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Cisco Unified Communications Manager SIP Trunk Messaging Guide (Standard)

For Cisco Unified Communications Manager Release 8.6(1)

Americas Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
http://www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 527-0883

Text Part Number: OL-25256-01

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Preface

This document describes the implementation of the Session Initiation Protocol (SIP) for trunk side devices in Cisco Unified Communications Manager.

The preface covers these topics:

- Audience
- Organization
- Conventions
- Obtaining Documentation, Support, and Security Guidelines

Audience

This document provides information for developers, vendors, and customers who are developing applications or products that integrate with Cisco Unified Communications Manager using SIP messaging.

Organization

This document consists of the following chapters.

Chapter	Description
Chapter 1, "SIP Standard Trunk Interface"	Provides an overview of SIP trunk messages and standards compliance.
Chapter 2, "SIP Trunk Call Flows"	Comprises a listing of all SIP trunk messages, including sequence charts and examples of call flows.

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Conventions

This document uses the following conventions:

<table>
<tr><th>Convention</th><th>Description</th></tr>
<tr><td>boldface font</td><td>Commands and keywords are in boldface.</td></tr>
<tr><td>italic font</td><td>Arguments for which you supply values are in italics.</td></tr>
<tr><td>[ ]</td><td>Elements in square brackets are optional.</td></tr>
<tr><td>{ x | y | z }</td><td>Alternative keywords are grouped in braces and separated by vertical bars.</td></tr>
<tr><td>[x | y | z]</td><td>Optional alternative keywords are grouped in brackets and separated by vertical bars.</td></tr>
<tr><td>string</td><td>A nonquoted set of characters. Do not use quotation marks around the string or the string will include the quotation marks.</td></tr>
<tr><td><code>screen</code> font</td><td>Terminal sessions and information the system displays are in <code>screen</code> font.</td></tr>
<tr><td><code>boldface screen</code> font</td><td>Information you must enter is in <code>boldface screen</code> font.</td></tr>
<tr><td>italic screen font</td><td>Arguments for which you supply values are in italic screen font.</td></tr>
<tr><td>→</td><td>This pointer highlights an important line of text in an example.</td></tr>
<tr><td>^</td><td>The symbol ^ represents the key labeled Control—for example, the key combination ^D in a screen display means hold down the Control key while you press the D key.</td></tr>
<tr><td>< ></td><td>Nonprinting characters, such as passwords are in angle brackets.</td></tr>
</table>

Notes use the following conventions:

Note Means *reader take note*. Notes contain helpful suggestions or references to material not covered in the publication.

Caution Means *reader be careful*. In this situation, you might do something that could result in equipment damage or loss of data.

Tip Means *the following information might help you solve a problem*.

Obtaining Documentation, Support, and Security Guidelines

For information on obtaining documentation, obtaining support, providing documentation feedback, security guidelines, and also recommended aliases and general Cisco documents, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation, at:

http://www.cisco.com/en/US/docs/general/whatsnew/whatsnew.html

CHAPTER 1

SIP Standard Trunk Interface

This document describes the standard external interface for Cisco Unified CM SIP trunk device. It highlights the SIP primitives that are supported across the SIP trunk and also describes basic call flow scenarios that can be used as a guide for technical support.

This chapter includes the following sections:

- New and Changed Information
- Backward Compatibility
- Interface Compliance Summary
- SIP Message Fields
- SIP Trunk Supported Features
- Troubleshooting

New and Changed Information

The new features in this release are as follows:

- SIP OPTIONS ping, page 1-54
- Static Call Routing, page 1-54
- SIP Header Enhancements for Recording, page 1-54
- Third Party HD video support, page 1-55
- Early Offer support for SIP Trunk, page 1-55
- Secure Icon Enhancement over SIP Trunk, page 1-58
- Support for Image Attribute in SDP, page 1-59
- Support for Initial Invite Requri Parameter Passthrough, page 1-59
- Support for Blind Transfer Refer Parameter Passthrough, page 1-60
- Support for Contact Header Parameter Passthrough, page 1-60
- MAX-FPS attribute support for H264 Codec, page 1-60
- AMR/AMR-WB Codec Support, page 1-60
- BFCP Support, page 1-61
- CUCM G.722.1 Codec Support, page 1-63
- CUCM AAC-LD MP4A-LATM Codec Support on SIP, page 1-63

Backward Compatibility

The features that are introduced in this release do not impose any backward compatibility implications on previous versions of the SIP trunk.

Interface Compliance Summary

Cisco Unified CM SIP compliance on the SIP trunk depends on the portable SIP stack itself, which is based on the RFC3261 standard.

The current stack supports the following items in RFC3261:

- Can process UPDATE method.
- Support for generating branch and sent-by parameters in Via header used to identify transactions.
- Implementation of loose-routing based on lr parameter in Record-Route header.
- A UAS that receives a second INVITE before it sends the final response to a first INVITE with a lower Cseq sequence number on the same dialog must return a 500 (Server Internal Error) response to the second INVITE and must include a Retry-After header field with a random value of between 0 and 10 seconds.
- If the non-2xx final response to a mid-call INVITE is a 481 (Call/Transaction Does Not Exist), or a 408 (Request Timeout), or no response at all is received for the re-INVITE (that is, a timeout is returned by the INVITE client transaction), the UAC will terminate the dialog.
- If the UAC receives a reliable provisional response with an answer, it may generate an additional offer in the PRACK. If the UAS receives a PRACK with an offer, it must place the answer in the 2xx to the PRACK.
- If a reliable provisional response is retransmitted for 32 seconds without reception of a corresponding PRACK, the UAS should reject the original request with a 5xx response.

