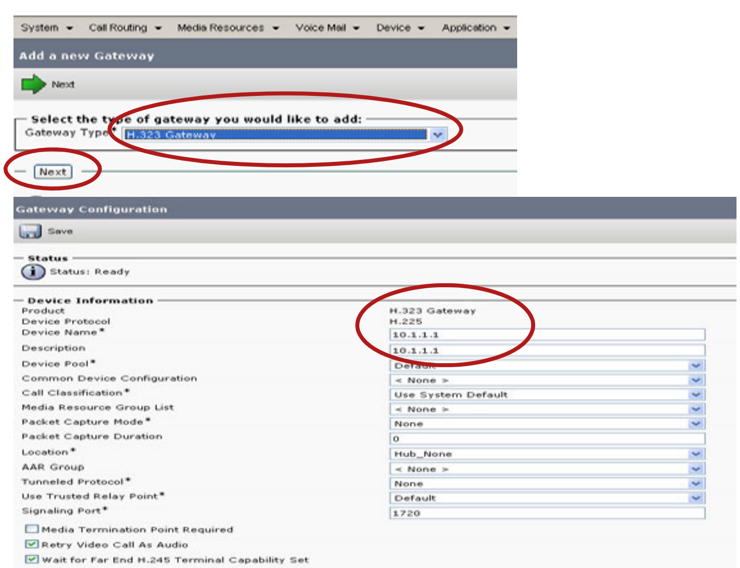
Try this dial-peer first if 2… is match because it has the highest preference: 0; default preference value, therefore invisible in dial-peer configuration

If the IP host in dial-peer 1 (20.1.1.1) does not response H.225 setup in 5 seconds, try this dial-peer as it has lower preference

CUCM H.323 Gateway Configuration

1

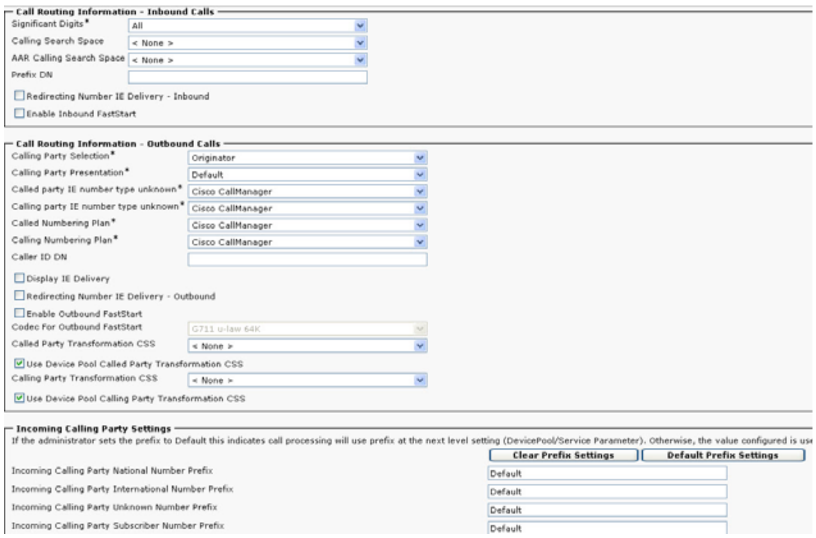
2a



CUCMH.323 Gateway Configuration (Cont)

Continued from CUCM H.323 Gateway Configuration Page

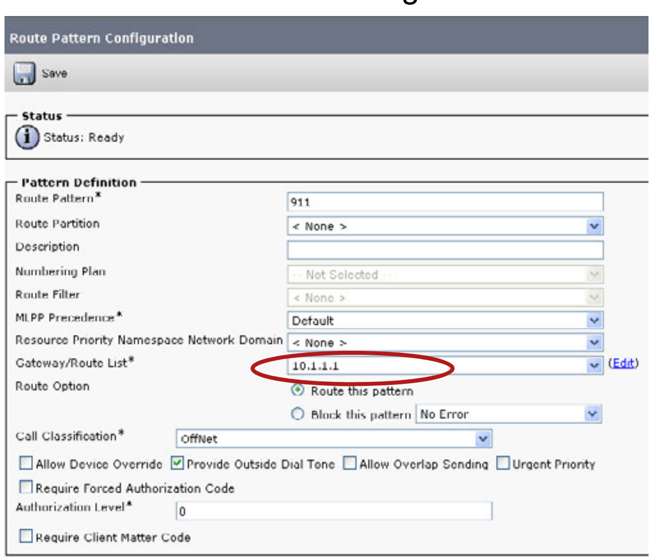
2b



CUCM H.323 Gateway Configuration (Cont)

Define a Route Pattern Pointing to the H.323 Gateway

3



Useful Cisco IOS Verification

Commands: H.323

```
H323-gateway#sh call active voice brief

  Telephony call-legs: 1
  SIP call-legs: 0
  H323 call-legs: 1
  MGCP call-legs: 0
  Total call-legs: 2

131E : 1452845022hs.1 +144 pid:1234 Answer 51234 active
 dur 00:00:12 tx:671/107360 rx:603/96480
 IP 20.1.1.20:19886 rtt:0ms pl:8310/0ms lost:0/1/0 delay:64/64/65ms g711ulaw

131E : 1452845025hs.1 +141 pid:408 Originate 14083132001 active
 dur 00:00:12 tx:603/96480 rx:672/107520
 Tele 1/0:23 (8617): tx:13440/1344/0ms g711ulaw noise:0 acom:19  i/0:-56/-38 dBm
```

```
H323-gateway#sh call active voice
<SNIP><SNIP><SNIP><SNIP><SNIP>
VOIP:
RemoteIPAddress=20.1.1.1
RemoteUDPPort=19886
RemoteSignallingIPAddress=20.1.1.1
RemoteSignallingPort=3139
RemoteMediaIPAddress=20.1.1.20
tx_DtmfRelay=inband-voice
```

```
H323-gateway#sh call active voice
<SNIP><SNIP><SNIP><SNIP><SNIP>
ReceiveDelay=64 ms
LostPackets=0
EarlyPackets=1
LatePackets=0
VAD = enabled
CoderTypeRate=g711ulaw
CodecBytes=160
CallerName=Ben Ng
```

Useful Cisco IOS Debug

Commands: H.323

```
H323-gateway#debug cch323 ?
  CAPACITY   Enable Call Capacity debugging trace
  NXE        Enable NXE transport debugging trace
  RAS        Enable RAS State Machine debugging trace
  all        Enable all CCH323 debugging traces
  h225       Enable H225 State Machine debugging trace
  h245       Enable H245 State Machine debugging trace
  preauth    Enable CCH323 preauth debugging trace
```

```
H323-gateway#debug h245 ?
  asn1     H.245 ASN1 Library
  events   H.245 Events
```

```
H323-gateway#debug voip ccapi ?
  error   CCAPI error legs
  inout   CCAPI Funtion in (enter) and out (exit)
```

```
H323-gateway#csim start <destination-pattern-you-wish-to-test>
```

MGCP (Media Gateway Control Protocol)

- Media Gateway (MG) contains "simple" endpoints, which can be either analog voice-ports (FXS/FXO/E&M) or digital (T1-PRI/T1-CAS) voice trunks
- Call intelligence of these endpoints are provided by Media Gateway Controller (MGC) or Call Agent (CA), in our case, the Cisco Unified Communications Manager
- Master/Slave relationship between MGC/CA and MG
- MGCP messages are sent over IP/UDP between MGC and MG—signaling plane
- Voice traffic is carried over IP/RTP—data plane

MGCP Endpoints

- Endpoints are voice ports on a MGCP gateway
- Analog Endpoint Identifier

  AALN/S1/SU0/0@MGCP-GWY.cisco.com: the endpoint is voice port 1/0/0 on a gateway with hostname of MGCP-GWY and domain name of cisco.com

- Digital Endpoint Identifier

  S1/ds1-0/1@**MGCP-GWY**.cisco.com: the endpoint is b-channel #1 on T1 controller 1/0 on a gateway with hostname of **MGCP-GWY** and domain name of cisco.com

MGCP Messages (UDP Port 2427)

▪ End Point Configuration	EPCF	(CA → EP)
▪ Create Connection	CRCX	(CA → EP)
▪ Modify Connection	MDCX	(CA → EP)
▪ Delete Connection	DLCX(CA <-> EP)	
▪ Notification Request	RQNT	(CA → EP)
▪ Notify	NTFY	(CA ← EP)
▪ Audit Endpoint	AUEP	(CA → EP)
▪ Audit Connection	AUCX	(CA → EP)
▪ Restart In Progress	RSIP	(CA ← EP)

MGCP FXS Call Flow Explained

CallManager **MGCP Gateway**

(1) {Stn. Off-hook} “**NTFY O: L/hd**”

(2) “**RQNT R: L/hu,D/[0-9*#] S:dl**”
{dial-tone, send digit map}

(3) {Digit:} **“NTFY O: 4”**

(4) :**RQNT R: L/hu, D/[0-9*#] S:”**
{Turn off dial-tone}

(5) {Digit(s)...} “**NTFY O: 5”**

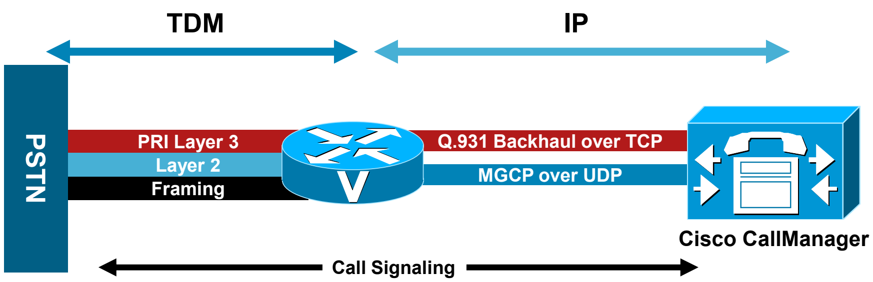
:

(6) CRCX {create connection}
Turns on ring tone

(7) Ack with local RTP addr/port

(8) MDCX {modify connection,
sends remote peer RTP info}

MGCP: PRI Backhaul



- Framing and Layer 2 signaling terminates at the gateway
- Q.921 status and Q.931 signal backhauled to the Cisco CallManager
- MGCP 0.1 with Cisco CallManager only
- MGCP messages over UDP, port 2427
- PRI Backhaul messages over TCP, port 2428

Cisco IOS MGCP PRI Backhaul

Configuration

```
hostname GW1
!
mgcp
mgcp call-agent 20.1.1.2
!
!
ccm-manager redundant-host 20.1.1.1
ccm-manager mgcp
!
controller T1 1/0
 linecode b8zs
 framing esf
 pri-group timeslots 1-24 service mgcp
!
interface Serial1/0:23
 no ip address
 no logging event link-status
 isdn incoming-voice voice
 isdn bind-l3 ccm-manager
!
dial-peer voice 101 pots
 service mgcp
 port 1/0:23
```

- Must match "Domain Name" on MGCP Gateway page on CCM
- Enables MGCP process globally
- Defines Primary Call-agent: the IP address of primary CCM
- Defines secondary call-agent
- MGCP version 0.1 with CCM
- Defines on the T1 controller that the PRI ports will be serviced by MGCP
- Under D-channel, binds L3 (Q.931) to call manager
- Defines MGCP as the call application under pots dial-peer

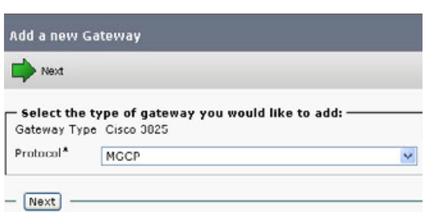
Additional Cisco IOS MGCP
Configuration Options

```
GW1(config)#ccm-manager ?
  application      application specific
  config           MGCP download configuration
  download-tones   Enable Tone Download from TFTP server
  fallback-mgcp    Enable Fallback from MGCP to H.323 mode if no CallManager is
                   available
  fax              Enable fax protocol for MGCP
  mgcp             Enable CallManager Application MGCP mode
  music-on-hold    Enable multicast Music-on-hold
  redundant-host   Redundant host list
  switchback       Configure switchback options for rehoming to higher-order
                   CallManager

GW1(config)#mgcp bind ?
  control   bind only MGCP control packets
  media     bind only media packets
```

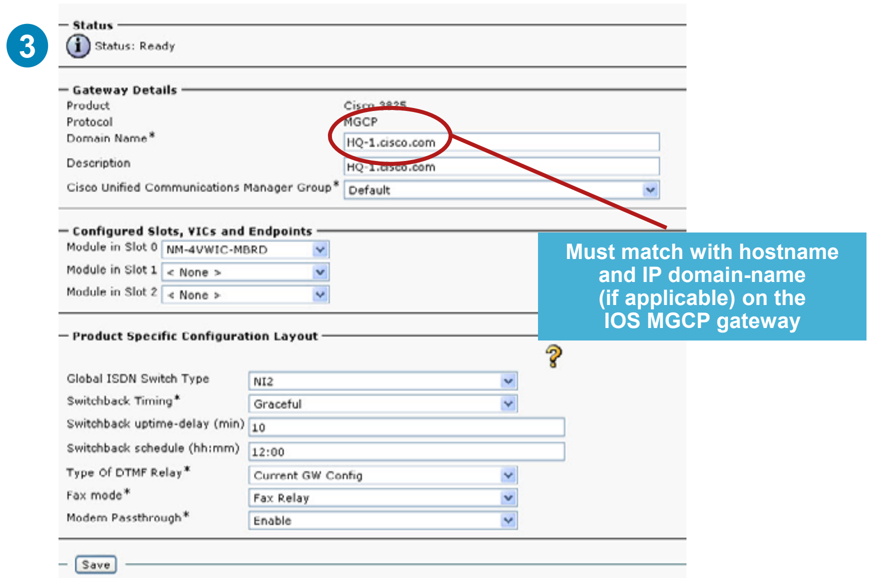
MGCP: CUCM Configuration

1

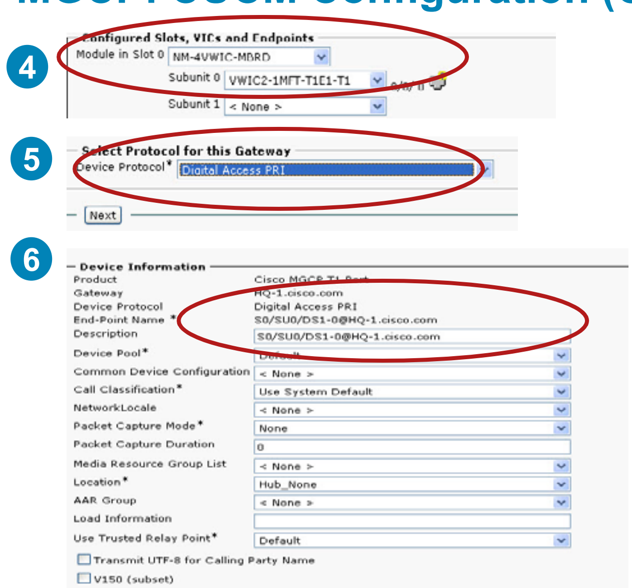
2



MGCP: CUCM Configuration (Cont)



MGCP: CUCM Configuration (Cont)



Useful Cisco IOS MGCP

Verification Commands

```
GW1#sh ccm-manager ?
  backhaul         Backhaul Info
  config-download  Automated Config download Info
  download-tones   XML Downloadable Tones
  fallback-mgcp    MGCP CM fallback
  hosts            Hosts Info
  music-on-hold    Music on hold Info
  redundancy       Redundancy Info
  <CR>


GW1#sh mgcp ?
  connection  Display MGCP connection
  endpoint    Display endpoints eligible for MGCP management
  nas         Display MGCP data channel information
  profile     Display MGCP profile
  statistics  Display MGCP statistics
```

Useful Cisco IOS MGCP

Verification Commands

```
GW1#sh isdn stat
Global ISDN Switchtype = primary-ni
ISDN Serial1/0:23 interface
        dsl 0, interface ISDN Switchtype = primary-ni
        L2 Protocol = Q.921  L3 Protocol(s) = CCM-MANAGER 0x0003
    Layer 1 Status:
        ACTIVE
    Layer 2 Status:
        TEI = 0, Ces = 1, SAPI = 0, State = MULTIPLE_FRAME_ESTABLISHED
    Layer 3 Status:
        0 Active Layer 3 Call(s)
    Active dsl 0 CCBs = 0
    The Free Channel Mask:  0x8000003F
    Number of L2 Discards = 2, L2 Session ID = 30
    Total Allocated ISDN CCBs = 0
```

Useful Cisco IOS MGCP

Debug Commands

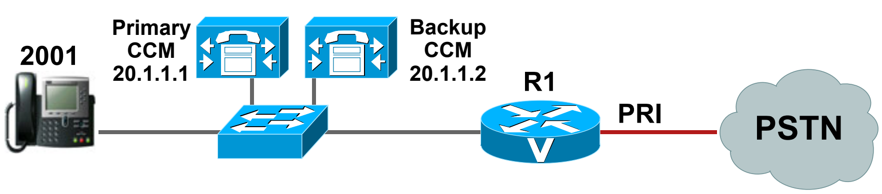
```
GW1#debug mgcp ?
  all       Enable all MGCP debug trace
  errors    MGCP errors
  events    MGCP events
  media     MGCP media
  nas       MGCP nas (data) events
  packets   MGCP packets
  parser    MGCP parser and builder
  src       MGCP System Resource Check CAC
  voipcac   MGCP VOIP CAC

GW1#debug ccm-manager ?
  backhaul          CallManager backhaul debug
  config-download   CallManager Automated config debug
  errors            CallManager errors
  events            CallManager events
  music-on-hold     CallManager music-on-hold
```

Proctor Case Studies VII:

MGCP Gateway #1

Lab Sample Question

Configure R1 as a MGCP Gateway for CUCM.
If the primary CUCM goes down, make sure all endpoints on the MGCP gateway re-registers to the backup CUCM.
Also ensure IP phones can send/receive calls to/from PSTN.



Candidate's Problem Statement

"I verified that my MGCP gateway worked, I even tested all inbound and outbound calls, why did I not receive points?"

Proctor Case Studies VII:

MGCP Gateway #1 (Cont)

Could you identify the mistake from the snippet of this “show ccm-manager” command?

```
R1#sh ccm-manager
MGCP Domain Name: R1
Priority         Status                        Host
============================================================
Primary          Registered                    20.1.1.1
First Backup     None
Second Backup    None

Current active CallManager:     20.1.1.1
Backhaul/Redundant link port:    2428
Failover Interval:               30 seconds
Keepalive Interval:              15 seconds
```

Candidate Missed the Following Command

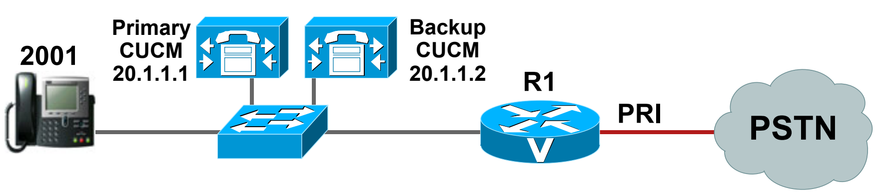
```
R1(config)#ccm-manager redundant-host 20.1.1.2
```

Proctor Case Studies VIII:

MGCP Gateway #2

Lab Sample Question

Configure R1 as a MGCP Gateway for CUCM. If the Primary CUCM goes down, make sure all endpoints on the MGCP gateway re-registers to the Backup CUCM. Also ensure IP phones can send/receive calls to/from PSTN.



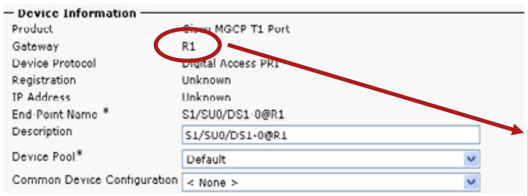
Candidate’s Problem Statement

“My MGCP gateway could not register to the CUCM. On the CUCM’s MGCP gateway configuration page, I see “Registration: Unknown”; on R1, I see “Registering” in “show ccm-manager”.

Proctor Case Studies VIII:

MGCP Gateway #2 (Cont)

Could you identify the mistake from the snippet of the following CUCM gateway configuration page and this “show ccm-manager” command?



Device Information

Product	MGCP T1 Port
Gateway	R1
Device Protocol	Digital Access PRI
Registration	Unknown
IP Address	Unknown
End-Point Name *	S1/SU0/DS1-0@R1
Description	S1/SU0/DS1-0@R1
Device Pool*	Default
Common Device Configuration	< None >

MGCP Domain Name mismatch between CCM and IOS MGCP gateway

```
R1#sh ccm-manager
MGCP Domain Name: R1.cisco.com
Priority        Status                   Host
============================================================
Primary         Registering               20.1.1.1
First Backup    Backup Ready              20.1.1.2
Second Backup   None

Current active CallManager:     None
Backhaul/Redundant link port:   2428
Failover Interval:              30 seconds
Keepalive Interval:             15 seconds
```

Registration, Authentication,

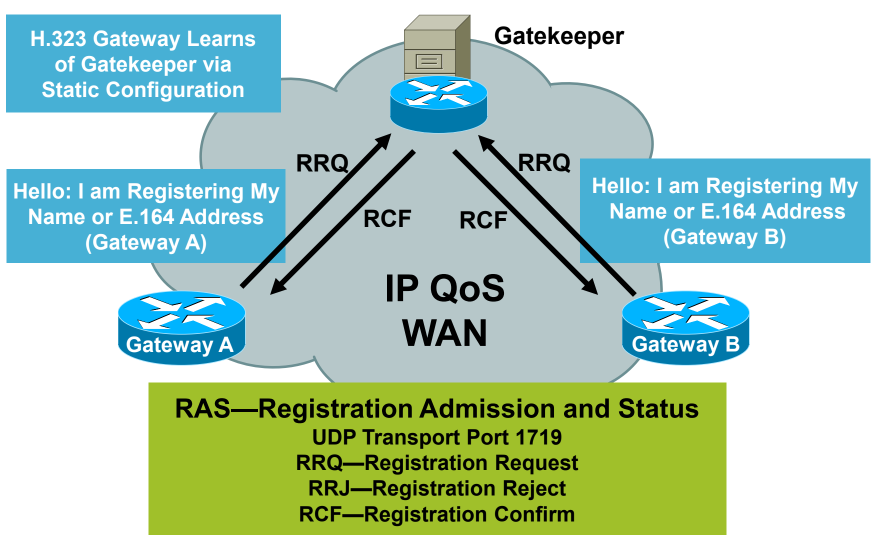
Status (RAS)

- Established between H.323 endpoint and gatekeeper
- Gateway initializes with full registration to gatekeeper
- Gateways sends lightweight registration, based on negotiated time-out, similar to keep-alive
- Unreliable transport—uses UDP
- Gateway could depend on gatekeeper to
  - e.164 address resolution
  - Call Admission Control

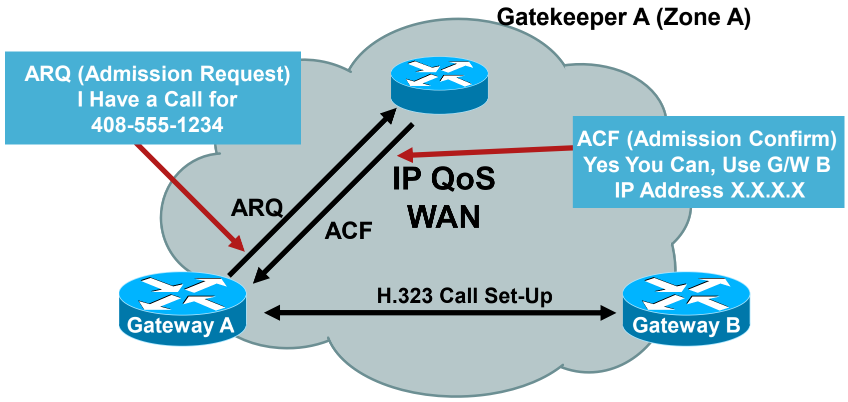
RAS Communication Messages

- GRQ/GCF/GRJ (discovery)
  - Unicast or multicast, find a gatekeeper
- RRQ/RCF/RRJ (registration)
  - Endpoint alias/IP address binding, endpoint authentication
- ARQ/ACF/ARJ (admission)
  - Destination address resolution, call routing
- LRQ/LCF/LRJ (location)
  - Inter-gatekeeper communication
- BRQ/BCF/BRJ (bandwidth modifications)
- DRQ/DCF/DRJ (disconnect)
  - Call termination

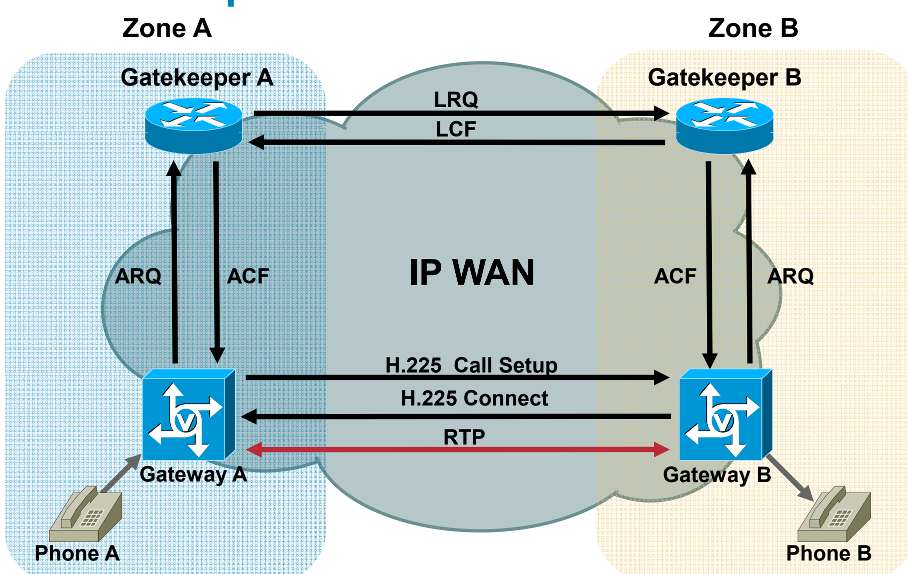
RAS Gatekeeper Registration Illustrated



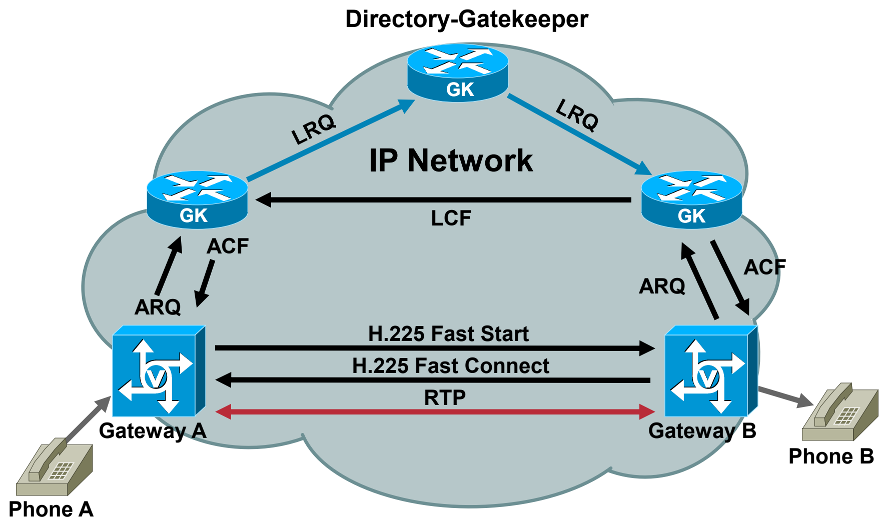
RAS Call Admission Illustrated



Gatekeeper Inter-zone Communication



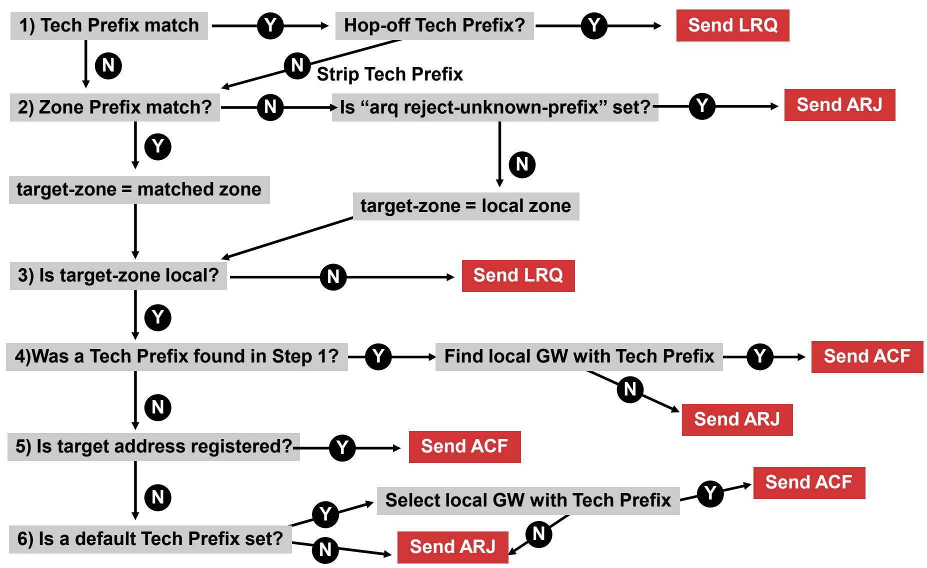
Directory Gatekeeper Call Flow Illustrated



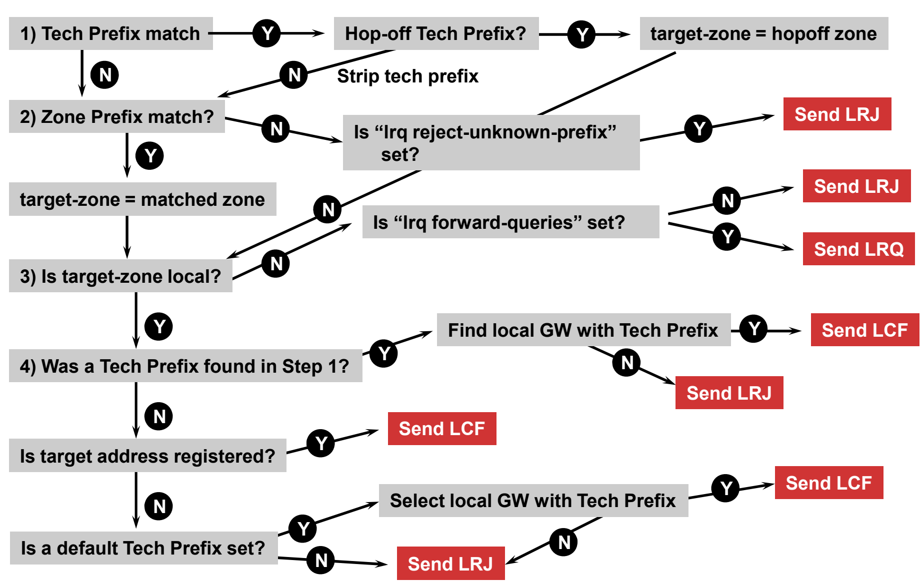
Cisco IOS Gatekeeper: Common Terms

- Zone: A collection of nodes for routing calls (can be H.323 clients, Cisco CallManager clusters, or H.323 Gateways); configure on gatekeepers and gateways/endpoints
- Zone Prefix: A unique number string configured on and used by gatekeepers to associate a dialed number to a zone
- Tech Prefix: A unique number string typically configured on gateways and presented to gatekeepers during registration; tech prefixes are then used by gatekeepers to group endpoints of the same type together; tech-prefix to gateway association could also be manually configured on GK
- Default Technology: Configured on gatekeepers for default routing of any unresolved E.164 addresses to gateways that registered with a specific tech prefix

