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SIP Trunking using CUCM and Cisco Session Border Controllers

BRKUCC-2073

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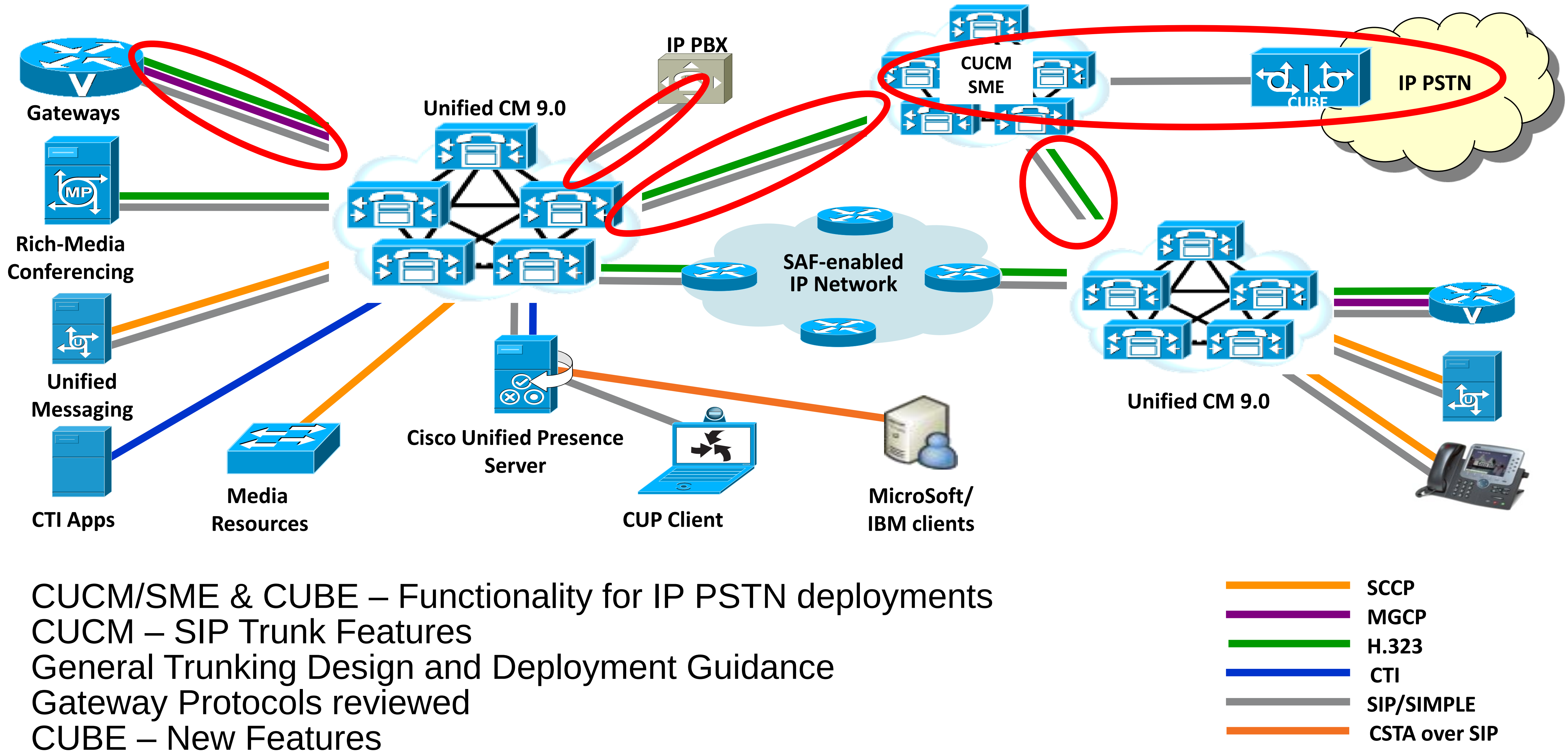
Aspects of SIP Trunking Covered in this Presentation



CUCM/SME & CUBE – Functionality for IP PSTN deployments
CUCM – SIP Trunk Features
General Trunking Design and Deployment Guidance
Gateway Protocols reviewed
CUBE – New Features

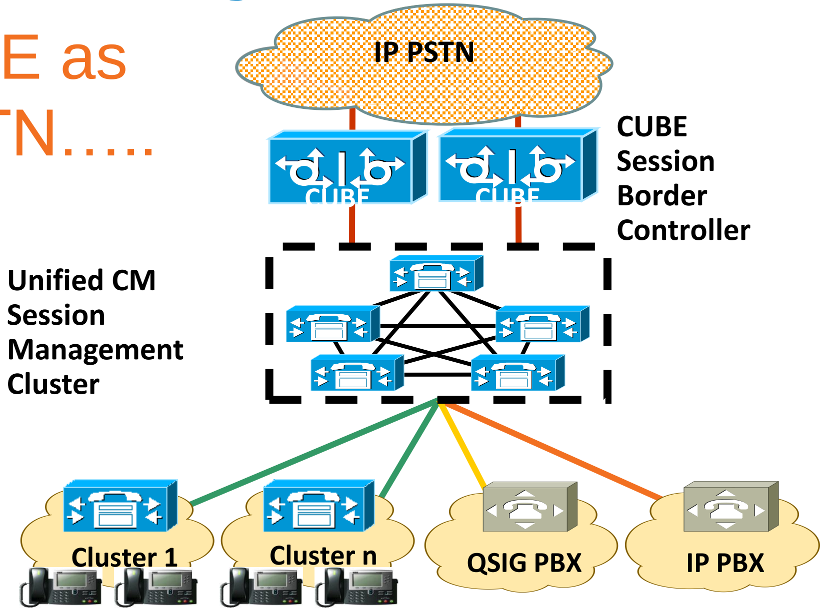
Agenda

- CUCM and CUBE – Functionality for IP PSTN Deployments
- CUCM SIP Trunking Features
  - SIP Trunk Signalling and Basic Operation
  - SIP Trunk – Load Balancing, Availability & Redundancy
  - SIP Trunk Codec Negotiation – Audio Codec Preference Lists
  - SIP Trunk Security
  - QSIG over SIP
  - SIP Normalisation and Transparency
- SIP Trunk Design Considerations
- Gateway Integration
- Cisco Unified Border Element (CUBE)

CUCM and CUBE – IP PSTN Connectivity

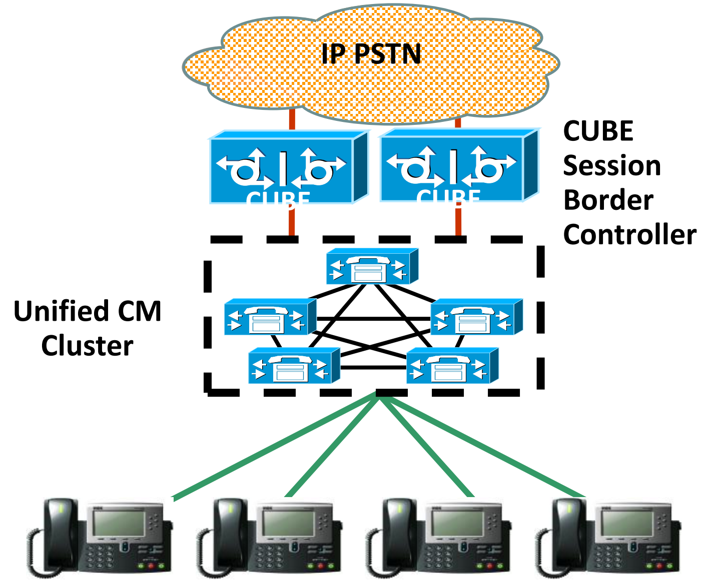
Cisco always recommends the deployment of CUBE as an SBC when connecting from CUCM to an IP PSTN…..

…….**Why ?**



1) Because Border Control and Phone/ Session Management have distinct differences in terms of their functional requirements and neither product can provide <u>all</u> of the functionality that the other product has…

2) The combined functionality of CUCM and IOS CUBE provides a extended feature set that benefits any UC deployment………..



CUCM and CUBE – Comparison

CUCM and SME – Unique Functional Characteristics

Reside in a trusted Environment i.e. In the Enterprise

GUI Based – Easy to use with highly extensible provisioning, call routing, dial plan and digit manipulation tools e.g. :

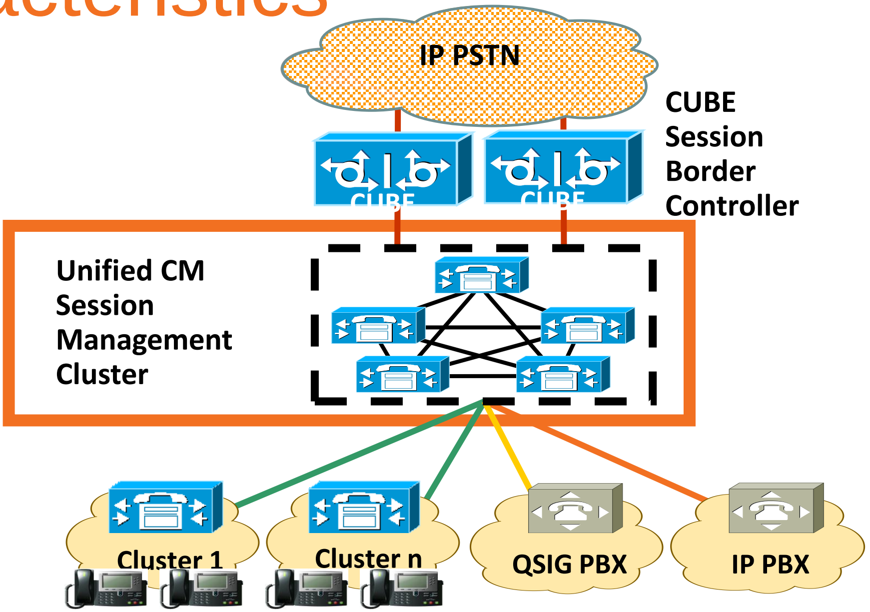
Can easily configure and manage 10's of 1000s of Route Pastterns, Number Translations & Transformations for Dial Plan Normalisation, Forced On Net Calls; Tail End Hop Off etc.

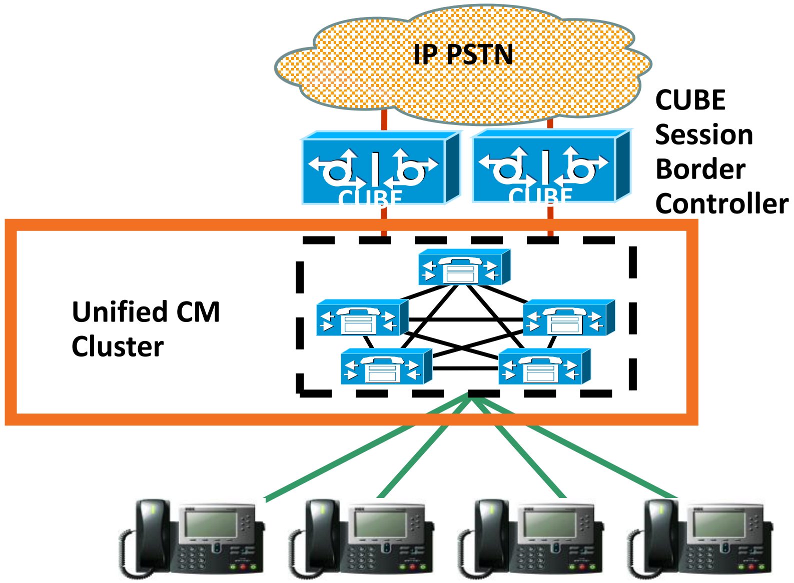
Capability to provision and manage up to 80,000 devices, 2100 Trunks – SCCP, H323, SIP, MGCP

URI based Call Routing (e.g. bob@cisco.com)

A call control entity – no media flow through – Combines Media Flow Around with sophisticated Call Admission Control mechanisms

Voice, Video, Encryption and QSIG feature support

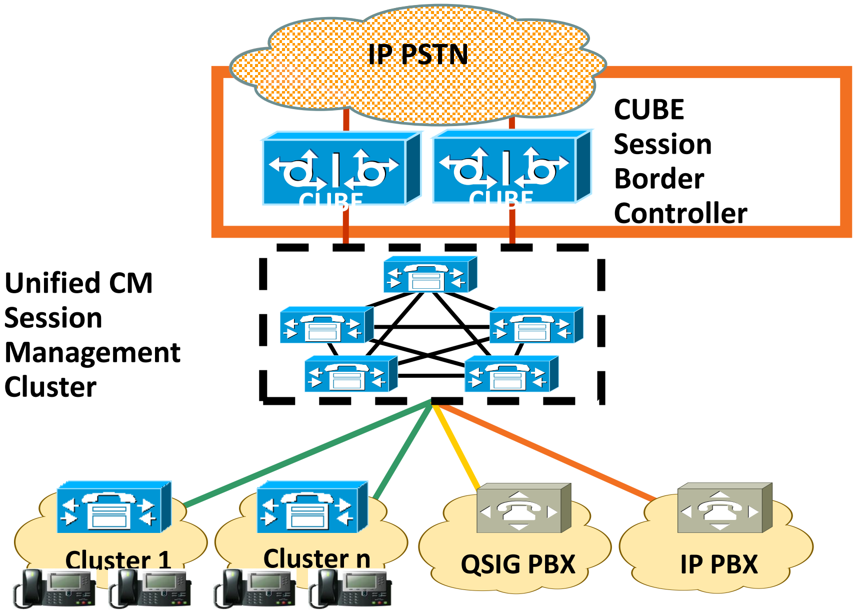


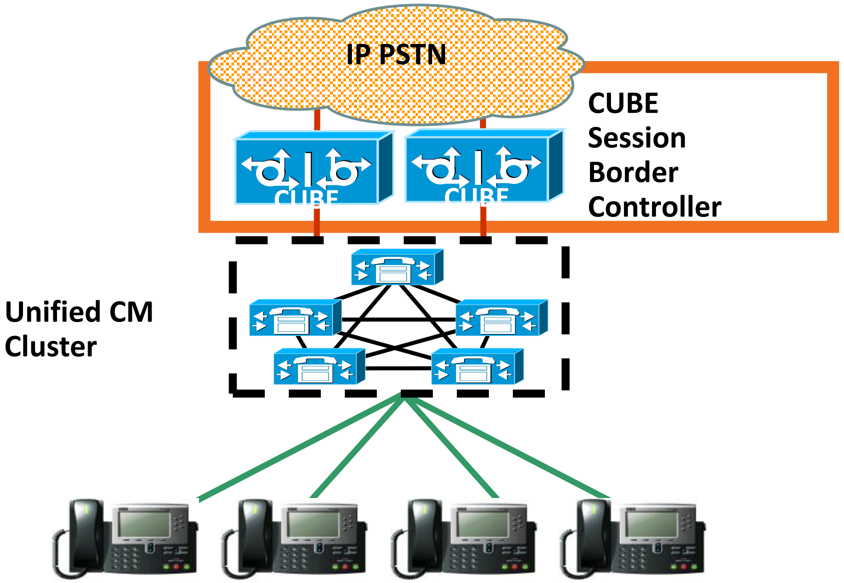


CUCM and CUBE – Comparison

CUBE – Unique Functional Characteristics

- Provides a security demarcation (border) between the trusted Enterprise network and un-trusted Public network
- Provides hiding of internal Enterprise IP addresses presenting a single IP address for signalling and media to the outside world
- Has built in tools to manage common vulnerability exploits, prevent Denial Of Service attacks and detect malformed packets. e.g. Intrusion Prevention System, IOS Firewall.
- A signalling and media control point -Typically uses Media Flow Through. Advanced Media inter-working features - background noise cancellation, QOS marking, sophisticated Interface Queuing mechanisms

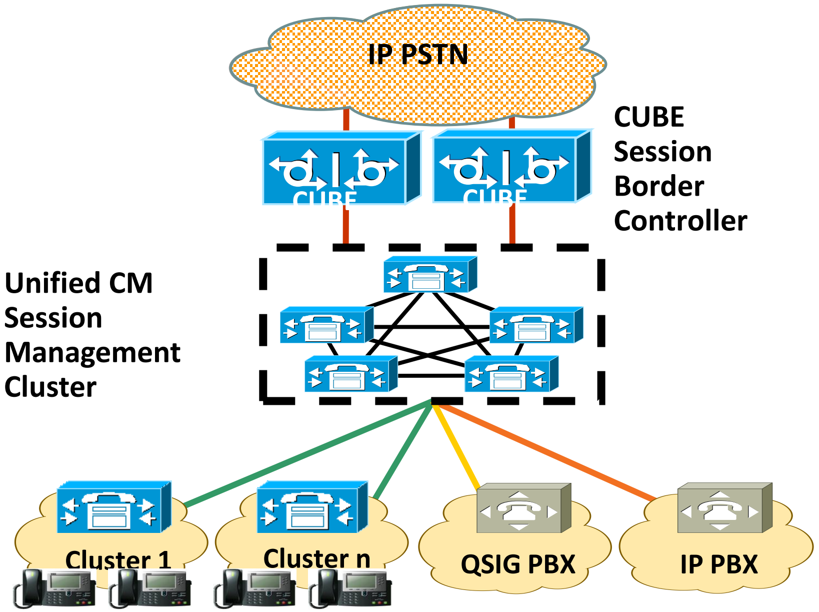


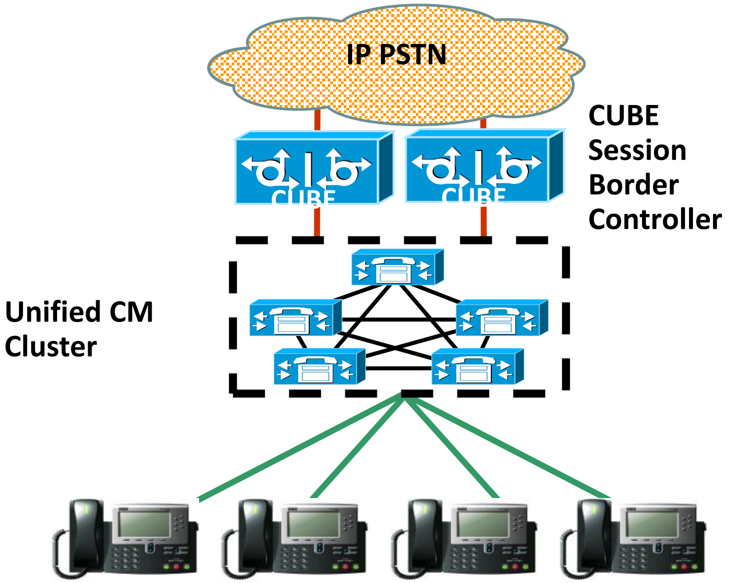


CUCM and CUBE

Common Feature Set

Feature	CUCM	CUBE
Protocol Translation	SIP H323 MGCP	SIP H323
<u>Scripting for SIP Normalisation and Transparency</u> Modification of SIP Messages Modification of SDP Content Storage and re-use of values	Very Powerful Yes Yes Yes	Good Yes Yes RegExp
<u>DTMF translation</u> RFC 2833 NTE, KPML, Unsolicited Notify, H245 Signal, H245 Alpha, Cisco NTE	Yes	Yes
Voice Transcoding	Yes	Yes

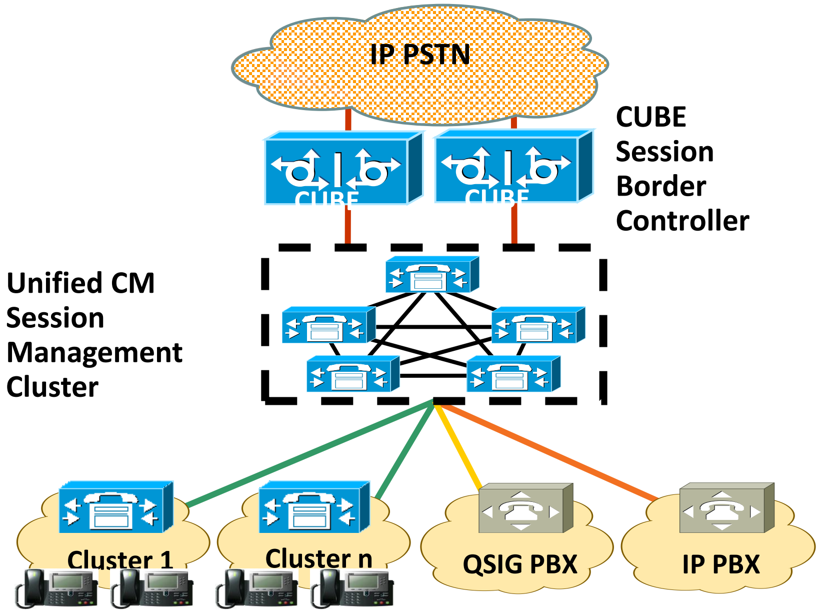


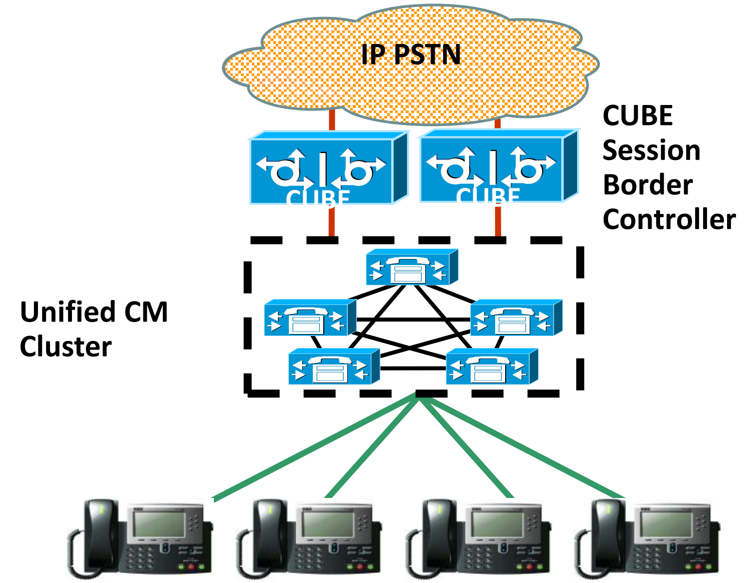


CUCM and CUBE

Common Feature Set

Feature	CUCM	CUBE
Media Flow Around (MFA)	Yes	Yes
Media Flow Through (MFT)	No	Yes
Call Admission Control		
Basic Call Counting	Yes	Yes
Per Call Bandwidth Aware Call Counting	Enhanced Locations CAC	Yes
RSVP	Yes	Yes (MFT only)
User Interface	GUI	GUI & CLI
Basic Tasks	GUI	GUI
Complex Tasks	GUI	CLI
Managing 1000s of entries	Very Good	Fair

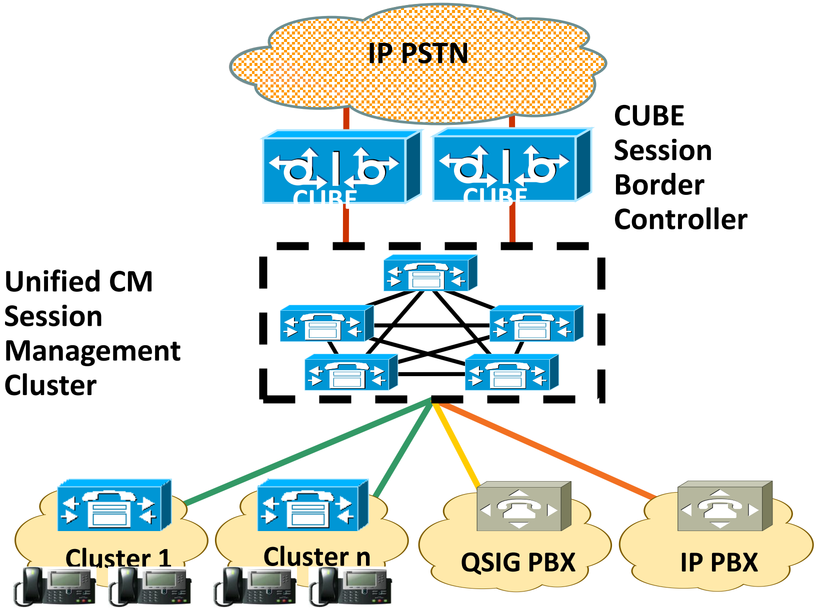


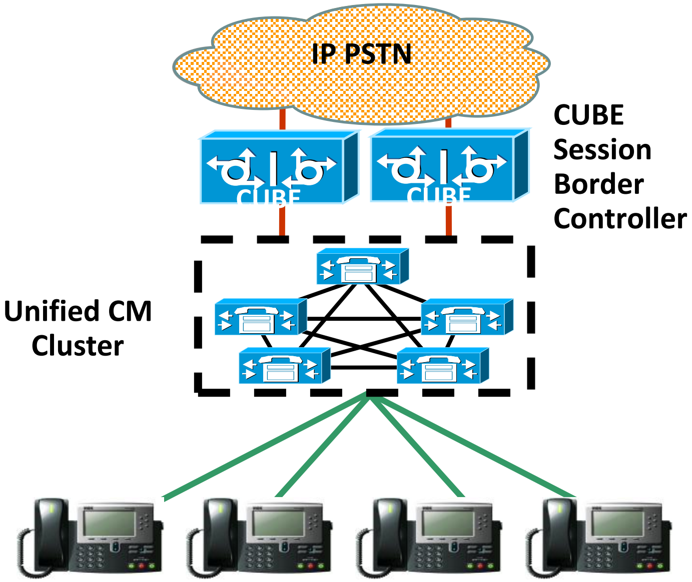


CUCM and CUBE

Common Feature Set

Feature	CUCM	CUBE
Path Replacement	QSIG based	SIP based
Secure RTP & TLS	Yes	Yes
Connections only from Trusted IP Addresses	Yes	Yes
OPTIONS Ping Send and Receive	Yes	Yes
Off Box stateful redundancy for SIP signalling	Planned	Yes
Digit Manipulation for Dial Plan Normalisation. Forced On Net, Tail End Hop Off Can easily manage very large dial plans	Very powerful Yes	Good No
External Call Control /Policy interface	Yes CURRI	Yes UC Services API

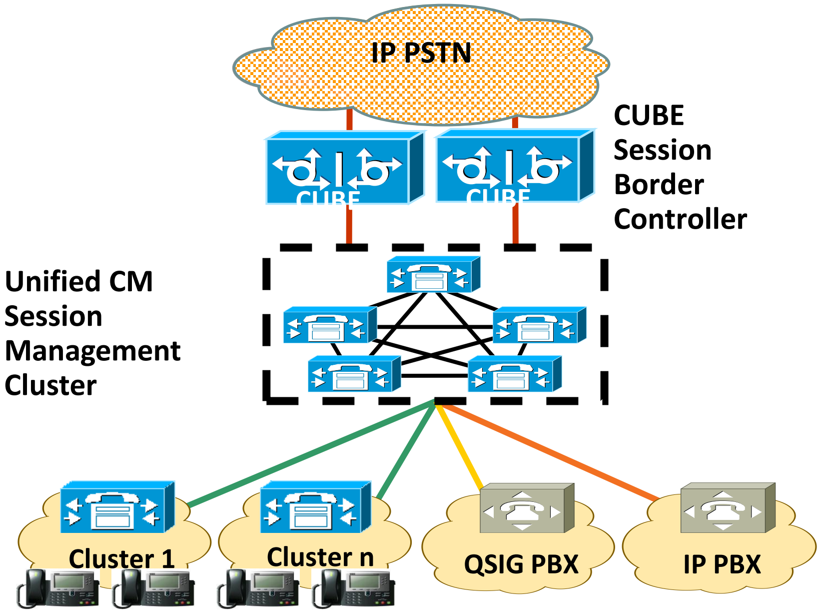


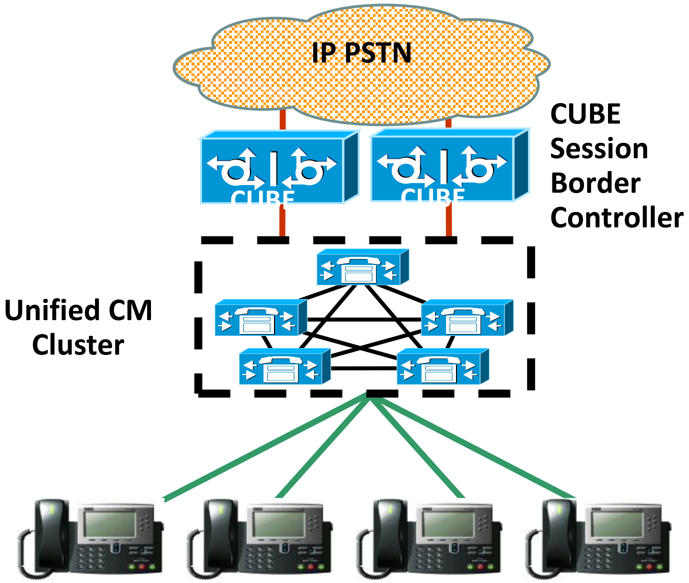


CUCM and CUBE

Extended Feature Set

Feature	CUCM	+ CUBE
Voice Conferencing	✓	✓
Video Trans-rating	✓	✓
Phone Registration	✓	✓
Advanced Media Interworking (Noise Cancellation etc)		✓
TDM Voice Interfaces (Analogue, Q931 and QSIG)		✓
WAN Interfaces		✓
QSIG Features e.g. Call back on Busy	✓	
Voice XML		✓

