Cisco TelePresence: Best Practices for Call Control Integration

BRKEVT-2801

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Agenda

- Introduction
- Unified Communication
- TelePresence networks
- Integration
- Scalability
- Redundancy and Load Balancing
- Security
- Integration case study
- Best practices
- Summary
Introduction
Agenda

- Introduction
  - Pervasive Video Collaboration
  - Unifying networks for Video Collaboration
- Unified Communication
- TelePresence networks
- Integration
- Scalability
- Redundancy and Load Balancing
- Security
- Integration case study
- Best practices
- Summary
Introduction
Pervasive Video Collaboration

Desktop clients and Personal endpoints

Business tablet

Web Collaboration

Video as intuitive and ubiquitous as voice

Multipurpose Telepresence Rooms

Immersive TelePresence

Video-Enabled Contact Center
Introduction
Cisco Video Solutions in the past
Unifying networks for Video collaboration

IP phones and Endpoints
- Soft phones and Mobility

Application Servers
- H.323 Gatekeeper
- H.323-to-SIP GW
- SIP services

Advanced Conferencing Services
- Expressway B2B
- Firewall traversal

Distributed Conferencing Services

3rd Party/H.323 video endpoints

B2B and Remote Worker

UC Manager Cluster

Immersive and Multipurpose
Introduction

TelePresence Architecture Overview
Agenda

- Introduction
- Unified Communication
  - Unified CM
  - Call processing
- TelePresence networks
- Integration
- Scalability
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- Best practices
- Summary
Unified CM
Endpoints and Applications

- CTMS
- MCU’s, TS
- MXE 5600
- Ad-hoc Conf
- CTS Series
- CUVA, Jabber, 99xx, Cius
- E20, EX, C-Series
- Profile Series

Unified CM

SIP trunk
3rd Party PBX
3rd Party Video Endpoints

SCCP
SIP
Unified CM Call Resolution

Route Patterns

**Route Pattern**
- Matches dialed number for external calls
- Points to a route list for routing
- Performs digit manipulation (optional)

**Route List**
- Points to prioritized route groups
- Performs digit manipulation (opt)

**Route Group**
- Points to the actual devices
  - Distribution algorithm

**Devices**
- Gateways (MGCP, SCCP, H.323)
- Gatekeeper (H.323)
- Trunk (H.323, ICT, SIP)

**Configuration Order**
- 1st Choice
- 2nd Choice
Unified CM Call Routing
Call search spaces, Partitions, Route lists and Route groups

CSS’s
- Internal
- Unrestricted (No Blocks)

Partitions
- BlockedPSTN
  - 9.[2-9]XXXXXX
  - 9.011!
  - 9.011!#

OnCluster
- All IP Phone DNs

Route Lists
- US_pstn_part
  - 911
  - 9.911
  - 9.[2-9]XXXXXX
  - 9.011!
  - 9.011!#

Route Groups
- JFK RG
- SFO RG

Location specific gateway selection through Local Route Group set on device pool
Site specific CSS still required for intra-site
Agenda

- Introduction
- Unified Communication
- TelePresence networks
  - Cisco Video Communication server (VCS)
  - Call processing
- Integration
- Scalability
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- Integration case study
- Best practices
- Summary
Video Communication Server (VCS)

Overview

- SIP proxy registrar and H.323 gatekeeper
- Bandwidth management
- Call admission and control (CAC)
- Protocol interworking
  - Between SIP and H.323
  - Between H.235 to BFCP screen or desktop sharing
  - Between IPv4 and IPv6 network interworking
- SIP Simple Presence support
Video Communication Server (VCS)
H.323 Gatekeeper & SIP Proxy/Registrar

- Registration
- Directory Lookup
- Number Translation
- Authentication
- Routing

Neighbor VCSs
- 3rd party devices
  - UCM v4.1 to Current
  - Microsoft OCS
  - Nortel
  - Other PBX mfrs
  - Other IP PBX mfrs

CUG Network

Internet
- DNS
- Enum
- Conference Bridge

CUG
Network
VCS Call Resolution

- Transforms
  - Uses priority
  - No domain in call information then used to add domain
  - Received calls are with IP address, then used to add domain
- Search rules
  - Uses priority
  - Source: Any, All Zones, Local Zone
  - Mode: Alias pattern match, any alias, any IP address
  - Target: Zones
- Call Policy
  - Call treatments
  - Call restrictions
VCS Call Routing

