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SIP Trunk Design and Deployment in Enterprise UC Networks

BRKUCC-2006

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Objectives of this Session

- Provide a quick overview of SIP basics

- To analyse real SIP traces and to explain how the various headers and SDP information relate to CUCM SIP Trunk configuration
  - We will look at how Voice, Video, Desktop Sharing and Far End Camera Control are set up over a SIP Trunk
  - We will describe the differences between voice and video media negotiation

- To explain how CUCM SIP Trunk features operate and how and when these features can be used

- To leave this audience with a good understanding of CUCM SIP Trunk operation

Agenda

Why choose SIP for UC Trunking?

SIP Basics

- User Agent Clients and servers, Proxy Servers, B2BUAs

Deep down into SIP

- Lots of detail here – I will focus on the relevant areas and cover other areas relatively quickly (slides are for your future reference)

Deep down into SDP

- Lots of detail here – I will focus on the relevant areas and cover other areas relatively quickly (slides are for your future reference)

CUCM SIP Trunk features and Call functions

- Feature explanations and call flows

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

Legend: Yes | Limited support | No

CUCM Trunks to IOS Gateways

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

	H.323	SIP	MGCP
Centralised Provisioning			
Centralised CDR (DS0 Granularity in Unified CM CDR)			
QSIG Tunnelling			
MLPP (Preemption)			
Hook-flash Transfer with Unified CM			
ISDN Overlap Sending			
Accept Audio Codec Preferences in Received Offer			
CUCM 8.5 “Run On All Nodes” feature	3 Active Nodes in a Call Manager Group		1 Active Node in a Call Manager Group
CUCM 8.5 “Up to 16 destination addresses” feature			
Mobility Manager VXML-Based Voice Profile Mgmt			
OPTIONS Ping			
TCL/VXML Apps (e.g. for CVP Integration)			
IPv6, Dual Stack, ANAT			

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 7.1 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
- UC 10.0 SDP Transparency




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

SIP Servers – 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 – 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.

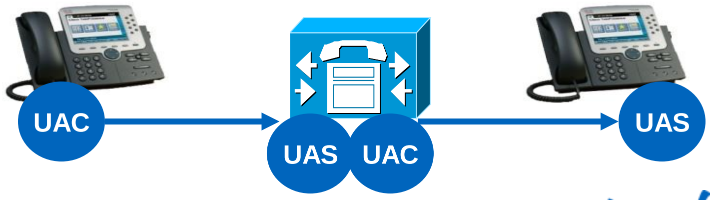
RFC 3621 does not define the specific functionality of a B2BUA….

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 recieved
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

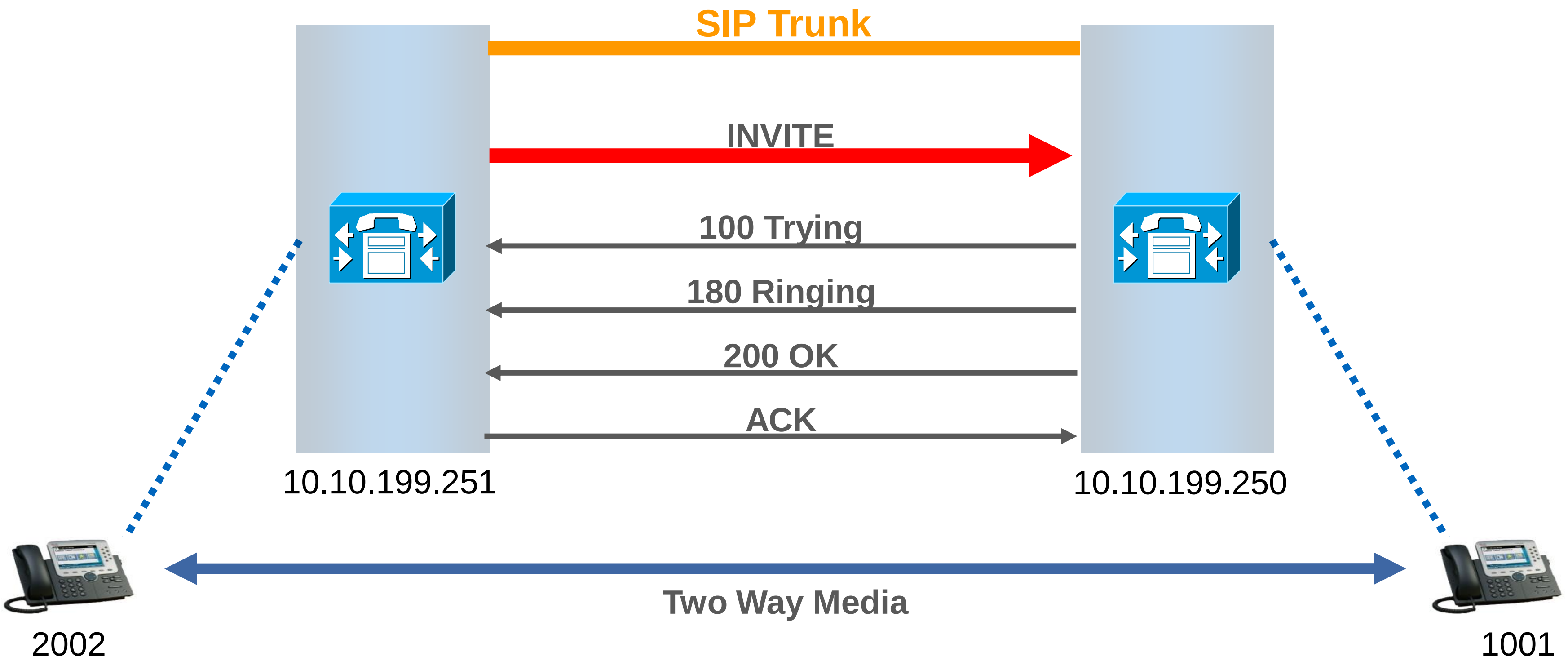
100 Trying - CUCM has received the INVITE
180 Ringing - Destination user agent has received the INVITE, and is alerting the user
183 Session in Progress - Used to send extra information for a call which is still being set up
200 OK - Indicates the request was successful
401 Unauthorised
404 Not Found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




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

<u>Request</u>

INVITE to 1001

<u>SIP Message Headers</u>

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 dialog 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

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)

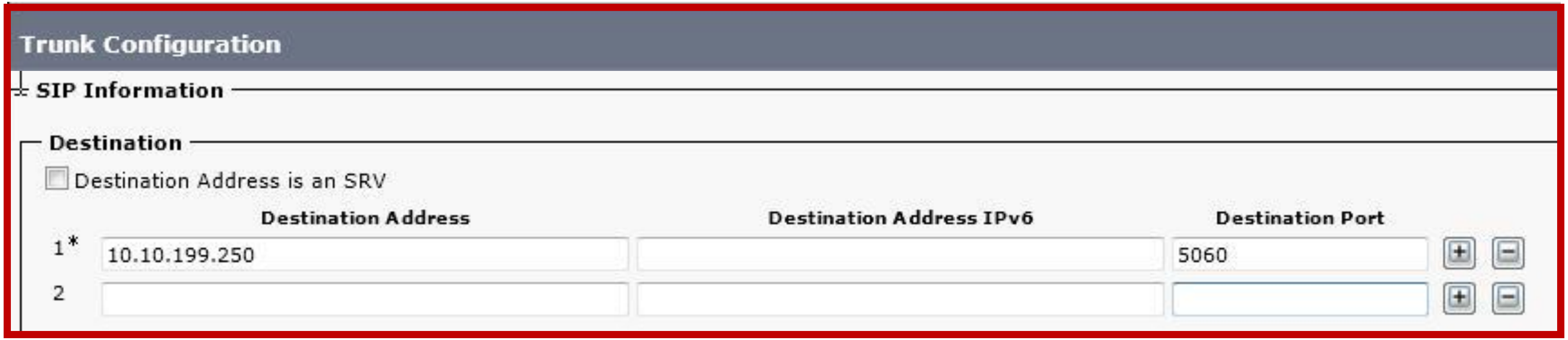
10.10.199.250 – This destination IP address is configured on a outbound CUCM SIP Trunk
5060 – TCP/UDP Port number for SIP signalling
SIP/2.0 – SIP protocol version

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
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

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

SIP INVITE – From and To Headers

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>

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
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

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

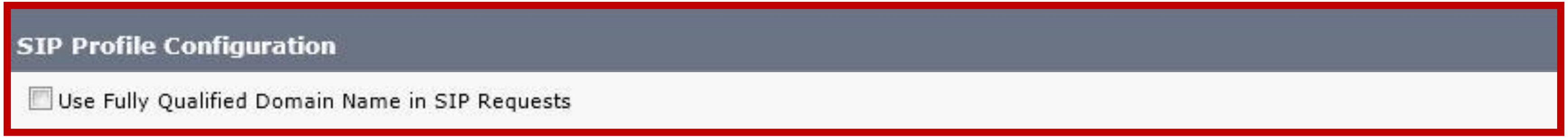
Transactions are tracked by the branch value in the VIA Header
Dialogues are tracked by the Call-ID, From Header tag and To Header tag

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
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>



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

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

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.

SIP INVITE

CUCM Config - P-Asserted-Identity – Asserted Type

Directory Number = 2002

Name = Bob Jones

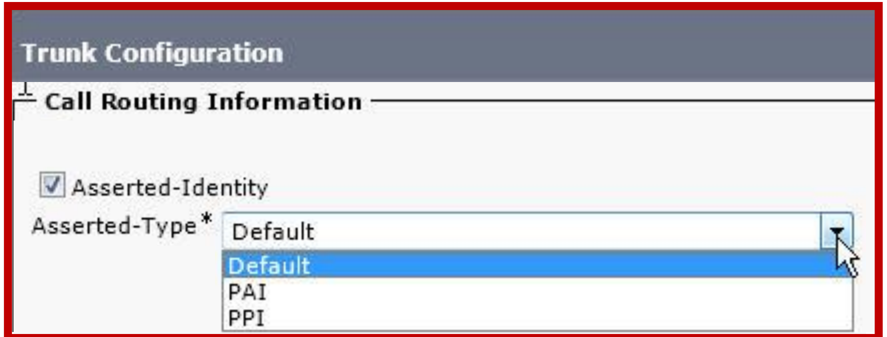
Default = P-Asserted-Identity: “Bob Jones” <sip:2002@10.10.199.251>

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

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

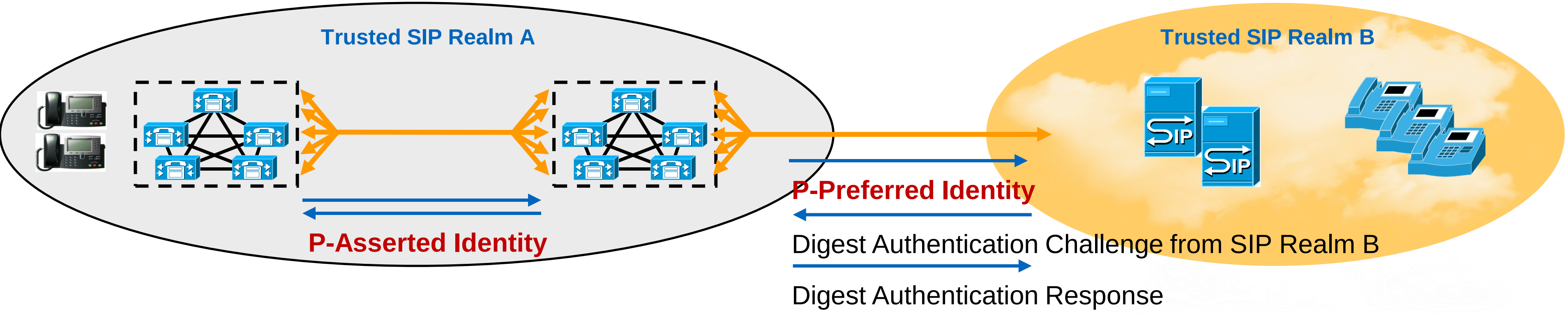
Default : PAI/ PPI value inherited from calling device/ trunk

Cisco Phone identity is Trusted so PAI 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.

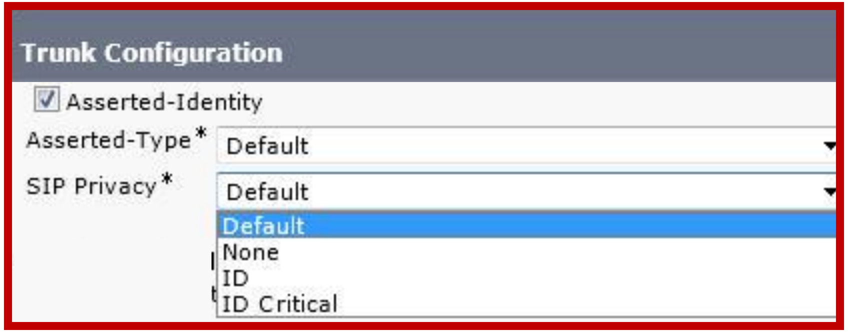
SIP INVITE

CUCM Config - 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

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

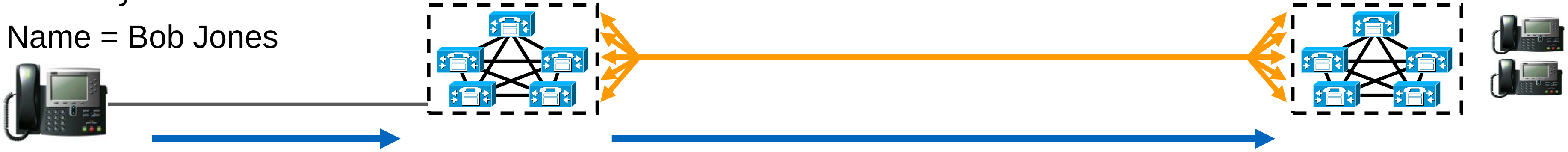
Optional Header - This option is checked by default on a CUCM SIP trunk
Remote Party ID can be used to provide the following services :
Calling identity delivery **From: "Bob Jones" <sip:2002@10.10.199.251>**
Calling identity blocking **From: "Anonymous" <sip:localhost>**
Tracing originator of call **Remote-Party-ID: "Bob Jones" <sip:2002@10.10.199.251>**
Screen=yes indicates the Remote-Party-ID was verified successfully by CUCM
Privacy value can be used to allow/restrict identity in the From Header – Privacy set by the Trunk
PAID and Remote Party-ID are independent mechanisms for the display of identity info –
Non Default PAI Privacy values always take precedence over RPID privacy values

SIP INVITE

CUCM Config - Remote Party ID

Directory Number = 2002

Name = Bob Jones

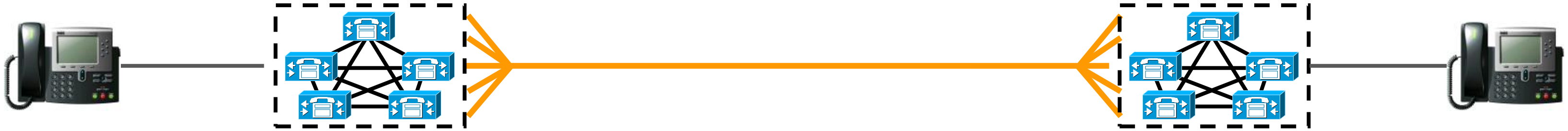


From: “Bob Jones" <sip:2002@10.10.199.251>

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

Remote-Party-ID differs from PAI in that it has no authentication challenge mechanism
Party = Calling/Called
Screen = Yes – ID from CUCM verified device – No if “Screen = No” received over Q931/SIP
Privacy = Name/ URI/ Full/ Off
Privacy values taken from Device or Trunk settings for ID Presentation and Restriction
Trunk Calling/ Connected Presentation/Restriction values over-ride Device settings

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

New CUCM SIP Trunk Features (UC10.0)

SIP Profile Settings – CLID Presentation

Calling Line Identification Presentation applies to inbound Requests and Responses

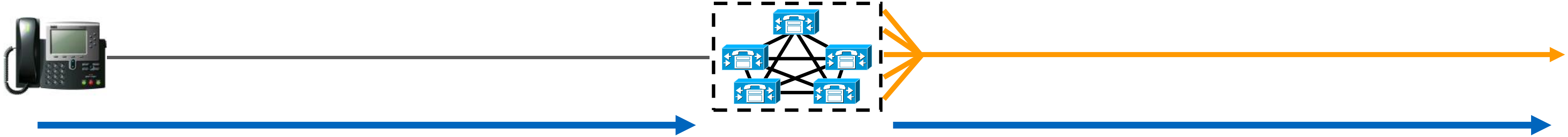
This feature affects

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

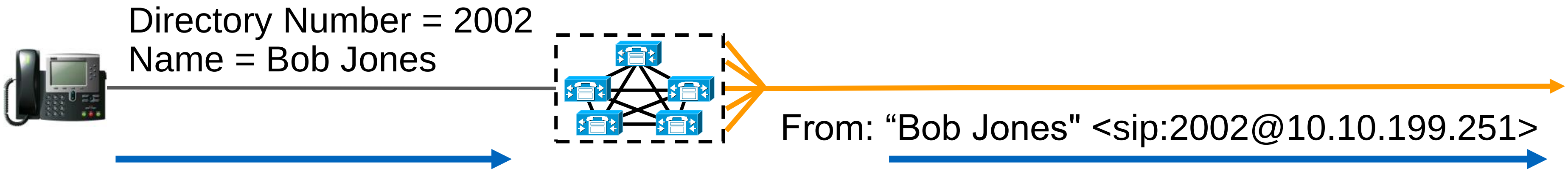
<u>Phone Caller ID Values</u> :

Default = Do not change ID/Name

Allowed

Restricted

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



Outbound Trunk Calling Line / Name Presentation config affects the values in the From header and the Privacy value in the RPID header in Requests

Trunk Configuration

Outbound Calls

Calling Line ID Presentation*	Default
Calling Name Presentation*	Default

If PAI Disabled or PAI Privacy = Default :
Name & Number presentation/ restriction is based on these Trunk settings :

Calling Line ID Presentation* — Default
- Default
- Allowed
- Restricted

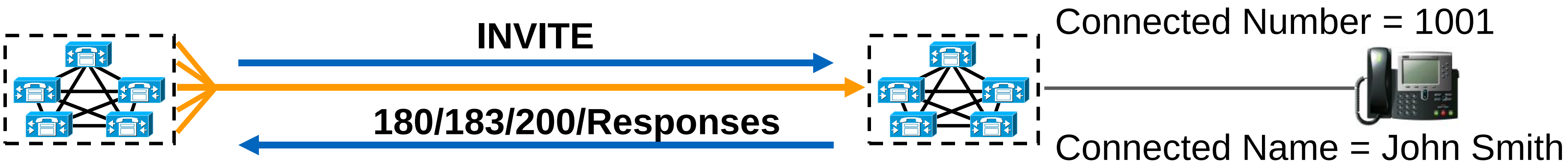
Calling Name Presentation* — Default
- Default
- Allowed
- Restricted

Default - Use values sent by calling UA
Allowed - RPID privacy value = Off
Restricted - RPID privacy value = Name/ URI/ Full

Trunk setting - Overrides Device settings - PAI Privacy value overrides Trunk setting

Cisco live!