GK Address Resolution on ARQ



GK Address Resolution on LRQ



Cisco IOS GK Configuration Example

```
gatekeeper
 zone local SJ cisco.com
 zone local SF cisco.com
 zone local DAL cisco.com
 zone remote RTP cisco.com 172.16.14.130 1719
 zone prefix SJ 1408*
 zone prefix SF 1415*
 zone prefix RTP 1919*
 zone prefix DAL 1972*
 gw-type-prefix 1#* default-technology
 bandwidth interzone default 512
 bandwidth remote 64
 no shutdown
```

Enter into gatekeeper configuration mode

Define local zone names

Defines remote zone names and IP address

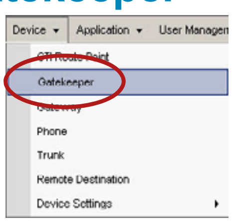
Define local and remote zone prefixes

Any gateways registered with a technology prefix of 1# are gateways of last resort if a called number is not resolved by gatekeeper's existing call routing rules

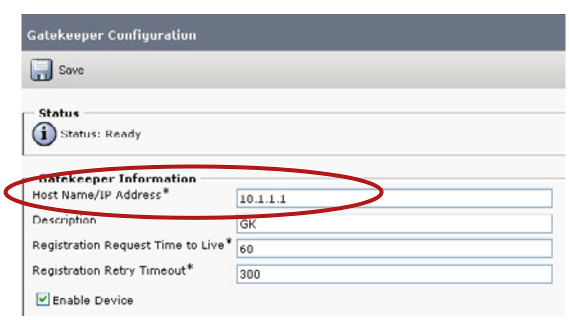
Allow up to four g711 (128x4=512) in local Zone and four g729 (16x4=64) to Remote Zones

CUCM Configurations Example for Gatekeeper

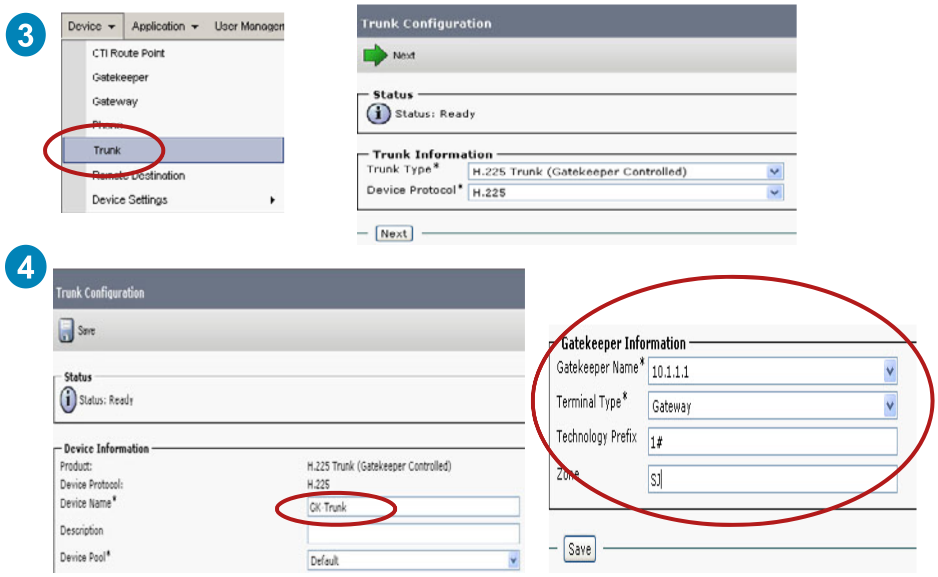
1



2



CUCM Configurations Example for Gatekeeper (Cont)



Cisco IOS GK Verification Commands (I)

```
GK#show gatekeeper ?
  calls           Display current gatekeeper call status
  circuits        Display current gatekeeper circuits
  clusters        Display gatekeeper cluster info
  endpoints       Display all endpoints registered with this gatekeeper
  gw-type-prefix  Display Gateway Technology Prefix Table
  performance     Display gatekeeper performance data
  servers         Display gatekeeper servers info
  status          Display current gatekeeper status
  zone            Display zone information
```

```
GK#show gatekeeper zone prefix
      ZONE PREFIX TABLE
      =================
GK-NAME                 E164-PREFIX
-------                 -----------
SJ                      1408*
SF                      1415*
RTP                     1919*
DAL                     1972*
```

Cisco IOS GK Verification Commands (II)

```
GK#show gatekeeper endpoint
                    GATEKEEPER ENDPOINT REGISTRATION
                    ================================
CallSignalAddr  Port  RASSignalAddr   Port  Zone Name   Type    Flags
--------------- ----- --------------- ----- ---------   ----    -----
20.1.1.1       61042   20.1.1.1       58267    SJ      VOIP-GW
    H323-ID: GK-Trunk_1
    Voice Capacity Max.=  Avail.=  Current.= 0
20.1.1.2       56628   20.1.1.2       54461    SJ      VOIP-GW
    H323-ID: GK-Trunk_2
    Voice Capacity Max.=  Avail.=  Current.= 0
20.30.1.254     1720   20.30.1.254    51112    SJ      VOIP-GW
    H323-ID: H323-Gateway-1
    Voice Capacity Max.=  Avail.=  Current.= 0
Total number of active registrations = 3
```

CUCM servers in a cluster register to gatekeeper using the Device name configured on the CUCM Trunk page; for purpose of having a unique H323-ID for each server in the cluster, CUCM attaches _1, _2, _3, etc., to the end of the configured Trunk Device Name

Cisco IOS GK Debug Commands

To see gatekeeper number matching logic, use “debug gate main 5”:
Note: This is a hidden command

```
GK#debug gate main 5
*Mar  8 18:30:08.577: gk_rassrv_arq: arqp=0x81B89578, crv=0x14, answerCall=0
*Mar  8 18:30:08.581: gk_dns_query: No Name servers
*Mar  8 18:30:08.581: rassrv_get_addrinfo: (19725552000) Tech-prefix match
failed.
*Mar  8 18:30:08.581: rassrv_get_addrinfo: (19725552000) Matched zone prefix
1972 and remainder 5552000
*Mar  8 18:30:08.601: gk_rassrv_arq: arqp=0x81AA488C, crv=0x8014,
answerCall=1
```

To see RAS messages and information contained within, use “debug h225 asn1”:

```
*Mar  7 21:03:57.339: RAS INCOMING PDU ::=
value RasMessage ::= admissionRequest :
destinationInfo
      dialedDigits : "19725552000"
      ip 'AC10F279'H
      port 4042
      bandWidth 1280
      callReferenceValue 14
      gatekeeperIdentifier {"SJ"}
    }
*Mar  7 21:03:57.355: ARQ (seq# 11652) rcvd
```

Lab Exam Case Studies IX:
Gatekeeper

Lab Sample Question

Register your CUCM servers and CUCME router to the gatekeeper. The CUCM servers should register with tech-prefix of "1" and the CCME router should register with tech-prefix of "2". When properly registered, the gatekeeper should produce the following "show gatekeeper gw-type-prefix":

```
Gatekeeper#show gatekeeper gw-type-prefix

GATEWAY TYPE PREFIX TABLE
=========================
Prefix: 2*
  Zone Gatekeeper master gateway list:
    120.100.1.1:1720 CCME

Prefix: 1*
  Zone Gatekeeper master gateway list:
    20.1.1.2:49296 CCM_2

    20.1.1.1:49824 CCM_1
```

Lab Exam Case Studies IX:
Gatekeeper (Cont)

Candidate's Problem Statement

Both of my CCM servers and the CCME routers are registered with the gatekeeper. I can even route calls between them. Why did I lose point on this section?

Let's take a look at the candidates "show gatekeeper gw-type-prefix" output:

Candidate's output:

```
Gatekeeper#show gatekeeper gw-type-prefix

GATEWAY TYPE PREFIX TABLE
=========================
Prefix: 2#*
  Zone Gatekeeper master gateway list:
    120.100.1.1:1720 CCME

Prefix: 1#*
  Zone Gatekeeper master gateway list:
    20.1.1.2:49296 CCM_2

    20.1.1.1:49824 CCM_1
```

Requested output:

```
Gatekeeper#show gatekeeper gw-type-prefix

GATEWAY TYPE PREFIX TABLE
=========================
Prefix: 2*
  Zone Gatekeeper master gateway list:
    120.100.1.1:1720 CCME

Prefix: 1*
  Zone Gatekeeper master gateway list:
    20.1.1.2:49296 CCM_2

    20.1.1.1:49824 CCM_1
```

SIP Basics

- SIP is Session Initiation Protocol
- SIP is a peer-to-peer protocol defined in RFC 3261
- SIP is human readable; (ASCII text-based; aids debugging)
- Uses UDP as well as TCP, flexibly connecting users independent of the underlying infrastructure
- SIP is extensible; (unrecognized headers are ignored)

SIP Endpoints and Dialogs

- SIP emphasizes a peer-to-peer model with end-to-end request/response transactions
- An issuer of a request is a User Agent Client (UAC)
- A responder to a request is a User Agent Server (UAS)
- An endpoint that incorporates a UAC and a UAS is termed a User Agent (UA)
- Transactions create dialogues

INVITE sip:2000@10.1.1.102:5060 SIP/2.0
From: “1000" <sip:1000@10.1.1.101>;tag=00120193edaa0fda62e313d6-2643faab
To: <sip:2000@10.1.1.102>
CallId: 00120193-edaa000d-2d230f76-44744f4d@10.1.1.101

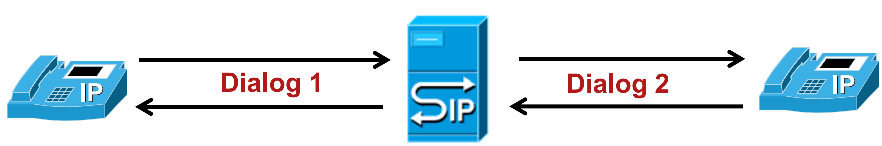
SIP/2.0 200 OK
From: “1000" <sip:1000@10.1.1.101>;tag=00120193edaa0fda62e313d6-2643faab
To: “2000" <sip:2000@10.1.1.102>;tag=ad611738-235c-4e04-8a1b-ef697b19fb06-22031740
CallId: 00120193-edaa000d-2d230f76-44744f4d@10.1.1.101

SIP Intermediate Components

- SIP Requests can be managed by intermediate components such as proxy servers
- Proxy servers have limited ability to modify SIP messages
  - Must obey strict rules regarding the modification of SIP headers
  - Can't touch SIP bodies, where the session's media is defined
- The dialog remains end-to-end

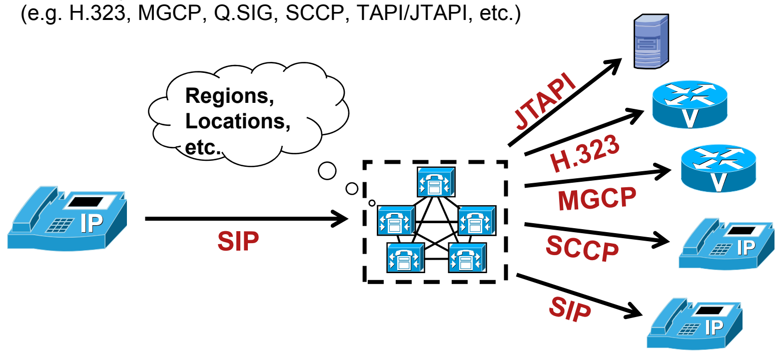
SIP B2BUA

- A commonly-adopted model, called a back-to-back user agent (B2BUA), combines a UAC and a UAS so that a request received by the UAS is reissued by the co-resident UAC
- The B2BUA generates a completely independent outgoing dialog, which affords it the ability to synthesize SIP headers and bodies of its choosing
- B2BUAs are inherently more stateful than proxy servers or redirect servers, and can more easily inter-work SIP with other protocols

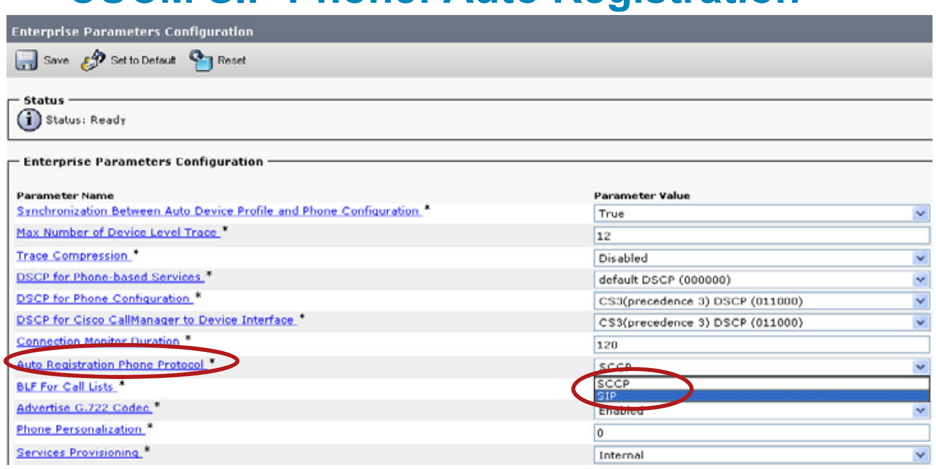


CUCM and B2BUA

- Cisco Unified Communications Manager 5.x/6.x/7.x uses the B2BUA model for all types of SIP calls (trunk-side and line-side). This allows Communications Manager to:
  - Fully support standards-based SIP while maintaining the centralized control and management capabilities of a PBX
  - Seamlessly inter-work SIP with all other supported protocols (e.g. H.323, MGCP, Q.SIG, SCCP, TAPI/JTAPI, etc.)



CUCM SIP Phone: Auto Registration


Enterprise Parameters Configuration

Save | Set to Default | Reset

Status

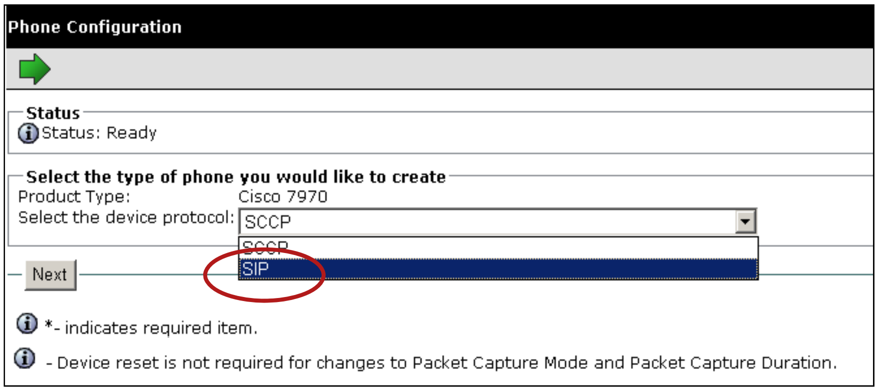
Status: Ready

Enterprise Parameters Configuration

Parameter Name	Parameter Value
Synchronization Between Auto Device Profile and Phone Configuration *	True
Max Number of Device Level Trace *	12
Trace Compression *	Disabled
DSCP for Phone-based Services *	default DSCP (000000)
DSCP for Phone Configuration *	CS3(precedence 3) DSCP (011000)
DSCP for Cisco CallManager to Device Interface *	CS3(precedence 3) DSCP (011000)
Connection Monitor Duration *	120
Auto Registration Phone Protocol *	SCCP (SCCP / SIP)
BLF For Call Lists *	
Advertise G.722 Codec *	Enabled
Phone Personalization *	0
Services Provisioning *	Internal

- Protocol choice automatically dictates what firmware filename gets specified in the default configuration file for each phone model 1
- When set to SIP, only applies to phones that can run SIP. SCCP-only phone models will still auto-register using SCCP

CUCM SIP Phone Provision



- Provisioning a SIP phone is just like provisioning a SCCP phone
- Protocol choice automatically dictates what firmware filename gets specified in the phones' configuration file 1

Cisco Unified Boarder Element (CUBE)

Formerly known as the Cisco Multiservice IP-to-IP Gateway

- CUBE facilitates end-to-end VoIP by interconnecting disparate VoIP networks
- CUBE provides secure, flexible, and reliable interconnect services
- CUBE interworks the following VoIP protocols:

  h323 to h323

  h323 to sip

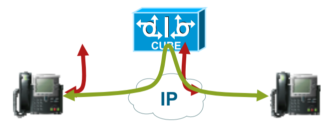
  sip to h323

  sip to sip

Cisco Unified Border Element Architecture

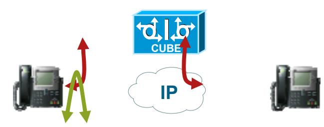
Formerly the Cisco Multiservice IP-to-IP Gateway

- Actively involved in the call treatment, signaling and media streams
  - SIP B2B User Agent
- Signaling is terminated, interpreted and re-originated
  - Provides full inspection of signaling, and protection against malformed and malicious packets
- Media is handled in two different modes
  - Media Flow-Through
  - Media Flow-Around
- Digital Signal Processors (DSPs) are required for transcoding (calls with dissimilar codecs)



Media Flow-Through

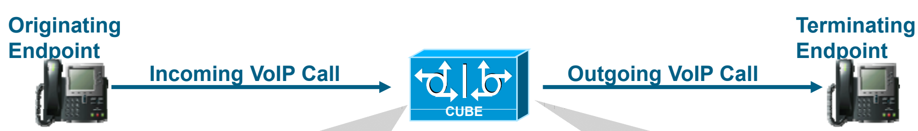
- Signaling and media terminated by the Cisco Unified Border Element
- Transcoding and complete IP address hiding require this model



Media Flow-Around

- Signaling and media terminated by the Cisco Unified Border Element
- Media bypasses the Cisco Unified Border Element

Cisco Unified Border Element Basic Call Flow




```
dial-peer voice 1 voip
  destination-pattern 1000
  incoming called-number .T
  session target ipv4:10.1.1.1
  codec g711ulaw
```

```
dial-peer voice 2 voip
  destination-pattern 2000
  session protocol sipv2
  session target ipv4:20.1.1.1
  codec g711ulaw
```

1. Incoming VoIP setup message from originating endpoint to the Cisco Unified Border Element
2. This matches inbound VoIP dial peer 1 for characteristics such as codec, VAD, DTMF method, protocol, etc.
3. The Cisco Unified Border Element then looks up the called number in the call setup and matches outbound VoIP dial peer 2
4. Outgoing VoIP setup message from the Cisco Unified Border Element to terminating endpoint

H.323 and SIP Interworking

- Requirement
  - Large installed base of H.323 applications, with an increasing number of SIP applications in the same enterprise network
  - Connect H.323 and SIP applications to SP SIP trunks
  - Incompatibilities and variations within same protocol
- The Cisco Unified Border Element supports H.323-H.323, SIP-SIP and H.323-SIP interworking
  - Voice supported for all combinations
  - Video supported for H.323-H.323 and SIP-SIP
- Define incoming and outgoing VoIP dial-peers with required parameters like protocol, transport, codec, CAC, QoS, etc.

```
voice service voip
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
```

H.323 and SIP Interworking

H.323-H.323

In Leg	Out Leg	Support
Fast Start	Fast Start	Bi-Directional
Slow Start	Slow Start	Bi-Directional
Fast Start	Slow Start	Bi-Directional

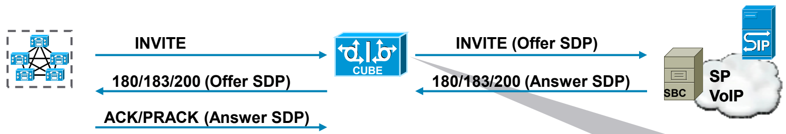
SIP-SIP

In Leg	Out Leg	Support
Early Offer	Early Offer	Bi-Directional
Delayed Offer	Delayed Offer	Bi-Directional
NEW Delayed Offer	Early Offer	Uni-Directional

H.323-SIP

In Leg	Out Leg	Support
Fast Start	Early Offer	Bi-Directional
Slow Start	Delayed Offer	Bi-Directional

Delayed Offer—Early Offer




```
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 …
dial-peer voice 4 voip
  destination-pattern 321....
  voice-class codec 1
  voice-class sip early-offer forced
  session target ipv4:x.x.x.x
```

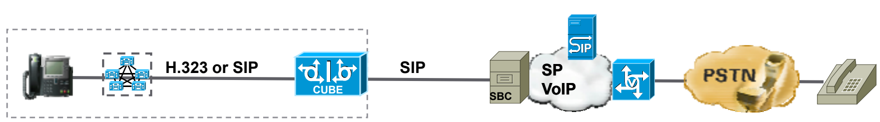
- SP SIP trunk Early Offer (EO) interconnect for enterprise apps that support only Delay Offer (DO)
- Flow-through required for DE-EO supplementary services

	Early	Delayed
Offer	SDP in INVITE	No SDP in INVITE
Answer	SDP in 180/183	SDP in 200

Global Configuration Also Supported:

```
voice service voip
  sip
    early-offer forced
```

Interconnecting Cisco Unified Communications Manager to a SIP Trunk

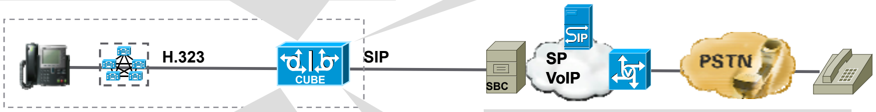


- Security via topology hiding and SIP signaling and media inspection
- Call admission control upon entry to network
- Meet SP UNI requirements via SIP message normalization
- Utilize SIP trunks with H.323 Unified CMs
- DTMF interworking and transcoding
- Co-resident with TDM GW, SRST GW and/or MTP
- Failover or backup via TDM trunks and Unified SRST

Cisco Unified Border Element Configuration: H.323 to Unified CM

```
voice service voip
  address-hiding
  allow-connections h323 to sip
  allow-connections sip to h323
```

```
sip-ua
  retry invite x
  retry bye x
  retry cancel x
```



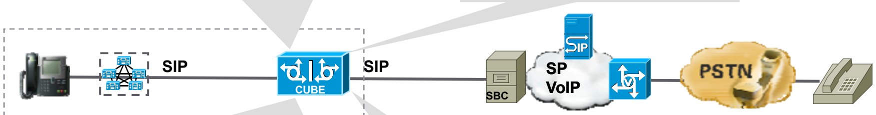

```
dial-peer voice 10 voip
  description incoming H323
  incoming called-number .T
  codec g711ulaw
  dtmf-relay h245-signal
!
dial-peer voice 2 voip
  description outgoing H323
  destination-pattern 8…
  session target ipv4:x.x.x.x
  codec g711ulaw
  dtmf-relay h245-signal
  ip qos dscp cs5 media
  ip qos dscp cs3 signaling
```

```
dial-peer voice 1 voip
  description incoming SIP
  incoming called-number 8…
  session protocol sipv2
  codec g711ulaw
  dtmf-relay rtp-nte
!
dial-peer voice 3 voip
  description outgoing SIP
  destination-pattern .T
  session protocol sipv2
  session target ipv4:x.x.x.x
  codec g711ulaw
  dtmf-relay rtp-nte
  ip qos dscp cs5 media
  ip qos dscp cs3 signaling
```

Cisco Unified Border Element Configuration: SIP to Unified CM

```
voice service voip
  address-hiding
  allow-connections sip to sip
```

```
sip-ua
  retry invite x
  retry bye x
  retry cancel x
```




```
dial-peer voice 10 voip
  description incoming SIP
  incoming called-number .T
  session protocol sipv2
  codec g711ulaw
  dtmf-relay rtp-nte
!
dial-peer voice 2 voip
  description outgoing SIP
  destination-pattern 8…
  session protocol sipv2
  session target ipv4:x.x.x.x
  codec g711ulaw
  dtmf-relay rtp-nte
  ip qos dscp cs5 media
  ip qos dscp cs3 signaling
```

```
dial-peer voice 1 voip
  description incoming SP SIP
  incoming called-number 8…
  session protocol sipv2
  codec g711ulaw
  dtmf-relay rtp-nte
!
dial-peer voice 3 voip
  description outgoing SP SIP
  destination-pattern .T
  session protocol sipv2
  session target ipv4:x.x.x.x
  codec g711ulaw
  dtmf-relay rtp-nte
  ip qos dscp cs5 media
  ip qos dscp cs3 signaling
```

Summary: Voice Gateway and Signaling

Be Familiar with the Following About Voice Gateways

- Telephony signaling configuration: T1/E1-PRI, T1-CAS, E1-R2
- VoIP signaling configuration: H.323, MGCP, RAS, SIP
- VoIP protocol interworking by CUBE
- VoIP signaling redundancy and fail-over options
- Verification and debugging of POTs and VoIP call legs

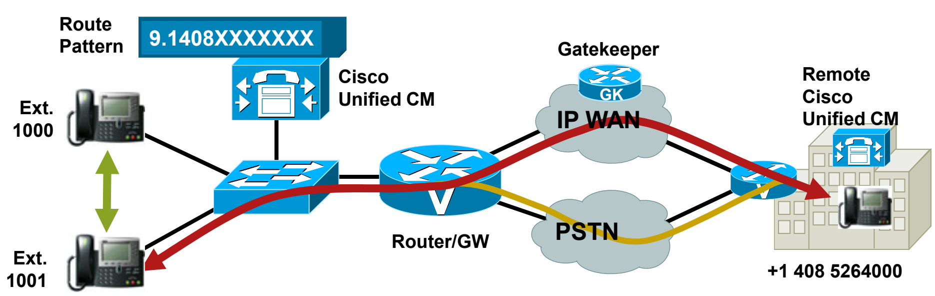
Q and A

Session 7

Dial-Plan Considerations

Dial Plan

The “IP Routing” of IP Telephony



Cisco Unified CM Routes Two Basic Call Types:

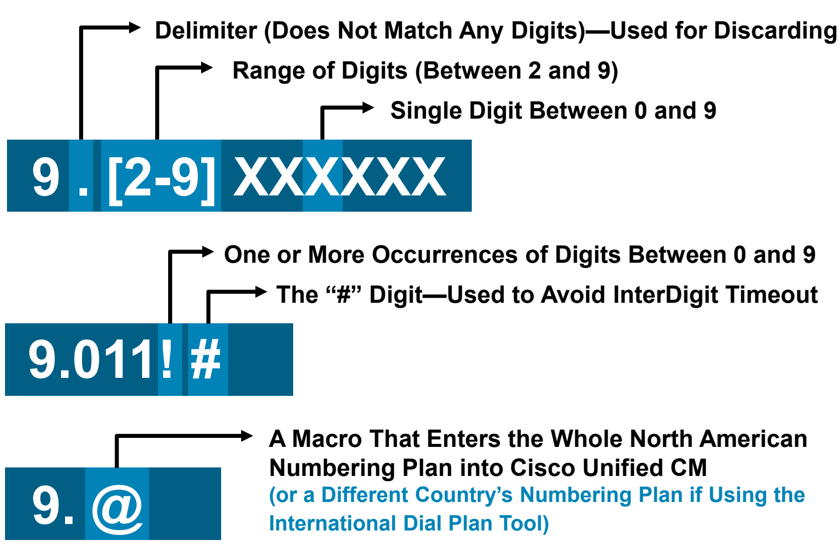
- On-Cluster Calls: Destination Directory Number (DN) is registered with Cisco Unified CM. DNs are considered “internal” routes.
- Off-Cluster Calls: Destination Number is *not* registered with Cisco Unified CM. Route Patterns are configured to allow for “external” routes.
- Alternate routes: Allow On-Cluster and Off-Cluster calls to attempt alternate paths to destination (e.g.: IP WAN not available, go through PSTN)

Cisco Unified CM Route Pattern Digits

Pattern	Description
0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #	Match Exactly One Keypad Button
X	Any Single Digit in the Range 0–9
[xy*z]	Exactly One of Any of the Keypad Buttons in the Brackets
[x-y]	Exactly One of Any Digit Between x and y Inclusively
[^x-y]	Any Digit That Is Not Between x and y Inclusively
!	One or More Digits in the Range 0–9
wildcard?	Zero or More Occurrences of the Previous Wildcard
wildcard+	One or More Occurrences of the Previous Wildcard
@	Numbering Plan Macro
<blank>	Immediately Route Call with No Digits

Cisco Unified CM Route Patterns Logic

Commonly Used Wildcards



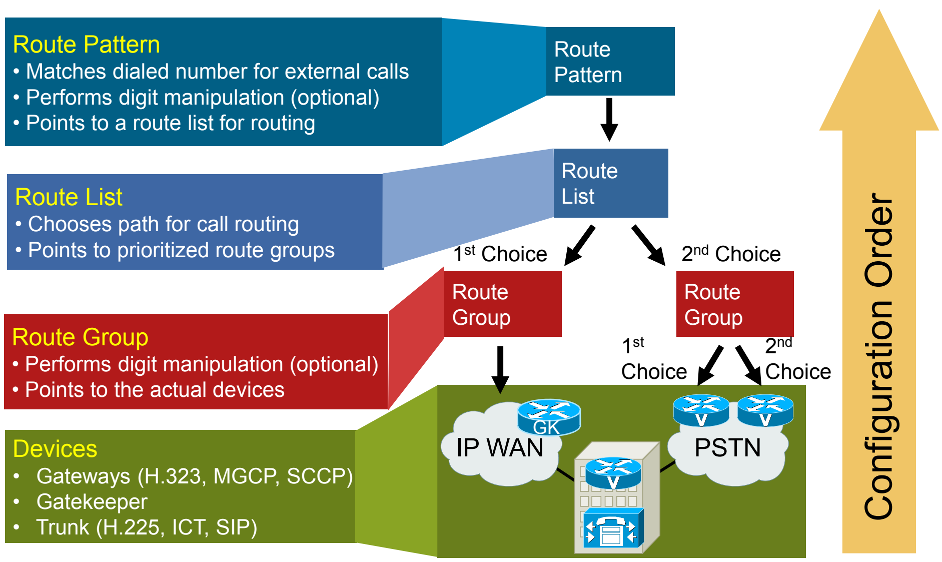
Cisco Unified CM Call Routing Logic

Matching Patterns

1111	**Matches 1111**
***1*1**	**Matches *1*1**
12XX	**Matches Numbers Between 1200 and 1299**
13[25-8]6	**Matches 1326, 1356, 1366, 1376, 1386**
13[^3-9]6	**Matches 1306, 1316, 1326, 13*6, 13#6**
13!#	**Matches Any Number That Begins with 13, Is Followed by One or More Digits, and Ends with #; 135# and 13579# Are Example Matches**