CPL: scripting for advanced routing rules

Search Rules: match based on pattern and route call to a destination Zone. Optional manipulation of dialed string

Transforms: preliminary manipulation of dialed string based on pattern

Transforms:
1. Transform: (\d{4})
2. Transform: (\d+)@1.1.1.1
3. RegEx: (\d+)
4. RegEx: (.*).+
5. RegEx: (\d{6})
6. RegEx: (\d)(\d)(\d)
7. RegEx: (.*).+@abc.com

Search Rules:
- AnyAlias
- AnyIPAddr
- RegEx: (\d+)
- RegEx: (.*).+
- RegEx: (\d{6})
- RegEx: (\d)(\d)(\d)
- RegEx: (.*).+@abc.com

Zones:
- Local Zone
  - 81234
  - 10.3.4.5
  - 202@a.com
- Neighbor Zone 1
  - 10.1.1.2
- Neighbor Zone 2
  - cucm.com
- DNS Zone 1
  - abc.com
- DNS Zone 2
  - 10.1.1.2

Remote VCS
CUCM
DNS SRV Lookup

Registered Endpoints

VCS incoming call
Agenda

- Introduction
- Unified Communication
- TelePresence networks
- Integration
  - Unified call control
  - Dial plan
  - Deployment Models
  - Trunks and zones
- Scalability
- Redundancy and Load Balancing
- Security
- Integration case study
- Best practices
- Summary
Unified CM Signaling
Architecture with VCS

- CTS Manager
- CTMS
- MXE 5600
- Ad-hoc Conf
- CTS Series
- CUVA, CUPC, 99xx, Cius
- CTS Series
- VCS
- 3rd Party Video Endpoints
- MXP, T, Profile Series
- SCCP
- SIP
- H.323

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Dial plan design
Call from endpoints in Unified CM to VCS endpoints

Caller: 10001
Called: 20001

Unified CM

Caller: 10001
Called: 20001

INVITE
To: 20001@10.10.10.100
From: 10001@10.10.10.1

INVITE
To: 20001@10.10.10.100
From: 10001@10.10.20.100

INVITE
To: 20001@10.10.20.100
From: 10001@10.10.10.1

INVITE
To: 20001@10.10.20.100
From: 10001@10.10.10.100

Server address or domain modifications, route calls on Called number or Domain

Called and Calling/Callback Number translations, simultaneous ring multiple endpoints

10.10.10.1
10.10.10.100
10.10.20.100
10.10.20.1
Dial plan design
Call from endpoints in VCS to CUCM endpoints

Call from endpoints in VCS to CUCM endpoints

INVITE
To: 10001@10.10.10.100
From:20001@vcs.cisco.com

INVITE
To: 10001@10.10.10.100
From:20001@vcs.cisco.com

INVITE
To: 10001@10.10.10.100
From:20001@vcs.cisco.com

Caller:20001
Called:10001

Unified CM
Cisco VCS

VCS routes based on Prefix or Domain

Called and Calling/Callback number transforms, DN based call routing, simultaneous ring

called and Called number or URI translations

10.10.10.1 10.10.10.100 10.10.20.100 10.10.20.1
## Dial plan considerations

<table>
<thead>
<tr>
<th>Numeric dial plan</th>
<th>URI based dial plan</th>
</tr>
</thead>
<tbody>
<tr>
<td>Unified CM, VCS, VCS Expressway</td>
<td>VCS, VCS Expressway</td>
</tr>
<tr>
<td>Numeric based call resolution</td>
<td>Alphanumeric and domain based call resolution</td>
</tr>
<tr>
<td>Number based manipulation</td>
<td>RegEx based manipulation</td>
</tr>
<tr>
<td>Called and Calling/Callback number</td>
<td>Contact header, To: and From: device URI manipulations</td>
</tr>
<tr>
<td>manipulation</td>
<td></td>
</tr>
<tr>
<td>PSTN, Cell phones and TDM calls</td>
<td>Supports IP networks</td>
</tr>
<tr>
<td>International, long distance call types</td>
<td>No concept of International or long distance</td>
</tr>
<tr>
<td>Domain concept is not always applicable</td>
<td>Requires Domain for calls and routing</td>
</tr>
<tr>
<td>for call routing</td>
<td></td>
</tr>
<tr>
<td>Class of service can be enforced to</td>
<td>Toll fraud prevented with dial rules</td>
</tr>
<tr>
<td>prevent toll fraud</td>
<td></td>
</tr>
<tr>
<td>Call restrictions enforced with class of</td>
<td>Call restrictions difficult (not possible to block</td>
</tr>
<tr>
<td>service (prevent 1-800 calls)</td>
<td><a href="mailto:hello@spam.com">hello@spam.com</a>)</td>
</tr>
</tbody>
</table>
Integration