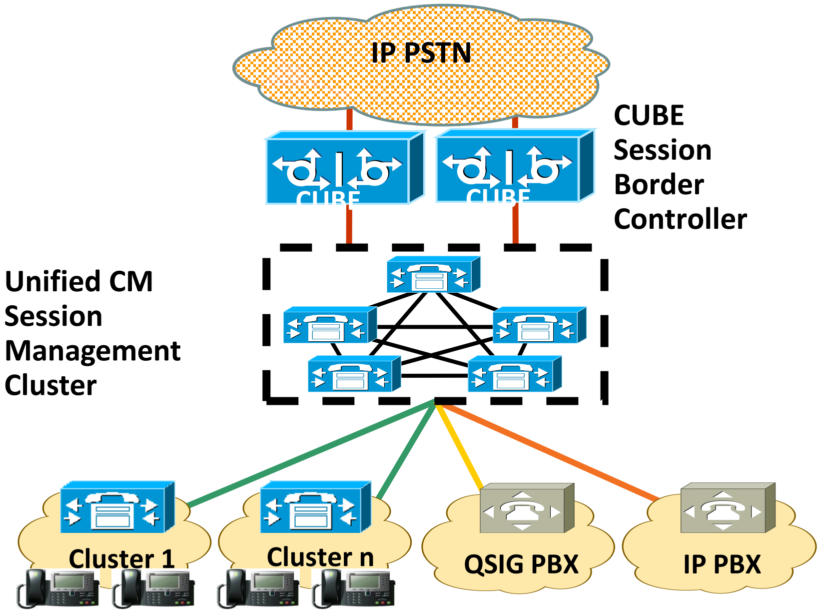


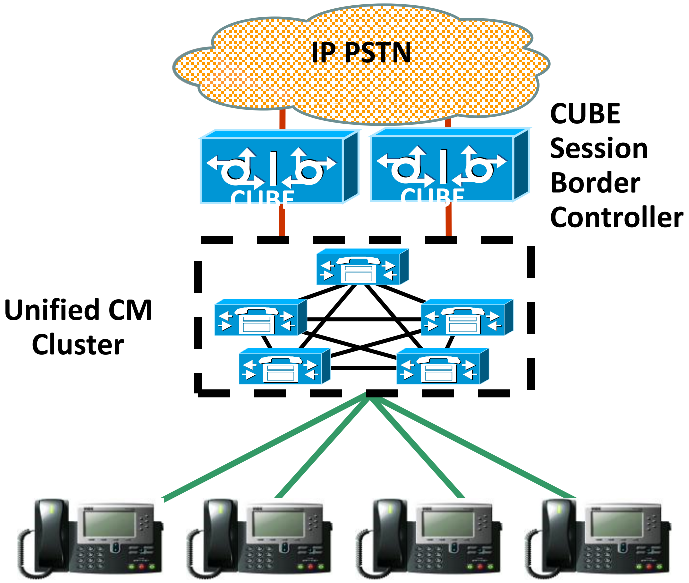


CUCM and CUBE

Extended Feature Set

Feature	CUCM +	CUBE
Registration and management of Internet based Phones with encryption	✓	
Survivable Remote Site Telephony (SRST)		✓
Gatekeeper Functionality		✓
Advanced IP Routing Suite		✓
Firewall		✓
Intrusion Prevention System		✓
Service Advertisement Framework (SAF)	✓	✓
SIP Pre-conditions for RSVP based CAC	✓	✓





CUCM SIP Trunking

Agenda

- –Overview of CUCM SIP Trunk Features
- –SIP Trunk Signalling and Basic Operation
- –SIP Trunk – Load Balancing, Availability & Redundancy
- –SIP Trunk Codec Support – Audio Codec Preference Lists
- –SIP Trunk Security
- –QSIG Over SIP
- –SIP Normalisation and Transparency

CUCM SIP Trunking Features - Overview

CUCM 8.5 SIP Trunk Features :

- Run on All Active Unified CM Nodes
- Up to 16 Destination Addresses
- SIP OPTIONS Ping
- SIP Early Offer for Voice & Video (Insert MTP if needed)
- QSIG over SIP
- SIP Normalisation and Transparency

CUCM 8.6 SIP Trunk Feature :

- REFER Transparency

CUCM 9.0 Features :

- Audio Codec Preference Lists
- SIP Trunk “Accept Received Codec Preference”

CUCM SIP Trunking

–Overview of CUCM SIP Trunk Features

–SIP Trunk Signalling and Basic Operation

–SIP Trunk – Load Balancing, Availability & Redundancy

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–SIP Trunk Security

–QSIG Over SIP

–SIP Normalisation and Transparency

SIP Trunk Signalling and Basic Operation

SIP Messaging – Delayed and Early Offer

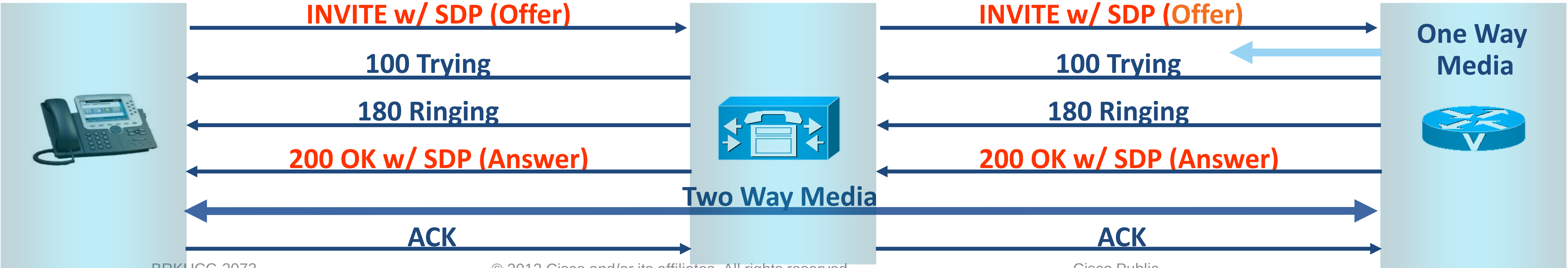
SIP Early Offer

Information about the calling device's media characteristics are sent with its initial SIP INVITE message – The media characteristics are contained in the Session Description Protocol (SDP) body sent with the SIP INVITE – The "Offer" in the SDP body will contain the IP Address, UDP Port number, list of codecs etc. supported by the calling device

The called device selects which of the offered codecs it wishes to use for the call and returns its "Answer" in the SDP body of a SIP response – The Answer also contains the IP address and UDP port number etc of the called device

Once the Answer has been received and acknowledged two way media can be established

Early Offer is widely used (particularly by Service Providers…….)



SIP Trunk Signalling and Basic Operation

SIP Messaging – Delayed and Early Offer

SIP Delayed Offer

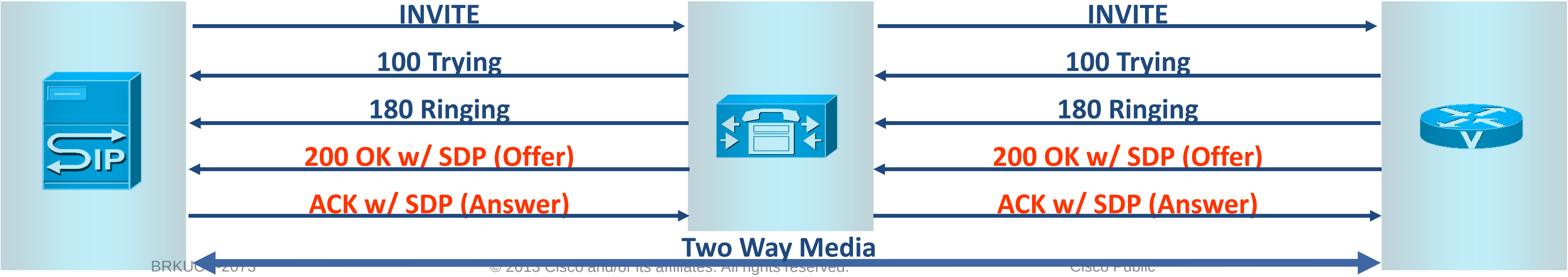
No information about the calling device's media characteristics are sent in the initial SIP INVITE

Instead the first set of media characteristics for the call are sent by the called device in the Session Description Protocol (SDP) body of the next reliable message (200 OK) – The called device's "Offer" will contain its IP Address, UDP Port number, list of codecs etc.

The calling device selects which of the offered codecs it wishes to use for the call and returns its "Answer" in the SDP body of a reliable SIP response (ACK) – The Answer also contains the IP address and UDP port number etc of the calling device

Delayed Offer is a mandatory part of the SIP standard – Most Service Providers prefer Early Offer

Ordinarily, the Offer or Answer cannot be sent with 100 Trying or 180 Ringing as 1XX messages are unreliable (unacknowledged)… This can be resolved using PRACK ….. discussed later…..



SIP Trunk Signalling and Basic Operation

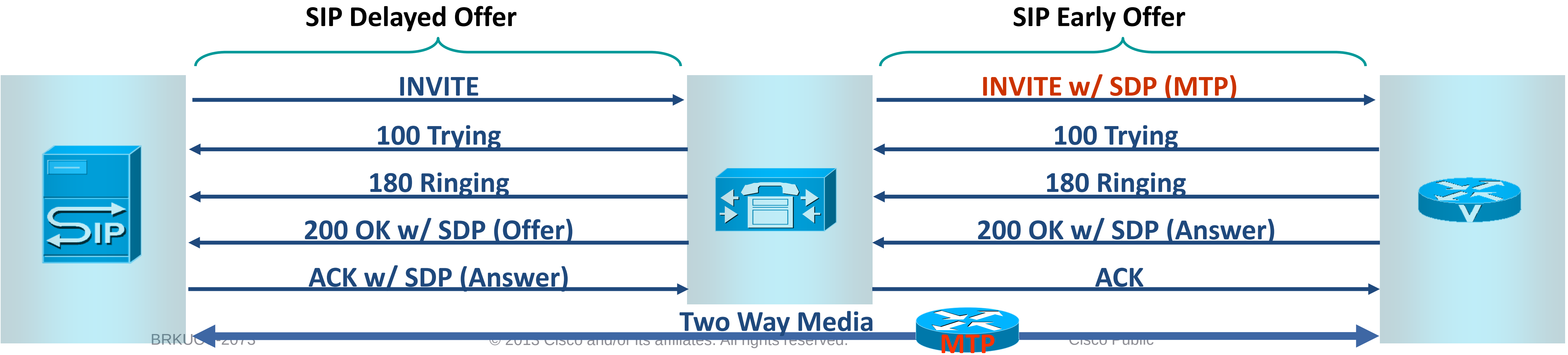
SIP Messaging – Delayed and Early Offer

Inbound SIP Delayed Offer to Outbound SIP Early Offer

So what happens when Unified CM receives an inbound call on a Delayed Offer Trunk and needs to onward route the call over a Early Offer Trunk ?

It does not have the calling device's media characteristics and it needs to send an Offer in SDP with the outbound INVITE…

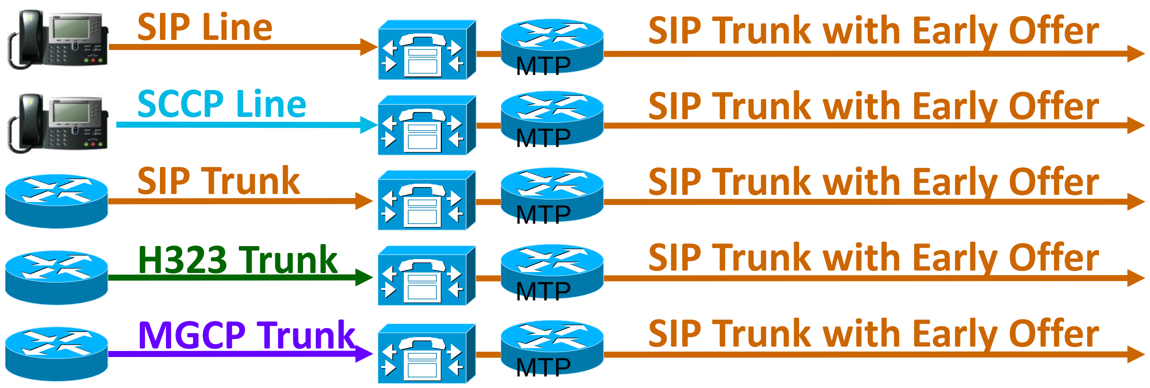
Solution – Insert a Media Termination Point (**MTP**) and use its media characteristics to create the Offer in SDP with the outbound INVITE

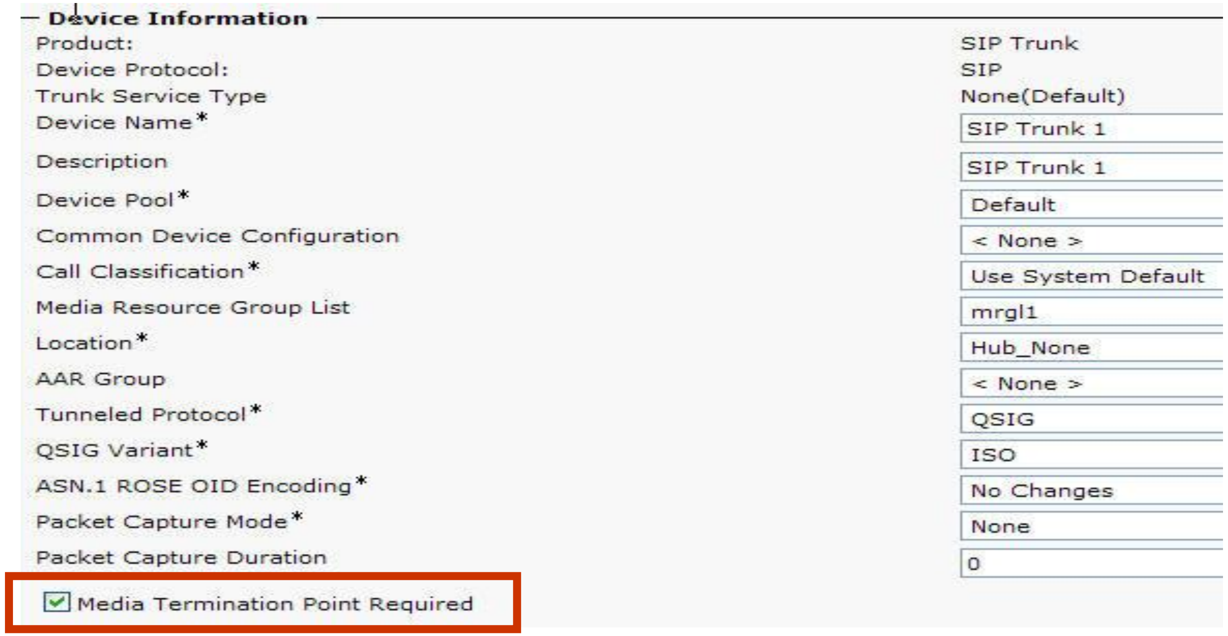


SIP Trunk Signalling and Basic Operation

SIP Messaging – Enabling SIP Early Offer – Method 1

SIP Trunk “MTP Required” Checkbox





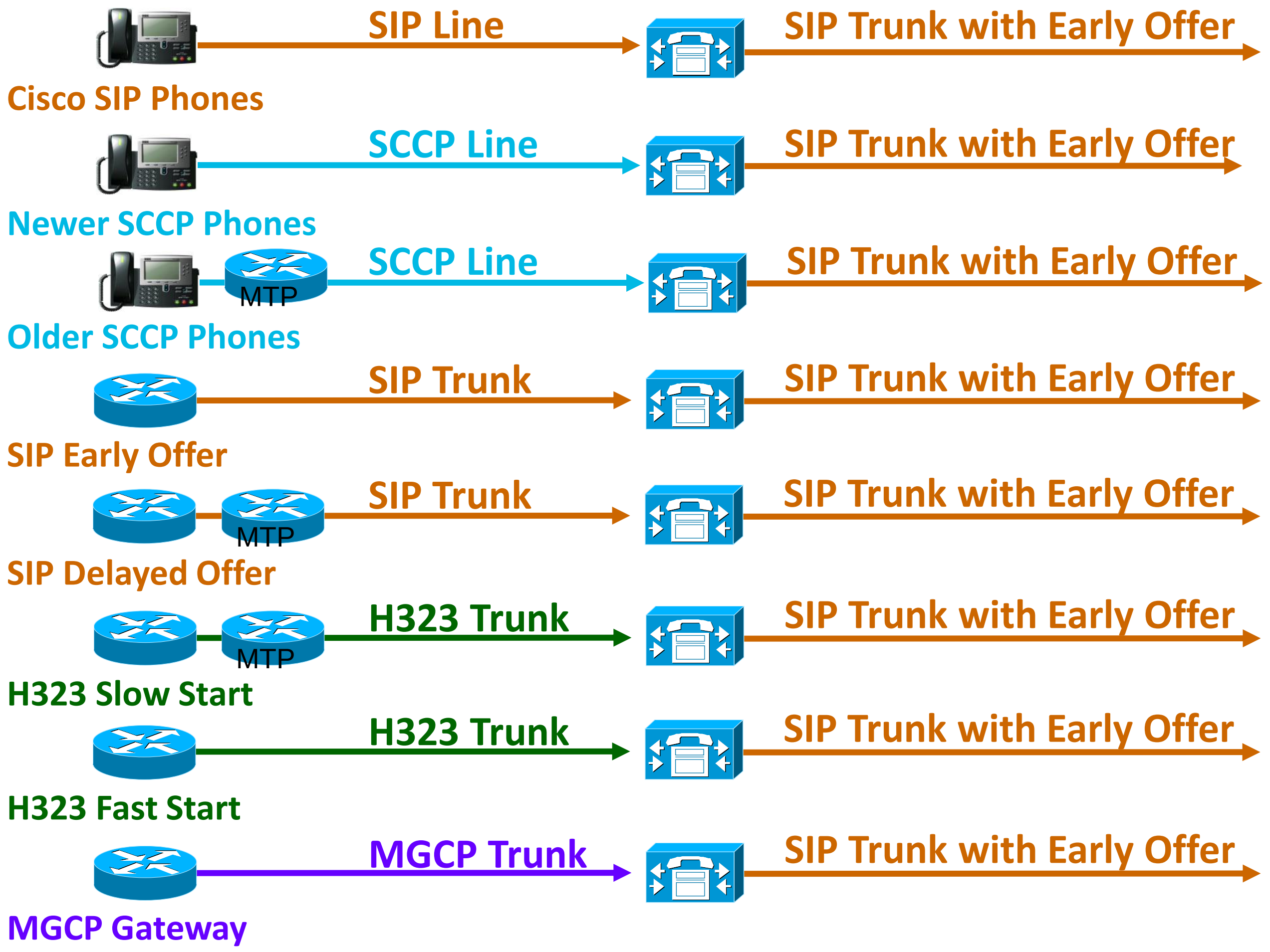
Using the “MTP Required” option :

SIP Early Offer Trunks use the Trunk’s Media Termination Point (MTP) resources, inserting an MTP into the media path for every outbound call – sending the MTP’s IP Address, UDP port number and codec in the SDP body of the initial SIP INVITE instead of those of the endpoint.

Disadvantages : MTPs support a single Audio codec only e.g. G711 or G729. The passthru codec is not supported excluding the use of SRTP and video calls. Since the Trunk’s MTPs are used rather than the calling device’s MTPs - The media path is forced to follow the signalling path.

SIP Trunk Signalling and Basic Operation

SIP Messaging – Enabling SIP Early Offer – Method 2



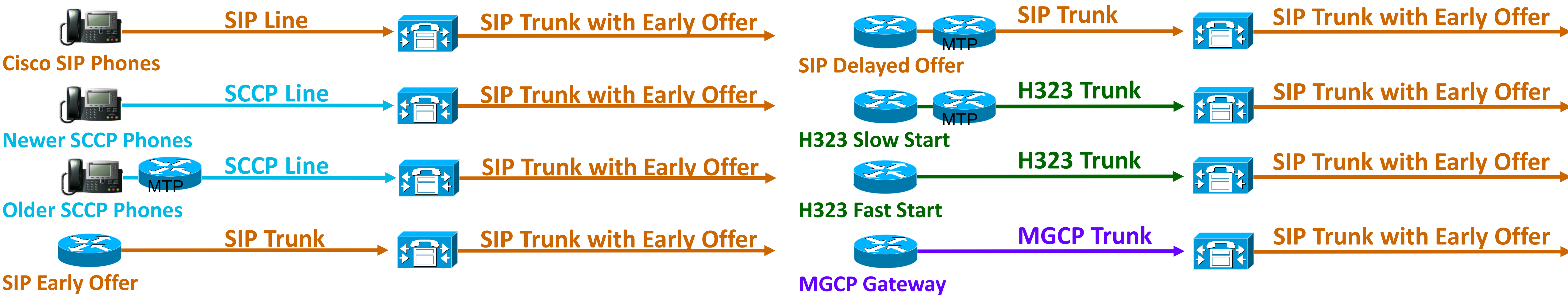
New SIP Profile checkbox **"Early Offer support for voice and video calls (insert MTP if needed)"**

For Calls from trunks and devices that can provide their IP Address, UDP port number and supported codecs - This information is sent in the SDP body of the initial SIP Invite on the outbound Early Offer Trunk. No MTP is used for the Early Offer

For Calls from trunks and devices that cannot provide Early Offer information – use the calling device's MTP resources (first) or the outbound trunk's MTPs (second) to create a SIP Offer for an unencrypted voice call. (SRTP and video can subsequently be initiated by the called device)

SIP Trunk Signalling and Basic Operation

SIP Messaging – Enabling SIP Early Offer – Method 2



Benefits of "Early Offer support for voice and video calls (insert MTP if needed)"

- Reduced MTP usage
- Single voice codec MTP limitation removed (by using the pass through codec)
- Voice codecs sent in SIP Offer based on calling device capabilities & region settings
- Video Calls supported
- Encryption supported
- Use of the Calling device's MTP rather than Trunk's MTP
  - Media does not have to follow the signalling path

SIP Trunk – CUBE DO to EO

SIP Messaging – Enabling SIP Early Offer – CUBE

- Configuration of codecs is required on CUBE to send in the outgoing Early Offer SIP INVITE when a Delayed Offer INVITE is received on the incoming leg
  - Early Offer on outgoing leg can be configured either globally or per dial-peer; dial-peer configuration overrides
  - Codec list to populate in outgoing Early Offer is configured on dial-peer
  - Important when CUCM cannot use Early Offer

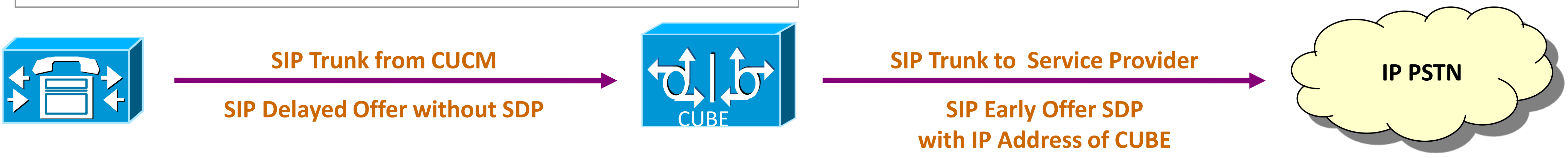
Dial-peer Configuration

```
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729a

dial-peer voice 4 voip
  destination-pattern 321....
  voice-class codec 1
  voice-class sip early-offer forced
  session target ipv4:9.6.3.22
```

Global Configuration

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



SIP Trunk Signalling and Operation – PRACK 1

SIP Early Media – Using Provisional Acknowledgement (PRACK)

SIP defines two types of responses: Final and Provisional.

Final responses convey the result of the processed request, and are sent reliably (i.e. they are acknowledged).

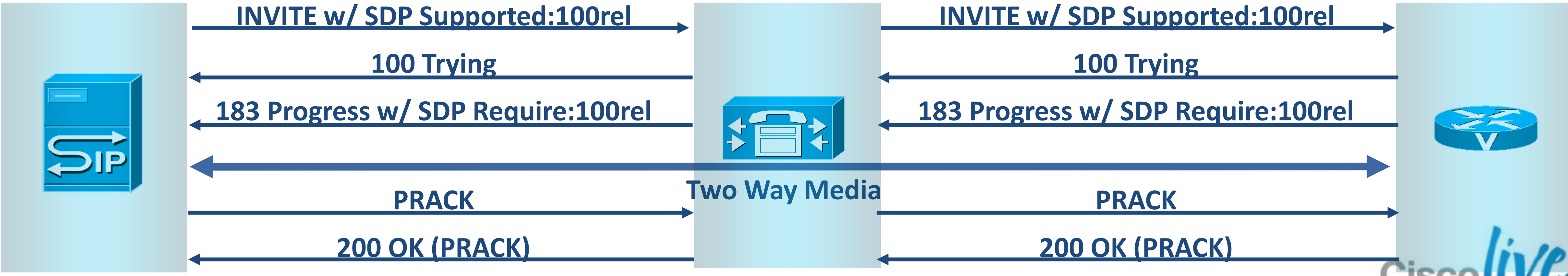
Provisional responses provide information on the progress of the request, but are not sent reliably – so the sender of a provisional response does know that it has been received.

To send an Offer or Answer with a provisional 1XX response – these responses must be sent reliably…..

PRACK – Provisional Reliable Acknowledgement is used to provide 1XX responses with reliability.

Diagram : Early Offer with Early Media

Early Offer ≠ Early Media



SIP Trunk Signalling and Operation – PRACK 2

SIP Early Media – Using Provisional Acknowledgement (PRACK)

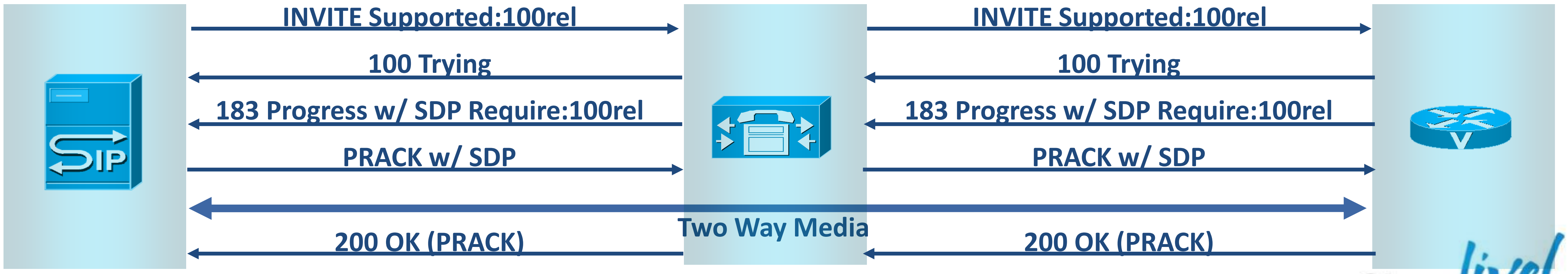
Like final responses, by using PRACK - 1XX messages will be periodically re-sent until their receipt is acknowledged by the receiver by sending a PRACK, which is also acknowledged by the 1XX sender.