Call Manager uses the portable SIP stack from IOS Gateway SunnyD project (EDCS-292452).

Table 1-1 identifies the RFC compliance for the SIP trunk.

Table 1-1 SIP Trunk RFC Compliance

RFC	Call Manager Supported	Comments
RFC2976 SIP INFO Method	Supported	Info method is used for video media channels; Picture Fast Update and Picture Freeze.
RFC2833 RTP Payload for DTMF Digits	Supported	
RFC2782 DNS SRV	Supported	

Table 1-1 SIP Trunk RFC Compliance (continued)

RFC	Call Manager Supported	Comments
RFC3261 SIP: Session Initiation Protocol	Supported	
RFC3262 SIP Reliability of Provisional Responses	Supported	
RFC3264 Offer/Answer Model for SDP	Supported	
RFC3265 Specific Event Notification	Supported	Packages supported: KPML, Presence.
RFC3311 SIP UPDATE Method	Supported	
RFC3515 SIP REFER Method	Partially supported	SIP Trunk accepts inbound REFER's (in-dialog and out-of-dialog). In this release CCM only supports method = INVITE in the Refer-To header. CCM does not support multiple Refer requests within the same dialog.
RFC3842 SIP MWI Package	Partially supported	SIP Trunk supports unsolicited NOTIFY events. It does not Subscribe for MWI events notification.
RFC3856 SIP PRESENCE Event Package	Supported	
RFC3859 Common Profile for Presence	Supported	
RFC3863 Presence Information Data Format	Supported	
RFC3891 SIP Replaces Header	Supported	INVITE w/Replaces and REFER w/Replaces.
RFC3903 SIP PUBLISH Method	Partially supported	SIP Trunk only supports outbound PUBLISH. Inbound PUBLISH is rejected with 405.
RFC4028 Session Timers in the SIP	Supported	For outgoing Invite's, the SIP Trunk indicates the support via Supported header. For incoming Invite's it accepts the Supported and Session-Expires headers.
RFC4480 RPID	Supported	RPID information is carried in the outbound messages if selected via SIP Trunk configuration.
Draft-ietf-sipping-kpml-07.txt	Supported	KPML Event Package (for OOB DTMF).

Table 1-1 SIP Trunk RFC Compliance (continued)

RFC	Call Manager Supported	Comments
Draft-ietf-sip-privacy-05.txt	Supported	Remote Party ID (RPID) Header.
Draft-levy-sip-diversion-08.txt	Supported	Draft-levy-sip-diversion-08. txt.
RFC3323	Supported	A Privacy Mechanism for SIP.
RFC3324	Supported	Short Term Requirements for Network Asserted Identity.
RFC3325	Supported	Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks.
RFC4411	Supported	SIP Trunk supports extending Reason header for Preemption Events.
RFC4412	Partially supported	SIP Trunk only supports outbound Resource-Priority header. A Resource-Priority header is added and will show up in "Require" header field when used.
RFC4040	Supported	A clear channel codec negotiation for SIP Trunk.
RFC4091	Supported	Alternative Network Address Types for advertising both IPv4 and IPv6 media in the SDP.
RFC4092	Supported	Usage of the Session Description Protocol (SDP) Alternative Network Address Types (ANAT) Semantics in the Session Initiation Protocol (SIP).
RFC3388	Partially supported	SIP Trunk only supports sections pertaining to grouping of M Lines for ANAT.
RFC3693	Partially supported	As part of Logical Partitioning feature, CUCM supports Geopriv Location Object specification described in this RFC.

Table 1-1 SIP Trunk RFC Compliance (continued)

RFC	Call Manager Supported	Comments
RFC4119	Partially supported	Extension to RFC3693. CUCM supports PIDF-LO specification as described in RFC 4119.
draft-ietf-sip-location-conveyance-10	Partially supported	CUCM supports Location Conveyance specification described in this ietf draft.
RFC3312	Partially supported	End to End qos is supported but Segmented qos is not supported.
RFC4032	Partially supported	Extension to RFC3312, CUCM supports changing precondition strength based on target changes.
RFC4574	Partially supported	Compliant to RFC 4574 in the context of BFCP application.
RFC4583	Partially supported	Partially compliant as there are deviations from the floorctrl attribute.
draft-sandbakken-xcon-bfcp-upd-01	Supported	SDP format for BFCP. Cisco Unified CM only supports UDP/BFCP implementation based on IETF draft - http://tool.ietf.org/html/draft-sandbakken-xcon-bfcp-udp-01
RFC4796	Partially supported	Compliant to RFC 4796 in the context of BFCP application.

This document identifies the SIP trunk compliance for SIP messages and headers, as described in Table 1-2 through Table 1-8.

Table 1-2 Compliance to SIP Requests

SIP Message	Cisco Unified CM Supported	Comments
INVITE	Yes	The system also supports re-INIVTE.
ACK	Yes	
OPTIONS	Yes	Supports basic OPTIONS ping fonctionality.
INFO	Yes	INFO method gets used for video support.
BYE	Yes	BYE can tunnel QSIG RELCOMP message.