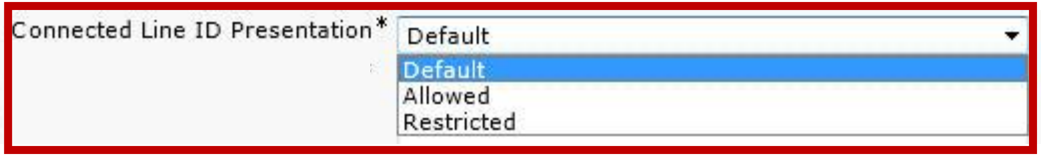
SIP Trunk - Connected Line ID and Connected Name Presentation/Restriction – Inbound Calls



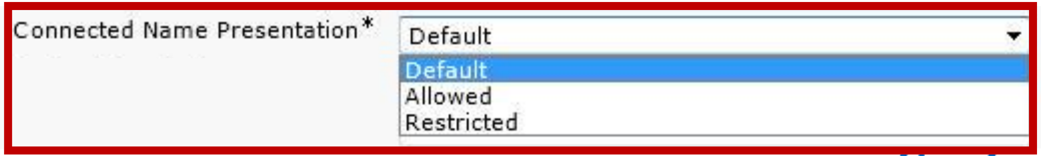
Inbound Trunk - Connected Number / Connected Name Presentation config affects the Privacy value in the RPID header sent in :

180, 183 Responses and 200 Responses

If PAI Disabled or PAI Privacy = Default :
<u>Name & Number presentation/ restriction is based on these Trunk settings</u> :



Default - Use values sent by calling UA
Allowed - RPID privacy value = Off
Restricted - RPID privacy value = Name/ URI/ Full



Trunk setting - Overrides Device settings - PAI Privacy value overrides Trunk setting

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

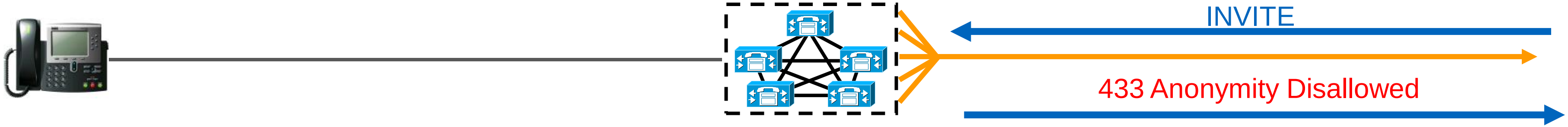
Device Lowest Precedence	Trunk Higher Precedence	RPID	PAI Highest Prec.	Presented User Info
Calling Line and Calling Name Presentation and Restriction setting	Calling Line and Calling Name Presentation and Restriction setting	Privacy field (Set by Trunk Presentation/ Restriction configuration)	Privacy 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
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

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 – This 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>
Date: Wed, 17 Feb 2010 18:37:57 GMT
Call-ID: 8fe4f600-b7c13785-3-fbc712ac@10.10.199.251

Supported: timer,resource-priority,replaces

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

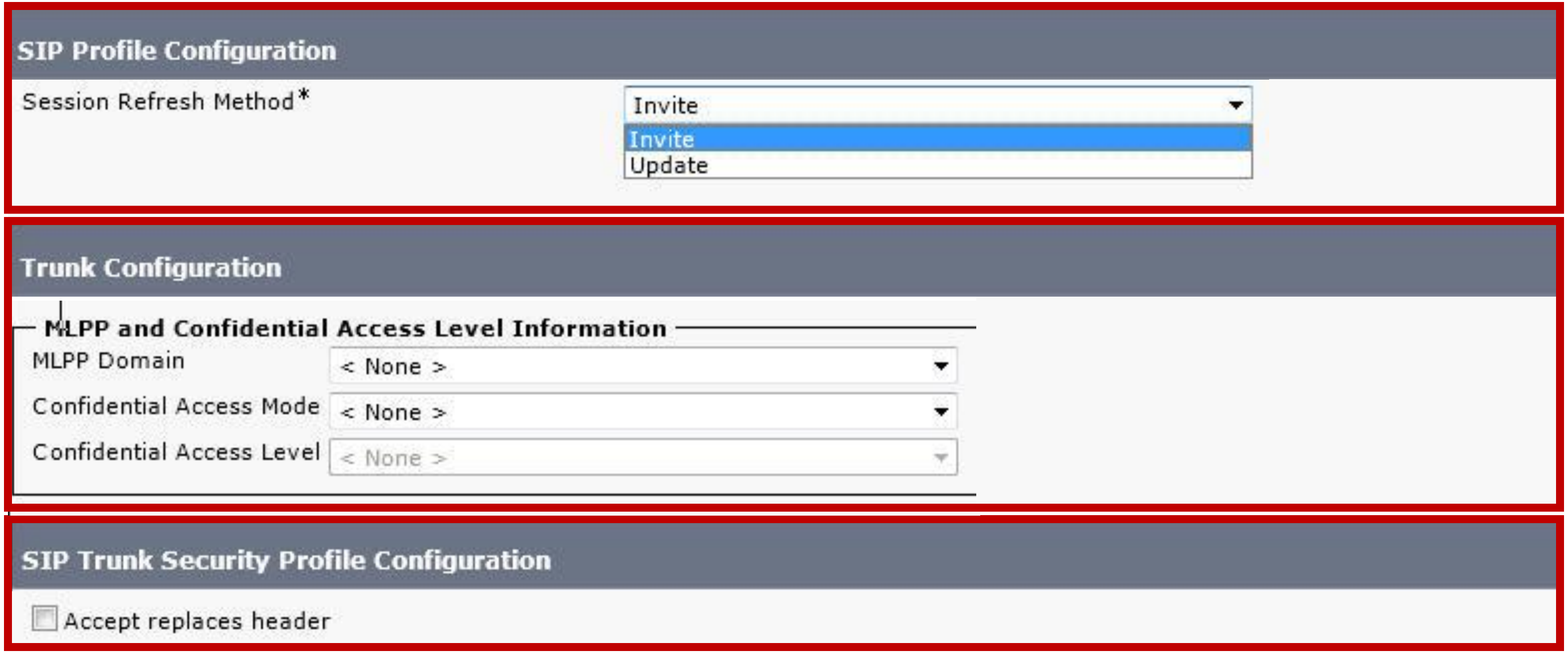
Options Supported : timer, resource-priority, replaces

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.

SIP INVITE – Supported Header

Related CUCM Configuration

Supported: timer, resource-priority, replaces



Note – 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

…
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

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>
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

Minimum Session Expires Header - Optional Header - Can be increased by intermediate Proxies
Support indicated via the “Supported: timer” header option
Used in conjunction with the “Session Expires” header as a session keep-alive mechanism

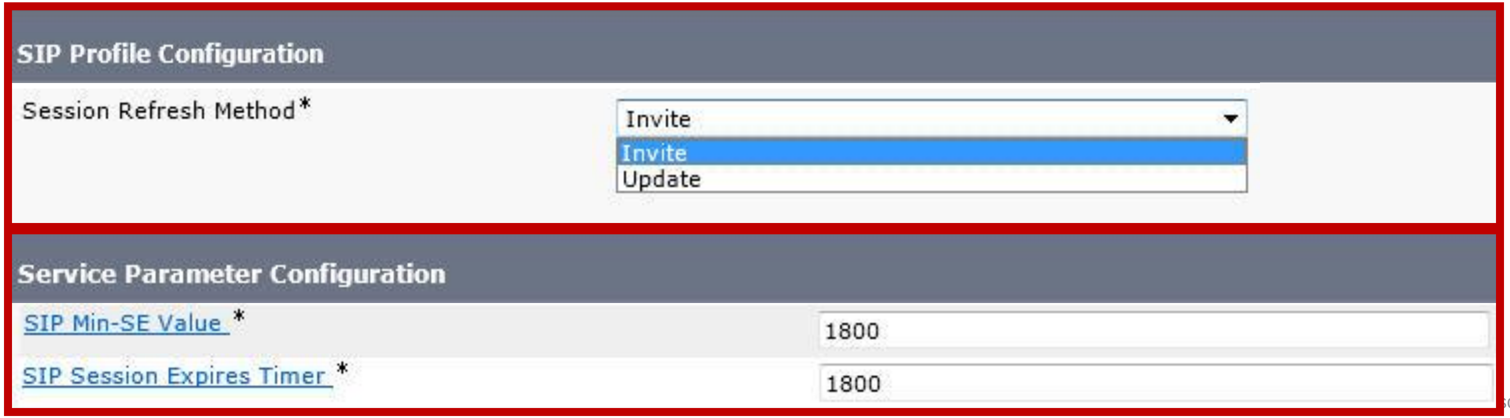
Min-SE: 1800 seconds (30 mins) – Default - Recommended value.
When Min-SE header is not present a default value of 90 seconds is used
Allows the sender to enforce a Minimum session timer when the call traverses multiple Proxies
Sessions can be refreshed with a Re-INVITE or UPDATE request

SIP INVITE – Min-SE Header, Session Expires Header

Related CUCM Configuration

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

“Minimum Session Expires” used with the “Session Expires” as a session keep-alive mechanism
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)

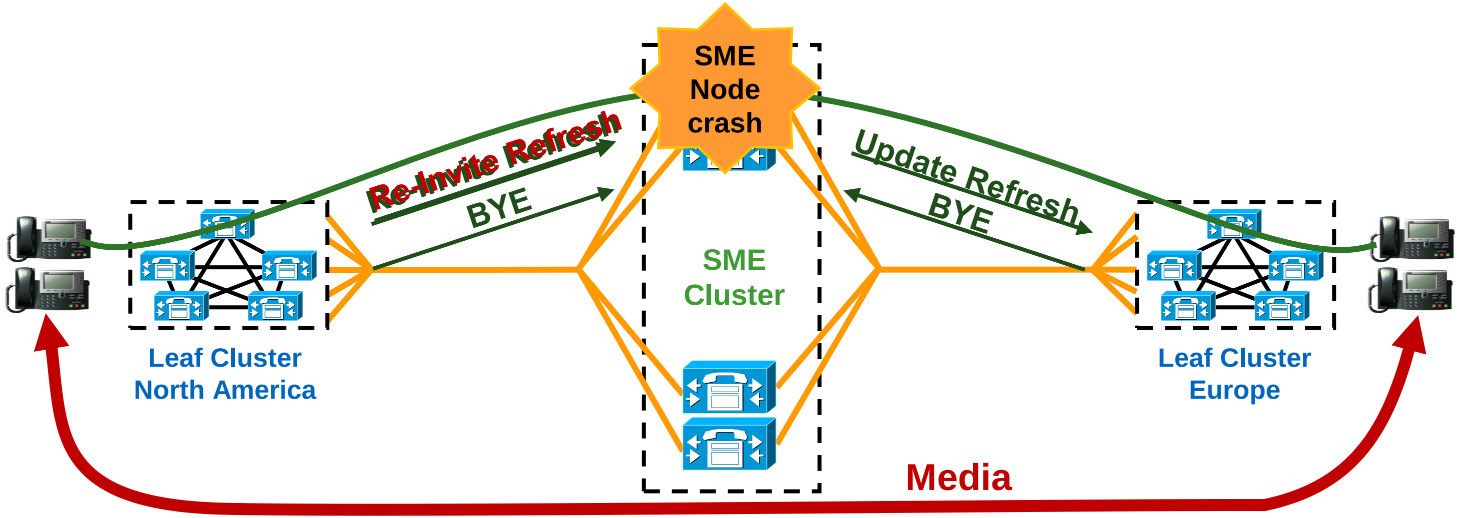


SIP INVITE – Session Expires and Min-SE Headers

CUCM Related Configuration

…
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 INVITE – 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>
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

Optional Header in INVITE Requests

The Expires header field gives the relative time that the message (INVITE in this case) remains valid in seconds. The expiration time of an INVITE does not affect the duration of the actual session that may result from the invitation. (See Session-Expires and Min-SE timers). If CUCM has not received a final answer for the INVITE before this timer expires, CUCM will retry the SIP INVITE up to the configured retry count (6) and if no response cancel the call.

SIP INVITE – Expires Header

Related CUCM Configuration

Expires: 180

Optional Header
The Expires header field gives the relative time that the message (INVITE in this case) remains valid in seconds. The expiration time in an INVITE does not affect the duration of the actual session that may result from the invitation. (See Session-Expires and Min-SE timers). If CUCM has not received an answer before this timer expires, Unified CM cancels the call.

Service parameter value in mS
Default value = 180000 mS = 3 mins
Max value = 3000000 = 5 mins

Used as the primary “no response timeout” timer for SIP INVITE messages

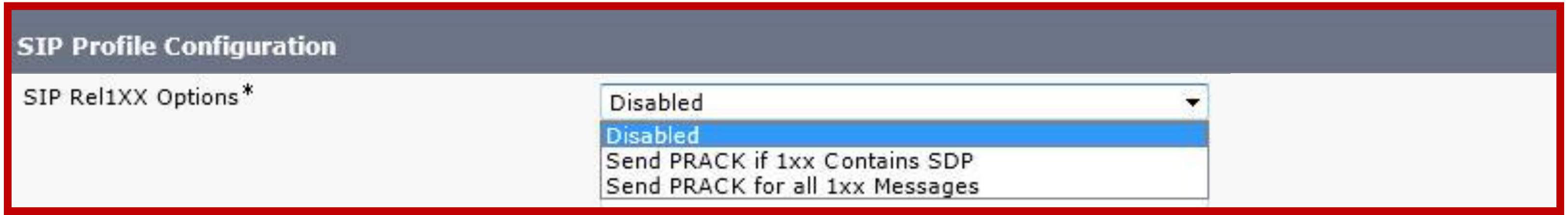
SIP Header Categories :
Methods and Events Supported by the UA

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>
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

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 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>
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

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 (DTMF)

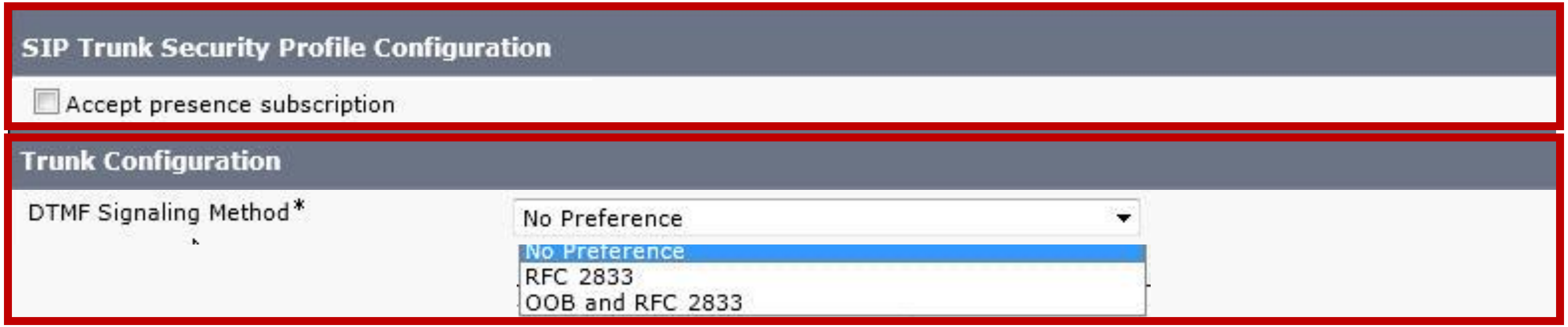
SIP INVITE Allow-Events Header

Related CUCM Configuration

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

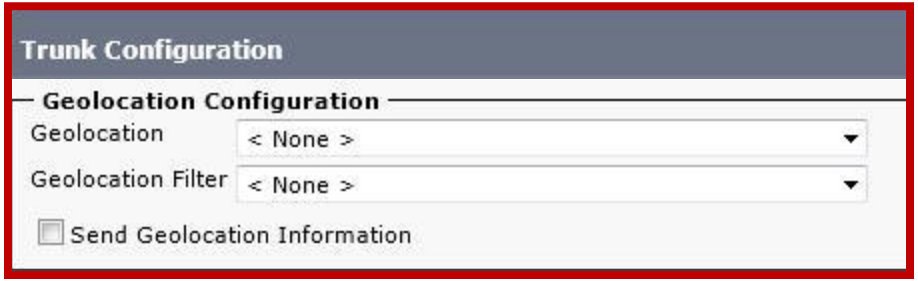
SIP INVITE – Supported Header

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>
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

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 – 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>
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



Optional Header
Geolocation – standardised method to convey geographical location information from one SIP entity to another SIP entity. Configurable on CUCM SIP Trunks – Used for Logical Partioning

Supported but needs to be configured on the SIP Trunk to be used




SIP Header Categories : Cisco and Other Headers

SIP INVITE – Call-Info Header

….

….

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“

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>
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

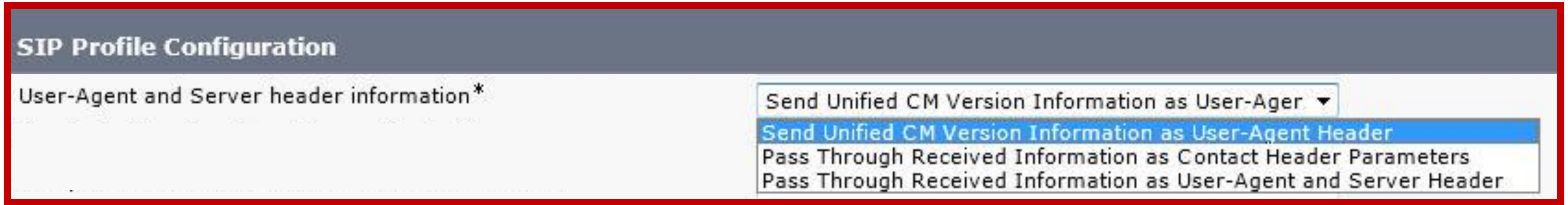
Optional Header
Contains information about the client User Agent originating the request

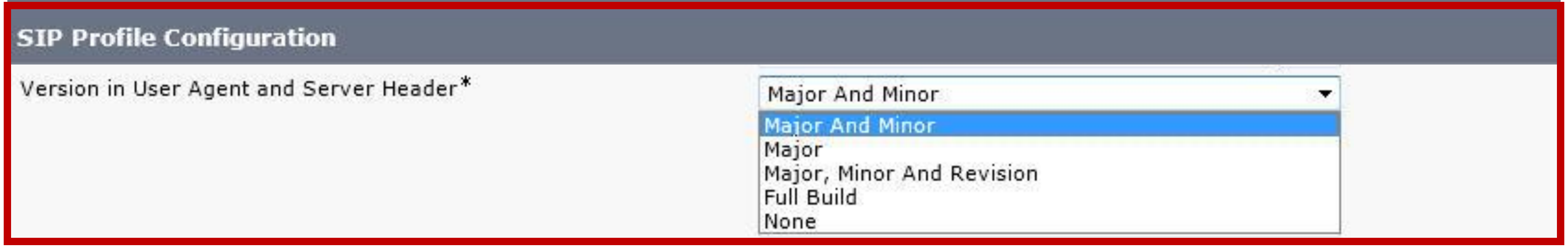
CUCM configurable : SIP Profile “User-Agent and Server header information”

- 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 – User Agent Header

Related CUCM Configuration





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"

SIP Profile "Version in User Agent and Server Header"

SIP INVITE

Cisco GUID Header – Globally Unique Identifier

…..
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

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 Trunk to Trunk calls on SME have different GUIDs for the inbound and outbound calls

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

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

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

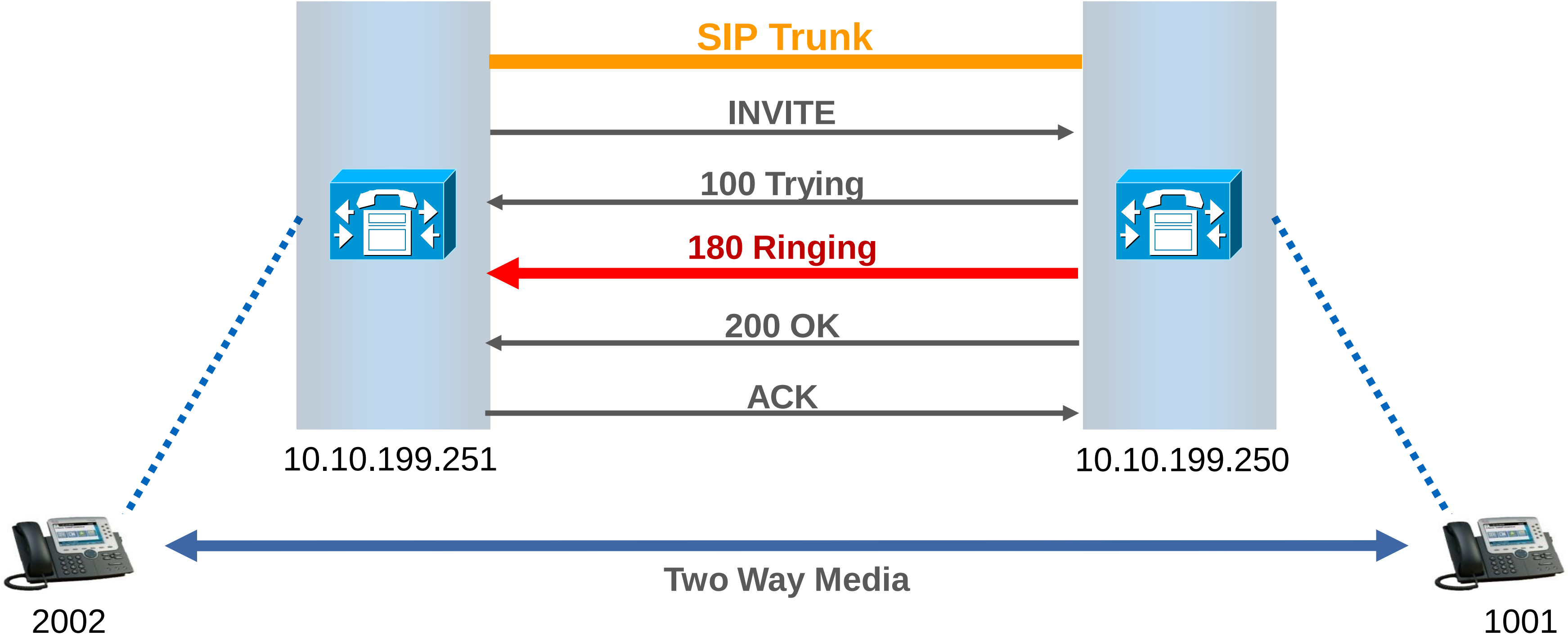
…
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: <sip:10.10.199.251:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 2414147072-3082893189-0000000002-4224127660
Session-Expires: 1800
P-Asserted-Identity: <sip:2002@10.10.199.251>
Remote-Party-ID: <sip:2002@10.10.199.251>;party=calling;screen=yes;privacy=off
Max-Forwards: 70
Content-Length: 0

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.

