
TOMORROW
starts here.

SIP Trunk Design and Deployment in Enterprise UC Networks

BRKUCC-2006

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#clmel

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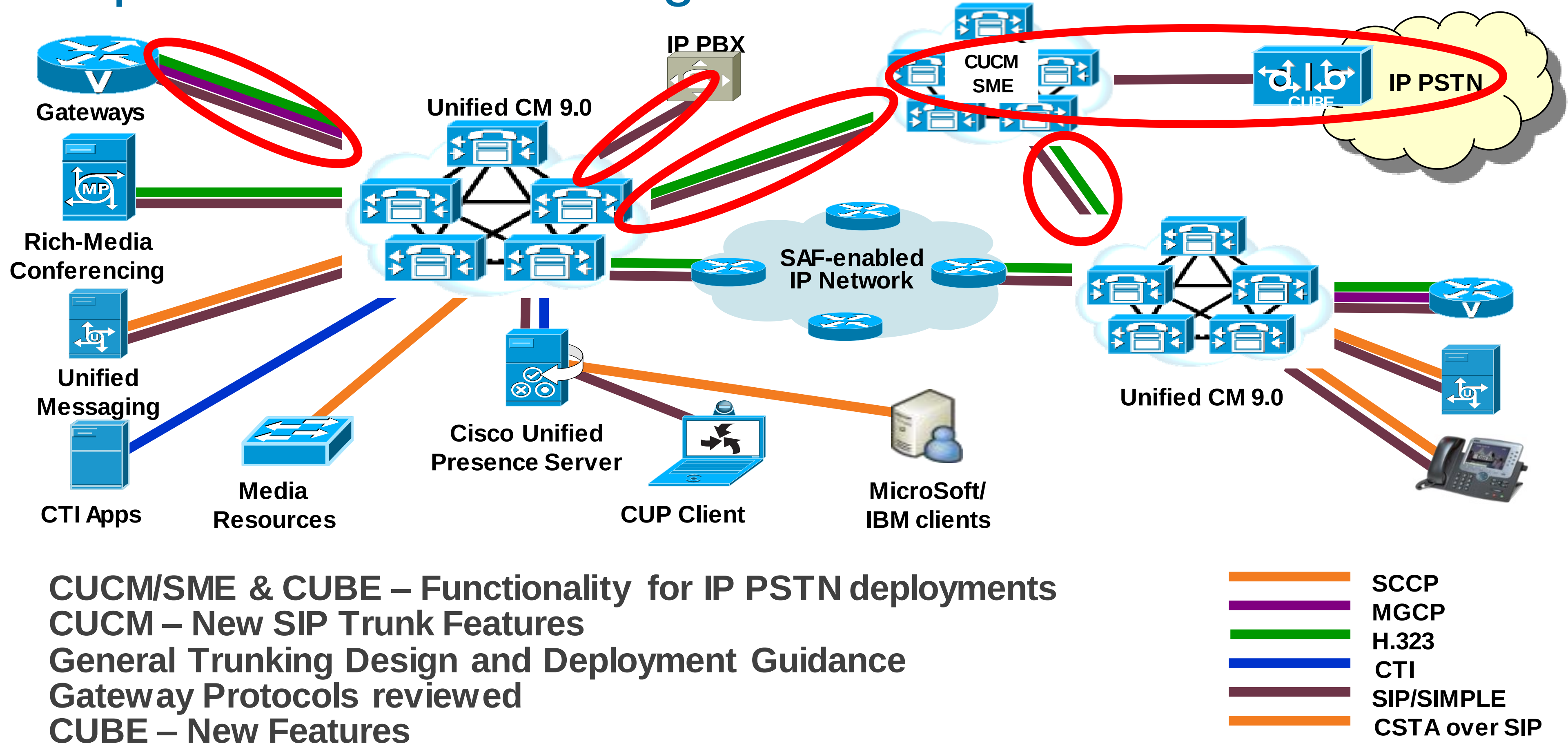
Aspects of SIP Trunking Covered in this Presentation



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

Objectives of this Session

a) Provide a quick overview of SIP basics
b) To give you a comprehensive overview of Cisco Unified Border Element (CUBE) Enterprise
c) To analyse real SIP traces and to explain how the various headers and SDP information relate to CUCM SIP Trunk configuration
d) To explain how CUCM SIP Trunk features operate and how and when these features can be use
e) To leave this audience with a good understanding of CUCM/CUBE SIP Trunk operation

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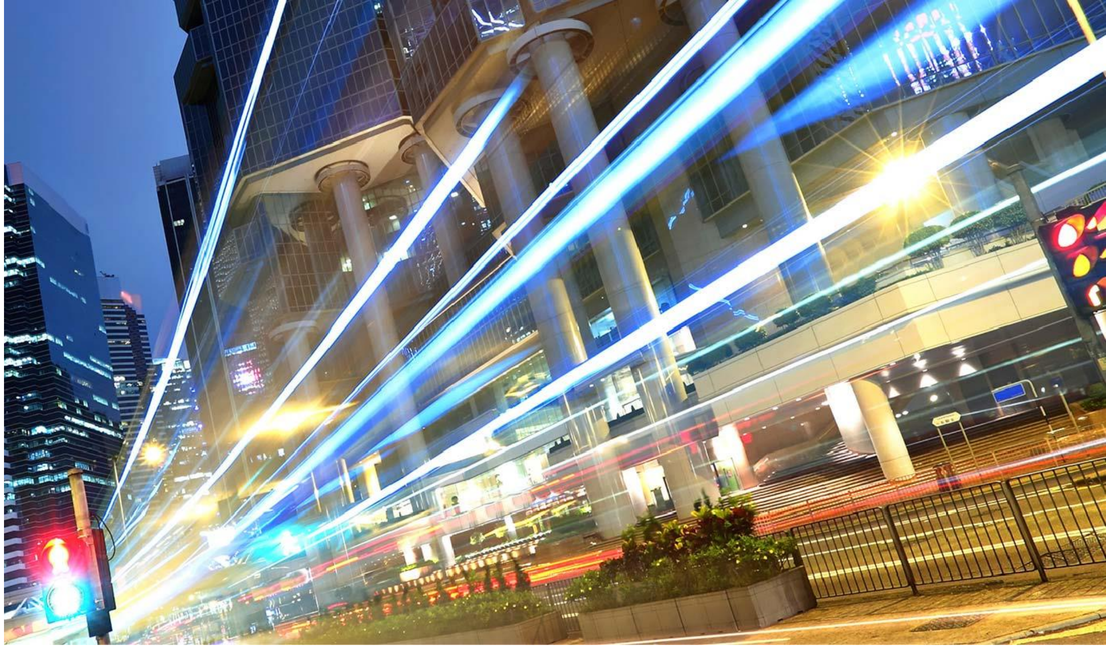
Agenda

- **Why choose SIP for UC Trunking ?**
- SIP Basics
- Cisco Unified Border Element (CUBE)
- Deep down into SIP
- Deep down into SDP
- CUCM SIP Trunk features and Call functions
- Key Takeaways



Why Choose SIP ?

Popularity – Industry demand
Multi-protocol Interoperability challenges
UC 8.5 + Protocol Feature Set Comparison

Unified CM Inter Cluster Trunks

SIP Trunks vs H.323 Trunks – Feature Comparison

	H.323	SIP
Support for “+” character		
Signalling Authentication and Encryption		TLS
Media Encryption		
“Run On All Nodes” feature		
“Up to 16 destination addresses” feature		
OPTIONS Ping		
SME CoW with extended Round Trip Times		
G.711, G.722, G.723, G.729 codec support		
SIP Subscribe / Notify, Publish – Presence		
Accept Audio Codec Preferences in Received Offer		
URI Call Routing, Global Dial Plan Replication		
IPv6, Dual Stack, ANAT		
BFCP – Video Desktop Sharing		

Legend: Yes | Limited support | No

Why Recommend SIP as The Preferred Trunk Protocol?

Using SIP Trunks only to interconnect UC systems provides real benefits in terms of feature support and design simplicity e.g. :

• UC 7.1	Support for IPv6
• UC 8.5	Simplified Call Routing
• UC 8.5	SME clusters which need no Media Resources (MTPs, Xcoders…)
• UC 8.5	H.264 Video support
• UC 8.5	Improved Interoperability – with powerful SIP Trunk LUA scripts
• UC 8.6	BFCP and Encrypted Video support
• UC 9.0	Codec Preference Lists and Codec Preference Pass through
• UC 9.0	ILS URI distribution and URI Call Routing over SIP Trunks
• UC 9.1	SME CoW with extended Round Trip Times
• UC 10.0	Support for ILS based Global Dial Plan Replication (GDPR)
• UC 10.0	SDP Transparency
• UC 10.5	Best Effort Early Offer

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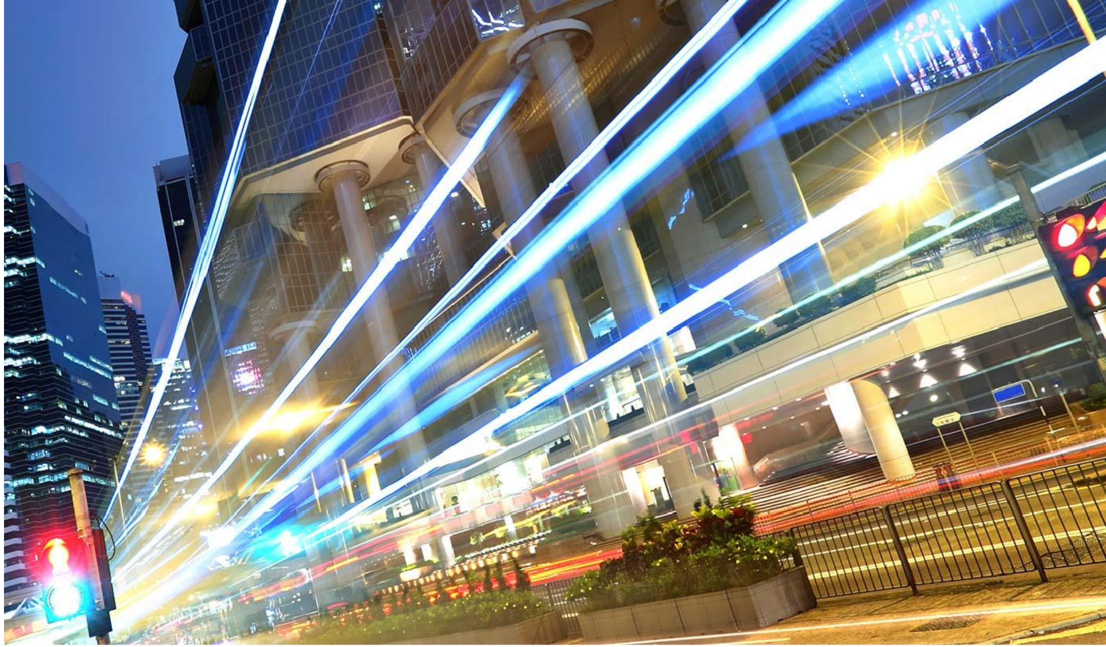


SIP Basics

SIP Basics

Session Initiation Protocol (SIP) is a text based signalling protocol for creating, modifying, and terminating sessions with one or more participants (SIP is described in RFC 3261)

SIP uses a Request/Response transaction model between endpoints (User Agents)

A User Agent (UA) – For example, a SIP Phone – performs two roles :

A User Agent Client (UAC) which sends SIP Requests

A User Agent Server (UAS) which receives SIP Requests and returns Responses

<u>SIP Servers</u> – Registrar, Redirect Server, Proxy Server,

Registrar – Provides location mapping for SIP User Agents

Redirect Server – Re-directs SIP Requests when a user has moved or is unavailable

Proxy Server – SIP message router – can be transaction stateful or stateless

SIP Proxy servers can either leave the signalling path when the call is connected or can enable "Record-route" to stay in the signalling path.

Proxies are designed to be mostly transparent to UAs (e.g. Can not terminate a call). Proxy servers can only change messages in specific and limited ways (e.g. Can not change call media info (e.g. codecs)). CUCM is not a Proxy server……

SIP Basics – CUCM/CUBE – Back to Back User Agent (B2BUA)

A B2BUA is similar to a stateful SIP Proxy Server in that it actively maintains call state, but does not have the same limitations as a SIP Proxy server. i.e. a B2BUA can disconnect a call and modify media information sent in the Session Description Protocol (SDP).

Why is CUCM a B2BUA ?

Because CUCM provides many other features beyond those of a SIP Proxy Server – Call Admission Control, Codec negotiation, interoperability with H323, MGCP and SIP, CTI etc.

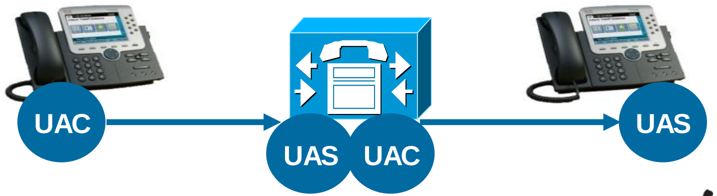
RFC 3621 does not define the specific functionality of a B2BUA….

How should a SIP call through CUCM be viewed ?

As two independent calls from a SIP perspective

An inbound call arriving at a User Agent and

An outbound call originating from another User Agent



SIP Basics – SIP Messages - Requests

INVITE	- Indicates a client is being invited to participate in a call session
ACK	- Confirms that the client has received a final response to an INVITE request
BYE	- Terminates a call and can be sent by either the caller or the callee
CANCEL	- Cancels any pending request
OPTIONS	- Queries the capabilities of servers (OPTIONS Ping)
REGISTER	- Registers the address in the To header field with a SIP server (Phones only)
PRACK	- Provisional acknowledgement
SUBSCRIBE	- Subscribes for an Event of Notification from the Notifier (Used for BLF)
NOTIFY	- Notify the subscriber of a new Event (Used for KPML and MWI)
PUBLISH	- Publishes an event to the Server
INFO	- Sends mid-session information that does not modify the session state
UPDATE	- Modifies the state of a session before a final response is received
MESSAGE	- Transports instant messages using SIP (XMPP also used for IM)
REFER	- Asks recipient to issue a SIP Request (Call Transfer)

SIP Basics – SIP Messages - Responses

Response Categories

Provisional	(1xx):	Request received and being processed (Unreliable – not ACK'ed)
Success	(2xx):	The action was successfully received, understood, and accepted.
Redirection	(3xx):	Further action needs to be taken to complete the request.
Client Error	(4xx):	The request contains bad syntax or cannot be fulfilled at the server.
Server Error	(5xx):	The server failed to fulfil an apparently valid request.
Global Failure	(6xx):	The request cannot be fulfilled at any server.

Commonly Used Responses

100 Trying - CUCM has received the INVITE
180 Ringing - Destination user agent has received the INVITE, and is alerting the user
183 Session in Progress - Used to send extra information for a call which is still being set up
200 OK - Indicates the request was successful
401 Unauthorised
404 Not Found - The server has definitive information that the user does not exist
503 Service Unavailable

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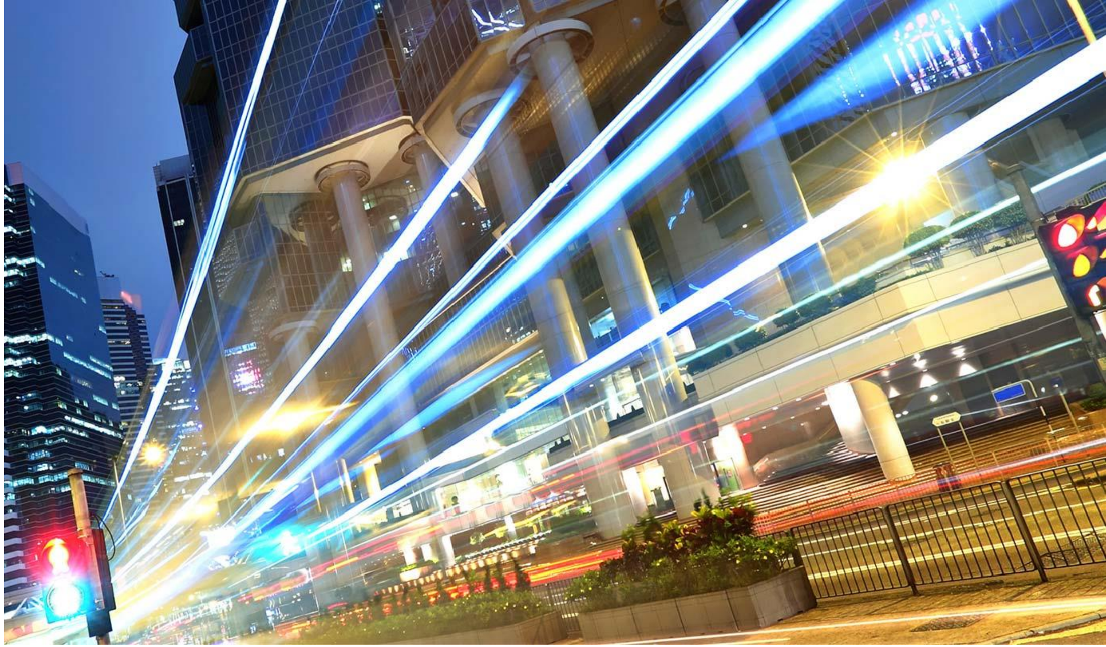
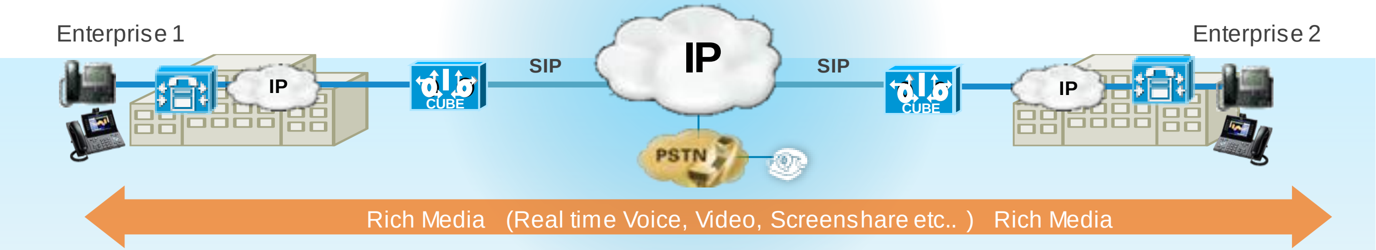
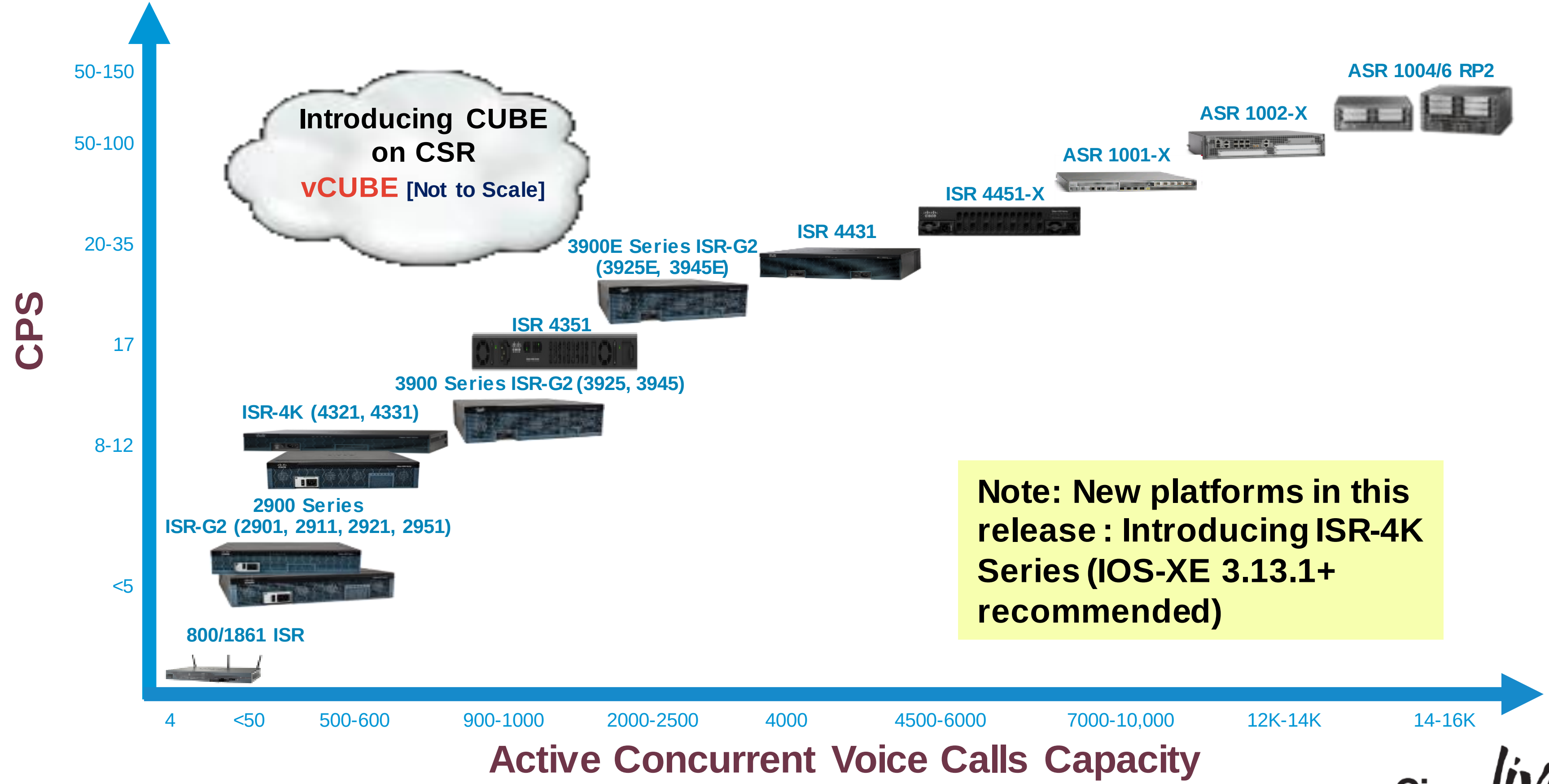


Why Does an Enterprise Need an SBC ?



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

CUBE (Enterprise) Product Portfolio



Note: New platforms in this release : Introducing ISR-4K Series (IOS-XE 3.13.1+ recommended)

CUBE Session Capacity Summary

For Your Reference

Platform	CUBE Sessions
NanoCUBE (8XX and SPIAD Platforms)	15 - 120
2901 – 4321	100
2911 – 2921	200 – 400
4331	500
2951	600
3925 – 3945	800 – 950
4351	1000
3925E – 3945E	2100 – 2500
4431	3000
4451	6000
ASR1001-X	12000
ASR1002-X	14000
ASR1004/1006 RP2	16000

CUBE Software Release Mapping

	ISR G2		CUBE Ent ASR Parity with ISR	ASR / ISR4400*			
CUBE Vers.	2900/ 3900	FCS		CUBE Vers.	IOS XE Release		FCS
8.9	15.2.2T	Nov 2011	>80%	1.4.4	3.6	15.2(2)S	Mar 2012
9.0	15.2.3T/ 15.2.4M	Mar 2012	>85%	9.0	3.7	15.2(4)S	July 2012
9.0.1	15.3.1T	Oct 2012	>95%	9.0.1	3.8	15.3(1)S	Oct 2012
9.0.2	15.3(2)T	Mar 2013	>95%	9.0.2	3.9	15.3(2)S	Mar 2013
9.5.1	15.3(3)M1	Oct 2013	>95%	9.5.1	3.10.1	15.3(3)S1	Oct 2013
10.0.0	15.4(1)T	Nov 2013	>95%	10.0.0	3.11	15.4(1)S	Nov 2013
10.0.1	15.4(2)T	Mar 2014	>95%	10.0.1	3.12	15.4(2)S	Mar 2014
10.0.2	15.4(3)M	July 2014	>95%	10.0.2	3.13	15.4(3)S	July 2014
10.5.0	15.5(1)T	Nov 2014	>95%	10.5.0	3.14	15.5(1)S	Nov 2014
11.0.0	15.5(2)T	Mar 2015	>95%	11.0.0	3.15	15.5(2)S	Mar 2015
11.0.1	15.5(3)M	July 205	>95%	11.0.1	3.16	15.5(3)S	July 2015

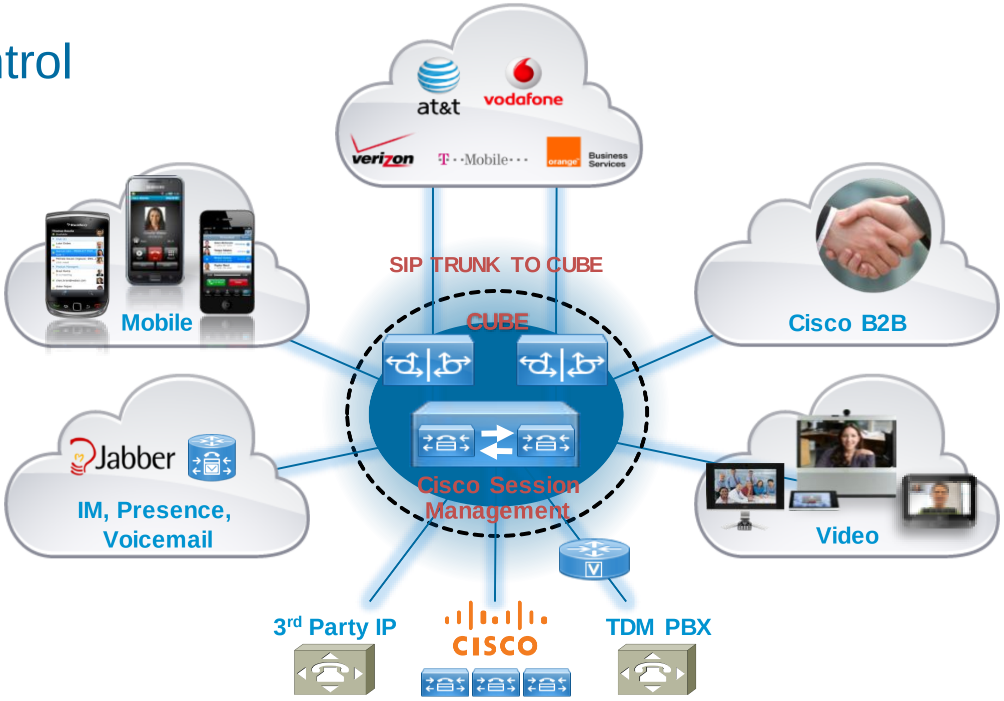
* IOS-XE3.13.1 or later recommended for all ISR 4400 series

SIP Trunking Design and Deployment Models

Cisco Session Management & CUBE:

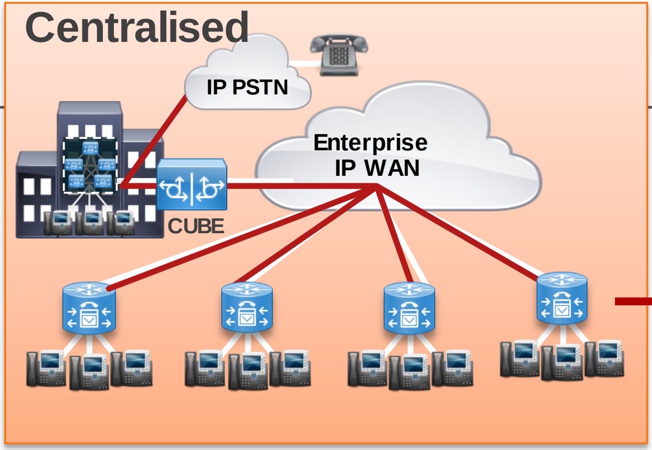
Essential Elements for Collaboration

- CUBE provides session border control between IP networks
  - Demarcation
  - Interworking
  - Session control
  - Security
- Cisco SME centralises network control
  - Centralises dial plan
  - Centralised applications
  - Aggregates PBXs



The Centralised Model

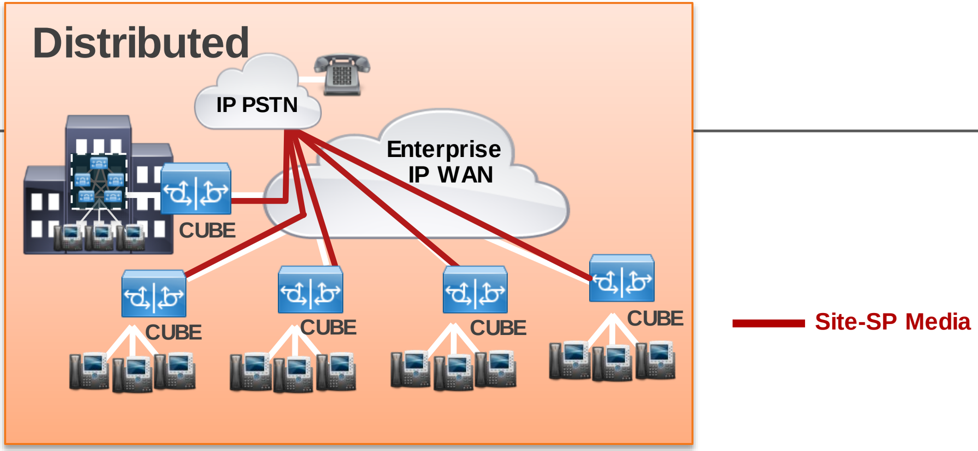
Characteristics of Centralised	Operational Benefits	Challenges
• Central Site is the only location with SIP session connectivity to IP PSTN • Voice services delivered to Branch Offices over the Enterprise IP WAN (usually MPLS) • Media traffic hairpins through central site between SP and branches	• Centralises Physical Operations • Centralises Dial-Peer Management • Centralises SIP Trunk Capacity	• Increased campus bandwidth, CAC, latency; media optimisation • HA in campus • Survivability at branch (PSTN connection at the branch) • Emergency services • Legal/Regulatory