Table 1-2 Compliance to SIP Requests (continued)

SIP Message	Cisco Unified CM Supported	Comments
CANCEL	Yes	CANCEL can tunnel QSIG RELCOMP message.
SUBSCRIBE	Yes	Supported events: kpml, presence
NOTIFY	Yes	Supported events: kpml, presence In addition, Cisco Unified CM supports unsolicited NOTIFY for DTMF and MWI.
REFER	Yes	Cisco Unified CM SIP trunk supports inbound REFER only, both in dialog and out of dialog.
REGISTER	No	The system sends 405 Method Not Allowed for Cisco Unified CM SIP trunk.
PRACK	Yes	The system provides three options: • To send PRACK for Only 180 w/sdp, or • for all 18x messages, or • To disable PRACK entirely via CCM Service Parameter
UPDATE	Yes	Cisco Unified CM supports receiving and generating UPDATE.
PUBLISH	Yes	Cisco Unified CM supports only generating PUBLISH.

Table 1-3 Compliance to SIP Responses

SIP Message	Cisco Unified CM Supported	Comment
1xx Response	Yes	
100 Trying	Yes	
180 Ringing	Yes	The system supports early media.
181 Call Forward	No	Cisco Unified CM sends 181 Call forwarded response when the call forward feature is invoked.
182 Queued	No	Stack drops this message.
183 Progress	Yes	The system supports early media.
2xx Response	Yes	
200 OK	Yes	
202 OK	Yes	For REFER
3xx Response	Yes	
300–302, 305, 380, 385	Yes	The system does not generate these messages, but contacts the new address in Contact header upon receiving.
4xx Response	Yes	Upon receiving, the system initiates a graceful call disconnect.

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Table 1-3 Compliance to SIP Responses (continued)

SIP Message	Cisco Unified CM Supported	Comment
401	Yes	Cisco Unified CM SIP trunk sends out 401 (Unauthorized) if authentication and authorization is enabled. Cisco Unified CM SIP trunk also responds to inbound 401 challenges.
403	Yes	Cisco Unified CM SIP trunk sends a 403 (Forbidden) message if a SIP method is on the Access Control List.
405	Yes	Cisco Unified CM SIP trunk sends a 405 message if the incoming SIP message is not supported.
407	Yes	Cisco Unified CM SIP trunk responds to inbound 407 (Proxy Authentication Required) message challenges.
412	Yes	Cisco Unified CM SIP trunk processes 412 response for PUBLISH.
415	Yes	Cisco Unified CM SIP Trunk sends out 415 (Unsupported Media Type) message when it does not support the media type received in incoming request's SDP.
420	Yes	Cisco Unified CM SIP Trunk sends out 420 (Bad Extension) message when it receives sdp-anat tag in the Require/Supported header and it is not configured for ANAT.
424	Yes	Cisco Unified CM SIP Trunk sends out 424 (Bad Location Information) message when it receives INVITE requests carrying Location Conveyance info with errors against SIP compliance like: • Geolocation Headers indicate inclusion of PIDF-LO, but message body doesn't carry it. • Geolocation header has a cid header referring to a URI for which there is no corresponding Content-ID header with the same URI. • Geolocation header having URI other than cid header e.g SIP or SIPS URI for LbyR.
5xx Response	Yes	Upon receiving, the system sends a new request if additional address is present. Otherwise, it initiates a graceful disconnect.

Table 1-3 Compliance to SIP Responses (continued)

SIP Message	Cisco Unified CM Supported	Comment
580	Yes	CUCM get INVITE with precondition and its local RSVP layer sends reject because the bandwidth is not available, then CUCM responds with 580 Precondition Failed. Upon receiving the 580 response, CUCM will either continue the call with no RSVP involved or call fails based on local RSVP policy configuration.
6xx Response	Yes	The system does not generate this response; upon receiving, the system initiates a graceful disconnect.

Table 1-4 SIP Header Fields

SIP Header	Cisco Unified CM Supported	Comments
Accept	No	
Accept-Encoding	No	
Accept-Language	No	
Alert-Info	No	
Allow	Yes	
Authentication-Info	No	
Authorization	Yes	
Allow-Events	Yes	kpml, presence
Call-Info	Yes	Call-Info is used to transmit the overall security level of the trunk call. For example: Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated Call-Info header is also used for figuring out the IP address of the originating device for SIP trunk identification, if it contains the tag, purpose=x-cisco-origIP. For example: Call-Info: <sip:172.18.200.127>; PURPOSE=x-cisco-origIP
Call-ID	Yes	
Contact	Yes	
Content-Disposition	Yes	Signal; handling=optional.
Content-Encoding	No	
Content-Language	No	
Content-Length	Yes	

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Table 1-4 SIP Header Fields (continued)