Using PRACK can reduce the number of SIP messages that need to be sent before two way media can be established

PRACK is useful in situations where long Round Trip Times between SIP devices can cause a delay to media cut through or media clipping

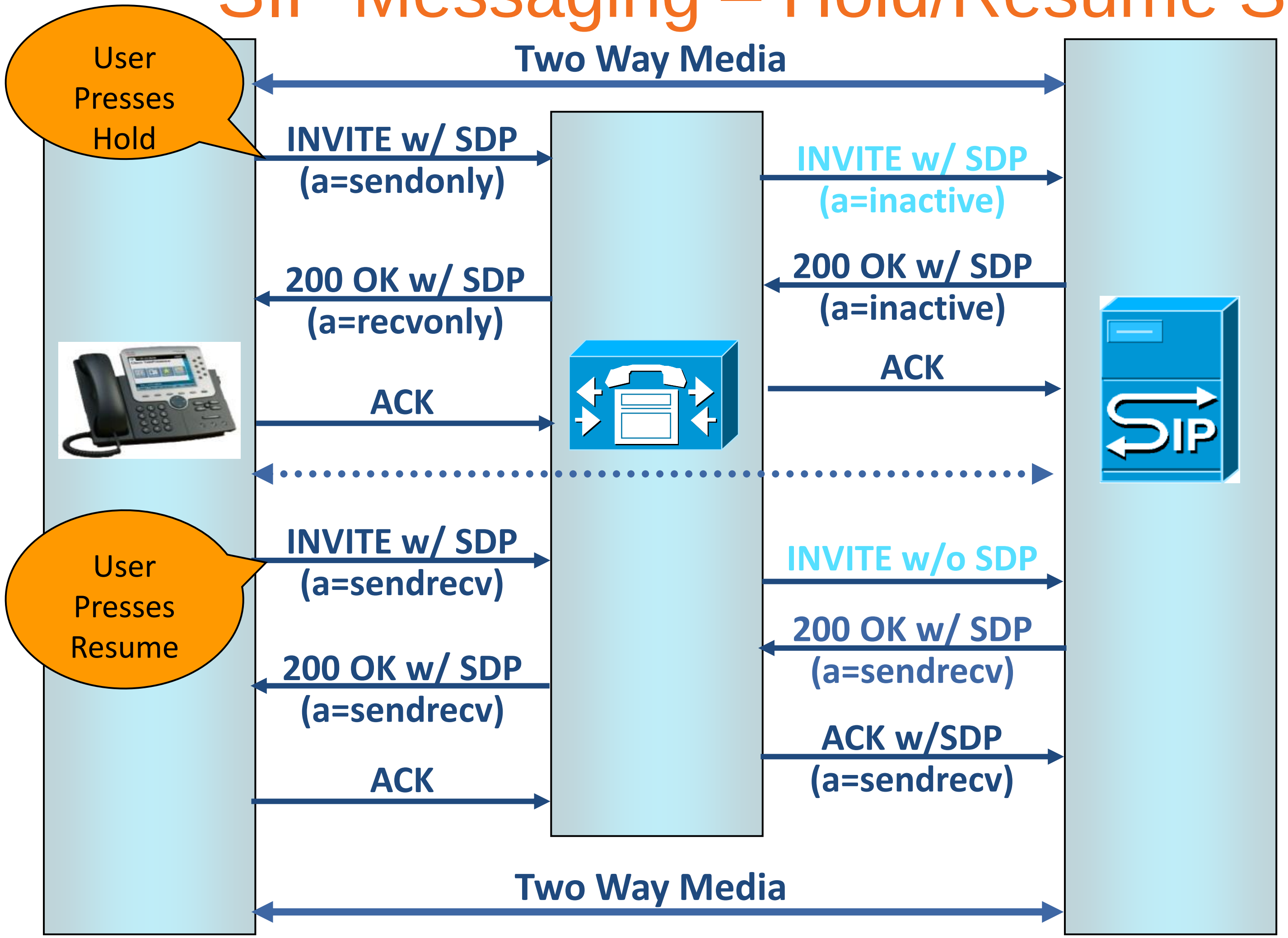
PRACK can be enabled on the SIP Trunk Profile by setting “SIPRel1XX Options” to enabled

Diagram : Delayed Offer with Early Media



SIP Trunk Signalling and Basic Operation

SIP Messaging – Hold/Resume Signalling



- To Hold and Resume a call CUCM sends :
  - a=inactive in SDP to stop media
  - Delayed Offer to start MoH or resume media
- The remote entity must respond with its full codec list and a=sendrecv when it receives Delayed Offer for MoH or to resume held call

The SDP media attributes are interpreted from the senders perspective :

<u>sendrecv</u> - Used to establish a 2-way media stream

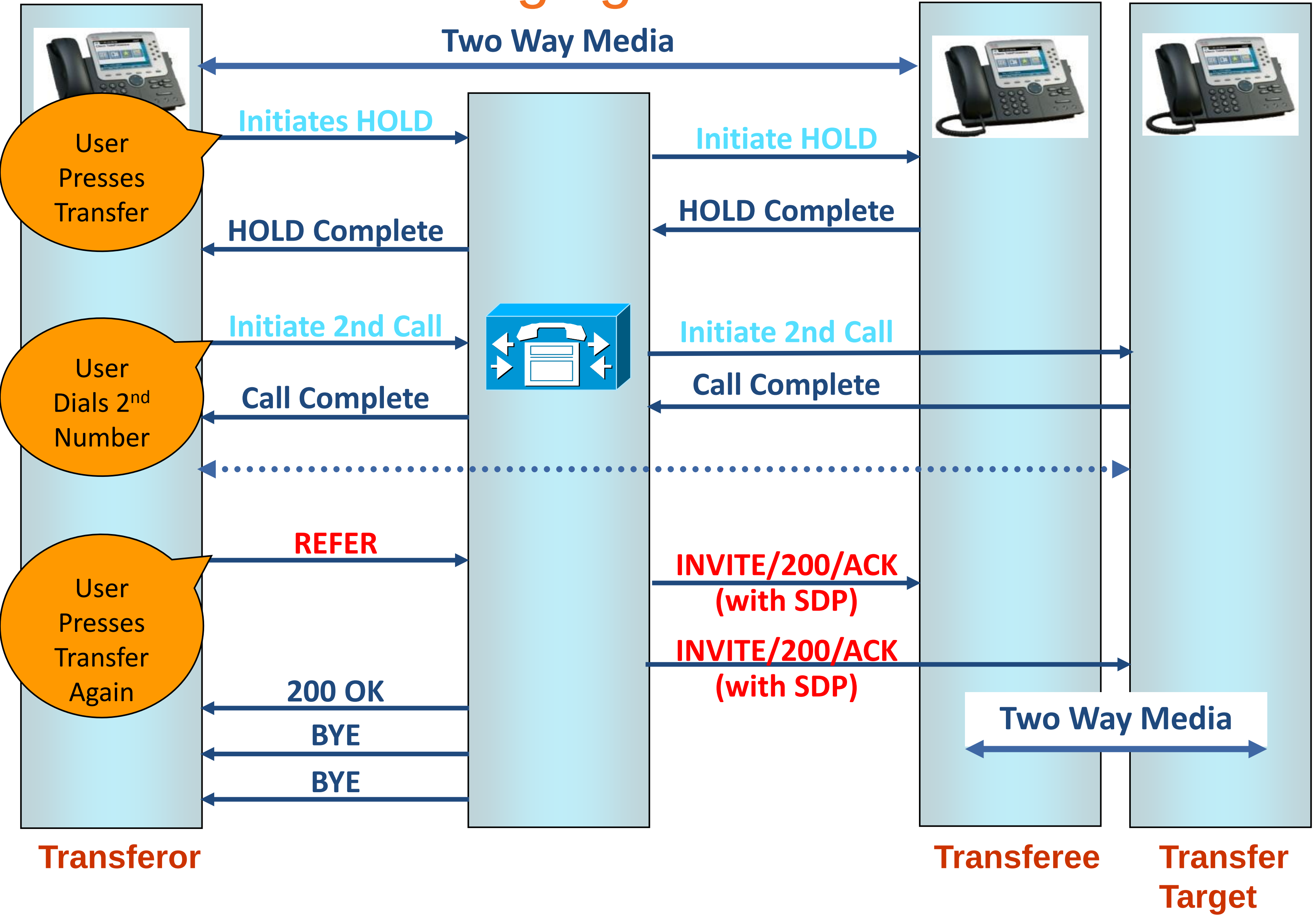
<u>recvonly</u> - Endpoint will only receive media (listen) and not send media

<u>sendonly</u> - Endpoint will only send and not receive media

<u>Inactive</u> - Endpoint will neither send nor receive media

SIP Trunk Signalling and Basic Operation

SIP Messaging – Transfer



To Transfer a call - Unified CM :

- Places the active call on hold and plays MoH
- Connects to Transfer Target
- Updates the connected name/number for each call
- Updates the SDP information for each call leg
- Disconnects the Transferor

Media flows directly between the connected parties

By default Unified CM does NOT forward REFERs – but sends an INVITE instead and maintains call signalling control

With CUCM 8.6 – REFER Transparency LUA script can be enabled on SIP Trunks allowing call control to be relinquished

CUCM SIP Trunking

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– SIP Normalisation and Transparency

Unified CM SIP Trunks

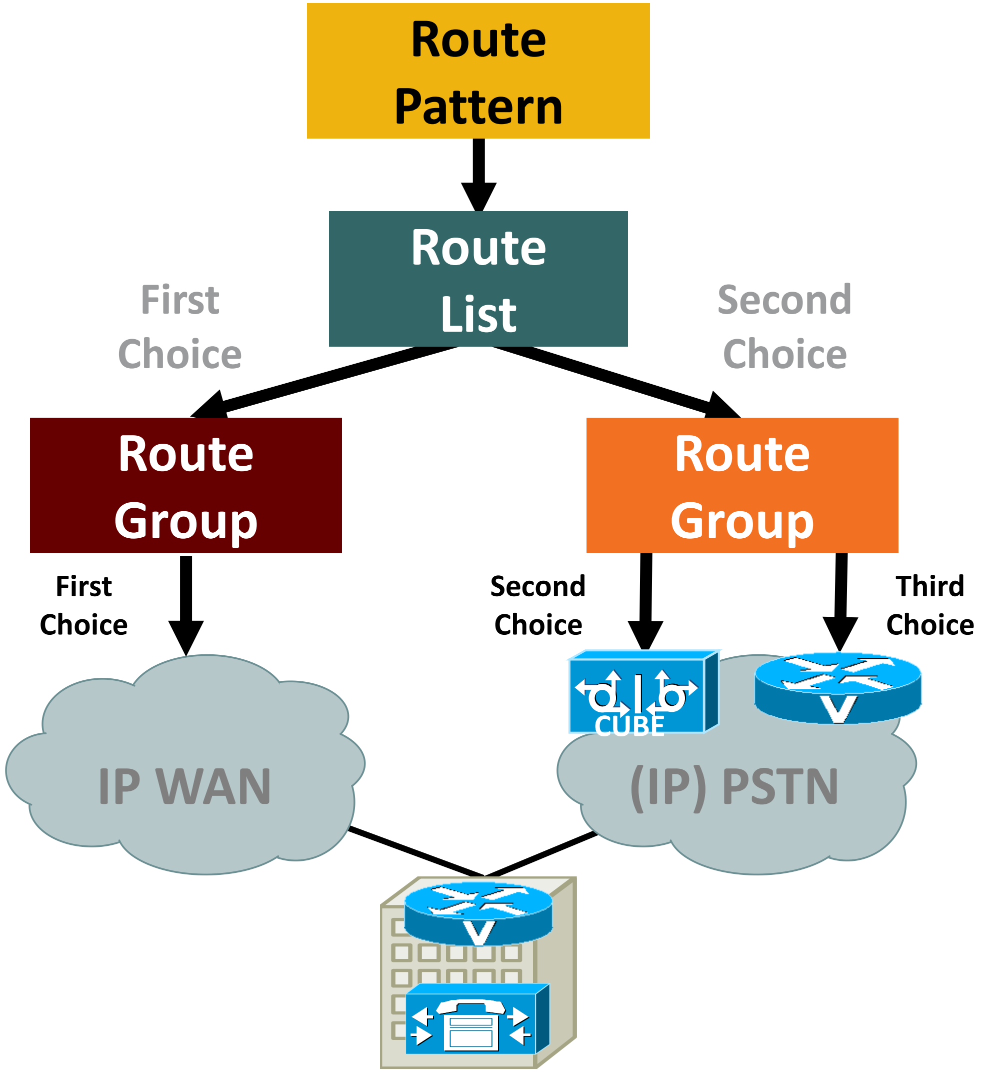
Load Balancing, High Availability & Redundancy – Introduction

UC 8.5 Features

HA Features affecting Calls originating from a CUCM cluster		HA Features for SIP Trunk destinations
Calls over single SIP Trunks	**Calls over multiple SIP Trunks**	
Call Manager Groups	Call Manager Groups	Up to 16 Destination IP Addresses
	Route Lists and Route Groups	DNS SRV
Run on All Nodes	Run on All Nodes	SIP OPTIONS Ping
Route Local	Route Local	

SIP Trunk Load Balancing, Availability & Redundancy

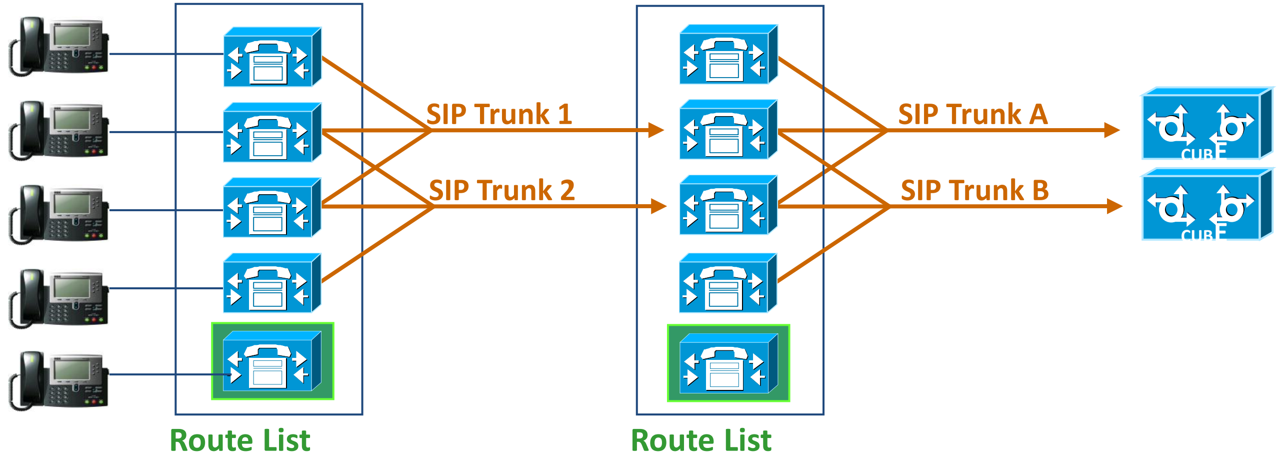
Redundancy and Load Balancing – Multiple Trunk Routes



- Create redundant routes for the same destination
  - –A trunk denotes a unique route
- Trunks are placed in Route Groups
  - –A distribution algorithm in Route Group determines which trunk will be used next
- Route Groups are placed in priority order into a Route List
- A Route Pattern contains the Route List

SIP Trunk Load Balancing, Availability & Redundancy

Call Routing Pre CUCM 8.5 – Source and Destination IP Address Limits



SIP Trunks

Up to 3 source IP addresses (Call Manager Group)
1 destination IP address or a single DNS SRV Record

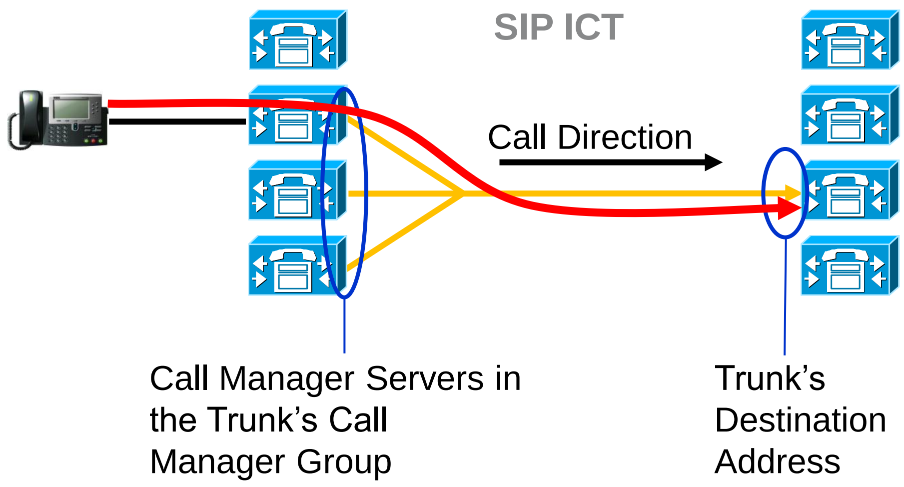
Route Lists

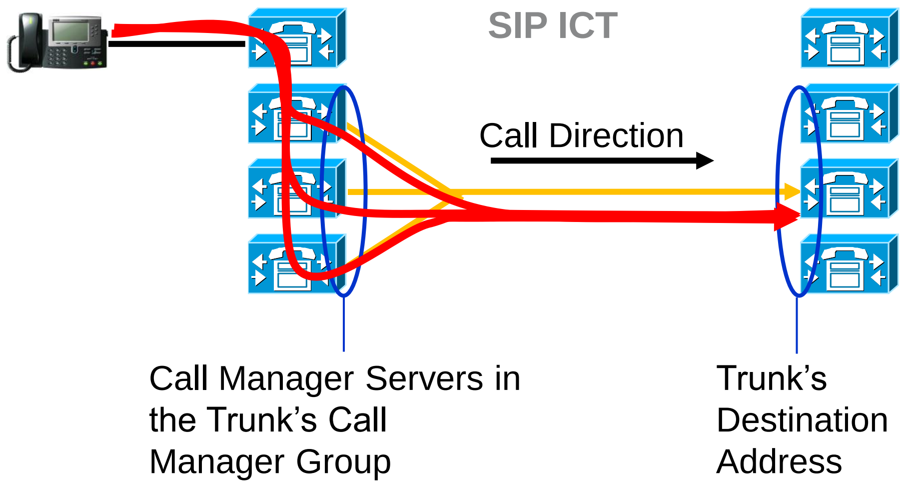
The Route List is active only on the primary node on its Call Manager Group
The Route List's primary node should not be co-resident on a node used by associated outbound Trunks as this can limit the choice of server for outbound Trunk calls

Pre CUCM 8.5 – Trunk Call Routing – “Route Local” (1)

The Route Local feature – Single Outbound Trunk

Route Local has an influence on which server is used to initiate outbound calls





Single ICT – No Route Lists and Route Groups

Phone is registered to same node as the Trunk

Route Local Behaviour

If the calling device is registered to a CUCM server that is also a server in the selected outbound Trunk’s Call Manager Group…..

Then use this server to initiate the outbound Trunk call

Single ICT – No Route Lists and Route Groups

Phone is NOT registered to same node as Trunk

Route Local Behaviour

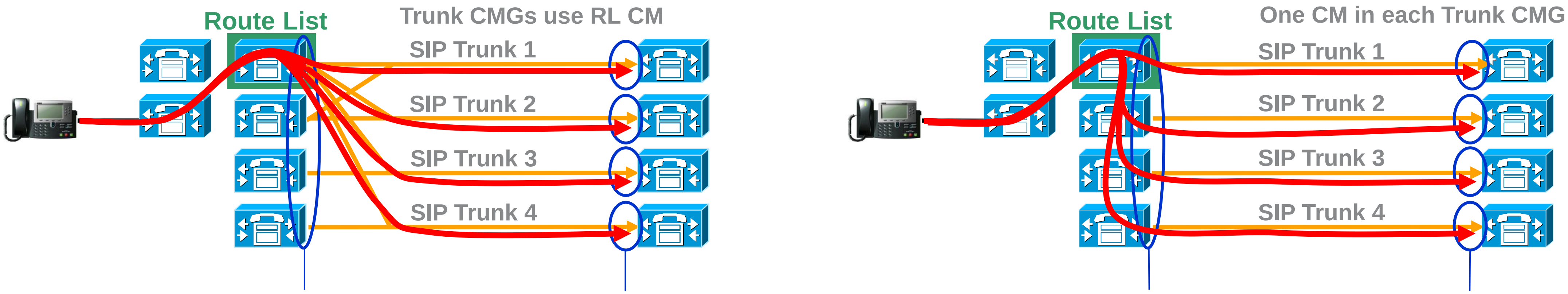
If the calling device is NOT registered to a CUCM server that is a server in the selected outbound Trunk’s Call Manager Group…..

Randomly distribute calls across the servers in the Trunk’s Call Manager Group for outbound Trunk calls

Pre CUCM 8.5 Trunk Call Routing – "Route Local" (2a)

The Route Local feature – Multiple Outbound Trunks with Route List and Route Groups

Route Local has an influence on which server is used to initiate outbound calls



Route Local affects the choice of server used to originate outbound calls

Call distribution to Trunk Destination Addresses controlled by Route Group

Route Local does not affect the choice of server used to originate outbound calls

Call distribution to Trunk Destination Addresses controlled by Route Group

Route Local with Route Lists

When multiple Trunks are used with Route Lists and Route Groups – The Route List is considered to be the "calling device". When a call arrives at the Route List it will select an outbound Trunk from its Route Groups (The Route List is active only on the primary UCM in the RL's Call Manager Group)

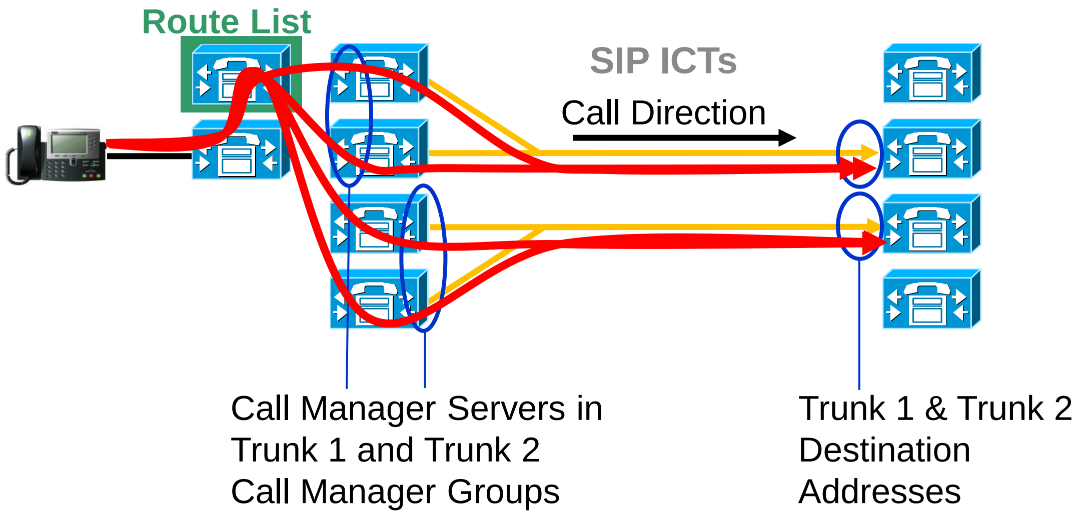
If the calling device (Route List) is registered to a CUCM server that is also a server in the selected outbound Trunk's Call Manager Group - Then use this server to initiate the outbound Trunk call

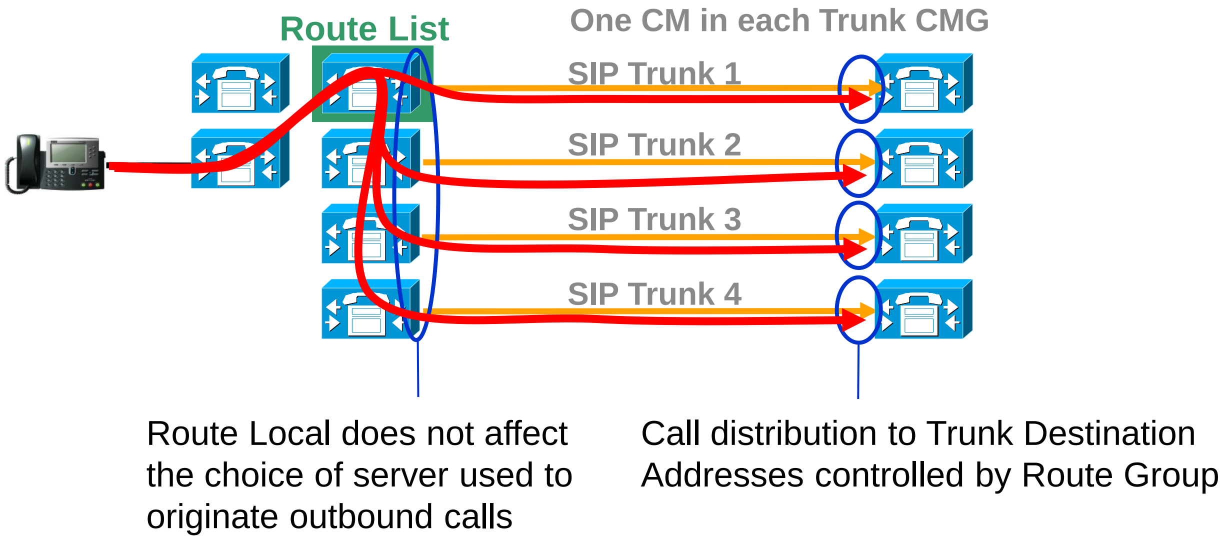
If the calling device (Route List) is not registered to a CUCM server that is a server in the selected outbound Trunk's Call Manager Group - Then randomly distribute calls across the servers in the Trunk's Call Manager Group to initiate outbound Trunk calls

Pre CUCM 8.5 – Trunk Call Routing – "Route Local" (2b)

The Route Local feature – Multiple Outbound Trunks with Route List and Route Groups

Route Local has an influence on which server is used to initiate outbound calls





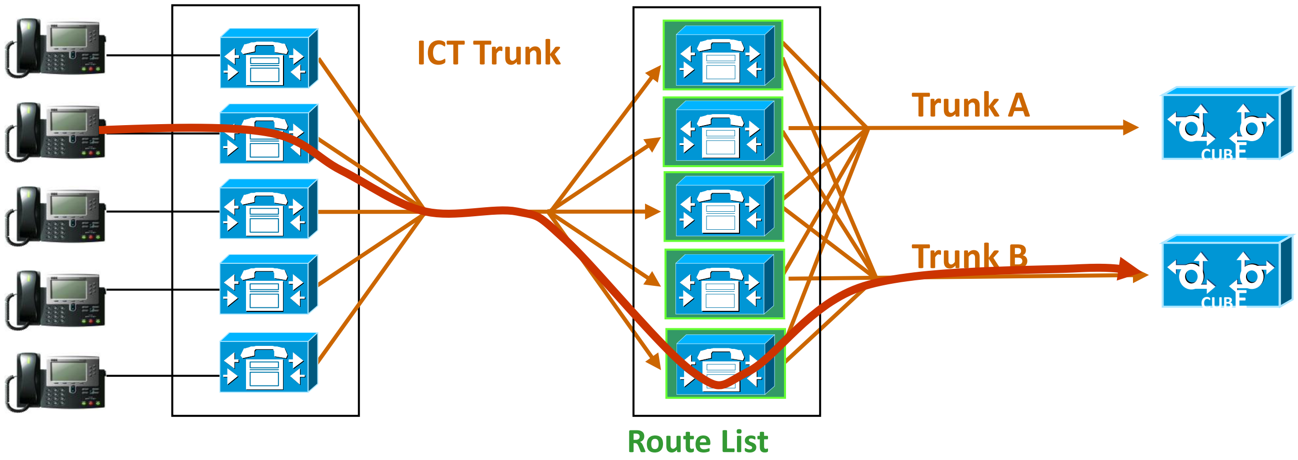
Route Local with Route Lists – Recommendation do not co-locate RL and Trunks