Site-SP Media

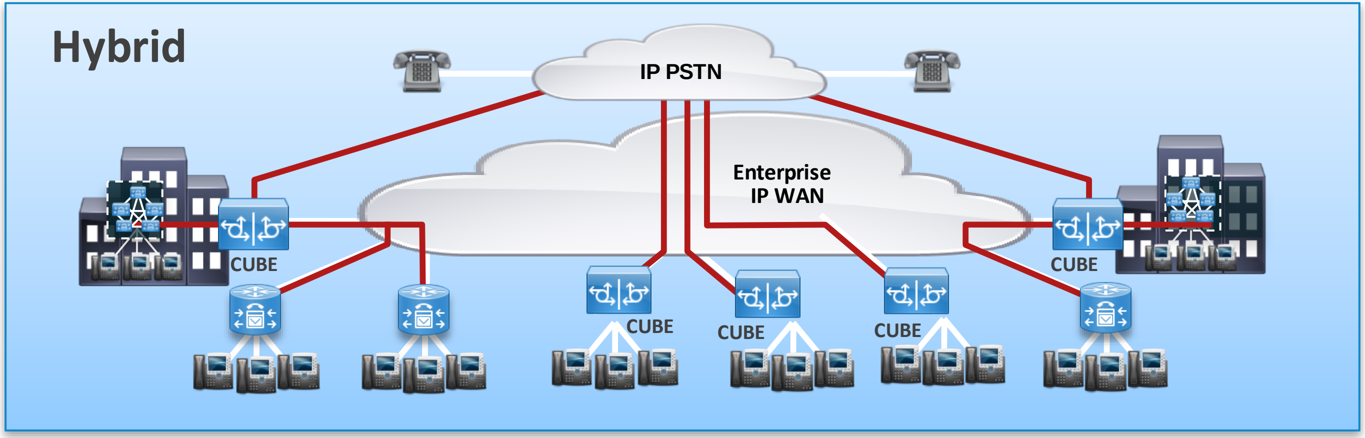
The Distributed Model

Characteristics of Distributed	Operational Benefits	Challenges
• Each site has direct connection for SIP sessions to SP • Takes advantage of SP session pooling, if offered by SP • Media traffic goes direct from each branch site to the SP	• Leverages existing branch routers • No media hair-pinning thru any site • Lower latency on voice or video	• Distributed dial-peer management • Distributed operational overhead • IP addressing to Service Provider from branch



.. and The Hybrid Model

Characteristics of Hybrid	Benefits
• Connection to SP SIP service is determined on a site by site basis to be either direct or routed through a regional site. • Decision to route call direct or indirect based on various criteria • Media traffic goes direct from site to SP or hairpins through another site, depending on branch configuration.	• Adaptable to site specific requirements • Optimises BW use on Enterprise WAN • Adaptable to regional SP issues • Built-in redundancy strategy



CUBE Call Flow

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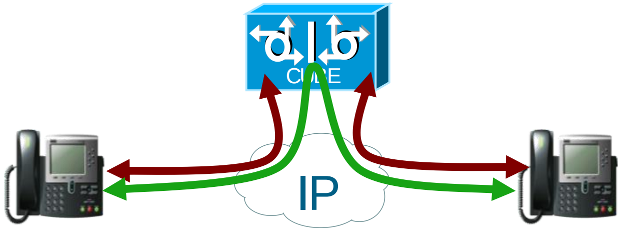
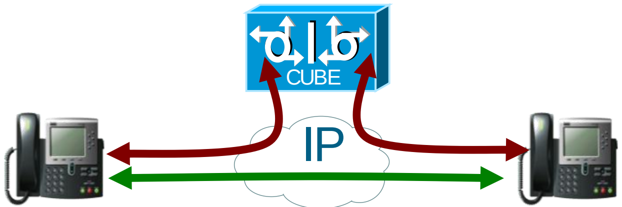
CUBE Call Processing

- 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

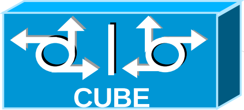
- Signalling and media terminated by the Cisco Unified Border Element
- Media bypasses the Cisco Unified Border Element

Cisco Unified Border Element Basic Call Flow

```
voice service VoIP
  mode border-element
  allow-connections h323 to h323
  allow-connections h323 to sip
  allow-connections sip to h323
  allow-connections sip to sip
```

Originating Endpoint - 1000

Incoming VoIP Call



Outgoing VoIP Call

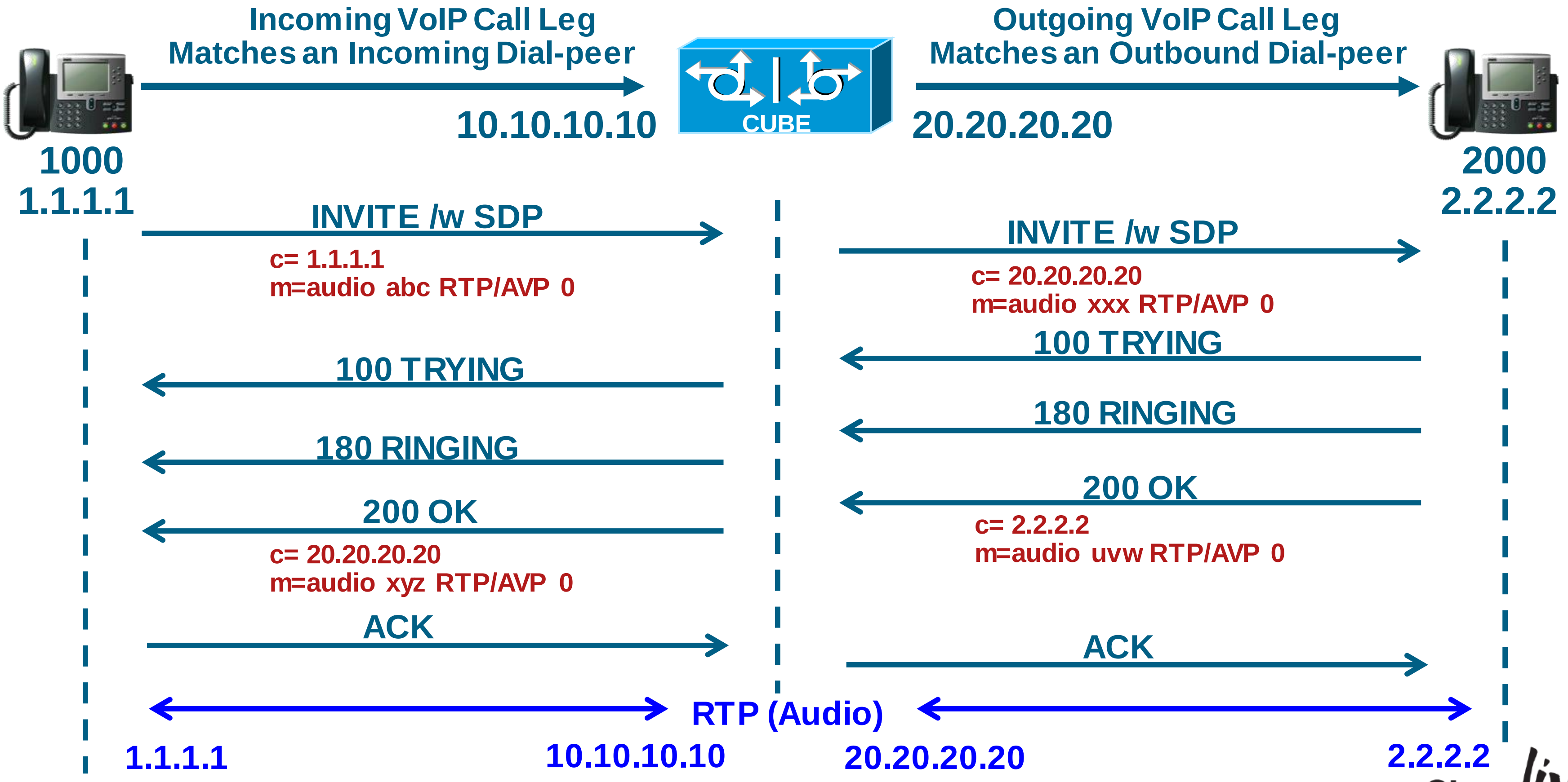
Terminating Endpoint – 2000

```
dial-peer voice 1 VoIP
  incoming called-number 1000
  session protocol sipv2
  codec g711ulaw
  no vad
```

```
dial-peer voice 2 VoIP
  destination-pattern 2000
  session protocol sipv2
  session target ipv4:2.2.2.10
  codec g711ulaw
```

1. Incoming VoIP setup message from originating endpoint
2. This matches inbound VoIP dial peer 1 for characteristics such as codec, VAD, DTMF method, protocol, etc.
3. Match the called number to outbound VoIP dial peer 2
4. Outgoing VoIP setup message

Understanding The CUBE Call Flow



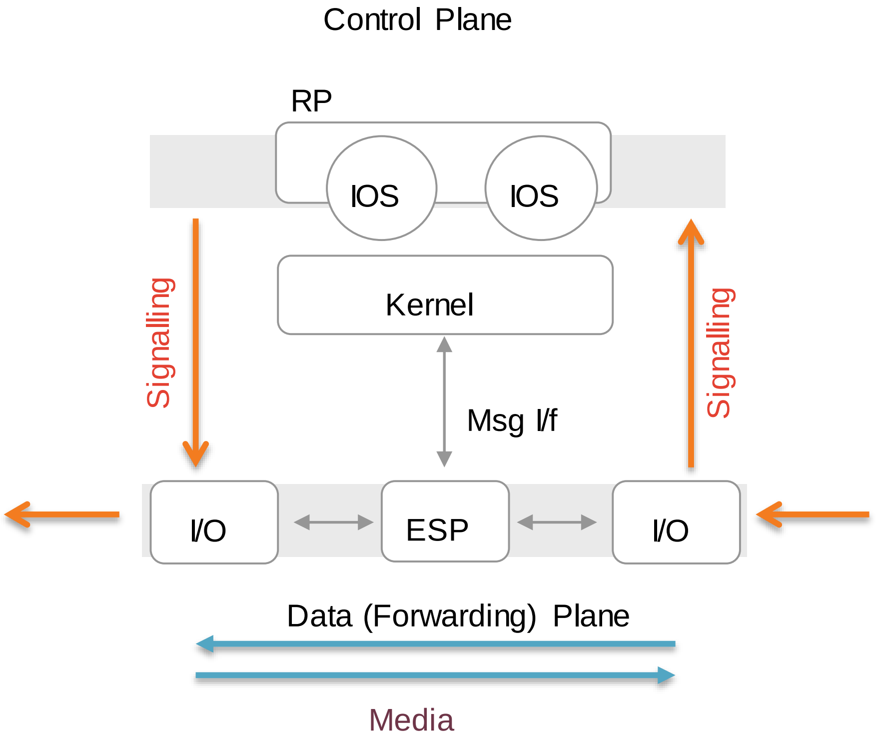
Basic Show Commands for Active Calls

```
CUBE# show call active voice brief

121A : 17 13:02:24.215 IST Mon Jun 27 2011.1 +2040 pid:2 Answer 2000 active
 dur 00:00:14 tx:0/0 rx:0/0
 IP 2.2.2.2:6001 SRTP: off rtt:0ms pl:0/0ms lost:0/0/0 delay:0/0/0ms g729r8 TextRelay: off
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a
 long duration call detected:n long duration call duration:n/a timestamp:n/a

121A : 18 13:02:24.225 IST Mon Jun 27 2011.1 +2020 pid:1 Originate 1000 active
 dur 00:00:14 tx:0/0 rx:0/0
 IP 1.1.1.1:6000 SRTP: off rtt:0ms pl:0/0ms lost:0/0/0 delay:0/0/0ms g729r8 TextRelay: off
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a
 long duration call detected:n long duration call duration:n/a timestamp:n/a

Telephony call-legs: 0
SIP call-legs: 2
H323 call-legs: 0
Call agent controlled call-legs: 0
SCCP call-legs: 0
Multicast call-legs: 0
Total call-legs: 2

CUBE# show VoIP rtp connections

VoIP RTP active connections :
No. CallId     dstCallId  LocalRTP  RmtRTP      LocalIP       RemoteIP
1     17         18          17474    6000     10.10.10.10     1.1.1.1
2     18         17          17476    6001     20.20.20.20     2.2.2.2

Found 2 active RTP connections
```

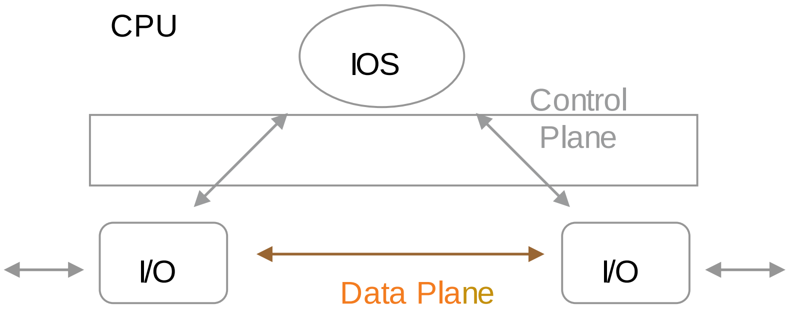
CUBE Architecture
ISR G2 vs ASR1K vs ISR 4400

ASR/4400 & ISR-G2 Architecture Comparison

ASR/4400 Architecture



ISR G2 Architecture



- ISR: Pkt fwd’ing and signalling are handled by the same CPU
- ASR: Pkt fwd’ing and signalling are handled by different CPUs
  - ESP must be programmed or instructed by the control plane to do specific media functions
  - Performed by Forwarding Plane Interface (FPI)

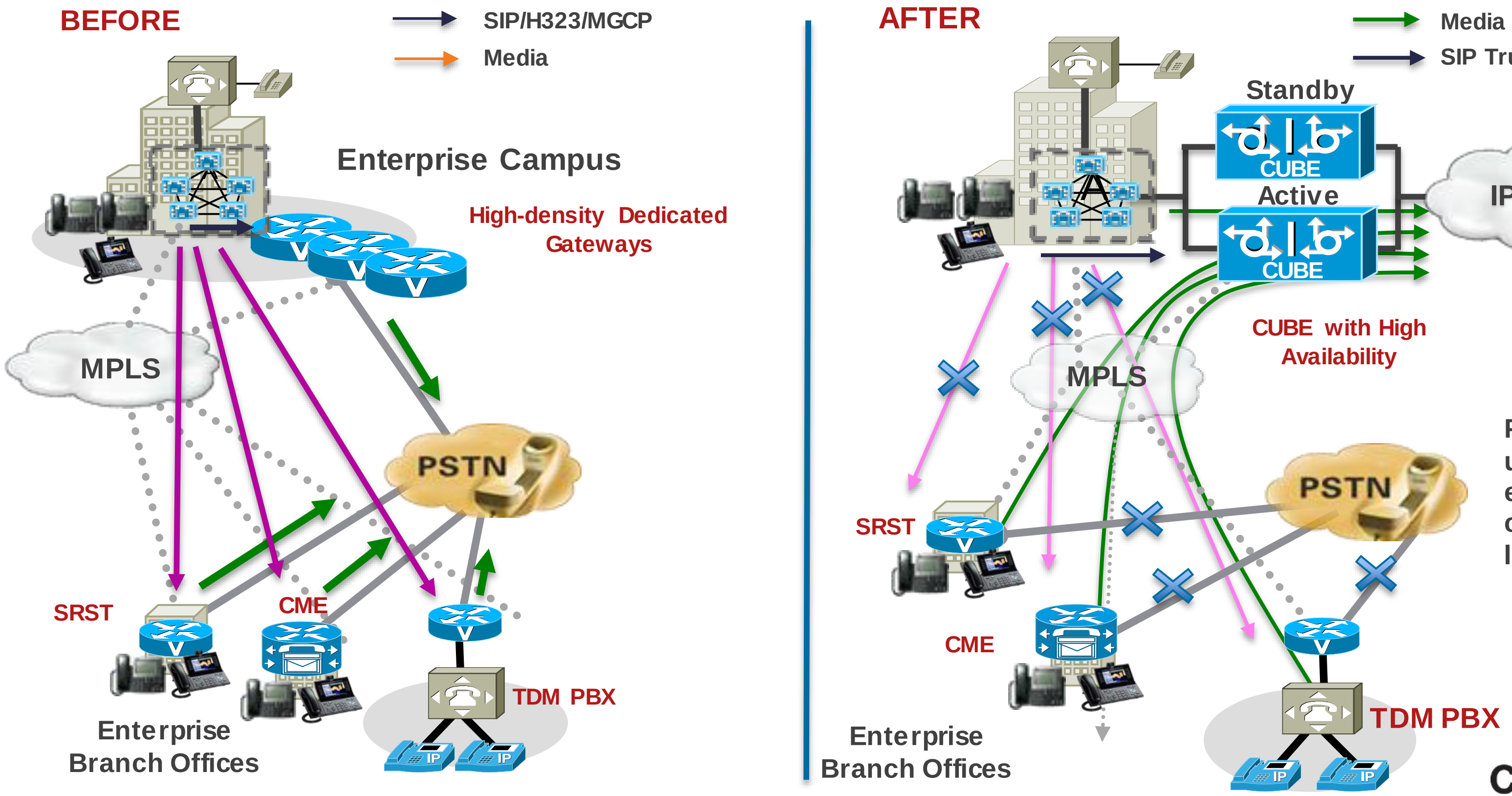
ASR & ISR-G2/4400 Feature Comparison

General SBC Features	ASR	ISR-G2	4400
High Availability Implementation	Redundancy-Group Infrastructure	HSRP Based	Redundancy-Group Infrastructure
TDM Trunk Failover/Co-existence	Not Available	Exists	Exists
Media Forking	XE3.8 (Thousands of calls)	15.2.1T (Upto 1250 calls)	XE3.10
Software MTP registered to CUCM (Including HA Support)	XE3.6	Exists	Exists
DSP Card	SPA-DSP	PVDM2/PVDM3	PVDM4
Transcoder registered to CUCM	Not Available	Exists via SCCP	Exists via SCCP **(XE3.11)**
Transcoder Implementation	Local Transcoder Interface (LTI)	SCCP or LTI (starting IOS 15.2.3T)	SCCP and LTI
Embedded Packet Capture	Exists	Exists	Exists
Web-based UC API	XE3.8	15.2.2T	Exists
Noise Reduction & ASP	Exists	15.2.3T	Exists
Call Progress Analysis	XE3.9	15.3.2T	Exists
CME/SRST and CUBE co-existence	Not Available	Exists	XE3.11
SRTP-RTP Call flows	Exists (NO DSPs needed)	Exists (DSPs required)	Exists (NO DSPs needed)

Transitioning to SIP Trunking Using CUBE

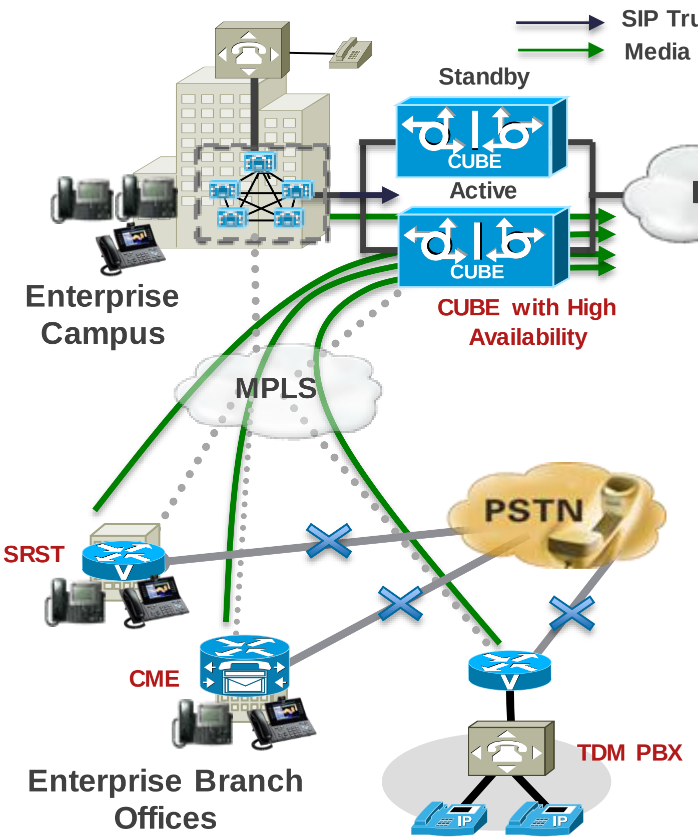
Transitioning to SIP Trunking...

Re-purpose your existing Cisco voice gateway's as Session Border Controllers



PSTN is now used only for emergency calls over FXO lines

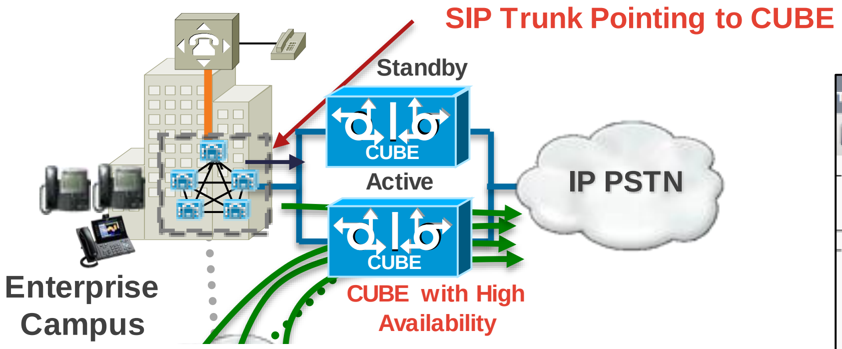
Steps to Transitioning...

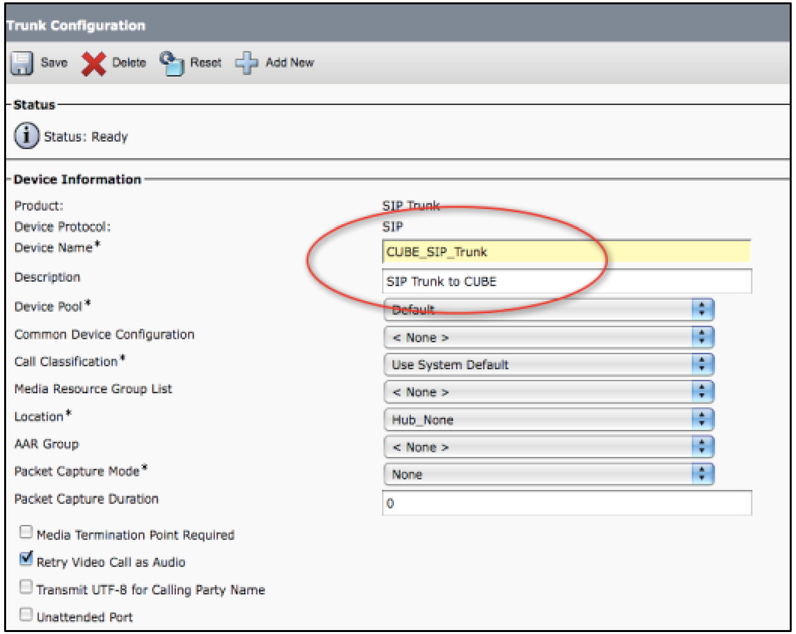


PSTN is now used only for emergency calls over FXO lines

- Step 1 – Configure IP PBX to route all calls (HQ and branch offices) to the edge SBC
- Step 2 – Get SIP Trunk details from the provider
- Step 3 – Enable CUBE application on Cisco routers
- Step 4 – Configure call routing on CUBE (Incoming & Outgoing dial-peers)
- Step 5 – Normalise SIP messages to meet SIP Trunk provider's requirements
- Step 6 – Execute the test plan

Step 1: Configure CUCM to Route Calls to the Edge SBC





- Configure CUCM to route all PSTN calls (central and branch) to CUBE via a SIP trunk
- Make sure all different patterns of calls – local, long distance, international, emergency, informational etc.. are pointing to CUBE

Step 2: Get Details from SIP Trunk Provider

Item	SIP Trunk service provider requirement	Sample Response
1	SIP Trunk IP Address (Destination IP Address for INVITES)	20.1.1.2 or DNS
2	SIP Trunk Port number (Destination port number for INVITES)	5060
3	SIP Trunk Transport Layer (UDP or TCP)	UDP
4	Codecs supported	G711, G729
5	Fax protocol support	T.38
6	DTMF signalling mechanism	RFC2833
7	Does the provider require SDP information in initial INVITE (Early offer required)	Yes
8	SBC's external IP address that is required for the SP to accept/authenticate calls (Source IP Address for INVITES)	20.1.1.1
9	Does SP require SIP Trunk registration for each DID? If yes, what is the username & password	No
10	Does SP require Digest Authentication? If yes, what is the username & password	No

Step 3: Enable CUBE Application on Cisco Routers

1. Enable CUBE Application

```
voice service VoIP
  mode border-element license capacity 200
  allow-connections sip to sip
```

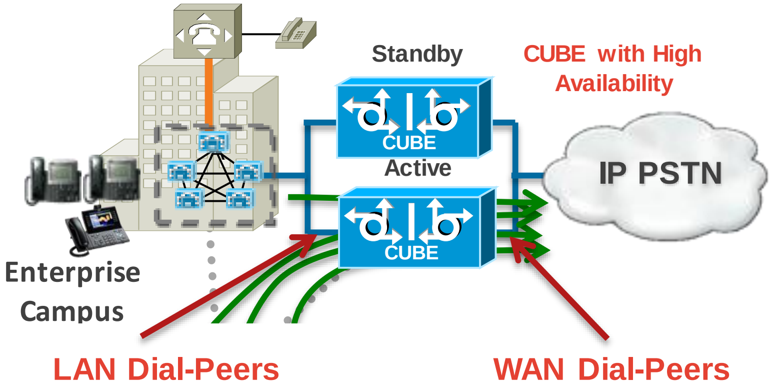
2. Configure any other global settings to meet SP's requirements

```
voice service VoIP
  sip
    early-offer forced
      header-passing
      error-passthru
```

3. Create a trusted list of IP addresses to prevent toll-fraud

```
voice service VoIP
 ip address trusted list
  ipv4 10.1.1.50
  ipv4 20.20.20.20
 sip
  silent discard-untrusted  → Default configuration starting XE 3.10.1 /15.3(3)M1 to
                               mitigate TDoS Attack
```

Step 4: Configure Call Routing on CUBE



- Dial-Peer – "static routing" table mapping phone numbers to interfaces or IP addresses
- LAN Dial-Peers – Dial-peers that are facing towards the IP PBX for sending and receiving calls to & from the PBX
- WAN Dial-Peers – Dial-peers that are facing towards the SIP Trunk provider for sending & receiving calls to & from the provider

WAN Dial-Peer Configuration

Inbound Dial-Peer for calls from SP to CUBE

```
dial-peer voice 100 VoIP
    description *** Inbound WAN side dial-peer ***
    incoming called-number [2-9].........
    session protocol sipv2
    codec g711ulaw
    dtmf-relay rtp-nte
```

Catch-all for all inbound PSTN calls

Outbound Dial-Peer for calls from CUBE to SP

```
dial-peer voice 200 VoIP
    description *** Outbound WAN side dial-peer ***
    translation-profile outgoing Digitstrip
    destination-pattern 9[2-9].........
    session protocol sipv2
    voice-class sip bind control source gig0/1
    voice-class sip bind media source gig0/1
    session target ipv4:<SIP_Trunk_IP_Address>
    codec g711ulaw
    dtmf-relay rtp-nte
```

Dial-peer for making long distance calls to SP

Note: Separate outgoing DP to be created for Local, International, Emergency, Informational calls etc.