SIP Header	Cisco Unified CM Supported	Comments
Content-Type	Yes	Supported as: "sdp", "kpml-request+xml", "media_control+xml", "text/plain", "pidf+xml", "simple-message-summary", and "application/qsig". The system only supports "multipart" for cases where both the SDP and QSIG content is to be tunneled.
CSeq	Yes	
Date	Yes	
Diversion	Yes	The system uses this header for RDNIS information. If it is present, it is always the Original Called Party information. The receiving side of this header always assumes that it is the Original Called Party information if present. In case of chained-forwarding to a voice messing system, the system leaves the message to the Original Called Party.
Encryption	No	
Error-Info	No	
Expires	Yes	
Event	Yes	
From	Yes	Cisco Unified CM adds x-nearend, x-farend, x-refci, x-nearenddevice, x-farenddevice, x-farendaddr, and isfocus (for conference) parameters for Recording INVITEs. and UPDATEs.
Geolocation	Yes	Cisco Unified CM supports this header as part of Logical Partitioning feature.
Geolocation-Error	Yes	Cisco Unified CM supports this header as part of Logical Partitioning feature. This is an informatory header which is sent in SIP response messages.
Hide	No	
In-Reply-To	No	
Max-Forwards	Yes	Cisco Unified CM sets to 70 for outgoing INVITE and does not increment/decrement it.
Min-Expires	Yes	
MIME-Version	No	
Organization	No	
Priority	No	
Proxy-Authenticate	Yes	Cisco Unified CM SIP trunk supports receiving this header in 407 responses.
Proxy-Authorization	Yes	Cisco Unified CM SIP trunk supports sending new request with this header after receiving 407 responses.
Proxy-Require	No	

Table 1-4 SIP Header Fields (continued)

SIP Header	Cisco Unified CM Supported	Comments
P-Asserted-Identity	Yes	
P-Preferred-Identity	Yes	
RAck	Yes	
Record-Route	Yes	
Remote-Party-ID	Yes	The system uses this header for ID services including Connected Name and ID. This is a non-standard header from a draft specification.
Replaces	Yes	For INIVTE and REFER.
Require	Yes	
Response-Key	No	
Retry-After	Yes	The system sends it but ignores receiving it.
Resource-Priority	Yes	
Route	Yes	
RSeq	Yes	
Server	Yes	
SIP-If-Match	Yes	For PUBLISH
SIP-Etag	Yes	For PUBLISH
Subject	No	
Supported	Yes	
Subscription-State	Yes	
Timestamp	Yes	
To	Yes	
Unsupported	Yes	
User-Agent	Yes	
Via	Yes	
Warning	Yes	
WWW-Authenticate	Yes	

Table 1-5 Supported Audio Media Types

Type	Encoding Name	Payload Type	Comments
G.711 u-law	PCMU	0	
GSM Full-rate	GSM	3	
G.723.1	G723	4	
G.711 A-law	PCMA	8	
G.722	G722	9	
G.728	G728	15	
G.729	G729	18	The system supports all combinations of annex A and B.
AAC	mpeg4-generic	Dynamically Assigned	Acceptable range is 96–127.
ILBC	iLBC	Dynamically Assigned	Acceptable range is 96–127.
RFC2833 DTMF	Telephony-event	Dynamically Assigned	Acceptable range is 96–127.
G.Clear	G.Clear	Dynamically Assigned	Acceptable range is 96–127. Cisco Unified CM default is 125.
iSAC	iSAC	Dynamically Assigned	Acceptable range is 96–127.
NoAudio (negociable in v1.50 MER)	NoAudio	Dynamically Assigned	Currently hard-coded to 126.
AAC-LD	MP4A-LATM	Dynamically Assigned	Acceptable range is 96-127.
G.722.1	G7221	Dynamically Assigned	Acceptable range is 96-127

Table 1-6 Supported Video Media Types

Types	Encoding Name	Payload Type
H.261	H261	31
H.263	H263	34
H.263+	H263-1998	Acceptable range is 96–127.
H.263++	H263-2000	Acceptable range is 96–127.
H.264	H264	Acceptable range is 96–127.

Table 1-7 Supported Application Media Type

Types	Encoding Name	Payload Type
H.224 FECC	H224	Acceptable range is 96–127

Table 1-8 Supported T38fax Payload Type

Types	Encoding Name	Payload Type
T38fax	Not applied	Not applied

SIP Message Fields

The SIP trunk supports SIP request and SIP response messages. The request messages include INVITE, ACK, OPTIONS, BYE, CANCEL, PRACK, SUBSCRIBE, UPDATE, and REFER methods. The response message consists of a status-line with various status codes (1xx, 2xx, 3xx, 4xx, 5xx and 6xx). The SIP trunk supports all mandatory fields from the SIP standard. See Table 1-9 through Table 1-24.

Request Messages

INVITE

Table 1-9 INVITE Message Fields

Field	Example	Notes
Request-Line	`INVITE sip:cdpn@destIP:destPort; phone-context=cisco.com; tgrp=ccdata;user=phone SIP/2.0`	destIP=resolved IP address of configured DestAddr under SIPTrunk; it also could be FQDN SRV instead of destIP. If the address is an IPv6 address then it will be specified within [ ] square brackets. destPort=configured destPort or resolved port from DNS SRV. cgpn for outgoing INVITE, and cdpn for incoming INVITE. “phone-context=cisco.com; tgrp=ccdata;”= When MTP is not enabled and the G.Clear feature is enabled, these tags are added to the Request URI. Both of these values are user provisionable.