- Unified call control
- Dial plan

Deployment Models

- Centralized call processing
  - Unified CM with VCS cluster
- Distributed call processing
  - Unified CM and multiple VCS clusters
  - Multiple Unified CM and multiple VCS clusters

- Trunks and zones
Deployment Models
Unified CM with VCS cluster

- Unified CM for CTS and UC endpoints
- VCS clusters for Video endpoints that support H323 and SIP
- SIP trunk between Unified CM and VCS
- Application services through Unified CM (Voicemail, PSTN)
- Video services through VCS (Multiway)
- VCS for H.323 network
- Interworking through VCS
- VCS for H.235, sRTP encryption
- VCS for H.239, BFCP presentation sharing
- DN based dial plan with Unified CM
- DN or URI based dial plan with VCS
- TMS for managing VCS and endpoints
Deployment Models
Unified CM and multiple VCS clusters

- Unified CM for CTS and UC endpoints
- VCS clusters for Video endpoints that support H323 and SIP
- Local VCS for Video endpoints
- Local MCU for local conferencing
- DN based dial plan with Unified CM
- DN and URI based dial plan with VCS
- Branch VCS neighbored to HQ Unified CM and VCS
- VCS provides protocol interworking
- Interworking for calls at local VCS
- Unified CM Regions and Locations CAC for CM endpoints
- VCS Pipes and links for bandwidth management
- Local VCS for VCS Expressway
Deployment Models
Multiple Unified CM and multiple VCS clusters

- Unified CM for CTS and UC endpoints
- Local VCS for Video endpoints that support H323 and SIP
- Local conferencing through Call agents
- DN based dial plan with Unified CM
- DN and URI based dial plan with VCS
- VCS provides protocol interworking
- Interworking for calls at local VCS
- Separate CAC within the call agents
- Local VCS for VCS Expressway
CUCM with mesh connection
Multiple Unified CM and multiple VCS clusters

- DN dial plan with Unified CM
- DN and URI dial plan with VCS
- Non overlapped dial plan with Unified CM and VCS
- Interworking to be done at the local VCS to optimize media path and for VCS Expressway connectivity
## Deployment models

### Enterprises using Deployment models

<table>
<thead>
<tr>
<th>Deployment Models</th>
<th>Enterprises who deploy them</th>
</tr>
</thead>
</table>
| Unified CM with VCS cluster                         | • Campus networks  
• Enterprises wanting to add Video to existing Unified CM networks  
• Educational institutions  
• Legal and consulting organizations                   |
| Unified CM and multiple VCS clusters                | • Large distributed enterprises,  
• Enterprises integrating video to existing Unified CM networks  
• Financial institutions  
• Healthcare deployment                                     |
| Multiple Unified CM and multiple VCS clusters       | • Large distributed enterprises, Globally distributed organizations  
• Enterprises integrating video to existing Unified CM networks  
• Global financial institutions  
• Global Multinational organizations that have manufacturing, services or other similar facilities |
Integration

- Unified call control
- Dial plan
- Deployment Models
- Trunks and zones
  - Trunks
  - Unified CM trunk parameters
  - Zones
Call routing between Unified CM and VCS

CUCM cluster

Translators
- Route Pattern
- Route List
- Route Group
- SIP Route Pattern

- Domain based routing
- Circular/Top down logic for route groups

Trunk 1
Trunk 2
Trunk 3

VCS cluster

Transforms
- Search rule 1 (priority 1)
- Search rule 2 (priority 2)
- Option ping for neighboring call agents