LAN Dial-Peer Configuration

Inbound Dial-Peer for calls from CUCM to CUBE

```
dial-peer voice 100 VoIP
   description *** Inbound LAN side dial-peer ***
   incoming called-number 9T
   session protocol sipv2
   codec g711ulaw
   dtmf-relay rtp-nte
```

CUCM sending 9 + All digits dialed

Outbound Dial-Peer for calls from CUBE to CUCM

```
dial-peer voice 200 VoIP
   description *** Outbound LAN side dial-peer ***
   destination-pattern [2-9].........
   session protocol sipv2
   session target ipv4:<CUCM_Address>
   codec g711ulaw
   dtmf-relay rtp-nte
```

SP will be sending 10 digits inbound

Note: If more than 1 CUCM cluster exists, you will have to create multiple such LAN dial-peers with “preference CLI” for CUCM redundancy/load balancing as the traditional way to accommodate multiple trunks

Step 5: SIP Normalisation

SIP profiles is a mechanism to normalise or customise SIP at the network border to provide interop between incompatible devices

SIP Profile testing Tool now available
http://www.cisco.com/web/tsweb/tools/sip-profile/index.html

SIP incompatibilities arise due to:

- A device rejecting an unknown header (value or parameter) instead of ignoring it
- A device expecting an optional header value/parameter or can be implemented in multiple ways
- A device sending a value/parameter that must be changed or suppressed ("normalised") before it leaves/enters the enterprise to comply with policies
- Variations in the SIP standards of how to achieve certain functions
- **With CUBE 10.0.1 SIP Profiles can be applied to inbound SIP messages as well**

Add user=phone for INVITEs

Incoming → CUBE → Outgoing

Incoming	Outgoing
INVITE sip:5551000@sip.com:5060 SIP/2.0	INVITE sip:5551000@sip.com:5060 user=phone SIP/2.0

```
voice class sip-profiles 100
  request INVITE sip-header SIP-Req-URI modify "; SIP/2.0" ";user=phone SIP/2.0"
  request REINVITE sip-header SIP-Req-URI modify "; SIP/2.0" ";user=phone SIP/2.0"
```

Modify a "sip:" URI to a "tel:" URI in INVITEs

Incoming → CUBE → Outgoing

Incoming	Outgoing
INVITE sip:2222000020@9.13.24.6:5060 SIP/2.0	INVITE tel:2222000020 SIP/2.0

```
voice class sip-profiles 100
  request INVITE sip-header SIP-Req-URI modify "sip:(.*)@[^ ]+" "tel:\1"
  request INVITE sip-header From modify "<sip:(.*)@.*>" "<tel:\1>"
  request INVITE sip-header To modify "<sip:(.*)@.*>" "<tel:\1>"
```

Normalise Inbound SIP Message (Example 1)

For Your Reference

15.4.2T/ XE3.12

CUBE Requirement

SIP Diversion header must include a user portion

SIP INVITE received by CUBE

```
Sent:
INVITE  sip:2000@9.44.44.4:5060 SIP/2.0
.........
User-Agent: SP-SBC
.........
Diversion:  <sip:9.44.44.4>;privacy=off;
reason=unconditional;screen=yes
……...
m=audio 6001 RTP/AVP  0 8 18 101
a=rtpmap:0 PCMU/8000
……...
```

SIP INVITE CUBE expects

```
Sent:
INVITE  sip:2000@9.44.44.4:5060 SIP/2.0
..........
User-Agent: SP-SBC
..........
Diversion:  <sip:1234@abc.com>;
            privacy=off;reason=unconditional;screen=yes
……….
m=audio 32278 RTP/AVP  18 8 101
a=rtpmap:0 PCMU/8000
………..
```

Enable Inbound SIP Profile feature

```
voice service VoIP
  sip
    sip-profiles inbound
```

Configure Inbound SIP Profile to add a dummy user part

```
voice class sip-profiles 400
  request INVITE sip-header Diversion modify "sip:" sip:1234@
```

Apply to Dial-peer or Globally

```
dial-peer voice 4000 VoIP
 description Incoming/outgoing SP
 voice-class sip profiles 400 inbound
```

```
voice service VoIP
  sip
    sip profiles 400 inbound
```

Step 6: Execute the Test Plan

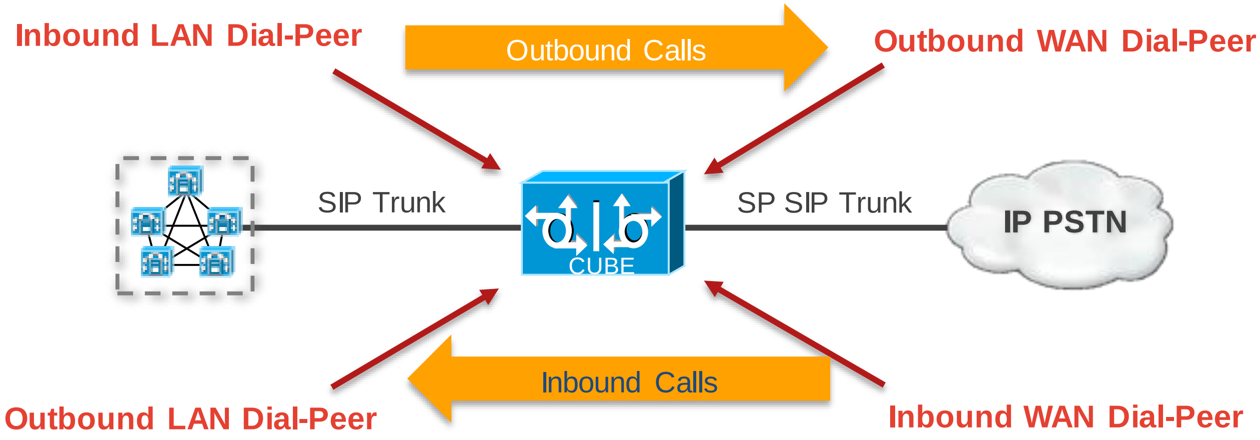
- Inbound and outbound Local, Long distance, International calls for G711 & G729 codecs (if supported by provider)
- Outbound calls to information and emergency services
- Caller ID and Calling Name Presentation
- Supplementary services like Call Hold, Resume, Call Forward & Transfer
- DTMF Tests
- Fax calls – T.38 and fallback to pass-through (if option available)

CUBE Dial-Peers Call Routing

Understanding Dial-Peer Matching Techniques:

LAN & WAN Dial-Peers

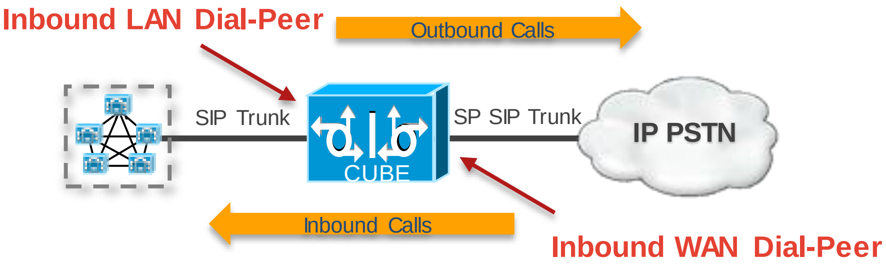
- LAN Dial-Peers – Dial-peers that are facing towards the IP PBX for sending and receiving calls to & from the PBX
- WAN Dial-Peers – Dial-peers that are facing towards the SIP Trunk provider for sending & receiving calls to & from the provider



Understanding Inbound Dial-Peer Matching Techniques

Priority

1. **Match Based on URI of an incoming INVITE message**
   - Exact Pattern match
   - Host Name/IP Address
   - User portion of URI
   - Phone-number of tel-uri
2. **Match based on Called Number**
3. **Match based on Calling number**
4. **Default Dial-Peer = 0**

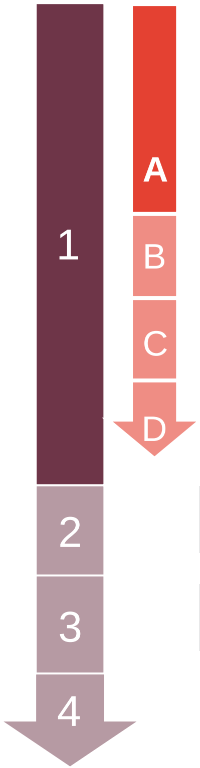


Received:

```
INVITE sip:654321@10.2.1.1 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.1:5060;x-route-
tag="cid:orange@10.1.1.1";;branch=z9hG4bK-23955-1-0
From: "555" <sip:555@10.1.1.1:5060>;tag=1
To: ABC <sip:654321@10.2.1.1:5060>
Call-ID: 1-23955@10.1.1.1
CSeq: 1 INVITE
Contact: sip:555@10.1.1.1:5060
Supported: timer
Max-Forwards: 70
Subject: BRKUCC-2934 Session
Content-Type: application/sdp
Content-Length:   226
........
```

Understanding Inbound Dial-Peer Matching Techniques

Priority



voice class uri 1001 sip
  host ipv4:10.1.1.1

voice class uri 2001 sip
  host ipv4:10.2.1.1

dial-peer voice 1 VoIP
  incoming uri via 1001

dial-peer voice 2 VoIP
  incoming uri request 2001

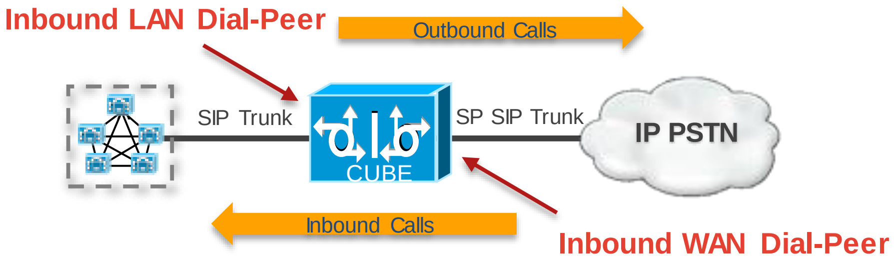
dial-peer voice 3 VoIP
  incoming uri to 2001

dial-peer voice 4 VoIP
  incoming uri from 1001

dial-peer voice 5 VoIP
  incoming called-number 654321

dial-peer voice 6 VoIP
  answer-address 555

dial-peer voice 7 VoIP
  destination-pattern 555



Received:
INVITE sip:654321@10.2.1.1 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.1:5060;x-route-tag="cid:orange@10.1.1.1";;branch=z9hG4bK-23955-1-0
From: "555" <sip:555@10.1.1.1:5060>;tag=1
To: ABC <sip:654321@10.2.1.1:5060>
Call-ID: 1-23955@10.1.1.1
CSeq: 1 INVITE
Contact: sip:555@10.1.1.1:5060
Supported: timer
Max-Forwards: 70
Subject: BRKUCC-2934 Session
Content-Type: application/sdp
Content-Length: 226
....

Cisco live!

Understanding Inbound Dial-Peer Matching Techniques

Priority

1 — A, **B**, C, D

2

3

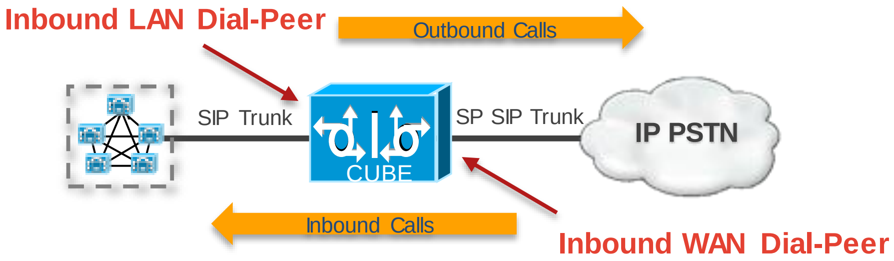
4

```
voice class uri 1001 sip
  host ipv4:10.1.1.1

voice class uri 2001 sip
  host ipv4:10.2.1.1

dial-peer voice 1 VoIP
  incoming uri via 1001

dial-peer voice 2 VoIP
  incoming uri request 2001

dial-peer voice 3 VoIP
  incoming uri to 2001

dial-peer voice 4 VoIP
  incoming uri from 1001
```

```
dial-peer voice 5 VoIP
  incoming called-number 654321
```

```
dial-peer voice 6 VoIP
  answer-address 555
```

```
dial-peer voice 7 VoIP
  destination-pattern 555
```



Received:

```
INVITE sip:654321@10.2.1.1 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.1:5060;x-route-tag="cid:orange@10.1.1.1";;branch=z9hG4bK-23955-1-0
From: "555" <sip:555@10.1.1.1:5060>;tag=1
To: ABC <sip:654321@10.2.1.1:5060>
Call-ID: 1-23955@10.1.1.1
CSeq: 1 INVITE
Contact: sip:555@10.1.1.1:5060
Supported: timer
Max-Forwards: 70
Subject: BRKUCC-2934 Session
Content-Type: application/sdp
Content-Length:   226
........
```

Cisco live!

Understanding Inbound Dial-Peer Matching Techniques

Priority

1 — A, B, C, D

2

3

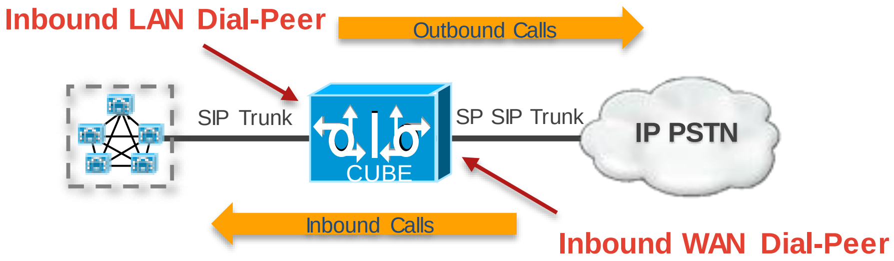
4

```
voice class uri 1001 sip
 host ipv4:10.1.1.1

voice class uri 2001 sip
 host ipv4:10.2.1.1

dial-peer voice 1 VoIP
 incoming uri via 1001

dial-peer voice 2 VoIP
 incoming uri request 2001

dial-peer voice 3 VoIP
 incoming uri to 2001

dial-peer voice 4 VoIP
 incoming uri from 1001
```

```
dial-peer voice 5 VoIP
 incoming called-number 654321
```

```
dial-peer voice 6 VoIP
 answer-address 555
```

```
dial-peer voice 7 VoIP
 destination-pattern 555
```



Received:

```
INVITE sip:654321@10.2.1.1 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.1:5060;x-route-
tag="cid:orange@10.1.1.1";;branch=z9hG4bK-23955-1-0
From: "555" <sip:555@10.1.1.1:5060>;tag=1
To: ABC <sip:654321@10.2.1.1:5060>
Call-ID: 1-23955@10.1.1.1
CSeq: 1 INVITE
Contact: sip:555@10.1.1.1:5060
Supported: timer
Max-Forwards: 70
Subject: BRKUCC-2934 Session
Content-Type: application/sdp
Content-Length:   226
........
```

Cisco live!

Understanding Inbound Dial-Peer Matching Techniques

Priority

1
- A
- B
- C
- **D**

2

3

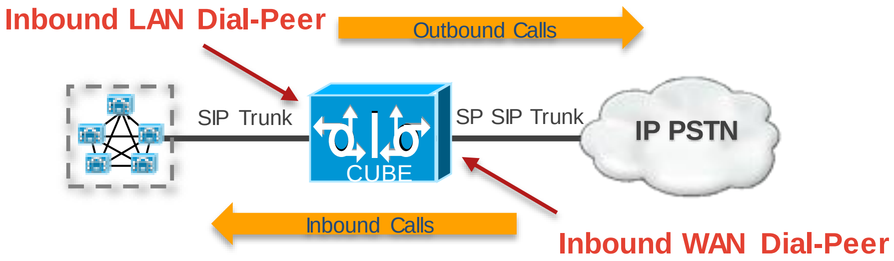
4

```
voice class uri 1001 sip
  host ipv4:10.1.1.1

voice class uri 2001 sip
  host ipv4:10.2.1.1

dial-peer voice 1 VoIP
  incoming uri via 1001

dial-peer voice 2 VoIP
  incoming uri request 2001

dial-peer voice 3 VoIP
  incoming uri to 2001

dial-peer voice 4 VoIP
  incoming uri from 1001
```

```
dial-peer voice 5 VoIP
  incoming called-number 654321
```

```
dial-peer voice 6 VoIP
  answer-address 555
```

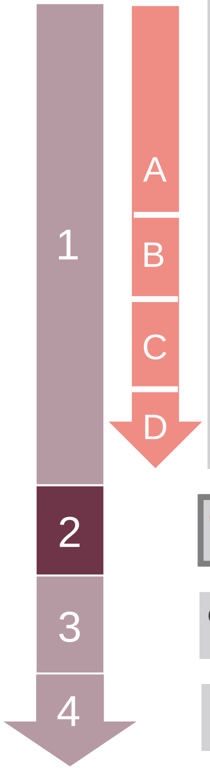
```
dial-peer voice 7 VoIP
  destination-pattern 555
```



Received:
INVITE sip:654321@10.2.1.1 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.1:5060;x-route-tag="cid:orange@10.1.1.1";;branch=z9hG4bK-23955-1-0
From: "555" <sip:555@10.1.1.1:5060>;tag=1
To: ABC <sip:654321@10.2.1.1:5060>
Call-ID: 1-23955@10.1.1.1
CSeq: 1 INVITE
Contact: sip:555@10.1.1.1:5060
Supported: timer
Max-Forwards: 70
Subject: BRKUCC-2934 Session
Content-Type: application/sdp
Content-Length: 226
....

Understanding Inbound Dial-Peer Matching Techniques

Priority

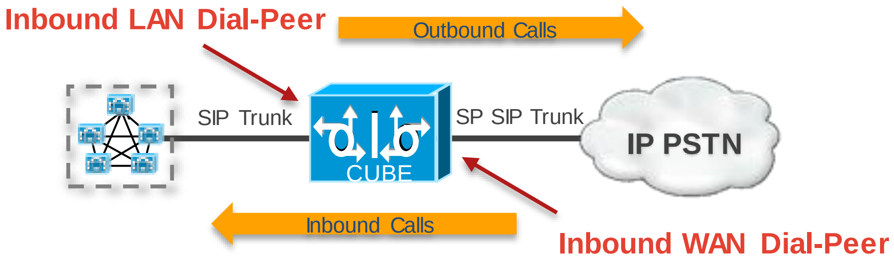



```
voice class uri 1001 sip
  host ipv4:10.1.1.1

voice class uri 2001 sip
  host ipv4:10.2.1.1

dial-peer voice 1 VoIP
  incoming uri via 1001

dial-peer voice 2 VoIP
  incoming uri request 2001

dial-peer voice 3 VoIP
  incoming uri to 2001

dial-peer voice 4 VoIP
  incoming uri from 1001
```

```
dial-peer voice 5 VoIP
  incoming called-number 654321
```

```
dial-peer voice 6 VoIP
  answer-address 555
```

```
dial-peer voice 7 VoIP
  destination-pattern 555
```

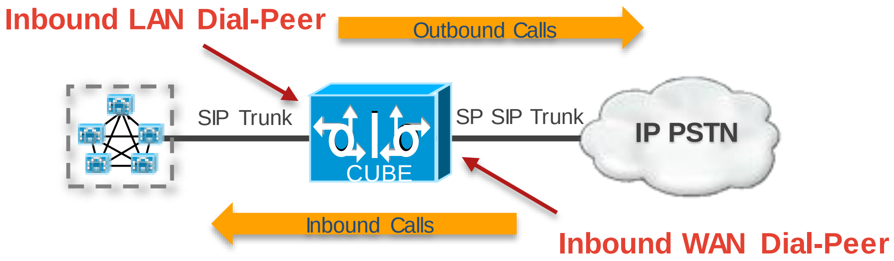


Received:
```
INVITE sip:654321@10.2.1.1 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.1:5060;x-route-
tag="cid:orange@10.1.1.1";;branch=z9hG4bK-23955-1-0
From: "555" <sip:555@10.1.1.1:5060>;tag=1
To: ABC <sip:654321@10.2.1.1:5060>
Call-ID: 1-23955@10.1.1.1
CSeq: 1 INVITE
Contact: sip:555@10.1.1.1:5060
Supported: timer
Max-Forwards: 70
Subject: BRKUCC-2934 Session
Content-Type: application/sdp
Content-Length:   226
........
```


Understanding Inbound Dial-Peer Matching Techniques

Priority

Priority	Order	Configuration
1		voice class uri 1001 sip host ipv4:10.1.1.1 voice class uri 2001 sip host ipv4:10.2.1.1
1	A	dial-peer voice 1 VoIP incoming uri via 1001
1	B	dial-peer voice 2 VoIP incoming uri request 2001
1	C	dial-peer voice 3 VoIP incoming uri to 2001
1	D	dial-peer voice 4 VoIP incoming uri from 1001
2		dial-peer voice 5 VoIP incoming called-number 654321
3		**dial-peer voice 6 VoIP answer-address 555**
4		dial-peer voice 7 VoIP destination-pattern 555




```
Received:
INVITE sip:654321@10.2.1.1 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.1:5060;x-route-
tag="cid:orange@10.1.1.1";;branch=z9hG4bK-23955-1-0
From: "555" <sip:555@10.1.1.1:5060>;tag=1
To: ABC <sip:654321@10.2.1.1:5060>
Call-ID: 1-23955@10.1.1.1
CSeq: 1 INVITE
Contact: sip:555@10.1.1.1:5060
Supported: timer
Max-Forwards: 70
Subject: BRKUCC-2934 Session
Content-Type: application/sdp
Content-Length:   226
........
```


Understanding Inbound Dial-Peer Matching Techniques

Priority

1 — A, B, C, D

```
voice class uri 1001 sip
  host ipv4:10.1.1.1

voice class uri 2001 sip
  host ipv4:10.2.1.1

dial-peer voice 1 VoIP
  incoming uri via 1001

dial-peer voice 2 VoIP
  incoming uri request 2001

dial-peer voice 3 VoIP
  incoming uri to 2001

dial-peer voice 4 VoIP
  incoming uri from 1001
```

2

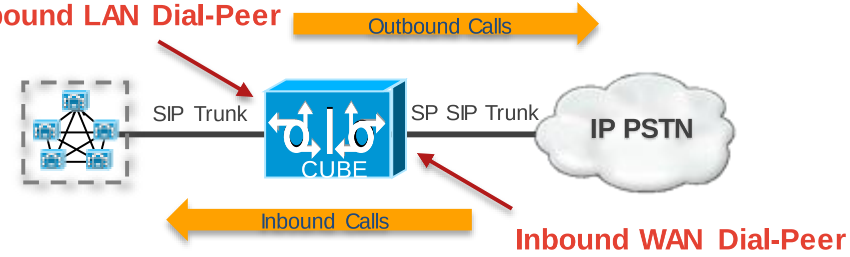
```
dial-peer voice 5 VoIP
  incoming called-number 654321
```

3

```
dial-peer voice 6 VoIP
  answer-address 555
```

4

```
dial-peer voice 7 VoIP
  destination-pattern 555
```

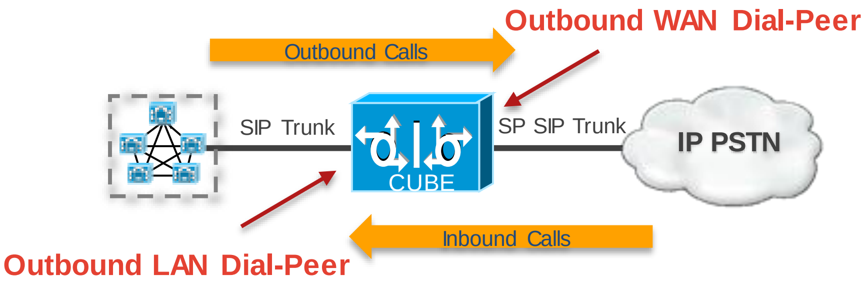


Received:
INVITE sip:654321@10.2.1.1 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.1:5060;x-route-tag="cid:orange@10.1.1.1";;branch=z9hG4bK-23955-1-0
From: "555" <sip:**555**@10.1.1.1:5060>;tag=1
To: ABC <sip:654321@10.2.1.1:5060>
Call-ID: 1-23955@10.1.1.1
CSeq: 1 INVITE
Contact: sip:555@10.1.1.1:5060
Supported: timer
Max-Forwards: 70
Subject: BRKUCC-2934 Session
Content-Type: application/sdp
Content-Length: 226
....

Understanding Outbound Dial-Peer Matching Techniques

Priority

1. **Match Based on URI of incoming INVITE message & carrier-id target**
   - Exact Pattern match
   - Host Name/IP Address
   - User portion of URI
   - Phone-number of tel-uri
2. **Match based on Called Number & carrier-id target**
3. **Match based on URI of an incoming INVITE message**
   - Exact Pattern match
   - Host Name/IP Address
   - User portion of URI
   - Phone-number of tel-uri
4. **Match based on Called number**



Received:
INVITE sip:654321@10.2.1.1 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.1:5060;x-route-tag="cid:orange@10.1.1.1";;branch=z9hG4bK-23955-1-0
From: "555" <sip:555@10.1.1.1:5060>;tag=1
To: ABC <sip:654321@10.2.1.1:5060>
Call-ID: 1-23955@10.1.1.1
CSeq: 1 INVITE
Contact: sip:555@10.1.1.1:5060
Supported: timer
Max-Forwards: 70
Subject: BRKUCC-2934 Session
Content-Type: application/sdp
Content-Length: 226
....

Understanding Outbound Dial-Peer Matching Techniques

Priority

1

```
voice class uri 2001 sip
  host ipv4:10.2.1.1

dial-peer voice 1 VoIP
  destination uri 2001
  carrier-id target orange
```

2

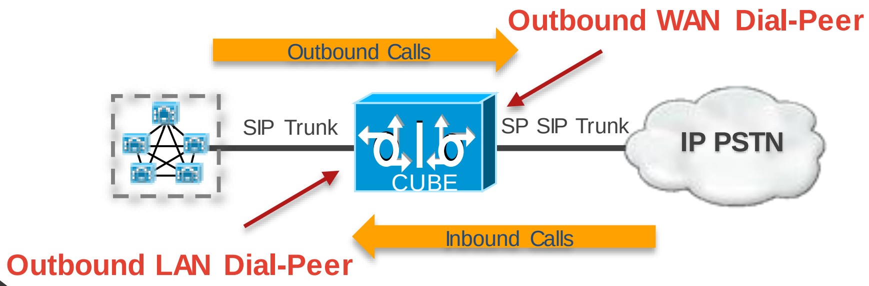
```
dial-peer voice 2 VoIP
  destination-pattern 654321
  carrier-id target orange
```

3

```
voice class uri 2001 sip
  host ipv4:10.2.1.1

dial-peer voice 3 VoIP
 destination uri 2001
```

4

```
dial-peer voice 4 VoIP
  destination-pattern 654321
```




```
Received:
INVITE sip:654321@10.2.1.1 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.1:5060;x-route-
tag="cid:orange@10.1.1.1";branch=z9hG4bK-23955-1-0
From: "555" <sip:555@10.1.1.1:5060>;tag=1
To: ABC <sip:654321@10.2.1.1:5060>
Call-ID: 1-23955@10.1.1.1
CSeq: 1 INVITE
Contact: sip:555@10.1.1.1:5060
Supported: timer
Max-Forwards: 70
Subject: BRKUCC-2934 Session
Content-Type: application/sdp
Content-Length:   226
........
```

Understanding Outbound Dial-Peer Matching Techniques

Priority

1

```
voice class uri 2001 sip
 host ipv4:10.2.1.1

dial-peer voice 1 VoIP
 destination uri 2001
 carrier-id target orange
```

2

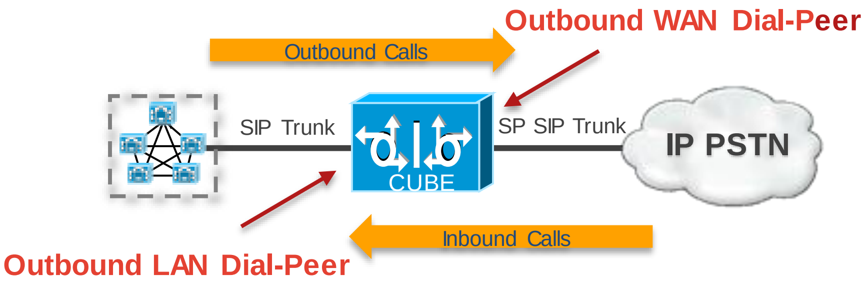
```
dial-peer voice 2 VoIP
 destination-pattern 654321
 carrier-id target orange
```

3

```
voice class uri 2001 sip
 host ipv4:10.2.1.1

dial-peer voice 3 VoIP
destination uri 2001
```

4

```
dial-peer voice 4 VoIP
 destination-pattern 654321
```



Received:

```
INVITE sip:654321@10.2.1.1 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.1:5060;x-route-
tag="cid:orange@10.1.1.1";branch=z9hG4bK-23955-1-0
From: "555" <sip:555@10.1.1.1:5060>;tag=1
To: ABC <sip:654321@10.2.1.1:5060>
Call-ID: 1-23955@10.1.1.1
CSeq: 1 INVITE
Contact: sip:555@10.1.1.1:5060
Supported: timer
Max-Forwards: 70
Subject: BRKUCC-2934 Session
Content-Type: application/sdp
Content-Length:   226
........
```

Understanding Outbound Dial-Peer Matching Techniques

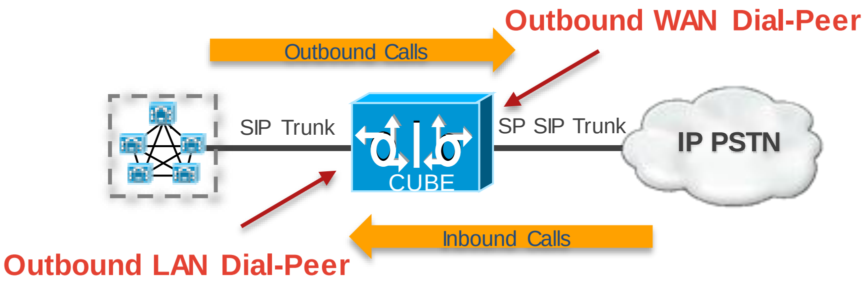
Priority

1
```
voice class uri 2001 sip
 host ipv4:10.2.1.1

dial-peer voice 1 VoIP
 destination uri 2001
 carrier-id target orange
```

2
```
dial-peer voice 2 VoIP
 destination-pattern 654321
 carrier-id target orange
```

3
```
voice class uri 2001 sip
 host ipv4:10.2.1.1

dial-peer voice 3 VoIP
 destination uri 2001
```

4
```
dial-peer voice 4 VoIP
 destination-pattern 654321
```



Received:

```
INVITE sip:654321@10.2.1.1 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.1:5060;x-route-tag="cid:orange@10.1.1.1";branch=z9hG4bK-23955-1-0
From: "555" <sip:555@10.1.1.1:5060>;tag=1
To: ABC <sip:654321@10.2.1.1:5060>
Call-ID: 1-23955@10.1.1.1
CSeq: 1 INVITE
Contact: sip:555@10.1.1.1:5060
Supported: timer
Max-Forwards: 70
Subject: BRKUCC-2934 Session
Content-Type: application/sdp
Content-Length:   226
........
```

Understanding Outbound Dial-Peer Matching Techniques

Priority

1

```
voice class uri 2001 sip
 host ipv4:10.2.1.1

dial-peer voice 1 VoIP
 destination uri 2001
 carrier-id target orange
```

2

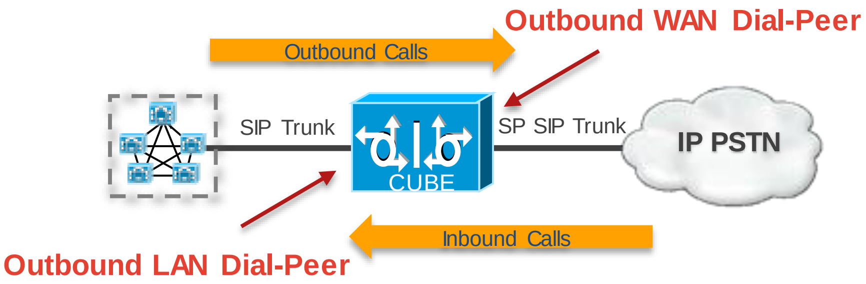
```
dial-peer voice 2 VoIP
 destination-pattern 654321
 carrier-id target orange
```

3

```
voice class uri 2001 sip
 host ipv4:10.2.1.1

dial-peer voice 3 VoIP
 destination uri 2001
```

4

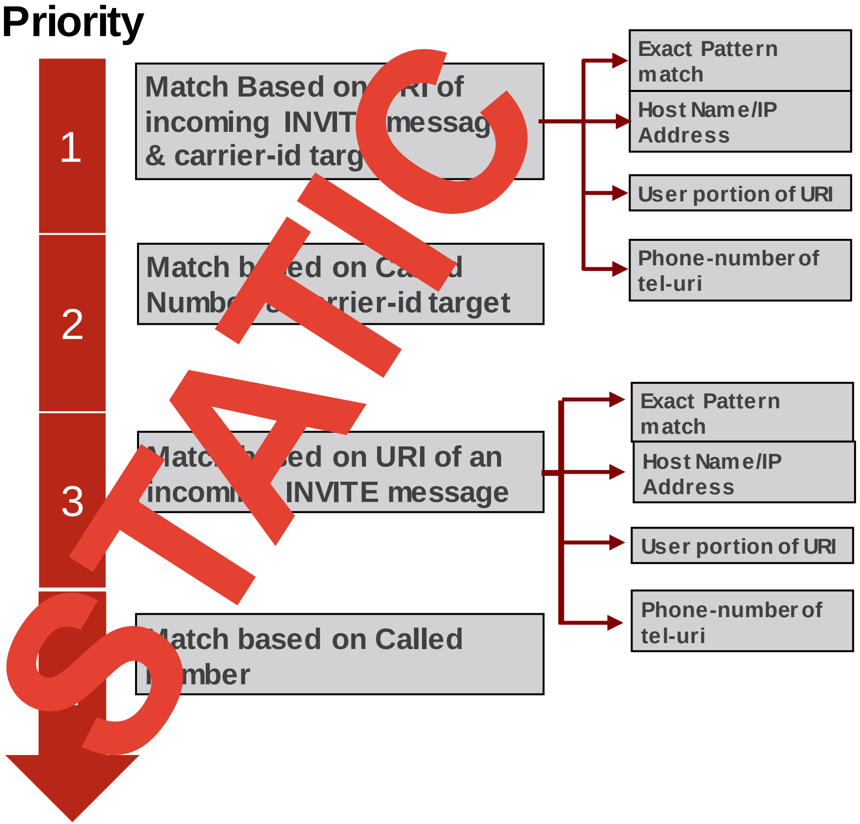
```
dial-peer voice 4 voip
 destination-pattern 654321
```

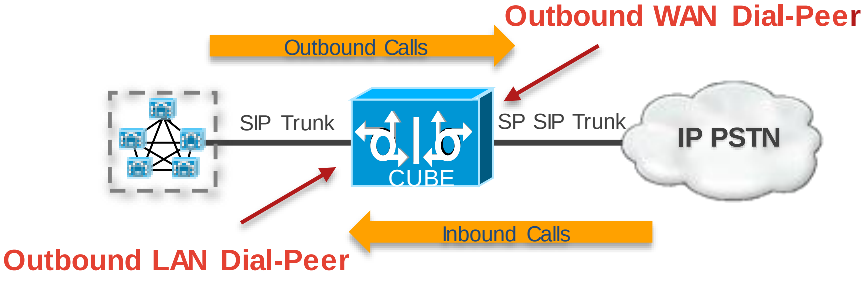



```
Received:
INVITE sip:654321@10.2.1.1 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.1:5060;x-route-tag="cid:orange@10.1.1.1";branch=z9hG4bK-23955-1-0
From: "555" <sip:555@10.1.1.1:5060>;tag=1
To: ABC <sip:654321@10.2.1.1:5060>
Call-ID: 1-23955@10.1.1.1
CSeq: 1 INVITE
Contact: sip:555@10.1.1.1:5060
Supported: timer
Max-Forwards: 70
Subject: BRKUCC-2934 Session
Content-Type: application/sdp
Content-Length:   226
........
```

CUBE Advanced Call Routing

Understanding Outbound Dial-Peer Matching Techniques





Received:
INVITE sip:654321@10.2.1.1 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.1:5060;x-route-tag="cid:orange@10.1.1.1";;branch=z9hG4bK-23955-1-0
From: "555" <sip:555@10.1.1.1:5060>;tag=1
To: ABC <sip:654321@10.2.1.1:5060>
Call-ID: 1-23955@10.1.1.1
CSeq: 1 INVITE
Contact: sip:555@10.1.1.1:5060
Supported: timer
Max-Forwards: 70
Subject: BRKUCC-2934 Session
Content-Type: application/sdp
Content-Length: 226
....