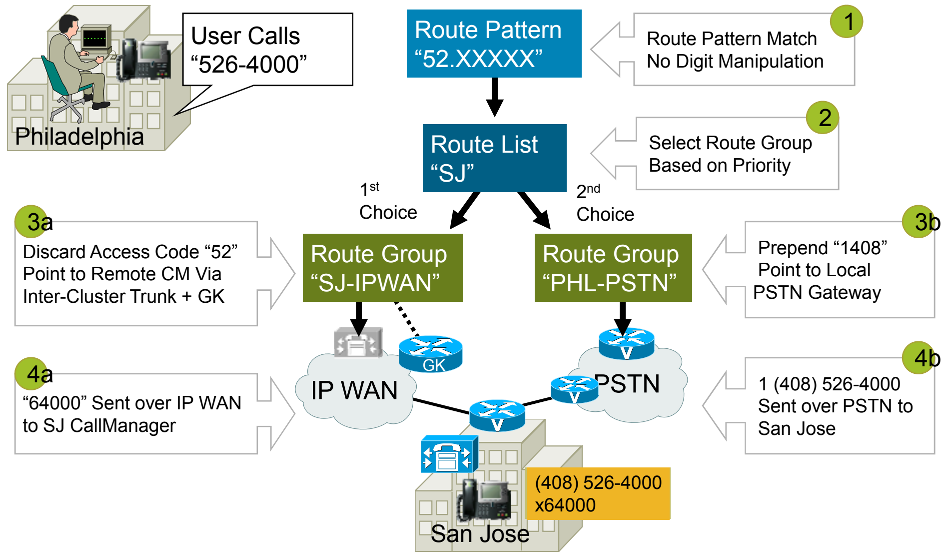
External Routes in Cisco Unified CM

Overall Structure



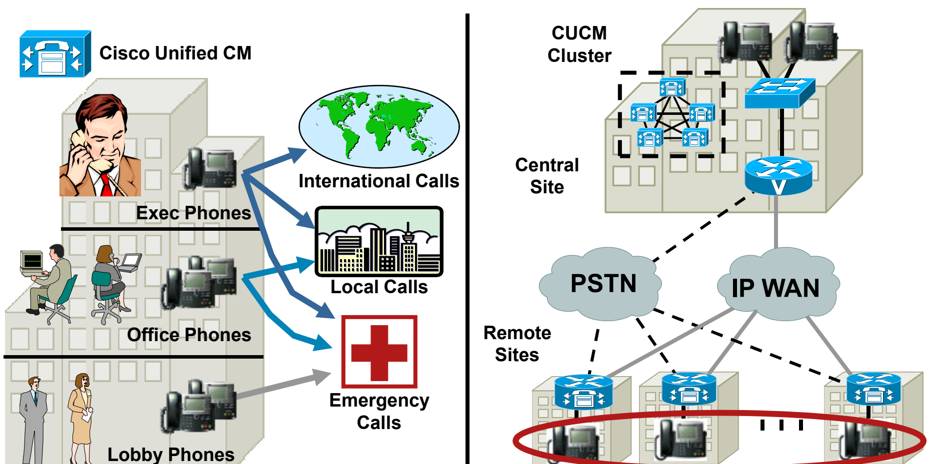
Defining External Routes

Example: PHL to SJ



Building Classes of Service

Routing by User Class or Location



Create "Dial Plan Policy Groups" to Define Calling Restrictions

Instruct these Phones to Use Their Local Gateway for PSTN Access

Building Classes of Service

Understanding Partitions and Calling Search Space (CSS)

Partition:

"Where You Are"

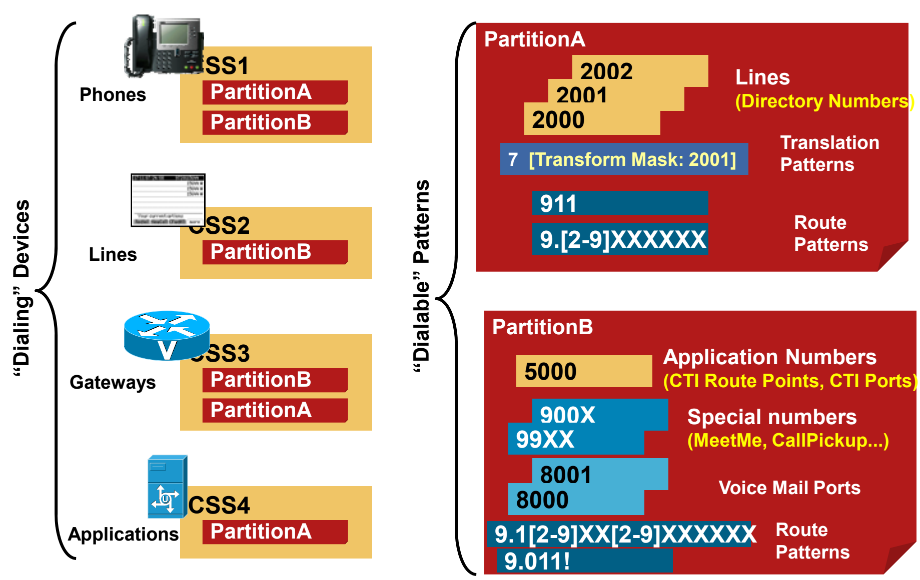
- A group of devices or patterns with similar accessibility
- Items placed into partitions: Directory Numbers (DN), Route Patterns, Voice Mail Ports, etc.

Calling Search Space:

"Where You May Call"

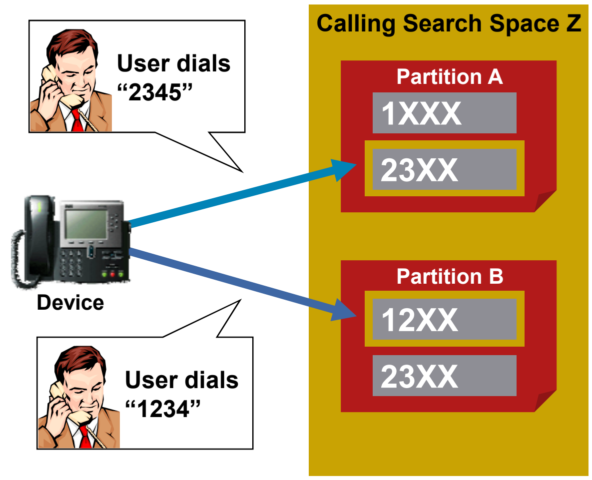
- A collection of partitions which a device can access
- CSS is assigned to IP phones, Gateways, etc

Partitions and Calling Search Spaces



Partitions and Calling Search Spaces

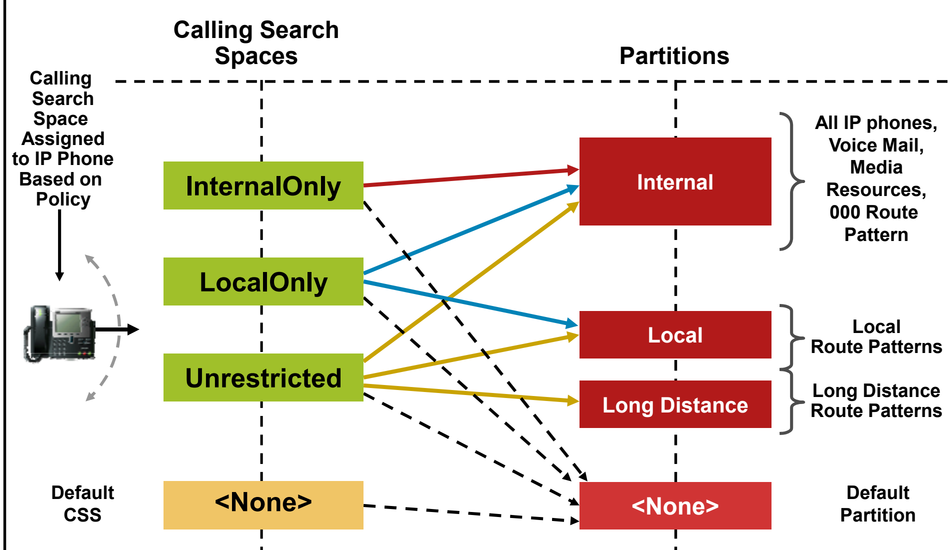
Impact of the Partition Order in a CSS



- Most specific patterns are chosen irrespective of partition order
- Partition order is only used as a tie-breaker in case of equal matches

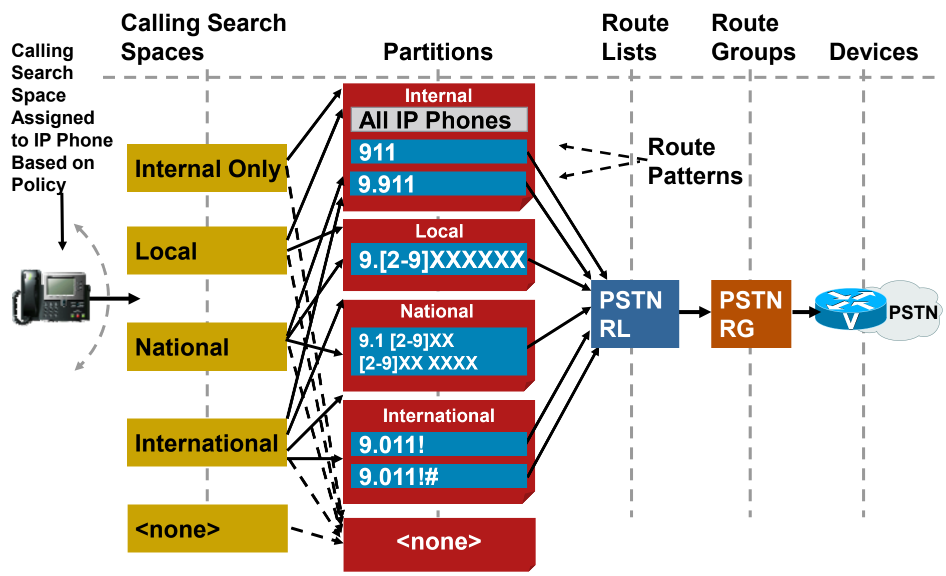
Building Classes of Service

Example of Classes



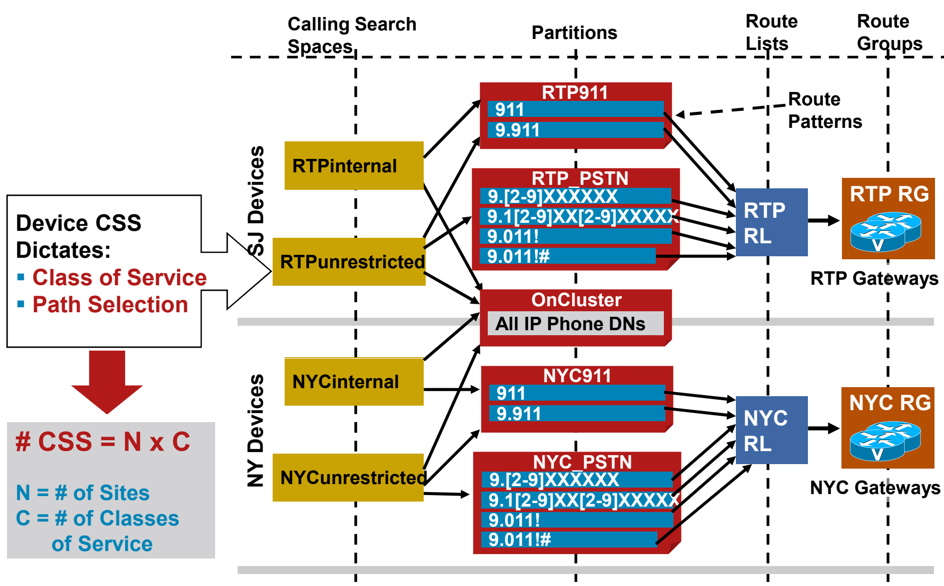
Building Classes of Service

Single Site Deployment Example: Composite Dial-Plan View



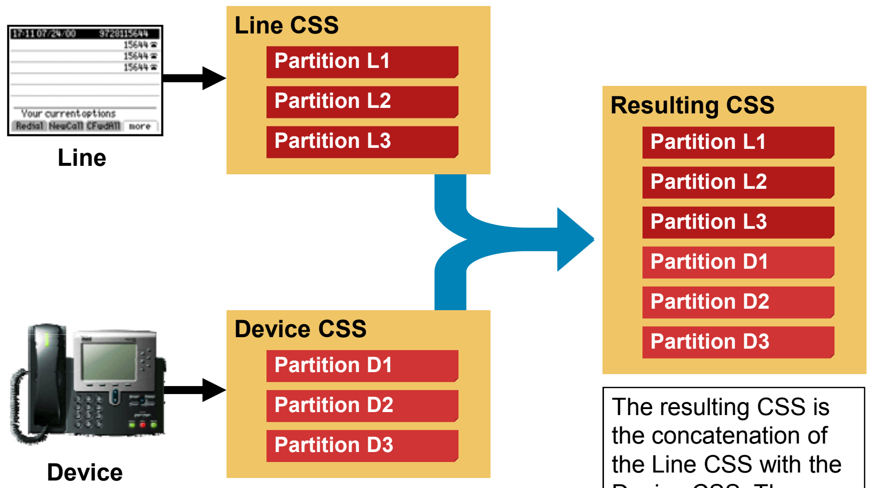
Building Classes of Service

Traditional CSS Approach for Centralized Deployments



Building Classes of Service

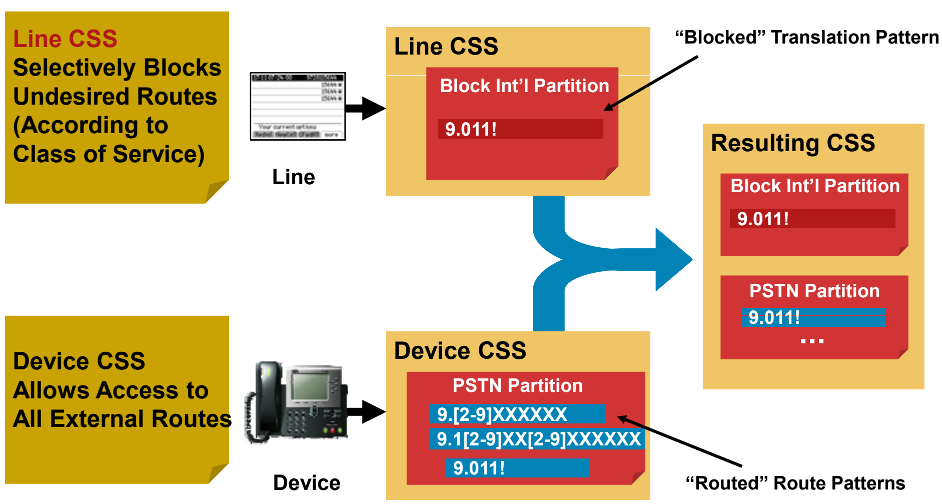
Device-Line CSS Interaction



The resulting CSS is the concatenation of the Line CSS with the Device CSS. The <none> CSS is always implied at the end.

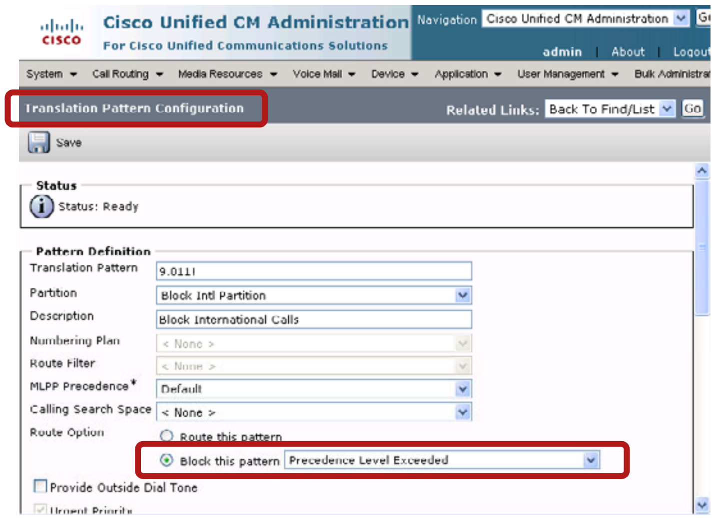
Building Classes of Service

Device-Line CSS Approach: Key Concept



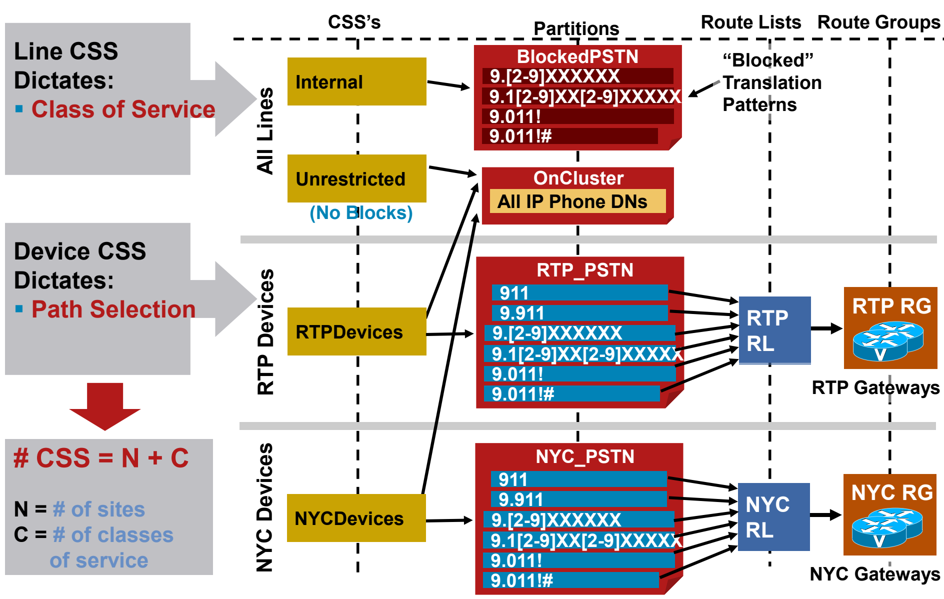
Building Classes of Service

Device-Line CSS Approach: Translation Pattern Config



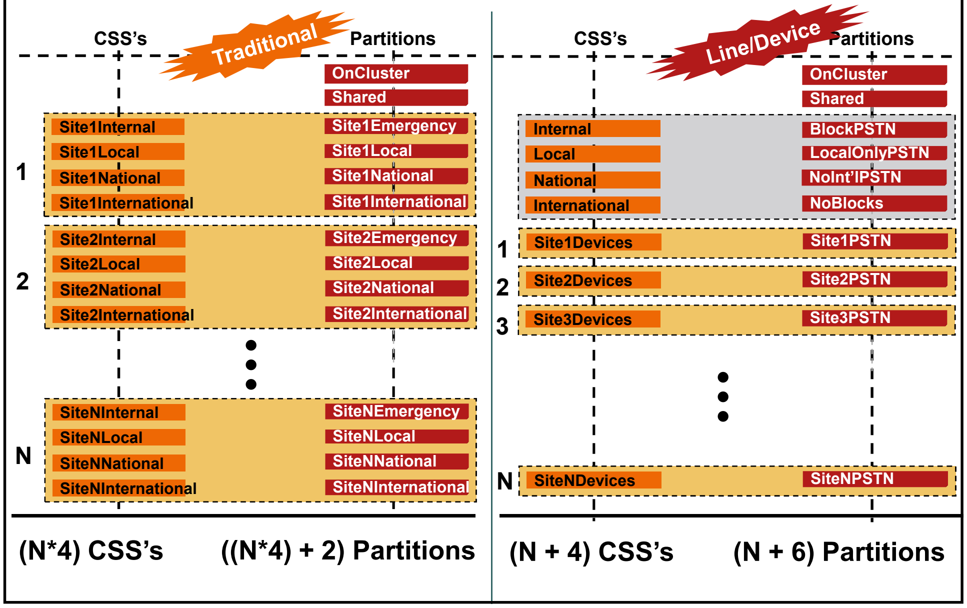
Building Classes of Service

Device-Line CSS Approach for Centralized Deployments



Building Classes of Service

Comparison of the Two Methods



Local Route Group

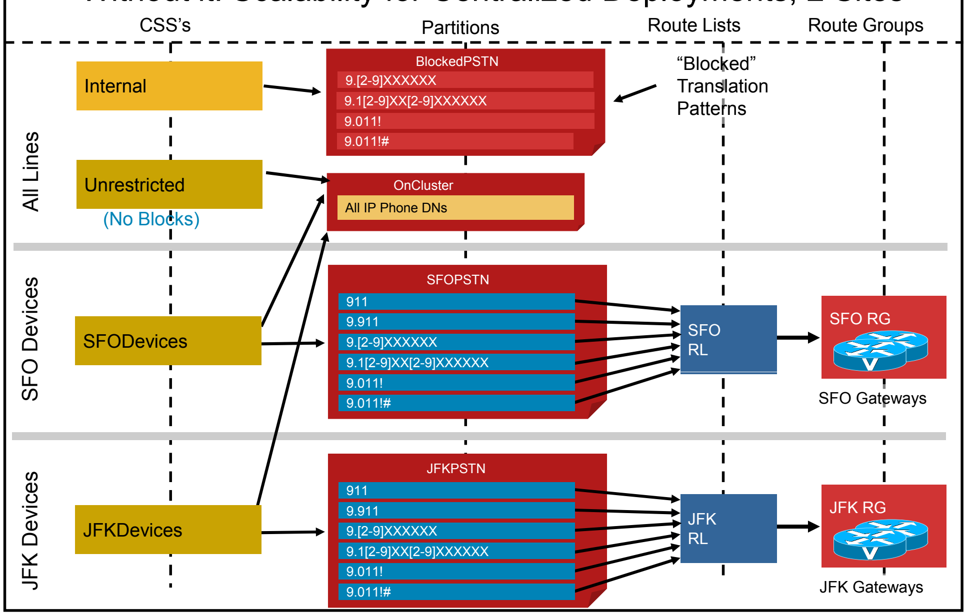
Introduced in CUCM 7.0

- Local Route Group decouples the location of a PSTN gateway from the route patterns used to access it.
- Allow the site-specificity of call routing to be established by the calling device's location (as derived from device pool).
- Different endpoints in different sites would be associated with different local route groups: they can all call the same patterns, and the calls will be routed differently, based on the caller's currently associated local route group.
- Route patterns are no longer site-specific and can be used for callers of different sites.

Local Route Group
Without it: Scalability for Centralized Deployments

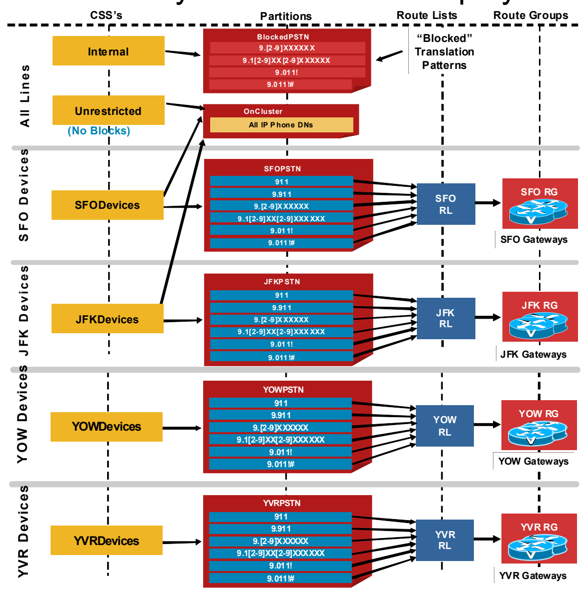
- Providing site-specific routing of patterns requires:
  - 1 route pattern per pattern per site
  - 1 partition per site (assuming flat addressing)
  - 1 calling search space per site (for call routing (device CSS), assuming the line-device approach)
  - At least 1 route list per site (more if some patterns use centralized gateways or GK)
  - At least 1 route group per site (perhaps more)
- For 1000 sites and 6 patterns, we need 6000 route patterns, 1000 partitions, 1000 CSSes, 1000 route lists, and 1000 route groups: 10,000 things

Local Route Group
Without it: Scalability for Centralized Deployments, 2 Sites



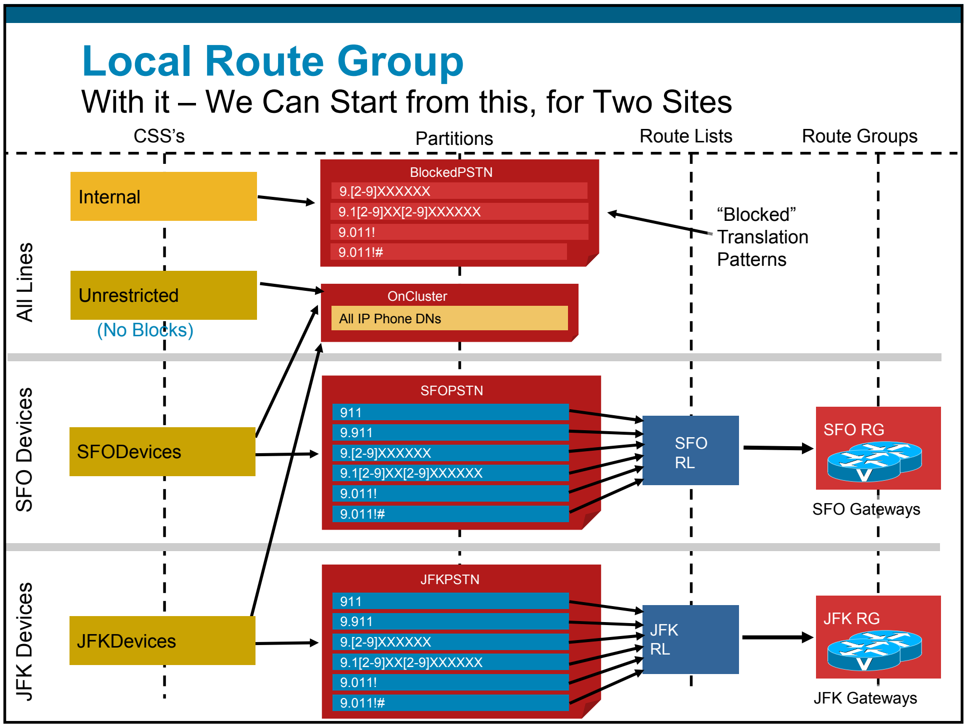
Local Route Group

Without it: Scalability for Centralized Deployments, 4 Sites

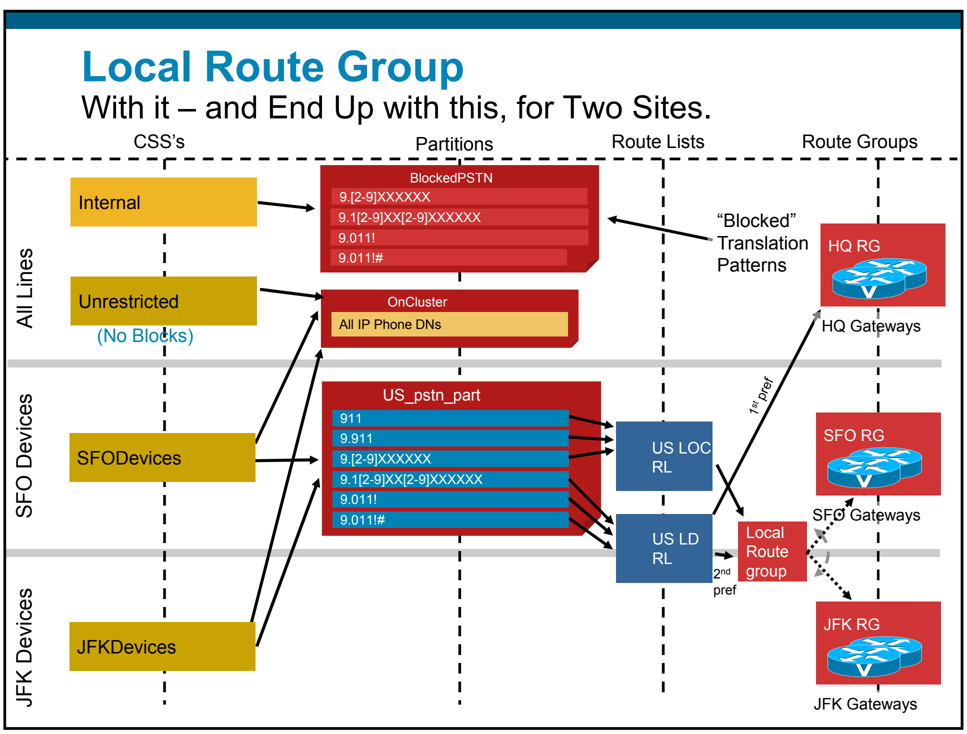


Local Route Group

Without it: Scalability for Centralized Deployments, 1000 Sites



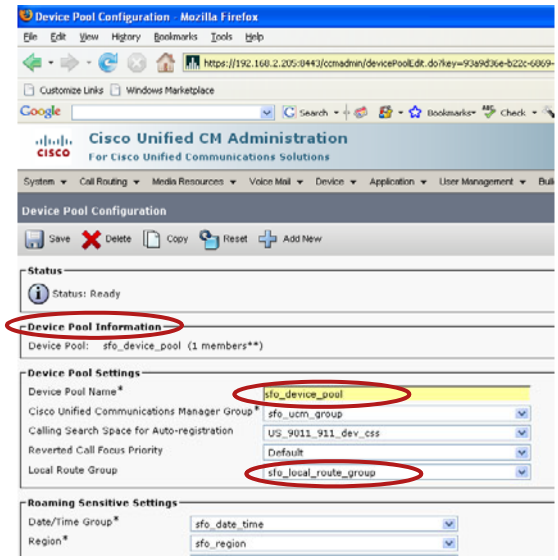
Local Route Group
With it – We Can Start from this, for Two Sites
CSS's
Partitions
Route Lists
Route Groups
All Lines
SFO Devices
JFK Devices
Internal
Unrestricted
(No Blocks)
SFODevices
JFKDevices
BlockedPSTN
9.[2-9]XXXXXX
9.1[2-9]XX[2-9]XXXXXX
9.011!
9.011!#
OnCluster
All IP Phone DNs
"Blocked" Translation Patterns
SFOPSTN
911
9.911
9.[2-9]XXXXXX
9.1[2-9]XX[2-9]XXXXXX
9.011!
9.011!#
SFO RL
SFO RG
SFO Gateways
JFKPSTN
911
9.911
9.[2-9]XXXXXX
9.1[2-9]XX[2-9]XXXXXX
9.011!
9.011!#
JFK RL
JFK RG
JFK Gateways