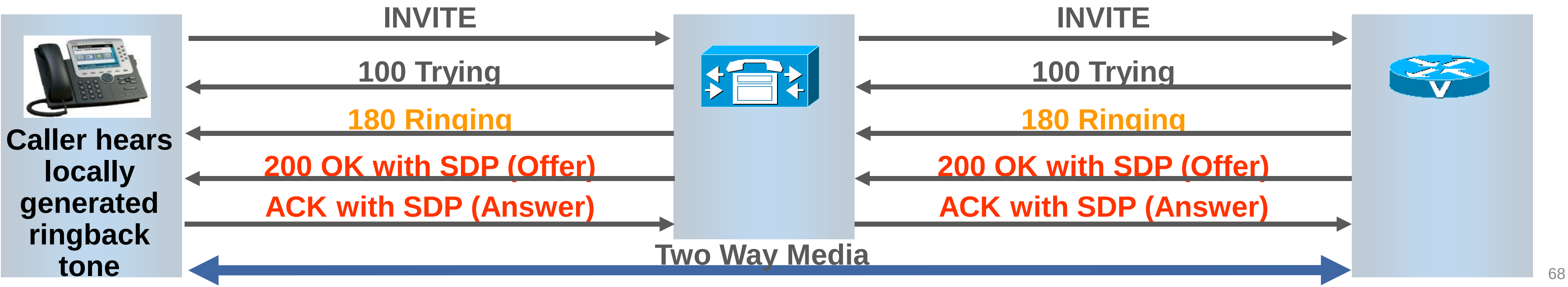
SIP Basics – Typical Call Set Up

SIP Message Exchange



CUCM SIP Trunk Signalling

180 Ringing Response - Ringback



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 of a Request 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
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

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
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 dialog 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
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
Date: Wed, 17 Feb 2010 18:25:39 GMT

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 dialog.
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 dialog

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

SIP 180 Ringing
Identity Headers

….
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

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
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
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

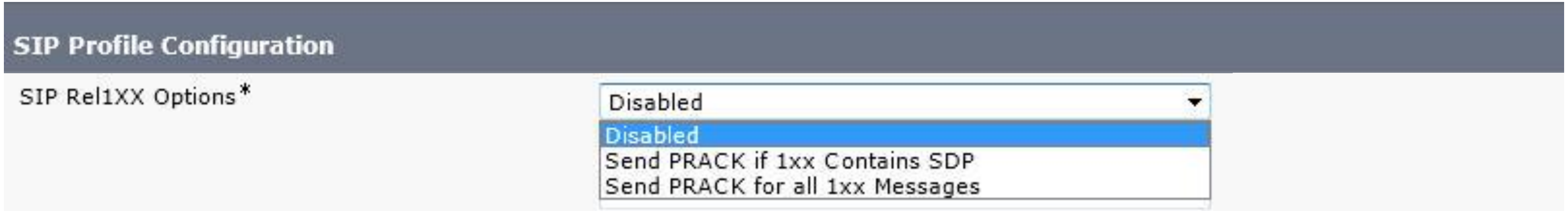
Optional in 1XX Responses (Mandatory in 2XX Responses)
A Contact header field value can contain a display name, a URI with URI parameters, and header parameters
In a Request the contact field contains the address at which the calling UA can be reached
In a Response the contact field contains the address at which the called UA can be reached
With CUCM – a B2BUA – This is the address of the CUCM server - not the phone

SIP 180 Ringing
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
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

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

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
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

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

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
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“

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

Cisco live!

SIP 180 Ringing

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
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

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

Content Header

….
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

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.




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

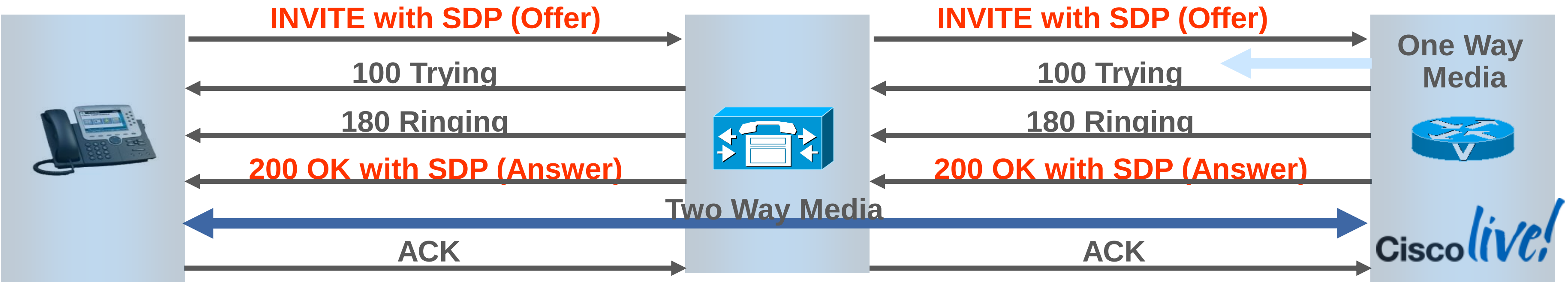
The Offer/Answer Model - SIP Early Offer

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

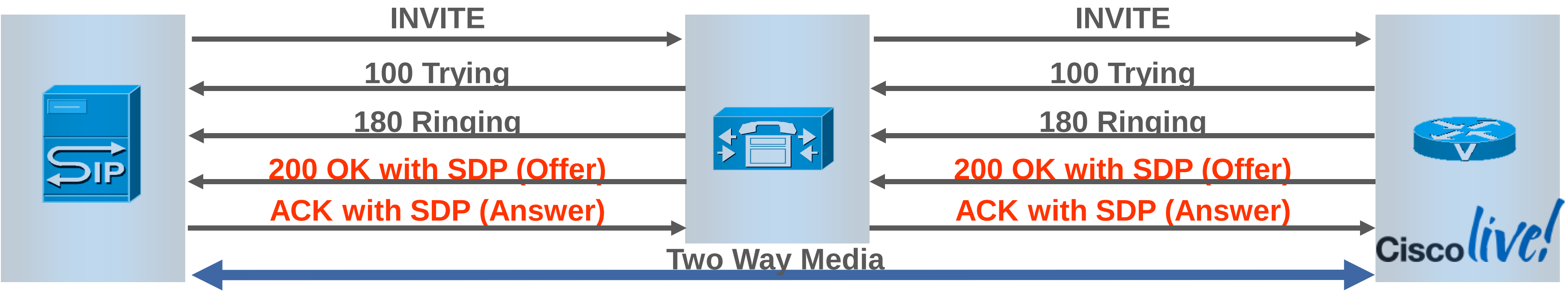
No information about the calling device's media characteristics are sent in the initial SIP INVITE

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…..



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Deep Down into SDP – Voice Call Media Negotiation

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

SIP Message Headers

Content-Type : application/SDP

Content-Length : 337 Bytes

SDP Message Body

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

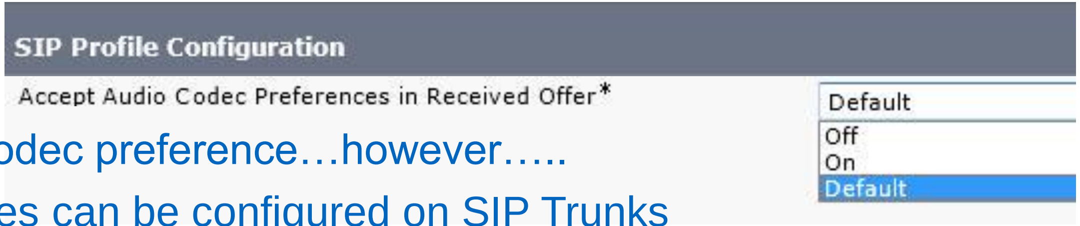
v=	Version =	Version of SDP protocol – currently only version “0”	- Required
o=	Origin =	<Username> <Session-ID> <Session Version> <Network Type> <Address Type> <Unicast Address>	- Required
s=	Session Name =	Text based session name or “s= “	- Required
c=	Connection Data =	<Network Type> <Address Type> <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

....
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 used as "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 us 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.

SIP Trunk Signalling

Media Negotiation Voice calls – SDP Answer

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

SIP Message Headers

Content-Type : application/SDP

Content-Length : 228 Bytes

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

SDP Message Body

Describes the media characteristics of the endpoint answering the SDP offer

Includes :

Endpoint IP address

Codec selected

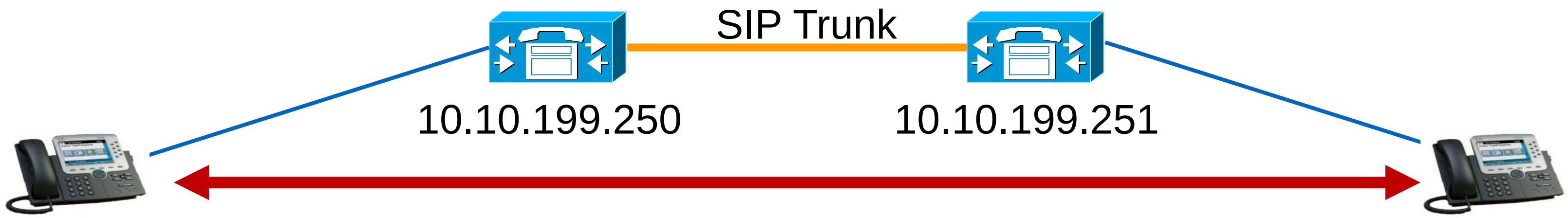
UDP Port number for RTP

In Band DTMF Support details

The codec sent 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




Deep Down into SDP – Video Call Media Negotiation

SIP Trunk Signalling

Video Calls

Video is fundamentally different from voice in the sense that there are many use cases where asymmetric media flows are desirable –

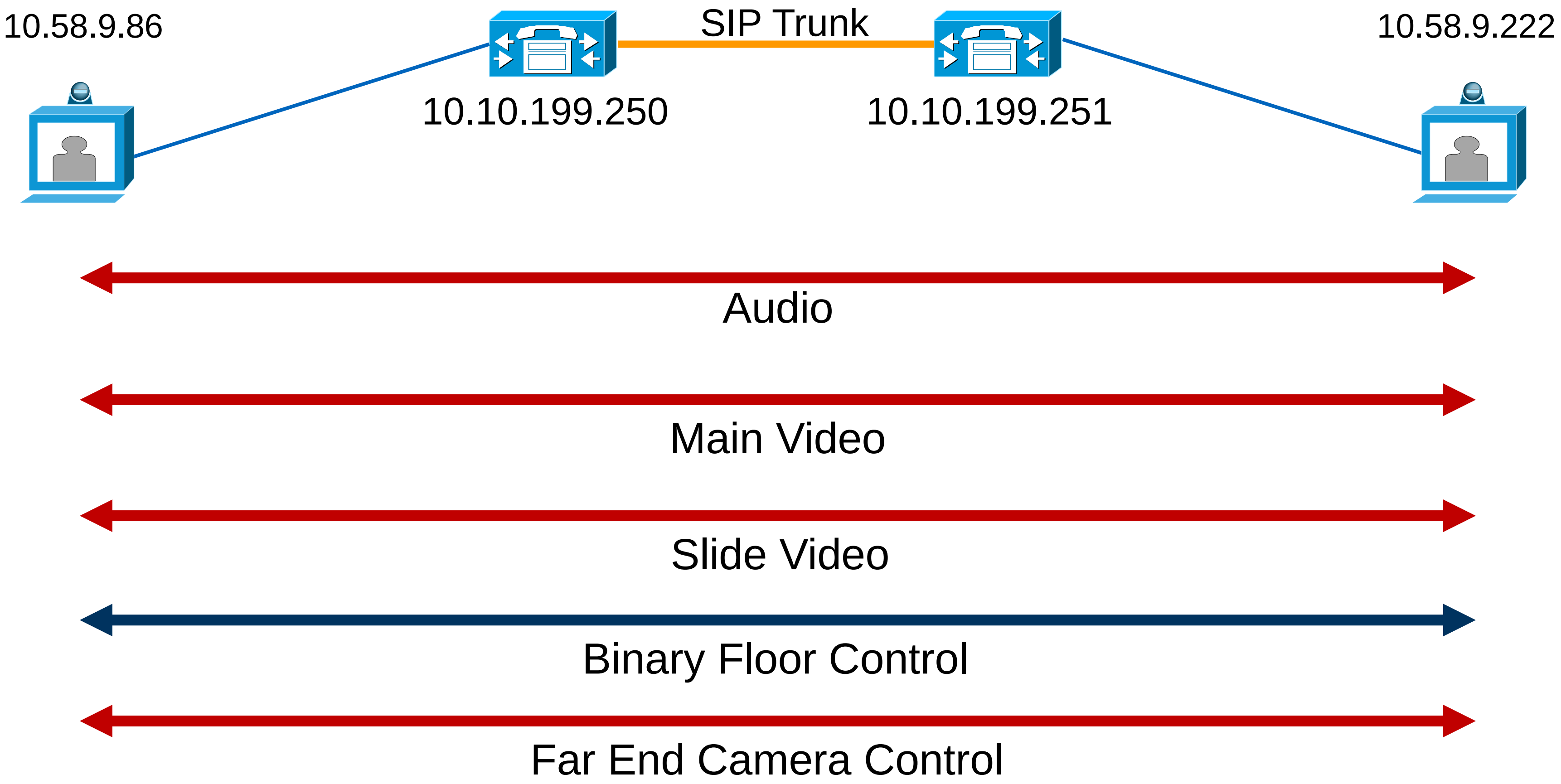
For example, broadband services where the upload and download speeds are different – often by an order of magnitude.

Also because encoding video is more CPU intensive than decoding video - Video endpoints can typically decode at a higher resolution than they can encode.

Because of these requirements – the video codec capabilities sent in an SDP Offer and Answer should be considered as the receive capabilities of the respective endpoints rather than the negotiated capabilities in common with both devices

SIP Trunk Signalling and Basic Operation

Voice and Video call with BFCP and FECC



SIP Trunk Signalling

Video calls – SDP Offer – Detail - Video

v=0
o=CiscoSystemsCCM-SIP 161095 1 IN IP4 10.58.9.6
s=SIP Call
b=TIAS:6000000
b=AS:6000
t=0 0
m=audio 16444 RTP/AVP 102 103 104 9 105 106 0 8 101
c=IN IP4 10.58.9.86
b=TIAS:64000
….attributes of multiple audio codecs in the offer

m=video 16446 RTP/AVP 98 99
c=IN IP4 10.58.9.86
b=TIAS:6000000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=428016;packetization-mode=1;max-mbps=245000;max-fs=9000;max-cpb=200;max-br=5000;max-rcmd-nalu-size=3456000;max-smbps=245000;max-fps=6000
a=rtpmap:99 H263-1998/90000
a=fmtp:99 QCIF=1;CIF=1;CIF4=1;CUSTOM=352,240,1
a=rtcp-fb:* nack pli
a=rtcp-fb:* ccm tmmbr

SIP Trunk Signalling

Video Calls – SDP Offer – Bandwidth in this Offer

o=CiscoSystemsCCM-SIP 161095 1 IN IP4 10.58.9.6
s=SIP Call
b=TIAS:6000000 **Transport Independent Application Specific bandwidth (RTP) in bits/sec**
b=AS:6000 **Application Specific bandwidth (RTP/UDP/IP) in kbps**
t=0 0
m=audio 16444 RTP/AVP 102 103 104 9 105 106 0 8 101
b=TIAS:64000
….attributes of multiple audio codecs in the offer
…
m=video 16446 RTP/AVP 98 99
b=TIAS:6000000

For this endpoint – the maximum media stream bandwidths that can be received :

= 6 Mbps for all voice and video streams including UDP and IP headers (AS session bandwidth)
= 64kbps for voice RTP traffic – not including UDP and IP headers (TIAS audio)
= 6 Mbps for video RTP traffic – not including UDP and IP headers (TIAS video)

The bandwidth values in the SDP Answer do not have to be the same

SIP Trunk Signalling - Video Calls

SDP Offer – H.264 and H.263 Video Codecs

…
m=video 16446 RTP/AVP **98 99**
c=IN IP4 10.58.9.86
b=TIAS:6000000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=428016;packetization-mode=1;max-mbps=245000;max-fs=9000;max-cpb=200;max-br=5000;max-rcmd-nalu-size=3456000;max-smbps=245000;max-fps=6000
a=rtpmap:99 H263-1998/90000
a=fmtp:99 QCIF=1;CIF=1;CIF4=1;CUSTOM=352,240,1
a=rtcp-fb:* nack pli
a=rtcp-fb:* ccm tmmbr

The Codecs (formats) in the Offer must be listed in preference order. H.264 preferred over H.263

The video capabilities sent in the SDP body should be considered as the receive capabilities of the sending endpoint.

The codecs used by video streams are more complex than audio codecs, particularly for H.264 which is a more recent codec standard that offers significant improvements when compared with H.263. Today H.263 is considered to be a legacy codec, but can be used as a “lowest common denominator” codec between various video endpoints, albeit at lower quality and resolution for a given bandwidth than H.264

SIP Trunk Signalling

Video Calls – SDP Offer – H.264 Video Codec

a=rtpmap:98 **H264**/90000
a=fmtp:98 **profile-level-id=428016;packetization-mode=1;max-mbps=245000;max-fs=9000;max-cpb=200;max-br=5000;max-rcmd-nalu-size=3456000;max-smbps=245000;max-fps=6000**

profile-level-id=428016

The Profile-Level-ID describes the minimum set of features/capabilities that are supported by this endpoint

packetization-mode=1
max-mbps=245000
max-fs=9000
max-cpb=200
max-br=5000
max-rcmd-nalu-size=3456000
max-smbps=245000
max-fps=6000

These parameters describe the features and capabilities beyond those of the profile-level-id that are supported by this endpoint

SIP Trunk Signalling

Video calls – SDP Offer – H.264 Video Codec - RTCP

…
m=video 16446 RTP/AVP 98 99
…
…

a=rtcp-fb:* nack pli	“rtcp-fb”	RTP Control Protocol (RTCP) - Feedback
	“*”	RTCP-Feedback for any of the offered video codecs
	NACK	– Negative Acknowledgement – indicates the loss of one or more RTP packets
	PLI	– Picture Loss Indication
a=rtcp-fb:* ccm tmmbr	“rtcp-fb”	RTCP-Feedback
	“*”	RTCP-Feedback for any of the offered video codecs
	“ccm”	indicates support of codec control using RTCP feedback messages
	"tmmbr"	indicates support of the Temporary Maximum Media Stream Bit Rate Request/Notification

RTCP is used for video rate adaption when congestion/ packet loss encountered

SIP Trunk Signalling

H.264 Video Codec - Offer/Answer Compared

Offer **H.264 and H.263 Offered**

a=rtpmap:98 H264/90000
a=fmtp:98 **profile-level-id=428016;packetization-mode=1;max-mbps=245000;max-fs=9000**;max-cpb=200;
max-br=5000; **max-rcmd-nalu-size=3456000;max-smbps=245000**;max-fps=6000
a=rtpmap:99 H263-1998/90000
a=fmtp:99 QCIF=1;CIF=1;CIF4=1;CUSTOM=352,240,1

Answer **H.264 selected – Symmetric Attributes - Asymmetric attributes**

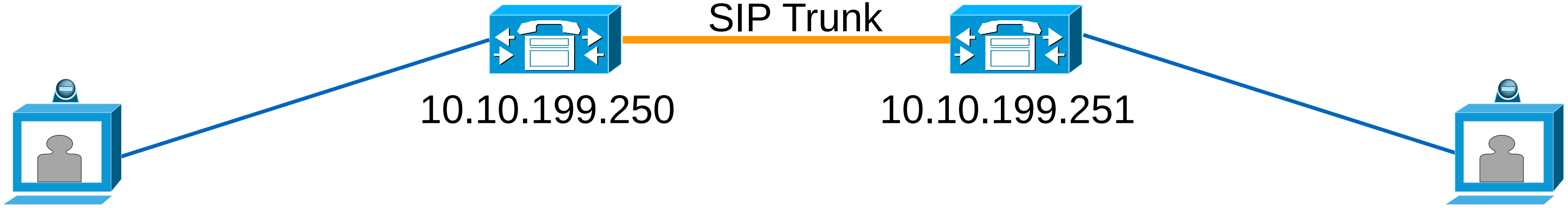
a=rtpmap:98 H264/90000
a=fmtp:98 **profile-level-id=428016;packetization-mode=1**;**max-mbps=108000;max-fs=3600**;max-cpb=200;
max-br=5000; **max-rcmd-nalu-size=1382400;max-smbps=108000**;max-fps=6000

For the selected H.264 Codec :

- The Profile-level-IDs are the same for both endpoints (428016 = Baseline Profile, Level 2.2)
- The Packetisation Mode (=1) is the same for both endpoints
- Note that each device supports different receive values for Max-Macroblocks/second, Max Frame Size, Max Recommended NALU Size, Max Static Macroblock processing rate.