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Additional Headers for Outbound Dial-Peer Matching

Match Based on URI of incoming INVITE message with or without carrier-id target

Match based on CALLED Number with or without carrier-id target

Match Based on FROM Header of incoming INVITE

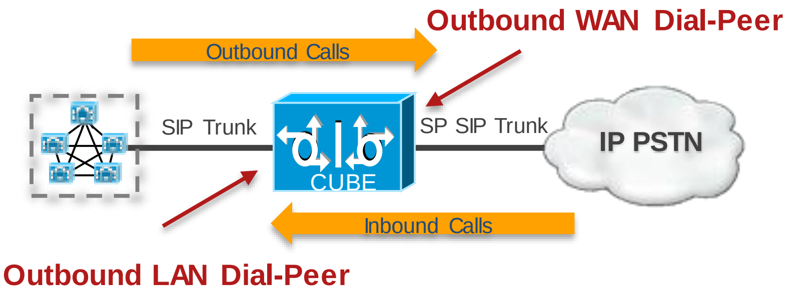
Match Based on TO Header of incoming INVITE

Match Based on VIA Header of incoming INVITE

Match based on DIVERSION Header of incoming INVITE

Match based on REFERRED-BY Header of incoming INVITE

Match based on CALLING Number



Received:
INVITE sip:654321@10.2.1.1 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.1:5060;x-route-tag="cid:orange@10.1.1.1";;branch=z9hG4bK-23955-1-0
From: "555" <sip:555@10.1.1.1:5060>;tag=1
To: ABC <sip:654321@10.2.1.1:5060>
Call-ID: 1-23955@10.1.1.1
CSeq: 1 INVITE
Contact: sip:555@10.1.1.1:5060
Supported: timer
Max-Forwards: 70
Subject: BRKUCC-2934 Session
Content-Type: application/sdp
Content-Length: 226
....

Destination Server Group

15.4.1T/ XE3.11

- Supports multiple destinations (session targets) be defined in a group and applied to a single outbound dial-peer
- Once an outbound dial-peer is selected to route an outgoing call, multiple destinations within a server group will be sorted in either round robin or preference [**default**] order
- This reduces the need to configure multiple dial-peers with the same capabilities but different destinations. E.g. Multiple subscribers in a cluster

```
voice class server-group 1
 hunt-scheme {preference | round-robin}
 ipv4 1.1.1.1 preference 5
 ipv4 2.2.2.2
 ipv4 3.3.3.3 port 3333 preference 3
 ipv6 2010:AB8:0:2::1 port 2323 preference 3
 ipv6 2010:AB8:0:2::2 port 2222
```

*** DNS target not supported in server group**

```
dial-peer voice 100 voip
 description Outbound DP
 destination-pattern 1234
 session protocol sipv2
 codec g711ulaw
 dtmf-relay rtp-nte
 session server-group 1
```

Multiple Incoming Patterns Under Same Incoming/Outgoing Dial-peer

15.4.1T/ XE3.11

```
voice class e164-pattern-map 100
  e164 919200200.
  e164 510100100.
  e164 408100100.

dial-peer voice 1 voip
  description Inbound DP via Calling
  incoming calling e164-pattern-map 100
  codec g729r8
```

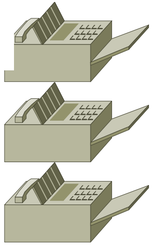
Provides the ability to combine multiple incoming called OR calling numbers on a single inbound voip dial-peer, reducing the total number of inbound voip dial-peers required with the same routing capability

Site A (919)200-2000

Site B (510)100-1000

Site C (408)100-1000

G729 Sites



Site A (919)200-2010

Site B (510)100-1010

Site C (408)100-1010

G711 Sites

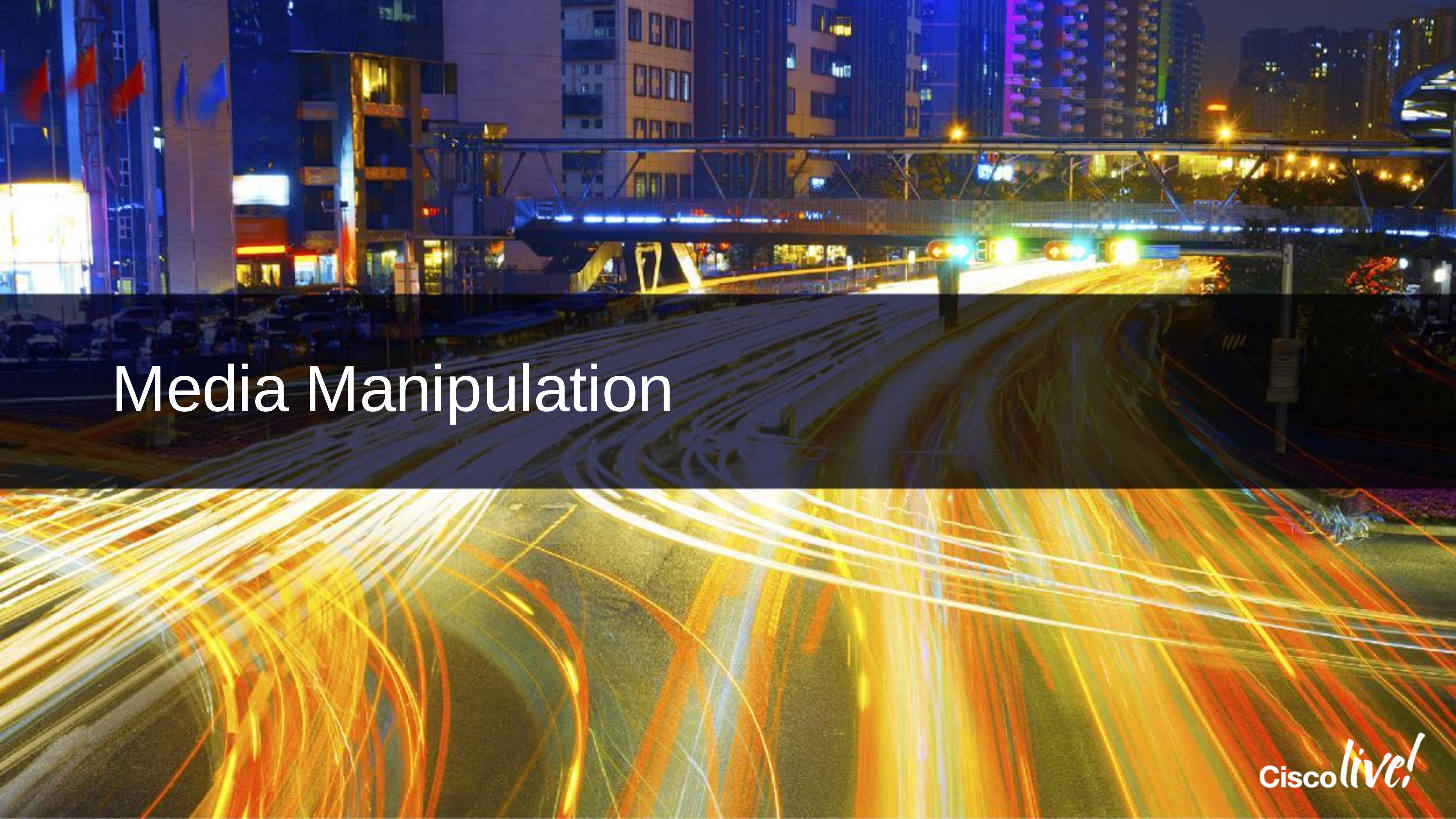
```
voice class e164-pattern-map 200
 url flash:e164-pattern-map.cfg

dial-peer voice 2 voip
  description Inbound DP via Called
  incoming called e164-pattern-map 200
  codec g711ulaw
```

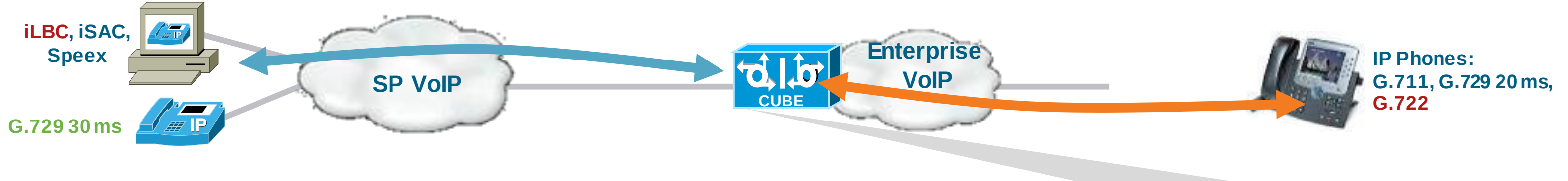
! This is an example of the contents of E164 patterns text file stored in flash:e164-pattern-map.cfg

9192002010
5101001010
4081001010

Media Manipulation



Audio Transcoding and Transrating



- Transcoding: G.711, G.723.1, G.726, G.728, G.729/a, iLBC, G.722
- Transrating: G.729 20ms ↔ 30ms (AT&T)

- Transcoding (12.4.20T)
  - One voice codec to any other codec E.g. iLBC-G.711 or iLBC-G.729
  - Support for H.323 and SIP
  - CUCM 7.1.5 or later supports universal Transcoding
- Transrating (15.0.1M)
  - Different packetisations of the same codec
  - E.g. G.729 20ms to G.729 30ms
  - Support for SIP-SIP calls
  - No sRTP support with transrating

Supported Codecs	Packetization (ms)
G.711 a-law 64 Kbps	10, 20, 30
G.711 µlaw 64 Kbps	10, 20, 30
G.723 5.3/6.3 Kbps	30, 60
G.729, G.729A, G.729B, G.729AB 8 Kbps	10, 20, 30, 40, 50, 60
G.722—64 Kbps	10, 20, 30

```
dial-peer voice 2 voip
 codec g729r8 bytes 30 fixed-bytes
```

```
!Call volume (gain/loss) adjustment
dial-peer voice 2 voip
  audio incoming level-adjustment x
  audio outgoing level-adjustment y
```

Configuration for SCCP Based Transcoding
(ISR-G2/4400)

For Your Reference

1. Enabling dspfarm services under voice-card

```
voice-card 1
    dspfarm
    dsp services dspfarm
```

2. telephony-service configuration

```
telephony-service
    sdspfarm units 1
    sdspfarm transcode sessions 128
    sdspfarm tag 1 CUBE-XCODE
    max-ephones 10
    max-dn 10
    ip source-address
<CUBE_internal_IP> port 2000
```

3. sccp configuration

```
sccp local GigabitEthernet0/0
sccp ccm <CUBE_internal_IP>
                    identifier 1 version 4.0
sccp
sccp ccm group 1
associate ccm 1 priority 1
associate profile 1 register CUBE-XCODE
```

4. dspfarm profile configuration

```
dspfarm profile 1 transcode
    codec g711ulaw
    codec g711alaw
    codec g729r8
    maximum sessions 10
    associate application SCCP
```


Configuration for LTI Based Transcoding

(ISR-G2/4400 & ASR)

1. Enabling dspfarm services under voice-card

```
voice-card 0/1
      dspfarm
      dsp services dspfarm
```

2. dspfarm profile configuration

```
dspfarm profile 1 transcode
     codec g711ulaw
     codec g711alaw
     codec g729abr8
     codec g729ar8
     codec ilbc
     maximum sessions 100
     associate application CUBE
```

Feature Notes:

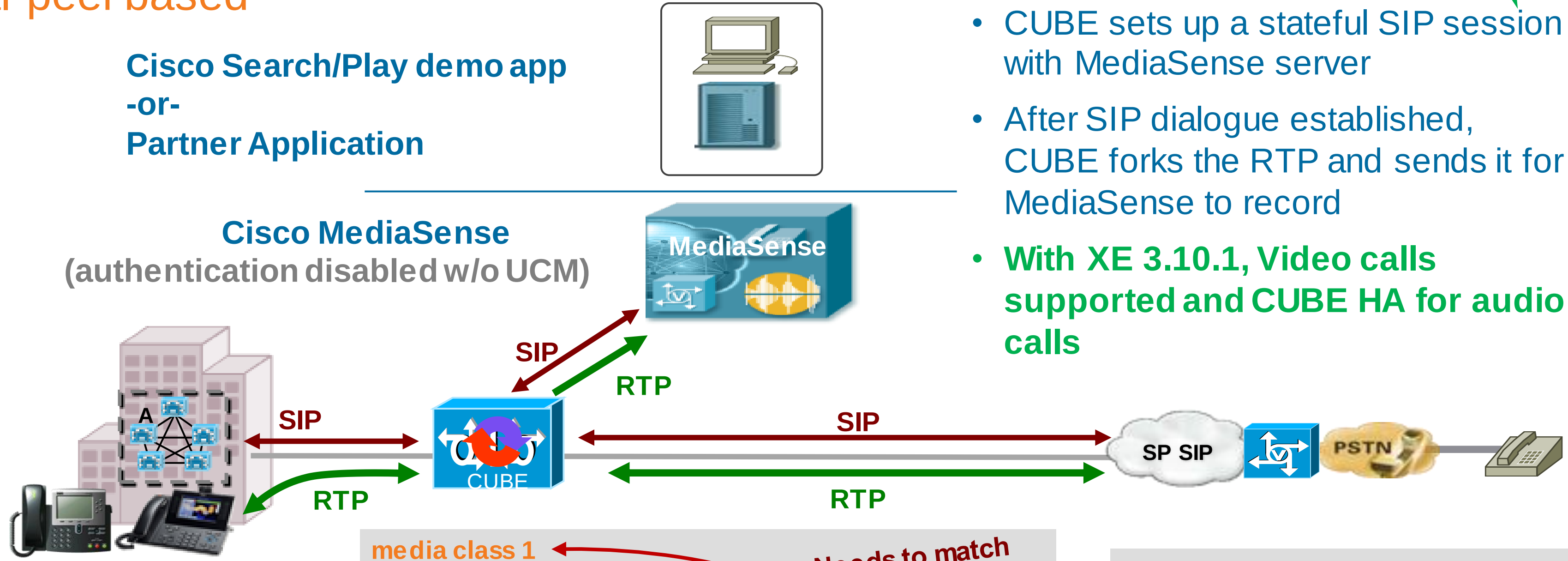
- This uses Local Transcoding Interface to communicate between CUBE and DSPs
- Also available on ISR-G2 starting IOS 15.2.3T
- **Can only be used if CUBE invokes the DSP for media services**
- **CUCM cannot invoke DSPs using this LTI interface**

Call Recording

CUBE Controlled Recording Option – Media Forking

15.2.1T/ XE3.8

Dial-peer based



- CUBE sets up a stateful SIP session with MediaSense server
- After SIP dialogue established, CUBE forks the RTP and sends it for MediaSense to record
- **With XE 3.10.1, Video calls supported and CUBE HA for audio calls**

```
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

Needs to match

```
dial-peer voice 20 voip
  description dial-peer pointing to MediaSense
  session protocol sipv2
  session target ipv4:<Mediasense_IP>
```

- **Call agent independent**
- **Configured on a per Dial-peer level**

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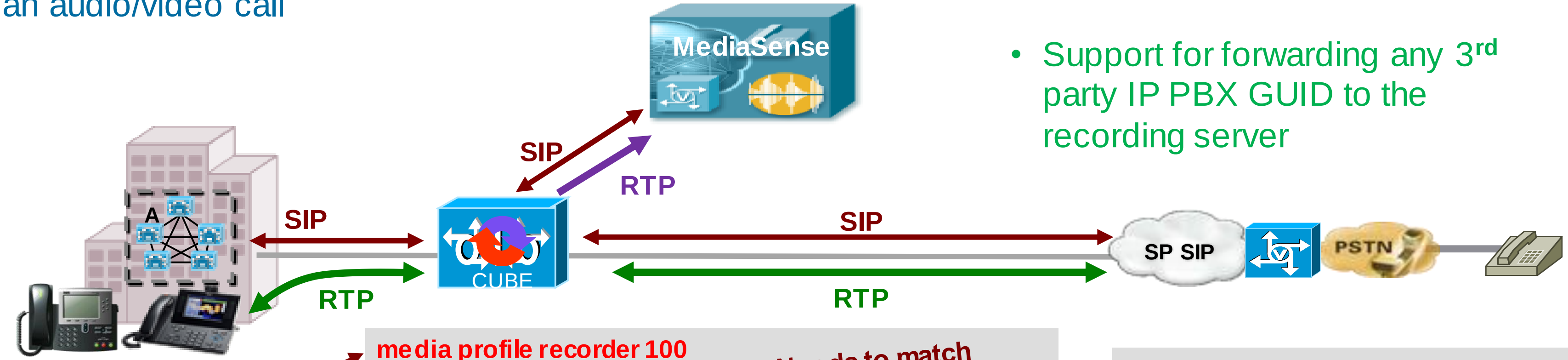
Audio only Media Forking for an Audio/Video Call

15.4.3M/ XE3.13

CUBE Controlled Recording

- MediaSense 10+ or any recording server can decline the video stream and choose to have only the audio stream recorded by setting the video port as 0 in the SDP answer
- CUBE can be configured to offer only audio streams to be recorded even if the call that is being recorded is an audio/video call



- Support for forwarding any 3rd party IP PBX GUID to the recording server

```
media-class 1
  recorder profile 100
```

```
media profile recorder 100
  media-type audio
    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 target ipv4:<Mediasense_IP>
```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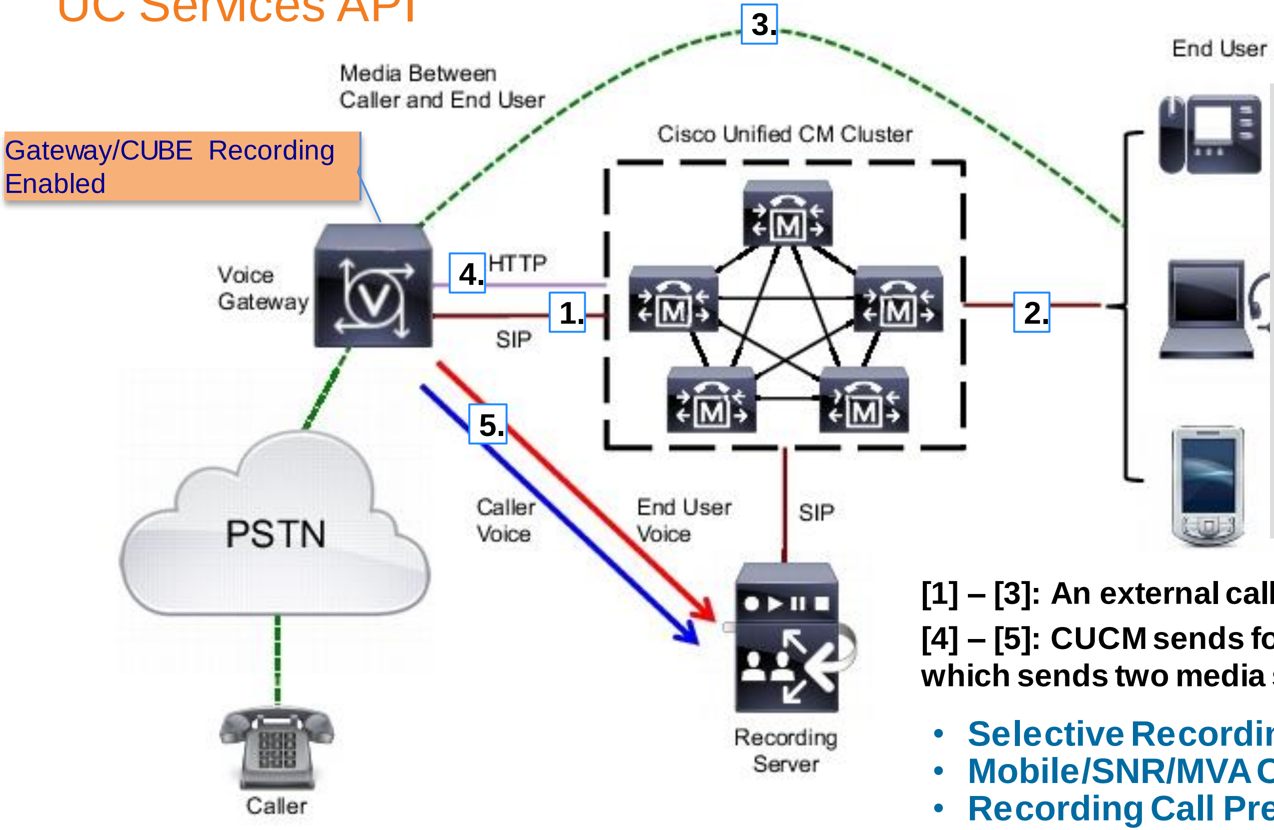
Also see BRKUCC-2250

CUCM (10.X or later) Controlled Recording

UC Services API

15.3.3M1
XE3.10.1
CUCM 10.0



1. Enable HTTP on IOS
ip http server
http client persistent

2. Enable the API on IOS
uc wsapi
source-address [IP_Address_of_CUBE]

3. Enable XMF service within the API
provider xmf
remote-url 1 http://CUCM:8090/ucm_xmf
no shutdown

[1] – [3]: An external call is answered by user with IP phone

[4] – [5]: CUCM sends forking request over HTTP to CUBE, which sends two media streams towards the Recording Server

- **Selective Recording**
- **Mobile/SNR/MVA Calls**
- **Recording Call Preservation**

- **With XE3.13/IOS15.4(3)M, CUBE supports SRTP-SRTP, SRTP-RTP, RTP-SRTP recording. Feature on CUCM roadmap**

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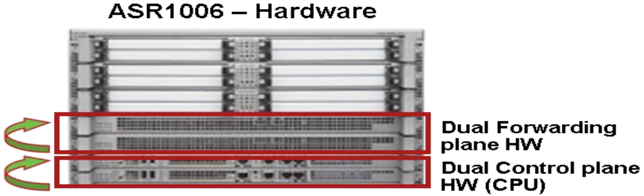
High Availability

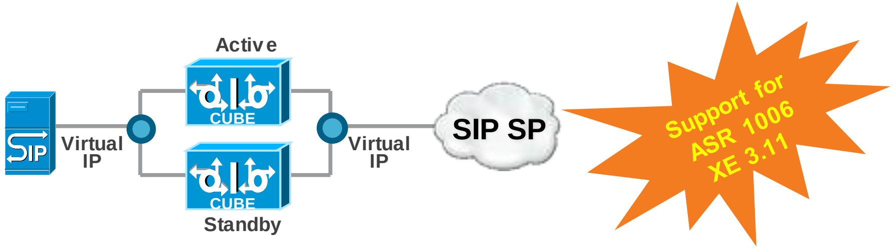
CUBE High Availability Options

- **Inbox redundancy**
  - ASR 1006
  - Stateful failover
  - Local redundancy

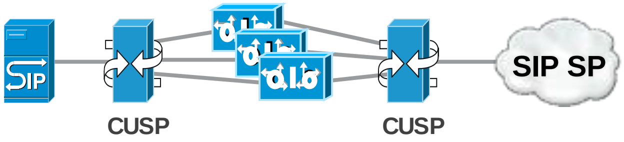
```
ASR(config)#redundancy
ASR-RP2(config-red)#mode sso
ASR-RP2(config-red)#end
```



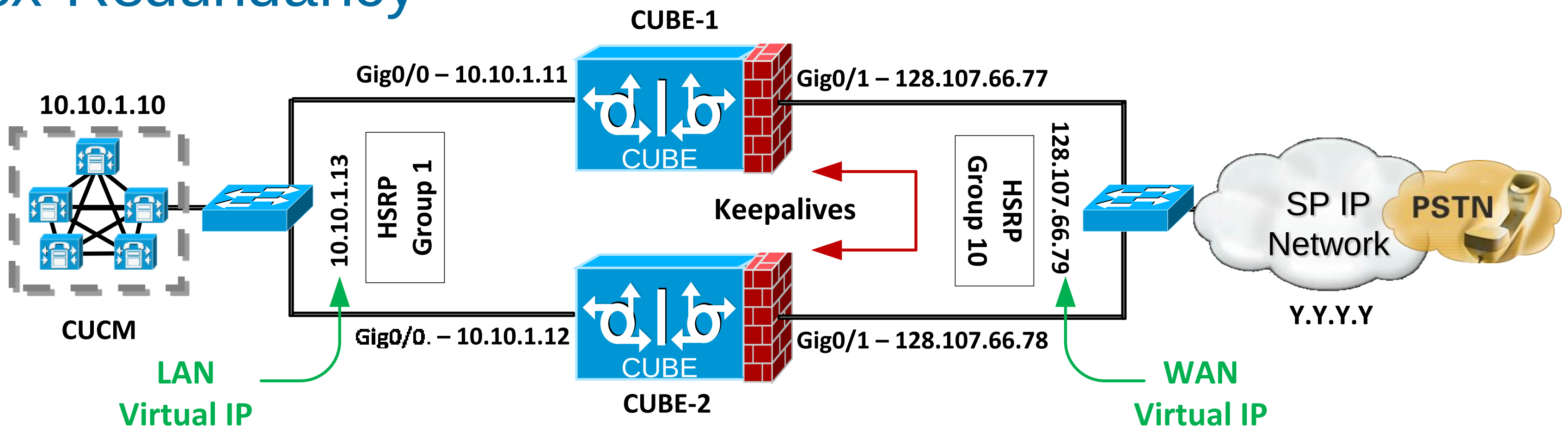
- **L2 Box-to-Box redundancy**
  - ISR G2/4451-X (Stateful failover)
  - ASR 1001/2/4/6 (Stateful failover)
  - Local redundancy (Both routers must be physically located on the same Ethernet LAN)
  - Not supported across data centres
  - Only 1 RP and 1 ESP in ASR1006



- **Clustering with load balancing**
  - All platforms
  - Load balancing by
    - SP call agent
    - Cisco Unified SIP Proxy
  - Local and geographical redundancy



CUBE HA Design Considerations on ISR-G2 for Box-to-Box Redundancy



- All signalling is sourced from/to the Virtual IP Address
- Lower address for both the interfaces (Gig0/0 and Gig0/1) should be on the same platform, which is used as a tie breaker for the HSRP Active state
- HSRP Group number should be unique to a pair/interface combination on the same L2
- Both interfaces of the same group have to be configured with the same priority
- Multiple HSRP interfaces require preemption with interface tracking to be configured
- No media-flow around, SDP-Passthru, or UC Services API support for CUBE HA

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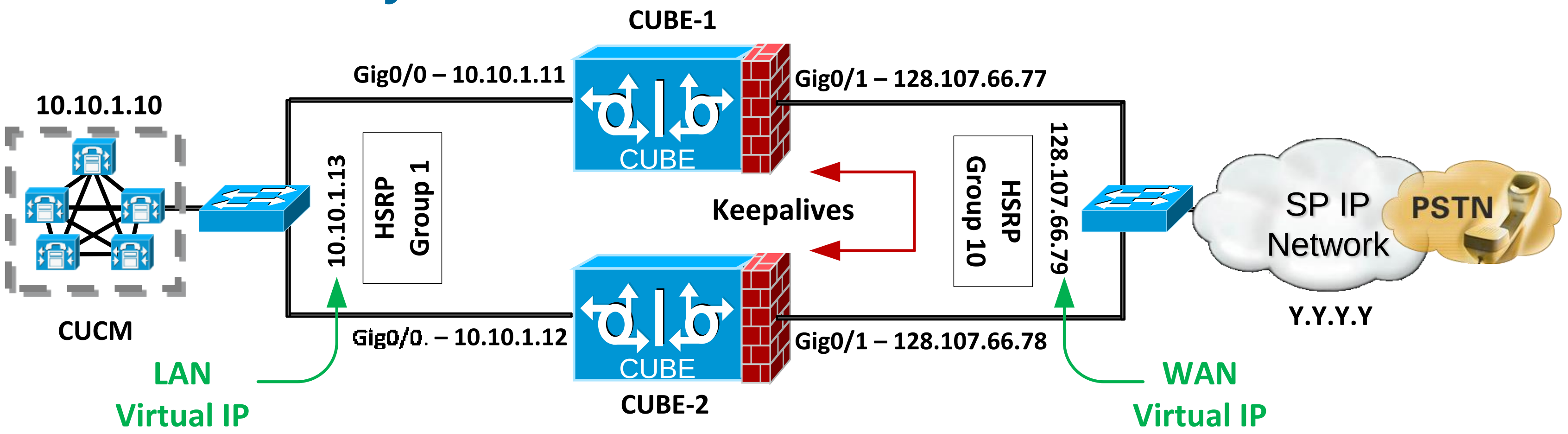
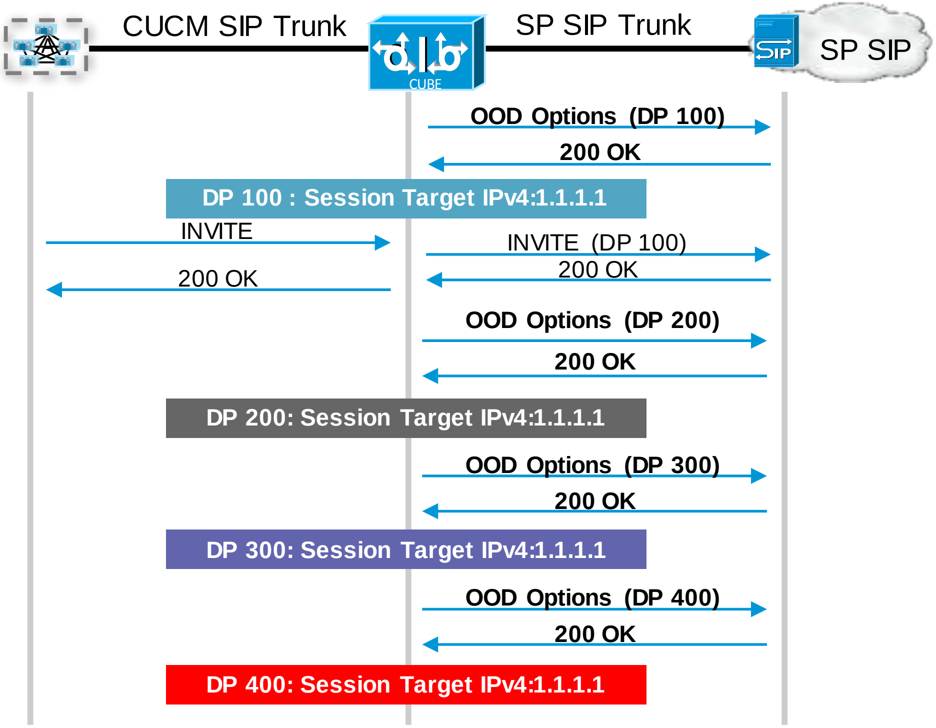
CUBE HA Design Considerations on ISR-G2 for Box-to-Box Redundancy – Cont'd



- Both platforms must be connected via a Switch for CUBE HA to work. Cannot have WAN terminated on CUBEs directly or Data HSRP on either side
- TDM or VXML GW cannot be collocated with CUBE HA
- Both the CUBEs must be running on the same type of platform and IOS version and identical configuration. Loopback interfaces cannot be used as they are always up.
- Some call flows requiring DSPs will be preserved in a future release [15.5(2)T] – March 2015
- **Upon failover, the ACTIVE CUBE goes through a reload**

OOD OPTIONS Ping Keepalive Enhancement

15.4.1T/ XE3.11



- Each dial-peer that has OPTIONS message configured sends out a separate message, even if the session targets are same
- Network bandwidth and process runtime are wasted in CUBE and remote targets to sustain duplicate OOD OPTIONS Ping heartbeat keepalive connection
- Consolidate SIP OOD Options Ping connections by grouping SIP dial-peers with same OOD Options Ping setup
- **New CLI :** "`voice class sip-keepalive-profile` <tag>" is used to define OOD OPTIONS Ping setup
- Consolidated SIP OOD Options Ping connection will then be established with a target for multiple SIP dial-peers with the same target and OOD Options Ping profile setup

OOD OPTIONS Ping Keepalive Enhancement - Configuration