When multiple Trunks are used with Route Lists and Route Groups – The Route List is considered to be the "calling device". When a call arrives at the Route List it will select an outbound Trunk from its Route Groups

If the calling device (Route List) is not registered to a CUCM server that is a server in the selected outbound Trunk's Call Manager Group – Then randomly distribute calls across the servers in the Trunk's Call Manager Group to initiate outbound Trunk calls

SIP Trunk Load Balancing, Availability & Redundancy

CUCM 8.5+ Call Routing – Source & Destination IP Address Enhancements



SIP Trunks

Up to 16 source addresses – by enabling “Run on all Active Unified CM Nodes”
Up to 3 source addresses using standard CM Groups
Up to 16 configured destination IP addresses or a single DNS SRV Record
Random distribution of calls over configured destination IP addresses

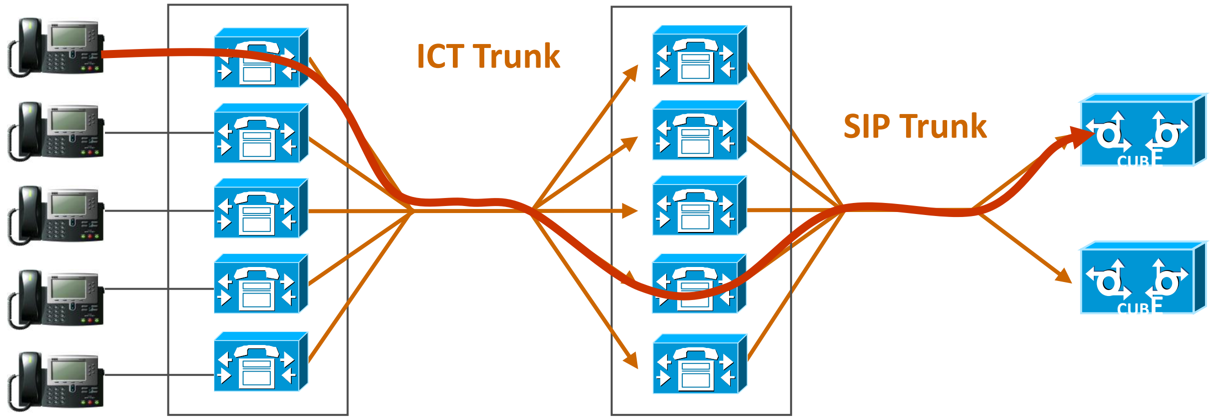
Route Lists

Enable “Run on all Active Unified Nodes” to activate Route List on all Unified CM nodes

“Run on All Nodes” in conjunction with the “Route Local” rule reduces intra-cluster traffic
In effect, the node that the inbound call arrives on is the node that is used to initiate the outbound call

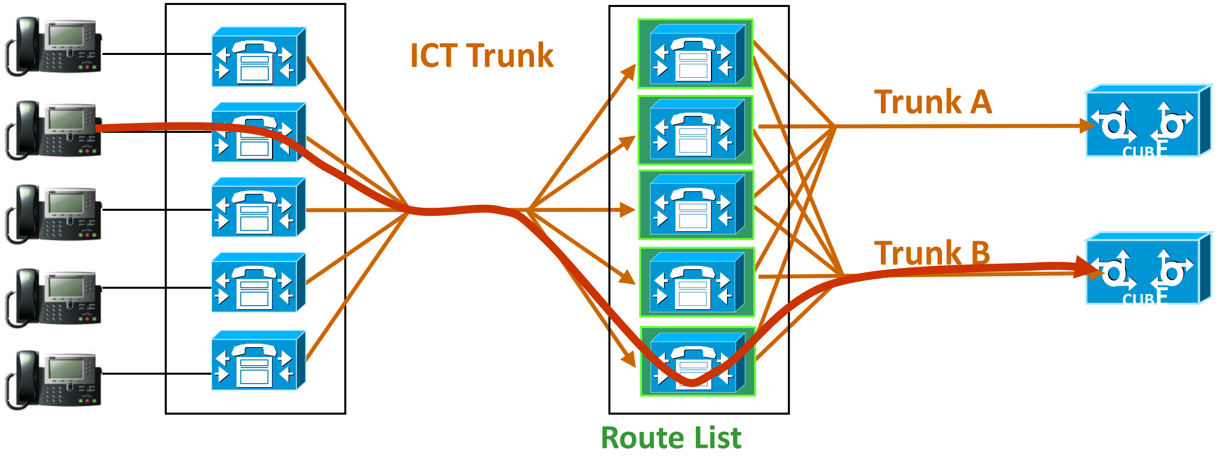
SIP Trunk Load Balancing, Availability & Redundancy

The Route Local Rule and “Run on all nodes” Option for Trunks and RLs



For single Trunks

The Route Local rule operates in conjunction with the “Run on all Nodes” feature enabled on the Trunk - Such that outbound Trunk calls originate from the same node that the inbound call arrives on



Multiple Trunks using Route Lists

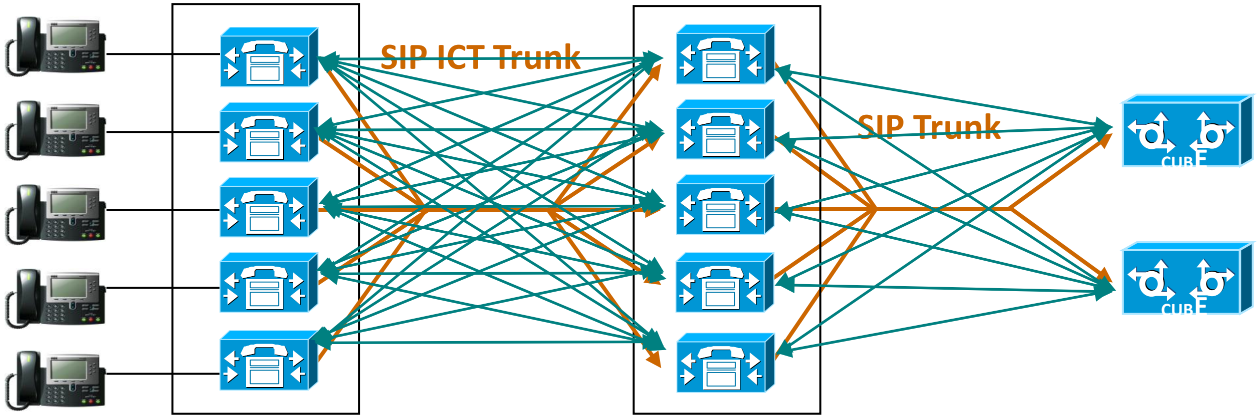
The Route Local rule operates in conjunction with the “Run on all Unified CM Nodes” feature enabled on the Route List and Trunks - Such that outbound Trunk calls originate from the same node that the inbound call arrives on

Benefit – Calls more evenly distributed across all nodes within a cluster

Cisco live!

SIP Trunk Load Balancing, Availability & Redundancy

CUCM 8.5 SIP OPTIONS Ping – Improved Failover…….



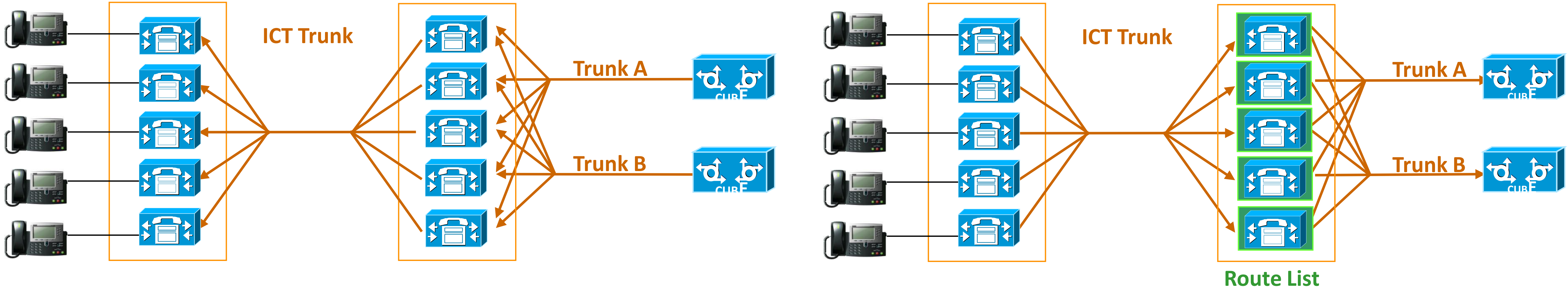
OPTIONS Ping is activated on a per SIP Trunk basis
Each node running the SIP Trunk daemon in the originating cluster uses OPTIONS Ping to determine the availability of each defined destination IP address

- CUCM will not attempt to establish a call to an unavailable remote peer
- SIP Trunk - "In Service" whilst one remote peer is reachable
- SIP Trunk - "Out Of Service" state when all remote peers are unreachable

- CUCM 8.5 – Dynamic reach-ability detection
- Pre CUCM 8.5 - Per call time out

SIP Trunk Load Balancing, Availability & Redundancy

CUCM 8.5 – Benefits of New Call Routing Features



<u>Run on All Active Unified CM Nodes, Up to 16 Destination Addresses</u>
- Fewer Trunks, fewer Route Lists and Route Groups required
- Calls evenly distributed across all destination addresses
- Outbound calls originate from the node that the call arrived on

<u>OPTIONS Ping</u>
- Dynamic reachability detection
- Calls only sent to those SIP Trunk destinations known to be alive

SIP Trunk Load Balancing, Availability & Redundancy

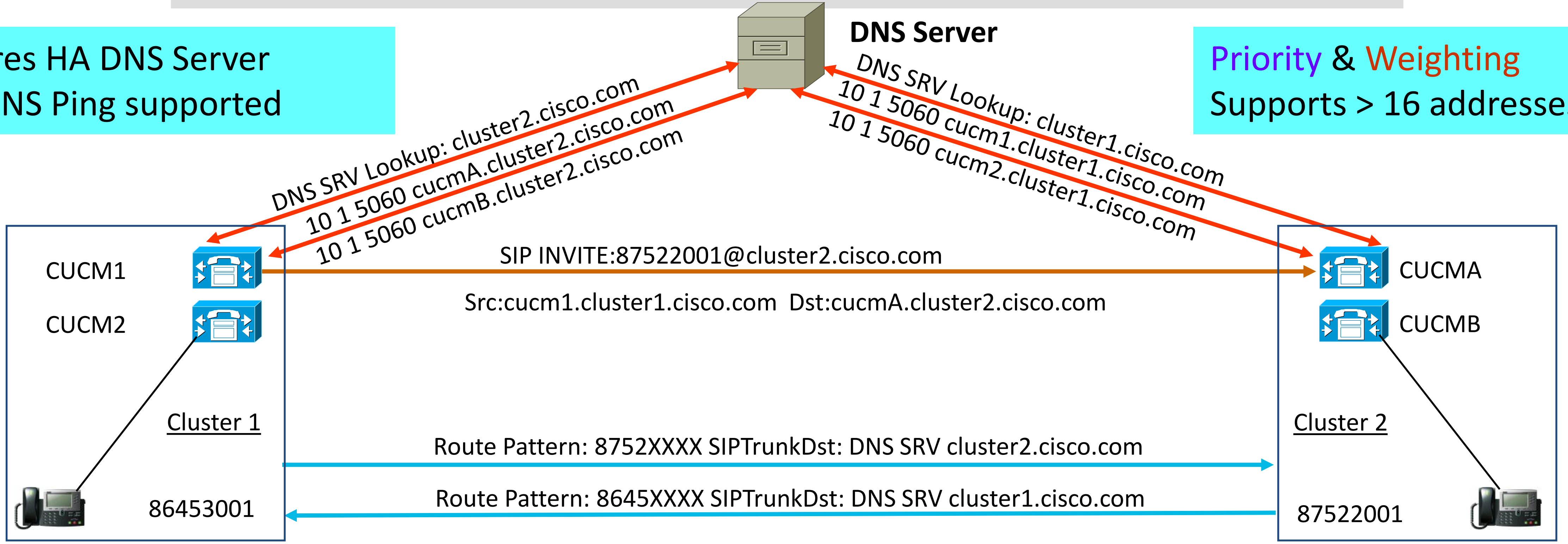
Redundancy and Load Balancing – Using DNS

```
_sip._tcp.cluster1.cisco.com     IN  SRV  10  1  5060  cucm1.cluster1.cisco.com
                                 IN  SRV  10  1  5060  cucm2.cluster1.cisco.com
_sip._tcp.cluster2.cisco.com     IN  SRV  10  1  5060  cucmA.cluster2.cisco.com
                                 IN  SRV  10  1  5060  cucmB.cluster2.cisco.com

cucm1.cluster1.cisco.com         IN    A     50.50.50.5
cucm2.cluster1.cisco.com         IN    A     50.50.50.6
cucmA.cluster2.cisco.com         IN    A     10.10.10.1
cucmB.cluster2.cisco.com         IN    A     10.10.10.2
```

Requires HA DNS Server
OPTIONS Ping supported

Priority & Weighting
Supports > 16 addresses



Calls 87522001

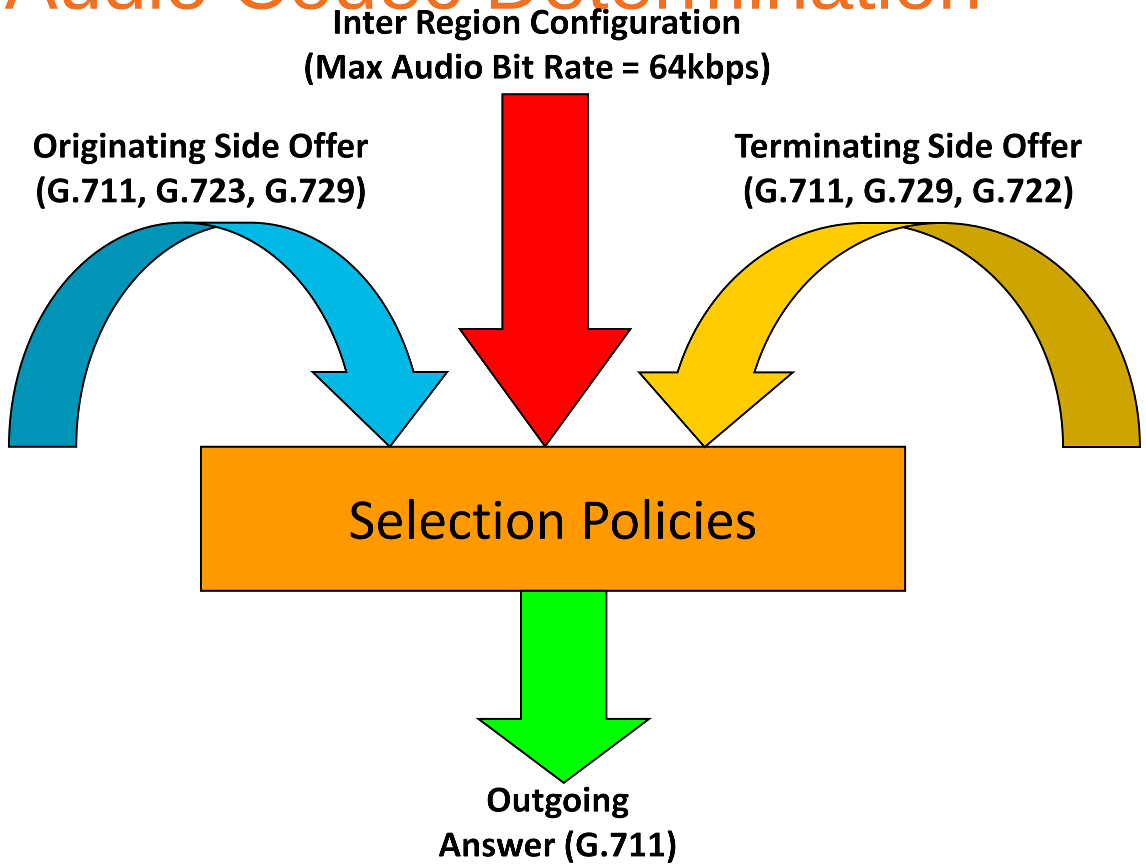
DNS A Look Up not shown

CUCM 8.5 SIP Trunking

- –Overview of CUCM SIP Trunk Features
- –SIP Trunk Signalling and Basic Operation
- –SIP Trunk – Load Balancing, Availability & Redundancy
- –SIP Trunk Codec Support – Audio Codec Preference Lists
- –SIP Trunk Security
- –QSIG Over SIP
- –SIP Normalisation and Transparency

Unified CM SIP Trunks – Pre UC 9.0

Audio Codec Determination



- Codec selection determined by an intersection of codecs in originating and destination offers and inter-region configuration
- For a given per call Max Audio Bit Rate – The best quality codec in the intersection is selected for the call
- e.g. For 64 kbps max audio bit rate G.711 preferred over G.729

Region Relationships

Region	Max Audio Bit Rate	Max Video Call Bit Rate (Includes Audio)	Link Loss Type
Applications	64 kbps (G.722, G.711)	384	Lossy
Audio-32k Video-256k	32 kbps (iSAC, G.722.1)	256	Lossy
Audio-64k Video-2M	64 kbps (G.722, G.711)	384	Lossy
Audio-64k Video-384k	64 kbps (G.722, G.711)	384	Lossy
Audio-8k	8 kbps (G.729)	None	Lossy
Audio-8k Video-128k	8 kbps (G.729)	128	Lossy
CORE H323	64 kbps (G.722, G.711)	384	Lossy
CORE H323 G729	64 kbps (G.722, G.711)	384	Lossy
CORE Voicemail	64 kbps (G.722, G.711)	384	Lossy
CORE XCODE	64 kbps (G.722, G.711)	384	Lossy
Default	64 kbps (G.722, G.711)	384	Lossy
NOTE: Regions(s) not displayed	Use System Default	Use System Default	Use System Default

Unified CM SIP Trunks – UC 9.0 +

Audio Codec Determination

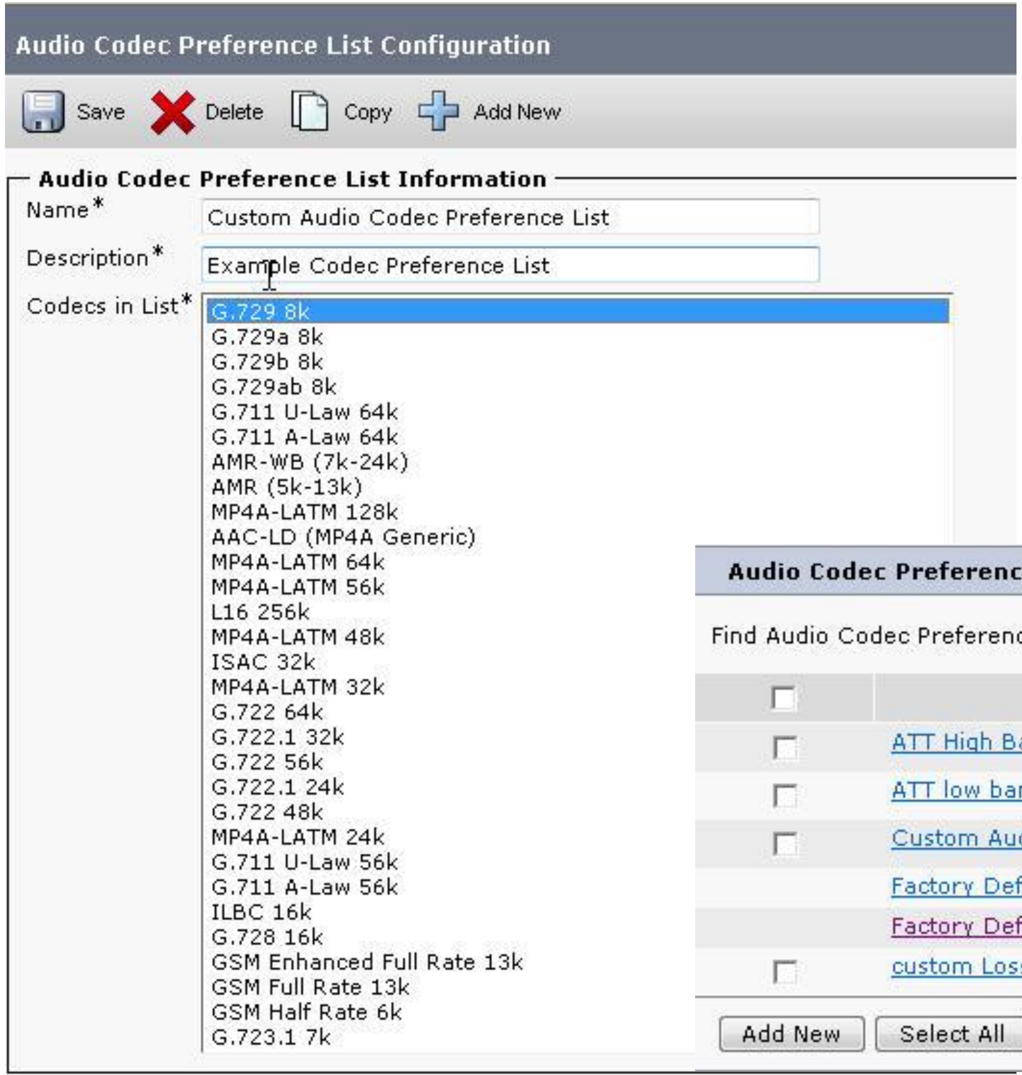
- In UC 9.0 customisable Codec Preference Lists are added to CUCM regions
- In addition to the default "Lossy" and Low "Loss" audio codec preference lists (previously known as the "Link Loss Type"), multiple custom Audio Codec Preference Lists can now be configured.
- Audio Codec Preference lists can be used for codec selection for calls within a region and between regions.
- The "Maximum Audio Bit rate" is still applied for calls within a region and between regions, but rather that using the highest audio quality codec (as per earlier Unified CM releases); the codec selection is now made based on the codec order in the Audio Codec Preference List and the codecs that the endpoints support.

Unified CM SIP Trunks – UC 9.0+

New Features

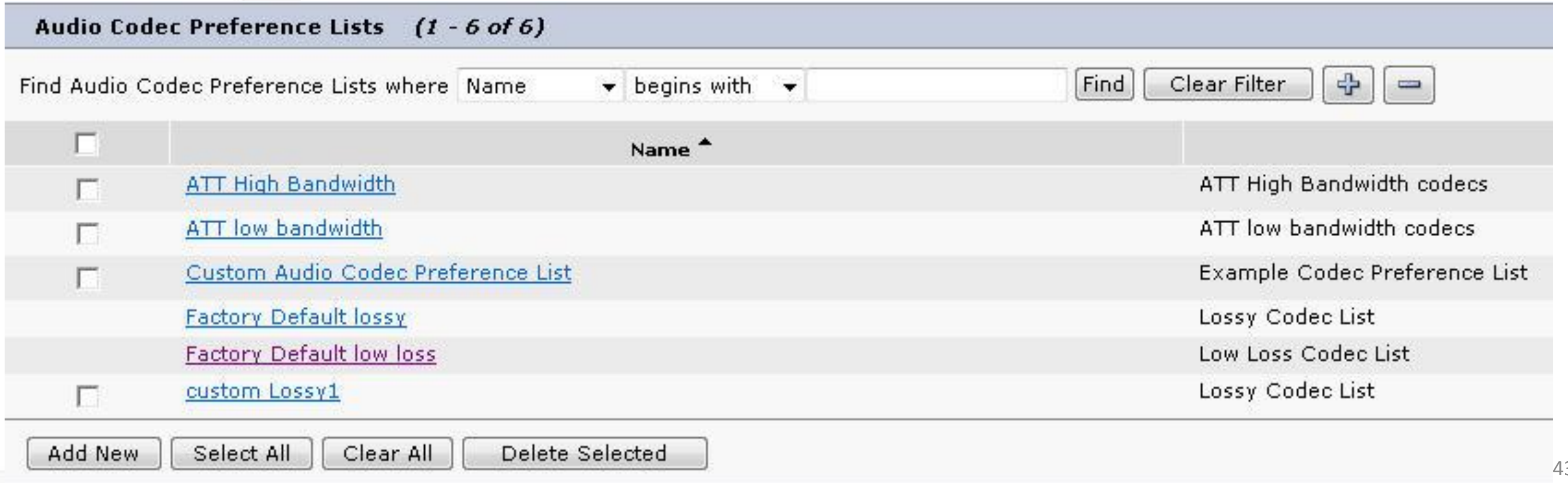
Audio Codec Preference Lists

Cisco Unified CM Administration>System>Region Information>Audio Codec Preference List



An Audio Codec Preference List is a list of all the codec types supported by Unified CM.

The preference order of this list of codecs can be modified and saved as custom preference list (Note that codecs cannot be removed from the Audio Codec Preference List).