SIP Trunk Signalling

Voice and Video Call – Negotiated Media



10.58.9.86
Audio RTP UDP Port 16444
MP4A-LATM Audio codec
Bandwidth 64kbps
RFC 2833 DTMF

Audio

10.58.9.222
Audio RTP UDP Port 2346
MP4A-LATM Audio codec
Bandwidth 64kbps
RFC 2833 DTMF

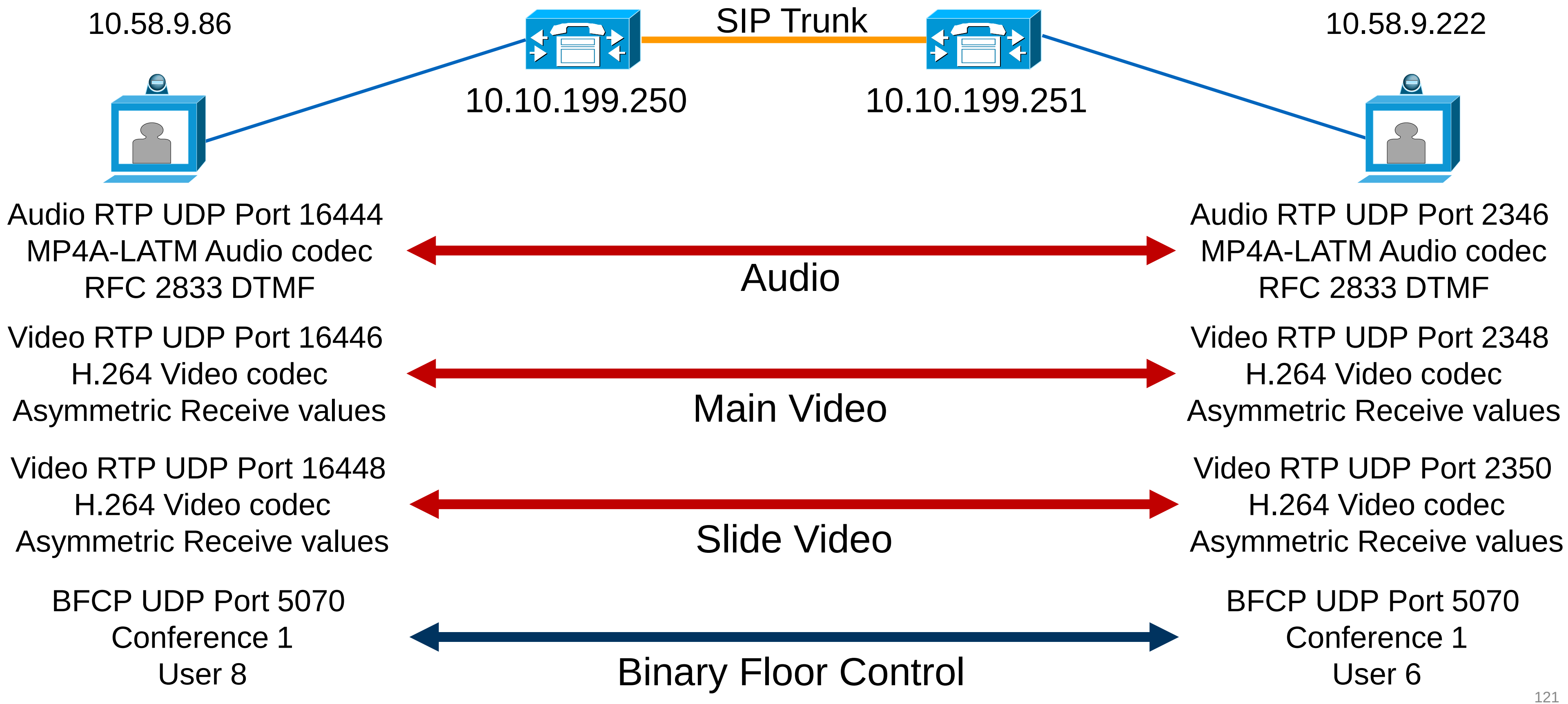
10.58.9.86
Video RTP UDP Port 16446
H.264 Video codec
Asymmetric Receive values
Bandwidth 6Mbps

Video

10.58.9.222
Video RTP UDP Port 2348
H.264 Video codec
Asymmetric Receive values
Bandwidth (6Mbps - 64kbps)

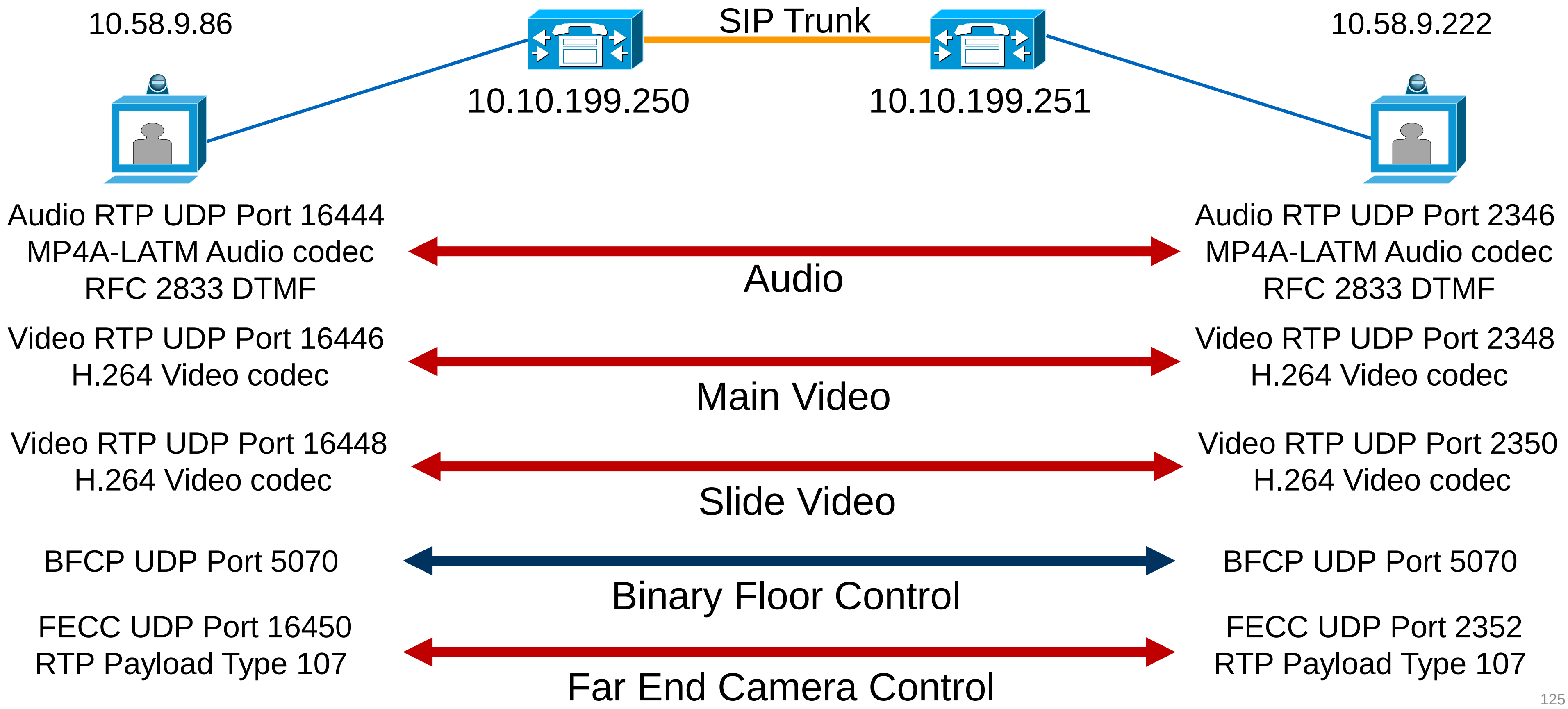
SIP Trunk Signalling

Voice and Video Call with BFCP – Negotiated Media



SIP Trunk Signalling and Basic Operation

Voice and Video Call with BFCP & FECC – Negotiated Media

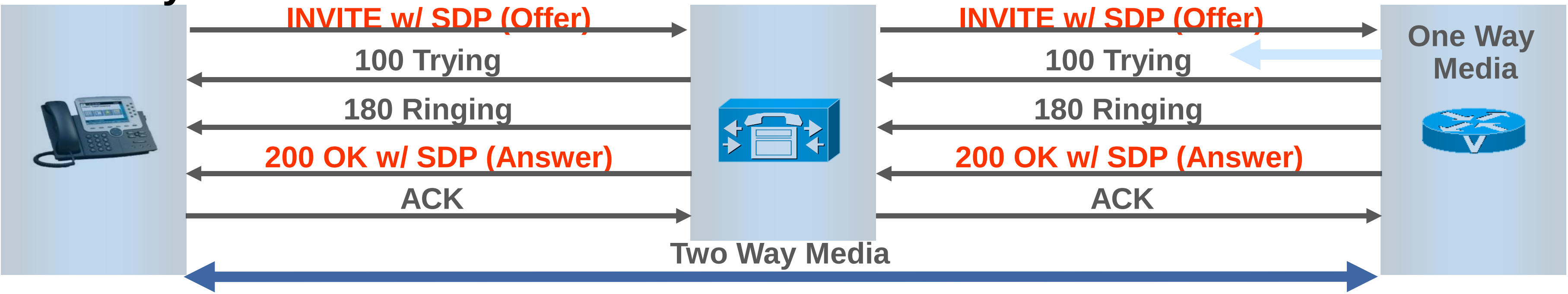





CUCM SIP Trunk Features and Call Functions

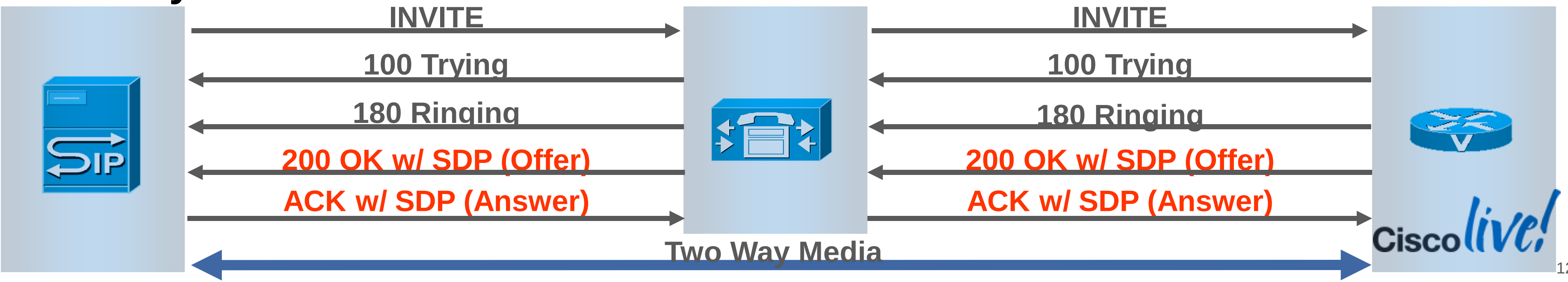
SIP Messaging – Delayed and Early Offer

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).

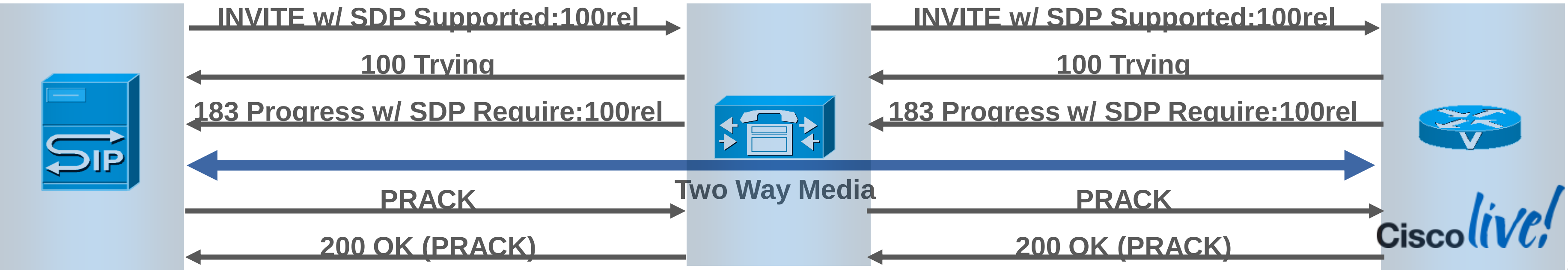
Provisional responses provide information on the progress of the request, but are not sent reliably – so the sender of a provisional response does know that it has been received.

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

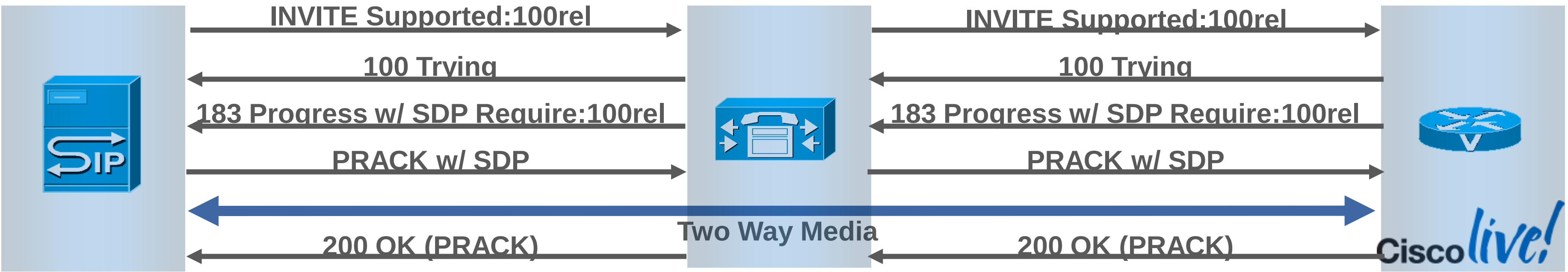
Like final responses, by using PRACK - 1XX messages will be periodically re-sent until their receipt is acknowledged by the receiver by sending a PRACK, which is also acknowledged by the 1XX sender.

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

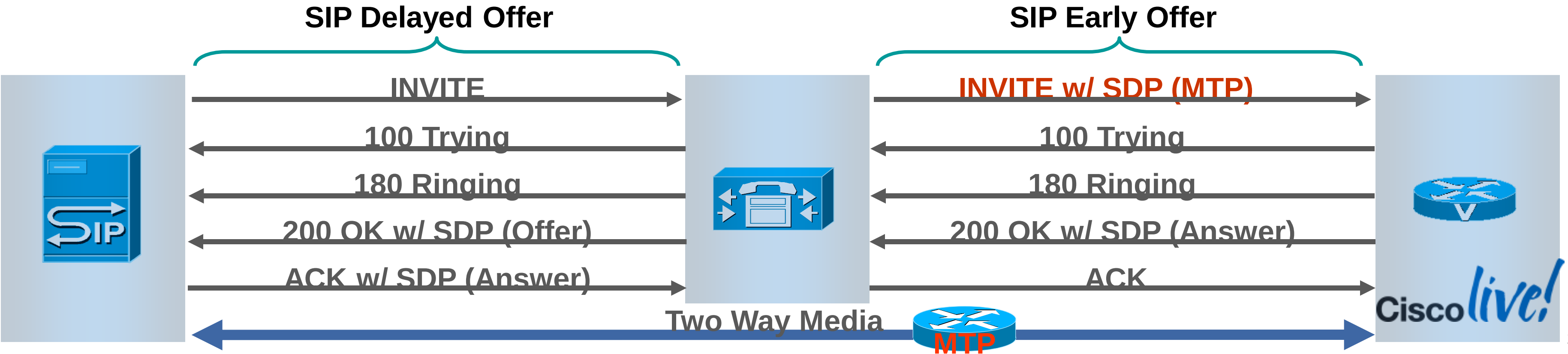
Delayed Offer to Early Offer calls

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…

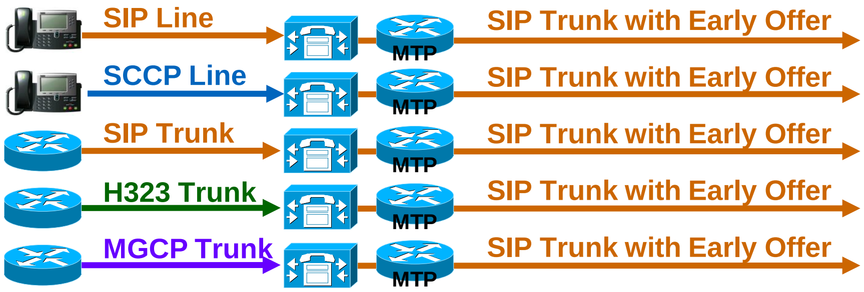
Solution – Insert a Media Termination Point (**MTP**) and use its media characteristics to create the Offer in SDP with the outbound INVITE



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 :

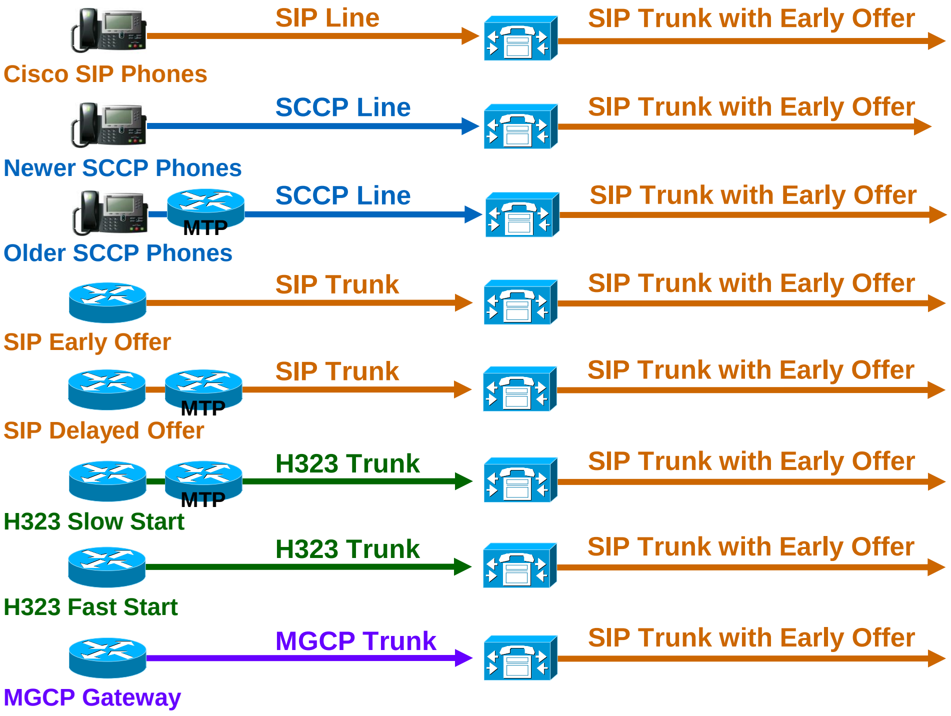
SIP Early Offer Trunks use the Trunk’s Media Termination Point (MTP) resources, inserting an MTP into the media path for every outbound (and inbound) call – sending the MTP’s IP Address, UDP port number and codec in the SDP body of the initial SIP INVITE instead of those of the endpoint.