```
voice class sip-options-keepalive 1
 description UDP Options consolidation
 down-interval 49
 up-interval 180
 retry 7
 transport udp

dial-peer voice 1 voip
 destination-pattern 6666
 session protocol sipv2
 session target ipv4:10.104.45.253
 voice-class sip options-keepalive profile 1

dial-peer voice 2 voip
 destination-pattern 5555
 session protocol sipv2
 session target ipv4:10.104.45.253
 voice-class sip options-keepalive profile 1
```

Single OOD Option Ping Group applied to multiple dial-peers with same session targets

Sample Show command output

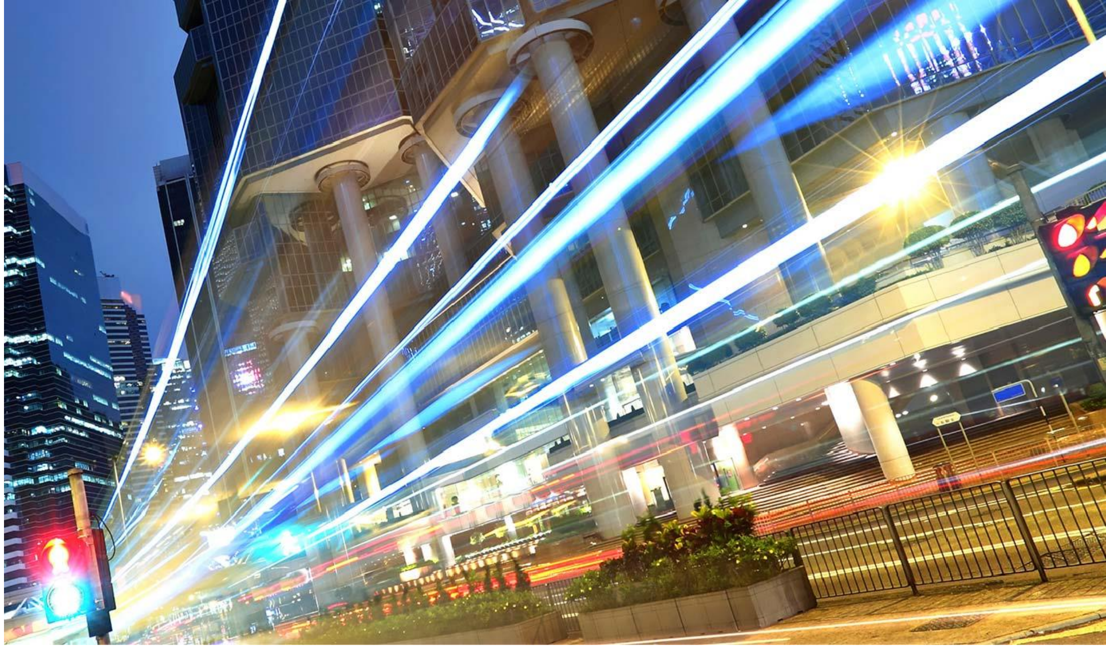
```
CUBE#sh voice class sip-options-keepalive 1
Voice class sip-options-keepalive: 1            AdminStat: Up
 Description: UDP Options consolidation
 Transport: udp          Sip Profiles: 0
 Interval(seconds) Up: 180              Down: 49
 Retry: 7

  Peer Tag      Server Group    OOD SessID     OOD Stat     IfIndex
  --------      ------------    ----------     --------     -------
  1                             4              Active       9
  2                             4              Active       10
 OOD SessID: 4                   OOD Stat: Active
  Target: ipv4:10.104.45.253
  Transport: udp                 Sip Profiles: 0
```

- With OOD Options Ping Keepalive group, an options ping keepalive connection is established on per remote target base as opposed an options ping keepalive connection established per dial-peer basis
- Up to 10,000 “voice class sip-options-keepalive <tag>” can be defined per system
- Either legacy “sip options-keepalive” or the new “sip options-keepalive profile <tag>” can be configured on a dial-peer

Agenda

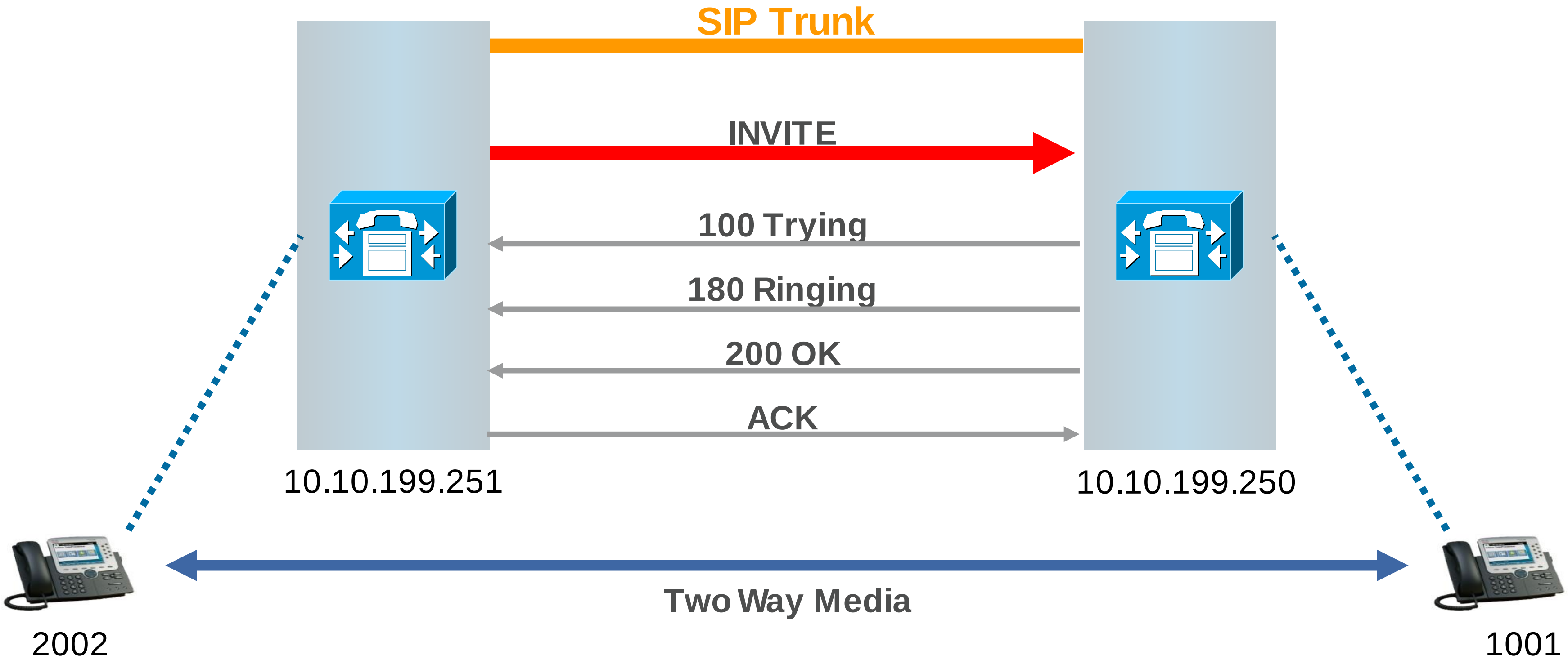
- Why choose SIP for UC Trunking ?
- SIP Basics
- Cisco Unified Border Element (CUBE)
- **Deep down into SIP**
- Deep down into SDP
- CUCM SIP Trunk features and Call functions
- Key Takeaways



Deep Down into SIP

SIP Basics – Typical Call Set Up

SIP Message Exchange



SIP – Messages - INVITE