Table 1-9 INVITE Message Fields (continued)

Field	Example	Notes
From	From: "callerName" <sip:cgpn@CCM_IP_addr>; x-nearend; x-farend; x-refCI=CI_NUMBER; x-devicename=SEP_NAME	callerName = caller display i.e. IP phone Cgpn = calling party number CCM_IP_addr = CCM IP address (Depending on if this is a IPv4 or an IPv6 call, the appropriate Cisco Unified CM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets. x-nearend = Recorded party is the near-end. x-farend = Recorded party is the far-end x-refCI = Cisco Unified CM call identifier for recorded party x-devicename = Device name of recorded party.
To	To: "calledName" <sip:cdpn@destIP;user=phone>	The system includes calledName if available. destIP=resolved IP address of configured DestAddr/DestV6Addr under SIPTrunk; it could also be FQDN SRV instead of destIP.
Via	Via:SIP/2.0 IP addr:Port;Branch=number	IP addr = Cisco Unified CM IP (Depending on if this is a IPv4 or an IPv6 call, the appropriate CCM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets. Port = CCM Port Branch = Unique number
Call-ID	Call-ID: number@CCM_IP_addr	Cisco Unified CM generates the number internally.
Contact	Contact: <sip:cgpn@CCM_IP_addr:localPort;user=phone>	localPort = configured "Incoming port" of the indicated SIPTrunk
Cseq	Cseq:number method	Number = a traditional sequence number that is incremented for each new request within a dialog Method = INVITE
Max-Forwards	Max-Forwards:number	Number = 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 by sip proxy. CCM sets it 6 for outgoing INVITE.

Table 1-9 INVITE Message Fields (continued)

Field	Example	Notes

Remote-Party-Id	**IPv4 Example**
	
	`Remote-Party-ID:"Alice Smith" <sip:9728135111@161.44.147.67;user=phone>; party=calling;screen=no;privacy=off`
	
	IPv6 Example
	
	`Remote-Party-ID:"Alice Smith" <sip:9728135111@[2001:db8:1:101::12];user=phone>; party=calling;screen=no;privacy=off`
		Cisco Unified CM uses this SIP extension for more detailed description of Caller Identify and Privacy. It is also used to convey Connected Name and ID in a re-Invite message.

Diversion	**IPv4 Example**
	
	`Diversion: <sip:23222@172.18.193.123>;reason=no-answer`
	
	IPv6 Example
	
	`Diversion:<sip:23222@[2001:db8:1:101::12]>; reason=no-answer`
		Cisco Unified CM uses Diversion header to carry RDNIS. In this case, 23222 will be carried as the Original Called Party ID.

Call-Info	**IPv4 Example**
	
	`Call-Info: <sip:172.18.199.211>;purpose=x-cisco-origIP`
	
	IPv6 Example
	
	`Call-Info: <sip:[2001:db8:1:101::12]>;purpose=x-cisco-origIP`
		If this header is present in an incoming INVITE with the tag, purpose=x-cisco-origIP, and the trunk is configured to route the call based on the end device's IP Address, then this header will be used to route the call to the Trunk pointing to the IP Address in the Call-Info header.

P-Asserted-Identity	**IPv4 Example**
	
	`P-Asserted-Identity: "Alice" <sip:4762424@172.18.199.211>`
	
	IPv6 Example
	
	`P-Asserted-Identity: "Alice" <sip:4762424@[2001:db8:1:101::12]>`
		Cisco Unified CM uses this header to convey caller name/number information.
		
		In the re-Invite message, it is used to convey the connected name/number.
		
		Cisco Unified CM adds the PAI header if the SIP trunk is configured with Asserted-Type=PAI or Asserted-Type=Default and the call control in Cisco Unified CM provides the screening indication of UserProvided “VerifiedAndPassed” or NetworkProvided.
		
		PAI header values are displayed per the user configuration.

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Table 1-9 INVITE Message Fields (continued)

Field	Example	Notes

P-Preferred-Identity	**IPv4 Example**
	`P-Preferred-Identity: "Alice" <sip:4762424@172.18.199.211>`
	IPv6 Example
	`P-Preferred-Identity: "Alice" <sip:4762424@[2001:db8:1:101::12]>`
		Cisco Unified CM uses this header to convey caller name/number information.
		In the re-Invite message, it is used to convey the connected name/number.
		Cisco Unified CM adds the PAI header if the SIP trunk is configured with Asserted-Type=PPI or Asserted-Type=Default and when the call control in Cisco Unified CM provides the screening indication of “Not Screened” or “VerifiedAndFailed” values.
		PPI header values are displayed as per the user configuration.

Resource-Priority	`Resource-Priority: DRSN.9`	If the MLPP feature is enabled, the Resource-Priority header is added to the initial INVITE.

Supported	`Supported: option-tag`
	`Example:`
	`Supported: 100rel,timer,resource-priority,replaces`
		Cisco Unified CM adds Supported header in outgoing INVITE request when additional options like PRACK, Presence, Resource-Priority etc. are configured to indicate other side about Cisco Unified CM’s capability.

Geolocation	`Geolocation: cid-url;inserted-by=hostport`
	`Example:`
	`Geolocation: <cid:4900@10.10.10.10>;inserted-by="10.10.10.10`
		SIP Trunk only supports sending and processing of cid-url in Geolocation header, though specification allows SIP, SIPS & pres URI's.

Content-Disposition	`Signal; handling=optional`	The system adds this header when QSIG tunneling is enabled.