- Option ping for reachability
- Trunk can run on all nodes
Unified CM SIP trunk
SIP trunk configuration

• Allows trunks to send and receive calls on all CM node servers
• Provides optimized call processing
Unified CM SIP trunk Configuration

- **SIP trunk security profile**
  - System -> Security -> SIP trunk security profile
  - Fine tune trunk security configuration

- **SIP Profile**
  - Device -> Device settings -> SIP profile
  - Fine tune configuration for SIP line and trunk
Unified CM SIP trunk security profile
Configuration

- Security Mode
- Transport protocol
- Protocol ports
- Mobility and voicemail scenarios
- Voicemail MWI and other notifications
- Missed call list and voicemail mailbox redirect
Unified CM SIP profile
Call Bandwidth accounting

• Limitation: Unified CM didn’t accurately account call bandwidth

- Unified CM now can account call bandwidth using TIAS and AS
- Provides more granular visibility for call
- Various media streams can now be accounted by Unified CM for calls
Unified CM SIP profile
Header Information pass through

- Limitation: Unified CM only used information in the From header of the SIP Invite, calls from <name>@<IP address> were difficult to call back

```
INVITE sip:c9ff9cac-b763-a289-4573-4548c14d583f@10.35.201.228:49630;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 172.19.236.55:5060;branch=z9hG4bKddcba4b7560d7
From: <sip:89080007@172.19.236.55>;tag=532455-31190bdc-efd6-4005-b89d-c1245ccf75eb-25451912
To: <sip:89080004@172.19.236.55>
Remote-Party-ID: <sip:89080007@172.19.236.55;x-cisco-callback-number=89080007>;party=calling;screen=yes;privacy=off
Contact: <sip:89080007@172.19.236.55:5060;transport=tcp>;video;audio
```

- Solution: Unified CM 8.6 and later
  - Unified CM now can use the optional header (PAI, PPI, RPID) for calls
  - Missed call entry now can populate the actual URI of calling party
  - Provides ease for users to call back based on calling party information
Unified CM SIP profile
Domain in SIP calls

• Limitation: Unified CM only used IP address or host name of Unified CM in SIP requests, domain information was lost for calls from it

• Solution: Unified CM 8.6 and later
  • Unified CM now can replace IP address with Organizational Top Level Domain (Enterprise parameters) information in call signaling
  • Preserves domain information in URI for calls
Domain information in SIP calls

Caller: 10001
Called: kate@acme.com

Unified CM

INVITE
To: kate@acme.com
From: 10001@10.10.10.1

Endpoint sends call to CUCM

CUCM adds RHS of @ from OTLD configuration

•Unified CM adds domain in URI for calling party

INVITE
To: kate@acme.com
From: 10001@cisco.com

Caller URI can be used to callback

VCS

kate@acme.com
Callback: 10001@cisco.com

INVITE
To: kate@acme.com
From: 10001@10.10.10.1

INVITE
To: kate@acme.com
From: 10001@10.10.10.100

INVITE
To: kate@acme.com
From: 10001@10.10.20.100
Unified CM SIP profile
MTP if needed for Early offer calls

• Limitation: Unified CM added MTP for Early offer calls resulting in disabling video

• Solution: Unified CM 8.6 and later
  • Early Offer video calls possible and no MTP was added
  • MTP only included in case of DTMF mismatch
Unified CM SIP profile
Mid-call Send-receive attribute in SDP

• Limitation: No video possible after invoking supplementary services like Hold/Resume during call

Solution: Unified CM 8.6 and later
  • Send-Receive attribute allows to resume media after invoking supplementary services
Unified CM SIP profile
BFCP support

• Limitation: Unified CM did not support BFCP. BFCP information in call signaling had to be masked by far end

\[ \text{Trunk Specific Configuration} \]
- Reroute Incoming Request to new Trunk based on: Never
- RSVP Over SIP: Local RSVP
- Fall back to local RSVP
- SIP Rel1XX Options: Disabled
- Deliver Conference Bridge Identifier
- Early Offer support for voice and video calls (insert MTP if needed)
- Send send-receive SDP in mid-call INVITE
- **Allow Presentation Sharing using BFCP**