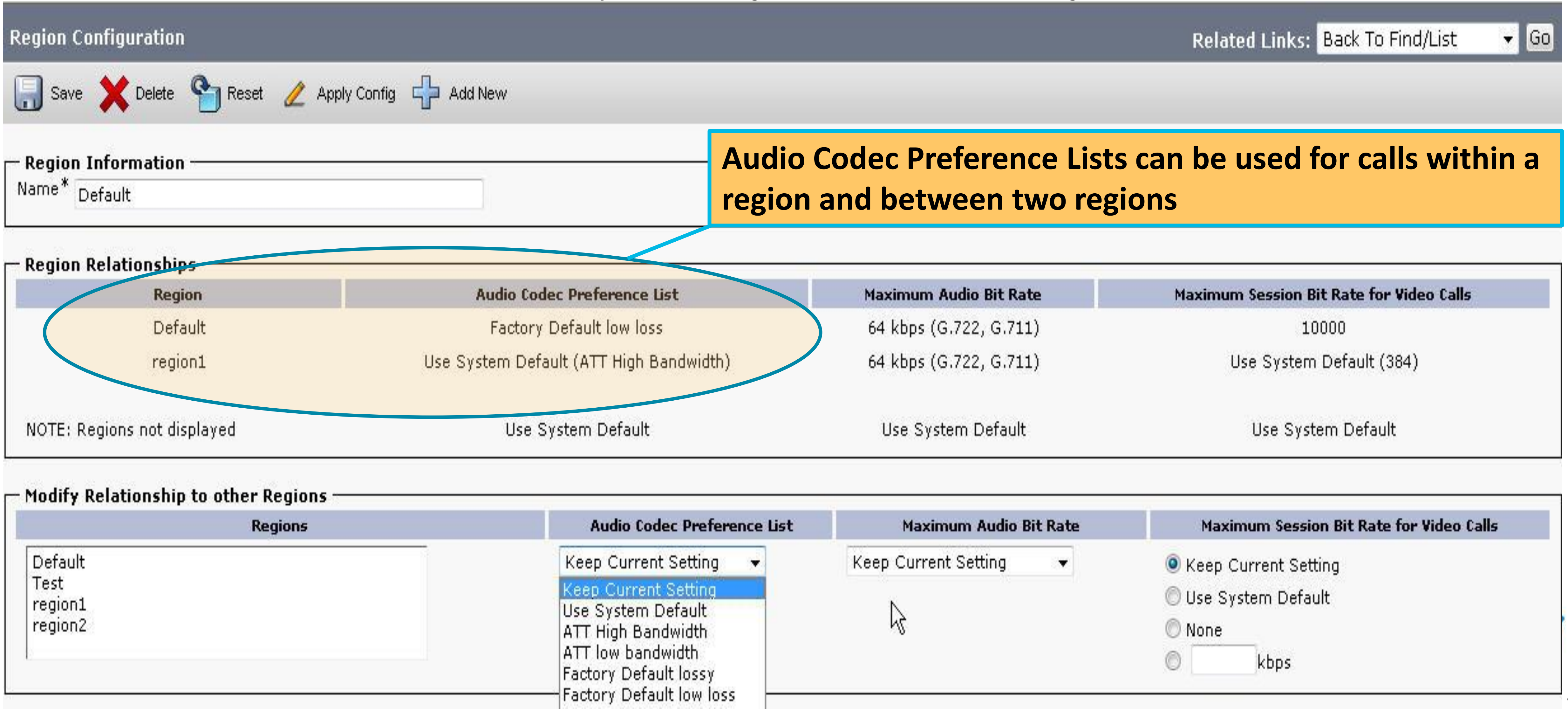


Unified CM SIP Trunks – UC 9.0+

New Features

Regions with Audio Codec Preference Lists

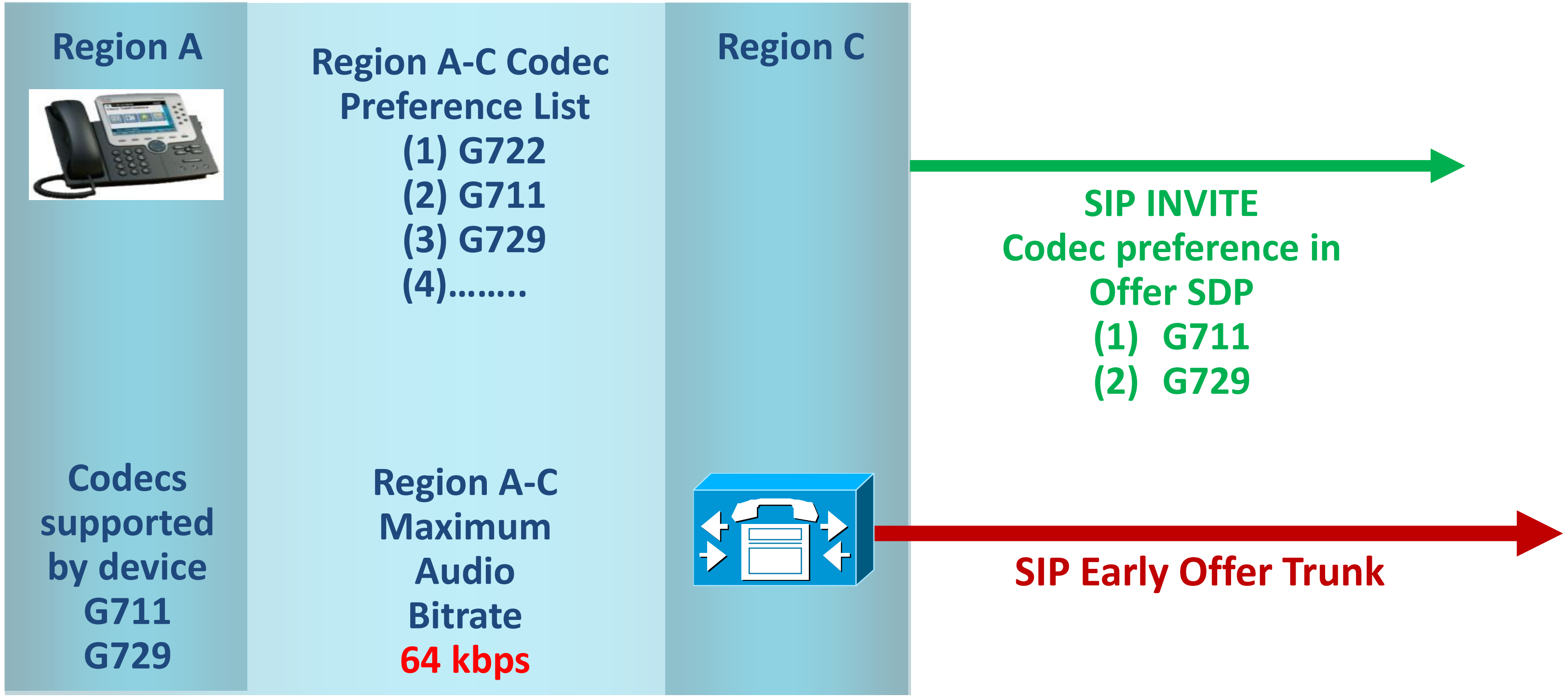
Cisco Unified CM Administration>System>Region Information>Region



Unified CM SIP Trunks – UC 9.0 +

Audio Codec Determination

The list of codecs used for codec negotiation during call set up is the subset of codecs supported by the device and those in the codec preference list, limited by the maximum audio bit rate for the region/ region pair.



Unified CM SIP Trunks – UC 9.0 +

Audio Codec Determination

The list of codecs used for codec negotiation during call set up is the subset of codecs supported by the device and those in the codec preference list, limited by the maximum audio bit rate for the region/ region pair.

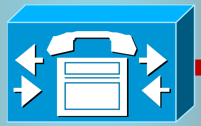
Region A

Codecs supported by device
G711
G729

Region A-C Codec Preference List
(1) G722
(2) G711
(3) G729
(4)……..

Region A-C Maximum Audio Bitrate
40 kbps

Region C

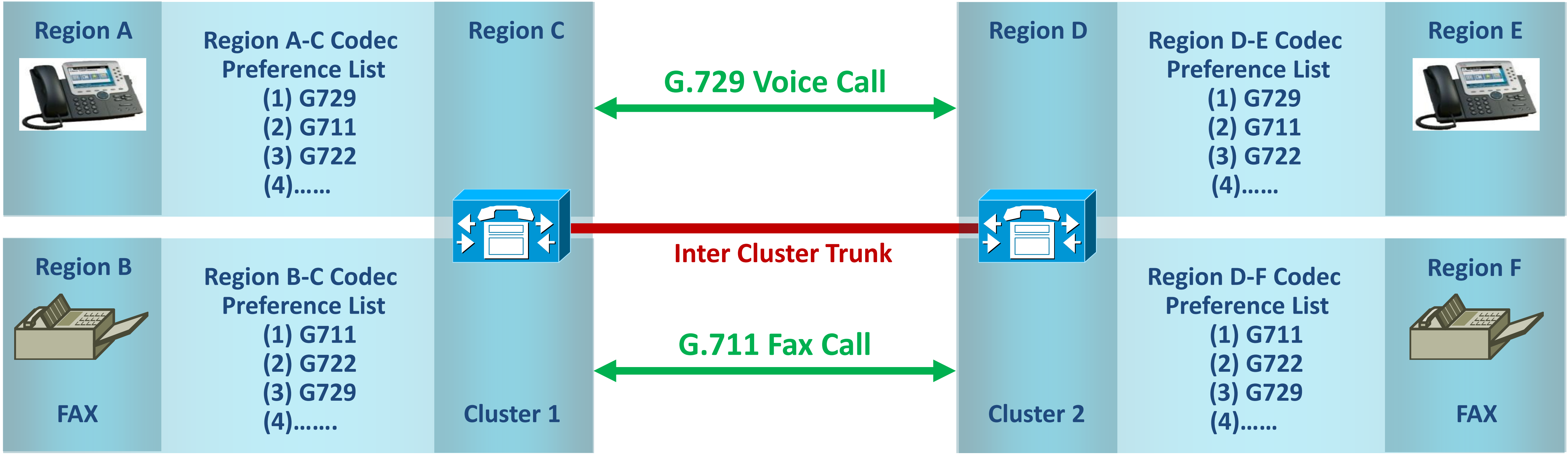


SIP INVITE
Codec preference in Offer SDP
(1) G729

SIP Early Offer Trunk

Unified CM SIP Trunks – UC 9.0 +

Audio Codec Preference Lists – Deployment Recommendations



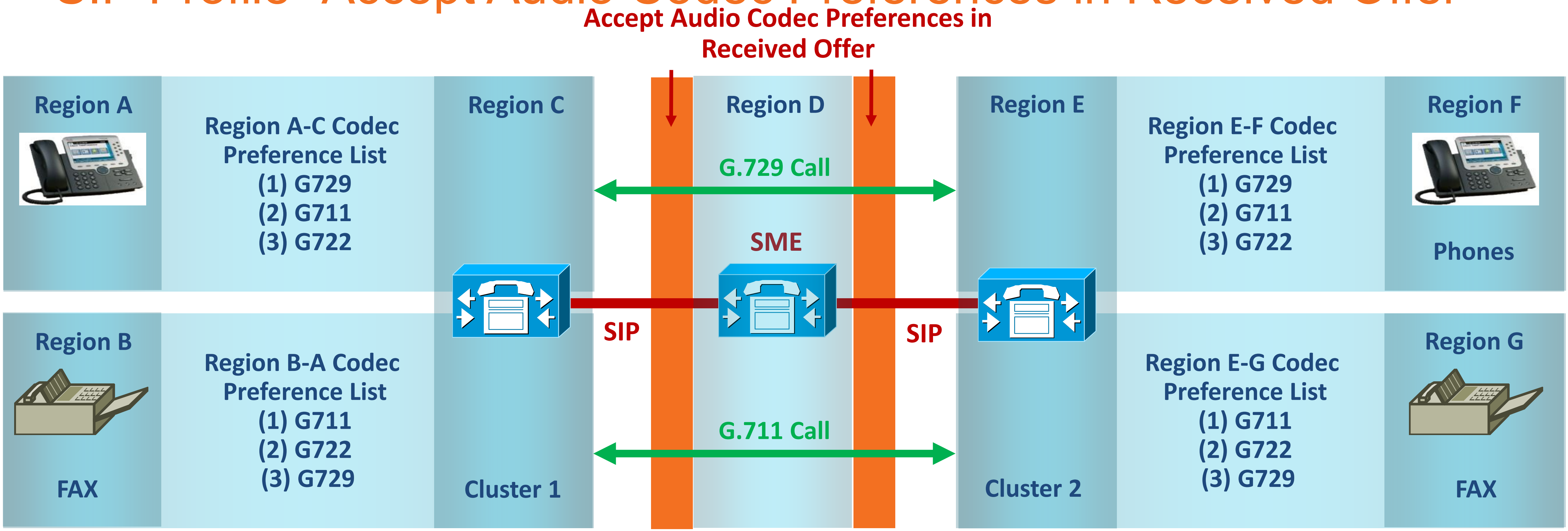
For calls between two Unified CM clusters via SIP or H323 inter cluster Trunks, Audio Codec Preference Lists allow the codec used for a call to be selected based upon the calling and called devices codec preference.

Note – Equivalent Audio Codec Preference Lists for each device type region should be configured in each cluster to ensure that a common codec is selected for each device types, irrespective of call direction or Trunk configuration.

Unified CM SIP Trunks – UC 9.0 +

9.0 Features

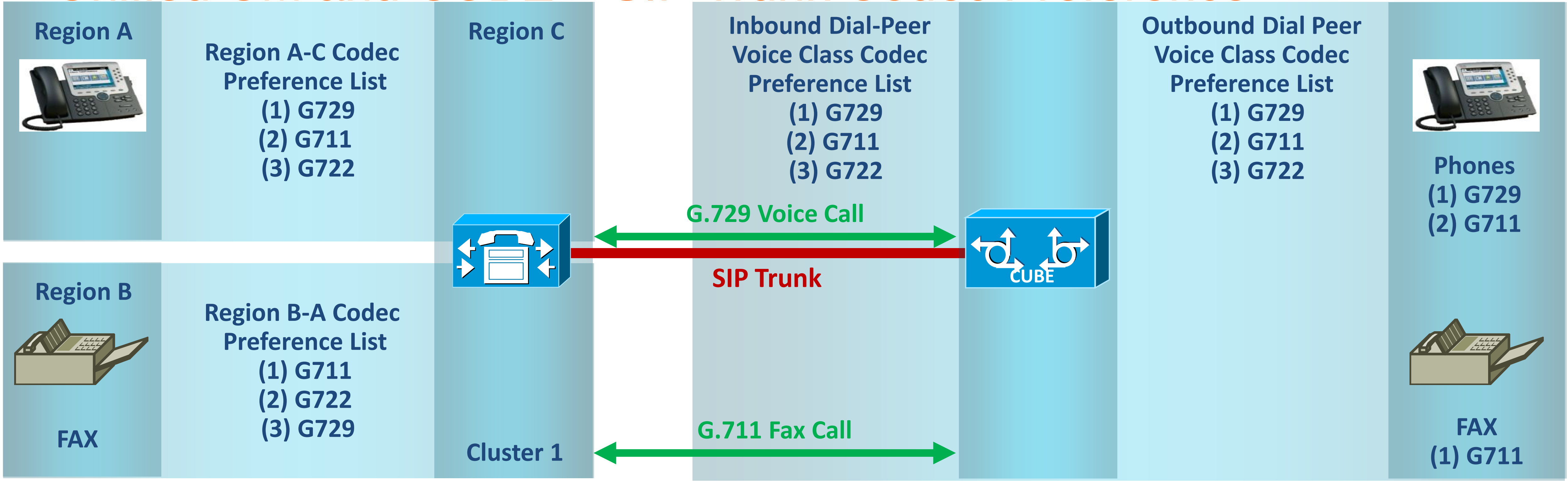
SIP Profile "Accept Audio Codec Preferences in Received Offer"



In deployments where calls can pass through more than one Unified CM cluster, for example SME deployments; the inter-region Audio Codec Preference List of the intermediary Unified CM cluster can over-ride the preferred codec selection between the calling and called device. To ensure that the endpoints' codec preferences are honored as calls pass through SME, enable the SIP Profile feature "Accept Audio Codec Preferences in Received Offer" on all SME SIP Trunks.

Unified CM SIP Trunks – UC 9.0 +

Unified CM and CUBE – SIP Trunk Codec Preference



Instead of using dedicated SIP Trunks to CUBE for Voice and Fax calls, a single SIP Trunk and a single inbound & outbound Dial-Peer can be configured on CUBE for all device types. It's recommended to use the same voice class codec preference list for both the inbound & outbound dial-peer containing the list of codecs that you want to negotiate with the service-provider.

The order of the codecs will be dictated first, by the order received in the inbound offer and then by the order defined under voice class codec.

CUCM 8.5 SIP Trunking

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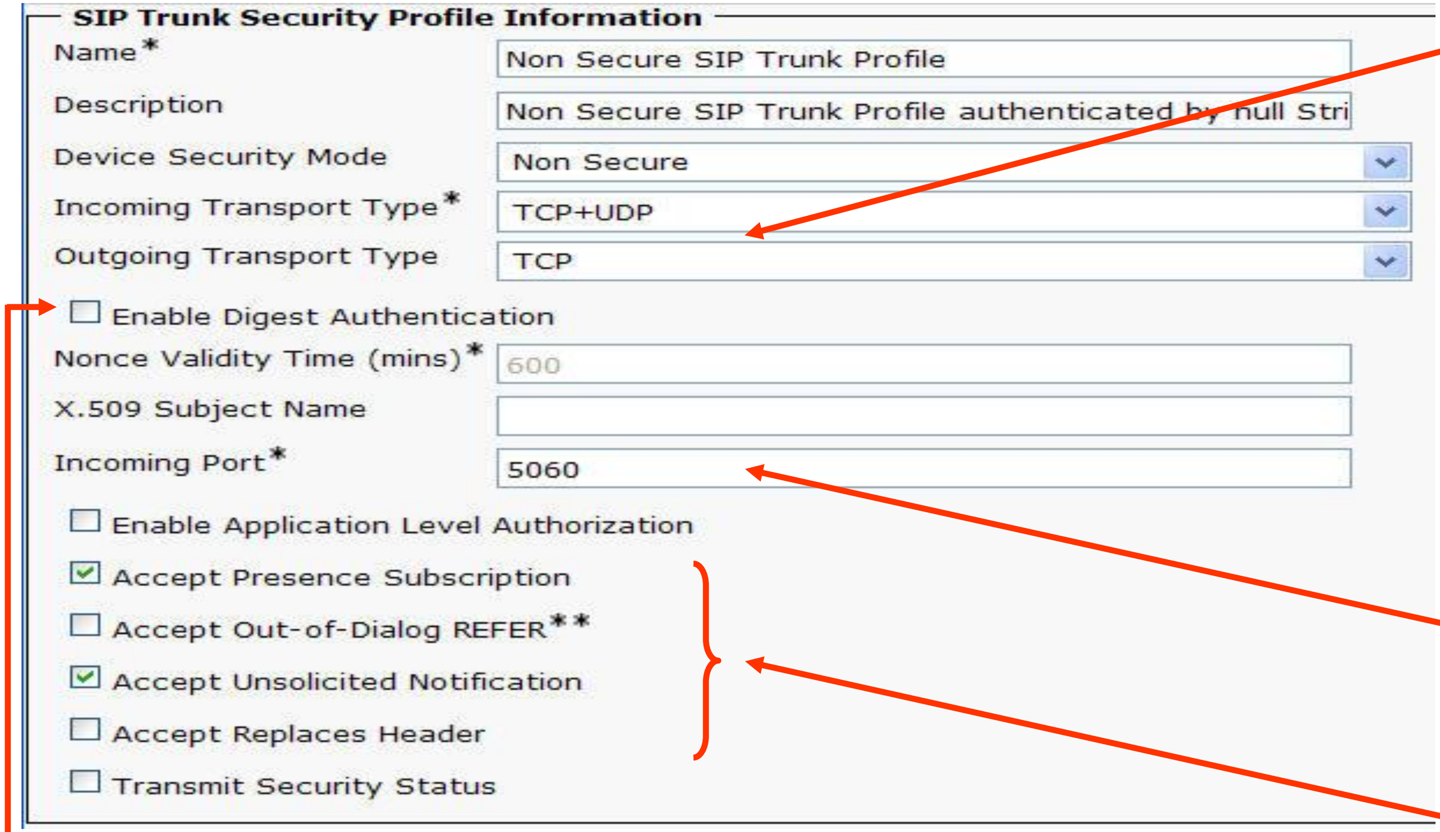
- –SIP Normalisation and Transparency

Unified CM SIP Trunks

SIP Trunk Dynamics – Trunk Security Profile Information

Cisco Unified CM Administration>System>Security>


SIP Trunk Security Profile Information
Name* Non Secure SIP Trunk Profile
Description Non Secure SIP Trunk Profile authenticated by null Stri
Device Security Mode Non Secure
Incoming Transport Type* TCP+UDP
Outgoing Transport Type TCP
Enable Digest Authentication
Nonce Validity Time (mins)* 600
X.509 Subject Name
Incoming Port* 5060
Enable Application Level Authorization
Accept Presence Subscription
Accept Out-of-Dialog REFER**
Accept Unsolicited Notification
Accept Replaces Header
Transmit Security Status

Incoming and Outgoing transport protocol types :

- **UDP** – Connectionless – Uses SIP Session Timers to determine session state
- **TCP** – Connection Oriented – TCP maintains connection state
- **TLS** – Connection Oriented Encrypted Signalling – Uses X.509 Certificates to authenticate connections – should be used if encrypted media (SRTP) is required

Incoming port

- The Transport protocol port number CUCM uses to listen on for inbound SIP connections

SIP Messages that the trunk will accept

Digest Authentication
Less secure than TLS (does not provide integrity or confidentiality) Uses a challenge – response mechanism based on a shared username and digest credentials to authenticate connections. Challenger sends Realm (ClusterID) and a nonce value (a random number) to the calling device. Calling device uses the nonce value to create and return an MD5 hash of the Realm's username and digest credentials

Unified CM SIP Trunks

Technology Basics – Security - TLS

X.509 Certificates can be :

- Imported to each server using Platform Administration
  - Certificates may be obtained directly from peer entities (self signed certificates)
- Signed by an external Certificate Authority
  - More scalable, Easier to manage

- The Certificate authenticates the remote trunk to a server / cluster
- The X.509 Subject Name field in the SIP Trunk Security profile – matches the name on a certificate and authorises its use with the Trunk
- A TLS Session – May be shared for multiple calls
- Secure Interworking with UCME, CUBE, and Gateways
- Media Encryption (SRTP) can be set up without secure signalling but in this case security keys are sent in the clear

CUCM SIP Trunking

–Overview of CUCM SIP Trunk Features

–SIP Trunk Signalling and Basic Operation

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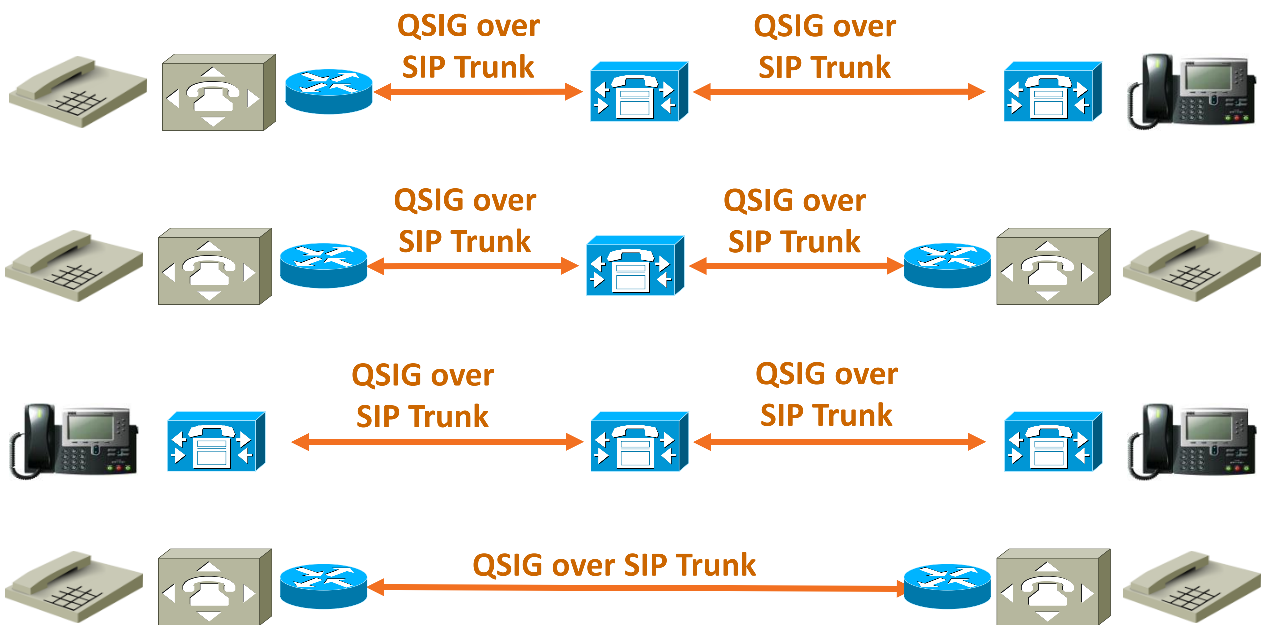
–SIP Trunk Codec Support – Audio Codec Preference Lists

–SIP Trunk Security

–QSIG Over SIP

–SIP Normalisation and Transparency

CUCM 8.5 SIP Trunking – QSIG Over SIP Trunks



<u>Unique QSIG Features</u>

- Call Back on No Answer/ No Reply
- Path Replacement

<u>Pre CUCM 8.5</u>

QSIG over H323 Annex M1

- Used over Inter Cluster Trunks

QSIG over MGCP from IOS gateways

- Used to interconnect QSIG TDM PBX and CUCM clusters

<u>With CUCM 8.5</u> – When the “QSIG over SIP Trunks” feature is considered in conjunction with the other features that SIP Trunks offer – e.g. SIP Options Ping, Normalisation Scripts, SIP Preconditions for RSVP Call Admission Control, simpler configuration for Signalling Authentication and Encryption, a greater range of codecs supported in comparison with other Trunk types etc……

SIP is likely to become the Trunk protocol of choice for the majority of UC customers

Unified CM 8.5 - Inter Cluster Trunks

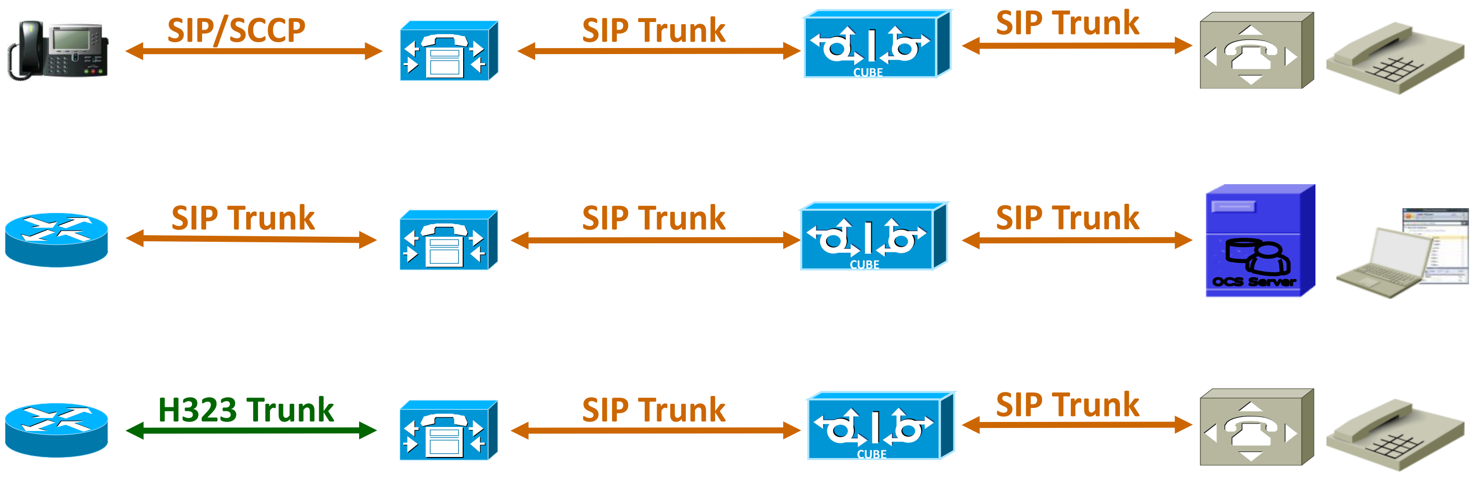
SIP Trunks vs H.323 Trunks – Feature Comparison

	H.323 (Q.SIG)	SIP (QSIG)
Support for "+" character		
Signalling Authentication and Encryption		TLS
Media Encryption		
"Run On All Nodes" feature		
"Up to 16 destination addresses" feature		
Calling Line ID / Name – Presentation / Restriction		
Connected Line ID / Name – Presentation / Restriction		
OPTIONS Ping		
iLBC, AAC, ISAC and G.Clear Support		
G.711, G.722, G.723, G.729 Support		
SIP Subscribe / Notify, Publish – Presence		
Accept Audio Codec Preferences in Received Offer		
QSIG Call Completion – No Reply / Busy Subscriber		
Topology Aware - RSVP Based Call Admission Control – SIP Pre-Conditions		
IPv6, Dual Stack, ANAT		
BFCP – Video Desktop Sharing		

Legend: Yes | Limited support | No

CUCM 8.5 SIP Trunks – Normalisation & Transparency

Pre CUCM 8.5 - SIP Trunk Interoperability with 3rd Party UC Systems



CUCM has some configurable SIP parameters that provide basic SIP interop e.g.