Local Route Group
With it – and End Up with this, for Two Sites.
CSS's
Partitions
Route Lists
Route Groups
All Lines
SFO Devices
JFK Devices
Internal
Unrestricted
(No Blocks)
SFODevices
JFKDevices
BlockedPSTN
9.[2-9]XXXXXX
9.1[2-9]XX[2-9]XXXXXX
9.011!
9.011!#
OnCluster
All IP Phone DNs
"Blocked" Translation Patterns
HQ RG
HQ Gateways
US_pstn_part
911
9.911
9.[2-9]XXXXXX
9.1[2-9]XX[2-9]XXXXXX
9.011!
9.011!#
US LOC RL
US LD RL
1st pref
2nd pref
Local Route group
SFO RG
SFO Gateways
JFK RG
JFK Gateways

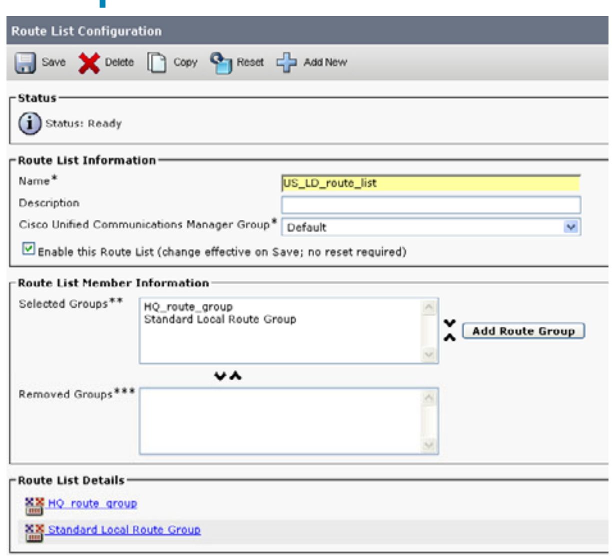
Local Route Group

- Device pool is site-specific
- Local route group is associated with device pool
- Local route group is thus associated with all devices using a given device pool: e.g.: phones, gateways



Local Route Group

- Route lists can now refer to Local Route Groups as well as regular route group
- Allows for simple local failover
- In this example, calls go to the centralized US GW (in site HQ), and fallback to the local route group

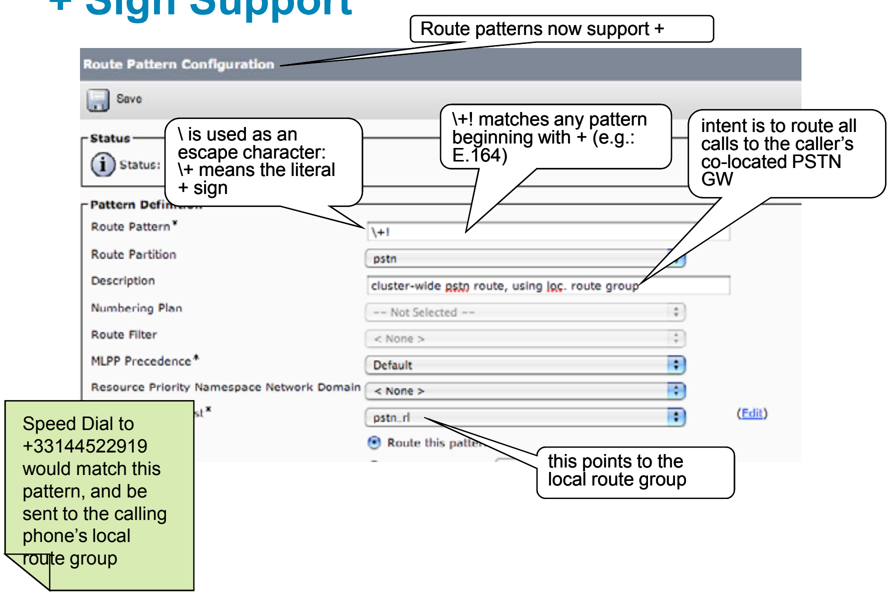


E.164: "+" Sign Support

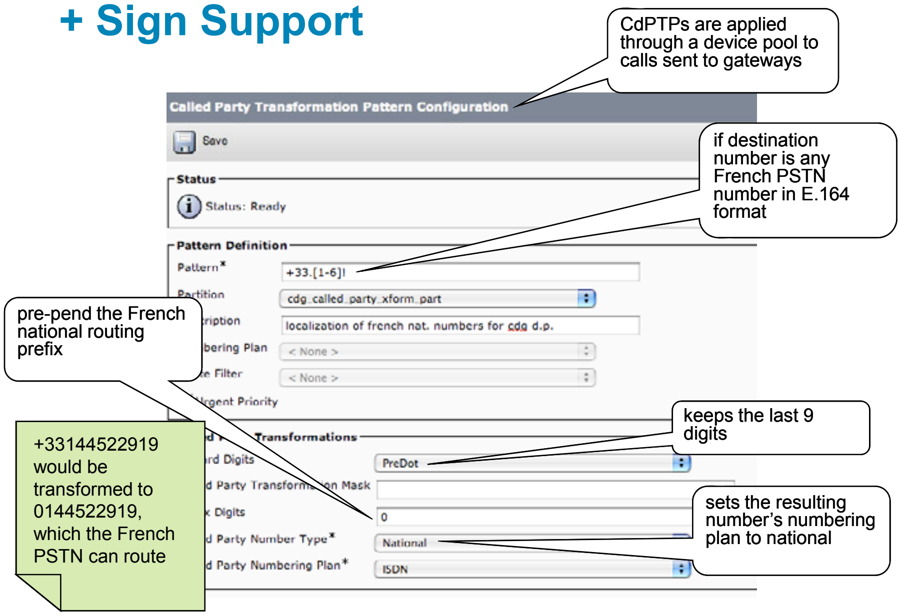
Introduced in CUCM 7.0

- E.164 support includes the use of + to "wildcard" international access codes.
- From "anywhere", by sending +33144522919, into a network that can digest it. For Example: most mobile GSM carriers, and now CUCM 7.0
- Phones do not support the + sign for keypad entry, but support the + sign in display and missed/received calls menus

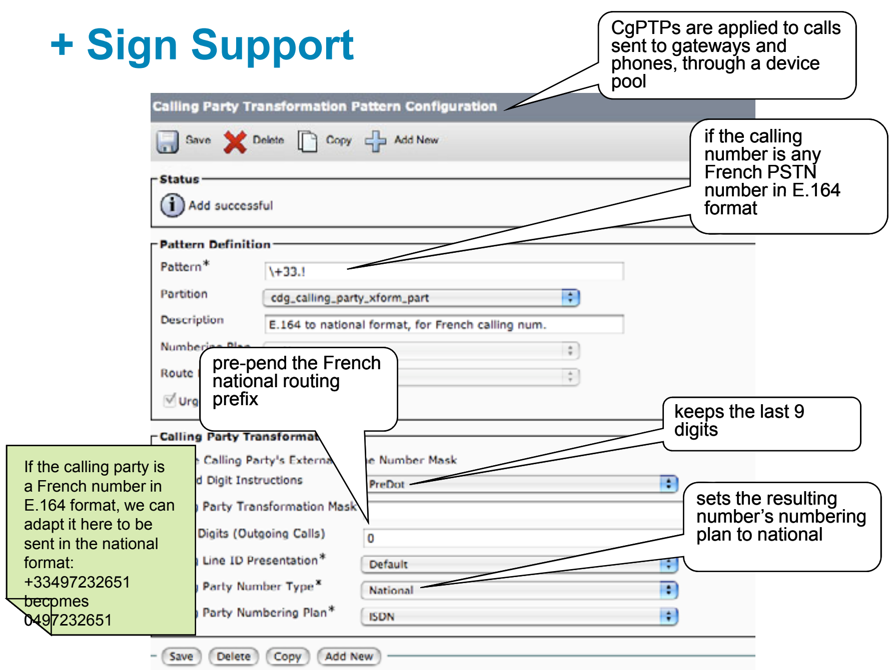
+ Sign Support



+ Sign Support



+ Sign Support



Summary: Dial Plan

Dial Plan Must-knows:

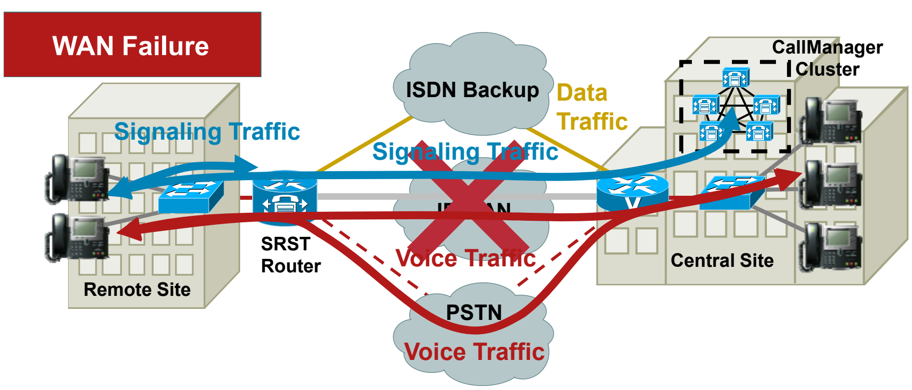
- Solid understanding of CCM partition and CSS
- Route patterns and wild cards
- Translation patterns and implications
- Route-list, Route-groups, Local Route Group, and digit manipulation checkpoints
- Calling/Called Party Translation Patterns

Q and A

Session 8

High Availability

Survivable Remote Site Telephony (SRST)



- SRST router needs minimal configuration
- Remote site IOS router take over SCCP call processing for local ip phones in case of WAN failure
- Basic call functions and features are preserved

Basic SRST Configurations

```
Call-manager-fallback
  ip source-address 10.1.1.1 port 2000 [any-match | strict-match]
  max-dn 48
  max-ephone 24
  max-conferences 8
  time-format 24
  limit-dn 7960 2
  call-forward pattern 9192345000
  dialplan-pattern 1 9192345... extension-length 4
```

Global command to enable SRST

Mandatory command to enable router to receive and process SCCP msgs

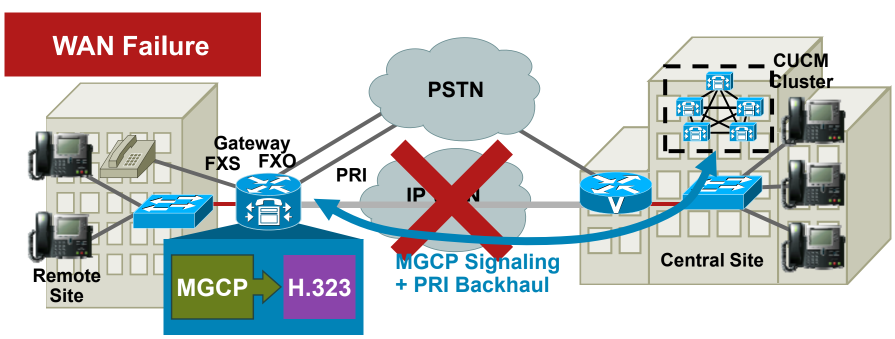
Mandatory commands which define the max. # of IP phones and directory numbers (DNs) supported by SRST. Default is “0”

Limit maximum # of DNs assignable to particular types of phone

Global prefix which maps full e.164 called number to local ip phone extensions. In this case, if the DID of an inbound call is 9192345001, it will be routed to a registered DN of 5001; also used to construct full e.164 caller ID for calls originated from SRST router

strict-match” option enables strict IP address verification of IP phones trying to register to SRST router

SRST MGCP Fallback to H.323



- Under normal operation, the gateway translates FXS/FXO signaling into MGCP and backhauls L3 PRI signaling to Cisco CallManager
- When the WAN fails, the gateway reverts to H.323 operation—SRST provides backup for the IP phones

SRST MGCP Fallback to H.323 Configuration

```
application
 global
  service alternate default
!
ccm-manager fallback-mgcp
!
dial-peer voice 1 pots
 service mgcp
 incoming called-number .
 direct-inward-dial
 port 1/0:23
!
call-manager-fallback
 ip source-address 10.1.1.1 port 2000
 max-dn 48
 max-ephone 24
 dialplan-pattern 1 9192345... extension-length 4
```

Enables gateway to fall back to default call application (H.323) when mgcpapp is not available

Allows MGCP gateway to fall back to H.323 mode

“direct-inward-dial” and “incoming called-number .” needs to be associated with a POTS dial-peer pointing to the voice-port. Otherwise, inbound PRI calls will get a secondary dial-tone

SRST Verification Commands

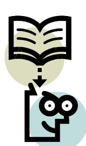
```
SRST#sh call-manager-fallback ?
  all          Show call-manager fallback details
  dial-peer    Show call-manager fallback dialpeers
  ephone-dn    Show call-manager fallback ephone-dn
  voice-port   Show call-manager fallback voice ports
  <cr>
!
SRST#sh ephone ?
  <snip>
  7960               7960 phone status
  H.H.H              mac address
  ata                ata phone status
  dn                 Dn with tag assigned
  offhook            Offhook phone status
  overlay            registered ephones with overlay DNs
  phone-load         Ephone phoneload information
  registered         Registered ephone status
  remote             non-local phones (with no arp entry)
  ringing            Ringing phone status
  summary            Summary of all ephone
  telephone-number   Telephone number assigned
  unregistered       Unregistered ephone status
  <cr>
  <snip>
```

SRST Debug Commands

```
SRST#debug ephone ?
  alarm        Enable ephone alarm message debugging
  detail       Enable ephone detail debugging
  error        Enable ephone error debugging
  keepalive    Enable ephone keepalive debugging
  loopback     Enable ephone loopback debugging
  moh          Enable ephone music-on-hold debugging
  mwi          Enable ephone mwi debugging
  pak          Enable ephone packet debugging
  qov          Enable ephone voice quality debugging
  raw          Enable ephone raw protocol debugging
  register     Enable ephone registration debugging
  state        Enable ephone state debugging
  statistics   Enable ephone statistics debugging
```

Proctor Case Studies XI:

SRST #1



Lab Sample Question

Implement the necessary configurations on R1 at Remote Site #1. Ensure that the local IP phones can call each other and send/receive calls to the PSTN even when IP connectivity to the CallManager is lost. Allow a maximum of 4 IP phones to register.

Candidate's Problem Statement

My IP phones are not registering when I shut the WAN link down, they just keep on "configuring ip"; I configured max-ephone 4 under "call-manager-fallback" too.

Proctor Case Studies XI:

SRST #1 (Cont)

Could you identify what's missing from the snippet of this "show run" on R1?

```
call-manager-fallback
 ip source-address 10.1.1.1 port 2000
 max-ephone 4
 max-conferences 8
 time-format 24
 limit-dn 7910 2
 call-forward pattern 9192345000
 dialplan-pattern 1 9192345... extension-length 4
```

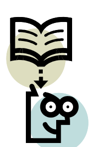
Candidate missed the "max-dn" mandatory command, by default max-dn is zero

```
call-manager-fallback
 ip source-address 10.1.1.1 port 2000
 max-ephone 4
 max-dn 10
 dialplan-pattern 1 9192345... extension-length 4
```

Proctor Case Studies XII:

SRST #2

Same Lab Sample Question



Implement the necessary configurations on R1 at Remote Site #1. Ensure that the local IP phones can call each other and send/receive calls to the PSTN even when IP connectivity to the CallManager is lost. Allow a maximum of 4 IP phones to register. Also allow a maximum of three 3-party conferences.

Candidate's Problem Statement

My phones are registered in SRST mode, I verified that I can place and receive all PSTN calls. I also have the "max-conferences 3" command. Why did I lose points in my SRST section?

Proctor Case Studies XII:

SRST #2 (Cont)

Could you identify what's missing from the snippet of this "show run" on R1?

```
call-manager-fallback
 ip source-address 10.1.1.1 port 2000
 max-ephone 4
 max-dn 16
 max-conferences 3
 time-format 24
 call-forward pattern 9192345000
 dialplan-pattern 1 9192345... extension-length 4
```

Candidate forgot to add the dual-line option in the max-dn command, resulting in single channel DN which does not have an additional channel to put the first call on hold and initialize a conference

```
call-manager-fallback
 ip source-address 10.1.1.1 port 2000
 max-ephone 4
 max-dn 16 dual-line
 max-conferences 3
 dialplan-pattern 1 9192345... extension-length 4
```

Alternate Routing

Basics

- Mechanisms that allow Unified CM to route a call through an alternate path if the preferred path is not available

  e.g.: IP path not usable/not enough bandwidth/phone unregistered—then reroute the call through the PSTN

- Alternate Routing is not triggered on events happening mid-call
- Alternate Routing for on-cluster routes:

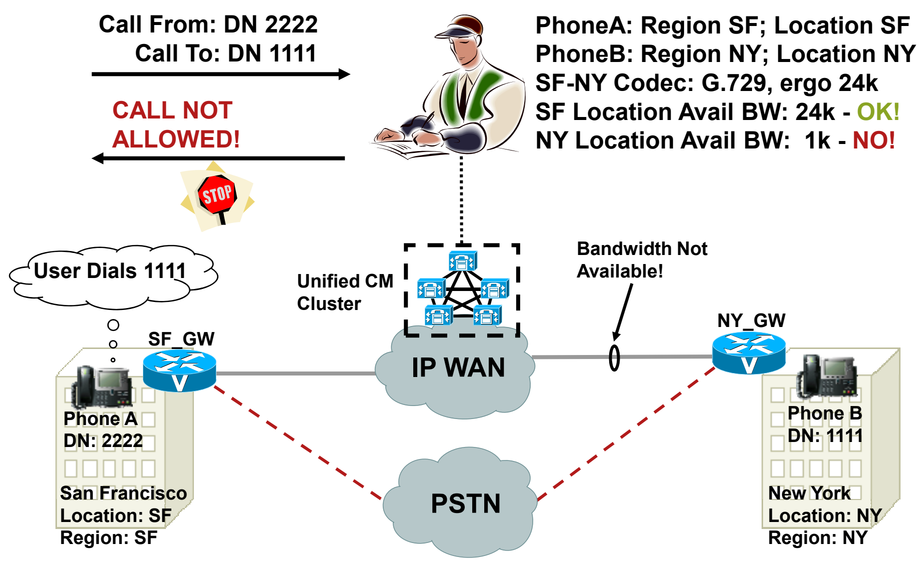
  Automated Alternate Routing (AAR) for calls to on-cluster IP endpoints when there is not enough bandwidth

  Call Forward Un-Registered (CFUR) for calls to IP endpoints when the destination is unreachable (e.g.: a remote site in SRST)

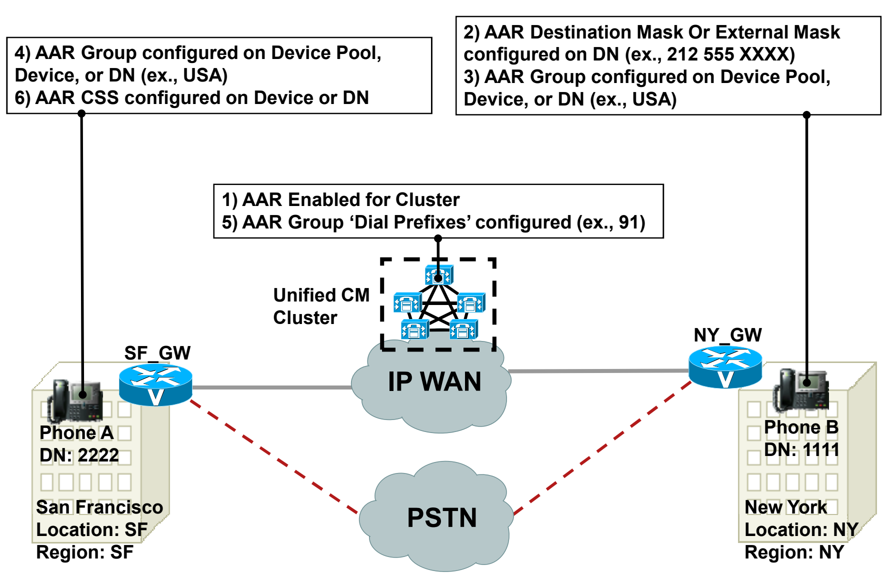
- Alternate Routing for off-cluster routes:

  The Route List/Route Groups construct provides alternate routing for external routes

Alternate Routing for Internal Routes
Call Admission Control (CAC) without AAR

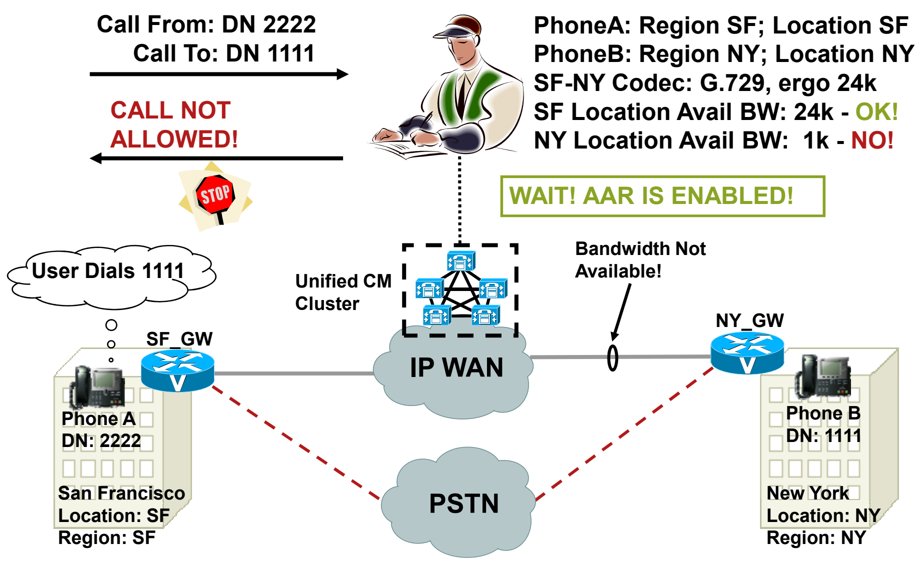


Alternate Routing for Internal Routes
AAR Components



Alternate Routing for Internal Routes

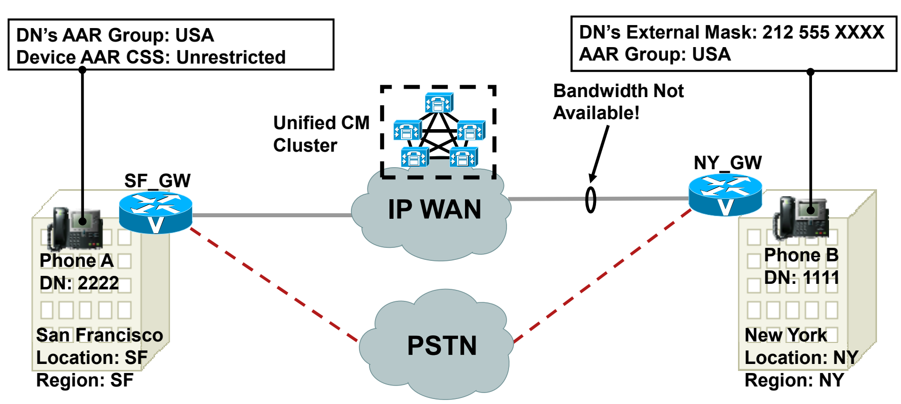
Call Admission Control (CAC) with AAR



Alternate Routing for Internal Routes

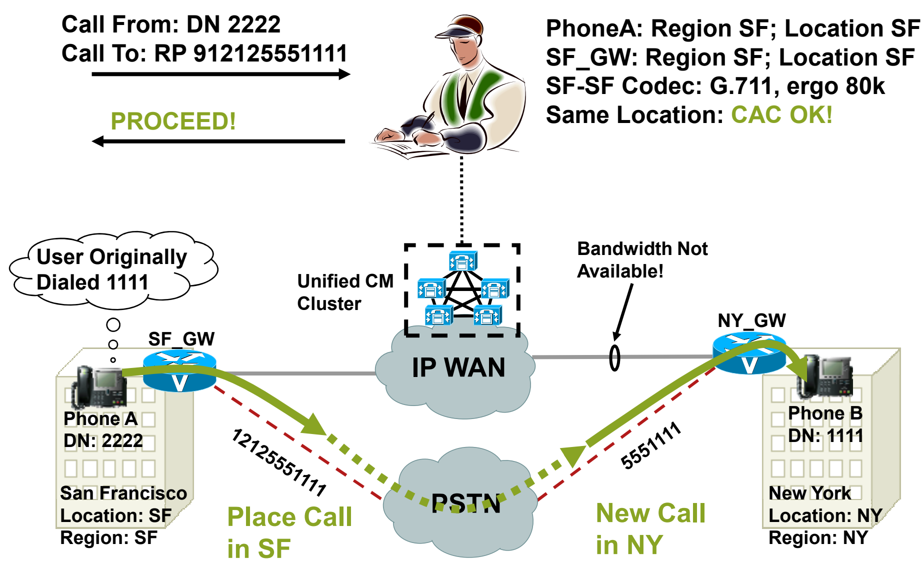
Call Admission Control (CAC) with AAR

- DN 1111’s External Phone Number Mask = 212555XXXX
  - => New Destination = 2125551111
- Within AAR Group ‘USA’, Prefix Dialing = 91
  - => New Destination = 912125551111
- AAR CSS of PhoneA Contains RP 9.1[2-9]XX[2-9]XXXXXX which Points to SF_GW
- Call Is Now Attempted From 2222, To 912125551111, Via the SF_GW



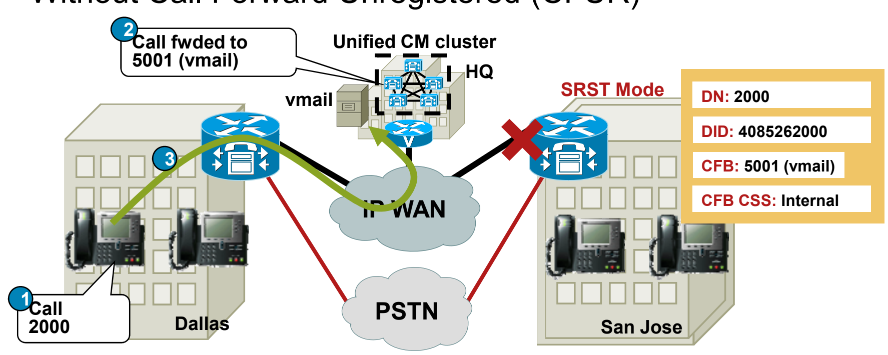
Alternate Routing for Internal Routes

Call Admission Control (CAC) with AAR



Alternate Routing for Internal Routes

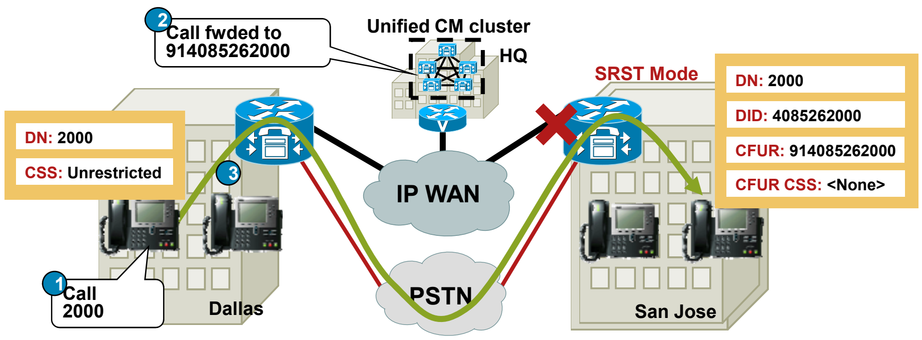
Without Call Forward Unregistered (CFUR)



- Prior to CFUR, Call Forward Busy (CFB) used when phones unregistered
- Issue: Phone is still able to receive calls via PSTN

Alternate Routing for Internal Routes

With CFUR



- Reroutes calls to unregistered DN’s using number specified in “Call Forward Unregistered” (CFUR) field
- Number in CFUR field needs to include PSTN access codes
- If leave CFUR CSS as <None>, calling party’s CSS is used (Calling phone’s class of service must allow call)

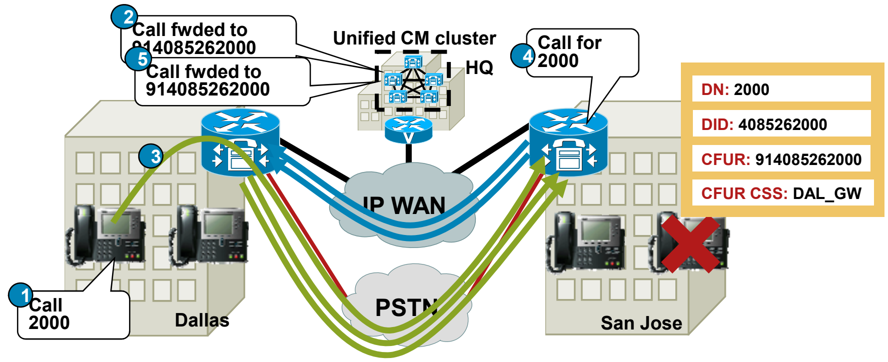
Alternate Routing for Internal Routes

CFUR Caveats

- CFUR Destination number same irrespective of calling phone’s PSTN dialing requirements based on calling site
  - An issue for previous example if say, calling phone was in Europe: the dialed number should be 0 00 1 408 526 2000
- CFUR CSS same irrespective of calling phone’s dial plan. Ie., not able to use different GW based on calling site
  - As noted, if CFUR CSS is left to <none>, calling phone’s CSS is used and the Calling phone’s class of service must allow call. NOT A PROTECTED FEATURE!!!!
- BEWARE OF LOOPS: What happens if phone is “merely” un-registered (site not in SRST-mode)?
  - GWs should not be allowed to place calls to number ranges that deliver calls to the GW itself. Next slide illustrates this issue

Alternate Routing for Internal Routes

CFUR Caveats



- CFUR is invoked whenever DN is unregistered, including when EM is logged out or the phone is unplugged
- CFUR CSS cannot be expected to be able to avoid loops in this situation

Alternate Routing for Internal Routes

CFUR Caveats

- Set service parameter to 1 (or 2) to limit loops **(value may need to be higher if forwarding “chains” are used for voicemail or other applications)**
- When looping call is dropped, caller hears fast-busy

Summary: High Availability

Be Familiar with the Following

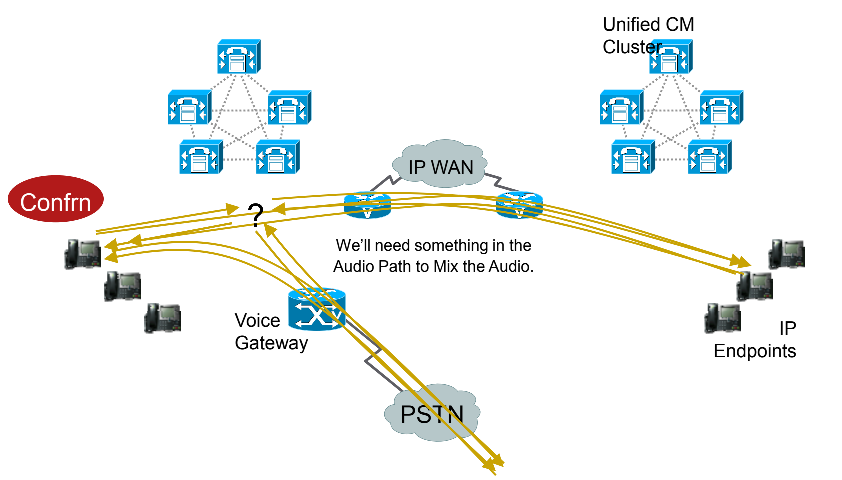
- SRST baseline and advanced configurations
- SRST fall back to H.323 gateway
- SRST show and debug commands
- Location-based CAC
- AAR configurations and components
- CFUR configurations

Q and A

Session 9

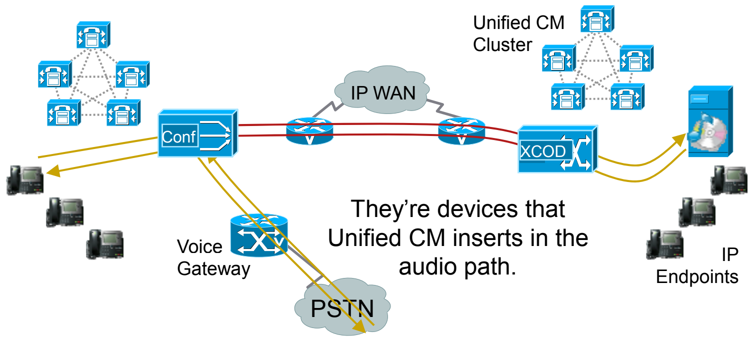
Media Resources

What Are Media Resources?