• Solution: Unified CM 8.6 and later
  • Unified CM supports BFCP for SIP line and trunk configurations
  • Only UDP based BFCP is supported
  • Unified CM only negotiates BFCP and does not terminate it
  • BFCP is established between endpoints
  • Interop script is needed to block TCP based BFCP
  • BFCP not supported for call through media devices
Unified CM SIP profile
Option ping

• Limitation: Unified CM did not support option ping. Far end availability was determined by SIP protocol timeouts

<table>
<thead>
<tr>
<th>Trunk Specific Configuration</th>
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</thead>
<tbody>
<tr>
<td>Reroute Incoming Request to new Trunk based on*</td>
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<td>Allow Presentation Sharing using BFCP</td>
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</tbody>
</table>

<table>
<thead>
<tr>
<th>SIP OPTIONS Ping</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable OPTIONS Ping to monitor destination status for Trunks with Service Type &quot;None (Default)&quot;</td>
</tr>
<tr>
<td>Ping Interval for In-service and Partially In-service Trunks (seconds)*</td>
</tr>
<tr>
<td>Ping Interval for Out-of-service Trunks (seconds)*</td>
</tr>
<tr>
<td>Ping Retry Timer (milliseconds)*</td>
</tr>
<tr>
<td>Ping Retry Count*</td>
</tr>
</tbody>
</table>

• Solution: Unified CM 8.6 and later
  • SIP Option Ping allowed to proactively determine if the far end is reachable
  • Verifies if the call agent is capable of servicing SIP call requests
**Unified CM SIP trunk Configuration**

<table>
<thead>
<tr>
<th>Destination Address</th>
<th>Destination Address IPv6</th>
<th>Destination Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>1* tgtmevcs1</td>
<td></td>
<td>5061</td>
</tr>
<tr>
<td>2 tgtmevcs2</td>
<td></td>
<td>5061</td>
</tr>
<tr>
<td>3 tgtmevcs3</td>
<td></td>
<td>5061</td>
</tr>
<tr>
<td>4 tgtmevcs4</td>
<td></td>
<td>5061</td>
</tr>
</tbody>
</table>

- Normalization script “VCS-interop” used to normalize SIP messages and mask TCP based BFCP signaling from VCS or other call agents
Zones, Subzones and Links
Relations between Local Zone and other VCS zones

VCS-C

Local Zone

Subzones

Traversal Subzone

Default Subzone

Neighbor Zone

Traversal Client Zone

Default Zone

DNS Zone

ENUM Zone

Neighbor Zone: Trunk to a remote system (VCS, CUCM, or other) using IP Address or domain name

Default Zone: For calls from an unknown zone

DNS Zone: Trunk to a remote system using DNS SRV

For VCS-E connectivity a client-server configuration is required
### VCS Zone Configuration

**Signaling port**
- **Mode**: On
- **Port**: 5060
- **Transport**: TCP
- **Accept proxied registrations**: Allow

**Authentication**
- **Authentication policy**: Do not check credentials
- **SIP authentication trust mode**: Off

**Location**
- **Peer 1 address**: 172.19.236.55
- **Peer 2 address**: 172.19.236.56
- **Peer 3 address**: 172.19.236.57
- **Peer 4 address**: 172.19.236.58
- **Peer 5 address**: 172.19.236.59
- **Peer 6 address**: 172.19.236.60

**Advanced**
- **Zone profile**: Cisco Unified Communications Manager

**Neighbor information**
- Peer 1: 172.19.236.55:5060
- Peer 2: 172.19.236.56:5060
- Peer 3: 172.19.236.57:5060
- Peer 4: 172.19.236.58:5060

**Neighbor availability status**
- Peer 1: Active
- Peer 2: Active
- Peer 3: Active
- Peer 4: Active

**Profile for different integrations**
• The highlighted configurations are the optimized zone profile settings for “Cisco Unified Communication Manager”
Agenda

- Introduction
- Unified Communication
- TelePresence networks
- Integration
- Scalability
  - SME
  - Directory VCS
- Redundancy and Load Balancing
- Security
- Integration case study
- Best practices
- Summary
SME Deployments
VCS deployments with SME