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)”

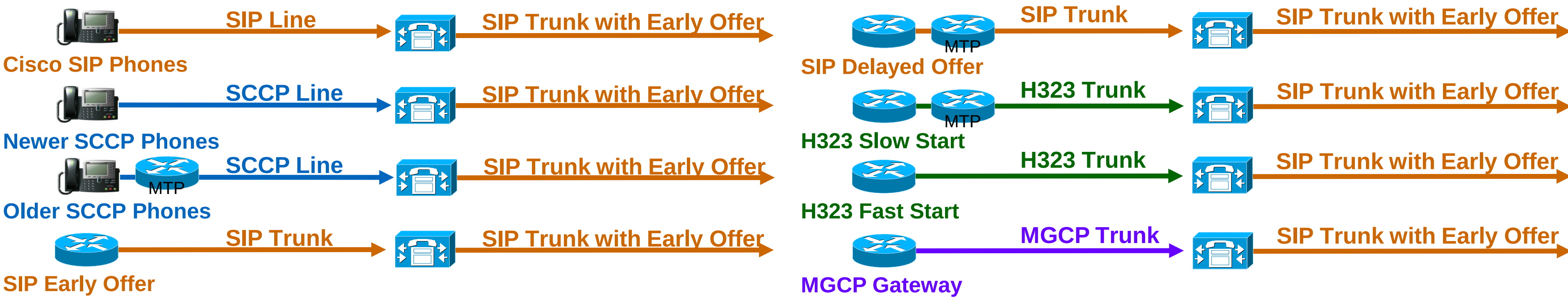


For Calls from trunks and devices that can provide their IP Address, UDP port number and supported codecs - This information is sent in the SDP body of the initial SIP Invite on the outbound Early Offer Trunk. No MTP is used for the Early Offer

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)

CUCM SIP Trunk Signalling

Enabling SIP Early Offer – Method 2 – Benefits



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

- Reduced MTP usage
- Single voice codec MTP limitation removed (by using the pass through codec (IOS MTPs only)
- Voice codecs sent in SIP Offer based on calling device capabilities & region settings
- Use of the Calling device's MTP rather than Trunk's MTP
  - Media does not hairpin through the Trunk's MTP

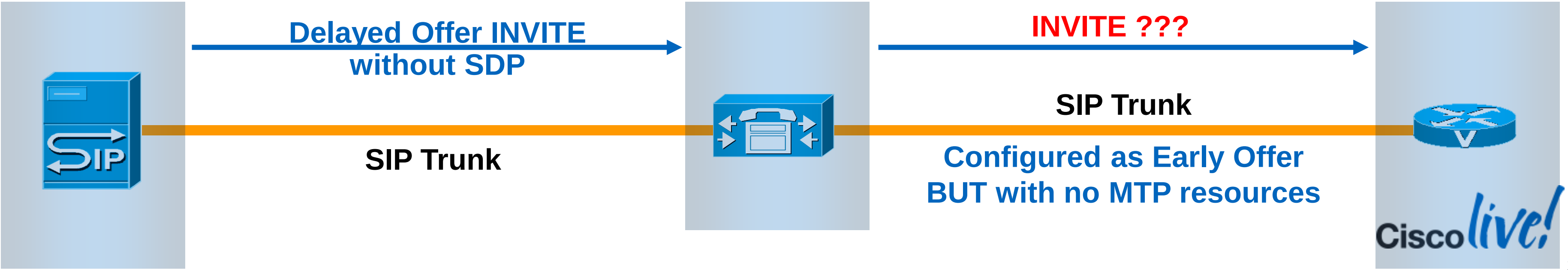
Outbound SIP Early Offer

SIP Profile Configuration

X Early Offer support for voice and video calls (insert MTP if needed)

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 ?

And No MTPs are available ??

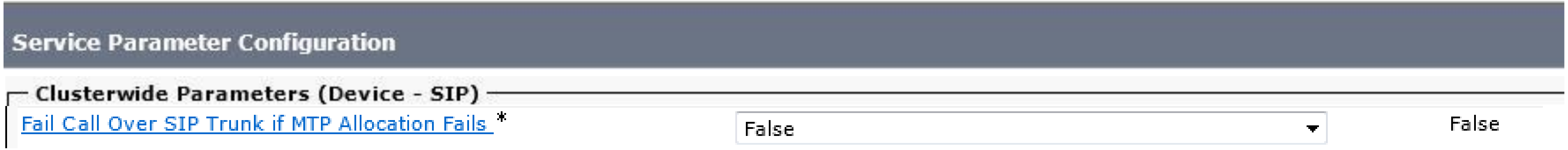


CUCM SIP Trunk Signalling

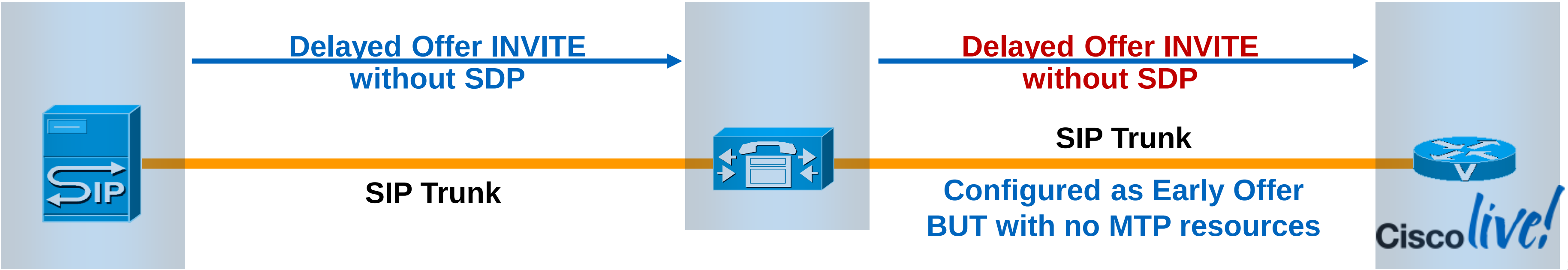
MTP-less Outbound Early Offer – Inbound Delayed Offer

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 ?

And No MTPs are available ? It depends on the value of this Service Parameter..
"Fail Call Over SIP Trunk if MTP allocation Fails"



Default value = False – Call proceeds without an MTP….. As a Delayed Offer call



CUCM SIP Trunk Signalling

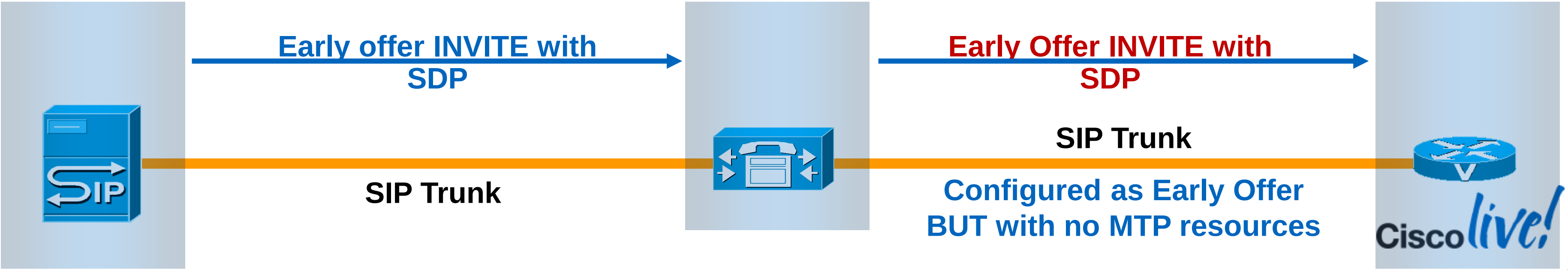
MTP-less Outbound Early Offer – Inbound Early Offer

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

SIP Profile Configuration

X Early Offer support for voice and video calls (insert MTP if needed)

As per the "Early Offer for Voice and Video (insert MTP if needed)" feature The call proceeds ….. The Early Offer received on the inbound Trunk is sent over the Outbound Early Offer Trunk



CUCM SIP Trunk Signalling

MTP-less Outbound Early Offer Trunks

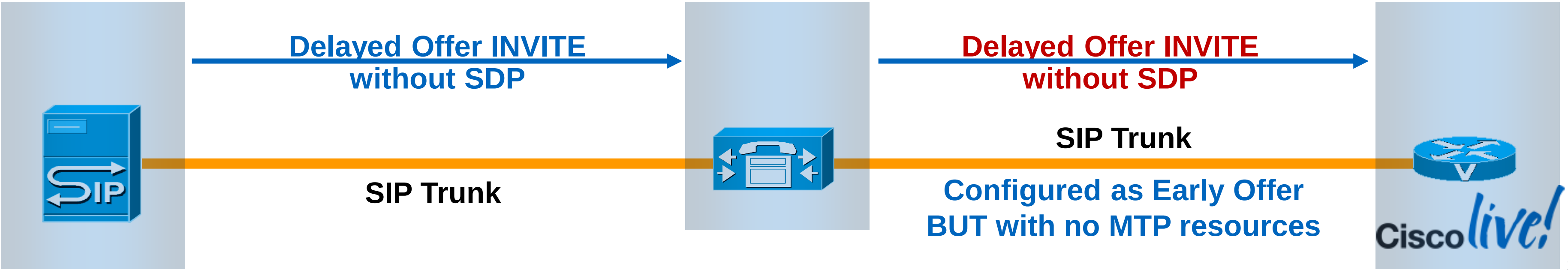
Service Parameter Configuration

Clusterwide Parameters (Device - SIP)

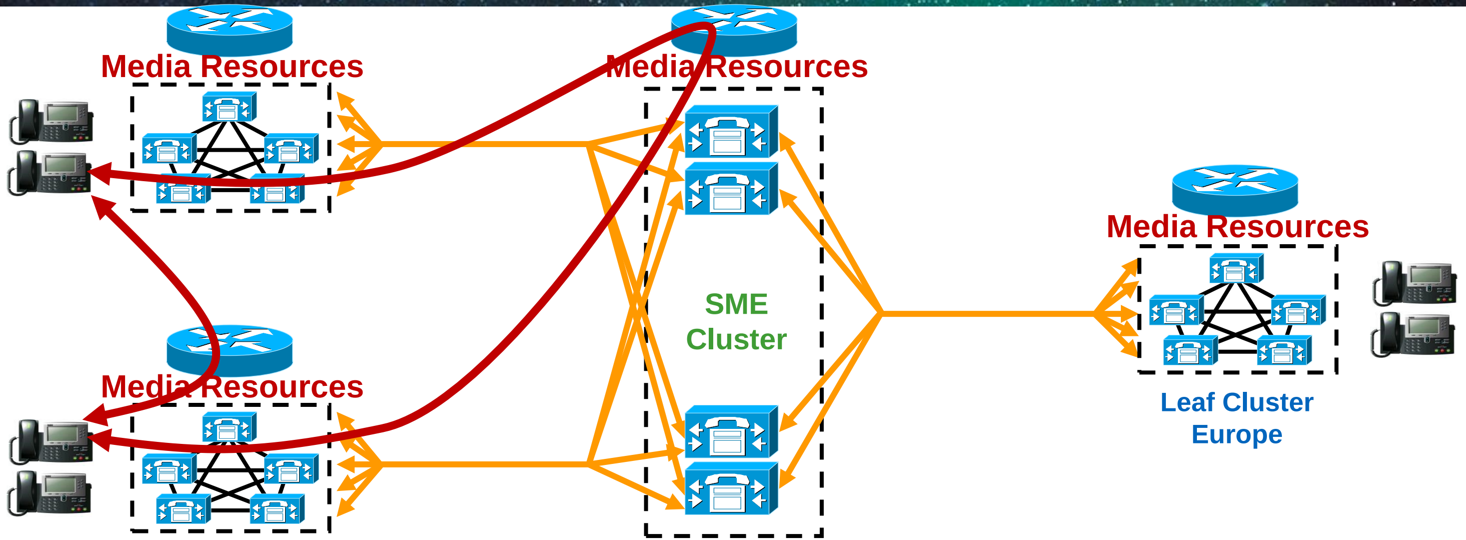
Fail Call Over SIP Trunk if MTP Allocation Fails *	False	False

- **Inbound Delayed Offer Call → Outbound Delayed Offer Call**
- **Inbound Early Offer Call → Outbound Early Offer Call**

Use cases - SME Trunk design – Media Transparent SME cluster
(For more details see SME breakout session BRKUCC 2450)
Requires that no MTPs/TRPs/RSVP Agents/Xcoders are associated with SME Trunks



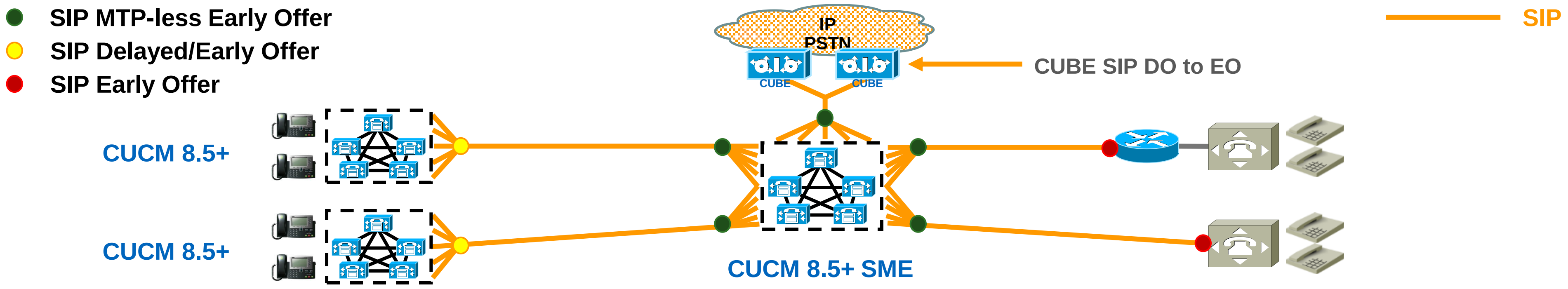
Reasons to use SIP Trunks with MTP-less EO on SME
SME Clusters with no Media Resources



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

Is this design possible ? Yes, but it requires the use of SIP Trunks only and a specific SIP Trunk configuration…. "MTP-less Early Offer"

SIP Trunk Design Recommendations
SME Cluster using MTP-less EO and no Media Resources



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 assigned to Trunks

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

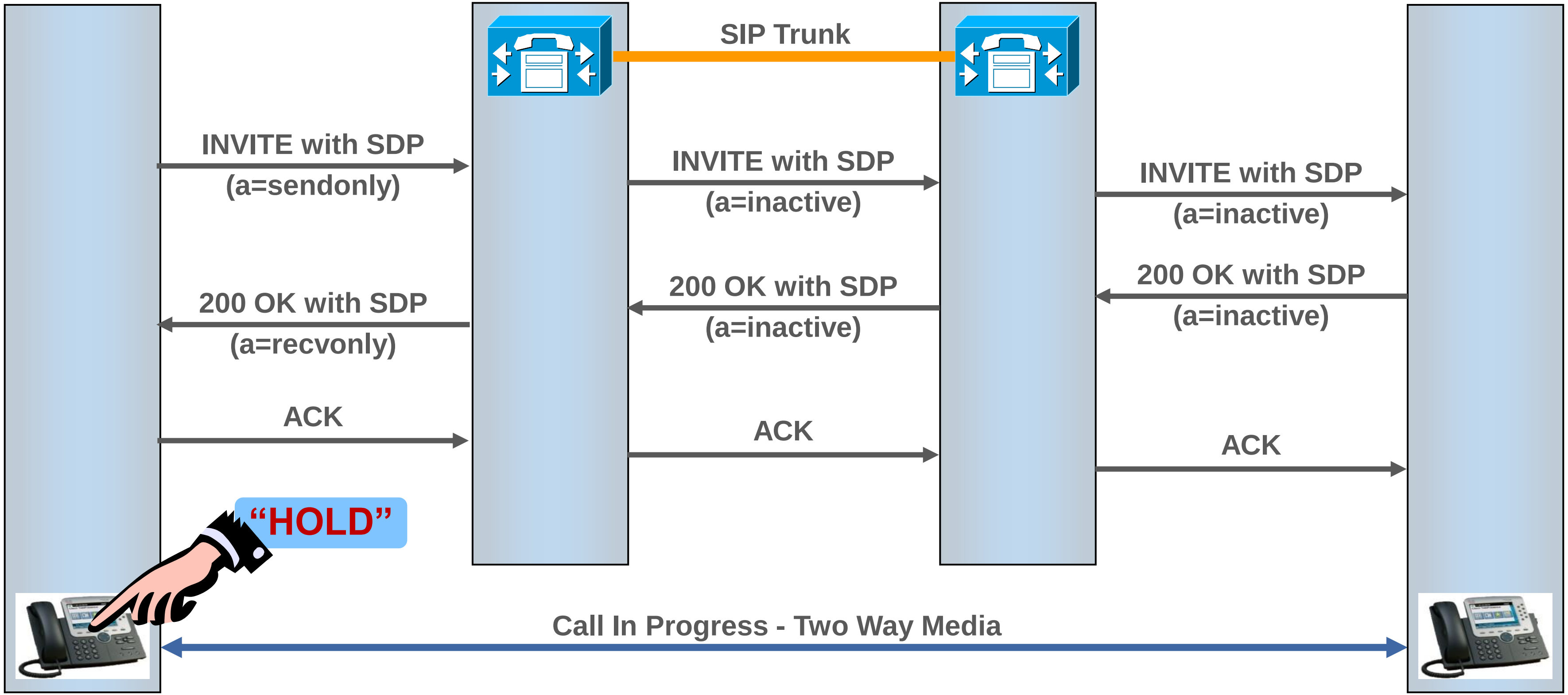
SIP Delayed Offer/Early Offer (EO commonly used), (EO by sent by the CUBE/ IOS Gateways) OPTIONS Ping, Early Offer usually required by Service Providers (Use CUBE SIP DO to EO)