What Are Media Resources?

- Cisco Unified CM uses them to change, create and modify or avoid having to modify Real Time Protocol (RTP) streams.



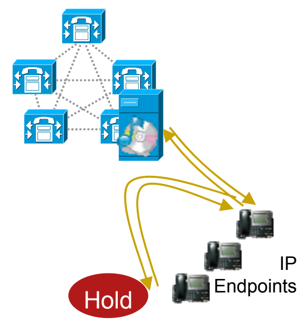
- Cisco Unified CM Uses them to cope with certain signaling scenarios that occur with H.245, Session Initiation Protocol (SIP) and other protocols.

Different Types of Media Resources

- Music on Hold **(MoH)** Resources
- Conferencing Resources
- Annunciator **(ANN)** Resources
- Media Termination Point **(MTP)** and Transcoding Resources **(XCODE)**

Music on Hold

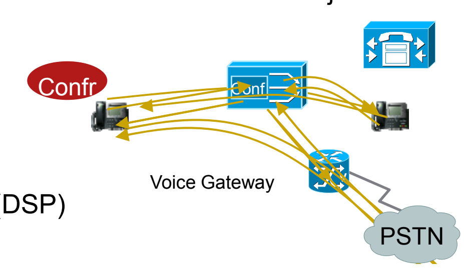
- MoH resources within Cisco Unified Communications Manager are provided (logically) solely by IP Voice Media Streaming App (IPVMS App).
- IPVMSA is a service that runs on a Unified CM server: software based.
- It is a single service that provides multiple types of logical entities, each of which registers with Unified CM. Each logical device may be enabled and disabled independently.



Conference Resources

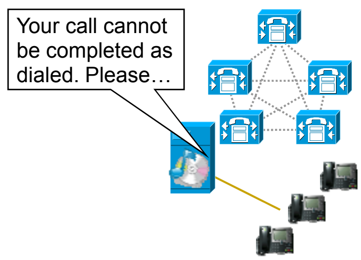
Conference Resource Uses

- Conference Bridges (CFB) provide a way for Unified CM to mix multiple audio or video streams into one, for transmission to a participant or "conferee"
- Allow for multiple parties to dial one number and all be joined to a single bridge
- Let a subscriber add his or her own selected parties to a call
- Can be software based (IPVMS App) or hardware (DSP)

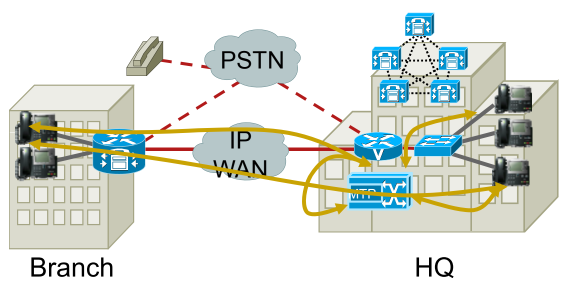


Annunciator Media Resources

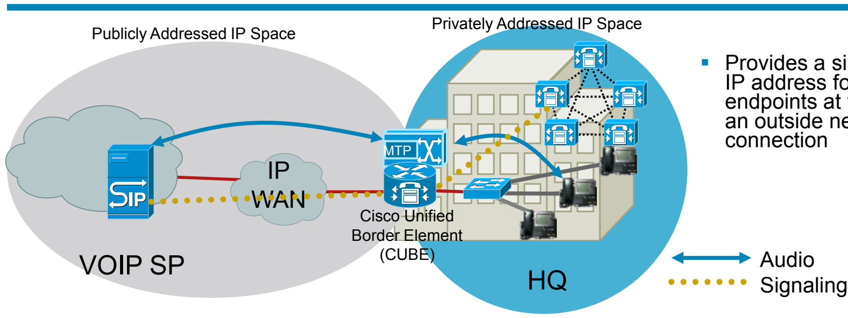
- Annunciator (ANN) Resources play announcements and tones to users.
- Annunciator plays call progress tones to off-net parties when we cannot signal those tones out of band.
- ANN is software based



MTP Media Resources

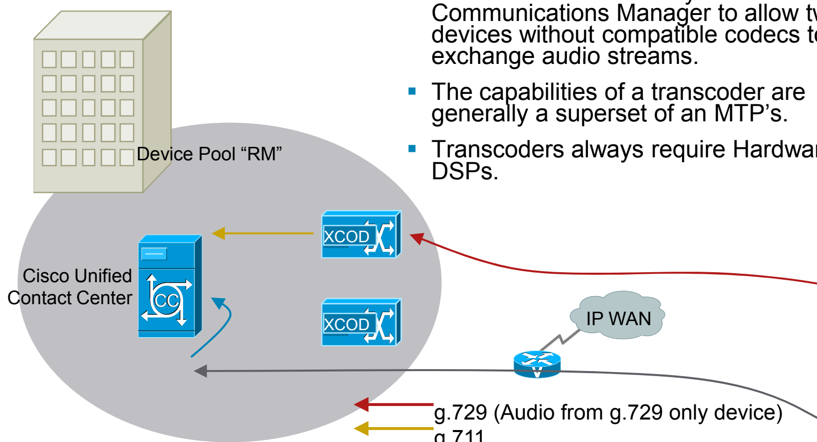


- An MTP anchors the RTP stream.
- Provides supplementary services for devices that cannot support H.245 EmptyCapabilitySet, RFC2833 DTMF



- Provides a single IP address for all endpoints at the site to an outside network connection

Transcoding Media Resources



- Transcoders are used by Cisco Unified Communications Manager to allow two devices without compatible codecs to exchange audio streams.
- The capabilities of a transcoder are generally a superset of an MTP's.
- Transcoders always require Hardware DSPs.

Media Resources Providers

- Unified CM IP Voice Media Streaming App (IPVMSApp)
- Integrated Services Routers (ISR) and Packet Voice Data Module 2 (PVDM2) Enhanced Conferencing and Transcoding and the IOS Software MTP
- Communications Media Module (CMM) with Advanced Conferencing and Transcoding module (ACT)
- Older Hardware such as 6608, VG200, NM-HDV, etc

IP Voice Media Streaming App (IPVMS App)

- The Media Streaming App Service provides:

Music on Hold—both Multicast and Unicast

	Music On Hold Server Name ^	Description	Device Pool	Status	IP Address
☐	MOH_2	MOH_ps-611	pslow-imaginary-HQ	Registered with ps-611	14.48.69.61

Annunciator

	Name ^	Description	Device Pool	Status	IP Address
☐	ANN_2	ANN_ps-611	pslow-imaginary-HQ	Registered with ps-611	14.48.69.61

Conference Resources

	Conference Bridge Name ^	Description	Device Pool	Status	IP Address
	CFB_2	CFB_ps-611	pslow-imaginary-HQ	Registered with ps-611	14.48.69.61

Media Termination Point Resources

		Name ^	Description	Device Pool	Status	IP Address	Copy
		MTP_2	MTP_ps-611	pslow-imaginary-HQ	Registered with ps-611	14.48.69.61	Not Allowed

- All of IPVMSApp’s services are run in software. They affect and are affected by the server’s CPU and Disk I/O usage.
- You can have as many IPVMS App server/services as you have MCS servers in your cluster.

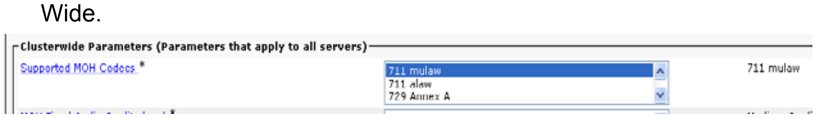
IP Voice Media Streaming App (IPVMS App)

Supported Codecs

- IPVMS MoH Can generate RTP streams in

  Wideband
  G.711a / G.711µ
  g.729a

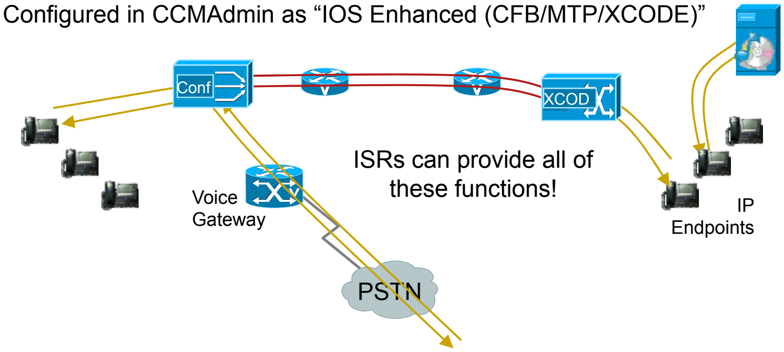
- For MoH, Each codec is enabled independently in IPVMS service Parameters and only µ-Law is enabled by default. The setting is Cluster-Wide.



- IPVMS ANN support for Wideband, G.711a, G.711µ and g.729a is always enabled.
- The IPVMS CFB supports G.711a, G.711µ and Wideband and does support a mixed-mode conference involving those codecs.

PVDM2 in IOS Voice Gateways

- This family of devices can provide:
  - Conferencing and Secure Conferencing
  - Transcoding / Media Termination
  - MoH from flash (covered at the end of this presentation)
- Performance of these services is minimally impacted by load – Mixing and transcoding done in hardware.
- Configured in CCMAdmin as "IOS Enhanced (CFB/MTP/XCODE)"



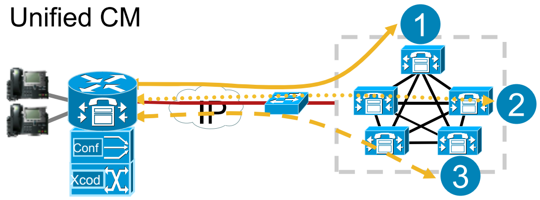
PVDM2 in IOS Voice Gateways

Supported Codecs

- The ISRs can support transcoding of the following codecs:
  - g711a/u-law
  - G.723.1 5300 bps
  - G.723.1 6300 bps
  - G.729ab
  - G.729a
  - G.729b
  - G.729
  - GSMAMR
  - ILBC codec
  - Pass-through (no transcoding)
- When used with Unified CM, ISRs can only be used to transcode from G.711 to low-bitrate codecs
- G.729 to G.729 MTP functionality is provided by the IOS Software MTP, which does not use DSPs

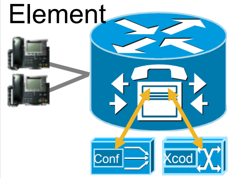
PVDM2 Configurations

Unified CM



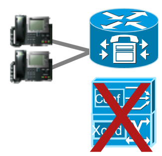

```
sccp local FastEthernet0/0
sccp ccm 10.1.1.1 identifier 1 version 6.0+
sccp ccm 10.1.1.2 identifier 2 version 6.0+
sccp ccm 10.1.1.3 identifier 3 version 6.0+
!
sccp ccm group 988
  associate ccm 1 priority 1
  associate ccm 2 priority 2
  associate ccm 3 priority 3
  …
  keepalive retries 5
  switchover method immediate
  switchback method immediate
  switchback interval 15
```

Unified CME, Unified Border Element




```
telephony-service
  ip source-address 20.1.1.1 port 2000
  sdspfarm units 1
  sdspfarm transcode sessions 16
  sdspfarm tag 1 MTP000f23cd6100
!
sccp local Vlan10
sccp ccm 20.1.1.1 identifier 1
sccp
!
sccp ccm group 1
  associate ccm 1 priority 1
  …
```

Unified SRST



Conference, Transcoding and MTP Resources Are Not yet Available During SRST Mode

DSP Conference Configuration Example

```
voice-card 1
  dsp services dspfarm
!
sccp local FastEthernet0/0
sccp ccm 10.1.1.1 identifier 1 version 6.0+
sccp ccm 10.1.1.2 identifier 2 version 6.0+
sccp ccm 10.1.1.3 identifier 3 version 6.0+
sccp ip precedence 3
sccp
!
dspfarm profile 10 conference
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 codec g729r8
 codec g729br8
 maximum sessions 6
 associate application SCCP
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate ccm 2 priority 2
 associate ccm 3 priority 3
 associate profile 10 register CFB123456789966
!
<some commands such as failover omitted here>
```

- **Enables DSPFARM feature for the hardware DSP module**
- **Defines local interface for SCCP**
- **Defines CUCM for registration. Associates ID to each CCM**
- **Enables SCCP globally**
- **Defines a conference bridge instance for the DSPFARM**
- **CODEC capability for this conference profile**
- **Maximum number of simultaneous conference sessions**
- **Associate profile to SCCP**
- **Defines a logical CCM group, add members into the group and associate priorities**
- **Assciate the conference bridge profile to register to CCM group**

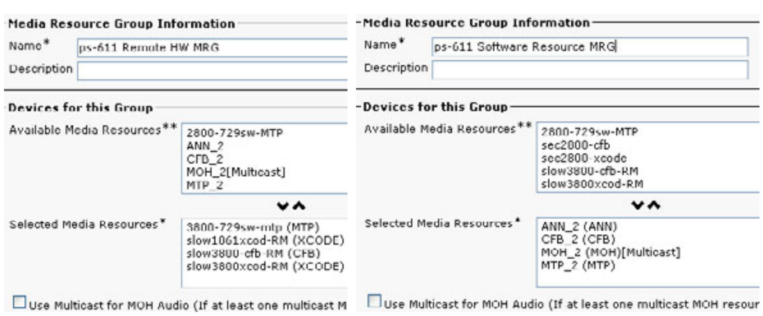
DSP Transcoding Configuration Example

```
voice-card 1
  dsp services dspfarm
!
sccp local FastEthernet0/0
sccp ccm 10.1.1.1 identifier 1 version 6.0+
sccp ccm 10.1.1.2 identifier 2 version 6.0+
sccp ccm 10.1.1.3 identifier 3 version 6.0+
sccp ip precedence 3
sccp
!
dspfarm profile 20 transcode
 codec g711ulaw
 codec g711alaw
 codec g729r8
 codec g729br8
 maximum sessions 10
 associate application SCCP
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate ccm 2 priority 2
 associate ccm 3 priority 3
 associate profile 20 register MTP123456789966
!
<some commands such as failover omitted here>
```

- Enables DSPFARM feature for the hardware DSP module
- Defines local interface for SCCP
- Defines CUCM for registration. Associates ID to each CCM
- Enables SCCP globally
- Defines a transcoder instance for the DSPFARM
- CODEC capability for this transcoder profile
- Maximum number of simultaneous transcoding sessions
- Associate profile to SCCP
- Defines a logical CCM group, add members into the group and associate priorities
- Assciate the transcoder profile to register to CCM group

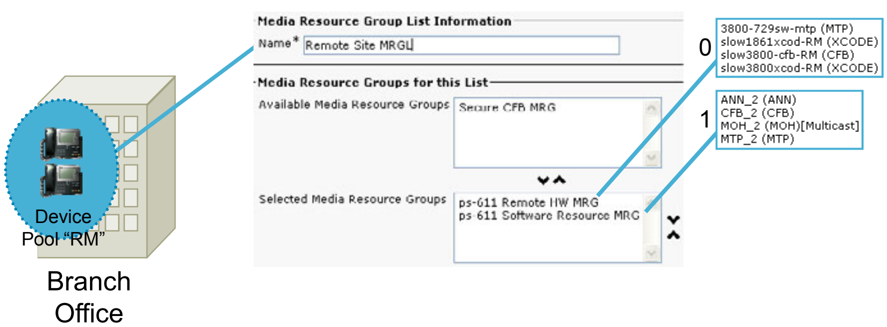
Media Resource Management with MRG and MRGL

- A Media Resource Group (MRG) is used to define a set of **same-priority** resources.

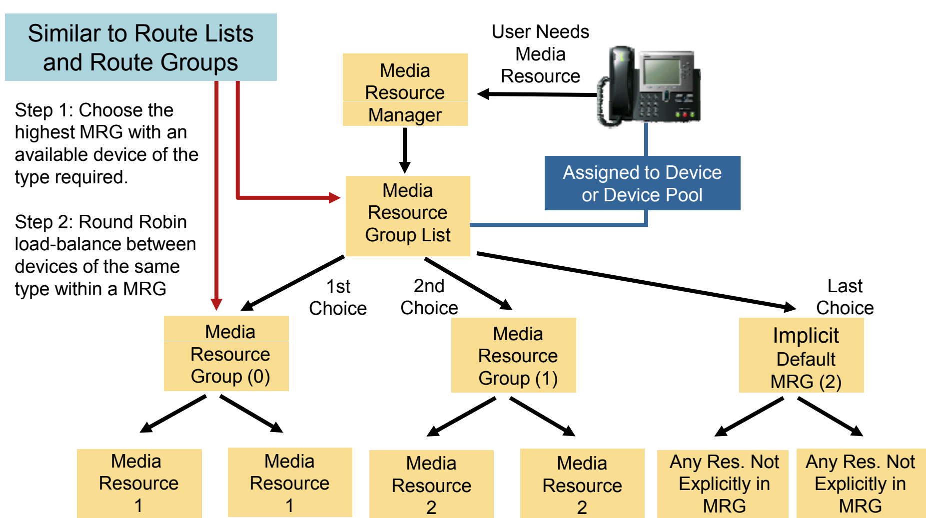


Media Resource Management with MRG and MRGL

- A Media Resource Group List is used to define a **prioritized** list of MRGs
- The MRGL is then used to associate those resources with a Station or Trunk device or Device Pool

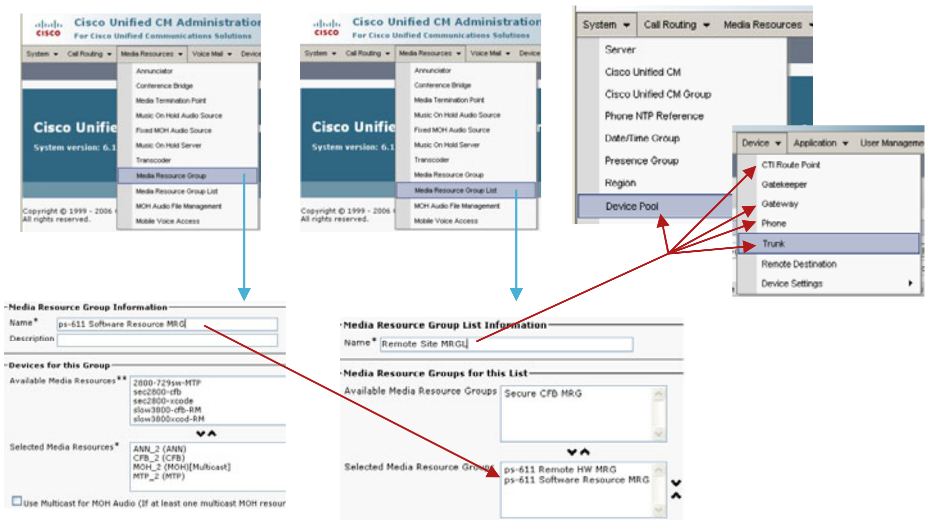


Unified CM Media Resource Selection



Note: Default Media Resource Group is the last choice of the MRGL (Contains devices not in any MRG)

MRG and MRGL Configuration and Assignment



MRG and MRGL Configuration and Assignment

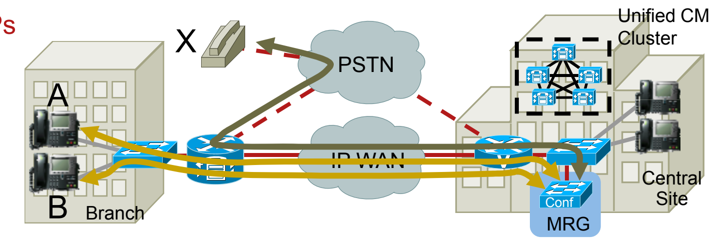
- When a Media Resource requires a Media Resource, the MRGL comes from it’s device pool.
- The device-level MRGL takes precedence over the device-pool level MRGL, if both are assigned.

Conference Resources

Centralized vs. Distributed DSPs

Centralized DSPs

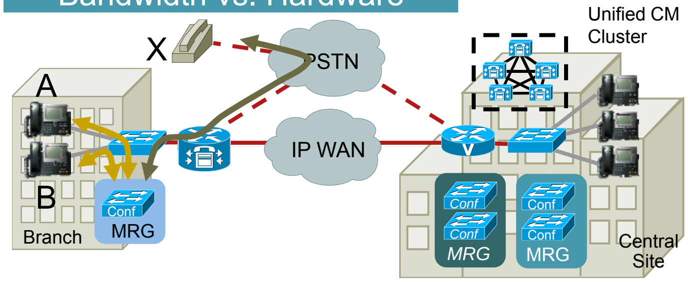
- Low-speed WAN links
- Remote sites are limited to the amount of bandwidth provisioned for conferencing



Bandwidth vs. Hardware

Distributed DSPs

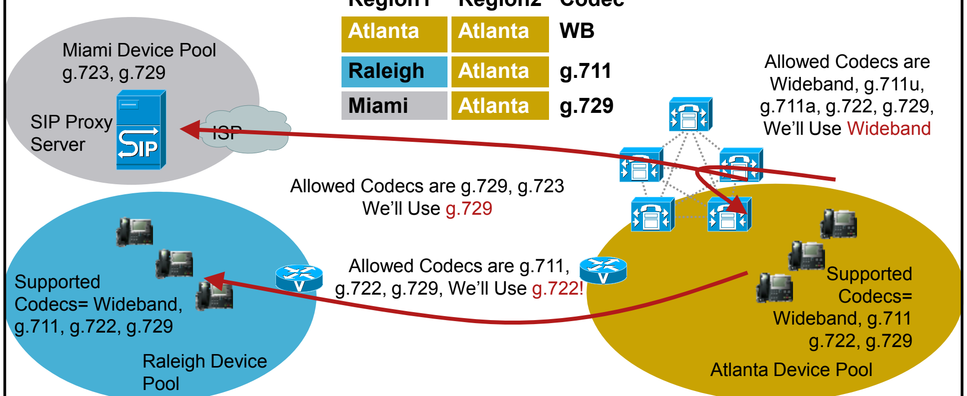
- Distribute CFBs and VCBs to large sites
- Endpoints use their local resource
- Single site calls stay local



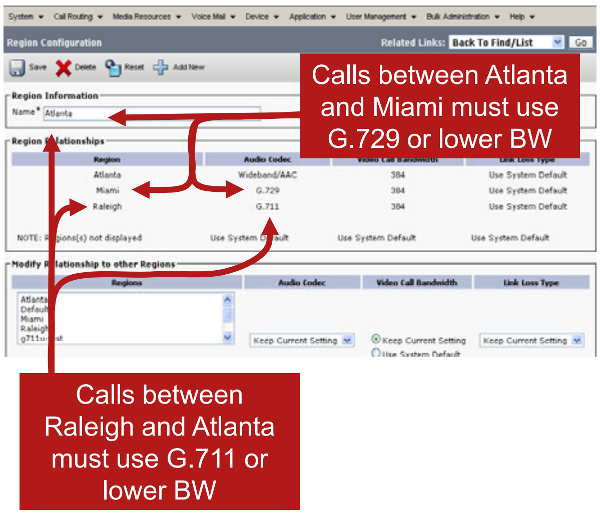
What Are Regions?

- Regions are used to **filter** the list of Codecs two devices can use to communicate
- A region **pair** defines the relationship.

Region1	Region2	Codec
Atlanta	Atlanta	WB
Raleigh	Atlanta	g.711
Miami	Atlanta	g.729



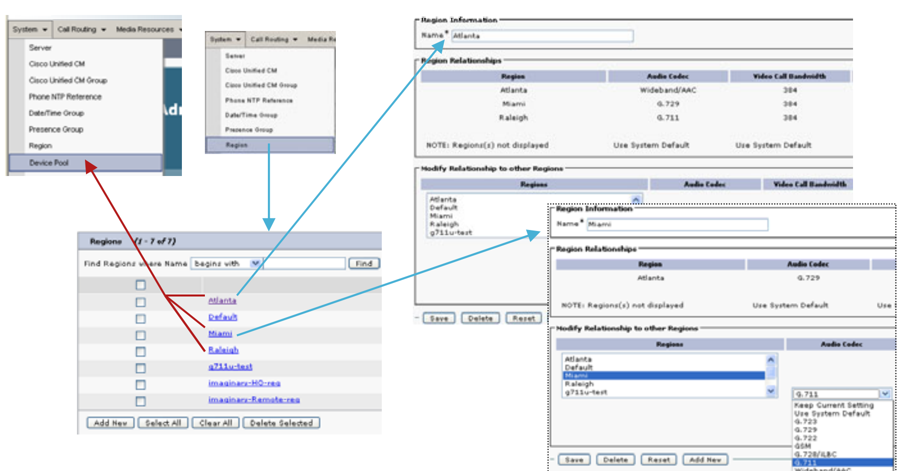
Interpreting Regions Configuration



- Each codec has a corresponding BW value within Unified CM.
- We can see what a call from Miami to Atlanta would use, but not from Miami to Raleigh.
- The relationships are Bi-Directional.

Regions Configuration

- Regions are an attribute assigned to device pools and configured relative to each other



The Barge Feature

- Call barge allows a user to conference his or herself with users in a normal call on a shared line. To Barge a call from a phone, that phone must indicate the shared line as remote-in-use.
- There is an internal bridge in Cisco Unified IP Phone 7940/60, 7941/61 and 7970/71 IP Phones.
  - When a Cisco Unified 7940 or 7960 is set to use encryption, the internal bridge is disabled. It cannot support simultaneous encrypted RTP streams.
- Requirements:
  - Subscriber must be using a line that is shared with your phone
  - You must have privacy disabled on the barged phone for the DN you are attempting to join a call on

Barge

- You must add a Barge softkey to a device's phone template for it to use this feature
- When the Barge feature is invoked using the Barge softkey, a Barge call is set up using the internal CFB present on the target phone
  - If the remote party (party that the Barge target was talking to) releases the call after it's been barged, the call terminates for all three users
  - If the Barge target holds the call, the Barge initiator's call will drop

Adding Yourself to a Shared-Line Call

Depending on how your phone is configured, you can add yourself to a call on a shared line using either Barge or cBarge.