1. VCS as SME leaf cluster as a common Video Call agent for Enterprise, similar to CUCM for CTS endpoints
2. VCS with the CUCM leaf cluster
3. VCS in remote branches of CUCM leaf clusters
Scaling Deployments
VCS deployments with Directory VCS

- Directory VCS for hierarchy
- No Media Processed by Directory VCS
- No interworking calls
- No endpoints or conference bridges registered to this VCS
- Not involved with call once call is resolved
Agenda

- Introduction
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- Integration
- Scalability

- Redundancy and Load Balancing
  - DNS SRV
  - Option Ping
  - Unified CM Trunks
  - VCS Zones

- Security
- Integration case study
- Best practices
- Summary
DNS SRV
Format SRV records for SIP and H.323 (RFC 2782)

- Name of the service
  - _sip._tcp.example.com

- Protocol and domain name (TCP, UDP...)
  - DNS Class. Always “IN”
  - Priority: Lowest priority means “preferred”. If connection fails, client fallback to the higher priority record
  - Weight: for records with same Priority, it is used for load-balancing
  - Port: TCP or UDP port for the service
  - Targeted: hostname or IP Address for the host Providing the service

- DNS Time-To-Live: how much time the server caches the record before it flushes the cache
  - 86400 IN SRV 10 60 5060 vcs.example.com
SIP Trunk with DNS SRV

- SIPS or SIP service
- TCP or UDP protocol
- DNS SRV records provide load balancing and redundancy
- DNS server needs to be highly available
- Options ping for rechability
SIP Trunk with Option Ping

- Option Ping for reachability
- Trunks In-Service if response received
- Trunks Out-of-service if 408 request timeout, 503 service unavailable or no response
- Calls from CUCM not sent to out-of-service servers
- Avoids SIP message retry and timeouts
- Can be used for all nodes in trunk
- DNS SRV queries and all hosts of the SRV responses
Unified CM Trunks
Call routing with Route lists and Route groups

- Route lists use Route Groups or Local route group
- Route lists can be used by Route patterns
- Route group members can be multiple trunks
- Distribution algorithm in Route Group is Top down or Circular
- Trunks can be SIP or H.323
- Trunk to the same destination provide basic redundancy
VCS zones
Call routing with Search rules and Zones

- Search rule priority for high availability
- Calls sent with circular logic if Search rule priority is same
- Multiple zones provide high availability
- Multiple Zones to reach all Unified CM servers
- Load balancing in Zones through bandwidth however uses only top down call logic
- External DNS servers with DNS or DNS SRV for load balancing
## High Availability

### Unified CM

<table>
<thead>
<tr>
<th><strong>Trunks</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>• Multiple nodes (IP address/DNS hosts)</td>
</tr>
<tr>
<td>• DNS SRV (multiple hosts)</td>
</tr>
<tr>
<td>• Different signaling ports</td>
</tr>
<tr>
<td>• Route local</td>
</tr>
<tr>
<td>• Option ping</td>
</tr>
</tbody>
</table>

### Zones

<table>
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<tr>
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<td>• Multiple Nodes ((IP address/DNS hosts)</td>
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<tr>
<td>• Option ping</td>
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</tbody>
</table>

### VCS

<table>
<thead>
<tr>
<th><strong>Route List</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>• Multiple trunks in route groups</td>
</tr>
<tr>
<td>• Route groups</td>
</tr>
<tr>
<td>• Local route groups (With AAR)</td>
</tr>
<tr>
<td>• Run on all nodes</td>
</tr>
<tr>
<td>• Option ping (on trunks)</td>
</tr>
</tbody>
</table>

### Search rules

<table>
<thead>
<tr>
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<tbody>
<tr>
<td>• Multiple rules</td>
</tr>
<tr>
<td>• Priority</td>
</tr>
</tbody>
</table>
## Load Balancing