Allow Presentation Sharing using BFCP
Accept Presence Subscription
Accept Unsolicited Notification

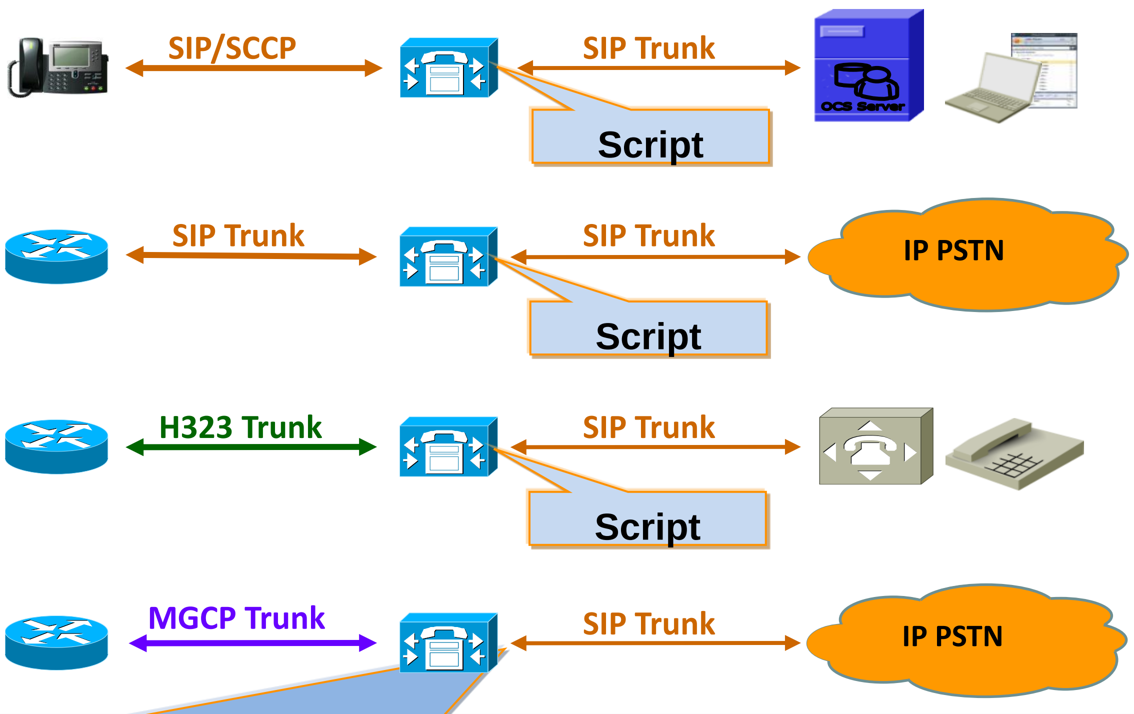
Outgoing T.38 INVITE include audio mline
Accept Out-of-Dialog REFER
Accept Replaces Header

For more complex interoperability issues where SIP messages need to be manipulated – An SBC is typically used on the SIP Trunk connection to a 3rd Party UC system

CUCM 8.5 SIP Normalisation removes the need for an SBC for this purpose……

CUCM 8.5 SIP Trunks – Normalisation & Transparency

SIP Trunk – Normalisation



Normalization Script

Remove "display name" from SIP Header

```
function fixInboundPai()
   local pai = Sip.getHeader("P-Asserted-Identity")
   local displayname = getDisplayName(pai)
   local uri = getUri(pai)
   local number = getUserPart(uri)
   if displayname == number
   then
      Sip.modifyHeader("P-Asserted-Identity", uri)
```

Normalisation allows incoming and outgoing SIP messages to be modified on their way through a CUCM SIP Trunk.

The Normalisation feature is designed to improve interoperability between CUCM SIP Trunks and SIP based 3rd Party SIP PBXs, Applications & IP PSTN services.

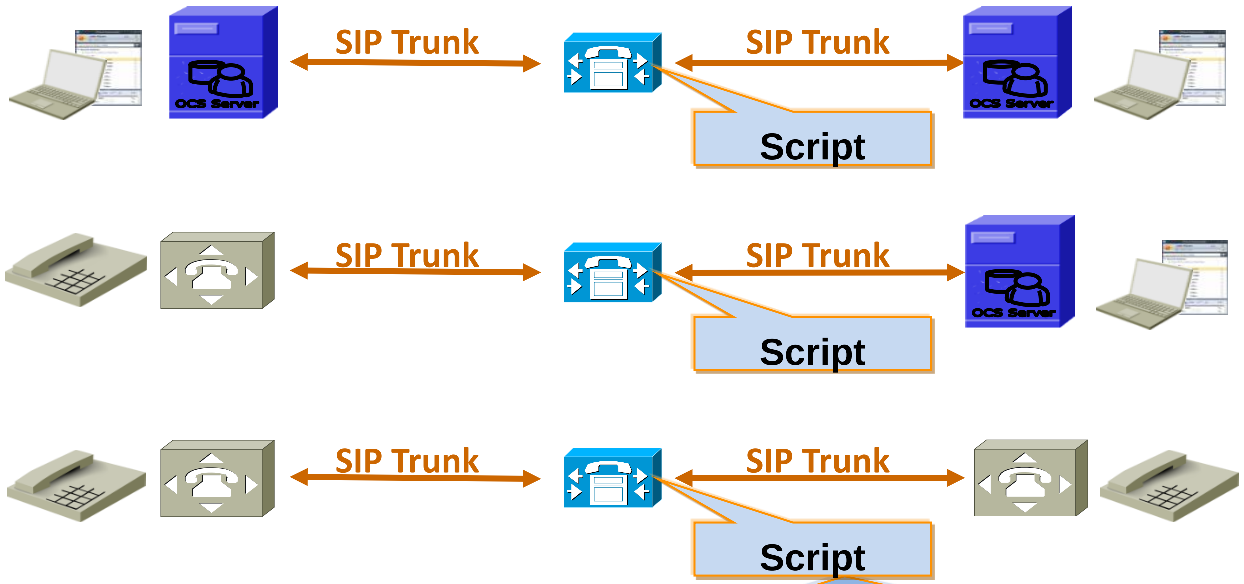
Normalisation is independent of what the SIP Trunk connects to on the other side of CUCM. e.g.

- SIP ⟷ SIP Trunk calls
- Skinny ⟷ SIP Trunk calls
- H.323 ⟷ SIP Trunk calls
- MGCP ⟷ SIP Trunk calls

Normalisation uses a scripting environment to allow customers to modify SIP messages and SDP content on a per trunk basis.

CUCM 8.5 SIP Trunks – Normalisation & Transparency

SIP Trunk – Transparency



Transparency Script

```
Sip.allowHeader("A-Callid")
Sip.allowHeader("A-ConversationId")
function A.inbound_INVITE
   Sip.passThroughHeader("A-Callid")
   Sip.passThroughHeader("A-ConversationId")
   Sip.passThroughHeaderValue("Supported",  "x-nortel")
   Sip.passThroughUriParameters("From", "uriparm1")
   Sip.passThroughHeaderParameters("From", "hparm1", "hparm2")
end
```

Transparency allows CUCM to pass headers, parameters, and SDP content from one SIP call leg to the other.

The Transparency feature is designed to improve the operation of and interoperability between 3rd Party SIP PBXs and Applications connected via CUCM/SME.

Transparent pass through is only applicable when the call through CUCM is from SIP Trunk to SIP Trunk.

- SIP Trunk ⟷ SIP Trunk calls

Transparency uses the same scripting environment as Normalisation to allow customers to pass SIP messages through CUCM. Transparency and Normalisation features can be combined.

Pre loaded CUCM - REFER Passthrough script

CUCM 8.5 SIP Trunking

Lua Scripting for Normalisation and Transparency

Normalisation and Transparency scripts use Lua - a powerful, fast, embeddable scripting language to modify SIP messages and SDP body content on SIP Trunks. (For more info on Lua see http://lua-users.org/wiki/LuaOrgGuide)

Cisco has created a library of Lua based SIP Message APIs that allow specified information in the SIP message and SDP body to be retrieved, modified, replaced, removed, passed through, ignored, appended to, transformed and so on…

The underlying Lua language allows retrieved information to be stored as variables and operated on using a series of operations such as : If, elseif, while, do, <, >, = etc

The scripting based approach naturally supports multiple variables and state specific contexts for making script decisions.

The combination of Cisco's SIP Message Library APIs and the functionality underlying the Lua language creates a very powerful scripting environment that allows almost any SIP message and/or its SDP body content to be modified.

CUCM 8.5 SIP Trunking

Lua Scripting for Normalisation and Transparency

Lua scripts use "callback functions" to request message types of interest e.g.

inbound_INVITE() outbound_INVITE() inbound_UPDATE()
inbound_3xx_INVITE() outbound_180_INVITE() outbound_SUBSCRIBE()

The Lua script then uses "APIs" defined in the Cisco SIP Message library to access and manipulate message parameters e.g.

getHeader(header-name) - returns header-value or ""
addHeaderValueParameter(header-name, parameter name, [parameter-value])
getUri(header-name) – retrieves the uri from the specified header
block() – blocks the specified SIP message
applyNumberMask(header-name, mask) – retrieves header & applies number mask
getSdp() – returns the SDP content
sdp:getLine(start of line, line contains) returns line in sdp that starts with "start of line" and also has string "line contains"
sdp:modifyLine(start of line, line contains, new-line) finds the line in sdp that starts with "start of line", the line matching "line contains" is replaced with the "new line" parameter

CUCM 8.5 SIP Trunking – Lua Script Example

Scripting Guide : http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/sip_tn/8_5_1/sip_t_n.html

- **SIP Message API - getRequestLine**
- getRequestLine() returns the : method, request-uri, and version

- This API returns three values:
- **• The method name**
- **• The request-uri, and**
- **• The protocol version**

- **Script Example :**
- **M = {}** — *- Initialises the set of call back functions to an empty value*
- **function M.outbound_INVITE(message)** — *- Callback Function executed when outbound INVITE is sent from CUCM*
- **local method, ruri, ver = message:getRequestLine()** — *- Gets method, Request-uri & version & stores values*
- **end**
- **return M**

- **Outbound Message**
- **INVITE sip:1234@10.10.10.1 SIP/2.0**

- **Function Output/Result**
- **method == "INVITE"**
- **ruri == "sip:1234@10.10.10.1"**
- **version == "SIP/2.0"**

Agenda

- CUCM and CUBE – Functionality for IP PSTN Deployments
- CUCM SIP Trunking
- Overview of CUCM SIP Trunking Features
  - SIP Trunk Signalling and Basic Operation
  - SIP Trunk – Load Balancing, Availability & Redundancy
  - SIP Trunk Codec Negotiation – Audio Codec Preference Lists
  - SIP Trunk Security
  - QSIG Over SIP
  - SIP Normalisation and Transparency
- SIP Trunk Design Considerations
- Gateway Integration
- Cisco Unified Border Element (CUBE)

SIP Trunk Design Considerations

Using standard Call Manager Groups, Run on all Active Unified CM Nodes and multiple destination IP addresses

Unified CM SIP Trunks will only accept inbound calls from a device with an IP address that has been defined as a destination IP address on the Trunk

Cluster 1 – SIP Trunk Configuration

The SIP Trunk has an active SIP daemon on Servers A, B, C, D and E

Servers F, G and H are defined as Trunk destinations

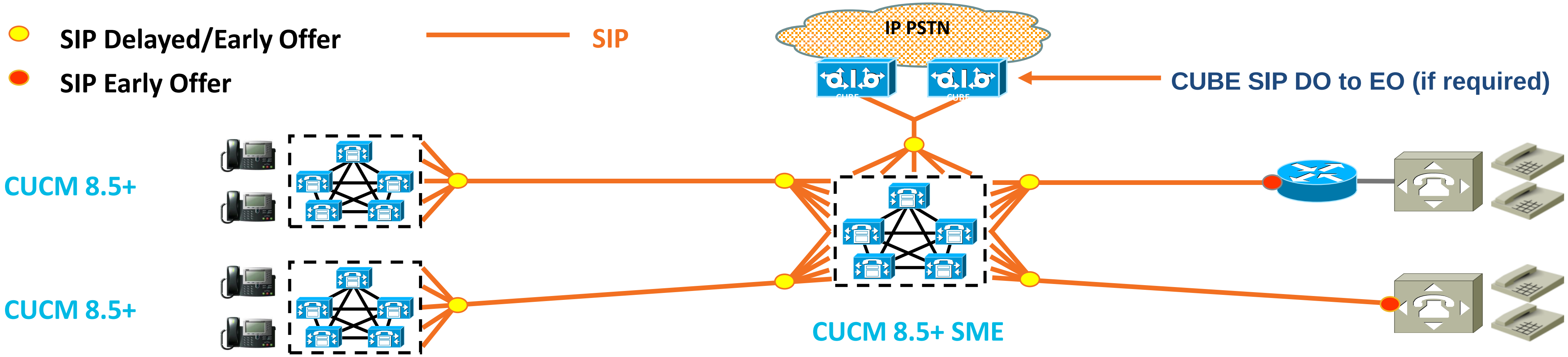
Cluster 2 – SIP Trunk Configuration

Servers F, G and H in SIP Trunk's Call Manager Group

Servers A, B, C, D and E are defined as Trunk destinations

SIP Trunk Design Considerations

Multi-Cluster Designs – with CUCM 8.5+ Leaf Clusters



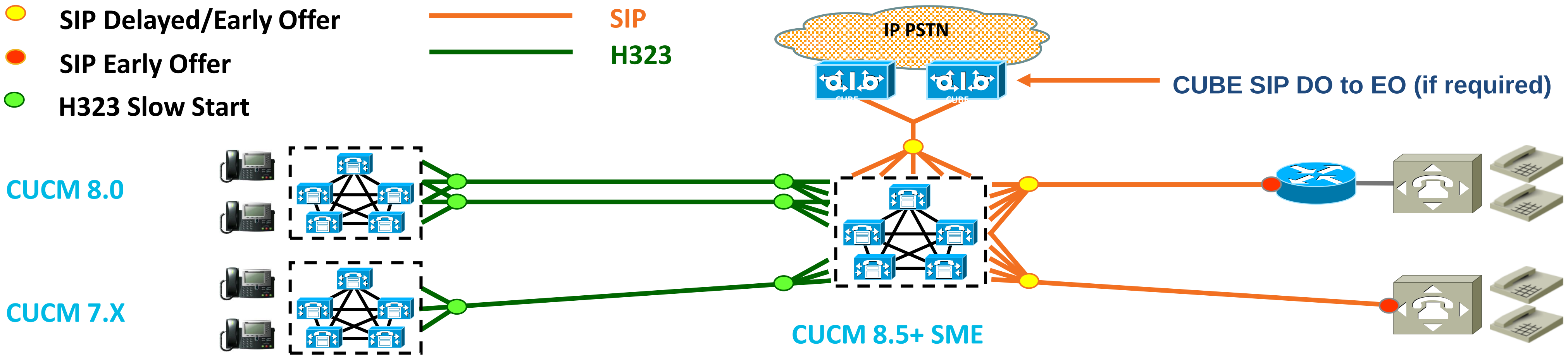
SIP ICT Trunks - Voice, Video and Encryption supported
OPTIONS Ping, SIP Delayed Offer/Early Offer (insert MTP if Needed), Run on All Nodes, Multiple Destination Addresses, QSIG over SIP (For Call Back & Path Replacement)

SIP IP PSTN Trunks – Typically Voice only – Early Offer usually required by Service Provider
OPTIONS Ping, SIP Delayed Offer/Early Offer (insert MTP if Needed), If required - CUBE can provide SIP DO to EO, Run on All Nodes, Multiple Destination Addresses. (EO should be sent by the SP)

SIP to 3rd Party UC Systems – Typically Voice – Video & Encryption also supported
OPTIONS Ping, SIP Delayed Offer/Early Offer (insert MTP if Needed), Run on All Nodes, If required - QSIG over SIP, If end device is capable it should send Early Offer.

SIP Trunk Design Considerations

Multi-Cluster Designs – with Pre 8.5 CUCM Leaf Clusters



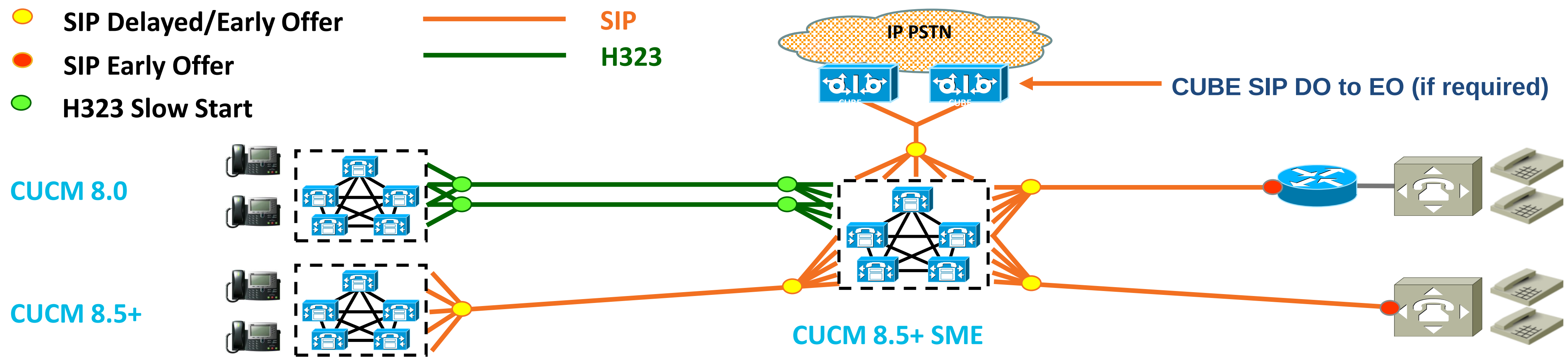
Pre 8.5 Clusters - H323 ICT Trunks - Voice, Video and Encryption supported
H323 Slow Start, Call Manager Groups, 3 Destination Addresses per Trunk, QSIG over H323.

SIP IP PSTN Trunks – Typically Voice only – Early Offer usually required by SP
OPTIONS Ping, SIP Delayed Offer/Early Offer (insert MTP if Needed), If required - CUBE can provide SIP DO to EO, Run on All Nodes, Multiple Destination Addresses. (EO should be sent by the SP)

SIP to 3rd Party UC Systems – Typically Voice – Video & Encryption also supported
OPTIONS Ping, SIP Delayed Offer/Early Offer (insert MTP if Needed), Run on All Nodes, If required – QSIG over SIP, If end device is capable it should send Early Offer.

SIP Trunk Design Considerations

Multi-Cluster Designs – Mixed Leaf Cluster CUCM Versions

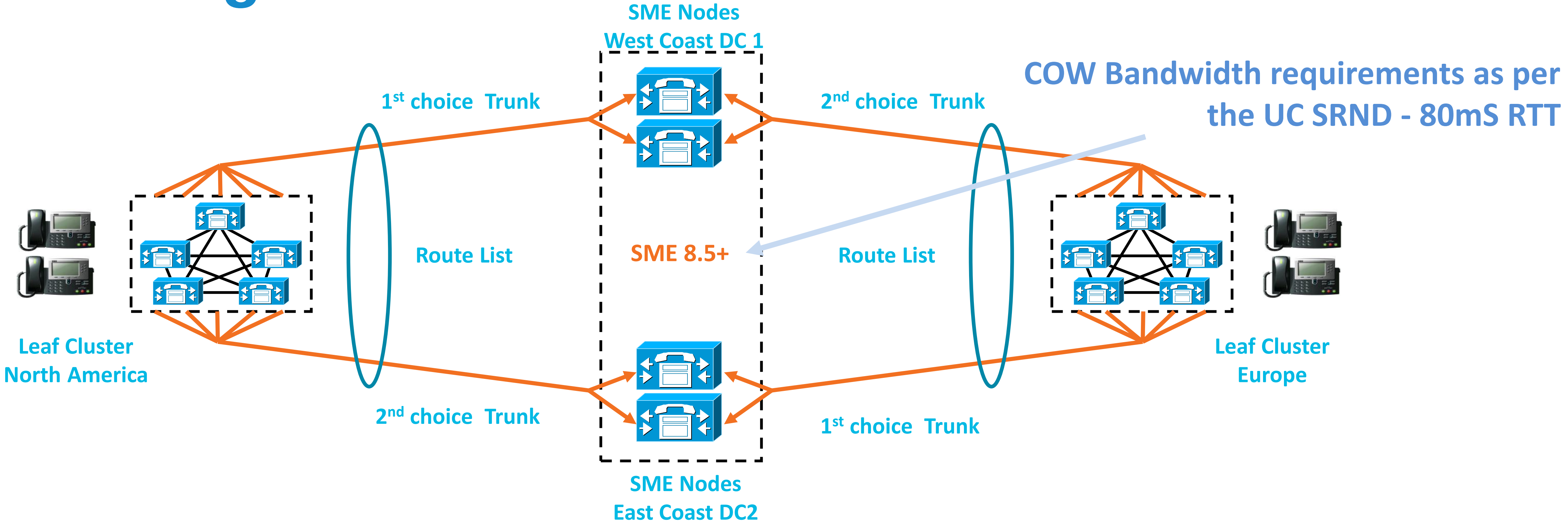


Pre 8.5 Leaf Clusters – H323 ICT Trunks – Voice, Video and Encryption supported
H323 Slow Start Trunks, Call Manager Groups, 3 Destination Addresses per Trunk, Route Lists & Route Groups (with considerations for active Route List node) QSIG over H323.

8.5+ Leaf Clusters – SIP ICT Trunks – Voice, Video and Encryption supported
SIP Delayed Offer, OPTIONS Ping, Run on All Nodes, Multiple Destination Addresses, QSIG over SIP.

SIP IP PSTN Trunks – Typically Voice only – Early Offer usually required by SP
OPTIONS Ping, SIP Delayed Offer, If required - CUBE can provide SIP DO to EO, Run on All Nodes, Multiple Destination Addresses. (EO should be sent by the SP)

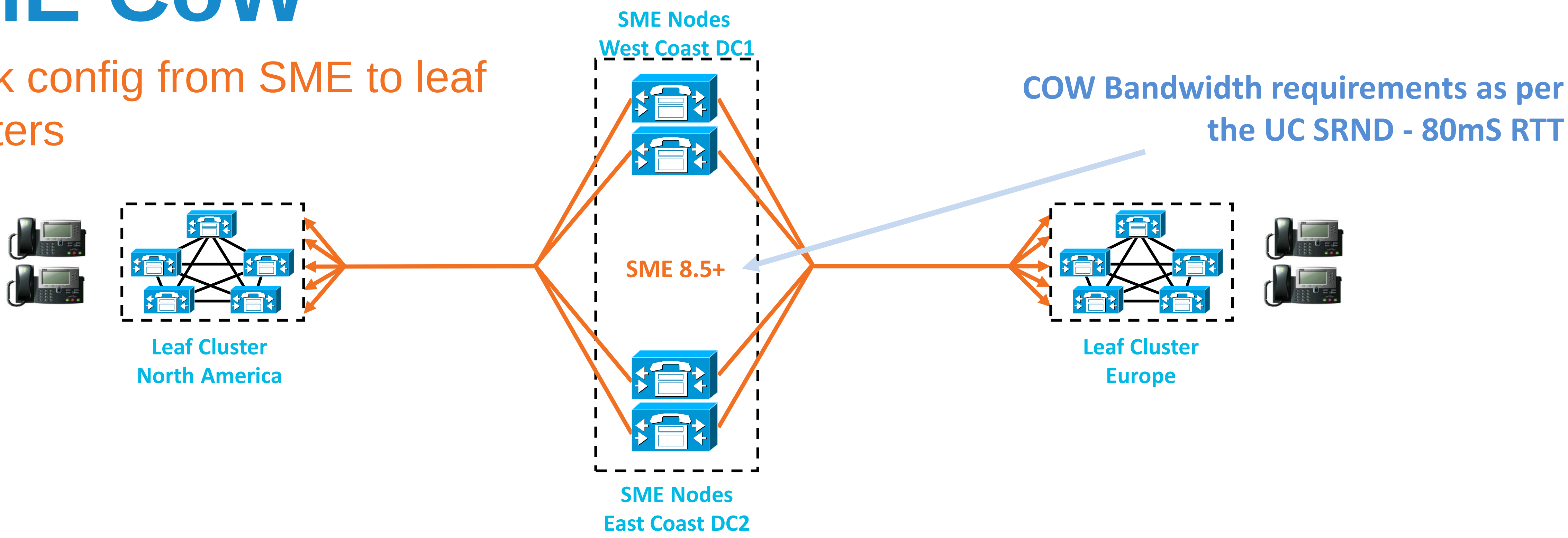
SME Clustering Over the WAN (CoW)
Trunk config from Leaf Cluster to SME



One Leaf cluster SIP Trunk to each pair of SME nodes in each regional data centre
Each Leaf Cluster SIP Trunk uses “Run on all Unified CM nodes”
Leaf Cluster SIP Trunk 1 - Uses multiple destination addresses pointing to each SME node DC 1
Leaf Cluster SIP Trunk 2 - Uses multiple destination addresses pointing to each SME node DC 2
Leaf Cluster Trunks placed into Route Lists and Route Groups for redundancy

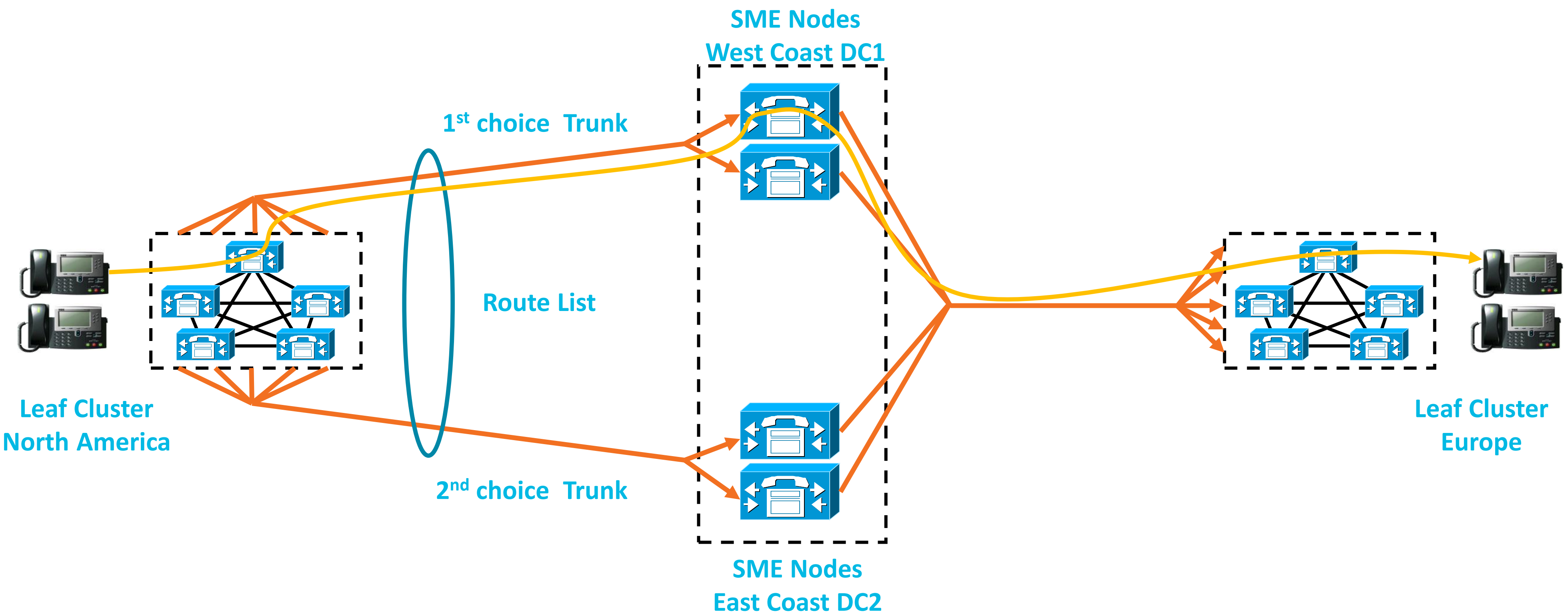
SME CoW

Trunk config from SME to leaf Clusters



One SIP Trunk from SME to each Leaf cluster
Each SME SIP Trunk uses “Run on all Unified CM nodes”
Each SME SIP Trunk uses multiple destination addresses pointing to every call processing node in the destination leaf cluster

SME CoW Cluster – Call Routing – Route Local



Leaf Clusters

Multiple Trunks in Route Groups provide ordered selection of SME nodes. Route List Call Distribution – priority order – nearest data centre 1st, second nearest data centre 2nd etc

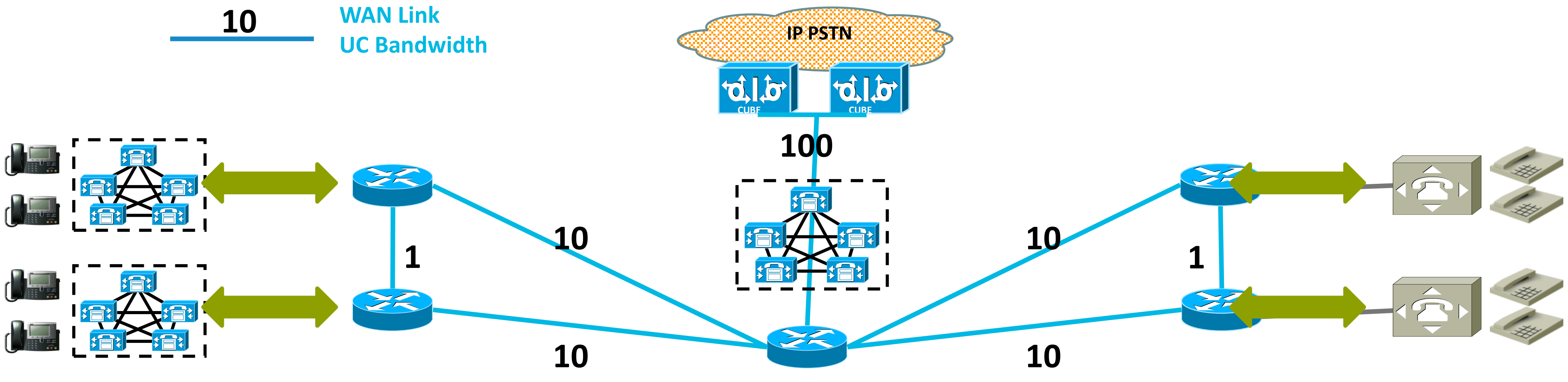
SME cluster

Trunks with "Run on all Nodes" enabled pointing to all nodes in each leaf cluster

Route local operates in the SME cluster – No inter node – intra cluster call routing

SIP Trunk Design Considerations

Partially Meshed Intercluster WAN – Call Admission Control



For CUCM Leaf Clusters

Using Locations based Call Admission Control….