If you want to...	Then...
See if the shared line is in use	Look for the remote-in-use icon next to a red line button.
View details about current calls on the shared line	Press the red line button for the remote-in-use line. All non-private calls appear in the call activity area of the touchscreen.
Add yourself to a call on a shared line using the Barge softkey	1. Highlight a remote-in-use call. 2. Press Barge. (You may need to press the more softkey to display Barge.) Other parties hear a beep tone announcing your presence.

cBarge

- You must add a cBarge softkey to a device’s phone template for it to use this feature
- When the cBarge feature is invoked using the cBarge softkey, a conference call using a CFB Resource is set up between the three involved parties
  - The cBarge target is treated as the conference initiator and can add more participants to the conference if he or she wishes
  - If any one party releases their call in a (three-party) cBarge scenario, the CFB is released and the remaining participants are joined on a point-to-point call
  - The cBarge target can hold and resume the call without dropping anyone

Summary: Media Resources

Be Familiar with the Following:

- Different types of SW/HW CFB, MTP, and transcoders, and their configurations
- MoH: Unicast and Multicast configurations
- Annunciator configuration
- Media resource group, media resource group list
- Regions
- Other media resource related features such as barge

Q and A

Session 10

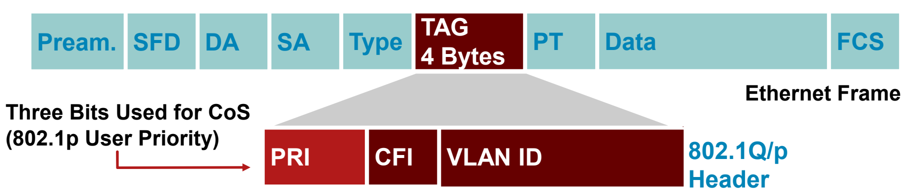
QoS

QoS Agenda

- Voice QoS Tools
- Campus QoS Considerations
- WAN QoS Considerations

Classification Tools—Layer 2

Ethernet 802.1Q Class of Service

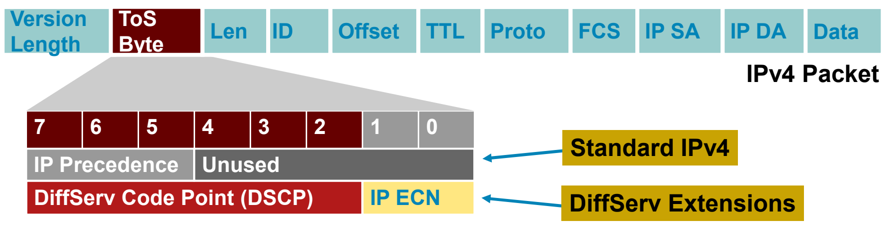


- 802.1p user priority field also called Class of Service (CoS)
- Different types of traffic are assigned different CoS values
- CoS 6 and 7 are reserved for network use

CoS	Application
7	Reserved
6	Routing
5	Voice
4	Video
3	Call Signaling
2	Critical Data
1	Bulk Data
0	Best Effort Data

Classification Tools—Layer 3

IP Precedence and DiffServ Code Points



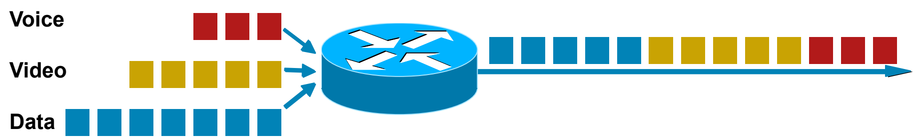
- IPv4: Three most significant bits of ToS byte are called IP Precedence (IPP)—other bits unused
- DiffServ: Six most significant bits of ToS byte are called DiffServ Code Point (DSCP)—remaining two bits used for flow control
- DSCP is backward-compatible with IP precedence

Classification and Marking Design

QoS Baseline Marking Recommendations

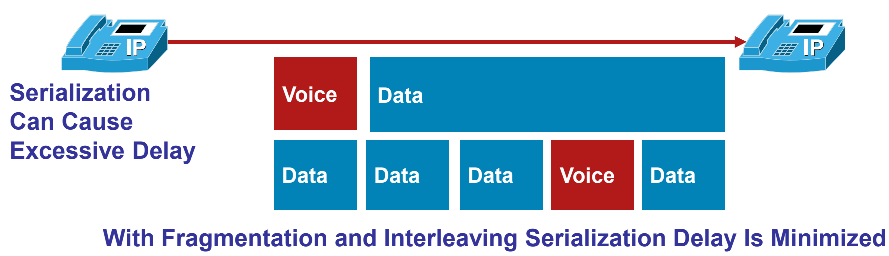
Application	L3 Classification			L2
	IPP	PHB	DSCP	CoS
Routing	6	CS6	48	6
Voice	5	EF	46	5
Video Conferencing	4	AF41	34	4
Streaming Video	4	CS4	32	4
Mission-Critical Data	3	AF31	26	3
Call Signaling	3	CS3	24	3
Transactional Data	2	AF21	18	2
Network Management	2	CS2	16	2
Bulk Data	1	AF11	10	1
Scavenger	1	CS1	8	1
Best Effort	0	0	0	0

Scheduling Tools
Queuing Algorithms



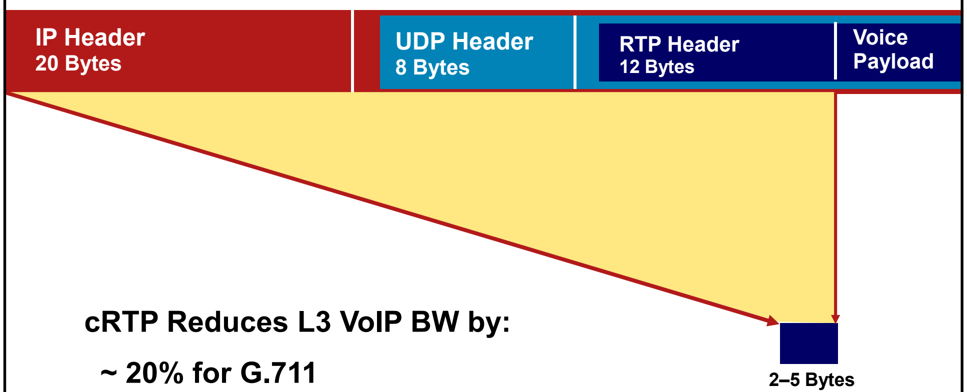
- Congestion can occur at any point in the network where there are speed mismatches
- Routers use Cisco IOS-based software queuing
  - Low-Latency Queuing (LLQ) used for highest-priority traffic (voice/video)
  - Class-Based Weighted-Fair Queuing (CBWFQ) used for guaranteeing bandwidth to data applications
- Cisco Catalyst switches use hardware queuing

Link-Specific Tools
Link-Fragmentation and Interleaving



- Serialization delay is the finite amount of time required to put frames on a wire
- For links ≤ 768 kbps serialization delay is a major factor affecting latency and jitter
- For such slow links, large data packets need to be fragmented and interleaved with smaller, more urgent voice packets

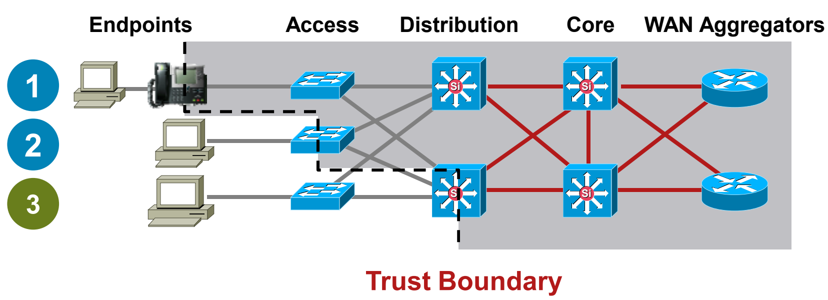
Link-Specific Tools
IP RTP Header Compression



cRTP Reduces L3 VoIP BW by:

- ~ 20% for G.711
- ~ 60% for G.729

Campus QoS Considerations
Establishing Trust-Boundaries



1. Optimal Trust Boundary: Trusted Endpoint
2. Optimal Trust Boundary: Untrusted Endpoint
3. Sub-Optimal Trust Boundary

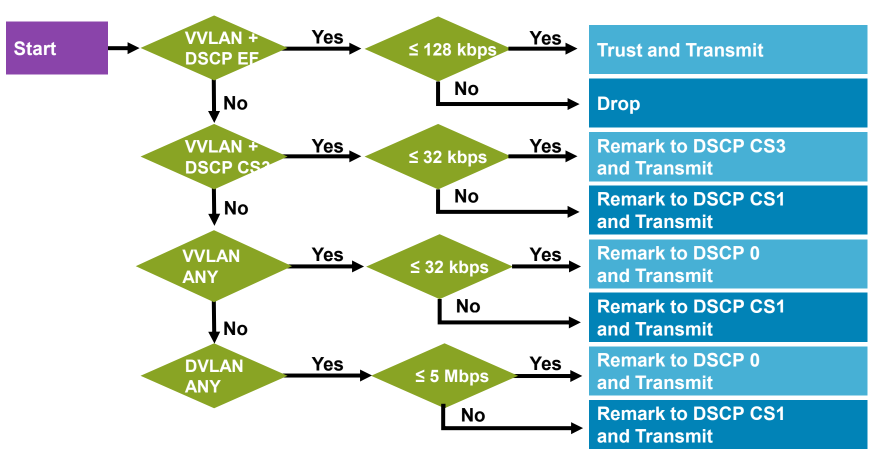
Campus QoS Considerations

Conditional-Trust Boundary Extension and Operation

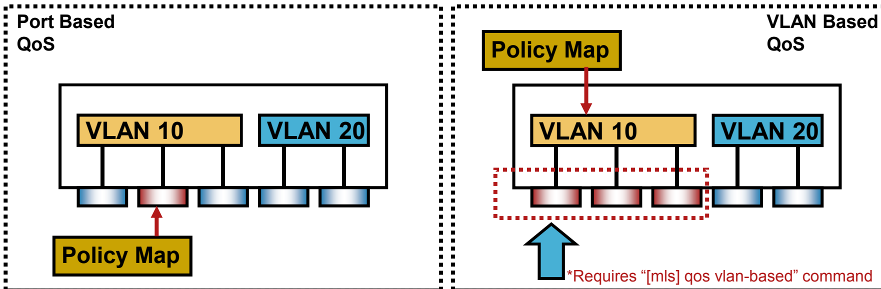
1. Switch and Phone Exchange CDP; Trust Boundary Is Extended to IP Phone
2. Phone Sets CoS 5 for VoIP and CoS 3 for Call-Signaling Traffic
3. Phone Rewrites CoS from PC Port to 0
4. Switch Trusts CoS from Phone and Maps CoS → DSCP for Output Queuing

Access-Edge Trust Models

Modular QoS CLI Based Model



Campus QoS Design Considerations
Port-Based v. VLAN-Based QoS



With **Port Based QoS**, QoS policies are applied to a physical interface. The policy manages traffic only the port the policy is applied.

With **VLAN Based QoS**, the QoS policy is applied to the VLAN interface. Traffic through all associated Switch ports is managed by that policy.

By default, Catalyst switches will refer to policies assigned to the physical port. Ports defined as a "switchport" can be told to use the policy attached to its parent VLAN interface – this is known as VLAN-based QoS

Cisco Catalyst 3750 QoS Design
Enabling QoS

```
CAT3750#show mls qos
QoS is disabled                        ! By default QoS is disabled


CAT3750#configure terminal
Enter configuration commands, one per line.  End with CNTL/Z.
CAT3750(config)#mls qos                ! Enables QoS globally for the Cat3550
CAT3750(config)#exit
CAT3750#


CAT3750#show mls qos
QoS is enabled                         ! Verifies that QoS is enabled globally
QoS ip packet dscp rewrite is enabled

CAT3750#
```

Cisco Catalyst QoS Deployment

Trust Boundary Policy—Access Edge (VLAN-Based Policy)

```
Catalyst(config)# ip access-list extended RealTime-Voice-ACL
Catalyst(config-ext-nacl)# permit udp any any range 16384 32767
Catalyst(config)# ip access-list extended Signaling-ACL
Catalyst(config-ext-nacl)# permit tcp any any range 1718 1721
Catalyst(config-ext-nacl)# permit tcp any any range 2000 2002
Catalyst(config-ext-nacl)# permit tcp any any range 2427 2428
Catalyst(config-ext-nacl)# permit tcp any any range 3230 3235
Catalyst(config-ext-nacl)# permit tcp any any eq 1731
Catalyst(config-ext-nacl)# permit tcp any any eq 1560
Catalyst(config-ext-nacl)# permit udp any any range 11000 11999

Catalyst(config)# class-map match-all Voice-Bearer
Catalyst(config-cmap)# match access-group name RealTime-Voice-ACL
Catalyst(config)# class-map match-all Voice-Signaling
Catalyst(config-cmap)# match access-group name Signaling-ACL

Catalyst(config)# policy-map Mark-VVLAN
Catalyst(config-pmap)# class Voice-Bearer
Catalyst(config-pmap-c)# police 12800000 400000 conform-action set-dscp-transmit ef exceed-
action drop
Catalyst(config-pmap)# class Voice-Signaling
Catalyst(config-pmap-c)# police 3200000 100000 confomr-action set-dscp-transmit cs3 exceed-
action drop
Catalyst(config-pmap)#  class class-default
Catalyst(config-pmap-c)# set dscp default

Catalyst(config)# policy-map Mark-DVLAN
Catalyst(config-pmap)# class class-default
Catalyst(config-pmap-c)# set dscp default
```

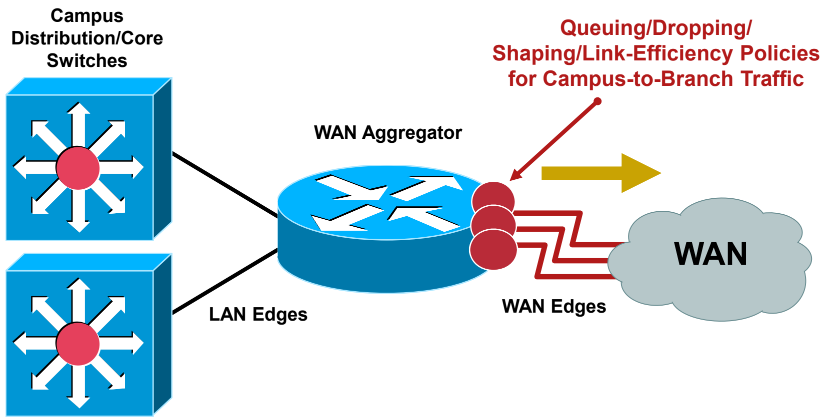
Cisco Catalyst QoS Deployment

Trust Boundary Policy—Access Edge (VLAN-Based Policy) (Cont)

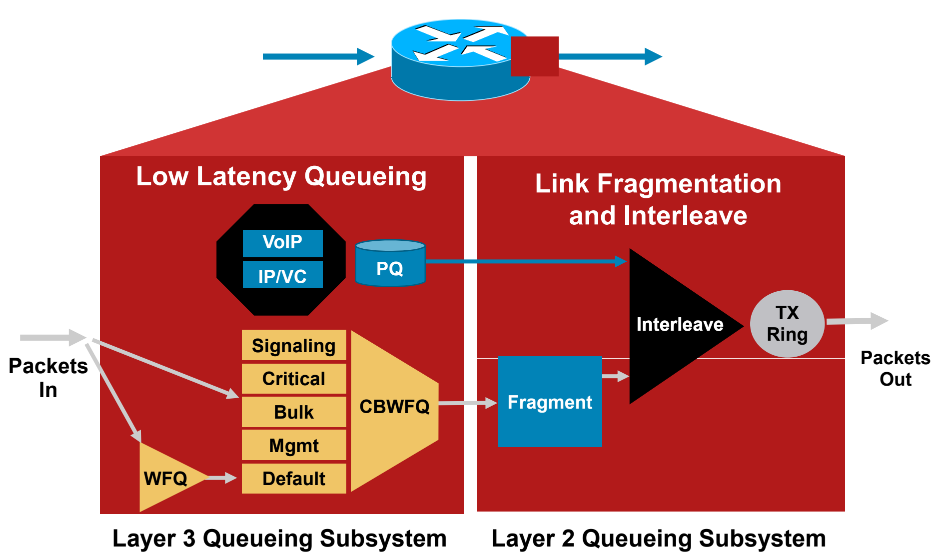
```
Catalyst(config)# interface FastEthernetx/y
Catalyst(config-if)# Description ***Access port with VLAN-based trust boundary**
Catalyst(config-if)# switchport access vlan 10
Catalyst(config-if)# switchport mode access
Catalyst(config-if)# switchport voice vlan 100
Catalyst(config-if)# mls qos vlan-based   ← ← ← ← ["qos vlan-based" for 4500]

Catalyst(config)# interface Vlan100
Catalyst(config-if)# service-policy input Mark-VVLAN

Catalyst(config)# interface Vlan10
Catalyst(config-if)# service-policy input Mark-DVLAN
```

WAN Edge QoS Design Considerations

QoS Requirements of WAN Aggregators



Scheduling/Queueing Tools



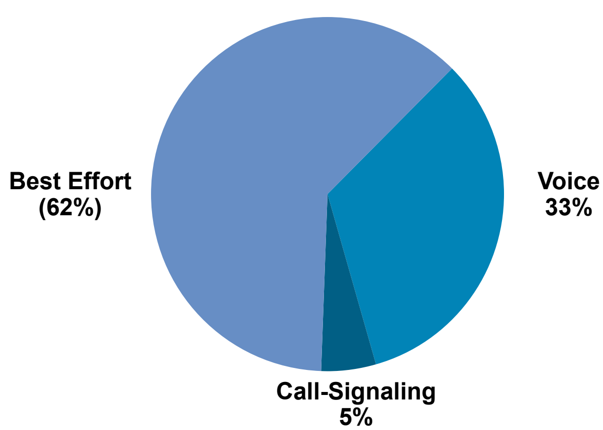
WAN Edge QoS Design Considerations

Link-Speed Considerations

- Slow-speed links (≤ 768 kbps)
  - Voice or video (not both)—3 to 5 class model
  - LFI mechanism required
  - cRTP recommended
- Medium-speed links (≤ T1/E1)
  - Voice or video (not both)—5 Class model
  - cRTP optional
- High-speed links (> T1/E1)
  - Voice and/or video—5 to 11 Class (QoS baseline) model
  - Multiple links require bundling or load-balancing

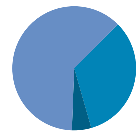
WAN Edge Bandwidth Allocation Models

Three-Class (VoIP and Data Only) WAN Edge Model



WAN Edge Bandwidth Allocation Models

Three-Class WAN Edge Model Configuration Example

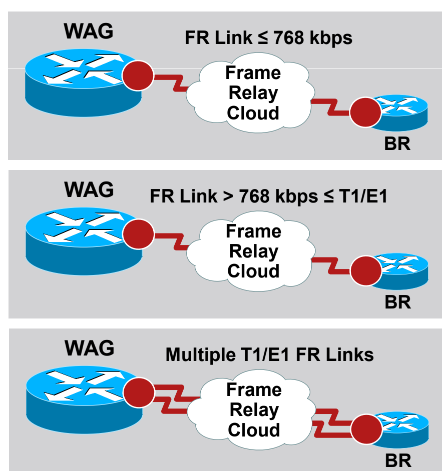


```
!
 class-map match-all VOICE
  match ip dscp ef                     ! IP Phones mark Voice to EF
 class-map match-any CALL-SIGNALING
  match ip dscp cs3                    ! Call-Signaling marking (new)
  match ip dscp af31                   ! Call-Signaling marking (old)
!
!
 policy-map WAN-EDGE
  class VOICE
   priority percent 33                 ! Recommended to keep LLQ ≤ 33%
   compress header ip rtp              ! Optional: Enables Class-Based cRTP
  class CALL-SIGNALING
   bandwidth percent 5                 ! Minimal BW guarantee for Call-Signaling
  class class-default
   fair-queue                          ! All other data gets fair-queuing
!
```

Frame Relay QoS Design

Frame Relay Recommendations

- Slow-speed (≤ 768 kbps)
  - FRTS or CB-FRTS
    - Set CIR to 95% PVC speed
    - Set Bc to CIR/100
    - Set Be to 0
  - FRF.12 or MLPoFR
  - cRTP
- Medium-speed (≤ T1/E1)
  - FRTS or CB-FRTS
    - Set CIR to 95% PVC speed
    - Set Bc to CIR/100
    - Set Be to 0
  - cRTP optional
- High-speed (Multiple T1/E1)
  - Use IP CEF per-packet load balancing



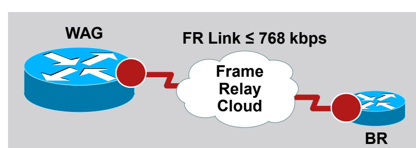
Frame Relay QoS Design

FRTS (+ FRF.12) Recommended Parameters Table

PVC Speed	CIR	Bc	Fragment Size
56 kbps	53200 bps	532 bits per Tc	70 Bytes
64 kbps	60800 bps	608 bits per Tc	80 Bytes
128 kbps	121600 bps	1216 bits per Tc	160 Bytes
256 kbps	243200 bps	2432 bits per Tc	320 Bytes
384 kbps	364800 bps	3648 bits per Tc	480 Bytes
512 kbps	486400 bps	4864 bits per Tc	640 Bytes
768 kbps	729600 bps	7296 bits per Tc	960 Bytes

Frame Relay QoS Design

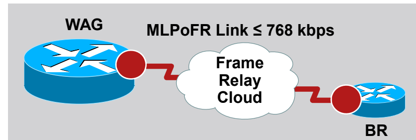
Slow-Speed Frame Relay Configuration Example




```
<a 3 to 5 Class Model can be used>
Optional: Enabling Class-Based cRTP

!
policy-map MQC-FRTS-768
  class class-default
   shape average 729600 7296 0          ! CIR=95% rate, Bc=CIR/100, Be=0
   service-policy WAN-EDGE              ! Queues packets before shaping
!
!
interface Serial2/0
 no ip address
 encapsulation frame-relay
 frame-relay traffic-shaping
!
interface Serial2/0.12 point-to-point
 frame-relay interface-dlci 102
  class FR-MAP-CLASS-768                ! Binds the map-class to the FR DLCI
!
!
map-class frame-relay FR-MAP-CLASS-768
 service-policy output MQC-FRTS-768     ! Attaches MQC policies to FR map-class
 frame-relay fragment 960               ! Enables FRF.12
!
```

MLPoFR QoS Design

Slow-Speed FR SIW Configuration Example at the Branch




```
<a 3 to 5 Class Model can be used>
Optional: Enabling Class-Based cRTP
!
interface Serial6/0
 no ip address
 encapsulation frame-relay
 frame-relay traffic-shaping
!
interface Serial6/0.60 point-to-point
 bandwidth 256
 frame-relay interface-dlci 60 ppp Virtual-Template60      ! Enables MLPoFR
  class FRTS-256kbps                          ! Binds the map-class to the FR DLCI
!
interface Virtual-Template60
 bandwidth 256
 ip address 10.200.60.2 255.255.255.252
 service-policy output WAN-EDGE               ! Attaches MQC policy to map-class
 ppp multilink
 ppp multilink fragment-delay 10              ! Enables MLP fragmentation
 ppp multilink interleave                     ! Enables MLP interleaving
!
map-class frame-relay FRTS-256kbps
 frame-relay cir 243200                       ! CIR is set to 95% of FR DLCI rate
 frame-relay bc 2432                          ! Bc is set to CIR/100
 frame-relay be 0                             ! Be is set to 0
 frame-relay mincir 243200                    ! MinCIR is set to CIR
 no frame-relay adaptive-shaping              ! Adaptive shaping is disabled
!
```

Proctor Case Studies XVI: QoS—LLQ

Lab Sample Question

Configure Low Latency Queuing on R2 so that voice media traffic gets strict priority queuing for up to 33%, voice signaling traffic receives 5% bandwidth guarantee, all other traffics should receive weighted fair queuing

Candidate's Problem Statement

I configured LLQ as requested, I even did "show policy interface" to verify that packets are matched and properly queued, why did I lose points in LLQ?

Proctor Case Studies XVI: LLQ (Cont)

From the output of a “show policy interface” from candidate’s router, could you spot what was misconfigured?

```
R2#sh poli int
 Serial0/0

  Service-policy output: LLQR2

    Class-map: media (match-all)
      1069 packets, 68416 bytes
      5 minute offered rate 0 bps, drop rate 0 bps
      Match: ip dscp ef
      Queueing
        Strict Priority
        Output Queue: Conversation 264
        Bandwidth 33 (%)
        Bandwidth 509 (kbps) Burst 12725 (Bytes)
        (pkts matched/bytes matched) 0/0
        (total drops/bytes drops) 0/0

!!!! Contiuned onto next slide
```

Proctor Case Studies XVI: LLQ (Cont)

From the output of a “show policy interface” from candidate’s router, could you spot what was misconfigured?

```
!!!!!! Continued from last page!!!!!!

Class-map: control (match-all)
      76941 packets, 4182091 bytes
      5 minute offered rate 0 bps, drop rate 0 bps
      Match: ip dscp af31
      Queueing
        Output Queue: Conversation 265
        Bandwidth 5 (%)
        Bandwidth 77 (kbps) Max Threshold 64 (packets)
        (pkts matched/bytes matched) 76941/4182091
        (depth/total drops/no-buffer drops) 0/0/0

    Class-map: class-default (match-any)
      95535 packets, 7385300 bytes
      5 minute offered rate 0 bps, drop rate 0 bps
      Match: any
```

No queuing policy for default class map, all other traffic should be weighted fair queued

Proctor Case Studies XVI: LLQ (Cont)

The Class-default section of the “show policy interface” should look like this

```
Class-map: class-default (match-any)
      95743 packets, 7400554 bytes
      5 minute offered rate 0 bps, drop rate 0 bps
      Match: any
      Queueing
        Flow Based Fair Queueing
        Maximum Number of Hashed Queues 256
        (total queued/total drops/no-buffer drops) 0/0/0
```

Here Is the Wrong Config

```
class-map match-all media
  match ip dscp ef
 class-map match-all control
  match ip dscp af31
!
 policy-map LLQR2
  class media
   priority percent 50
  class control
   bandwidth percent 5
  class class-default
```

Here Is the Good Config

```
class-map match-all media
  match ip dscp ef
 class-map match-all control
  match ip dscp af31
!
 policy-map LLQR2
  class media
   priority percent 50
  class control
   bandwidth percent 5
  class class-default
   fair-queue
```

Summary: Quality of Service

Be Familiar with the Following

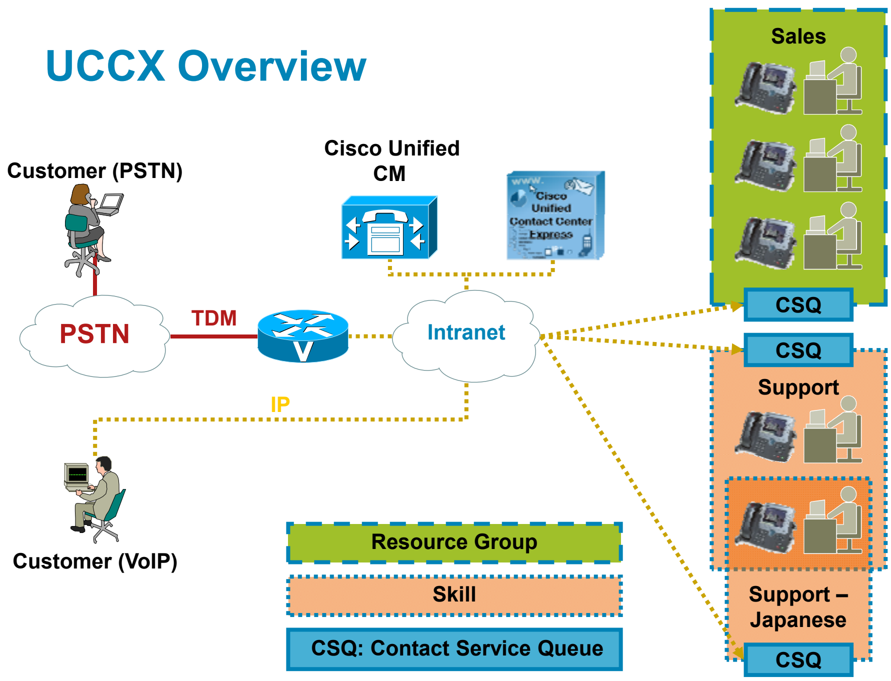
- QoS configuration such as queuing, classification, and policing in a campus environment
- Queuing, shaping and link efficiency configuration for WAN links such as Frame Relay and PPP
- Know different syntax across platforms

Q and A

Session 11

Unified Contact Center Express (UCCX) and B-ACD for CCME

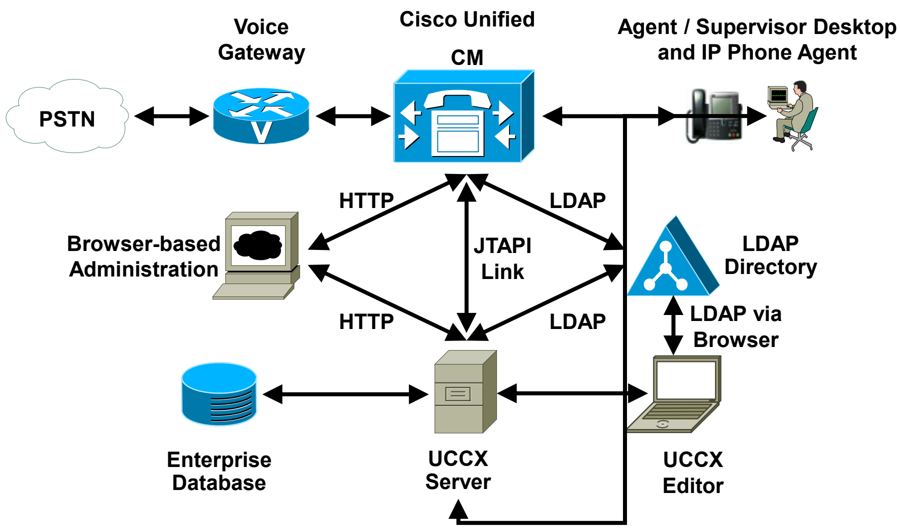
UCCX Overview



UCCX Components

- Cisco Unified Communications Manager (CUCM)
  - IP Phone control, routing calls to UCCX
- LDAP Directory
  - Repositories for UCCX configurations and scripts
- UCCX Server
  - Engine
- Cisco IP Phone Agent (IPPA) / Cisco Agent Desktop (CAD) / Cisco Supervisor Desktop (CSD)
  - Agent and Supervisor functions
- Cisco UCCX Editor
  - UCCX scripts creation and edition

UCCX Components (Cont)



IPCC Express Terminologies
IP-IVR

- Unified CM Telephony Subsystem

  Unified CM Telephony Trigger (CTI Route Point) – triggers Applications to start

  Unified CM Telephony Call Control Group (CTI Port) – group of CTI Ports

- Cisco Media Subsystem

  Cisco Media Termination Dialog Group – play prompts, collect DTMF digits

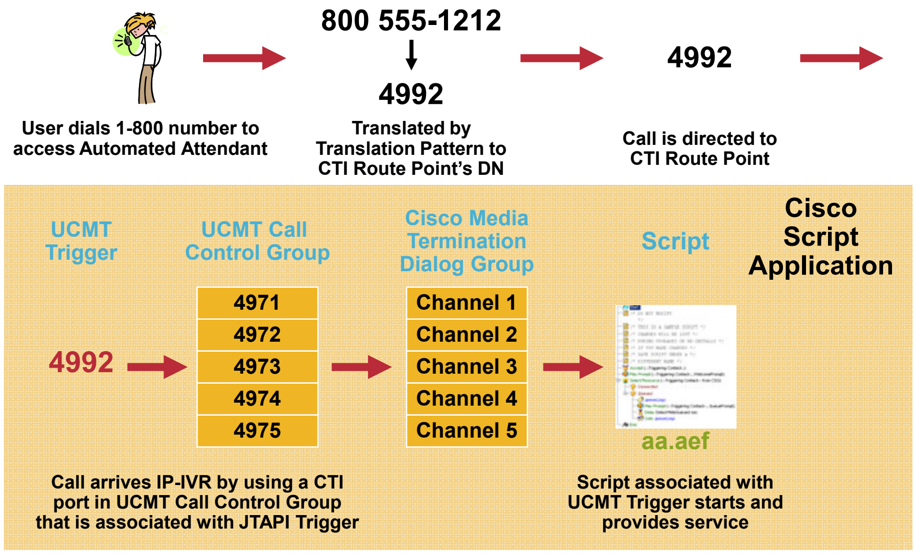
- Script

  Workflow created with UCCX Editor

- Cisco Script Application

  Service provided by combining all of the above

UCCX Call Flow Example



UCCX Terminologies

ICD

- RmCm Subsystem

  Resource – Agent / Supervisor that answers calls

  Resource Group – A group of Resources; Resource can only belong to 1 Resource Group

  Skill – Expertise that Resource have; Resource can be associated with multiple Skills

  CSQ (Contact Service Queue) – A queue of calls that is waiting to be serviced by Resources; CSQ can be associated with 1 Resource Group or multiple Skills