Table 1-9 INVITE Message Fields (continued)

Field	Example	Notes
SDP	**IPv4 SDP** v=0 o=CiscoSystemsCCM-SIP 2000 1000 IN IP4 10.89.79.203 s=SIP Call c=IN IP4 10.89.79.203 t=0 0 m=audio 32314 RTP/AVP 0 101 a=rtpmap:0 PCMU/8000 a=ptime:20 a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-15 m=audio 16604 RTP/AVP 125 a=rtpmap:125 CLEARMODE/8000 **IPv6 SDP** v=0 o=CiscoSystemsCCM-SIP 2000 1000 IN IP6 2001:db8:c18:1:21c:58ff:fe2a:23f8 s=SIP Call c=IN IP6 2001:db8:c18:1:21c:58ff:fe2a:23f3 t=0 0 m=audio 32314 RTP/AVP 0 101 a=rtpmap:0 PCMU/8000 a=ptime:20 a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-15 **ANAT SDP with both IPv4 and IPv6 addresses** v=0 o=CiscoSystemsCCM-SIP 2000 1 IN IP4 172.18.199.129 s=SIP Call t=0 0 a=group:ANAT 1 2 m=audio 18484 RTP/AVP 0 101 c=IN IP6 2001:db8:c18:1:21c:58ff:fe2a:23f8 a=rtpmap:0 PCMU/8000 a=ptime:20 a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-15 a=mid:1 m=audio 18282 RTP/AVP 0 101 c=IN IP4 172.18.199.55 a=rtpmap:0 PCMU/8000 a=ptime:20 a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-15 a=mid:2	If MTP is not enabled for the SIP trunk, SDP is not in the initial INVITE. If MTP is enabled, Cisco Unified CM always include telephone-event for RFC2833 DTMF in the SDP. This dynamic payload type is configurable under Cisco Unified CM Service Parameter with default value as 101. For a SIP Trunk in IPv4 only mode, the SDP will contain the IPv4 address of the MTP device For a SIP Trunk in IPv6 only mode, the SDP will contain the IPv6 address of the MTP device For a Dual Mode SIP Trunk, if ANAT is enabled then, the SDP will contain both the IPv4 address, and the IPv6 address using the ANAT format. If MTP is enabled and G.Clear feature is enabled, CCM sends G.Clear mode information in the SDP in the initial INVITE.
QSIG Content	**Binary body**	QSIG content in SIP message is encoded as binary.

ACK

Table 1-10 ACK Message Fields

Field	Example	Notes
Request-Line	`ACK sip:cdpn@destIP:destPort;SIP/2.0`	destIP = resolved IP address of configured DestAddr/DestAddrIPv6 under SIPTrunk, it could also be FQDN SRV instead of destIP; If the address is an IPv6 address then it will be specified within [ ] square brackets. destPort=configured destPort or resolved port from DNS SRV.
From	`From: "callerName" <sip:cgpn@CCM_IP_addr>`	callerName=caller display i.e. IP phone Cgpn = calling party number CCM_IP_addr=CCM IP address (Depending on if this is a IPv4 or an IPv6 call, the appropriate CCM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets.
To	`To: "calledName"<sip:cdpn@destIP;user=phone>`	calledName is included if available destIP=resolved IP address of configured DestAddr/DestAddrIPv6 under SIPTrunk; it could also be FQDN SRV instead of destIP; If the address is an IPv6 address then it will be specified within [ ] square brackets.
Via	`Via:SIP/2.0 IP addr:Port;Branch=number`	IP addr=CCM IP (Depending on if this is a IPv4 or an IPv6 call, the appropriate CCM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets. Port=CCM Port Branch=Unique number
Call-ID	`Call-ID: number@CCM IP addr`	Cisco Unified CM generates this number internally.
CSeq	`Cseq: number method`	Number=a traditional sequence number that is incremented for each new request within a dialog Method=ACK
Max-Forwards	`Max-Forwards:number`	Number= 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.

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BYE

Table 1-11 BYE Message Fields

Field	Example	Notes
Request-Line	`BYE sip:cdpn@destIP:destPort;SIP/2.0`	destIP = resolved IP address of configured DestAddr/DestAddrIPv6 under SIPTrunk, it could also be FQDN SRV instead of destIP; If the address is an IPv6 address then it will be specified within [ ] square brackets. destPort=configured destPort or resolved port from DNS SRV;
From	`From: "callerName" <sip:cgpn@CCM_IP_addr>`	callerName=caller display i.e. IP phone Cgpn = calling party number CCM_IP_addr=Cisco Unified CM IP address (Depending on if this is a IPv4 or an IPv6 call, the appropriate Cisco Unified CM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets.
To	`To: "calledName"<sip:cdpn@destIP;user=phone>`	calledName is included if available destIP=resolved IP address of configured DestAddr/DestAddrIPv6 under SIPTrunk; it could also be FQDN SRV instead of destIP; If the address is an IPv6 address then it will be specified within [ ] square brackets.
Via	`Via:SIP/2.0 IP addr:Port;Branch=number`	IP addr=Cisco Unified CM IP (Depending on if this is a IPv4 or an IPv6 call, the appropriate Cisco Unified CM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets. Port=CCM Port Branch=Unique number
Call-ID	`Call-ID: number@CCM IP addr`	Cisco Unified CM generates the number internally.
CSeq	`Cseq: number method`	Number=a traditional sequence number that is incremented for each new request within a dialog Method=BYE
Max-Forwards	`Max-Forwards:number`	Number = 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

Table 1-11 BYE Message Fields (continued)

Field	Example	Notes
Reason	`Reason: preemption;cause=<cause value>;text=<text string>`	Reason header is included when MLPP feature is enabled. Cause value and test string are defined as per RFC4411.
Content-Type	`Application/QSIG`	QSIG message REL COMP is tunneled in BYE only when QSIG tunneling is enabled.
Content-Disposition	`Signal;handling=optional`	Only when QSIG tunneling is enabled.