CUCM SIP Trunk Call Functions – Hold/Resume

CUCM SIP Trunk Signalling and Operation

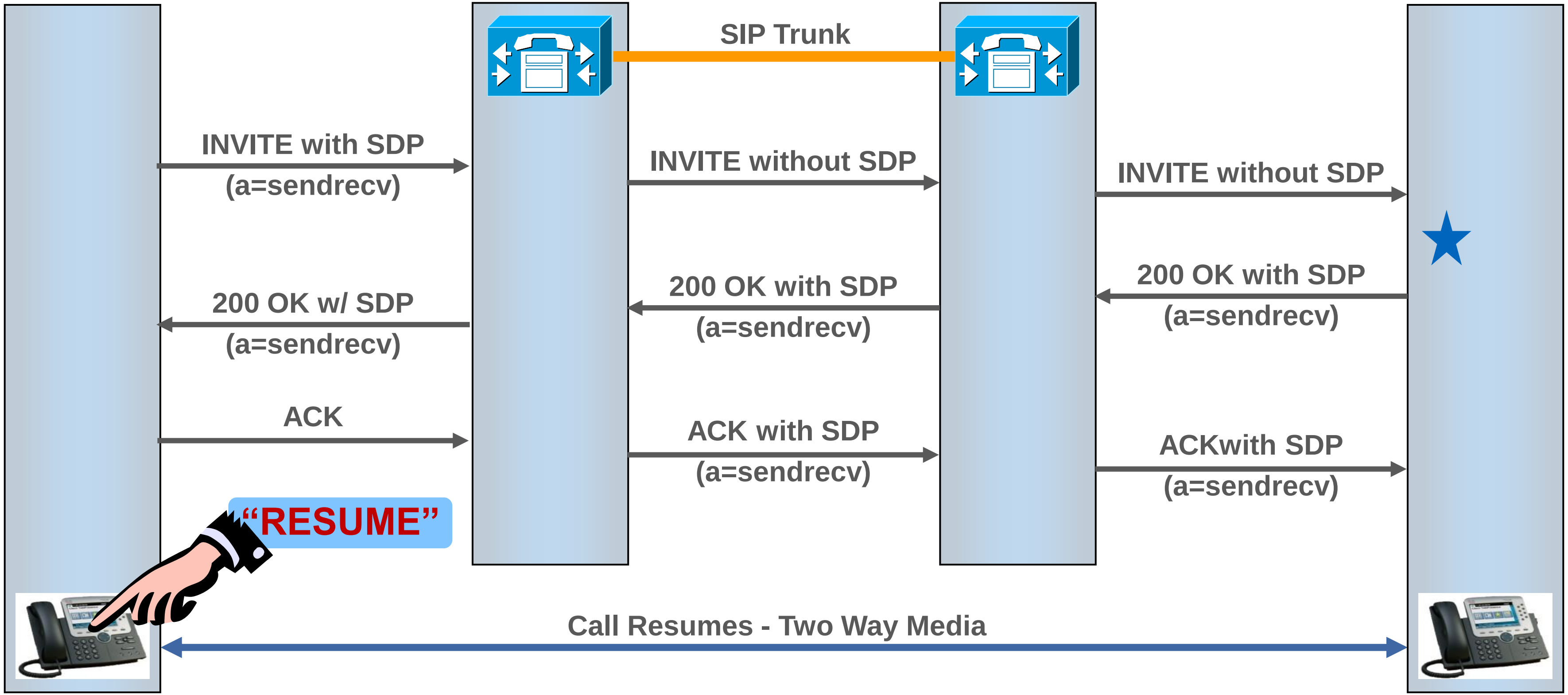
Hold/Resume Signalling - Hold



Note – CUCM SIP Trunks also send c= IN IP4 0.0.0.0 in SDP

(Older RFC 2543 Hold method)

CUCM SIP Trunk Signalling and Operation Hold/Resume Signalling - Resume



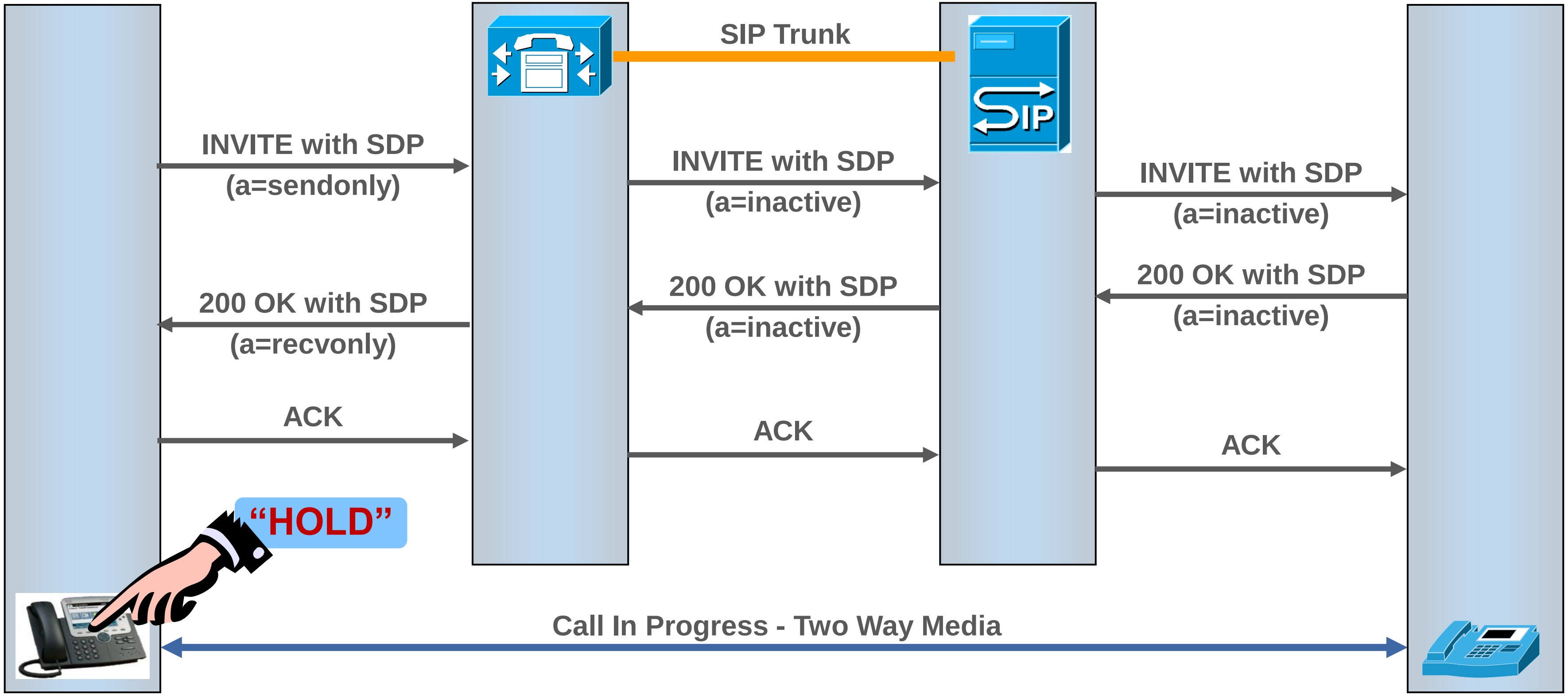
★ On receipt of a mid call INVITE without SDP the remote User Agent should respond with its full codec list and a=sendrecv




CUCM SIP Trunk Call Functions – Send “Send Receive” in Mid Call INVITE

CUCM SIP Trunk Interop Feature – The Issue (1)

"Send send-receive SDP in mid-call INVITE"

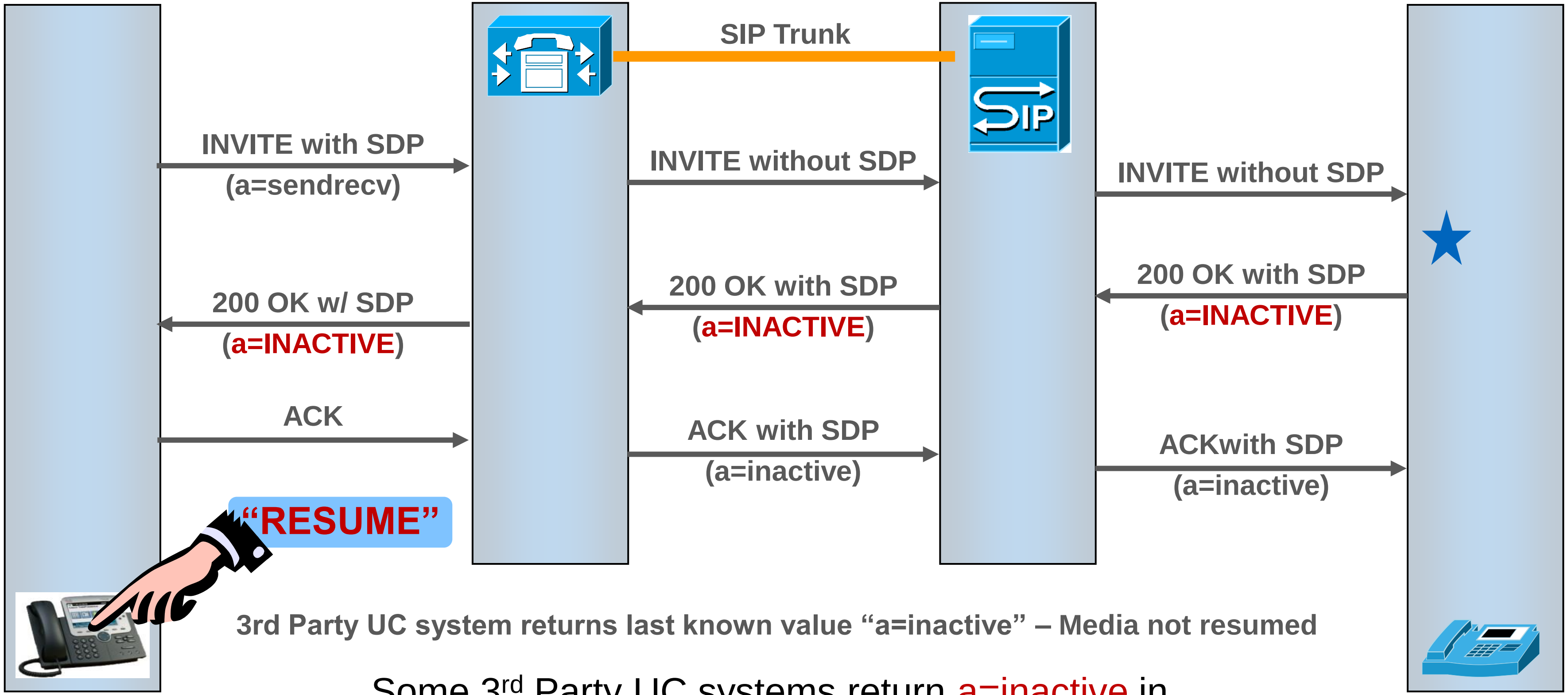


Used to address a=inactive issues with 3rd Party systems

(Issue not widely found - typically with some Service Providers)

CUCM SIP Trunk Interop Feature – The Issue (2)

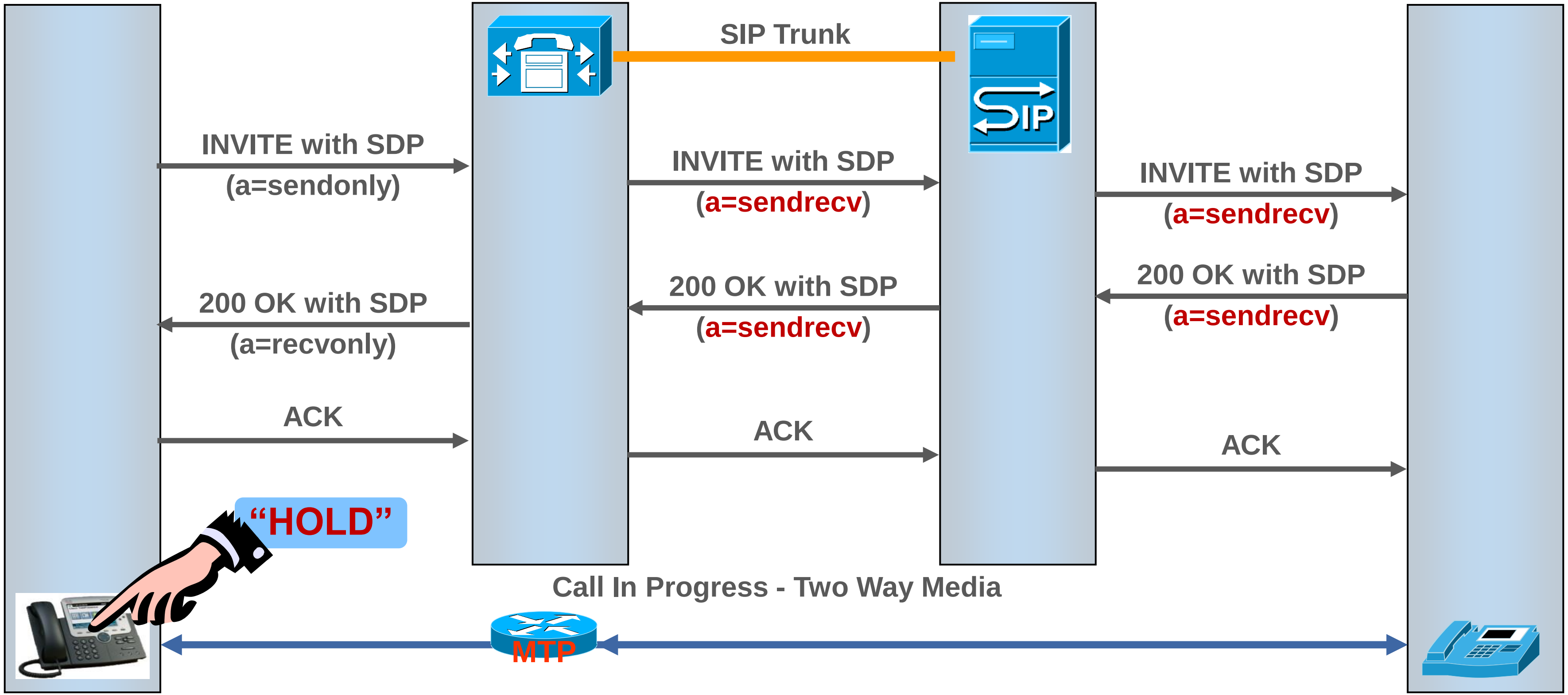
"Send send-receive SDP in mid-call INVITE"



Some 3rd Party UC systems return a=inactive in response to a mid call INVITE without SDP
Effect - Media cannot be re-established on Resume

CUCM SIP Trunk Interop Feature – The Fix (1)

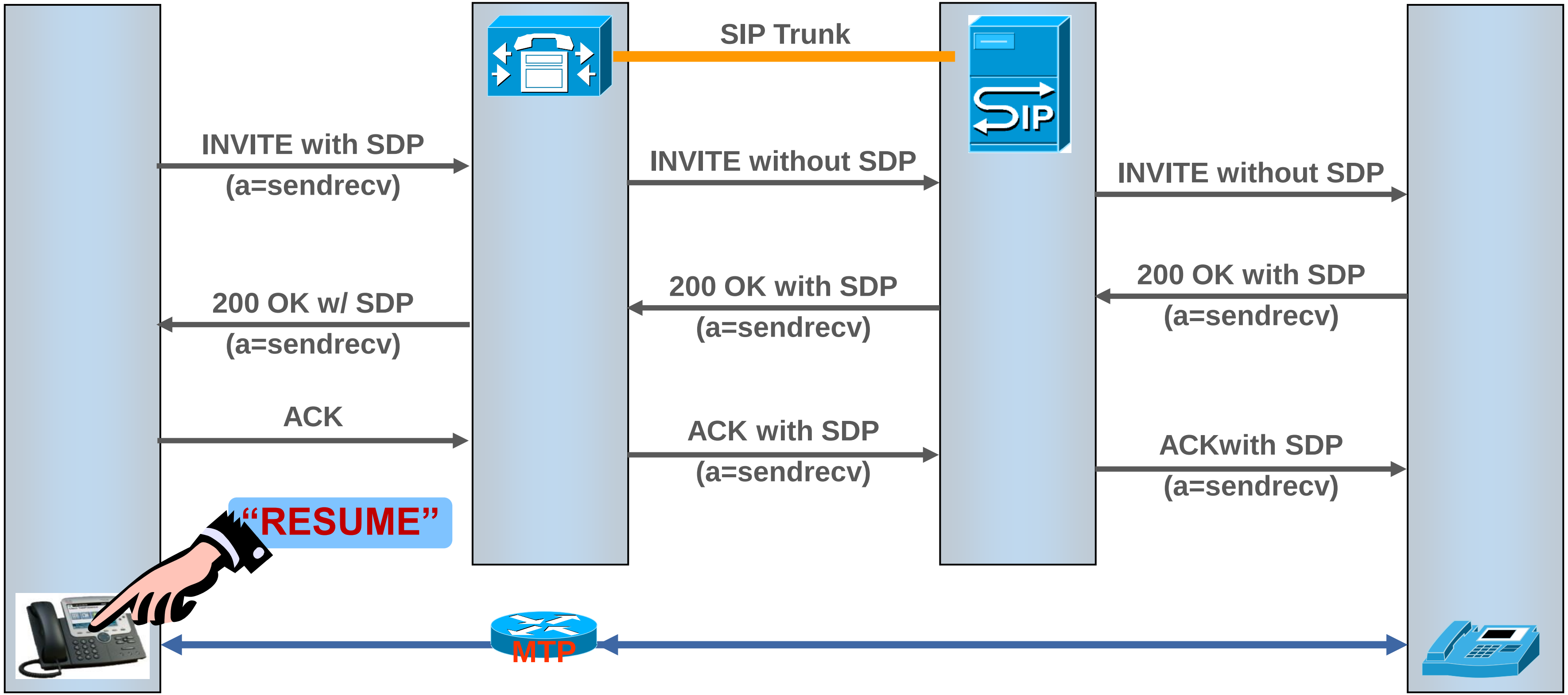
"Send send-receive SDP in mid-call INVITE" – Enabled



When the mid-call INVITE is sent – CUCM inserts an MTP which anchors the media to the held device and allows the holding device to disconnect its media (and optionally insert MOH)

CUCM SIP Trunk Interop Feature – The Fix (2)