UCCX User Accounts

- UCCX Administrator
  - Created on CUCM for logging on to AppAdmin page
- AXL Administrator
  - Used by UCCX to insert configurations (CTI RP, Ports, Application Users below) on CUCM
- Cisco Unified CM Telephony User (CTI Route Point and CTI Port are associated)
  - Connects to CTI Manager as JTAPI Client to route calls
- RMCM User (Agent/Supervisor IP Phone are associated)
  - Agent State monitoring, Call State monitoring, routes/queues calls
- Agent/Supervisor (his/her own IP Phone is associated)
  - Used for logging on to Agent or Supervisor applications

UCCX (IPIVR and ICD) Configuration Outline

All configuration done on UCCX except for CTI Manager Service activation on CUCM

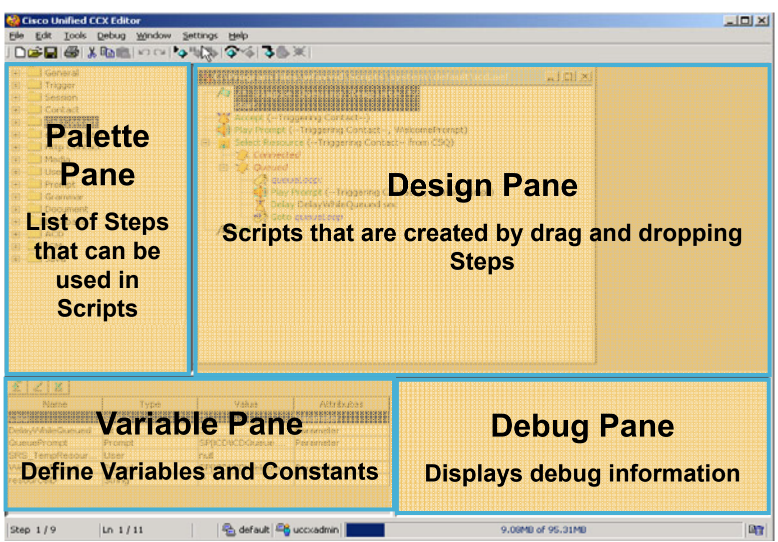
- On CUCM:
  - CTI Manager Service Activation
- On UCCX:
  - Cisco Unified CM Configuration
    1. Define AXL Service Provider Address, with Username and password
    2. Define Cisco Unified CM Telephony Provider (CTI Manager), User Prefix (Jtapi user) and password
    3. Define RMCM Provider, User ID (Rm user) and password
    4. Define NTP Server

UCCX (IPIVR and ICD)Configuration Outline (Continued)

- On UCCX (Continued from the previous page)
  - **Add a new Application and associate it with the ICD script**
  - **Cisco Media Termination Dialog Group Configuration**
  - **Cisco Unified CM Telephony Subsystems Configuration**
    1. Define Cisco Unified CM Telephony Provider (Done earlier but change if necessary)
    2. Define Cisco Unified CM Telephony Call Control Group
       - This is where you define CTI ports and all the relevant CUCM parameters
    3. Define Cisco Unified CM Telephony Call Control Trigger
       - This is where you define CTI Route Point and associate it with the application (e.g.icd.aef) and the Cisco Unified CM Telephony Call Control Group
  - **RMCM Subsystems Configuration**
    1. Define RMCM Provider (Done earlier but change if necessary)
    2. Define Resource Group, and then Resources (or skills)
    3. Define Contact Service Queue (CSQ)

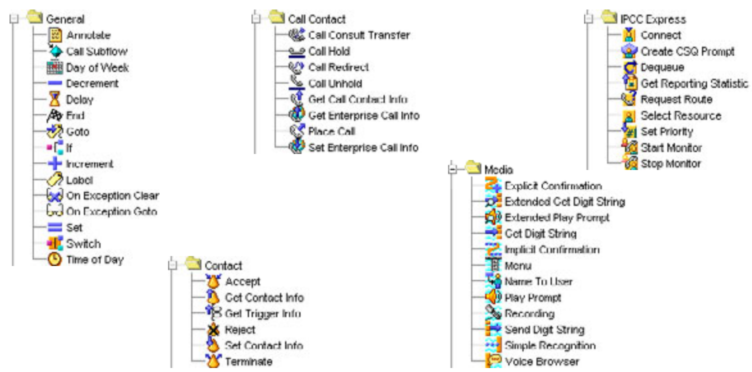
UCCX Editor

Panes



UCCX Editor (Key Steps)

- Drag and drop Steps from Palette Pane to Design Pane; Details are specified in Step’s property window



B-ACD Overview

- Consists of 2 TCL scripts, one is used as auto attendant, the other is used for call queuing. Both scripts required to make B-ACD to work
- Script and audio files can be hosted on flash, compact flash, or tftp server
- Call are queued using FIFO mechanism if unanswered. While call is queued caller will hear MoH from flash. Optional greeting indicating “all agents are busy” can be played at specific intervals
- If call was unanswered within retry timeout, or if all agents are logged out of hunt-group, call will forward to final/alternate destination (usually VM)

B-ACD Basic Configuration

```
application
 service acd flash:app-b-acd-2.1.2.2.tcl
  param number-of-hunt-grps 1
  param queue-manager-debugs 1
  param aa-hunt1 2000
!
 service aa flash:app-b-acd-aa-2.1.2.2.tcl
  paramspace english language en
  param number-of-hunt-grps 1
  paramspace english index 1
  paramspace english location flash:
  param max-time-vm-retry 1
  param voice-mail 1100
  param handoff-string aa
  param service-name acd
  param welcome-prompt _bacd_welcome.au
  param aa-pilot 4085552000
!
dial-peer voice 1000 pots
 service aa
 incoming called-number 4085552000
 direct-inward-dial
 port 2/0:23
!
ephone-hunt 1 sequential
 pilot 2000
 list 2001, 2002, 2003, 2004
 timeout 5
```

- Enters application configuration mode
- Declares name and location of call-queue script
- Defines max. # of hunt grps for call-queue script
- Associates a menu number with an ephone ephone hunt group pilot number
- Declares name and location of the aa script
- Defines max. # of hunt grps in the aa script
- Alternate extension for unanswered queue call
- Specifies the aa service name to be given to the call-queue script
- Associate aa script with the call-queue script
- Defines a pilot phone number for the aa script
- Dial-peer with incoming called number to ensure the aa tcl is invoked
- CCME ephone hunt groups with member extensions. The pilot number will be called when corrspondent menu number is selected in the call-queue script.

B-ACD Additional Configuration

```
application
 service acd flash:app-b-acd-2.1.2.2.tcl
  param number-of-hunt-grps 1
  param queue-manager-debugs 1
  param aa-hunt1 2000
  param queue-len 5
!
 service aa flash:app-b-acd-aa-2.1.2.2.tcl
  paramspace english language en
  param number-of-hunt-grps 1
  paramspace english index 1
  paramspace english location flash:
  param dial-by-extension-option 3
  param max-extension-length 7
  param voice-mail 1100
  param handoff-string aa
  param service-name acd
  param welcome-prompt _bacd_welcome.au
  param aa-pilot 4085552000
  param call-retry-time 10
  param max-time-call-retry 300
  param max-time-vm-retry 1
```

- Enable call statistics collection for debugging
- Maximum calls in the queue. Default is 10
- Specifies menu number for dial-by-extension
- Specifies maximum number of digit in dial-by-extension
- Defines wait interval (in seconds) before a queued call is resend to the hunt groups. Default is 15 seconds
- Maximum time (in seconds) in queue before second to the alternate number. Default is 600
- Maximum number of time the alternate number is attempted.

B-ACD Script Reload

- Use call application voice load <*scriptname*> command to reset AA script.

```
CME# call application voice load acd
```

- Since queue script is always running, you will need to stop the script before resetting. First, identify session ID of queue script:

```
CME# show call app sessions

Session ID 2
             App: acd
            Type: Service
             Url: flash:app-b-acd-2.1.0.0.tcl
```

- Next, stop the queue script using session ID and reset script:

```
CME# call application session stop id 2
CME# call application voice load acd
```

Other B-ACD Verifications:

```
CME# show call application session ?
  callid   ID of a call handled by a session
  id       ID of a session from show displays
  name     Name of a started session from start CLI
  |        Output modifiers
  <cr>

CME# show ephone-hunt <tag> statistics ?
  last    start of statistics ephone_hunt
  start   start time of statistics output
```

Summary: UCCX and B-ACD for CCME

Be Familiar with the Following:

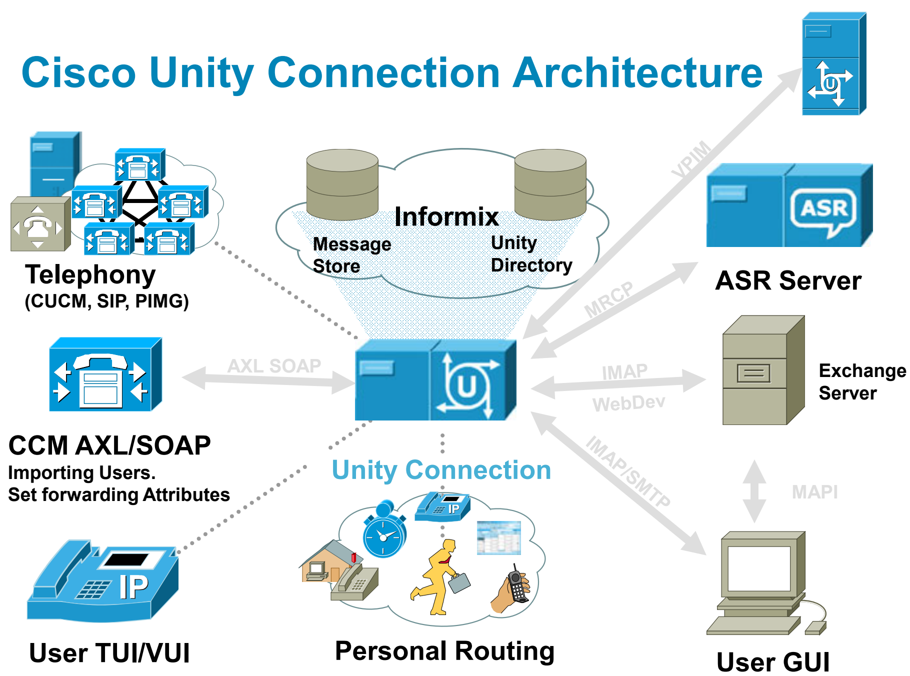
- UCCX server configuration
- UCCX trace file configuration and basic analysis
- Know how to customize and edit .aef scripts
- Phone services for UCCX applications
- B-ACD configuration for CCME
- TCL Script configuration for B-ACD

Q and A

Session 12

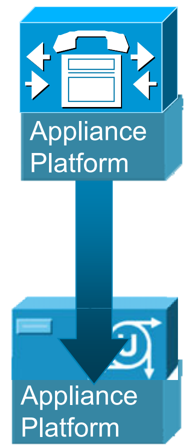
Cisco Unity Connection and Cisco Unity Express

Cisco Unity Connection Architecture



Linux Appliance Platform

- Same platform as Cisco Unified Communications Manager
  - Same version, same image
  - Pre-loaded on MCS servers
  - Released in lock step with UC Manager
- Common Platform Services
  - Diagnostics, traces, port status monitor
  - Backup and restore
  - Patch management
  - In-Place upgrades



Deployment Models

- Single site Centralized Messaging
- Multiple sites connected via VPIM

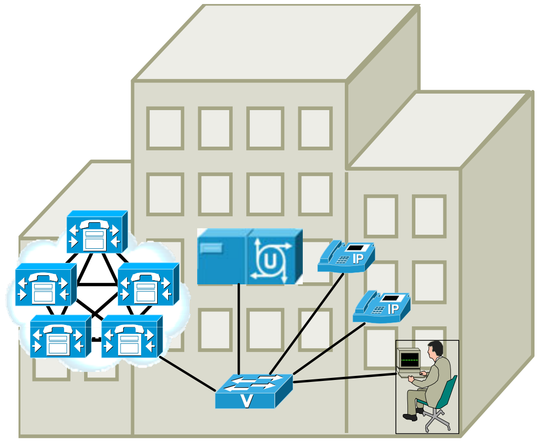
Single Site Centralized Messaging

A single Cisco Unity Connection supports up to 10,000 users and 144/288 voice ports

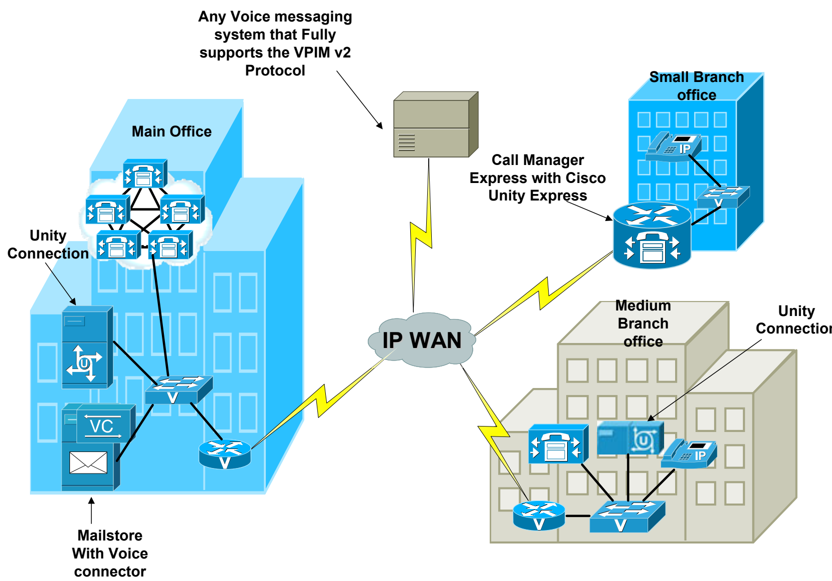
Many different Phone systems are supported for Cisco Unity Connection

System Administration

Cisco Personal Communications assistant for administration of your own mail box settings as well as Desktop messaging via the CPCA Inbox



Multiple Sites Connected via VPIM



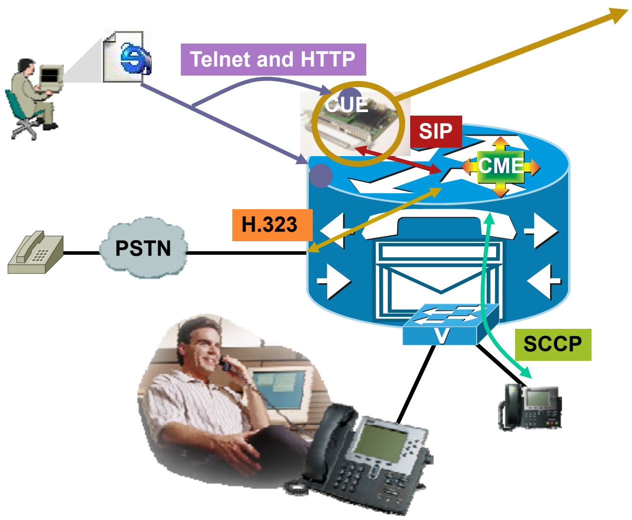
Integrating with CUCM

- Configuring CUCM
  - Create a new partition to be used by Unity Connection
  - Create a new Calling Search Space to be used by Unity Connection
  - Create a new Device Pool to be used by Unity Connection
  - Create Voicemail Ports using the Voice Mail Port Wizard
  - Create a Line Group and add Voicemail Ports to the list
  - Create a Hunt List and add the Line Group to the list
  - Create a Hunt Pilot with all the settings and route calls to Hunt List
  - Create Message Waiting Indicator Directory Numbers
  - Create Voicemail Pilot with all the appropriate settings
  - Create Voicemail Profile with all the appropriate settings
  - Assign the Voicemail Profile to the Users

Integrating with CUCM (Contd)

- Configuring Unity Connection
  - Create a new Phone System under Telephony Integration
  - Create a new Port Group with all the required settings
  - Create the actual Ports and enter all the required settings
  - Configure the CUCM AXL Servers under Phone System
  - Configure the Port Group with redundant CUCM Servers
  - Create or Modify User Templates, Class of Service etc
  - Add Users or Import Users from CUCM
  - Create or Modify Call Handlers with the required settings
  - Create or Modify Call Routing rules if required

CUE Architecture



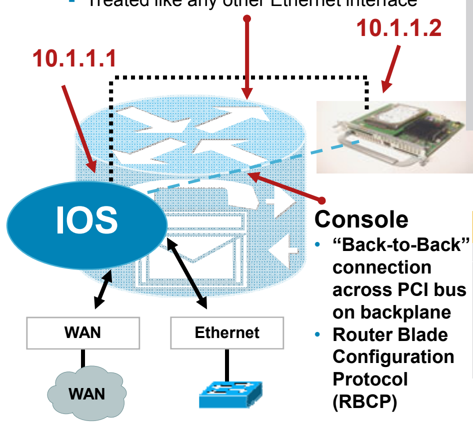
- VoiceMail and AA
  - Linux environment
  - LDAP directory
  - SQL database
  - VXML browser
  - CRS software

- VoiceMail and AA
  - Linux environment
  - LDAP directory
  - SQL database
  - VXML browser
  - CRS software

CUE IP Connectivity

Packet Communications

- "Back-to-Back" Ethernet connection across the PCI bus on the backplane
- Treated like any other Ethernet interface



Console

- **"Back-to-Back" connection across PCI bus on backplane**
- **Router Blade Configuration Protocol (RBCP)**

IP Unnumbered

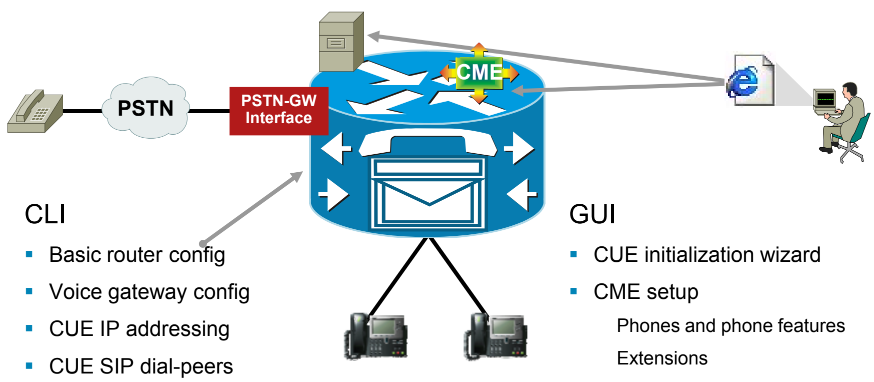
```
interface FastEthernet0/0
 ip address 10.1.1.1 255.255.255.0
interface Integrated-Service-Engine4/0
 ip unnumbered FastEthernet0/0
 service-module ip address 10.1.1.2 255.255.255.0
 service-module ip default-gateway 10.1.1.1
ip route 10.1.1.2 255.255.255.255 Integrated-Service-Engine4/0
```

Explicit IP Address

```
interface FastEthernet0/0
 ip address 10.1.1.1 255.255.255.0
interface Integrated-Service-Engine4/0
 ip address 20.1.1.2 255.255.255.0
 service-module ip address 20.1.1.2 255.255.255.0
 service-module ip default-gateway 20.1.1.1
ip route 10.1.1.2 255.255.255.255 Integrated-Service-Engine4/0
```

CUE Default-gateway MUST be the CME Router Address

CUCME/CUE Configuration: CLI and GUI



CLI

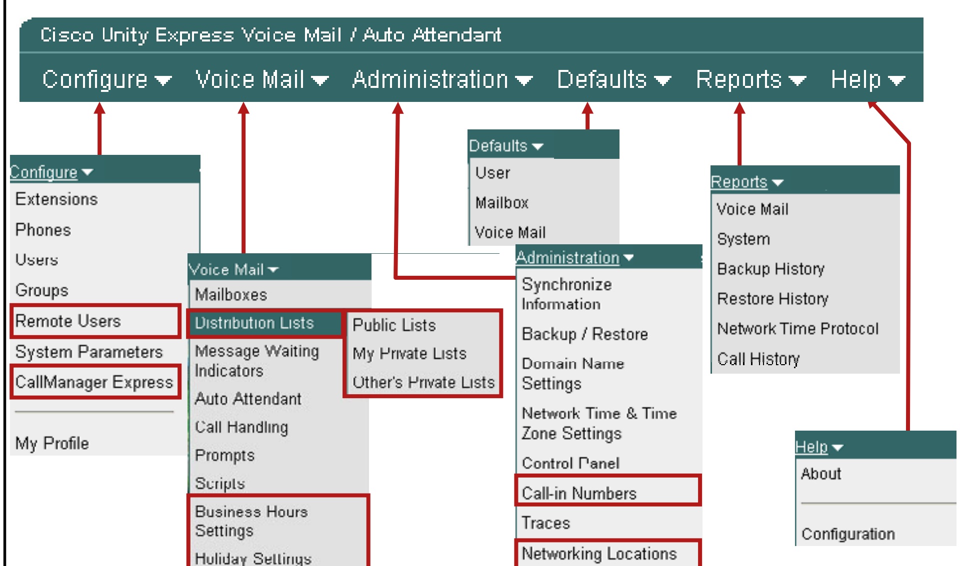
- Basic router config
- Voice gateway config
- CUE IP addressing
- CUE SIP dial-peers
- Basic CME admin login definition
- CME “Setup” utility
- Upgrades/Installs
- CUE backup and restore

**Both CLI and GUI Allowed in Lab

GUI

- CUE initialization wizard
- CME setup
  - Phones and phone features
  - Extensions
  - Dial-plans
- Vmail setup
  - Mailboxes
- AA setup
- Day-to-day moves, adds and changes

CUE GUI Navigation



Basic CUCME/CUE CLI Configuration

```
CCME#
!
interface FastEthernet0/0
 ip address 10.1.1.1 255.255.255.0
!
interface Integrated-Service-Engine4/0
 ip unnumbered FastEthernet0/0
 service-module ip address 10.1.1.2 255.255.255.0
 service-module ip default-gateway 10.1.1.1
!
ip route 10.1.1.2 255.255.255.255 Integrated-Service-Engine4/0
!
ip http server
!
dial-peer voice 5000 voip
 destination-pattern 5000
 session protocol sipv2
 session target ipv4:10.1.1.2
 codec g711ulaw
 no vad
!
```

Defines CUE IP routing properties

Static route to CUE module

SIP dial-peer to CUE for VM

Basic CUCME/CUE CLI Configuration (Cont)

```
CCME#
!
telephony-service
 max-ephones 8
 max-dn 48
 ip source-address 10.1.1.1 port 2000
 create cnf-files version-stamp Jan 01 2002 00:00:00
 dialplan-pattern 1 4085552... extension-length 4
 voicemail 5000
 max-conferences 8 gain -6
 transfer-system full-blind
 secondary-dialtone 9
 web admin system name cisco password cisco
 dn-webedit
!
ephone-dn  1
 number 2001
 description 4085552001
 call-forward busy 5000
 call-forward noan 5000 timeout 10
!
ephone-dn  50
 number 8000.... Secondary 8001....
 mwi on-off
```

Enables CCME Phone “voicemail” button

Enables GUI admin to CCME

Defines Call Forward destination per CCME dn

MWI

Basic CUCME/CUE CLI Configuration (Cont)

```
CME#service-module Integrated-service-Engine 4/0 session
Trying 10.1.1.1, 2033 ... Open
CUE>
CUE> en
Password:
CUE#
CUE# sh run
.
.
username jdoe create
username jdoe phonenumber "2001"
.
ccn application ciscomwiapplication
 description "ciscomwiapplication"
 enabled
 maxsessions 8
 script "setmwi.aef"
 parameter "strMWI_OFF_DN" "8001"
 parameter "strMWI_ON_DN" "8000"
 parameter "CallControlGroupID" "0"
 end application
```

Access CUE CLI from CCME

Define user and associates phone number

CUE MWI On and Off DN

Basic CUCME/CUE CLI Configuration (Cont)

```
ccn application voicemail
 description "voicemail"
 enabled
 maxsessions 8
 script "voicebrowser.aef"
 parameter "logoutUri"
"http://localhost/voicemail/vxmlscripts/mbxLogout.jsp"
 parameter "uri"
"http://localhost/voicemail/vxmlscripts/login.vxml"
 end application

ccn subsystem sip
 gateway address "10.1.1.1"
 end subsystem

ccn trigger sip phonenumber 5000
 application "voicemail"
 enabled
 maxsessions 8
 end trigger

voicemail mailbox owner "jdoe" size 3000
 end mailbox
```

Voicemail application properties

SIP subsystem properties

SIP calls to 5000 will trigger voicemail application

Voicemail box size

CUE CLI Verification Commands

```
CUE# sh ccn ?
  application  Telephony programs
  call         Active call related information
  engine       Common configuration parameters for all ccn subsystems
  prompts      Prompt files
  scripts      Workflow script files
  status       Runtime status for ccn subsystems
  subsystem    Subsystem specific configuration
  trace        Traces
  trigger      Telephony interconnects

CUE# sh  voicemail ?
  broadcast    broadcast features
  detail       Mailbox details
  limits       Default values for voicemail handling
  mailboxes    List the mailboxes on this system
  usage        Voicemail load information
  users        List the local voicemail users
```

Summary: Unity Connection and CUE

Be Familiar with the Following:

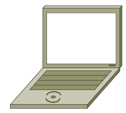
- Unity Connection integration
- Unity Connection call routing rules, call handlers, etc.
- Unity Connection users
- Unity Connection tools
- CUE configuration and integration with CUCME and CUCM

Q and A

Session 13

Cisco Unified Presence

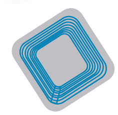
Presence Awareness



- What is "Presence"? Information about a person's willingness and availability to communicate
  - Typically represented by status: Available, In Meeting, On Mobile, At Lunch, Be Back Shortly, etc.
  - Includes details on user's preferred method to communicate: voice, video or Instant Message
  - "Find-Me", "Follow-Me", or "Hide-Me"

- Examples of presence in action today
  - IM "Buddy List" status indication
  - "Busy" tone on traditional phone
  - Contact Center Agent status

- Publish / Subscribe
  - Clients publish presence information to other users who are called subscribers



Presence Terminology - Presentity

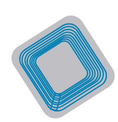
PERSON "A"

- A Person (PRESENTITY) may use multiple communication services/devices
- The status of these devices can be PUBLISHED to a presence Service.