<table>
<thead>
<tr>
<th>Unified CM</th>
<th>Trunks</th>
<th>Route List</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>• Multiple nodes (IP address/DNS hosts)</td>
<td>• Multiple trunks in route groups</td>
</tr>
<tr>
<td></td>
<td>• DNS SRV (Server configurations)</td>
<td>• Run on all nodes</td>
</tr>
<tr>
<td></td>
<td>• Route local</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>VCS</th>
<th>Zones</th>
<th>Search rules</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>• Multiple zones</td>
<td>• Multiple rules</td>
</tr>
<tr>
<td></td>
<td>• Capacity using bandwidth</td>
<td>• Priority</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• Equal priority provides circular logic</td>
</tr>
</tbody>
</table>
Agenda

- Introduction
- Unified Communication
- TelePresence networks
- Integration
- Scalability
- Redundancy and Load Balancing
- Security
- Integration case study
- Best practices
- Summary
Security

- **TLS**
  - Endpoint and trunk signaling

- **dTLS**
  - Endpoint Encryption negotiation
  - Applicable for CTS endpoints
  - Supports 23 bit or 80 bit encryption with Unified CM

- **SRTP**
  - Only for Voice with Unified CM
  - VCS supports voice and video calls
  - Supports 80 bit encryption for VCS endpoints

- **H.235**
  - VCS support
  - VCS does interworking for SIP SRTP calls
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Case Study

- Medium sized organization
- Unified CM cluster
- VCS Clusters with VCS Expressway

Voicemail: out-of-dialog Refer and header pass through
- MWI: Unsolicited Notify
- Supplementary services: Refer
- Callback: header pass through
- Presentation channel: enable BFCP
- Bandwidth accounting: TIAS and AS
- Calls: Early offer and no MTP
- DTMF: RFC2833
- Script: vcs-Interop
- Trunk availability: Option ping

Nat Traversal with SIP
- TURN/ICE for optimizing media

Zone Profile: Cisco Unified Call Manager
- Simultaneous ring: FindMe
- Presentation channel: BFCP, H239
- Security: SRTP, H235
Case Study

•Dialplan

•Dial plan: DN based
•Voicemail: out-of-dialog Refer and header pass through
•Callback: header pass through
•Domain: Enable use of OTLD
•DTMF: RFC2833

•External Connectivity with SRV records

•Dial plan: URI based
•DN to URI translation: FindMe
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Best practices

- Endpoints
  - Register Cisco IP phones, E20, EX series and C series devices to Unified CM
  - H323 endpoints register with VCS
  - If SRTP for calls is needed, register E20, EX series and C series endpoints to VCS

- Trunks
  - SIP trunk between the Unified CM and VCS

- Dial plan
  - Numeric dial plan with Unified CM
  - Alphanumeric dial plan with VCS

- FindMe
  - Translate Numbers to URI
  - Non overlapped DN numbers for VCS endpoints

- Number normalization
  - Unified CM does calling and called number normalization
  - VCS uses Regex and FindMe to normalize called or calling number to Unified CM
Best practices

- Secure calls
  - Register Endpoints to VCS

- VCS for Interworking
  - H323 to SIP interworking
  - H235 to SRTP interworking
  - H239 to BFCP interworking
  - IPv4 to IPV6 interworking

- PSTN
  - Numeric dial plan through Unified CM

- Conferencing
  - Unified CM based conferencing using MCUs
  - Supports Ad-hoc and Scheduled
  - Multiway conferencing with VCS
Summary

- Unified CM supports TelePresence, Video and UC endpoints with a large scale
- VCS supports H323 video devices
- SIP trunks between Unified CM and VCS clusters
- VCS does Interworking between protocols
- Unified CM and VCS enhancements provide calls between TelePresence and Video and UC endpoints
- The latest release of Unified CM 8.6.2 and VCS 7.0 provide the greatest solution for video and TelePresence
- Numeric dial plan is easy to normalize and provides PSTN connectivity
- URI dialing if needed is supported in the VCS network
- VCS Expressway for NAT traversal and external connectivity
Recommended Reading
BRKCCT-2030

SIP Trunking
Migrating from TDM to IP for Business to Business Communication

Christina Hattingh
Darryl Sladden
ATM Zakaria Swapan

ciscopress.com
Recommended Reading

Please visit the Cisco Store for suitable reading.
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- Or use the Cisco Live Mobile App to complete the surveys from your phone, download the app at www.ciscolivelondon.com/connect/mobile/app.html

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2. Download the app or access the mobile site
3. Log in to complete and submit the evaluations
Thank you.