"Send send-receive SDP in mid-call INVITE" Enabled

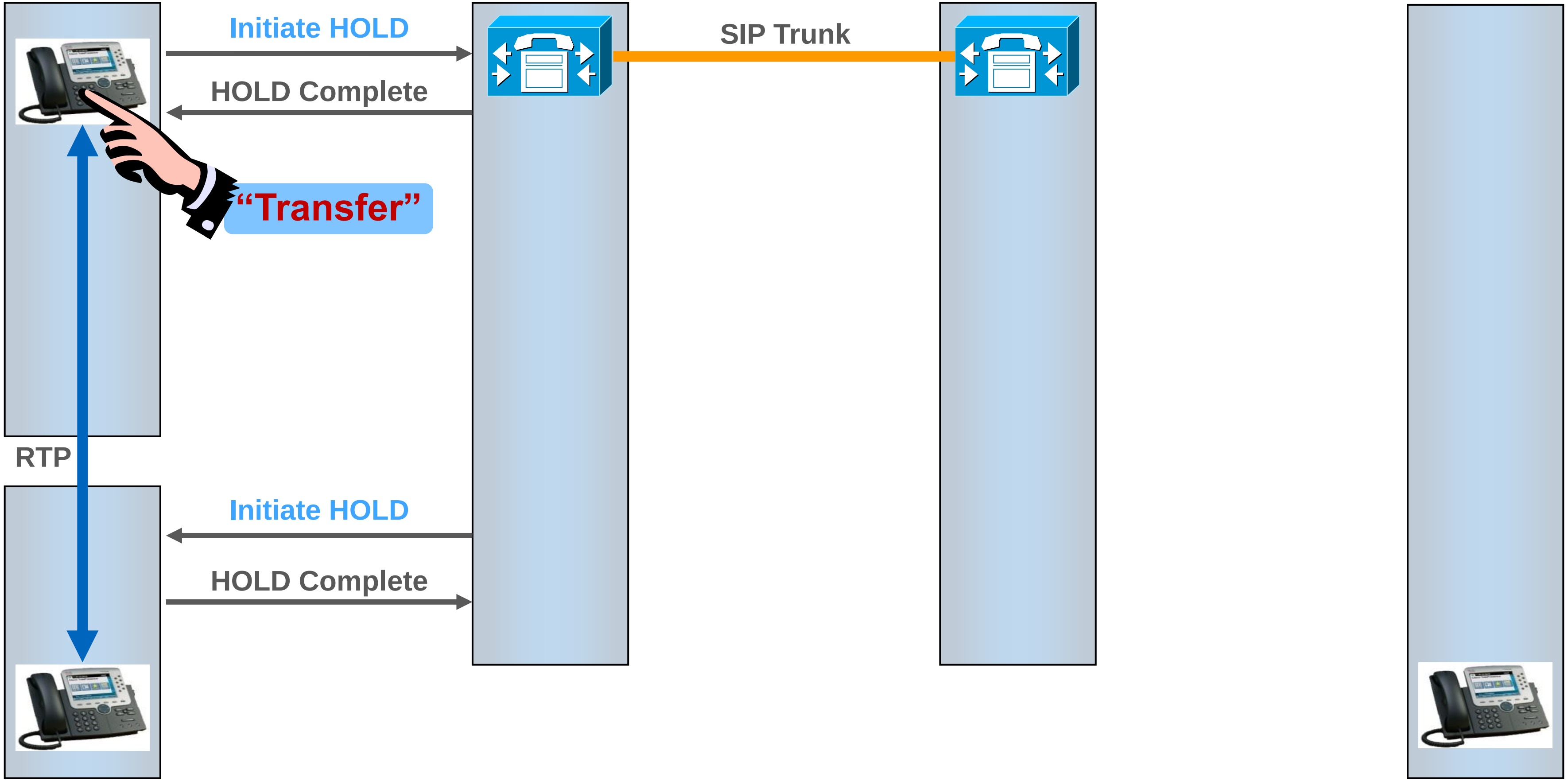





CUCM SIP Trunk Call Functions – Transfer

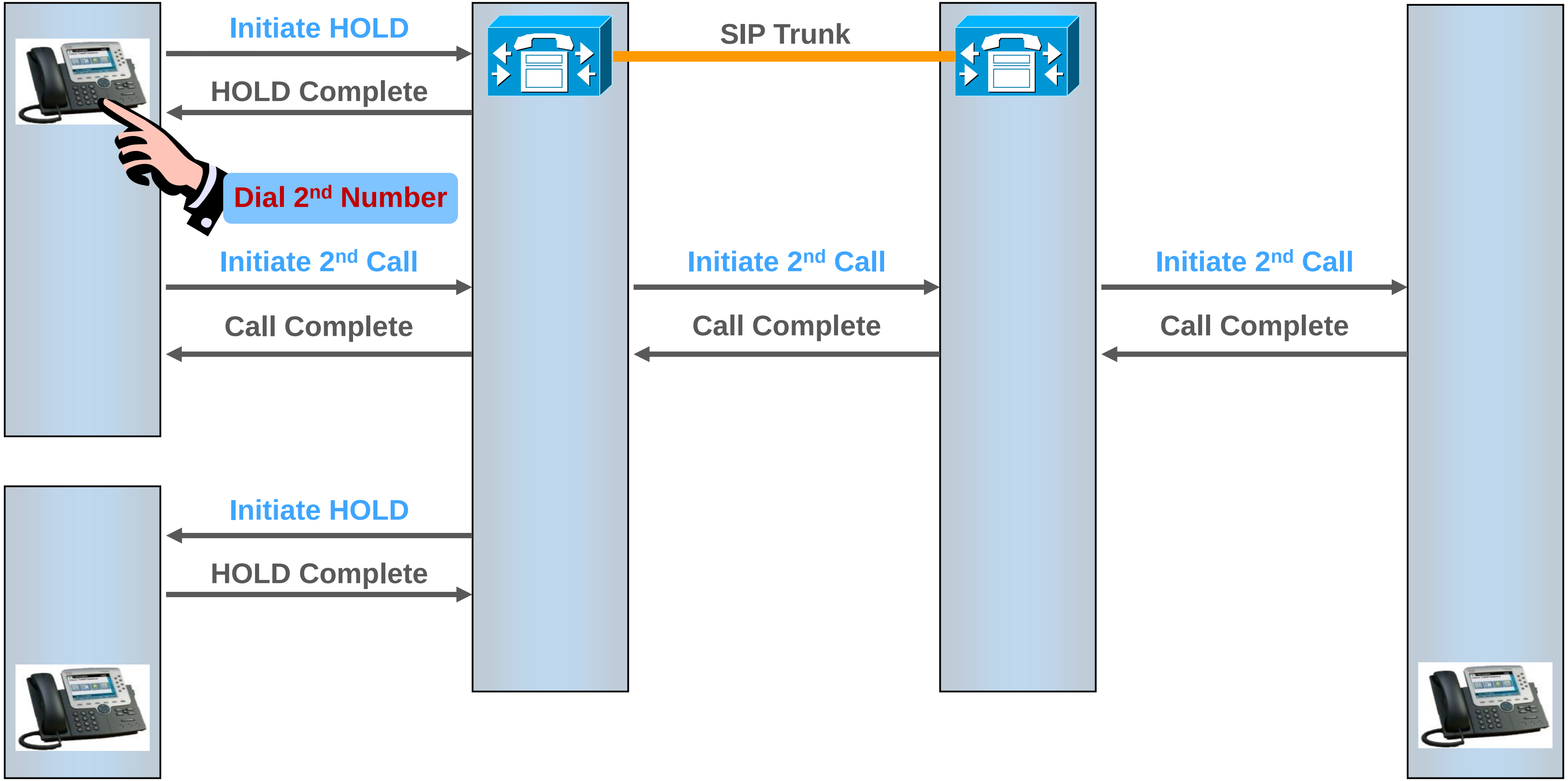
CUCM SIP Trunk Signalling and Operation

Transfer – Call in Progress – Hold Initiated



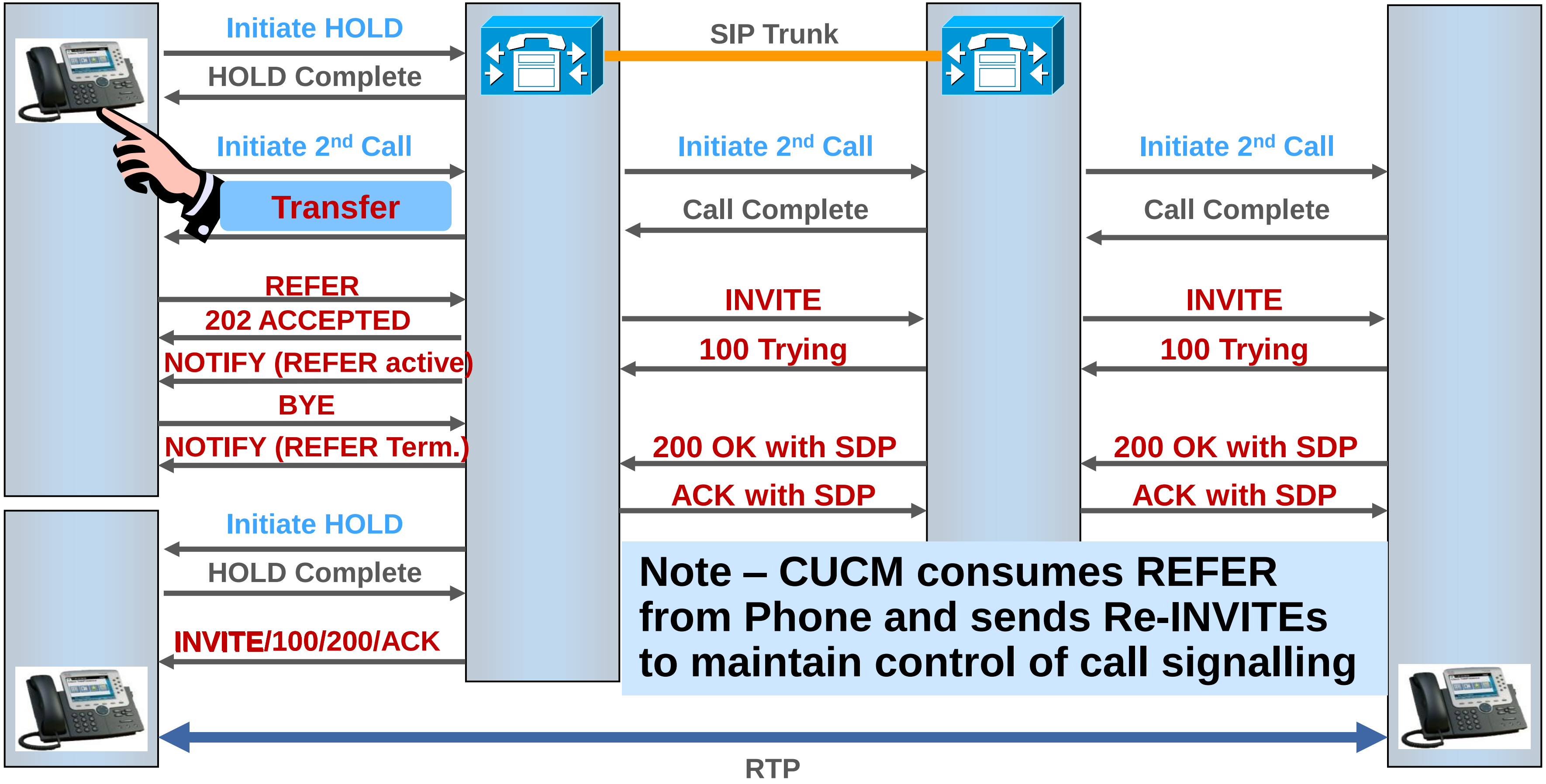
CUCM SIP Trunk Signalling and Operation

Transfer – Call Transfer Target

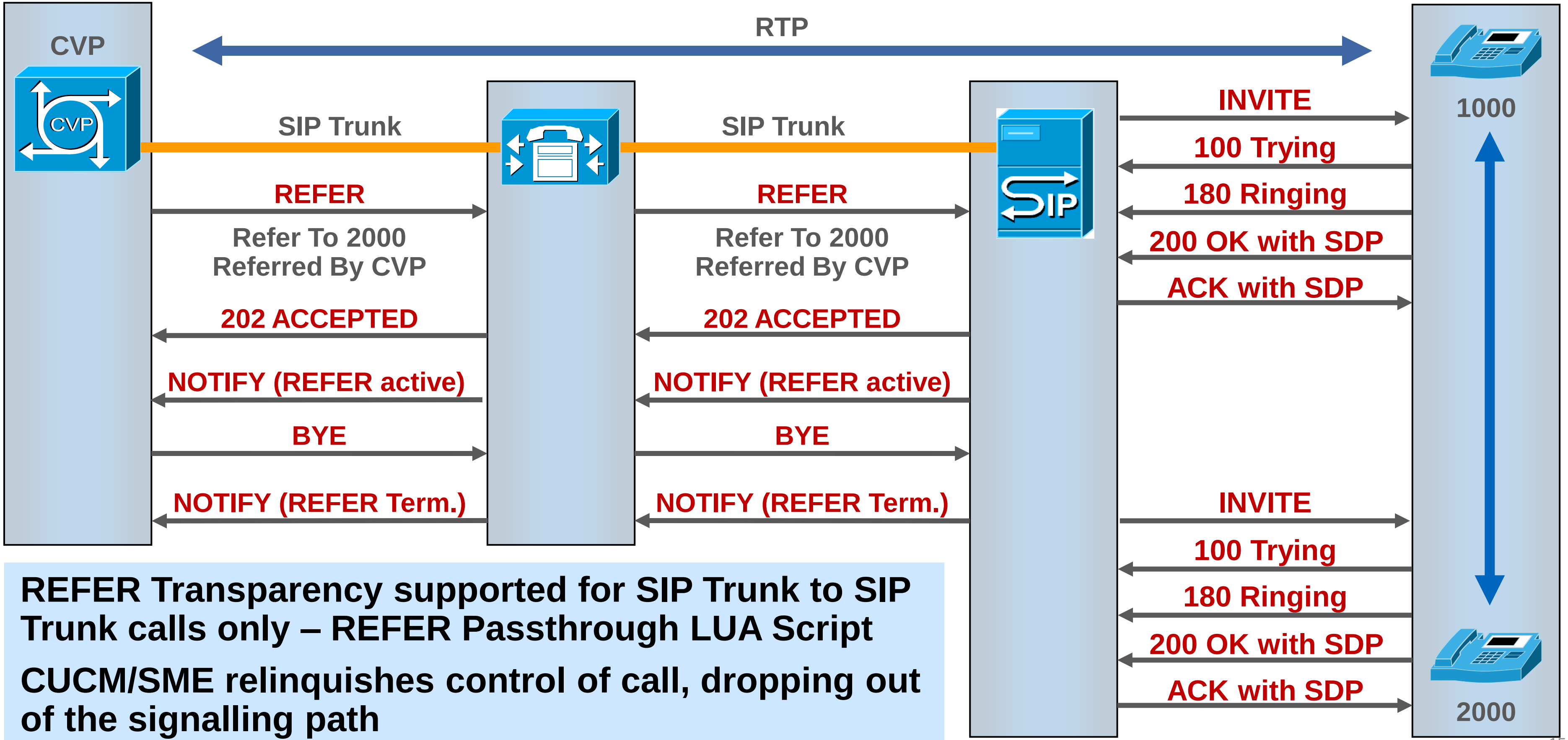


CUCM SIP Trunk Signalling and Operation

Transfer – Completing the Transfer



CUCM SIP Trunk Signalling Operation
SIP Messaging – Transfer – REFER Transparency



REFER Transparency supported for SIP Trunk to SIP Trunk calls only – REFER Passthrough LUA Script

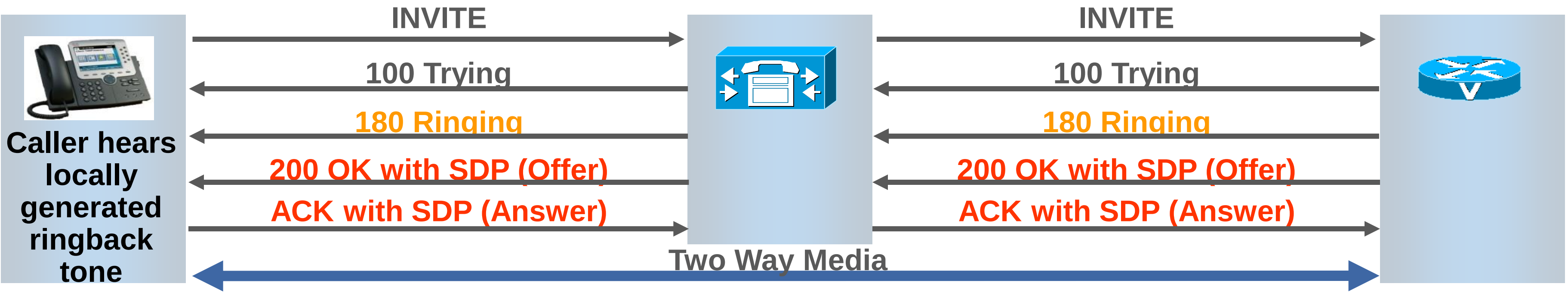
CUCM/SME relinquishes control of call, dropping out of the signalling path



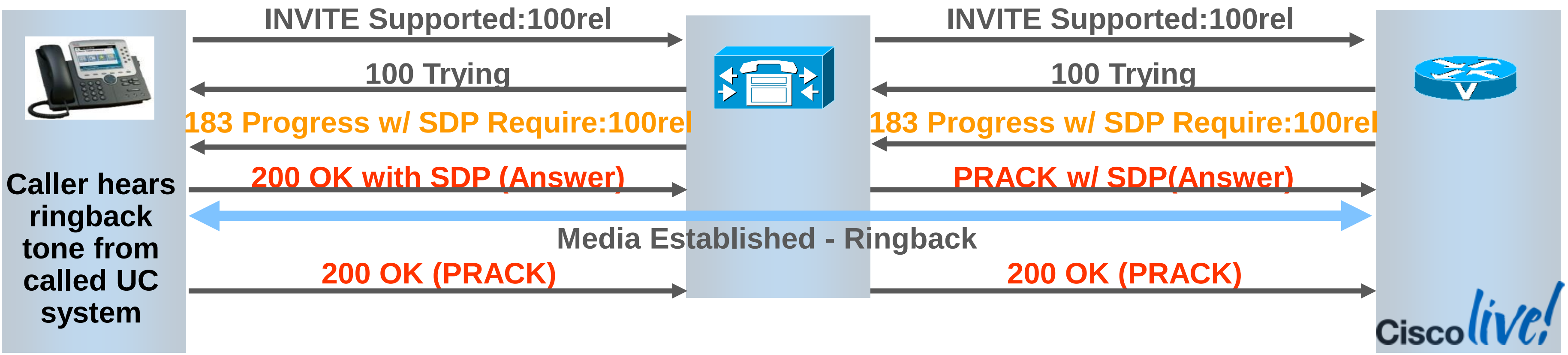

CUCM SIP Trunk Call Features Explained

CUCM SIP Trunk Signalling and Operation

The Offer/Answer Model - Ringback



SIP allows SDP to be optionally sent in 18X messages (MUST BE WITH PRACK)



CUCM SIP Trunk Features

SIP Profile Settings - Disable Early Media on 180

SIP Profile Configuration

☐ Disable Early Media on 180

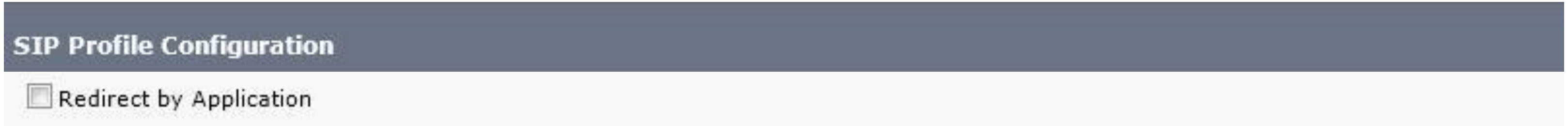
By default, Cisco Unified Communications Manager signals the calling phone to play local ringback if SDP is not received in the 180 or 183 response.

If SDP is included in the 180 or 183 response, instead of playing ringback locally, Cisco Unified Communications Manager connects media, and the calling phone plays whatever the called device is sending (such as ringback or busy signal).

If ringback is not received, the device to which you are connecting may be including SDP in the 180 response, but it is not sending any media before the 200 OK response. In this case, check this check box to play local ringback on the calling phone and connect the media upon receipt of the 200 OK response

CUCM SIP Trunk Features

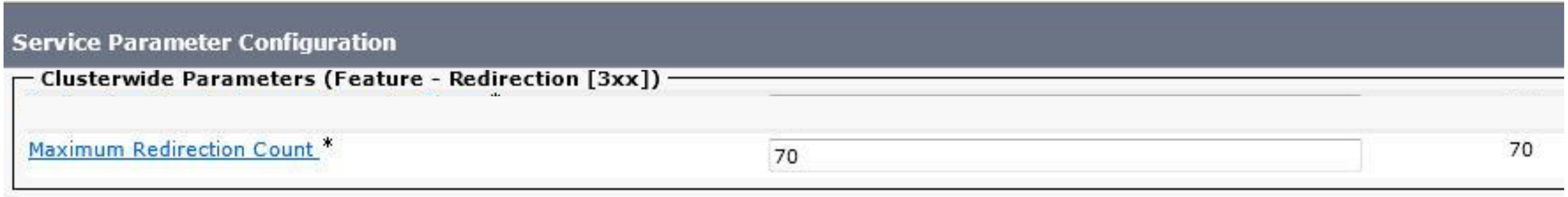
SIP Profile settings - Redirect by Application



Redirect by Application – Default setting = unchecked

When checked allows CUCM to apply digit analysis to the redirected contact number to :

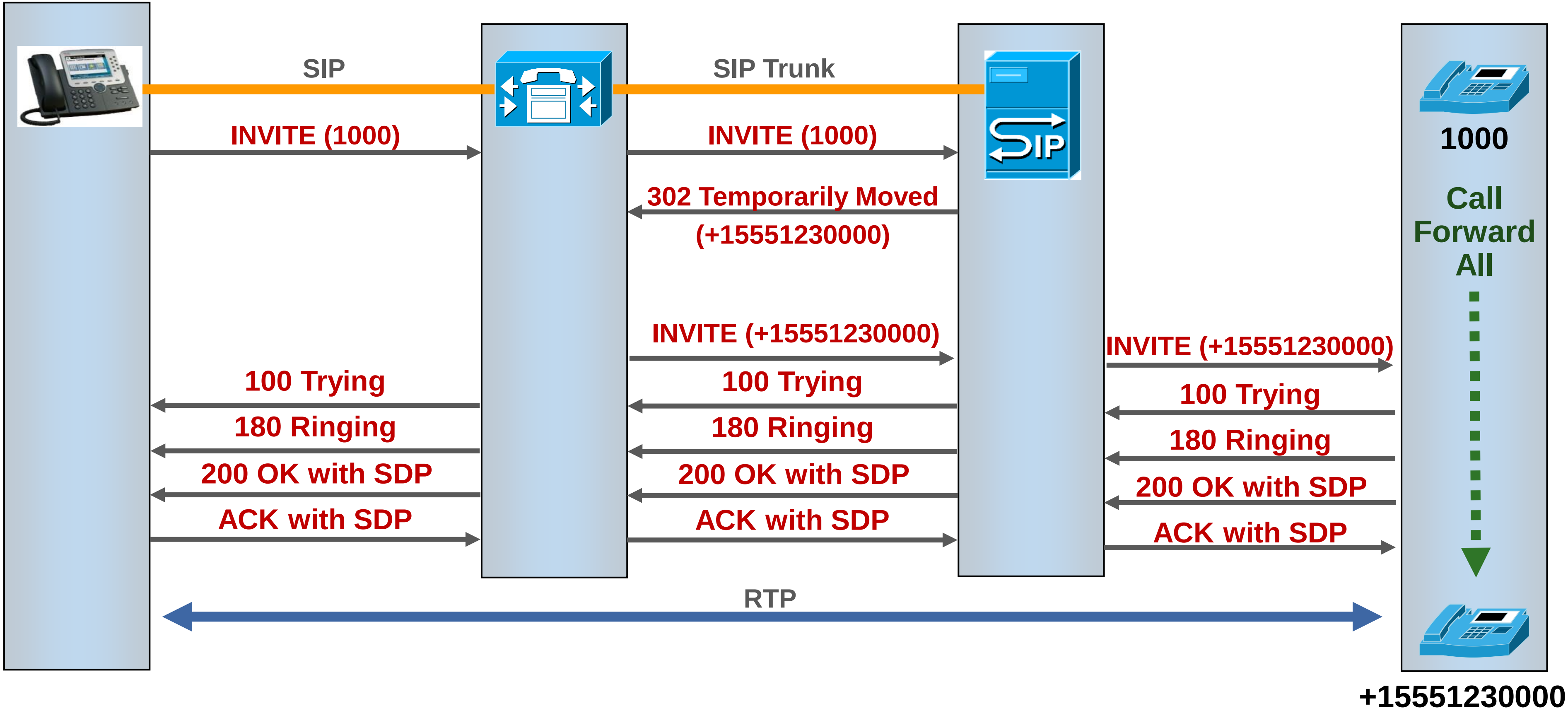
a) Make sure that the call gets routed correctly
b) Apply a specific Calling Search Space for Calls of Service/ Call Restriction
c) Prevent DOS attack by limiting the number of redirections
d) Allow other features to be invoked while the redirection is taking place