CANCEL

Table 1-12 CANCEL Message Fields

Field	Example	Notes
Request-Line	`CANCEL sip:cdpn@destIP:destPort;SIP/2.0`	destIP = resolved IP address of configured DestAddr/DestAddrV6 under SIPTrunk, it could also be FQDN SRV instead of destIP; If the address is an IPv6 address then it will be specified within [ ] square brackets. destPort=configured destPort or resolved port from DNS SRV.
From	`From: "callerName" <sip:cgpn@CCM_IP_addr>`	callerName=caller display i.e. IP phone Cgpn = calling party number CCM_IP_addr=Cisco Unified CM IP address (Depending on if this is a IPv4 or an IPv6 call, the appropriate Cisco Unified CM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets.
To	`To: "calledName"<sip:cdpn@destIP;user=phone>`	calledName is included if available destIP=resolved IP address of configured DestAddr/DestAddrIPv6 under SIPTrunk; it could also be FQDN SRV instead of destIP; If the address is an IPv6 address then it will be specified within [ ] square brackets.
Via	`Via:SIP/2.0 IP addr:Port;Branch=number`	IP addr=Cisco Unified CM IP (Depending on if this is a IPv4 or an IPv6 call, the appropriate CCM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets. Port=Cisco Unified CM Port Branch=Unique number

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Table 1-12 CANCEL Message Fields (continued)

Field	Example	Notes
Call-ID	`Call-ID: number@CCM IP addr`	Cisco Unified CM generates the number internally.
CSeq	`Cseq: number method`	Number=a traditional sequence number that is incremented for each new request within a dialog Method=CANCEL
Max-Forwards	`Max-Forwards:number`	Number= 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
Content-Type	`Application/QSIG`	QSIG message REL COMP is tunneled in BYE only when QSIG tunneling is enabled.
Content-Disposition	`Signal;handling=optional`	Only when QSIG tunneling is enabled.

PRACK

Table 1-13 PRACK Message Fields

Field	Example	Notes
Request-Line	`PRACK sip:cdpn@destIP:destPort;SIP/2.0`	destIP = resolved IP address of configured DestAddr/DestAddrIPv6 under SIPTrunk, it could also be FQDN SRV instead of destIP; If the address is an IPv6 address then it will be specified within [ ] square brackets. destPort=configured destPort or resolved port from DNS SRV;
From	`From: "callerName" <sip:cgpn@CCM_IP_addr>`	callerName=caller display i.e. IP phone Cgpn = calling party number CCM_IP_addr=CCM IP address (Depending on if this is a IPv4 or an IPv6 call, the appropriate CCM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets.
To	`To: "calledName"<sip:cdpn@destIP;user=phone>`	calledName is included if available destIP=resolved IP address of configured DestAddr/DestAddrIPv6 under SIPTrunk; it could also be FQDN SRV instead of destIP; If the address is an IPv6 address then it will be specified within [ ] square brackets.

Table 1-13 PRACK Message Fields (continued)

Field	Example	Notes
Via	`Via:SIP/2.0 IP addr:Port;Branch=number`	IP addr=CCM IP (Depending on if this is a IPv4 or an IPv6 call, the appropriate CCM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets. Port=CCM Port Branch=Unique number
Call-ID	`Call-ID: number@CCM IP addr`	Cisco Unified CM generates the number internally.
CSeq	`Cseq: number PRACK`	Number=a traditional sequence number that is incremented for each new request within a dialog Method=Method Name
Rack	`Rack:number1 number2`	Number1=value from the RSeq header in the provisional response that is being acknowledged Number2=The next number, and the method, are copied from the CSeq in the response that is being acknowledged.

UPDATE

Table 1-14 UPDATE Message Fields

Field	Example	Notes
Request-Line	`UPDATE sip:cdpn@destIP:destPort;user=phone SIP/2.0`	destIP = resolved IP address of configured DestAddr/DestAddrIPv6 under SIPTrunk, it could also be FQDN SRV instead of destIP; If the address is an IPv6 address then it will be specified within [ ] square brackets. destPort=configured destPort or resolved port from DNS SRV;
From	`From: "callerName" <sip:cgpn@CCM_IP_addr>`	callerName=caller display i.e. IP phone Cgpn = calling party number CCM_IP_addr=CCM IP address (Depending on if this is a IPv4 or an IPv6 call, the appropriate CCM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets.