```
INVITE  sip:1001@10.10.199.250:5060  SIP/2.0
Via: SIP/2.0/TCP  10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>
Date: Wed, 17 Feb 2010 18:37:57 GMT
Call-ID:  8fe4f600-b7c13785-3-fbc712ac@10.10.199.251
Supported:  timer,resource-priority,replaces
Min-SE:  1800
User-Agent:  Cisco-CUCM8.0
Allow:  INVITE,  OPTIONS,  INFO,  BYE,  CANCEL, ACK, PRACK, UPDATE, REFER,  SUBSCRIBE,  NOTIFY
CSeq: 101 INVITE
Contact:  <sip:2002@10.10.199.251:5060;transport=tcp>
Expires: 180
Allow-Events:  presence,  kpml
Supported:  X-cisco-srtp-fallback
Supported:  Geolocation
Call-Info:  <sip:10.10.199.251:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid:  2414147072-3082893189-0000000002-4224127660
Session-Expires:  1800
P-Asserted-Identity:  <sip:2002@10.10.199.251>
Remote-Party-ID:  <sip:2002@10.10.199.251>;party=calling;screen=yes;privacy=off
Max-Forwards:  70
Content-Length:  0
```

Request INVITE to 1001

SIP Message Headers

Some Mandatory

Some Optional

SIP Header Categories: Identity, Timers, Supported Methods and Events, Cisco Related Headers

SIP – Messages – INVITE – Headers Re-grouped

INVITE sip:1001@10.10.199.250:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
CSeq: 101 INVITE

Route and Transaction related

From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>
Call-ID: 8fe4f600-b7c13785-3-fbc712ac@10.10.199.251
P-Asserted-Identity: <sip:2002@10.10.199.251>
Remote-Party-ID: <sip:2002@10.10.199.251>;party=calling;screen=yes;privacy=off
Contact: <sip:2002@10.10.199.251:5060;transport=tcp>

Identity and dialogue related headers

Supported: timer,resource-priority,replaces
Session-Expires: 1800
Min-SE: 1800
Expires: 180

Timer related headers

Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence, kpml

Methods and Events

Supported: X-cisco-srtp-fallback
Supported: Geolocation
User-Agent: Cisco-CUCM8.0
Cisco-Guid: 2414147072-3082893189-0000000002-4224127660

Cisco related headers

SIP INVITE – Request Line

INVITE sip:1001@10.10.199.250:5060 SIP/2.0

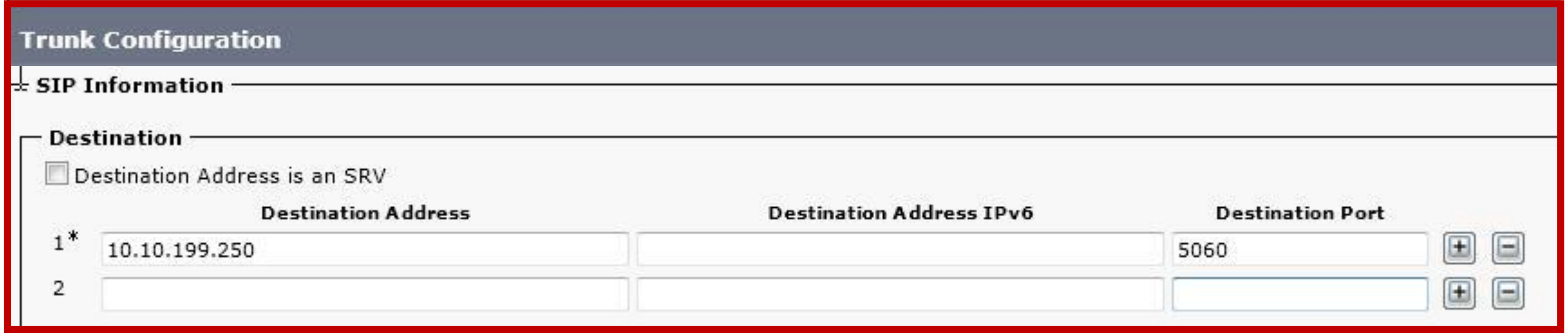
Request = INVITE
SIP URI = sip:user@host:port-number
User = 1001 - Can be a name or a number
Host = 10.10.199.250 - Can be an IP address, hostname or domain name (e.g. cisco.com)

10.10.199.250 – This destination IP address is configured on a outbound CUCM SIP Trunk
5060 – TCP/UDP Port number for SIP signalling
SIP/2.0 – SIP protocol version

How CUCM configuration affects this INVITE Request :

SIP Trunk destination configured using IP addresses - Host portion = IP address
SIP Trunk destination configured using FQDN or DNS SRV - Host portion = Name
SIP Trunk destination port number - Default = 5060 - Can be modified

SIP INVITE – Request Line
Related CUCM Configuration – INVITE and To Header



If an IP address is used = INVITE sip:1001@10.10.199.250:5060 SIP/2.0
To: <sip:1001@10.10.199.250>

If a FQDN /DNS SRV used = INVITE sip:1001@cisco.com:5060 SIP/2.0
To: <sip:1001@cisco.com>

FQDN /DNS SRV resolved to an IP address which is used at the IP Layer

SIP INVITE – Via Header

INVITE sip:1001@10.10.199.250:5060 SIP/2.0

…

Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb

A Mandatory Header in Requests and Responses

The Via header is used to record the SIP route taken by a Request and to route a Response back to the originator. A UA generating a Request records its own address in a Via header field. Multiple Via Headers can be used to record the route of a Request through several SIP switches

SIP/2.0 – SIP Protocol Version / TCP – Transport Protocol

10.10.199.251 – IP Address of CUCM generating the Request

5060 – TCP Port number for SIP signalling

Branch – Unique Identifier for this transaction

Exactly the same header is used by both client and server User Agents for this transaction

A transaction = An exchange of messages between User Agents to perform a specific task e.g. Call set up, or call tear down. A transaction consists of one request and all responses to that request. Transactions take place within a peer to peer Dialogue between two User Agents

SIP INVITE – Command Sequence Header

INVITE sip:1001@10.10.199.250:5060 SIP/2.0

…

Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb

…

CSeq: 101 INVITE

Mandatory Header in Requests and Responses

Command Sequence Header - **Identifies and Orders Transactions**

Consists of a sequence number and method

Method = method used in the Request – INVITE

Sequence number – arbitrary integer

The sequence number and method remain the same for each transaction in a dialogue

The method matches the Request

SIP Header Categories :
Identity and Dialogue Related Headers

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SIP INVITE – From and To Headers

INVITE sip:1001@10.10.199.250:5060 SIP/2.0
...
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>

Mandatory Headers in Requests and Responses
Can optionally include a display name
Calling UA appends the From tag
Called UA appends the To tag
Tags must be globally unique

The From and To tags are used with the Call ID to uniquely identify a Dialogue between two UAs

Note that the To and From header fields are not reversed in the response message as one might expect them to be. This is because the To and From header fields in SIP are defined to indicate the direction of the request, not the direction of the message

SIP INVITE – Call-ID Header

INVITE sip:1001@10.10.199.250:5060 SIP/2.0
…
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>
…
Call-ID: 8fe4f600-b7c13785-3-fbc712ac@10.10.199.251

Mandatory Header in all Requests and Responses
The Call-ID header field is an identifier used to keep track of a particular SIP Dialogue.
The originator of the request creates this unique string

The same Call-ID is used in all SIP messages (Requests and Responses) for all transactions within this dialogue

Transactions are tracked by the branch value in the VIA Header
Dialogues are tracked by the Call-ID, From Header tag and To Header tag

SIP INVITE – From Header (and Identity headers)
Related CUCM Config – **Use FQDN in SIP Requests**

INVITE sip:1001@10.10.199.250:5060 SIP/2.0

…

From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697

To: <sip:1001@10.10.199.250>

If this box is checked, CUCM will relay an alphanumeric hostname of a caller to the called endpoint as a part of the SIP header information. This enables the called endpoint to return the call using the received or missed call list.

If the call is originating from a line device on the CUCM cluster, and is being routed on a SIP trunk then the configured Organisational Top-Level Domain (e.g., cisco.com) will be used in the Identity headers, such as From, Remote-Party-ID, and P-Asserted-ID.

From: <sip:2002@cisco.com>

Headers – P-Asserted-ID and Remote-Party-ID

INVITE sip:1001@10.10.199.250:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
CSeq: 101 INVITE

Route and Transaction related

From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>
Call-ID: 8fe4f600-b7c13785-3-fbc712ac@10.10.199.251
P-Asserted-Identity: <sip:2002@10.10.199.251>
Remote-Party-ID: <sip:2002@10.10.199.251>;party=calling;screen=yes;privacy=off
Contact: <sip:2002@10.10.199.251:5060;transport=tcp>

Identity and dialogue related headers

Supported: timer,resource-priority,replaces
Session-Expires: 1800
Min-SE: 1800
Expires: 180

Timer related headers

SIP INVITE : P-Asserted-Identity Header

INVITE sip:1001@10.10.199.250:5060 SIP/2.0
…
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>
…
P-Asserted-Identity: <sip:2002@10.10.199.251>

Optional Header - This option is checked by default on a CUCM SIP trunk

The P-Asserted Identity and Privacy headers can be used to provide the following services :

Calling identity delivery	**From: “Bob Jones” <sip:2002@10.10.199.251>**
Calling identity blocking	**From: "Anonymous" <sip:localhost>**
Tracing originator of call	**P-Asserted-Identity: “Bob Jones” <sip:2002@10.10.199.251>**

The optional Privacy header can be sent to indicate whether or not privacy (identity delivery/ Identity blocking in the From header) is invoked for this call.

CUCM Config : P-Asserted-Identity – Asserted Type

Directory Number = 2002

Name = Bob Jones

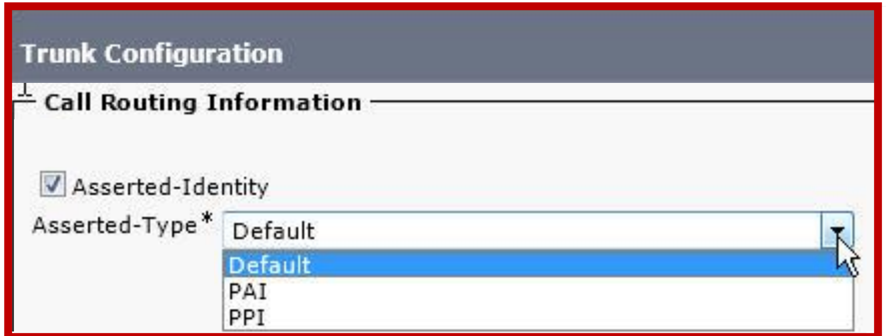
Default = P-Asserted-Identity: “Bob Jones” <sip:2002@10.10.199.251>

PAI = P-Asserted-Identity: “Bob Jones” <sip:2002@10.10.199.251>

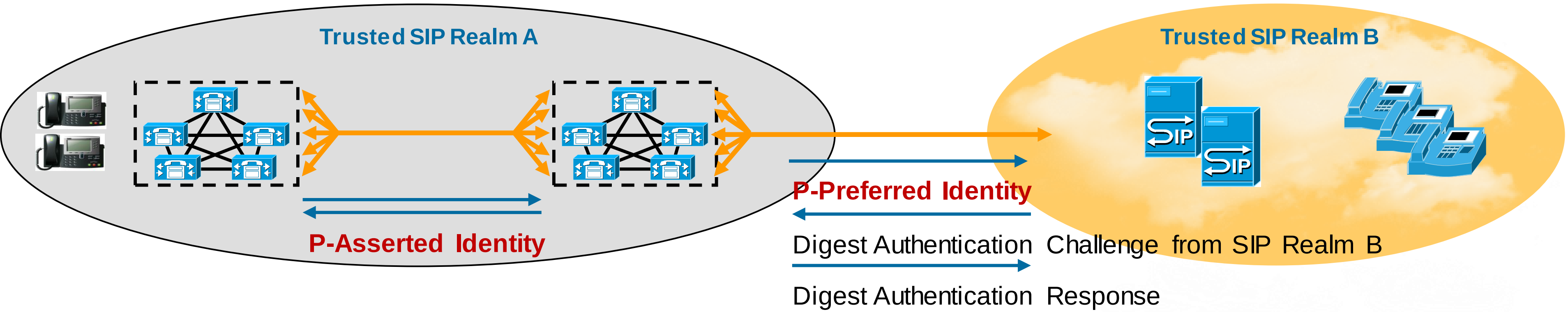
PPI = P-Preferred-Identity: “Bob Jones” <sip:2002@10.10.199.251>

<u>Default value</u>: PAI/ PPI value inherited from calling device/ trunk

Cisco Phone identity is Trusted so P-Asserted-ID sent



SIP INVITE : P-Asserted-Identity and P-Preferred-Identity



P-Asserted Identity is sent within a Trusted Realm

P-Preferred Identity is sent to/ received from an Untrusted Realm

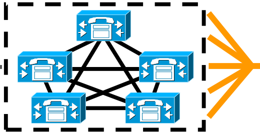
When CUCM sends P-Preferred-Identity, it will respond to a Digest Authentication Challenge from a Trunk peer in another SIP Realm. Digest Authentication takes place at the Trunk Level (Configure the remote Realm, User ID and Digest p/w via CUCM User Management)

CUCM does not send a Digest Authentication Challenge when a P-Preferred Identity is received. Not an issue - as connections to untrusted SIP Realms should always be via a Session Border Controller – which handles Authentication.

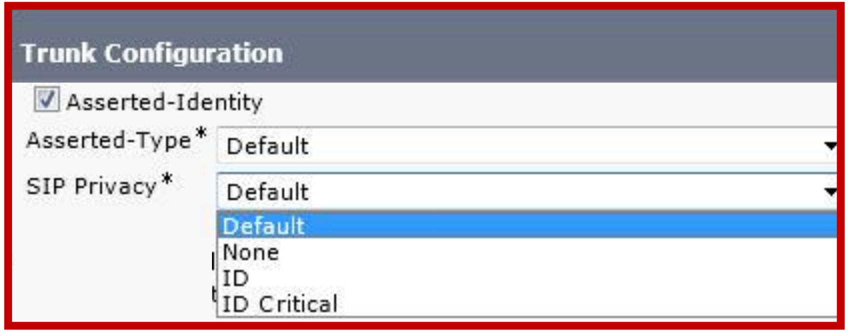
CUCM Configuration : PAID/PPID – SIP Privacy Header

Directory Number = 2002

Name = Bob Jones



From: "Anonymous" <sip:localhost>
P-Asserted-Identity: “Bob Jones” sip:2002@10.10.199.251
Privacy : ID



If non default – the PAI Privacy header value always overrides Device/ Trunk/ RPID Presentation/Restriction ID settings

Privacy :Default	Privacy values taken from Trunk/ Device - Presentation/Restriction settings
Privacy :None	Implies “Presentation Allowed” - No Privacy Header sent
Privacy :ID	Presentation restricted for name and number – Overrides device setting
Privacy :ID Critical	Presentation restricted – Must be supported by network, or call fails

SIP INVITE : Remote-Party-ID Header

INVITE sip:1001@10.10.199.250:5060 SIP/2.0

…

From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697

To: <sip:1001@10.10.199.250>

…

P-Asserted-Identity: <sip:2002@10.10.199.251>

Remote-Party-ID: <sip:2002@10.10.199.251>;party=calling;screen=yes;privacy=off

Optional Header - This option is checked by default on a CUCM SIP trunk

Remote Party ID can be used to provide the following services :

Calling identity delivery	**From: “Bob Jones” <sip:2002@10.10.199.251>**
Calling identity blocking	**From: "Anonymous" <sip:localhost>**
Tracing originator of call	**Remote-Party-ID: “Bob Jones” <sip:2002@10.10.199.251>**

PAID and Remote Party-ID are independent mechanisms for the display of identity info –
Non Default PAI Privacy header values always take precedence over RPID privacy values

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CUCM Configuration - Remote Party ID

Directory Number = 2002

Name = Bob Jones

From: “Bob Jones" <sip:2002@10.10.199.251>

Remote-Party-ID: “Bob Jones”<sip:2002@10.10.199.251>; party=calling; screen=yes; privacy=off

Trunk Configuration

Call Routing Information

☑ Remote-Party-Id

Remote-Party-ID differs from PAI in that it has no authentication challenge mechanism

Party value = Calling/Called

Screen value = Yes – ID from CUCM verified device

= No if “Screen = No” received over Q931/SIP

Privacy value = Name/ URI/ Full/ Off

Privacy values taken from Device or Trunk settings for ID Presentation and Restriction

Trunk Privacy setting values over-ride Device Privacy setting values

Number and Name Presentation Information From/ RPID/ PAI Header Priority

For Calling Name, Calling Number / Connected Name and Connected Number

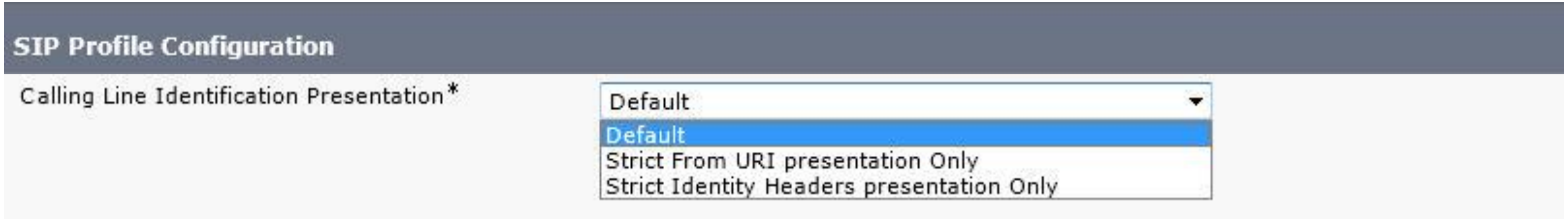
The following headers in priority order are used to select the presented user information

1) **PAI header**
2) **RPID header**
3) **From header**

UC 10.0 allows this order to be changed……

The Device, Trunk and PAI Privacy settings can affect the presentation and restriction of the Calling Name and Number / Connected Name in the From header

CUCM SIP Trunk Features (UC 10.0)
SIP Profile settings – CLID Presentation



Calling Line Identification Presentation applies to inbound Requests and Responses

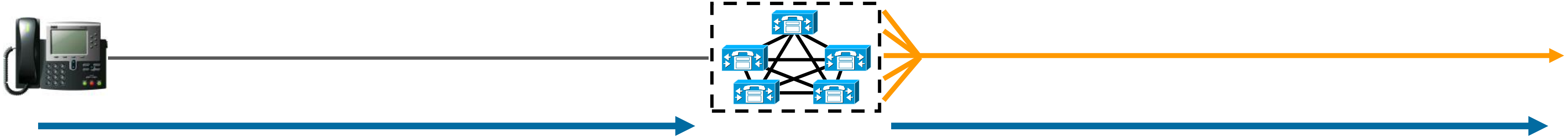
This feature affects

Calling Party Number and Name for inbound calls

Connected Party Number and Name for outbound calls

1) Select Strict From URI presentation to :
Process identity using the From header only
2) Select Strict Identity Headers presentation to :
Process identity using the PAI and RPID Identity headers

Line-side - Device- Presentation/Restriction of Calling Line ID and Calling Name



Directory Number = 2002

Name = Bob Jones

From: “Bob Jones" <sip:2002@10.10.199.251>
or From: “Bob Jones" <sip:localhost>
or From: "Anonymous" <sip:2002@10.10.199.251>
or From: "Anonymous" <sip:localhost>

Applied via Transformation Pattern /Translation Pattern

Calling Line ID Presentation* Default
Default
Allowed
Restricted

Applied Translation Pattern

Calling Name Presentation* Default
Default
Allowed
Restricted

Phone Caller ID Values :

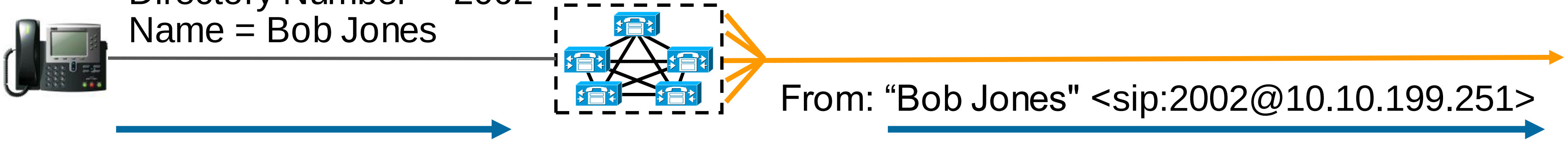
Default = Do not change ID/Name

Allowed

Restricted

SIP Trunk - Calling Line ID and Calling Name Presentation/Restriction – Outbound Calls

Directory Number = 2002
Name = Bob Jones



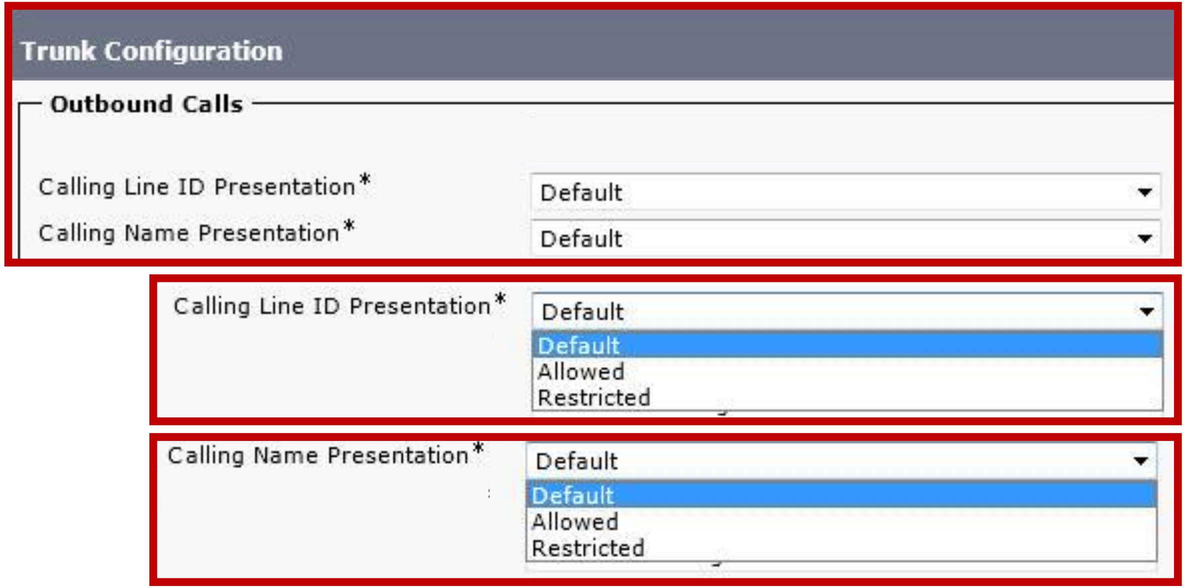
From: “Bob Jones" <sip:2002@10.10.199.251>

<u>Calling Line ID and Calling Name Presentation/ Restriction Trunk settings</u> :

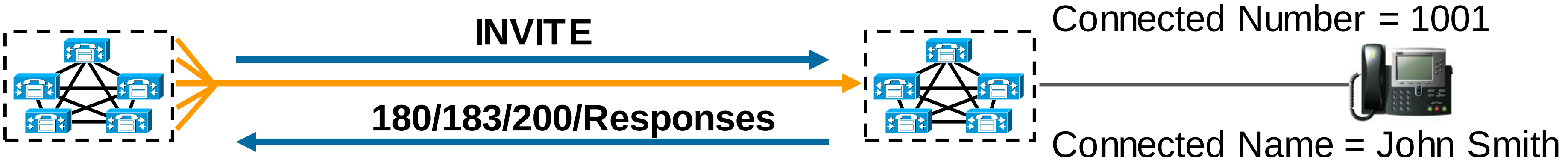
<u>Default</u> - Use calling device values
<u>Allowed</u> - RPID privacy value = Off
<u>Restricted</u> - RPID privacy value = Name/ URI/ Full

PAI Privacy if non Default - overrides Trunk and Device settings

Trunk Settings override Device settings



SIP Trunk - Connected Line ID and Connected Name Presentation/Restriction – Inbound Calls



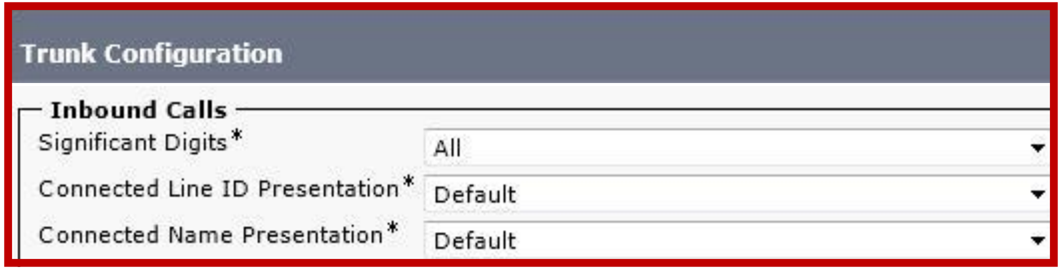
Connected Line ID and Connected Name Presentation/ Restriction affect the privacy value in the RPID header sent in :

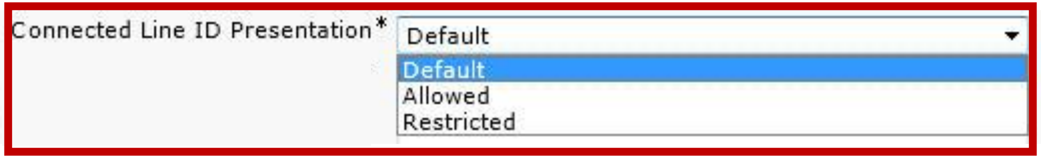
180, 183 Responses and 200 Responses

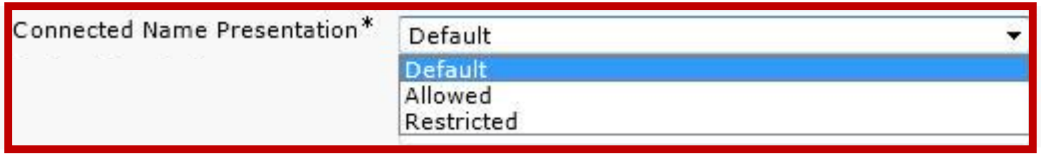
<u>Default</u> - Use calling device values
<u>Allowed</u> - RPID privacy value = Off
<u>Restricted</u> - RPID privacy value = Name/ URI/ Full

PAI Privacy if non Default overrides Trunk and Device settings

Trunk Settings override Device settings







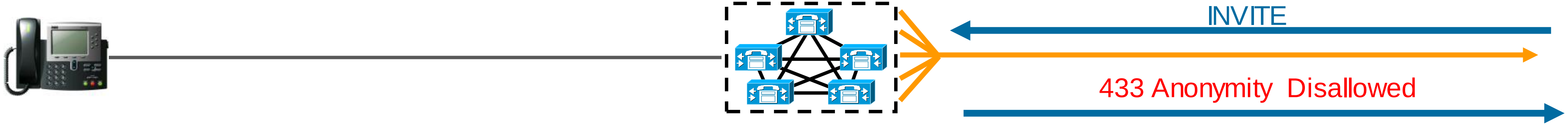
Number and Name – Presentation and Restriction Effects of Device, Trunk and PAI settings

Device Lowest Precedence	Trunk Higher Precedence	RPID	PAI Highest Prec.	Presented User Info
Calling Line and Calling Name Presentation and Restriction setting	Calling Line and Calling Name Presentation and Restriction setting	Privacy field (Set by Trunk Presentation/ Restriction configuration)	Privacy Header setting	User Details Presented or Restricted
Allowed	Restricted	Full	Default	Anonymous
Allowed	Restricted	Full	None	Presented
Allowed	Restricted	Full	ID/ ID Critical	Anonymous
Restricted	Allowed	Off	Default	Presented
Restricted	Allowed	Off	None	Presented
Restricted	Allowed	Off	ID/ ID Critical	Anonymous
Restricted	Default	Full	Default	Anonymous
Restricted	Default	Full	None	Presented
Restricted	Default	Full	ID/ ID Critical	Anonymous

CUCM SIP Trunk Features
SIP Profile Settings – Reject Anonymous Calls

SIP Profile Configuration

- [x] Reject Anonymous Incoming Calls
- [ ] Reject Anonymous Outgoing Calls



From: "Anonymous" sip:localhost
P-Asserted-Identity: “Jim Smith” sip:8888@10.10.10.1
Privacy : ID
Remote-Party-ID: “jim Smith”<sip:8888@10.10.10.1>;
party=calling;screen=yes;privacy=full

Trunk Configuration

- [x] Asserted-Identity

Asserted-Type* Default

SIP Privacy* Default

- Default
- None
- ID
- ID Critical

Note – This feature is based on Identity header settings, Not the From Header value i.e. If From header is Anonymous and PAI Privacy = None, or RPID Privacy = Off – the call is not rejected – the Call proceeds

SIP INVITE – Contact Header

INVITE sip:1001@10.10.199.250:5060 SIP/2.0
…
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>
…
P-Asserted-Identity: <sip:2002@10.10.199.251>
Remote-Party-ID: <sip:2002@10.10.199.251>;party=calling;screen=yes;privacy=off
…
Contact: sip:2002@10.10.199.251:5060;transport=tcp

Mandatory in INVITE Requests and 2XX Responses
A Contact header field value can contain a display name, a URI with URI parameters, and header parameters
In a Request the contact field contains the address at which the Calling UA can be reached
In a Response the contact field contains the address at which the Called UA can be reached
With CUCM – a B2BUA – The address in the contact header field is the address of the CUCM server, not the phone

SIP Header Categories: Timer Related Headers

SIP INVITE – Supported Header

INVITE sip:1001@10.10.199.250:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>
…
Supported: timer,resource-priority,replaces

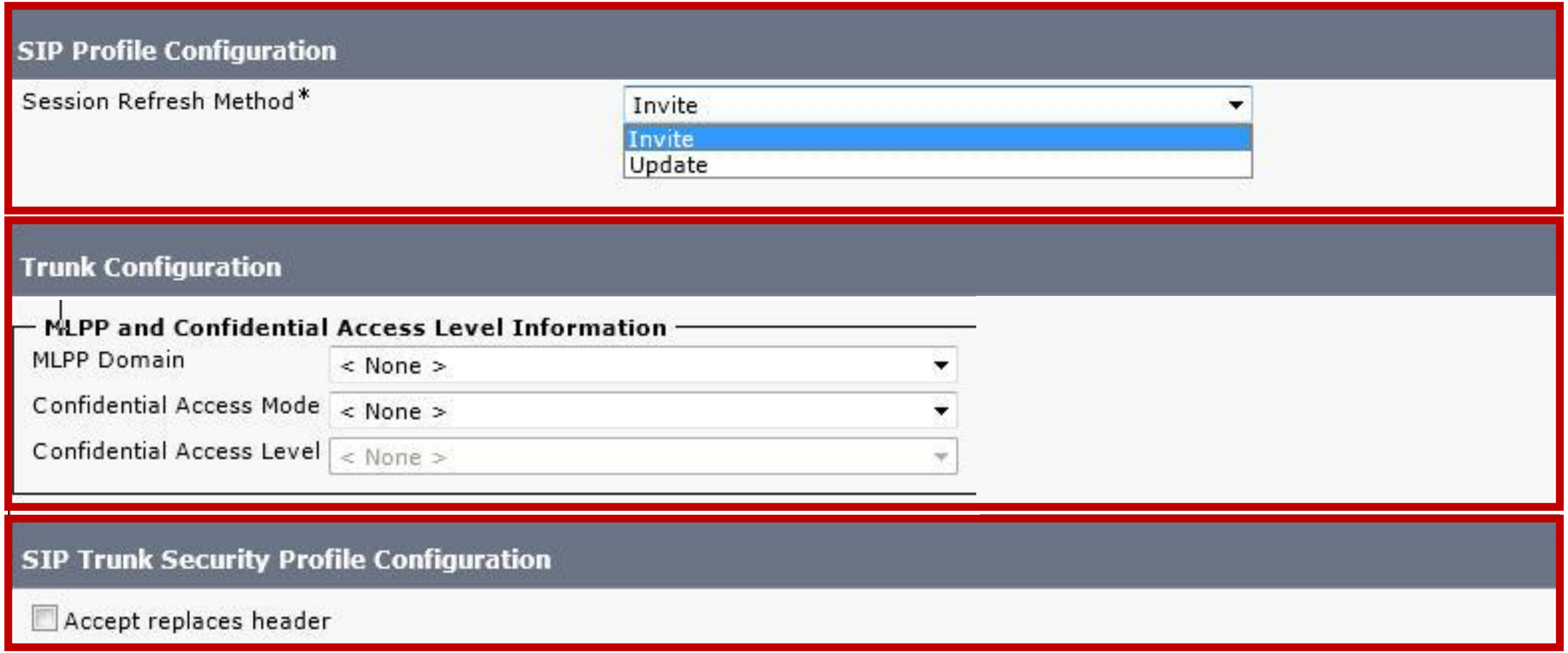
Should be sent in an INVITE
Indicates new SIP options supported by this UA

Options Supported : timer, resource-priority, replaces

Timer – indicates support for session timers as keep-alives to refresh sessions
Resource-priority – used for resource contention resolution, pre-emption
Replaces - Replaces header is used to logically replace an existing SIP dialogue with a new SIP dialogue. Can be used in attended Transfers, retrieve from Call Pick up etc.

Related CUCM Configuration: Supported Header

Supported: timer, resource-priority, replaces



This header indicates support only. i.e. The Trunk will not accept the “replaces” and “resource-priority” options if the corresponding Trunk settings have not been configured/ enabled

SIP INVITE – Session Expires Header

INVITE sip:1001@10.10.199.250:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>
…
Supported: timer, resource-priority, replaces
Min-SE: 1800
…
Session-Expires: 1800

Optional Header - Support indicated via the Supported: “timer” header option

Session-Expires Header used with the “Min-SE” header as a session keep-alive mechanism

Called UA responds with a Session-Expires header in a 2XX message and refresher parameter to indicate who (UAS or UAC) is doing the refreshing.

Sessions can be refreshed with a Re-INVITE or UPDATE request

SIP INVITE – Minimum Session Expires Header

INVITE sip:1001@10.10.199.250:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>
…
Supported: timer, resource-priority, replaces
Min-SE: 1800
…
Session-Expires: 1800

Minimum Session Expires Header - Optional Header –
Allows the sender to enforce a Minimum session timer when the call traverses multiple Proxies

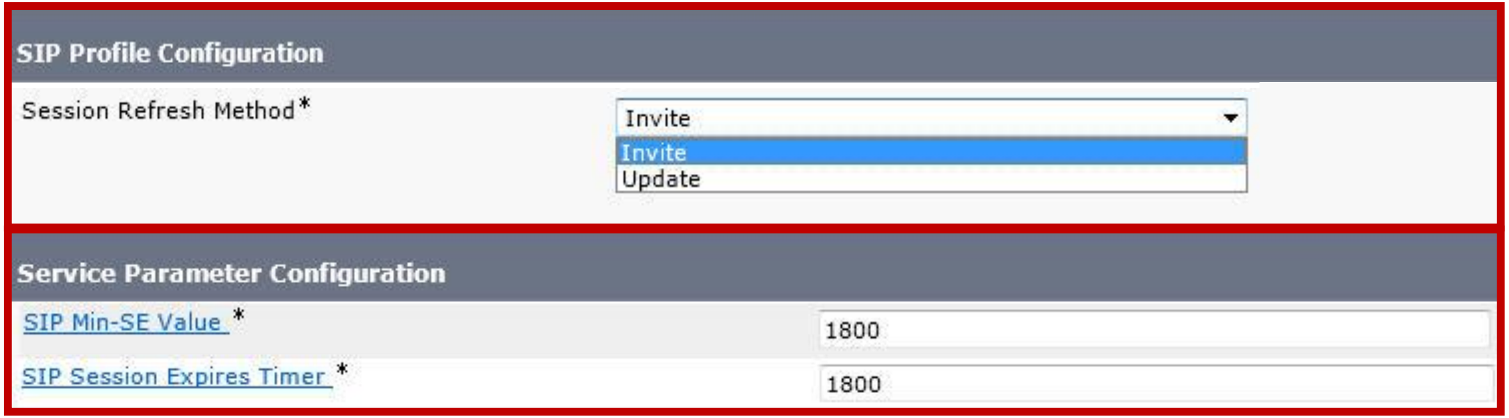
Session-Expires value can an be increased or decreased by intermediate Proxies
Min-SE value can only be increased by intermediate Proxies

Related CUCM configuration

Min-SE Header, Session Expires Header

Supported: timer
Min-SE: 1800
Session Expires: 1800

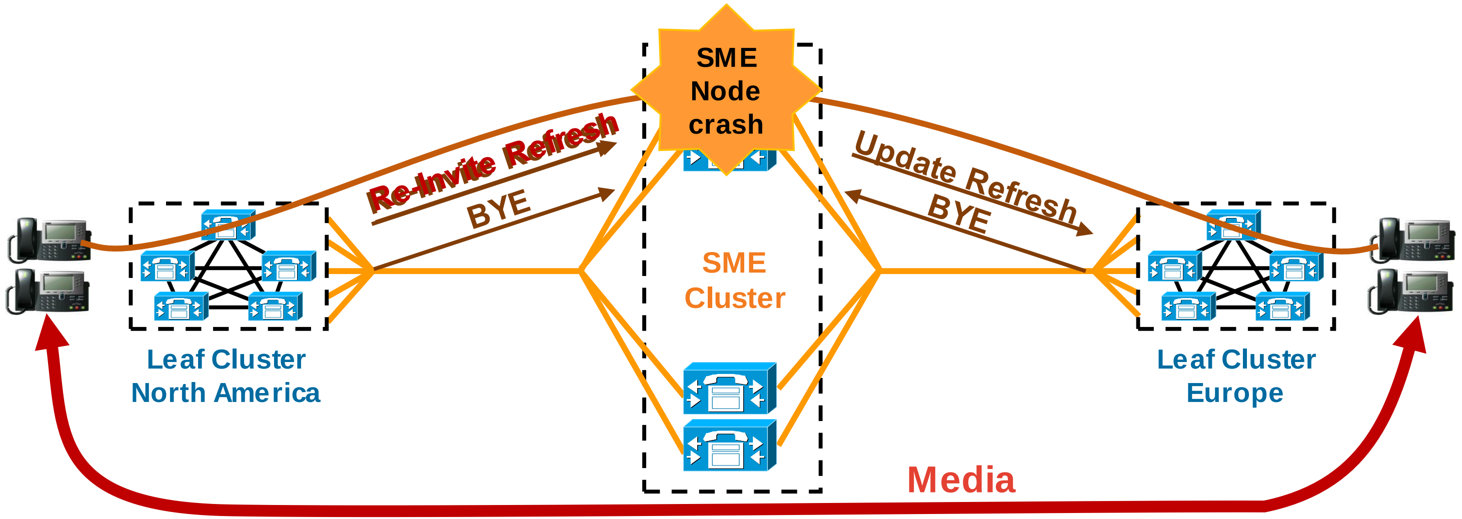
Session Refresh method is configurable (Invite/Update)
Min-SE: 1800 seconds (30 mins) – Default value (Min 60 secs, Max 86400 secs = 24 hours)
Allows the sender to enforce a minimum session timer when the call traverses multiple Proxies
Each Proxy processing this Request can raise the Min-SE value but cannot lower it
Session Expires: 1800 seconds (30 mins) – Default value (Min 90s, Max 86400s = 24 hours)



SIP Session Expires and Min-SE Headers - Operation

…
Supported: timer, resource-priority,replaces
Min-SE: 1800
Session-Expires: 1800

If no session refresh request or response is received before the session expires, the UA sends a BYE to terminate the session

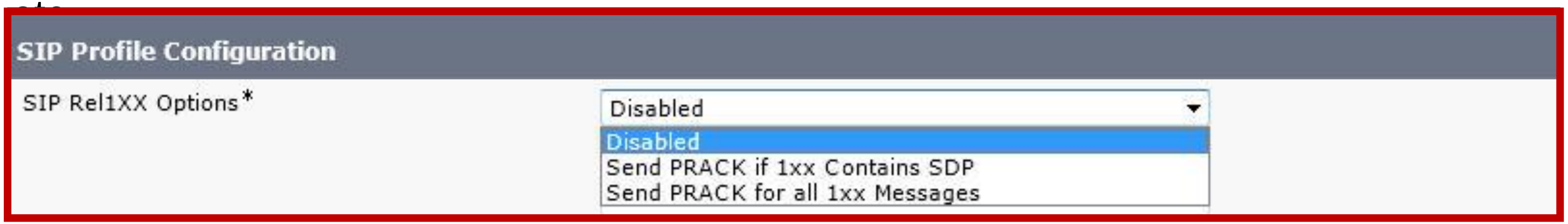


SIP Header Categories: Methods and Events Supported

SIP INVITE – Allow Header

INVITE sip:1001@10.10.199.250:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>
…
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
…

Optional Header - Lists the set of methods supported by the UA sending the message
Note – Although supported – To be used, some methods also need to be enabled on the SIP Trunk e.g. PRACK (shown below), Accept Presence Subscription, Accept Unsolicited NOTIFY etc.



SIP INVITE Allow-Events Header

INVITE sip:1001@10.10.199.250:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>
…
Allow-Events: presence, kpml
…

Optional Header

A UA sending an "Allow-Events" header is advertising that it can process SUBSCRIBE requests and generate NOTIFY requests for all of the event packages listed in that header.

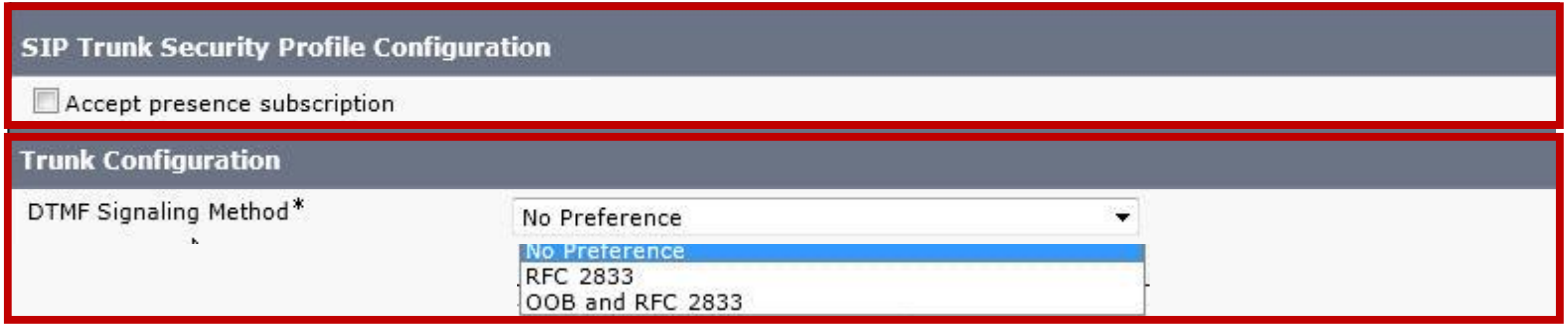
In the above case : Presence and KPML (Out of Band DTMF) event packages are supported

CUCM Configuration: Allow-Events Header

Allow-Events: presence, kpml

A UA sending an "Allow-Events" header is advertising that it can process SUBSCRIBE requests and generate NOTIFY requests for all of the event packages listed in that header. In the above case : Presence and KPML (DTMF)

Note – Although these events are supported by the UA the Trunk may need additional configuration to accept these events e.g.



Default = No Preference – Trunk supports either RFC 2833 or OOB DTMF – UA capabilities sent
RFC 2833 – will override Allow-Events values from UA
OOB and RFC 2833 - will override Allow-Events values from UA

SIP Header Categories: Cisco and Other Headers

SIP INVITE – Supported Header

INVITE sip:1001@10.10.199.250:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>
…
Supported: X-cisco-srtp-fallback
…

Optional Header
X-Cisco-srtp fallback – proprietary header (can be ignored by other vendors)
Allows an offered SRTP session to fall back to RTP if not supported by both UAs

SIP INVITE – Call-Info Header

INVITE sip:1001@10.10.199.250:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>
…
Call-Info: <sip:10.10.199.251:5060>;method="NOTIFY;Event=telephone-event;Duration=500“
…

Optional Header in Requests and Responses
The Call-Info header field provides additional information about the caller or callee, depending on whether it is found in a request or response. (In the above example - The Calling UA)

method="NOTIFY;Event=telephone-event;Duration=500“ indicates support for NOTIFY based out of band DTMF relay. Duration = time in mS between successive NOTIFY messages

Unsolicited NOTIFY used as a Cisco proprietary way to sent DTMF Out Of Band

SIP INVITE – User Agent Header

INVITE sip:1001@10.10.199.250:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>
…
User-Agent: Cisco-CUCM8.0
…

Optional Header
Contains information about the client User Agent originating the request

CUCM configurable : SIP Profile "User-Agent and Server header information"

- Send Unified CM Version Information as User-Agent Header (default)
- Pass Through Received User Agent and Server Information as Contact Header parameters
- Pass Through Received User Agent and Server Information as User-Agent and Server Header

SIP INVITE

Cisco GUID Header – Globally Unique Identifier

INVITE sip:1001@10.10.199.250:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>
…
Cisco-Guid: 2414147072-3082893189-0000000002-4224127660
…

Proprietary Header
Uniquely identifies the call on this Trunk
Typically used in INVITE messages
Maps to the Incoming/ Outgoing "ProtocolCallRef" in CUCM Call Detail Records

Note : Today Trunk to Trunk calls on SME have different GUIDs for inbound and outbound calls
Planning to address this by developing to the draft IETF standard "End-to-End Session Identification in IP-Based Multimedia Communication Networks" in UC 11.0

SIP INVITE – Date Header

INVITE sip:1001@10.10.199.250:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>
…
Date: Wed, 17 Feb 2010 18:37:57 GMT
…

An Optional Header
GMT only

SIP INVITE : Max-Forwards Header

INVITE sip:1001@10.10.199.250:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>
…
Max-Forwards: 70
…

Mandatory Header in all Requests
Not required in Responses

Max-Forwards serves to limit the number of hops a request can make on the way to its destination. It consists of an integer that is decremented by one at each hop. If the Max-Forwards value reaches 0 before the request reaches its destination, it will be rejected with a 483(Too Many Hops) error response. Can be used for loop detection

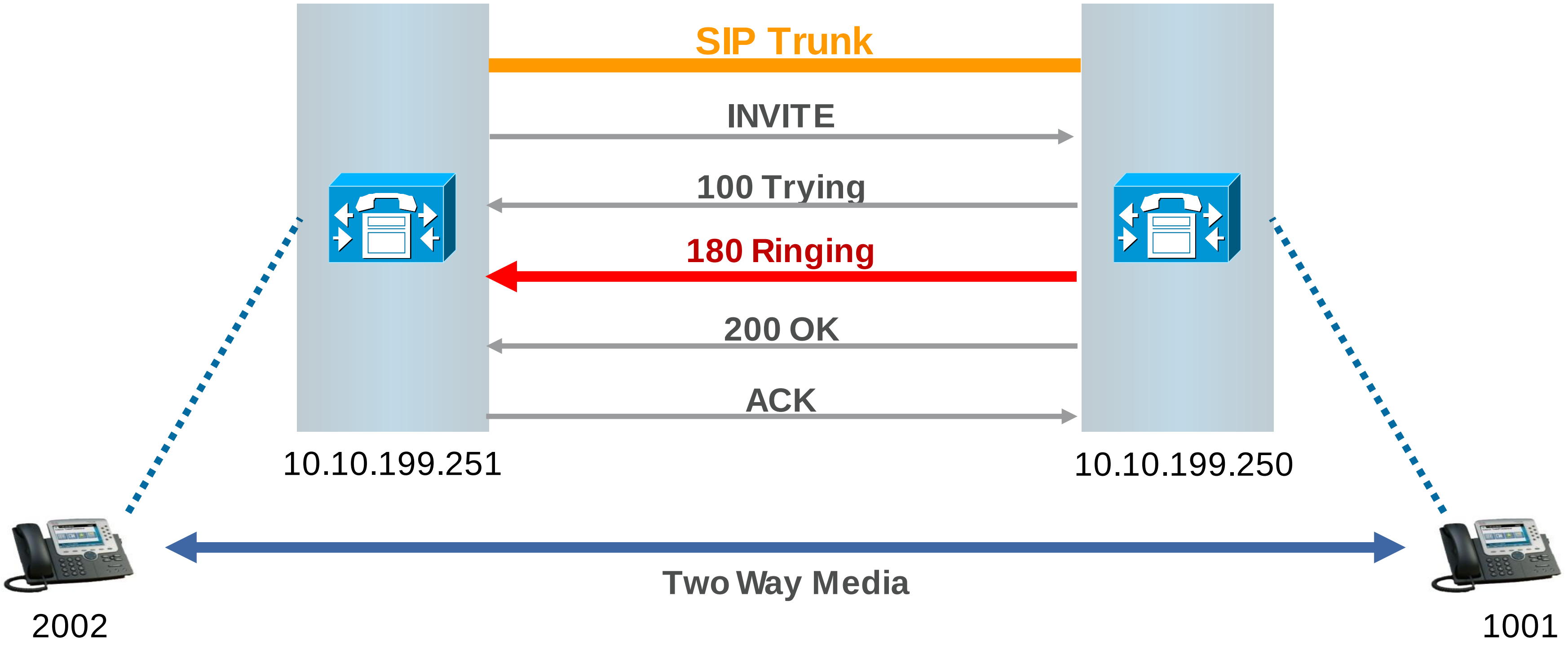
SIP INVITE : Content-Length Header

INVITE sip:1001@10.10.199.250:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>
…
…
Content-Length: 0

Mandatory Header if TCP transport used, Optional if UDP used
The Content-Length header indicates the size of the message-body sent to the recipient in decimal number of bytes.

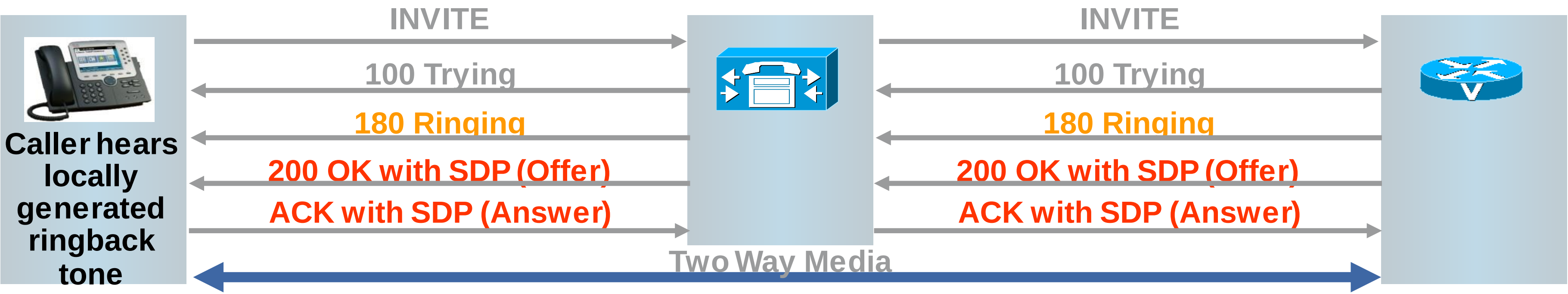
Message-Body – For example, the Session Description Protocol (SDP) message body, which if present would describe the media characteristics supported by the sender. The message body is appended after the Content-Length header.

SIP Basics – Typical Call Set-up SIP Message Exchange



CUCM SIP Trunk Signalling

180 Ringing Response - Ringback



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SIP/2.0 180 Ringing

Indicates that the destination User Agent has received the INVITE, and is alerting the user. Typically this is the first Response that contains information about the capabilities of the Called User Agent

1XX messages are Provisional responses that provide information on the progress of the request. Provisional messages are not sent reliably (i.e. They are not acknowledged) – So the sender of a provisional response does know that it has been received.

SIP Responses – 180 Ringing

SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>;tag=abee6e2b-75b0-4537-80f3-7a3a37d0fa55-32557664
Date: Wed, 17 Feb 2010 18:25:39 GMT
Call-ID: 8fe4f600-b7c13785-3-fbc712ac@10.10.199.251
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Contact: <sip:1001@10.10.199.250:5060;transport=tcp>
Call-Info: <sip:10.10.199.250:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:1001@10.10.199.250>
Remote-Party-ID: <sip:1001@10.10.199.250>;party=called;screen=yes;privacy=off
Content-Length: 0

SIP 180 Ringing
Via Header

SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb

A Mandatory Header in Requests and Responses
SIP/2.0 – SIP Protocol Version / TCP – Transport Protocol
10.10.199.251 – IP Address of CUCM generating the Request
5060 – TCP Port number for SIP signalling
Branch – Unique Identifier for this transaction
This Via header is used by both client and server User Agents for this transaction

Note - This Via Header is exactly the same as that sent in the INVITE and remains the same for all messages in this transaction

The Via header is used to record the SIP route taken by a Request and to route a Response back to the originator. A UA generating a Request records its own address in a Via header field. Multiple Via Headers can be used to record the route through several SIP switches

SIP 180 Ringing
Command Sequence Header

SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
…
CSeq: 101 INVITE

Mandatory Header in Requests and Responses
Command Sequence Header - Identifies and Orders Transactions
Consists of a sequence number and method
Method = method used in the Request – INVITE
Sequence number – arbitrary integer
The sequence number and method remains the same for each transaction in a dialogue
The method matches the request

SIP 180 Ringing Response: From and To Headers

SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>;tag=abee6e2b-75b0-4537-80f3-7a3a37d0fa55-32557664

Mandatory Headers in Requests and Responses
Can optionally include a display name
Calling UA appends the From tag
Called UA appends the To tag
Tags must be globally unique

The From and To tags are used with the Call ID to uniquely identify a dialogue between two UAs

Note that the To and From header fields are not reversed in the Response message as one might expect them to be. This is because the To and From header fields in SIP are defined to indicate the direction of the request, not the direction of the message

SIP 180 Ringing Response: Call-ID Header

SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>;tag=abee6e2b-75b0-4537-80f3-7a3a37d0fa55-32557664
…
Call-ID: 8fe4f600-b7c13785-3-fbc712ac@10.10.199.251
…
Mandatory Header in Requests and Responses
The Call-ID header field is an identifier used to keep track of a particular SIP dialogue.
The originator of the request creates this locally unique string

The same Call-ID is used in all messages (Requests and Responses) for all transactions within this dialogue

Transactions are tracked by the branch value in the VIA Header
Dialogues are tracked by the Call-ID, From Header tag and To Header tag

SIP 180 Ringing Response: Identity Headers

SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>;tag=abee6e2b-75b0-4537-80f3-7a3a37d0fa55-32557664
…
P-Asserted-Identity: <sip:1001@10.10.199.250>
Remote-Party-ID: <sip:1001@10.10.199.250> ;party=called;screen=yes;privacy=off

Optional Headers
These options are checked by default on a CUCM SIP trunk
The P-asserted Identity and Remote-Party-ID can be used to provide the following services :
Calling Identity delivery/ Calling Identity delivery blocking/ Tracing originator of a call.

P-Asserted Identity and Remote Party-ID are independent mechanisms for the display of identity information

SIP 180 Ringing Response: Contact Header

SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>;tag=abee6e2b-75b0-4537-80f3-7a3a37d0fa55-32557664
…
Contact: sip:1001@10.10.199.250:5060;transport=tcp

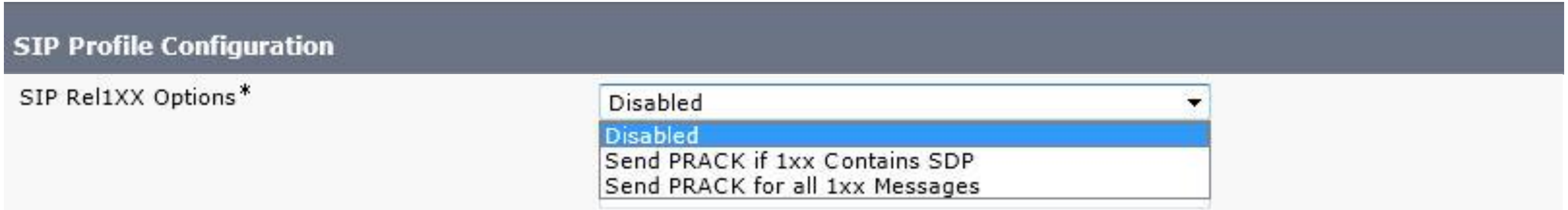
Optional in 1XX Responses (Mandatory in 2XX Responses)

A Contact header field value can contain a display name, a URI with URI parameters, and header parameters
In a Request the contact field contains the address at which the calling UA can be reached
In a Response the contact field contains the address at which the called UA can be reached
With CUCM – a B2BUA – The address in the contact header field is the address of the CUCM server, not the phone

SIP 180 Ringing Response: Allow Header

SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>;tag=abee6e2b-75b0-4537-80f3-7a3a37d0fa55-32557664
…
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY

Optional Header - Lists the set of methods supported by the UA sending the message
Note – Although supported – To be used, some methods also need to be enabled on the SIP Trunk e.g. PRACK, Accept Presence Subscription, Accept Unsolicited NOTIFY etc



SIP 180 Ringing Response: Allow-Events Header

SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>;tag=abee6e2b-75b0-4537-80f3-7a3a37d0fa55-32557664
…
Allow-Events: presence

Optional Header
A UA sending an "Allow-Events" header is advertising that it can process SUBSCRIBE requests and generate NOTIFY requests for all of the event packages listed in the header.
In this Response : Presence

Note – No KPML in this Response header – KPML was sent in Allow-Events header of the INVITE – This indicates that In Band DTMF (RFC 2833) is being used for this call. Implies that far end CUCM Trunk config for DTMF = No Preference or RFC 2833

SIP 180 Ringing Response: Call-Info Header

SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>;tag=abee6e2b-75b0-4537-80f3-7a3a37d0fa55-32557664
…
Call-Info: <sip:10.10.199.250:5060>;method="NOTIFY;Event=telephone-event;Duration=500“

Optional Header in Requests and Responses

The Call-Info header field provides additional information about the caller or callee, depending on whether it is found in a request or response. (In the above example - The Called UA)
method="NOTIFY;Event=telephone-event;Duration=500“ indicates support for NOTIFY based out of band DTMF relay. Duration = time in mS between successive NOTIFY messages

Unsolicited NOTIFY used as a Cisco proprietary way to sent DTMF Out Of Band

SIP 180 Ringing Response: Supported Headers

SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>;tag=abee6e2b-75b0-4537-80f3-7a3a37d0fa55-32557664
…
Supported: X-cisco-srtp-fallback
Supported: Geolocation

Optional Headers
X-Cisco-srtp fallback – proprietary header (can be ignored by other vendors)
Allows an offered SRTP session to fall back to RTP if not supported by both UAs
Geolocation – standardised method to convey geographical location information from one SIP entity to another SIP entity. Configurable on CUCM SIP Trunks

SIP 180 Ringing Response: Content Header

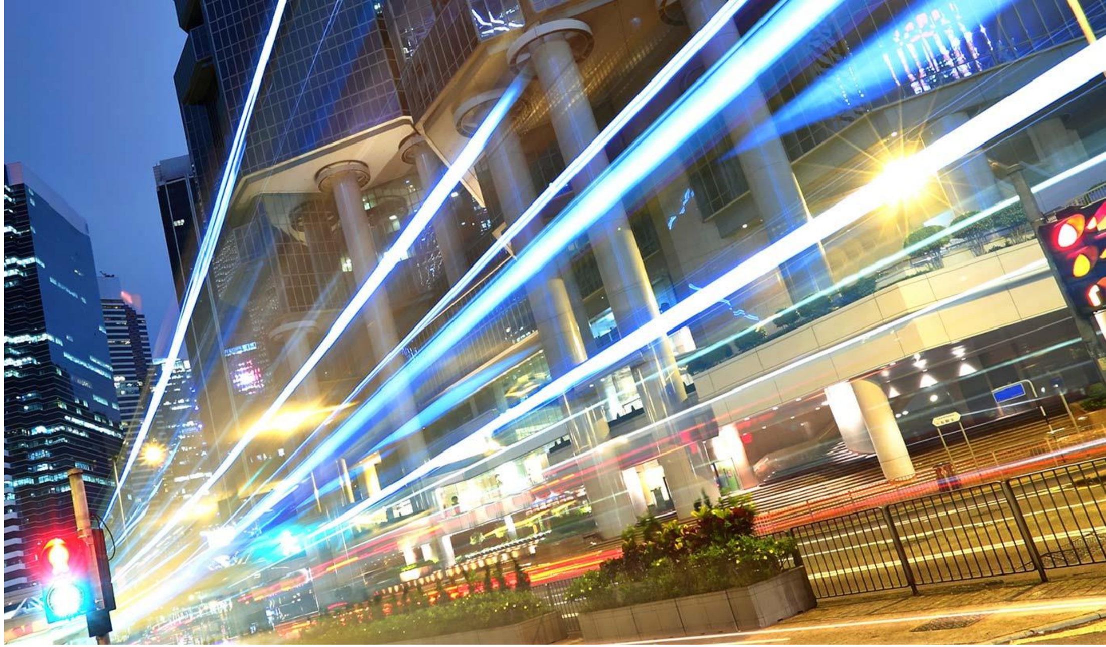
SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.199.251:5060;branch=z9hG4bK3395a5cdb
From: <sip:2002@10.10.199.251>;tag=1b1993ff-121d-4616-8dc5-353990242dfe-32552697
To: <sip:1001@10.10.199.250>;tag=abee6e2b-75b0-4537-80f3-7a3a37d0fa55-32557664
…
Content-Length: 0

Mandatory Header if TCP transport used, Optional if UDP used
The Content-Length header indicates the size of the message-body sent to the recipient in decimal number of bytes.

Message-Body – For example, the Session Description Protocol (SDP) message body. SDP is not usually sent in unreliable 1XX messages. The message body is appended after the Content-Length header.

Agenda

- Why choose SIP for UC Trunking ?
- SIP Basics
- Cisco Unified Border Element (CUBE)
- Deep down into SIP
- **Deep down into SDP**
- CUCM SIP Trunk features and Call functions
- Key Takeaways



Deep Down into SDP

SIP Trunk Signalling - Session Description Protocol (SDP)

The Offer / Answer Model

SDP is the companion protocol of SIP

SDP is used to describe media characteristics; it does not deliver media (for voice and video this is done using the Real-time Transport Protocol (RTP)), but is used to negotiate the media type, format and associated parameters of a multimedia session between endpoints.

SDP is described in RFC 4566

A media characteristics of a session are described by a series of one line fields in an SDP message. Within an SDP message there are three main sections, these detail the session name and purpose, the time the session is active, the media and information needed to receive the media (addresses, ports, formats, etc.). Additional information about bandwidth usage and contact information can also be sent.

Media negotiation using SDP is known as the Offer/ Answer model (described in RFC 3264)

Two key concepts in the Offer / Answer model are the “Early Offer” and “Delayed Offer”

SIP Trunk Signalling and Basic Operation

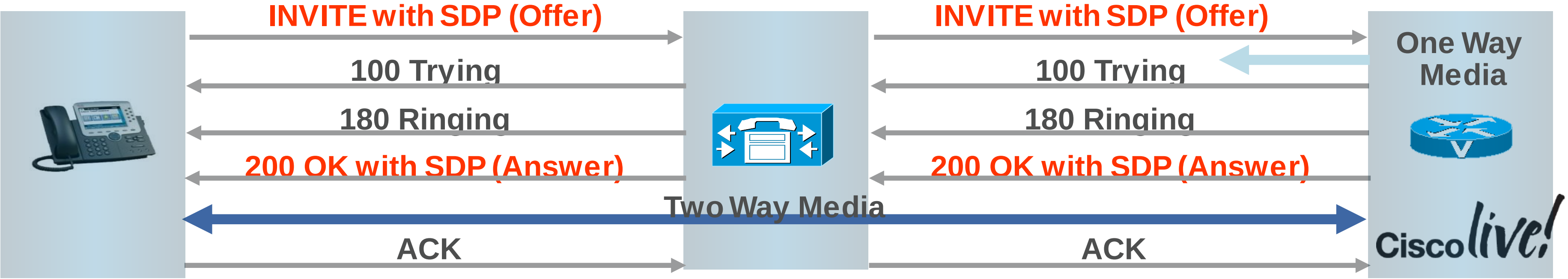
The Offer/Answer Model - SIP Early Offer

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 it in 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 two way media can be established

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



SIP Trunk Signalling and Basic Operation

The Offer/Answer Model - SIP Delayed Offer

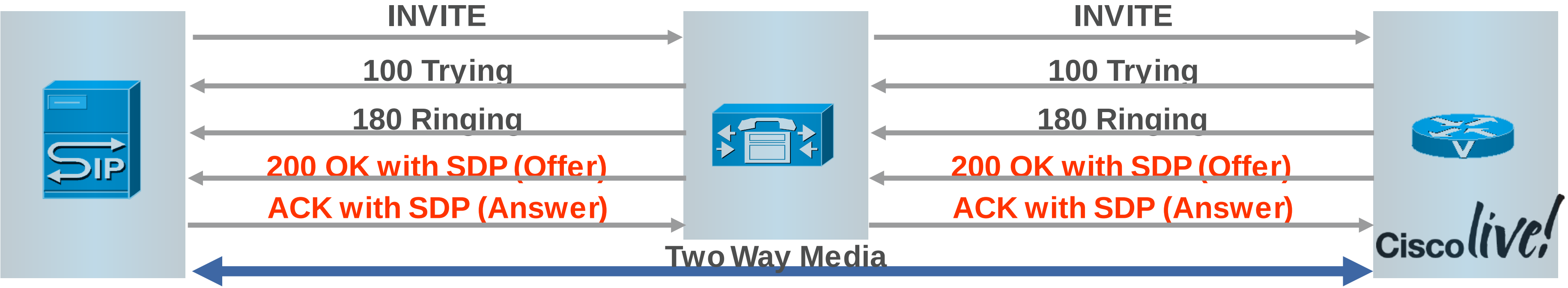
No information about the calling device's media characteristics are sent in the initial SIP INVITE

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 (but not supported by all vendors)

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




Deep Down into SDP
Media Negotiation for Voice Calls

SIP Trunk Signalling

Media Negotiation for Voice Calls – The SDP Offer