Desk Phone

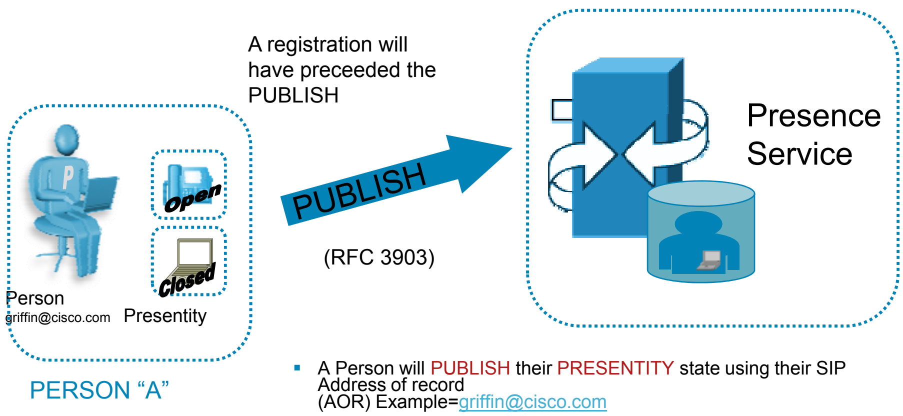
Smart Phone

IM Application



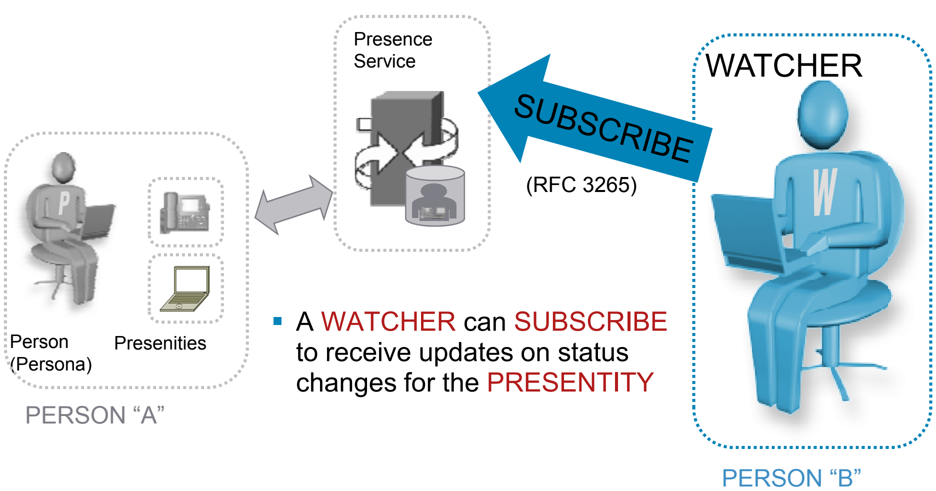
RFI Tag

Presence Terminology - Presentity



- A Person will PUBLISH their PRESENTITY state using their SIP Address of record (AOR) Example=griffin@cisco.com
- They will publish the status of communication Services/Devices to the PRESENCE SERVICE using their PRESENTITY

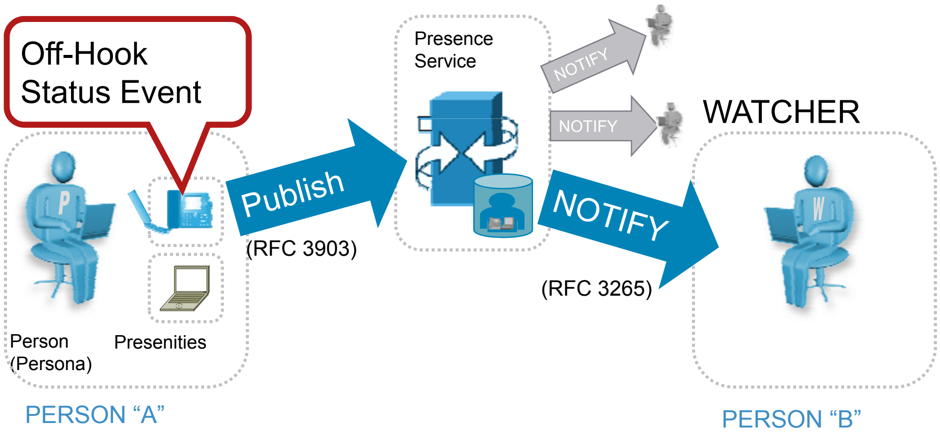
Presence Terminology - Watcher



- A WATCHER can SUBSCRIBE to receive updates on status changes for the PRESENTITY

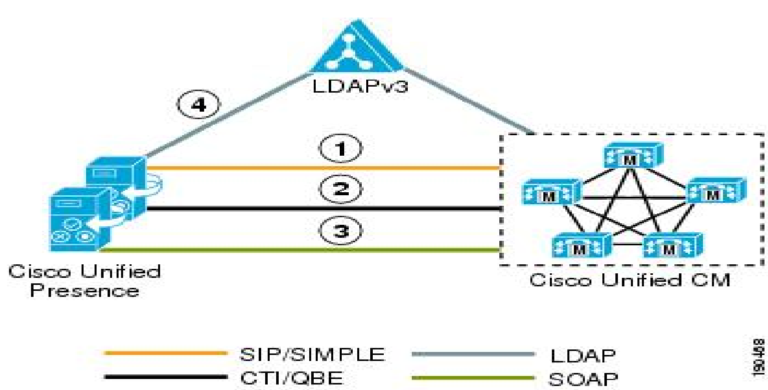
- A Watcher can also be a Presentity

Presence Terminology - Notification

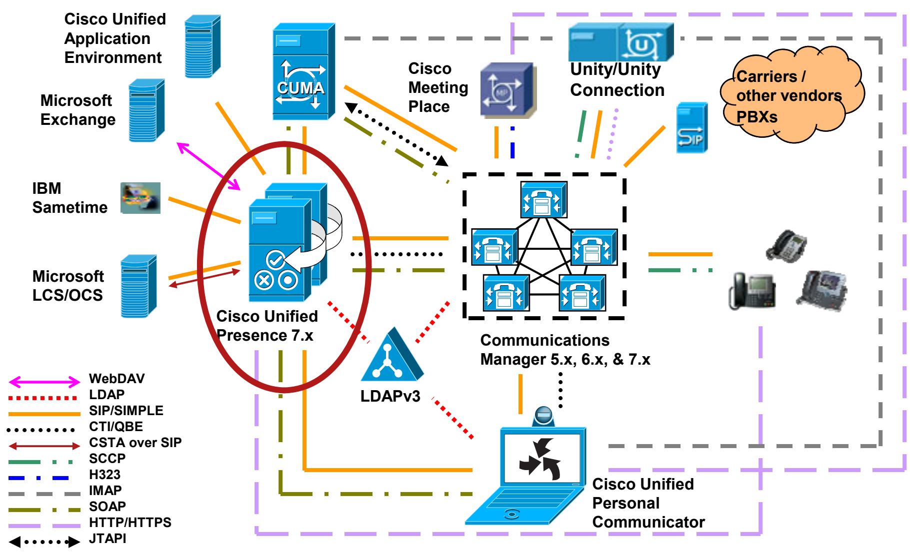


- On a Change of status the PRESENTITY is updated on the Presence Server.
- The Presence Server will Notify all the subscribers of the change in state for the PRESENTITY

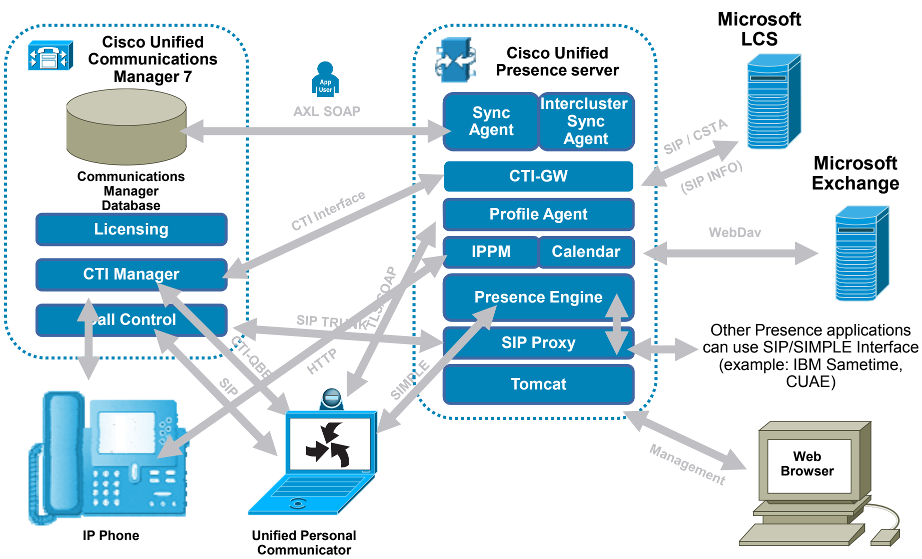
Basic Architecture Presence Server



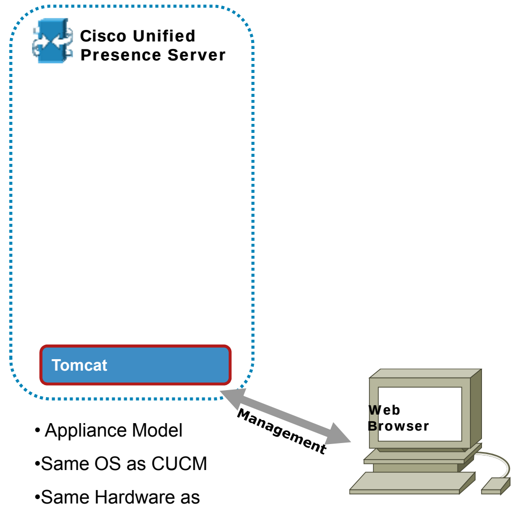
Cisco Unified Presence Components



CUP Appliance Overview

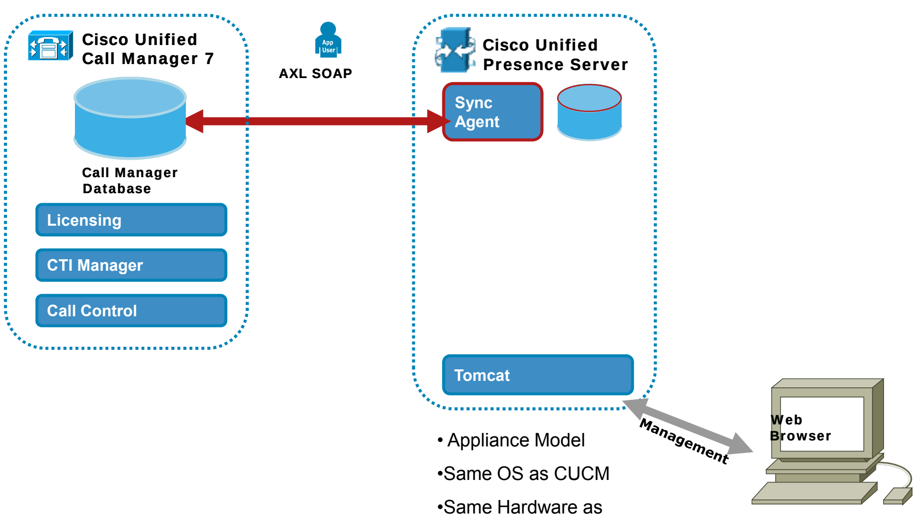


Overview of CUPS Appliance



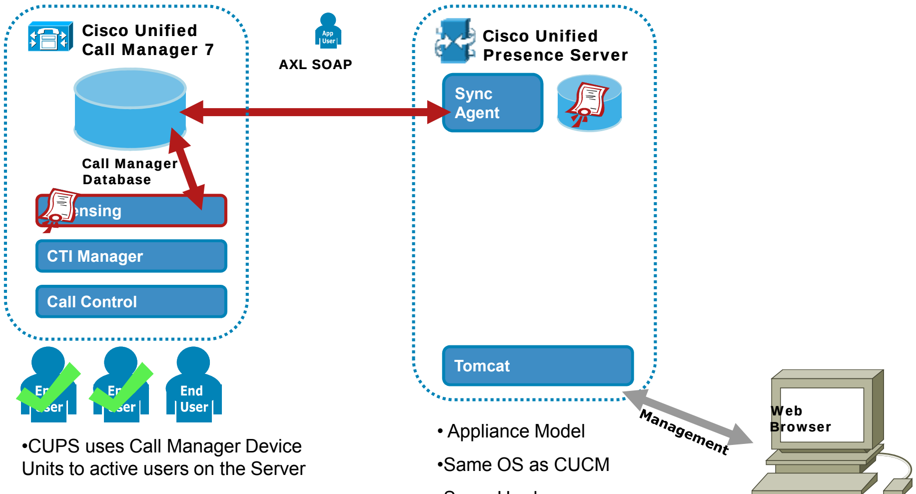
- Appliance Model
- Same OS as CUCM
- Same Hardware as CUCM

Sync Agent replication from Call Manager DB



- Appliance Model
- Same OS as CUCM
- Same Hardware as CUCM

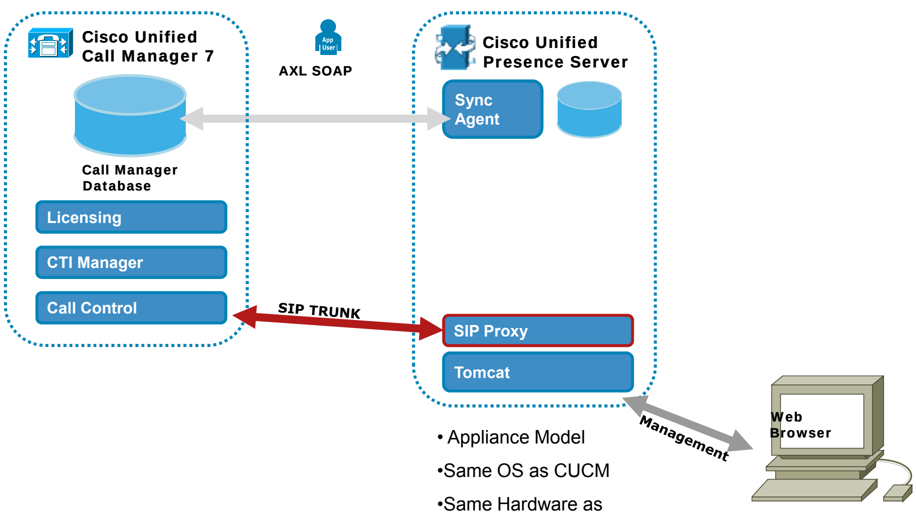
Licensing and User Activation on CUPS



- CUPS uses Call Manager Device Units to active users on the Server
- Additional Units are required for some configurations

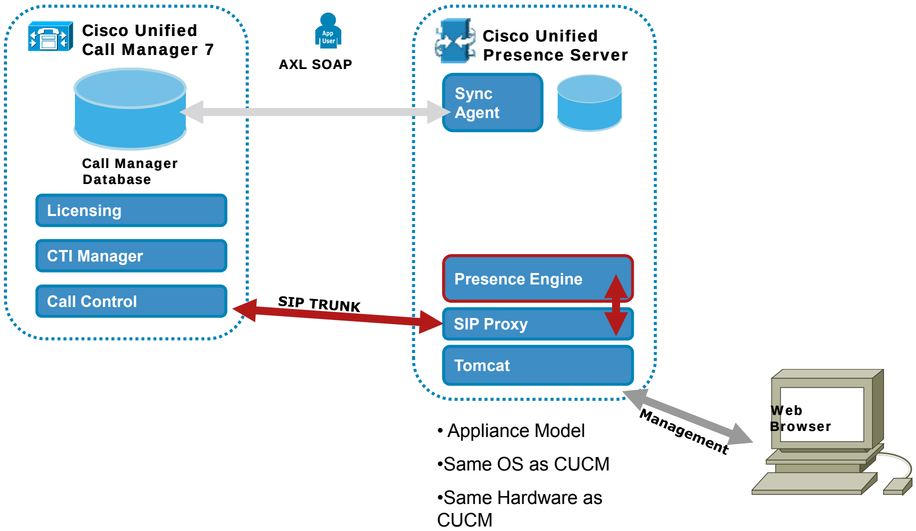
- Appliance Model
- Same OS as CUCM
- Same Hardware as CUCM

SIP Routing using the SIP Proxy Server

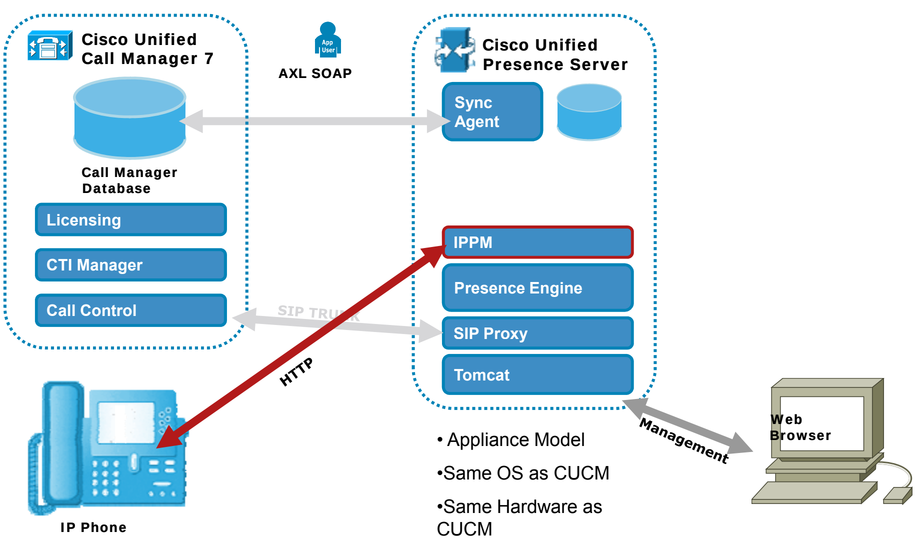


- Appliance Model
- Same OS as CUCM
- Same Hardware as CUCM

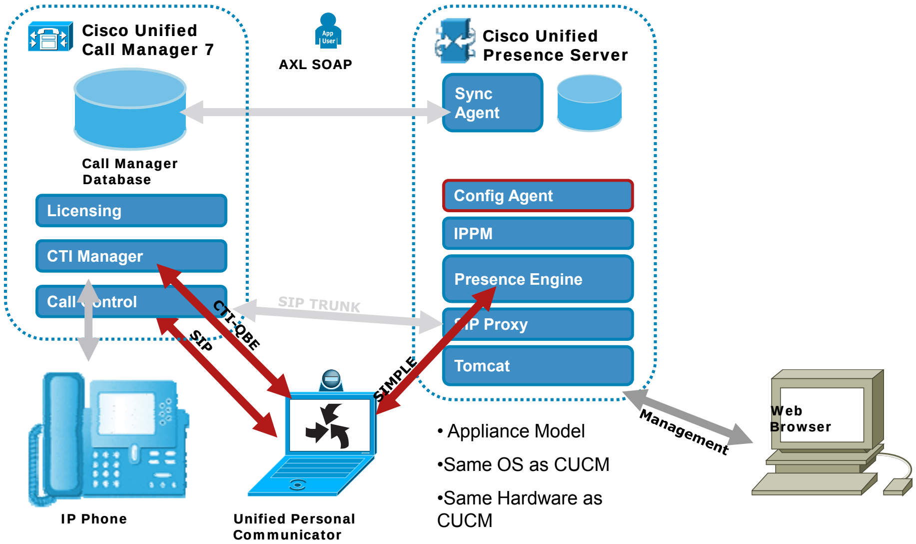
Presence Engine maintains readability information



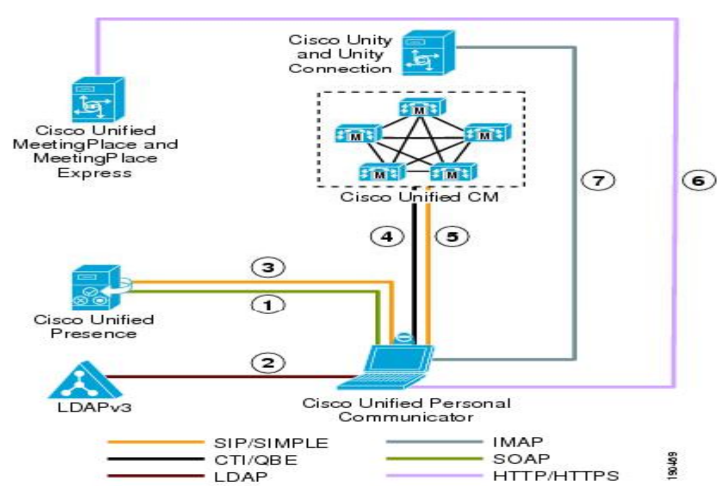
IPPM: HTTP / SIP/SIMPLE Gateway



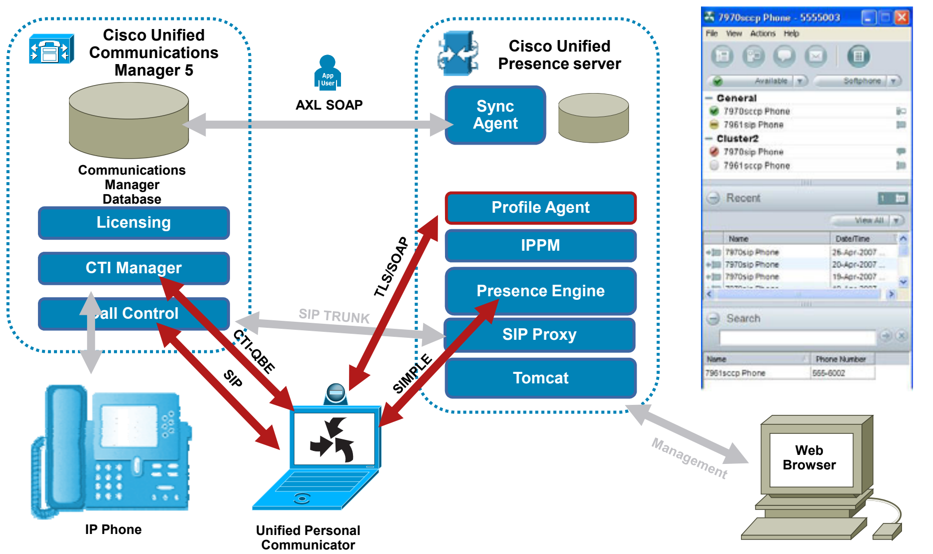
Unified Client Configuration download



Basic Architecture Cisco Unified Personal Communicator (CUPC)



Personal Communicator Configuration

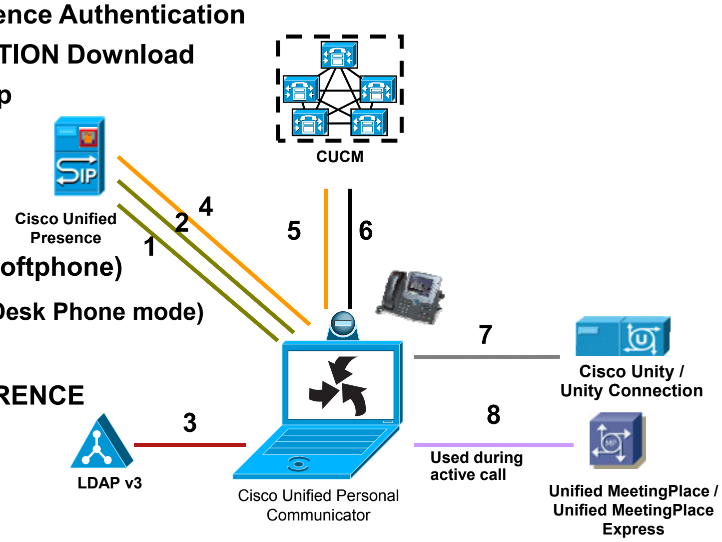


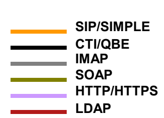
Cisco Unified Personal Communicator Login

1. Unified Presence Authentication
2. CONFIGURATION Download
3. LDAP Lookup
4. REGISTER

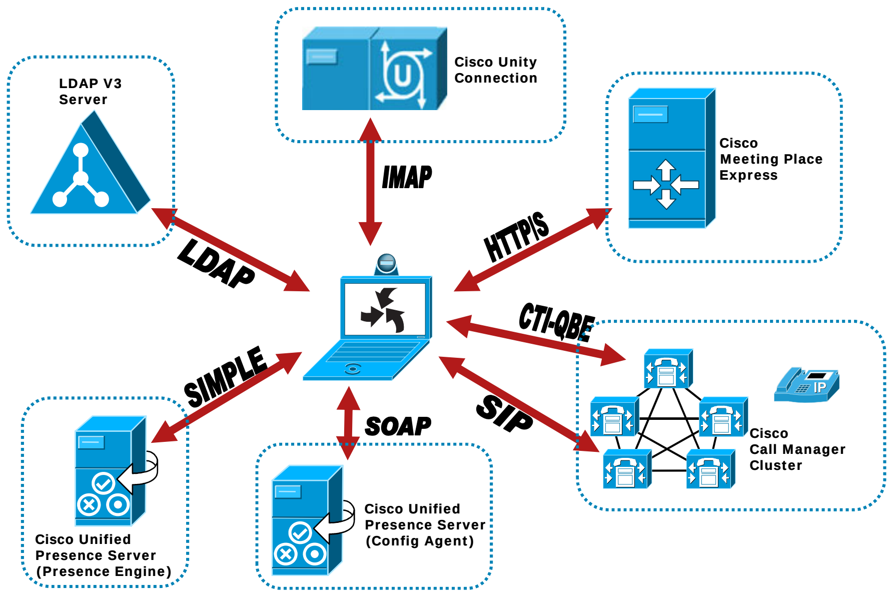
   SUBSCRIBE

   PUBLISH
5. REGISTER (Softphone)
6. CTI Control (Desk Phone mode)
7. IMAP
8. WEB CONFERENCE





Protocol Overview



Cisco Unified Personal Communicator

- Click-to-Call
  - Use the fully integrated softphone or your Cisco Unified IP Phone
  - Shared line, same number and dial plan
  - Same call detail records
- Integrated keypad dialer
- Ad hoc conferencing
  - Voice, video and Web conferencing
- Media Escalation
  - Add video telephony or share documents during active conversations
- Pop-up notifications
  - CLID/Name, call type, call options

Installation and Configuration

- Pre-Installation Steps
- Post-Installation Steps
- Cisco Unified Communications Manager Configuration
- Cisco Unified Presence Server Configuration

Pre-Installation Steps

- Add the Cisco Unified Presence Server as an application server on Communications Manager
- Create a new AXL User in Communications Manager to be used by Presence Server and assign it access to the appropriate group

Post-Installation Steps

- Add the appropriate license files to the Cisco Unified Presence Server
- Activate all the appropriate services on Cisco Unified Presence Server

Cisco Unified Communications Manager Configuration

- Configure the appropriate service parameters
- Configure a new Device Pool to be used by Presence Server
- Configure SIP Trunk Security Profile
- Configure SIP Trunk
- Configure CTI Gateway Application User and set the right permissions
- Configure Application Dial Rules
- Configure Directory Lookup Dial Rules
- Enable Users for Unified Presence Capabilities
- Create CUPC devices
- Associate the devices to the End Users
- Configure the End Users with appropriate settings

Cisco Unified Presence Server Configuration

- Configure the Service Parameters appropriately
- Configure Presence Settings
- Configure Gateway Settings
- Configure Routing Settings
- Configure Voicemail Server, Mail store and Voicemail Profile
- Configure the CTI Gateway Profile
- Configure LDAP Server and LDAP Profile
- Configure CTI Gateway
- Configure profiles for End Users

Summary: CUPS

Be Familiar with the Following:

- Presence Terminology
- CUPS and CUPC architecture
- CUPS integration with CUCM
- CUPS and CUPC Troubleshooting Tools
- CUPS Integration with Unity Connection

Q and A

Session 14

Preparation Resources and Tips

Voice Lab Exam Preparation (I)

Getting Started

- Use the Voice Lab content blueprint on the CCIE webpage as your guide
- Evaluate and determine your knowledge level and hands-on experience in the major topic areas
- Formulate a realistic study plan according to you own work/personal schedule, also customize it according to your technical strength and weaknesses
- Don't spend all your time and focus on collecting the exact replica of the Voice Lab equipment rack
- Seek advise, from other CCIE Voice certified engineers, on preparation plans and tips

Voice Lab Exam Preparation (II)

Study and Practice

- Books

  No single "All-in-One" book that covers it all; use the suggested book list as an reference and stay informed about any upcoming books www.cisco.com/warp/public/625/ccie/voice/book_list.html

- Online reading resources

  Abundant online documentations, write papers, and articles on www.cisco.com which compliments any books on latest technical developments

  Develop a habit of utilizing and navigating the UniverCD documentation at www.cisco.com/univercd/home/home.htm

- Online forums

  Online forums, such as NetPro on www.cisco.com, are excellent sources to seek technical supports and encouragements on questions that arise during your study process

Voice Lab Exam Preparation (III)

Study and Practice (Cont)

- Practice labs and scenarios
  1. You don’t need the exact lab replica to learn
  2. Use the equipment which you have access to and learn each technology thoroughly
  3. Form study groups to exchange ideas and share equipment
  4. Go beyond configuration, learn to debug and troubleshoot
  5. Stay with real world, applicable scenarios
  6. Focus on learning the technologies instead of learning only what you think (or what you’ve been told) is on the lab exam
  7. Stay aware and informed on up-coming new features

Voice Lab Exam Preparation (IV)

Study and Practice (Cont)

- Trainings and bootcamps
  1. To get the most out of trainings and bootcamps, learn as much material as you could _before_ you attend them
  2. Leverage bootcamps for practice time on hard-to-get equipments
  3. Leverage bootcamps to enhance your in-depth knowledge and hands-on skills on various voice technologies
  4. Leverage bootcamps to gauge you technical and mental readiness to take the lab exam
  5. Don’t attend bootcamps to learn the test

Session 14

Exam Tips and Test-Taking Strategies

Lab Exam Tips: Pre-lab

Before your lab exam:

- Visit lab test site the day before
- Don't schedule flights too close to the end of the exam: you should be thinking about the exam instead of catching your flight
- Avoid last minute lab material cramp
- Get some sleep the night before the exam

Lab Exam Tips: In-lab

Think the 4 “C”s:

- Calm
- Careful
- Confident
- Courteous

Lab Exam Tips In-lab (Cont):

Understanding Exam Requirements:

- Read the entire exam first
- Read the questions very carefully: every word is in there for a reason.
- Don’t assume requirements that aren’t mentioned in a question
- Some questions have multiple solutions, unless the test explicitly asked you to use one versus another, all are valid
- Excessive configurations are generally ignored during grading, unless they interfere with expected solution
- Ask the proctor for clarification

Lab Exam Tips In-lab (Cont):

Time Management:

- Use question point values to judge time
- Know time-saving configuration techniques
- Know when to move on – don’t spend too much time on a single task, no matter how important you think it is
- If you suspect hardware issues, notify the proctor immediately
- Don’t make any drastic changes towards the end of lab exam
- Save your configuration frequently

Lab Exam Tips In-lab (Cont):

Verification:

- Verify, verify, verify, and verify again
- Some prefer to verify after each question, others like to verify until they finish the whole test. It’s a matter of personal preference
- Verify against all requirements – not just basic functionalities.
- Makes notes and check-lists

Lab Exam Tips In-lab (Cont):

Troubleshoot:

- Troubleshooting skill is often the difference between failing and passing
- Know what and where to look for debugs and traces
- Look out for those seemingly "invisible" typos
- Remember the test lab is not your home lab – addressing scheme is different
- Troubleshooting is important but don't spend all your time on one problem
- Don't let a unresolved problem impact your confidence
- Again, seek the proctor's assistance

For More Information

- Beware of rumors!
- Visit the CCIE web page at: www.cisco.com/go/ccie
- Support: www.cisco.com/go/certsupport
- Post-lab Email: ccie-lab@cisco.com
- Cheating: ccie-nda-enforcement@cisco.com

FAQ

Q and A

FAQ #1:
How Do You Grade the Lab Exam?

- Proctors are responsible for grading all lab exams
- Automatic tools aid proctors with simple grading tasks; e.g. capturing candidate’s configuration in database, basic configuration verifications, ping tests, etc.
- Automatic tools are never solely responsible for lab exam grading. Proctors are.
- The proctor completes the grading of the exam and submits the final score
- Partial marks are not awarded for questions
- Points are awarded for working solutions only
- Some questions have multiple solutions

FAQ #2:
Affordable Equipment for Home Labs?

- Candidates should focus on learning and understanding of the voice technologies instead of building a home lab that is identical to the real lab
- Start with modular lab equipment and scenarios, go beyond the basic configuration tasks
- Learn to troubleshoot and debug with CLI or trace files, these knowledge are useful and applicable on most Cisco equipment
- Share gears with your study partners
- Leverage available rack rentals if necessary
- Seize any opportunity for relevant hands-on experiences, through daily work projects or even volunteering for projects out of your direct line of duty

FAQ #3:
It’s Discouraging to Fail the Exam

- Remember the knowledge you acquired while preparing for the exam is yours to keep
- Don’t compare with others
- Remember to enjoy the journey
- Tell us how we could improve, submit online feedbacks or write to us at: ccie-lab@cisco.com

References and Acknowledgements

- Various CCM and QoS SRNDs

  http://www.cisco.com/iam/unified/ipt1/Using_SRND_Documents.htm

- Various Networkers VVT Presentations
- Contributions from past/current CCIE Voice Techtorial speakers

Q and A

Recommended Readings

- Cisco IP Telephony: Planning, Design, Implementation, Operation and Optimization, ISBN: 1-58705-157-5
- Cisco QOS Exam Certification Guide, Second Edition, ISBN: 1-58720-124-0
- Cisco CallManager Best Practices: A Cisco AVVID Solution, ISBN: 1-58705-139-7
- Cisco CallManager Fundamentals: A Cisco AVVID Solution, Second Edition, ISBN: 1-58705-192-3
- Configuring CallManager and Unity: A Step-by-Step Guide, ISBN: 1-58705-196-6
- Troubleshooting Cisco IP Telephony, ISBN: 1-58705-075-7
- Cisco Unity Fundamentals, ISBN: 1-58705-098-6
- Cisco Unity Deployment and Solutions Guide, ISBN: 1-58705-118

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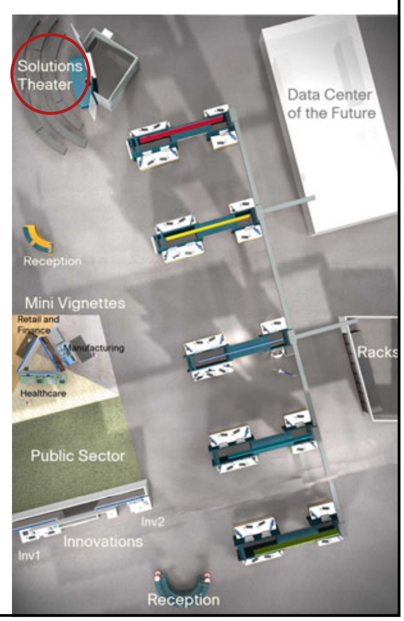

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