When unchecked (default) the redirection gets handled at the SIP stack level and the above features cannot be invoked

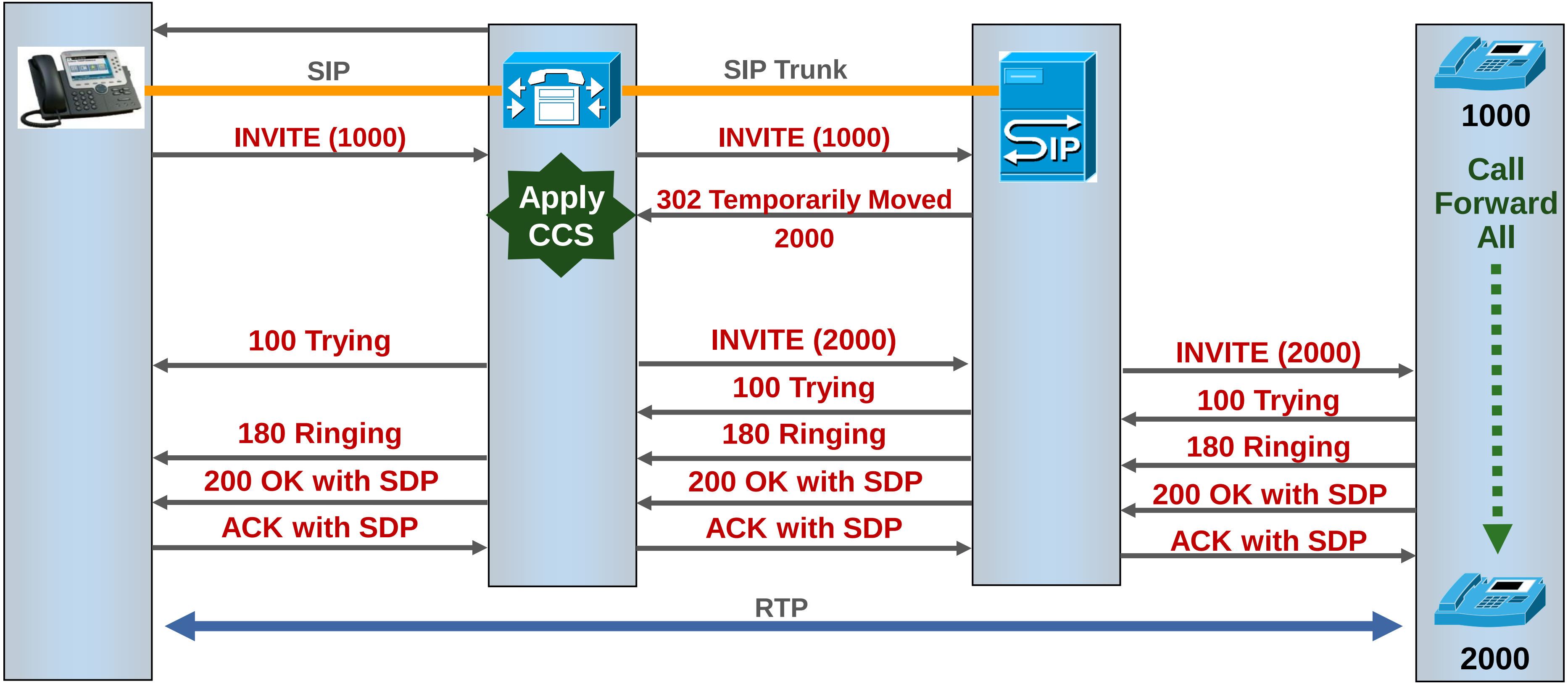
CUCM SIP Trunk Features

Redirect by Application – Disabled (Default setting)



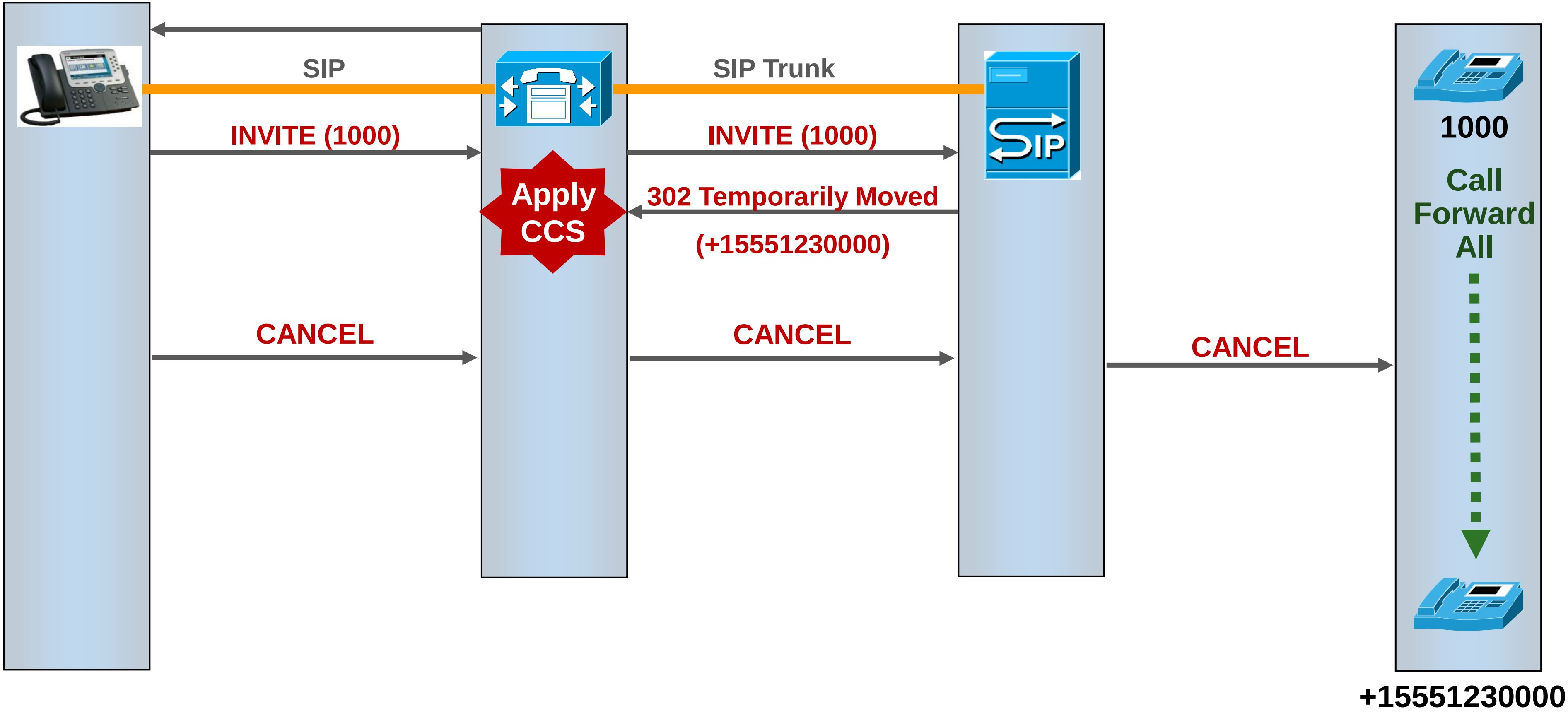
CUCM SIP Trunk Features

Redirect by Application – Enabled – Call Allowed



CUCM SIP Trunk Features

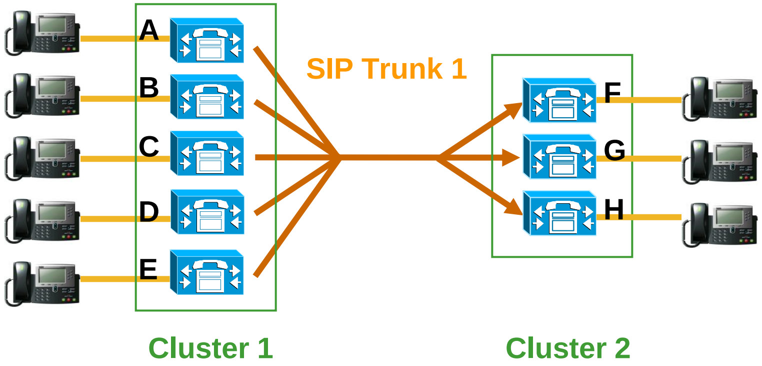
Redirect by Application – Enabled – Call Blocked

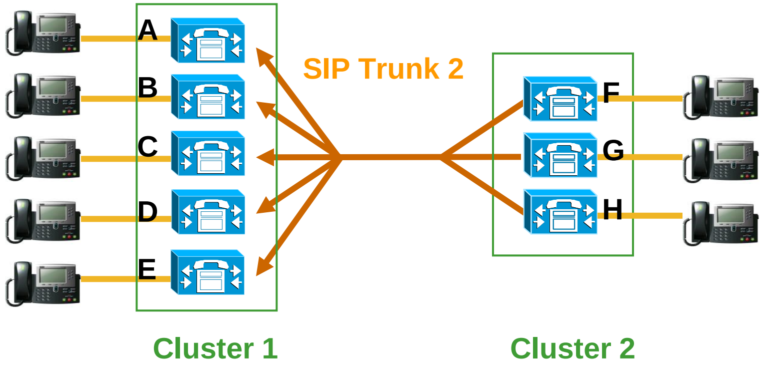


CUCM SIP Trunks

Matching inbound SIP calls to configured SIP Trunks

CUCM SIP Trunks will only accept inbound calls from a device with a source IP address and port number that has been defined on the matching Trunk





Cluster 1 – SIP Trunk 1 Configuration

SIP Trunk 1 has an active SIP daemon on servers A, B, C, D and E

Cluster 2 servers F, G and H are defined as Trunk destinations

Cluster 2 – SIP Trunk 2 Configuration

SIP Trunk 2 has an active SIP daemon on servers F, G and H

Cluster 1 servers A, B, C, D and E are defined as Trunk destinations

CUCM SIP Trunk Features

SIP Profile settings – Reroute request based on

Reroute Incoming Request to New Trunk based on :

Never (Default) - Match inbound call to SIP Trunk based on IP address and port number

Contact header - Match inbound call based on IP address and port number – but re-route the call to another SIP Trunk based on the IP address and port number received in the contact header

- Can be used to reroute calls from a SIP Proxy to an end user/system specific CUCM SIP Trunk

Call-Info Header with purpose=x-cisco-origIP

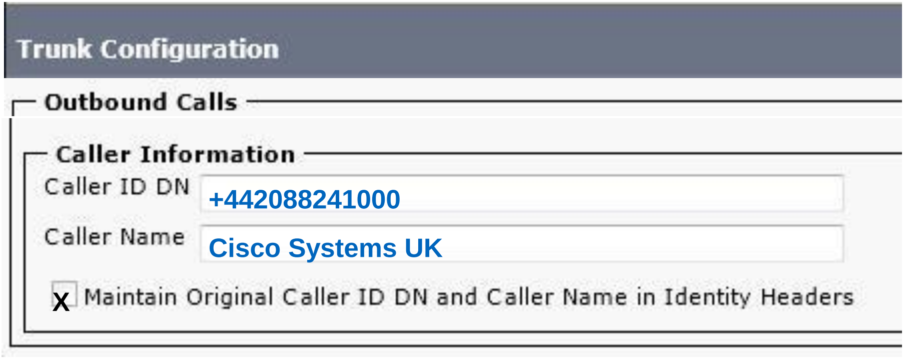
- Used to match inbound calls from CVP to a specific Trunk based on the IP address and port number contained in the Call-Info header – parameter "purpose=x-cisco-origIP" (Used for CAC)

CUCM SIP Trunk Features

Outbound Calls – Caller DN and Name Information

From: “Cisco Systems UK”<sip:+442088241000@10.10.199.251>
P-Asserted-Identity: “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
Contact: <sip:+442088241000@10.10.199.251:5060;transport=tcp>

Directory Number = 2002
Name = Bob Jones

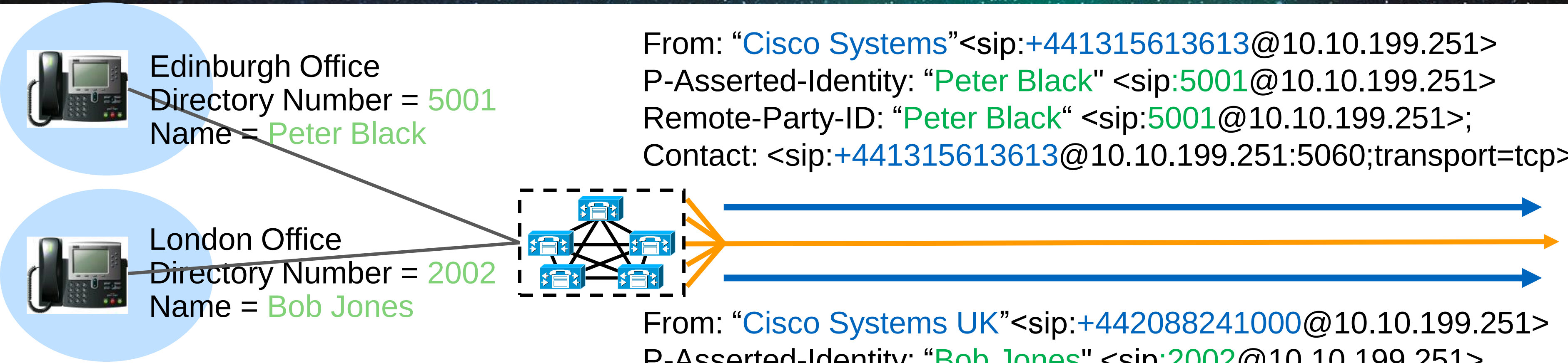


The Caller Information field allows the From header to be over written for outbound SIP Trunk calls

If “Maintain Original Caller ID DN and Name” is Not checked the PAID and RPID fields are also over written

CUCM SIP Trunk Features

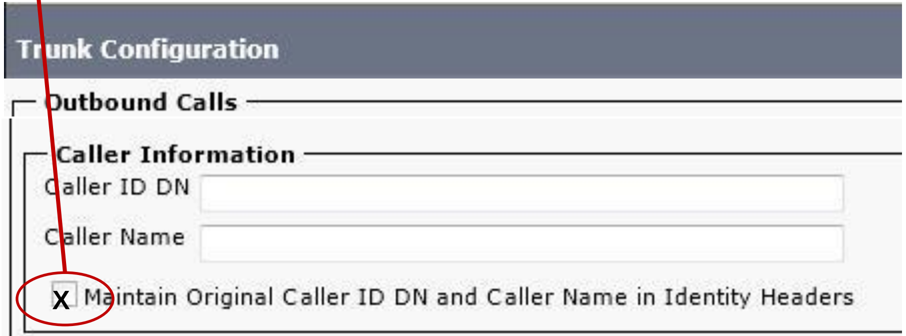
Outbound Calls –Caller DN and Name Information



From: “Cisco Systems”<sip:+441315613613@10.10.199.251>
P-Asserted-Identity: “Peter Black" <sip:5001@10.10.199.251>
Remote-Party-ID: “Peter Black“ <sip:5001@10.10.199.251>;
Contact: <sip:+441315613613@10.10.199.251:5060;transport=tcp>

From: “Cisco Systems UK”<sip:+442088241000@10.10.199.251>
P-Asserted-Identity: “Bob Jones" <sip:2002@10.10.199.251>
Remote-Party-ID: "Bob Jones“ <sip:2002@10.10.199.251>;
Contact: <sip:+442088241000@10.10.199.251:5060;transport=tcp>

Note this setting can still be used



Trunk Configuration

Outbound Calls

Caller Information

Caller ID DN

Caller Name

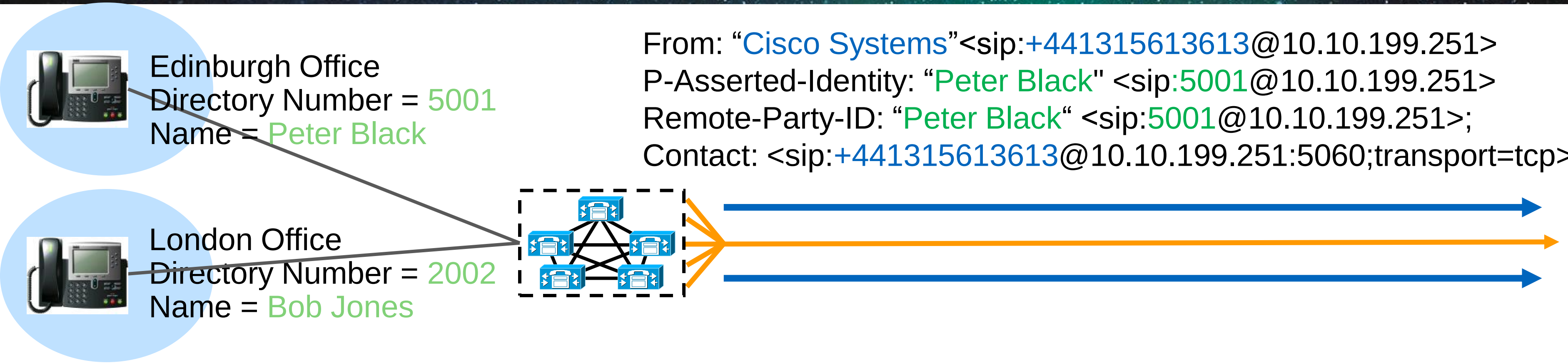
X Maintain Original Caller ID DN and Caller Name in Identity Headers

With multiple Remote Branches sharing a centralised PSTN egress. The Caller ID DN and Name can be configured per site (or per phone if needed) using SIP Profile – phones settings instead of Trunk settings

CUCM SIP Trunk Features

Outbound Calls –Caller DN and Name Information

From: “Cisco Systems”<sip:+441315613613@10.10.199.251>
P-Asserted-Identity: “Peter Black" <sip:5001@10.10.199.251>
Remote-Party-ID: “Peter Black“ <sip:5001@10.10.199.251>;
Contact: <sip:+441315613613@10.10.199.251:5060;transport=tcp>



SIP Profile Configuration — **For Phones in Edinburgh**

Parameters used in Phone

Incoming Requests FROM URI Settings

Caller ID DN: +332125551000

Caller Name: Cisco Systems FR

SIP Profile Configuration — **For Outbound Trunk**

X Allow Passthrough of Configured Line Device Caller Information

For each remote site create a SIP Profile and configure the Caller ID DN and Caller Name. Associate this profile with phones at this site

On the Outbound SIP Trunk SIP Profile

Check the box to “Allow Passthrough of configured Line Device Caller Information”

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