```
…..
Content-Type: application/sdp
Content-Length: 337

v=0
o=CiscoSystemsCCM-SIP 2000 1 IN IP4 10.10.199.250
s=SIP Call
c=IN IP4 10.10.199.130
t=0 0
m=audio 16444 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:8 PCMA/8000
a=ptime:20
a=rtpmap:18 G729/8000
a=ptime:20
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

<u>SIP Message Headers</u>

Content-Type : application/SDP

Content-Length : 337 Bytes

<u>SDP Message Body</u>

Describes the media characteristics of the endpoint offering the SDP

Includes :

Endpoint IP address

Codecs supported

UDP Port number for RTP

In Band DTMF support details

Media negotiation for Voice Calls – The SDP Offer

SDP Session Attributes

v=0
o=CiscoSystemsCCM-SIP 2000 1 IN IP4 10.10.199.250
s=SIP Call
c=IN IP4 10.10.199.130
t=0 0

Session Attributes

Some SDP lines are REQUIRED and some are OPTIONAL, but all MUST appear in exactly the order described in RFC 4566

v=	Version =	Version of SDP protocol – currently only version “0”	- Required
o=	Origin =	<Username> <Session-ID> <Session Version> <Network Type> <Address Type> <Unicast Address>	- Required
s=	Session Name =	Text based session name or “s= “	- Required
c=	Connection Data =	<Network Type> <Address Type> <Connection-Address> Defines the media address	- Optional
t=	Timing =	<start-time> <stop-time> 0 0 = permanent session	- Required

Media Negotiation for Voice calls – The SDP Offer

SDP Media Attributes – Voice Codecs Offered

…
c=IN IP4 10.10.199.130---------------------------- Phone's IP address
t=0 0
m=audio 16444 RTP/AVP 0 8 18 101 (UDP port 16444) – (RTP/AVP) – (RTP Payload Type numbers)
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:8 PCMA/8000
a=ptime:20
a=rtpmap:18 G729/8000
a=ptime:20

Media Attributes – voice codecs

m= Media Descriptions = <Media> <Port> <Protocol> <Format> ... -Required

<Media>	"audio"/ "video"/ "text"/ "application"/ "message"
<Port>	The transport port to which the media stream is sent
<Protocol>	The transport protocol – "UDP"/ "RTP/AVP" / "RTP/SAVP"
<Format>	Media format description. The fourth and any subsequent sub-fields describe the format of the media

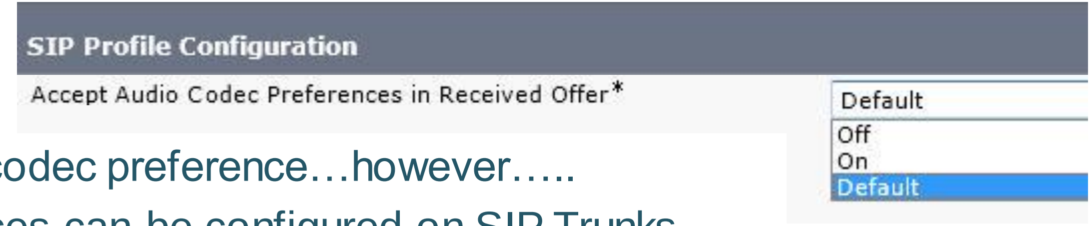
a= Attribute

Media Negotiation for Voice Calls – The SDP Offer

SDP Media Attributes – Voice Codecs Offered

….
m=audio 16444 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:8 PCMA/8000
a=ptime:20
a=rtpmap:18 G729/8000
a=ptime:20

The Codecs (formats) in the Offer must be listed in preference order. The recipient of the Offer <u>should</u> use the codec with the highest preference that is acceptable to it in its Answer



By Default CUCM does not honour codec preference…however…..

Accepting Received codec preferences can be configured on SIP Trunks

a=	Attribute = Attribute lines (in this case media attributes) - Optional May be "session-level" attributes, "media-level" attributes, or both.
a=rtpmap:	<payload type> <encoding name>/<clock rate> [/<encoding parameters>]
a=rtpmap:0 PCMU/8000	Payload Type = 0, Encoding Name = PCMU, Clock Rate – 8000 Hz
a=rtpmap:8 PCMA/8000	Payload Type = 8, Encoding Name = PCMA, Clock Rate – 8000 Hz
a=rtpmap:18 G729/8000	Payload Type = 18, Encoding Name = G729, Clock Rate – 8000 Hz
a=ptime:	<packet time> Time in mS represented by the media in a packet

Media Negotiation for Voice Calls – The SDP Offer

SDP Media Attributes - Audio Direction and DTMF

….
m=audio 16444 RTP/AVP 0 8 18 101 -------- RTP Payload Type (101 for DTMF)
…
a=sendrecv -------- Describes Audio Direction

a=rtpmap:101 telephone-event/8000 --------
a=fmtp:101 0-15 -------- In band DTMF Transport details

Audio Direction

a=sendrecv Media can be sent by this endpoint, media can be received on this endpoint
a=recvonly Media can only be received on this endpoint, it will not send media
a=sendonly Media can only be sent by this endpoint, it will not receive media
a=inactive Media can not be sent to or received from this device (used for "Hold")
If nothing is sent in SDP "a=sendrecv" is assumed

DTMF

a=rtpmap:101 telephone-event/8000 Used for In Band DTMF Transport (RFC 2833)
a=fmtp:101 0-15 DTMF tones (Events 0 through 15 = 0,1,2,3,4,5,6,7,8 ,9,*,#,A ,B,C,D)
a=fmtp: <format> <format specific parameters>
This attribute allows parameters that are specific to a particular format to be conveyed in a way that SDP does not have to understand them.

Cisco live!

SIP Trunk Signalling

Media Negotiation Voice Calls – SDP Answer

….
Content-Type: application/sdp
Content-Length: 228

v=0
o=CiscoSystemsCCM-SIP 2000 1 IN IP4 10.10.199.251
s=SIP Call
c=IN IP4 10.10.199.179
t=0 0
m=audio 28668 RTP/AVP 18 101
a=rtpmap:18 G729/8000
a=ptime:20
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

SIP Message Headers

Content-Type : application/SDP

Content-Length : 228 Bytes

SDP Message Body

Describes the media characteristics of the endpoint answering the SDP offer

Includes :

Endpoint IP address

Codec selected

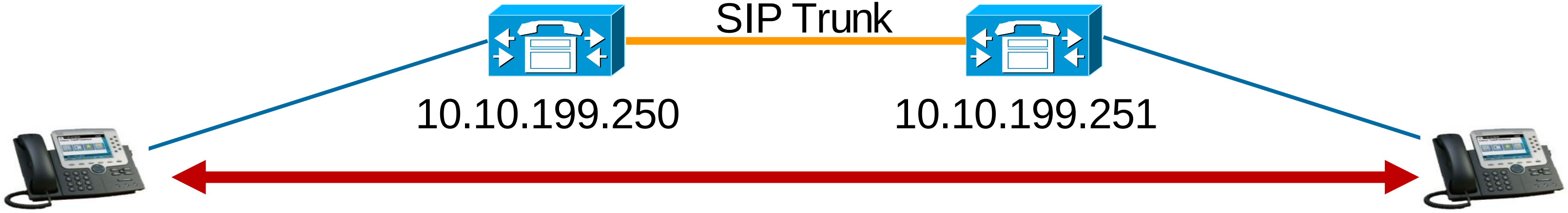
UDP Port number for RTP

In Band DTMF Support details

The codec in the SDP Answer is selected from the codecs sent in the SDP Offer

SIP Trunk Signalling

Media Negotiation Voice Calls – The Negotiated Session



10.10.199.130
RTP UDP Port 16444
G.729 codec
Two way Audio
RFC 2833 DTMF

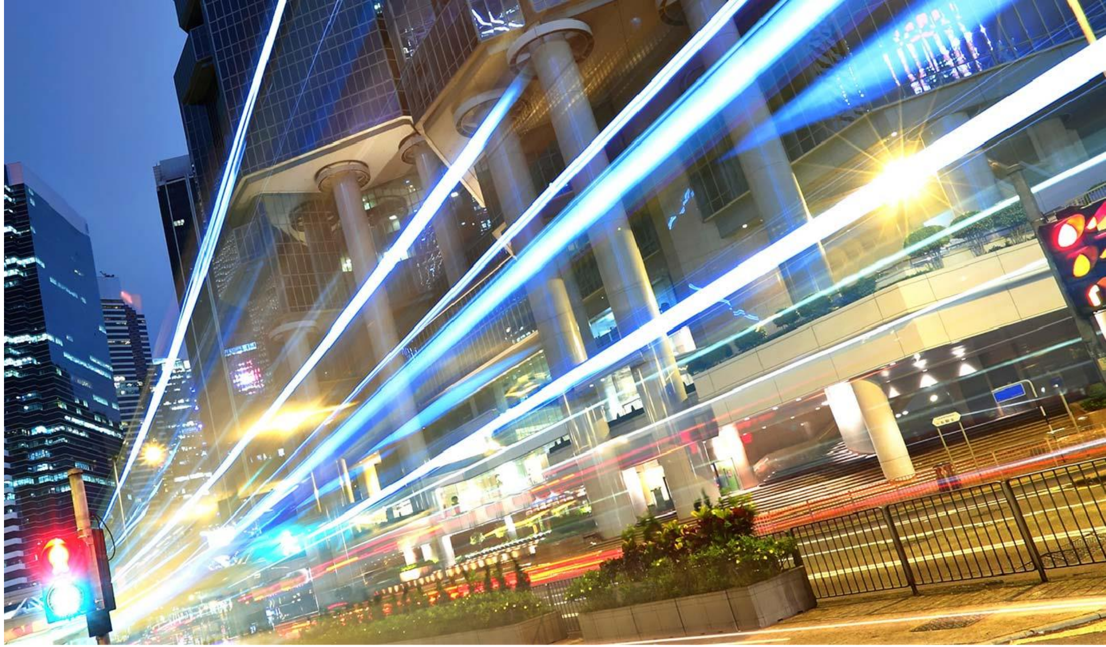
10.10.199.179
RTP UDP Port 28668
G.729 codec
Two way Audio
RFC 2833 DTMF

```
o=CiscoSystemsCCM-SIP  2000 1 IN  IP4  10.10.199.250
c=IN  IP4  10.10.199.130
m=audio  16444 RTP/AVP  18 101
a=rtpmap:18  G729/8000
a=ptime:20
a=sendrecv
a=rtpmap:101  telephone-event/8000
a=fmtp:101  0-15
```

```
o=CiscoSystemsCCM-SIP  2000 1 IN  IP4  10.10.199.251
c=IN  IP4  10.10.199.179
m=audio  28668 RTP/AVP  18 101
a=rtpmap:18  G729/8000
a=ptime:20
a=sendrecv
a=rtpmap:101  telephone-event/8000
a=fmtp:101  0-15
```

Agenda

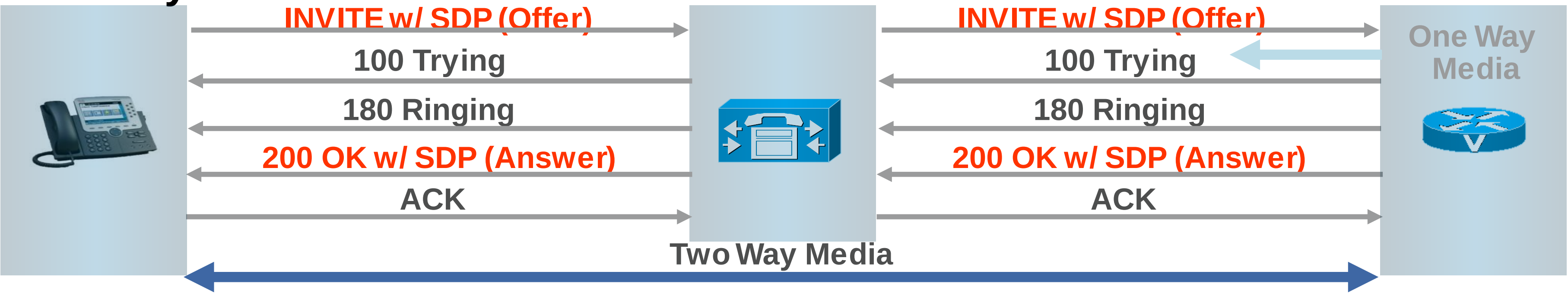
- Why choose SIP for UC Trunking ?
- SIP Basics
- Cisco Unified Border Element (CUBE)
- Deep down into SIP
- Deep down into SDP
- **CUCM SIP Trunk features and Call functions**
- Key Takeaways



CUCM SIP Trunk Features and Call Functions

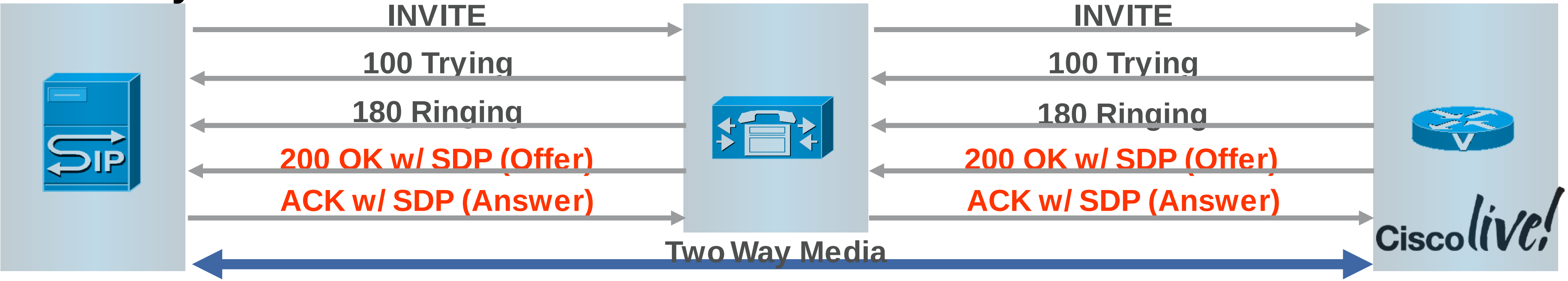
SIP Messaging – Delayed and Early Offer

SIP Early Offer



You can send SDP in 1XX messages but without PRACK these messages are unreliable and SDP must be sent in the next reliable message/response – Often seen – SDP in 18X and OK

SIP Delayed Offer



Cisco live!