What bandwidth value should be used for calls over SIP inter cluster Trunks into the WAN ?

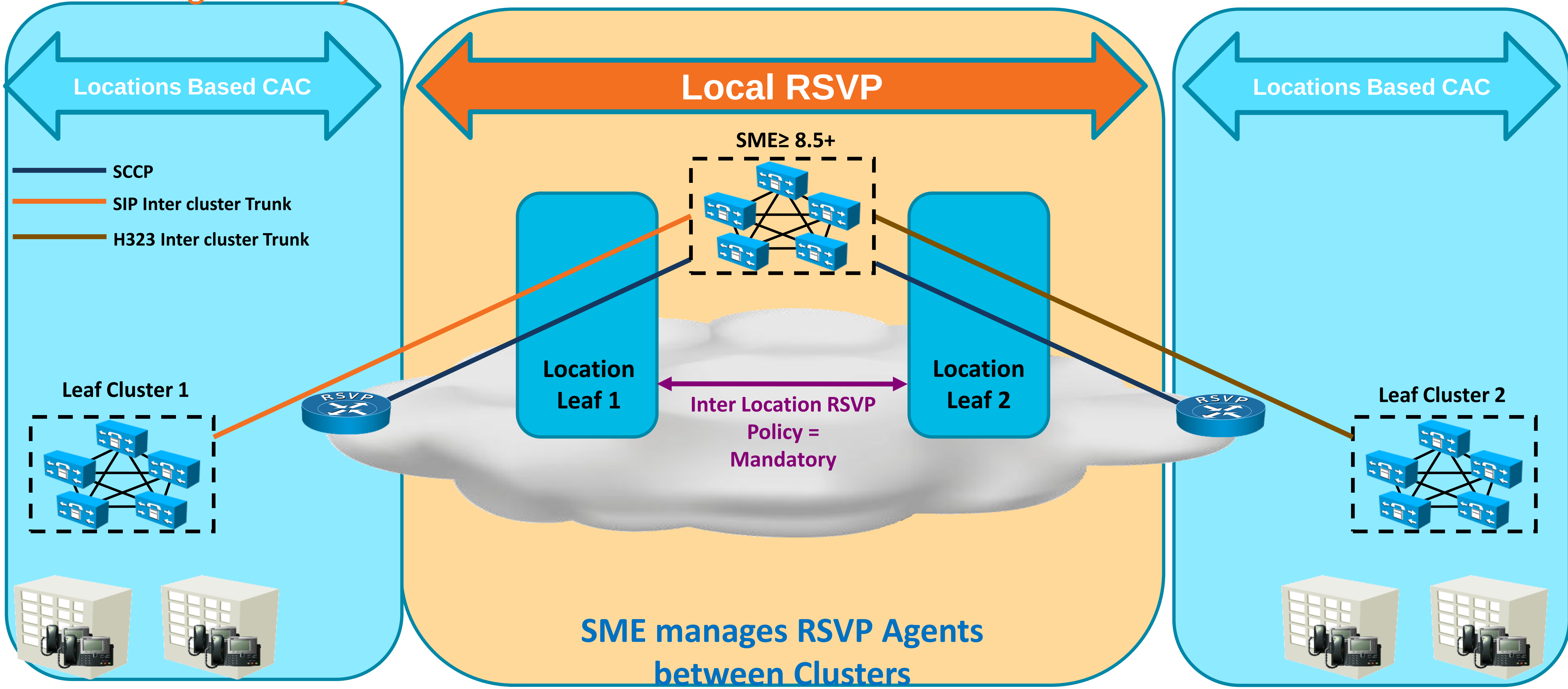
For Voice Gateways

Using Call Counting based Call Admission Control….

What value should be used for calls over SIP Trunks into the WAN ?

SIP Trunk Design Considerations

Addressing Partially Meshed Inter Cluster WAN Issues with RSVP



Using Local RSVP in the SME cluster has advantages where partially meshed WAN circuits are used, or where sites are dual homed to the WAN. RSVP does not need to be implemented in the leaf clusters

Agenda

- CUCM and CUBE – Functionality for IP PSTN Deployment
- CUCM SIP Trunking
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  - SIP Trunk Codec Negotiation
  - SIP Trunk Security
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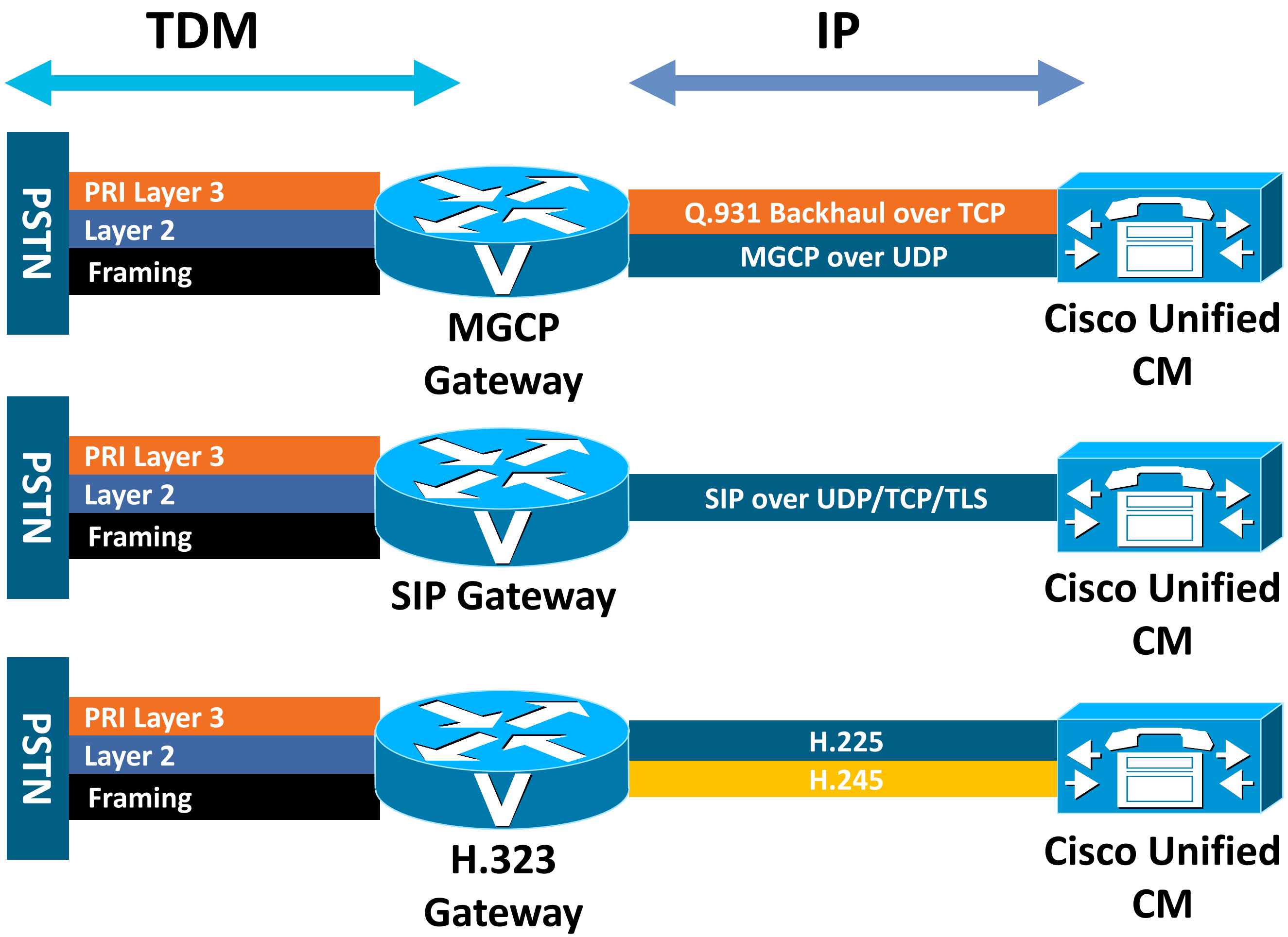
- Cisco Unified Border Element (CUBE)

Unified CM SIP Trunk — Gateway Integration

Technology Basics



Types of Gateways

– MGCP, H.323, and SIP

Gateway Deployment

– Central site

– Distributed at branch locations

Gateway Selection

– Route Patterns point to Gateways

– Route Patterns route calls to Gateways via Route Lists and Route Groups

IOS Gateway Trunks

SIP Trunks , H.323 Gateways, MGCP Gateways – Feature Comparison

	H.323	SIP	MGCP
Centralised Provisioning	No	No	Yes
QSIG Tunnelling	No	Yes	Yes
Centralised CDR (DS0 Granularity in Unified CM CDR)	No	No	Yes
MLPP (Preemption)	No	Yes	Yes
Hook-flash Transfer with Unified CM	No	Yes	Yes
ISDN Overlap Sending	Limited support	No	Yes
Accept Audio Codec Preferences in Received Offer	No	Yes	No
CUCM 8.5 "Run On All Nodes" feature	3 Active Nodes in a CMG	Yes	1 Active Node in a GMG
CUCM 8.5 "Up to 16 destination addresses" feature	No	Yes	No
Mobility Manager VXML-Based Voice Profile Mgmt	Yes	Yes	No
OPTIONS Ping	No	Yes	No
TCL/VXML Apps (e.g. for CVP Integration)	Yes	Yes	No
Voice & Data Integrated Access	Yes	Yes	No
Fractional PRI	Yes	Yes	No
TDM Variations: A-DID, E&M, PRI NFAS, CAMA, T1 FGD	Yes	Yes	No
ISDN Video Switching on GW	Yes	Yes	No
IPv6, Dual Stack, ANAT	No	Yes	No

Legend: Yes | Limited support | No

CMG – Call Manager Group

Agenda

- CUCM and CUBE – Functionality for IP PSTN Deployments
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- New CUCM SIP Trunking Features
  - SIP Trunking Configuration Overview
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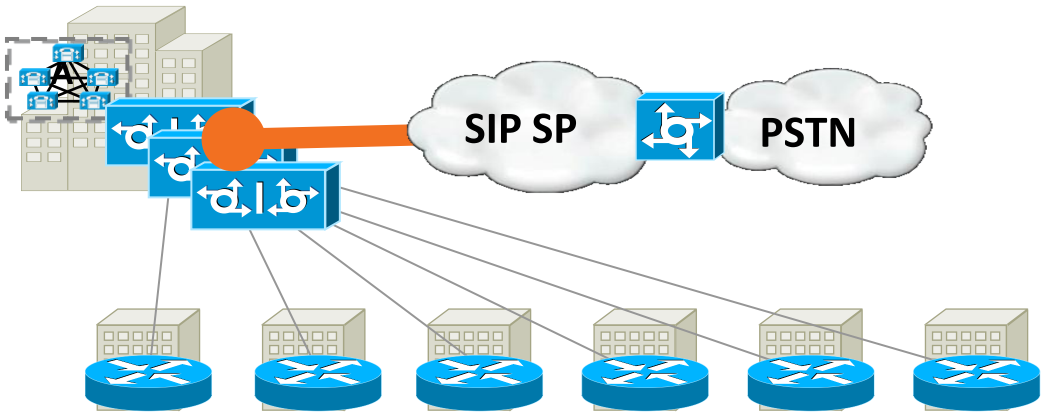
Cisco Unified Border Element

Feature Summary

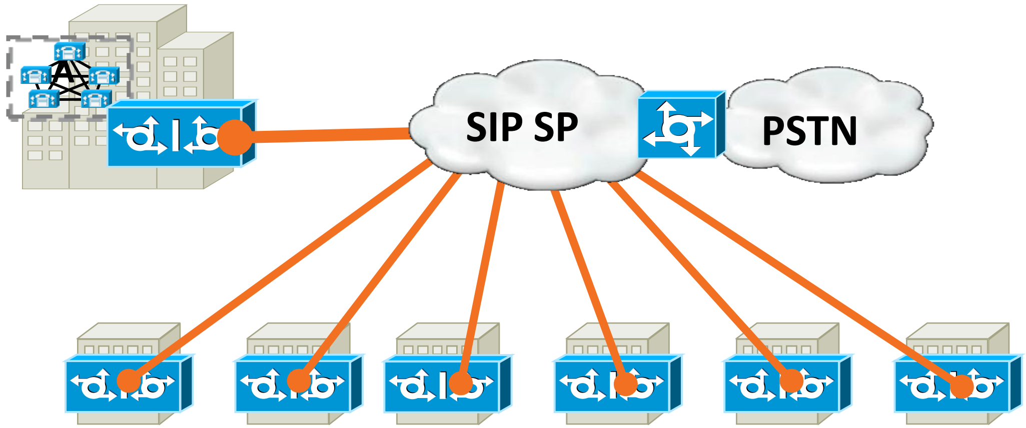
SESSION CONTROL	SECURITY	INTERWORKING	DEMARCATION
Call Admissions Control	Encryption	SIP - SIP	Fault Isolation
Ensuring QoS	Authentication	H.323 - SIP	Topology Hiding
Statistics and Billing	Registration	SIP Normalisation	Network Borders
Redundancy/ Scalability	SIP Protection	DTMF Interworking	L5/L7 Protocol Demarcation
	Firewall Placement	Transcoding	
	Toll Fraud	Codec Filtering	

SIP Trunk Designs & Capacity Requirements

Centralised SIP Trunk Design



Distributed SIP Trunk Design

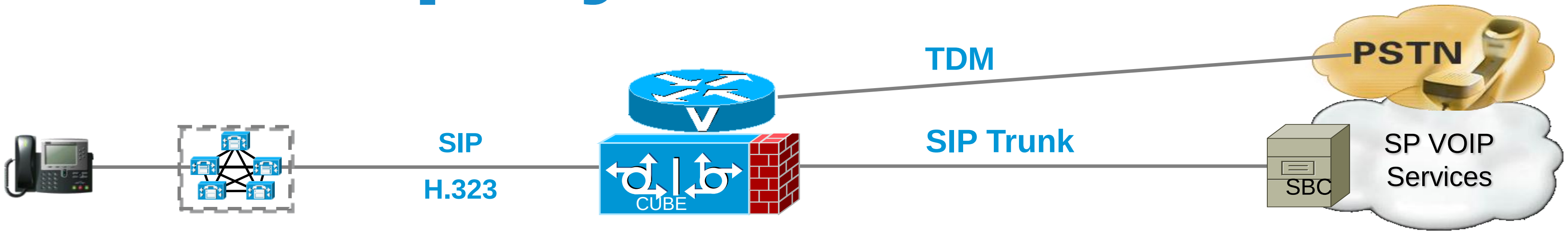


- **Centralised**: Calls from remote offices traverse the WAN and aggregate onto a large-scale SIP trunk at a central site
- Contact Centre
- Large session counts are typical
  - 400 to 10000+
  - A redundant design is critical

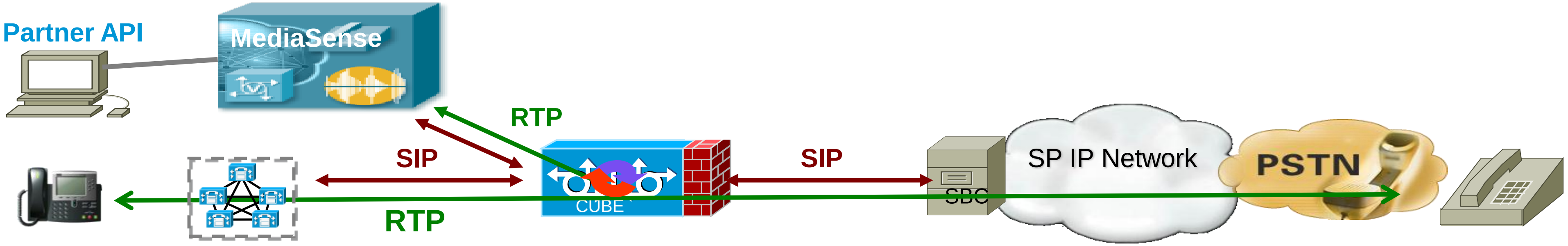
- **Distributed**: Calls from remote offices have individual SIP trunks and calls go directly to the SP from each site
- Small session counts are typical
  - 4 to ~200
  - A redundant design is typically not required, but possible

CUBE Deployment Scenarios

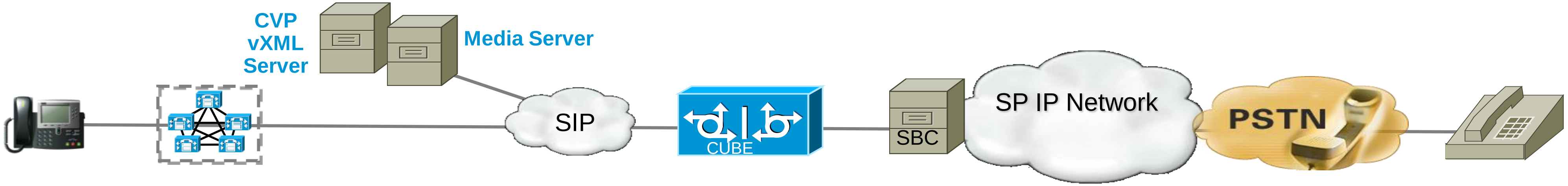
SIP Trunks for PSTN Access



Network-based Media Recording Solution

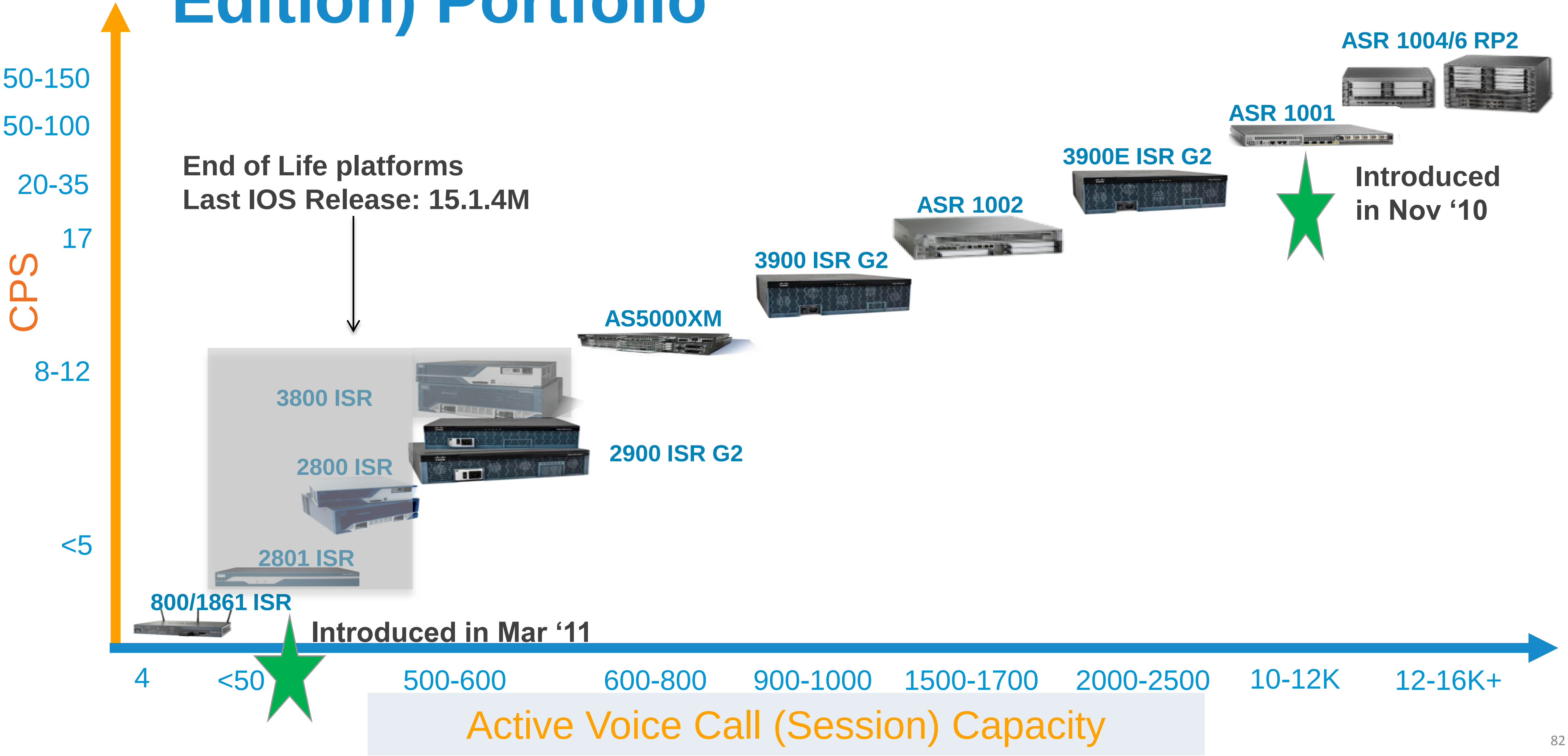


IVR Integration for Contact Centres



Business to Business Telepresence

Cisco Unified Border Element (Enterprise Edition) Portfolio

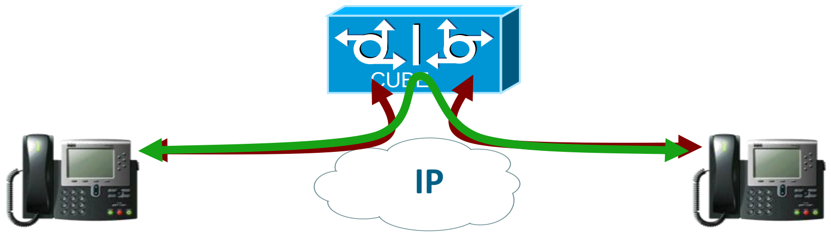


CUBE Sizing Recommendations Based on Concurrent SIP Sessions

Enterprise Size	SIP Trunk Sessions	Redundancy Recommen-dation	Platform Recommendation
Small	<100	None	Single 2901
	100-200	None	Single 2911
	200-500	Optional	Single 2951
Medium	500-1000	Recommended	No redundancy: Single 3900 Local redundancy: Dual Box2Box 3900 Geo Redundancy: Dual 3900
	1000-2500	Must-have	Local redundancy: Dual Box2Box 3900E Geo redundancy: Dual 3900E
Large	2500-10000	Must-have	Inbox redundancy: Single ASR1001/6 Local redundancy: Dual Box2Box ASR1001 Geo redundancy: Dual ASR1001/6
Very Large	10,000+	Must-have	Inbox redundancy: Single ASR1006 Local redundancy: Dual Box2Box ASR1004 Geo redundancy: Dual ASR1004/6

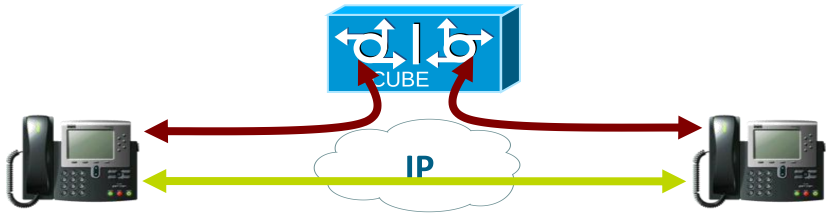
Cisco Unified Border Element Architecture

- Actively involved in the call treatment, signalling and media streams
  - SIP B2B User Agent
- Signalling is terminated, interpreted and re-originated
  - Provides full inspection of signalling, 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

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



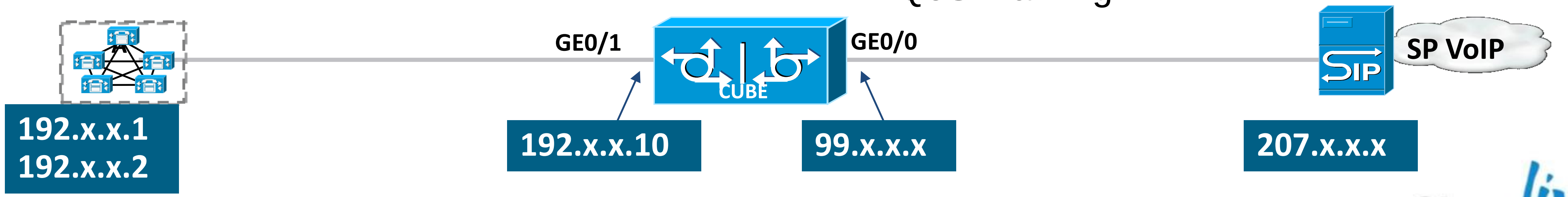
Media Flow-Around

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

CUBE Configuration Areas

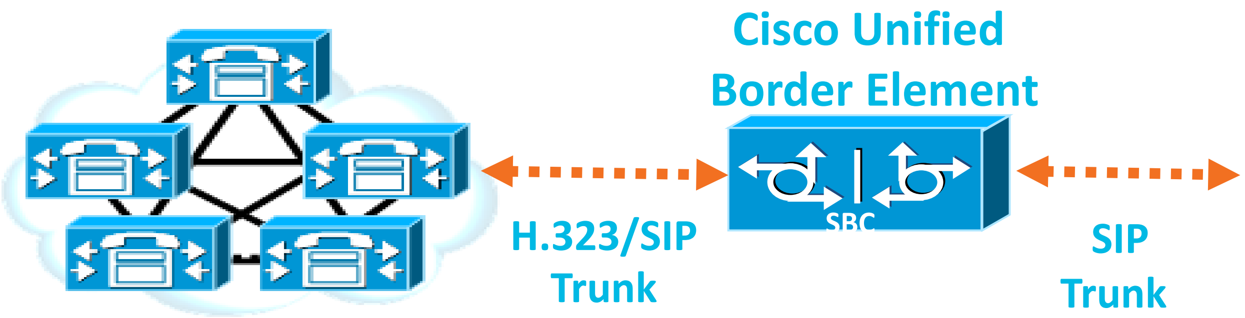
- Generic router capabilities
  - Routing, IP connectivity, Interfaces, ACLs
  - DHCP, QoS, FW…
- Global CUBE capabilities
  - Enable CUBE
  - CAC and SIP capabilities
  - Transcoding, codec classes and preferences
- Security Configuration
  - Secure peer IP addresses
  - Call Spike monitoring

- SIP Configuration
  - Message handling and interpretation
  - SIP Normalisation (all calls)
  - Fax (all calls)
  - Failover timers
- Dial-peer Configuration
  - Dial-plan; Digit Manipulation
  - SIP Normalisation (per destination)
  - DTMF settings
  - Fax (per destination)
  - QoS Marking



Unified CM SIP Trunk – CUBE Integration

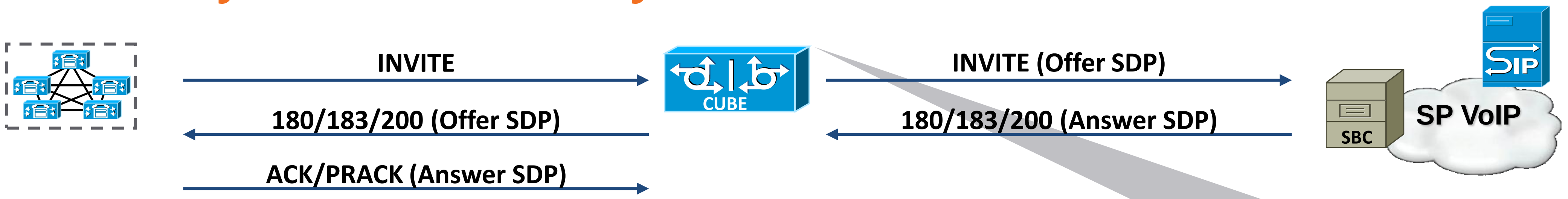
CUBE Features for SIP Calls



- H.323 to SIP Translation
- Delayed to Early Offer No MTP, Multiple Codecs
- Codec Filtering
- DTMF Relay
- Transcoding
- SIP Message Normalisation
- Connection Monitoring
- Call Spike Prevention
- Call Admission Control
- Address Hiding
- Dial Peer Hunt Groups

Unified CM SIP Trunk – CUBE Integration

SIP Delayed Offer—Early Offer



SP SIP trunk Early Offer (EO) interconnect for enterprise apps that support only Delay Offer (DO)

Flow-through required for DE-EO supplementary services

```
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
```

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

Global Configuration Also Supported:

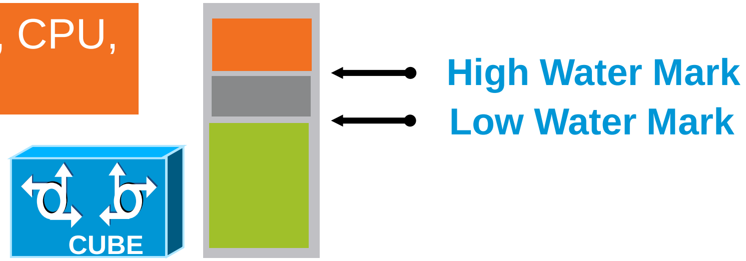
```
voice service voip
  sip
    early-offer forced
```

Call Admission Control at the Edge...