CUCM SIP Trunk Signalling – SIP Early Media

Using Provisional Acknowledgement (PRACK) - 1

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).

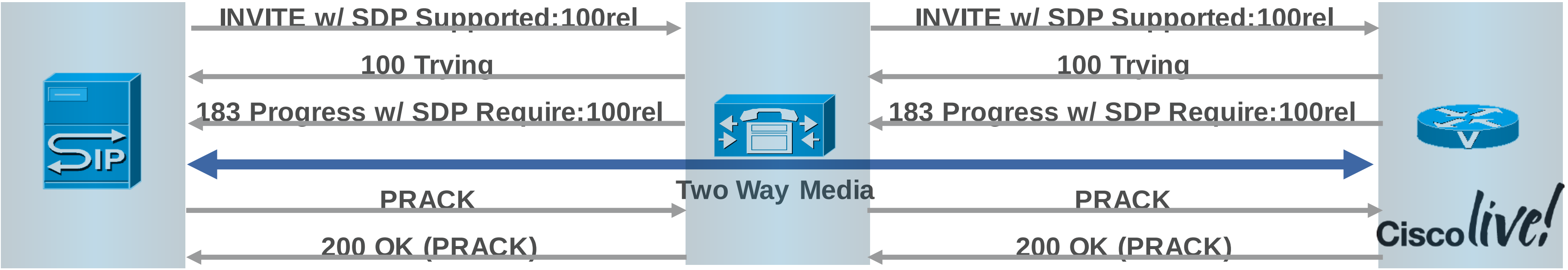
Provisional responses provide information on the progress of the request, but are not sent reliably – so the sender of a provisional response does know that it has been received.

To send an Offer or Answer in 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



CUCM SIP Trunk Signalling – SIP Early Media

Using Provisional Acknowledgement (PRACK) - 2

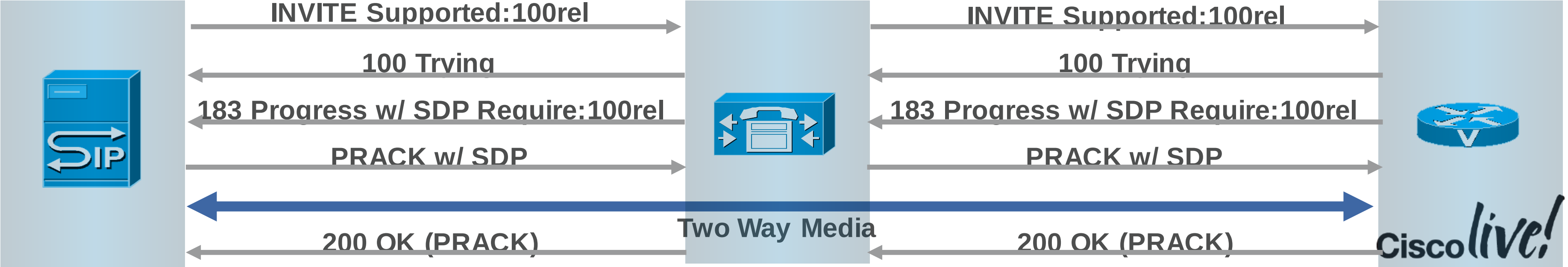
Like final responses, by using PRACK - 1XX messages will be periodically re-sent until their receipt is acknowledged by the receiver by sending a PRACK, which is also acknowledged by the 1XX sender.

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”

Diagram : Delayed Offer with Early Media



CUCM SIP Trunk Signalling

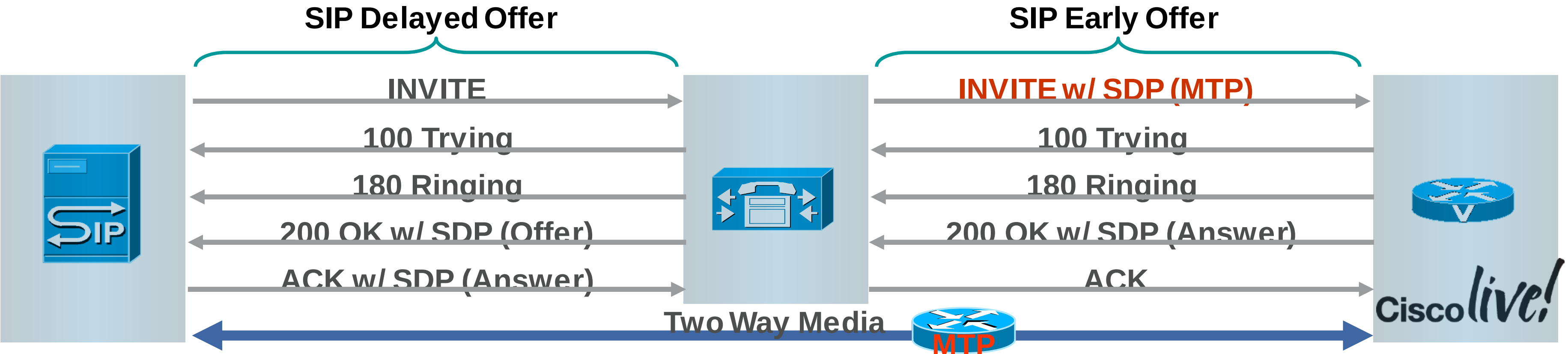
Delayed Offer to Early Offer Calls

Inbound SIP Delayed Offer to Outbound SIP Early Offer

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

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

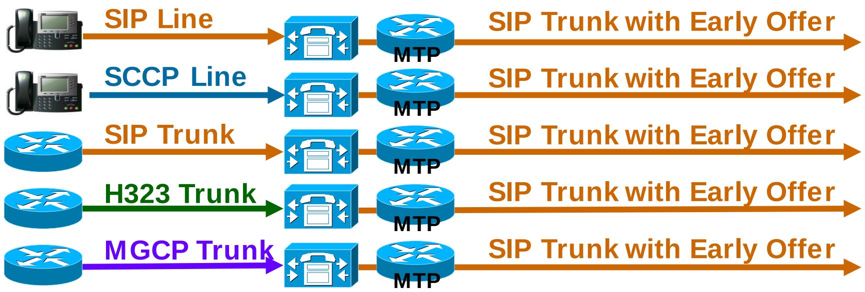
Solution – Insert a Media Termination Point (**MTP**) and use its media characteristics to create the Offer in SDP with the outbound INVITE



CUCM SIP Trunk Signalling

Enabling SIP Early Offer – Method 1 – Pre UC 8.5

SIP Trunk "MTP Required" Checkbox



MTP Recommendation – Always use IOS MTPs

CUCM based MTPs do not have feature parity with software and hardware based IOS MTPs

Using the "MTP Required" option :

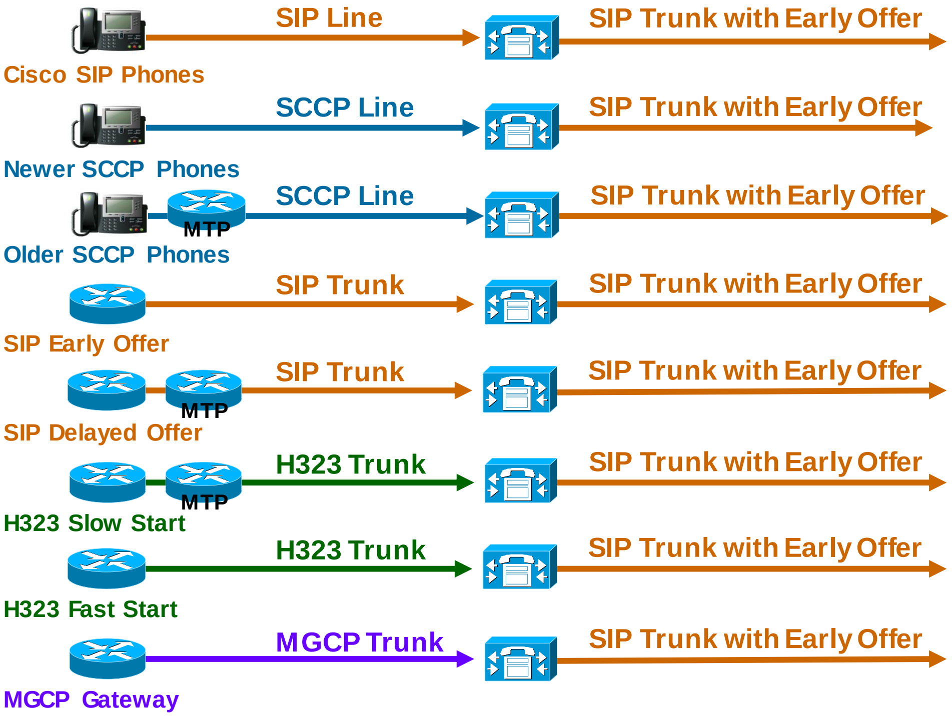
SIP Early Offer Trunks use the Trunk's Media Termination Point (MTP) resources, inserting an MTP into the media path for every outbound (and inbound) call – sending the MTP's IP Address, UDP port number and codec in the SDP body of the initial SIP INVITE instead of those of the endpoint.

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 - The media path is forced to follow the signalling path.

CUCM SIP Trunk Signalling

Enabling SIP Early Offer – Method 2 – UC 8.5+

SIP Profile “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)

SME/CUCM SIP Trunk Signalling – UC 10.5

Best Effort Early Offer

New SIP Profile configuration option

“Early Offer support for voice and video calls – Best Effort (no MTP inserted)”

Recommended configuration for all 10.5+ CUCM SIP Trunks

(Recommended for both CUCM clusters and Session Management Edition clusters)

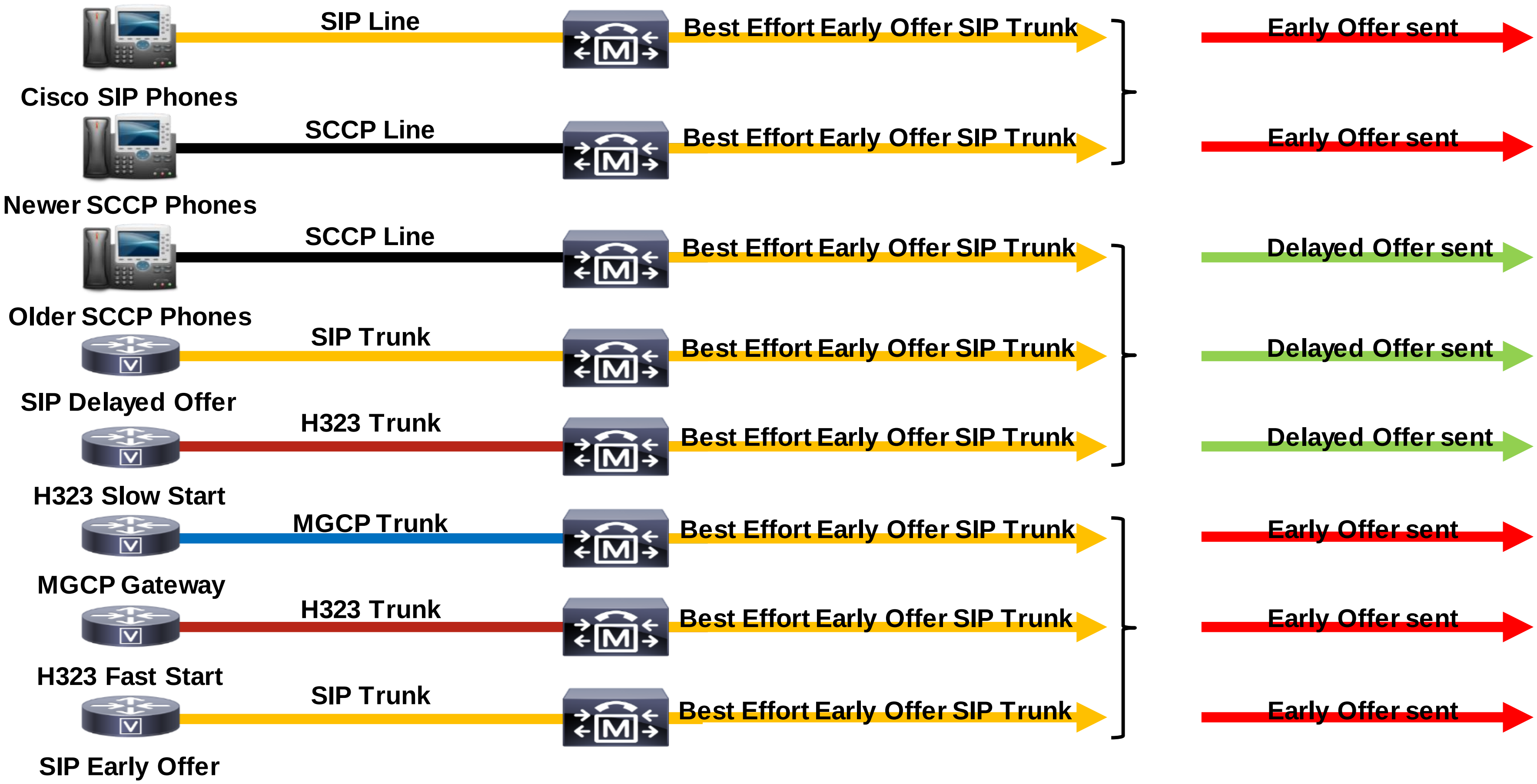
With Best Effort Early Offer – MTPs are never used to create an Offer

An Early Offer is sent only if the media characteristics of the calling device can be determined, if the media characteristics cannot be determined a Delayed Offer is sent.

Best Effort Early Offer is preferred over MTP-less Early Offer in SME clusters

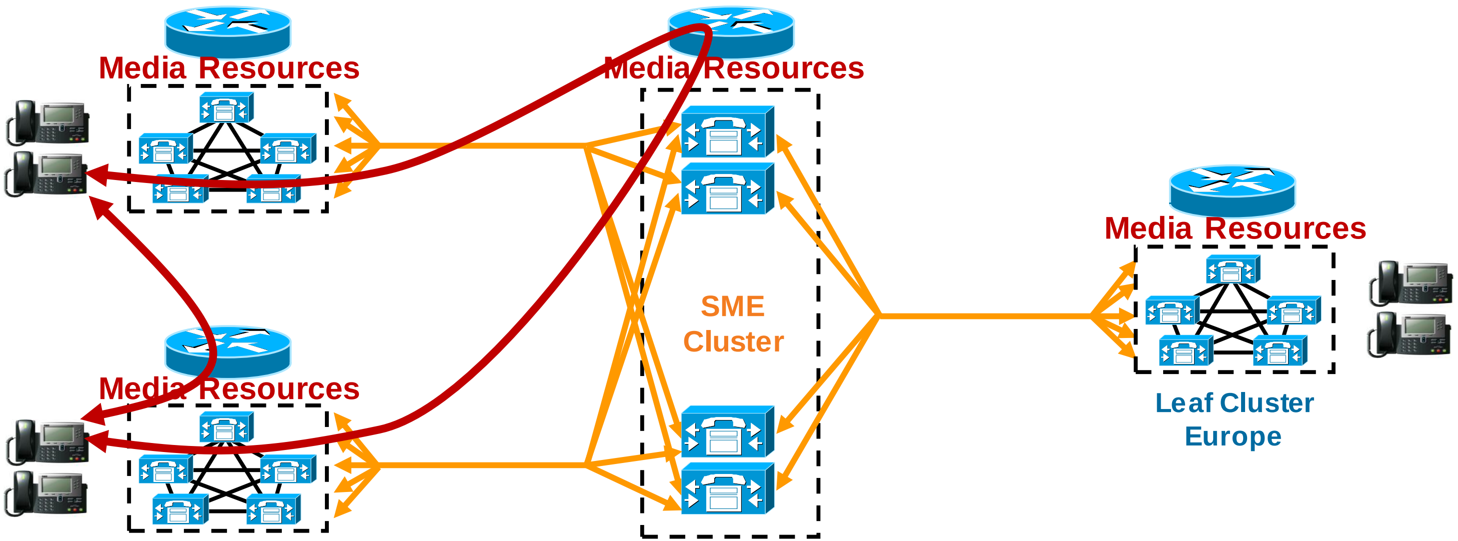
Best Effort Early Offer has the same media transparency effect as MTP-less Early Offer in SME clusters, but the feature is simpler and easier to configure

CUCM 10.5 SIP Trunks – Best Effort Early Offer



Best Effort Early Offer

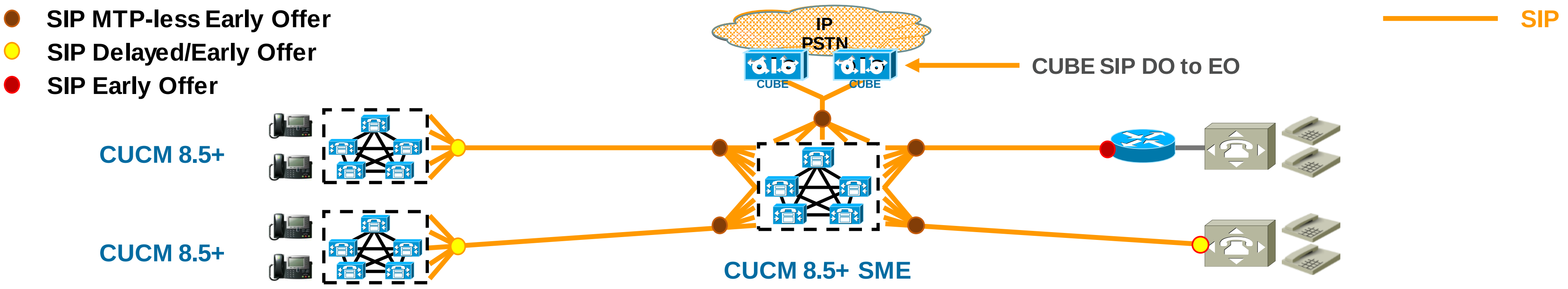
SME Clusters with No Media Resources



Ideally, Media Resources such as MTPs, Transcoders, Music on Hold, Conferencing Resources should never be utilised in the SME cluster – as this entails hair-pinning media via the media resource associated with the SME cluster

This design possible if the SME cluster uses of SIP Trunks only and Best Effort Early Offer Trunk configuration (or for pre UC 10.5 clusters "MTP-less Early Offer" see BRKUCC-2450/Collab SRND)

SIP Trunk Design Recommendations
– UC versions pre 10.5



Leaf Cluster SIP ICT Trunks - Voice, Video and Encryption supported

SIP Delayed Offer/Early Offer (insert MTP if Needed), Run on All Nodes, Multiple Destination Addresses, OPTIONS Ping

SME Cluster SIP ICT Trunks - Voice, Video and Encryption supported

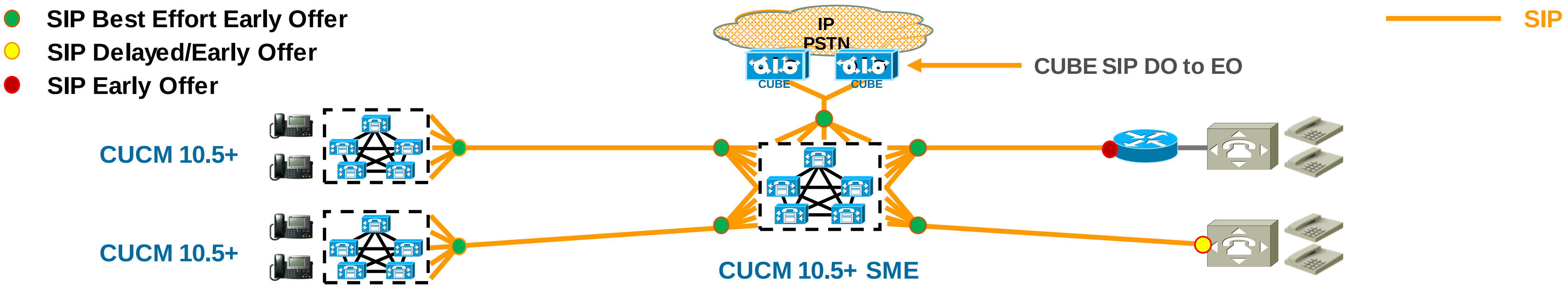
SIP MTP-less Early Offer, Run on All Nodes, Multiple Destination Addresses, OPTIONS Ping, No Media resources can be assigned to Trunks

CUBE/IOS Gateway/ IP PBX SIP Trunks – Typically Voice only, Video and Encryption possible

SIP Delayed Offer/Early Offer (EO commonly used), (EO by sent by the CUBE/ IOS Gateways) OPTIONS Ping, Early Offer usually required by Service Providers (Use CUBE SIP DO to EO)

SIP Trunk Design Recommendations

– UC versions 10.5+



Leaf Cluster SIP ICT Trunks - Voice, Video and Encryption supported
SIP Best Effort Offer/Early Offer, Run on All Nodes, Multiple Destination Addresses, OPTIONS Ping

SME Cluster SIP ICT Trunks - Voice, Video and Encryption supported
SIP Best Effort Early Offer, Run on All Nodes, Multiple Destination Addresses, OPTIONS Ping, Media resources not required

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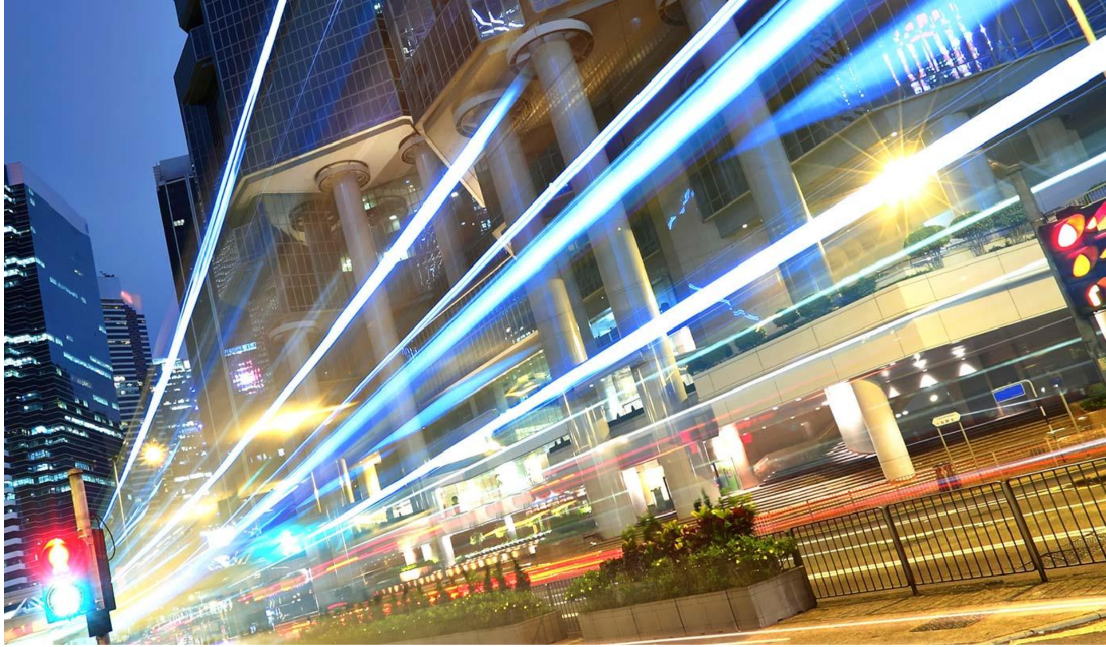

Unified CM SIP Trunk and CUBE

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 and CUBE 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**

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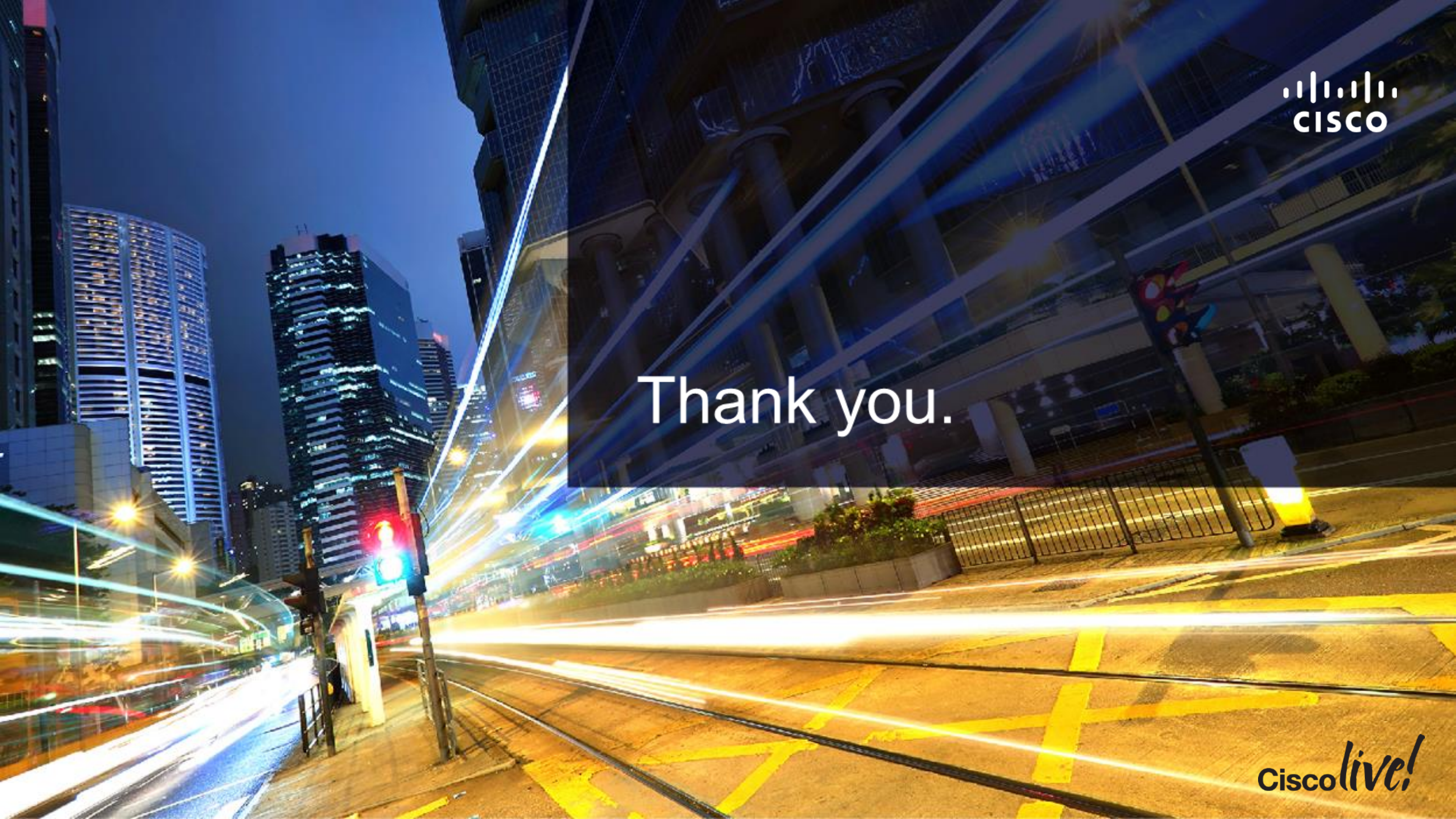
Thank you.

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