CUBE provides various CAC mechanisms to safeguard your network from SIP based attacks and to enforce policies based on:

- Total calls
- CPU & Memory
- Call spike detection
- Maximum connections per destination
- Dial-peer or interface bandwidth

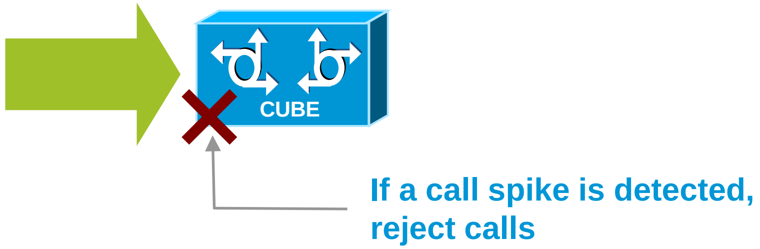
Total Calls, CPU, Memory



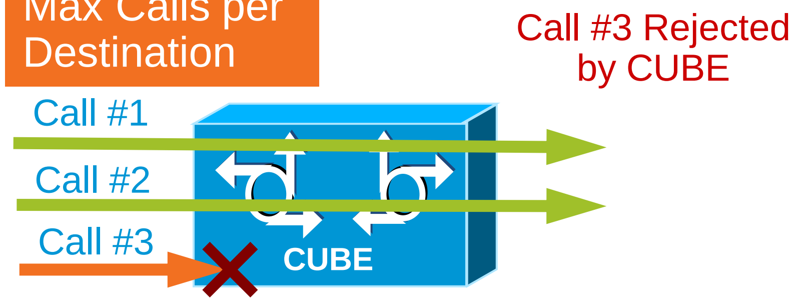
call threshold global [total/mem/cpu] calls low xx high yy
call treatment on

Call Spike Detection

call spike call-number [steps number-of-steps size milliseconds]
call spike 10 steps 5 size 200

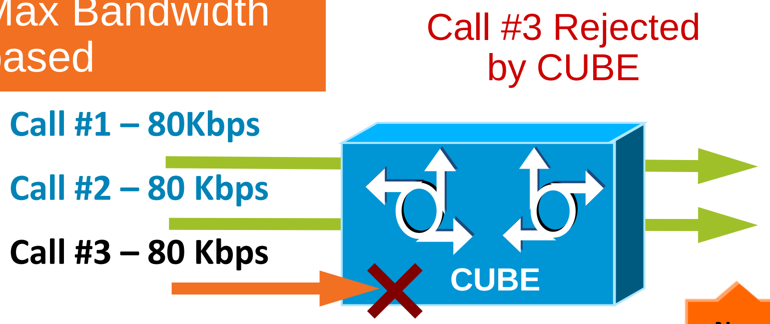


Max Calls per Destination



dial-peer voice 1 voip
max-conn 2

Max Bandwidth based



dial-peer voice 1 voip
max-bandwidth 160

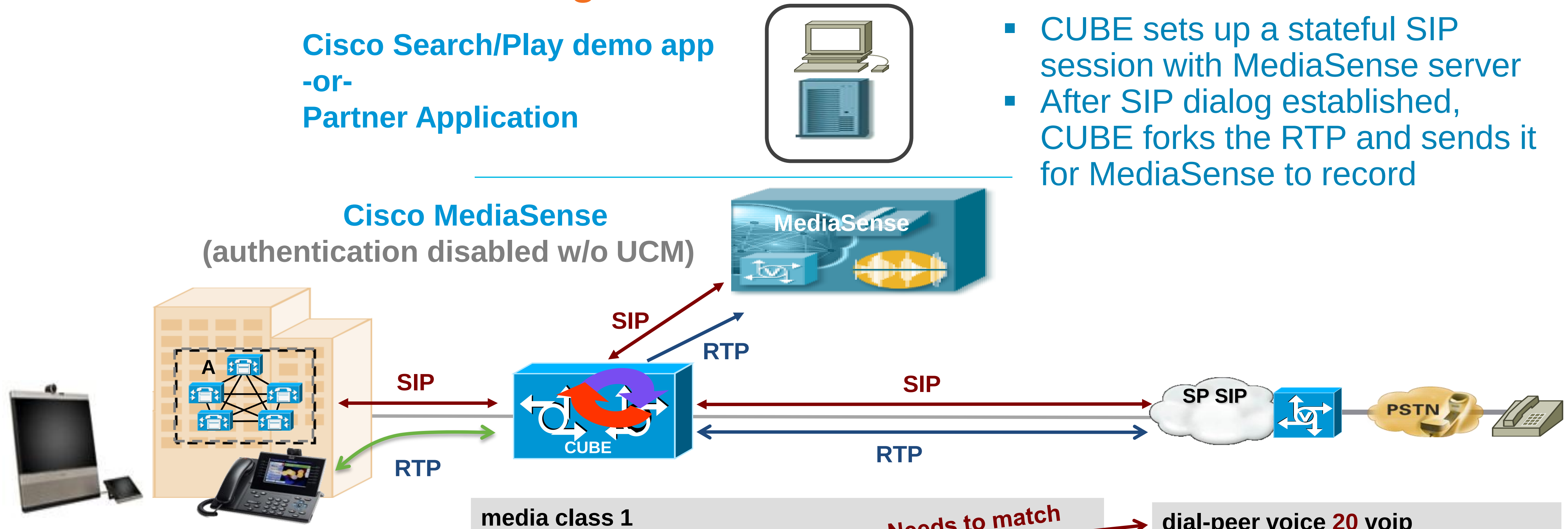
New Features

Media (Audio & Video) Forking

Network based recording solution

15.2.1T

- CUBE sets up a stateful SIP session with MediaSense server
- After SIP dialog established, CUBE forks the RTP and sends it for MediaSense to record



- Call agent independent
- Configured on a per Dialpeer level

```
media class 1
  recorder parameter
    media-recording 20

dial-peer voice 1 voip
  description dial-peer that needs to be forked
  session protocol sipv2
  media-class 1
```

Needs to match

```
dial-peer voice 20 voip
  description dial-peer pointing to
MediaSense
  session protocol sipv2
  session transport tcp
```

Toll Fraud Mitigation – Trusted IP addresses

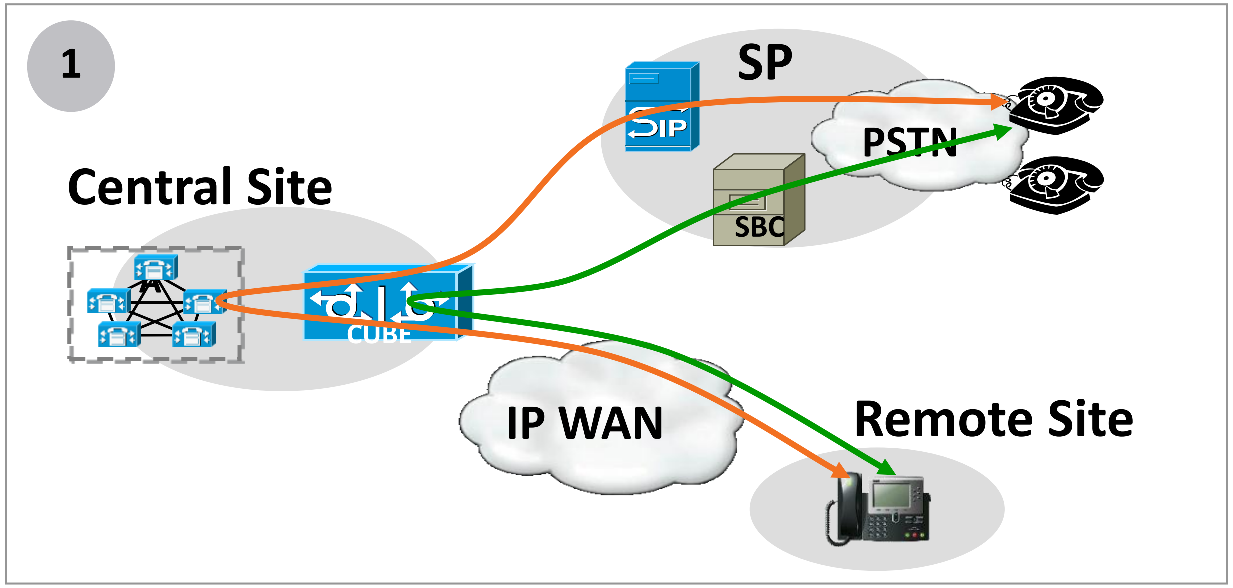
15.1.2T

- Default operation in 15.1.2T has changed
- As of 15.1.2T, by default, only calls from “trusted” source IP addresses will be accepted – similar to CUCM operation
- If you want to restore pre-15.1.2T default operation, use “voice service voip > no ip address trusted authenticate”. This is NOT RECOMMENDED.


```
voice service voip
 ip address trusted list
  ipv4 10.10.10.1
  ipv4 10.10.10.2
  ipv4 11.11.11.1
```

Toll Fraud Prevention – more info:
http://www.cisco.com/en/US/tech/tk652/tk90/technologies_tech_note09186a0080b3e123.shtml

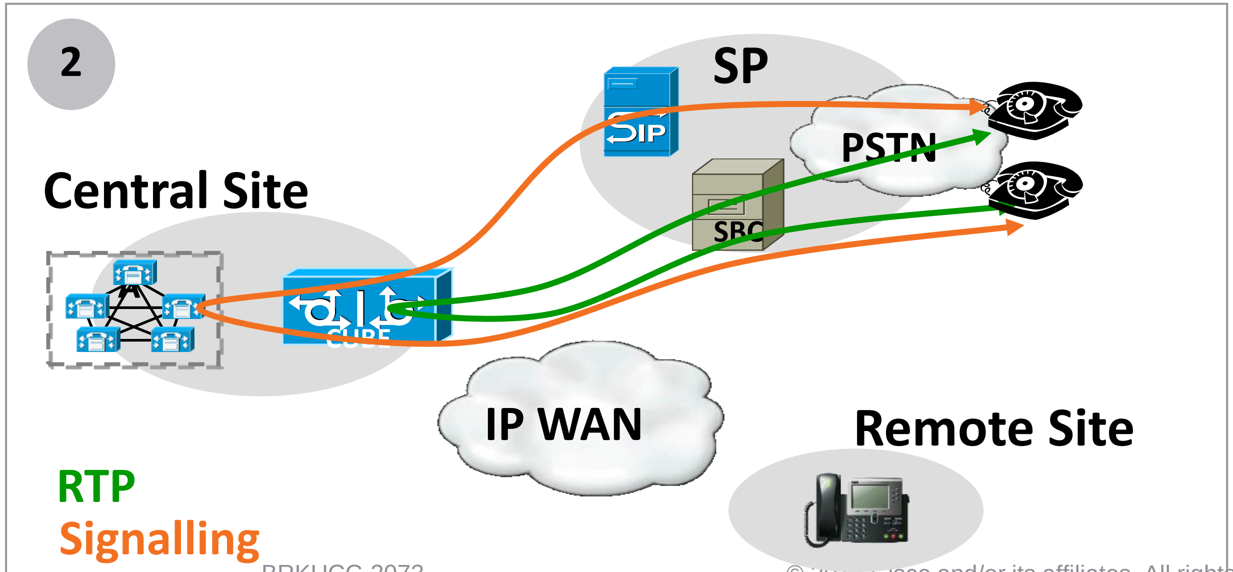
CUBE – Media Anti-Tromboning

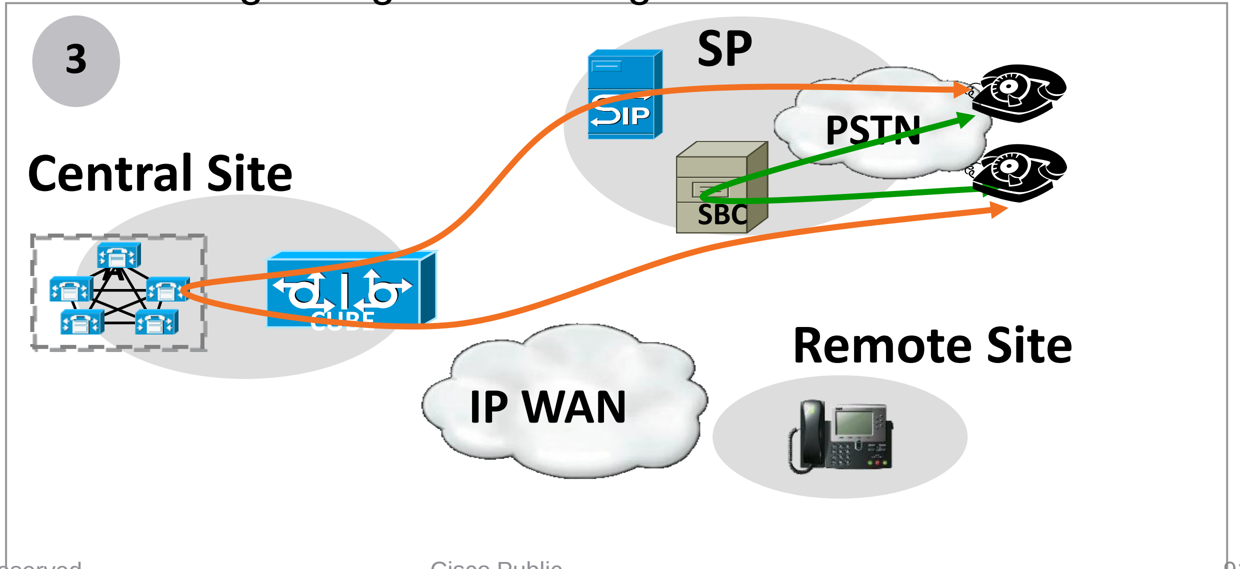


1
- Call arrives on a centralised SIP trunk to an enterprise endpoint
- Media and signalling flows through CUBE
- Signalling flows through CUCM

2
- Call is transferred back to a PSTN endpoint
  - Media and signalling flows through CUBE
  - Signalling flows through CUCM

3
- Media Anti- tromboning :
  - Media is released back to the PSTN
  - Signalling flows through CUCM and CUBE

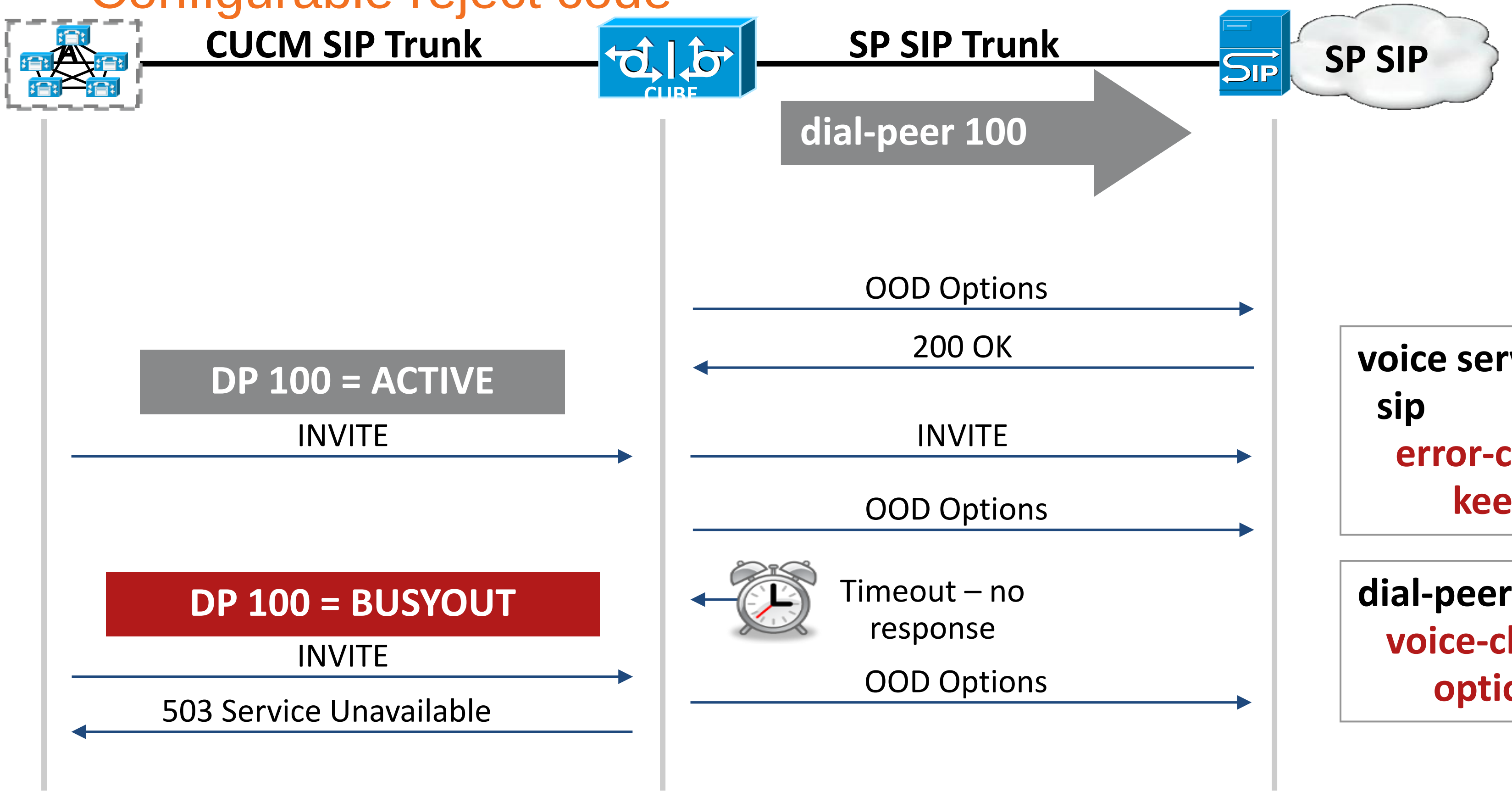




CUBE – SIP Trunk Monitoring with SIP Out Of Dialog (OOD) Options Ping

Dial-peer status based on SIP OOD Options Ping responses (or timeouts)

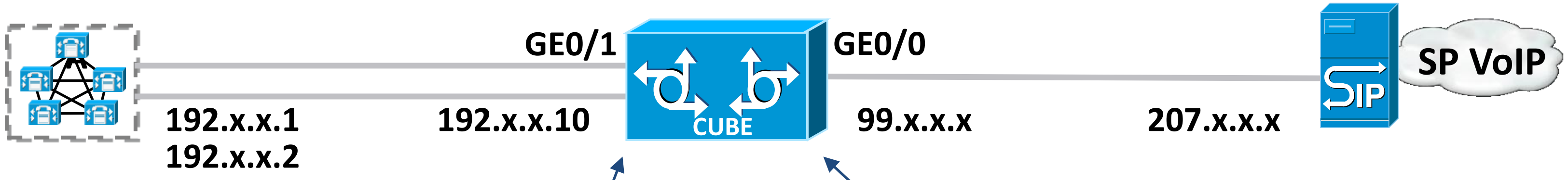
Configurable reject code




```
voice service voip
 sip
  error-code-override options-
     keepalive failure 503
```

```
dial-peer voice 12 voip
 voice-class sip error-code-override
     options-keepalive failure 503
```

CUBE Dial-Peer Configuration



DID range:
14085551000-5999

```
voice translation-rule 9
 rule 2 /^9\(.*\)/ /\1/
!
voice translation-profile DIGITSTRIP-9
 translate called 9
```

```
dial-peer voice 102 voip
 description to/from CUCM subscriber
 preference 2
 destination-pattern 408555[1-5]...
 voice-class codec 1
 session protocol sipv2
 session target ipv4:192.168.0.1
 incoming called-number 408555[1-5]...
 dtmf-relay rtp-nte
```

```
dial-peer voice 103 voip
 description to/from CUCM publisher
 preference 2
 destination-pattern 408555[1-5]...
 voice-class codec 1
 session protocol sipv2
 session target ipv4:192.168.0.2
 incoming called-number 408555[1-5]...
 dtmf-relay rtp-nte
 no vad
```

```
dial-peer voice 1 voip
 description OUTBOUND SIP calls to SP
 translation-profile outgoing DIGITSTRIP-9
 max-conn 50
 destination-pattern 91[2-9][1-9][1-9].......
 voice-class codec 1
 session protocol sipv2
 session target ipv4:207.x.x.x
 dtmf-relay rtp-nte
```

```
dial-peer voice 2 voip
 description INBOUND SIP calls from SP
 max-conn 50
 voice-class sip options-keepalive up-interval
          10 down-interval 60 retry 2
 session protocol sipv2
 session target sip-server
 incoming called-number 408555[1-5]...
 dtmf-relay rtp-nte
 no vad
```

Cisco Unified Border Element

Leverage all the advantages Cisco has to offer

MIGRATE WITH EXISTING EQUIPMENT

- Network devices are multipurpose
- Equipment inventory is simplified
- Leverage existing training
- Migration to SIP is phased

INTEROPERABILITY

- Tested with PBX's
- Validated with Service Providers
- Standards Based

CUBE ADVANTAGE

END TO END SUPPORT

- Safe, Trusted, Reliable
- Familiar interfaces and management
- Portfolio breadth

STATE OF THE ART TECHNOLOGY

- Largest R&D spending
- Revolutionary Platforms
- Broadest depth of protocols: SIP plus more

Unified CM SIP Trunk

Key Takeaways

- SIP is gaining popularity and SIP implementation in Cisco Unified Communications product portfolio is expanding rapidly
- A single protocol to inter-work all devices provides seamless integration and a richer user experience
- SIP Trunk on the Cisco Unified Communications Manager is the key to make this integration happen. It is not just for PSTN access
- Understanding the capabilities and working of this SIP Trunk is essential to building and deploying a Cisco Unified Communications network

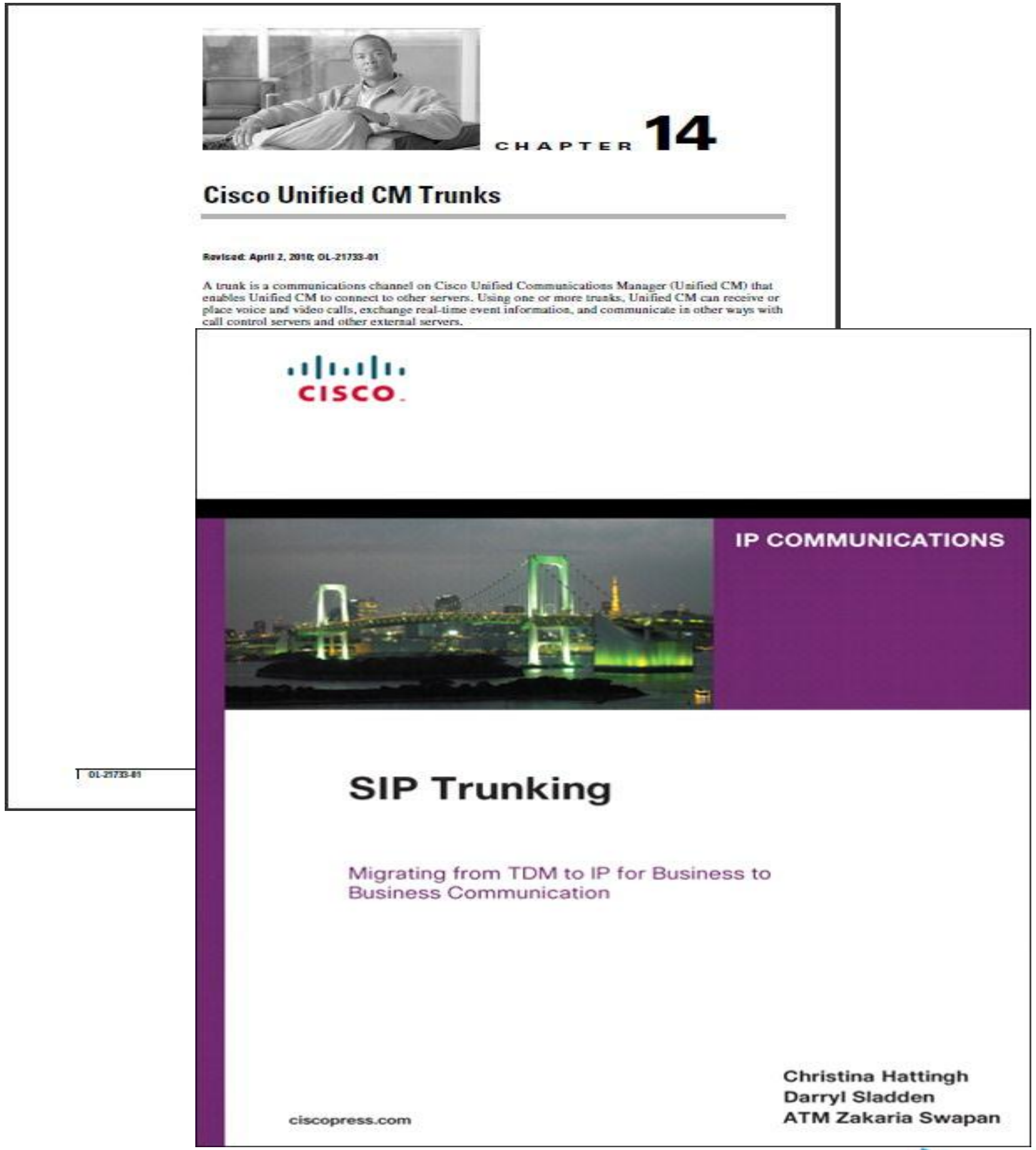
Unified CM SIP Trunk

Recommended Reading

- Cisco Unified Communications Solution Reference Network Design (SRND) for Cisco Unified Communications Manager Release 8.x, available online at:

www.cisco.com/go/designzone

- SIP Trunking: Migrating from TDM to IP for Business to Business Communications by Hattingh, Sladden, and Swapan (ISBN: 1-58705-944-4)



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