Table 1-14 UPDATE Message Fields (continued)

Field	Example	Notes
To	`To: "calledName"<sip:cdpn@destIP;user=phone>`	calledName is included if available destIP=resolved IP address of configured DestAddr/DestAddrIPv6 under SIPTrunk; it could also be FQDN SRV instead of destIP; If the address is an IPv6 address then it will be specified within [ ] square brackets.
Via	`Via:SIP/2.0 IP addr:Port;Branch=number`	IP addr=CCM IP (Depending on if this is a IPv4 or an IPv6 call, the appropriate CCM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets. Port=CCM Port Branch=Unique number
Call-ID	`Call-ID: number@CCM IP addr`	Cisco Unified CM generates the number internally.
Contact	`Contact: <sip:cgpn@CCM_IP_addr:localPort;user=phone>`	localPort = configured "Incoming port" of the indicated SIPTrunk
Cseq	`Cseq:number method`	Number=a traditional sequence number that is incremented for each new request within a dialog Method=UPDATE
Max-Forwards	`Max-Forwards:number`	Number= 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

Remote-Party-Id	`IPv4 Example`
	
	`Remote-Party-ID:"Alice Smith" <sip:9728135111@161.44.147.67;user=phone>; party=calling;screen=no;privacy=off`
	
	`IPv6 Example`
	
	`Remote-Party-ID:"Alice Smith" <sip:9728135111@[2001:db8:1:101::12];user=phone>; party=calling;screen=no;privacy=off`
		Cisco Unified CM uses this SIP extension for more detailed description of Caller Identify and Privacy. It is also used to convey Connected Name & ID in a re-Invite message.

P-Asserted-Identity	`IPv4 Example`
	
	`P-Asserted-Identity: "Alice" <sip:4762424@172.18.199.211>`
	
	`IPv6 Example`
	
	`P-Asserted-Identity: "Alice" <sip:4762424@[2001:db8:1:101::12]>`
		Cisco Unified CM uses this header for conveying the Connected Name/Number.
		Cisco Unified CM adds the PAI header, if SIP Trunk is configured with Asserted-Type=PAI or Asserted-Type=Default and when the call control in Cisco Unified CM provides the screening indication of UserProvided "VerifiedAndPassed" or NetworkProvided..

Table 1-14 UPDATE Message Fields (continued)

Field	Example	Notes

P-Preferred-Identity	`IPv4 Example`
	
	`P-Preferred-Identity: "Alice" <sip: 4762424@172.18.199.211>`
	
	`IPv6 Example`
	
	`P-Preferred-Identity: "Alice" <sip: 4762424@[2001:db8:1:101::12]>`
		Cisco Unified CM uses this header for conveying the Connected Name/Number.
		
		Cisco Unified CM adds the PPI header, if SIP Trunk is configured with Asserted-Type=PPI or Asserted-Type=Default and when the call control in Cisco Unified CM provides the screening indication of "Not Screened" or "VerifiedAndFailed" values.
		
		PPI header values are displayed as per the user configuration.

Resource-Priority	`Resource-Priority: DRSN.9`	If the MLPP feature is enabled, Resource-Priority header is added to the UPDATE message.

Geolocation	`Geolocation: cid-url;inserted-by=hostport`
	
	`Example:`
	`Geolocation: <cid:4900@10.10.10.10>;inserted-by="10.10.10.10`
		SIP Trunk only supports sending and processing of cid-url in Geolocation header, though specification allows SIP, SIPS & pres URI's.

SUBSCRIBE

Table 1-15 SUBSCRIBE Message Fields

Field	Example	Notes
Request-Line	`SUBSCRIBE sip:subscribee@destIP:destPort SIP/2.0`	destIP = resolved IP address of configured DestAddr/DestAddrIPv6 under SIPTrunk, it could also be FQDN SRV instead of destIP; If the address is an IPv6 address then it will be specified within [ ] square brackets. destPort=configured destPort or resolved port from DNS SRV;
From	`From: "callerName" <sip:cgpn@CCM_IP_addr>`	callerName=caller display i.e. IP phone Cgpn = calling party number CCM_IP_addr=CCM IP address (Depending on if this is a IPv4 or an IPv6 call, the appropriate CCM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets.
To	`To: "calledName"<sip:cdpn@destIP>`	calledName is included if available destIP=resolved IP address of configured DestAddr/DestAddrIPv6 under SIPTrunk; it could also be FQDN SRV instead of destIP; If the address is an IPv6 address then it will be specified within [ ] square brackets.

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Table 1-15 SUBSCRIBE Message Fields (continued)

Via	`Via:SIP/2.0 IP addr:Port;Branch=number`	IP addr=CCM IP (Depending on if this is a IPv4 or an IPv6 call, the appropriate CCM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets. Port=CCM Port Branch=Unique number
Call-ID	`Call-ID: number@CCM IP addr`	Cisco Unified CM generates the number internally.
Contact	`Contact: <sip:cgpn@CCM_IP_addr:localPort>`	localPort = configured "Incoming port" of the indicated SIPTrunk
Cseq	`Cseq:number method`	Number=a traditional sequence number that is incremented for each new request within a dialog Method=SUBSCRIBE
Expires	`Expires: number`	Number = The duration of the subscription, in seconds.
Max-Forwards	`Max-Forwards:number`	Number= 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 by sip proxy.
Event:	`kpml, presence`	The Event Type that the Subscribe if for. Unified CM supports kpml and presence event packages.
Content-Type:	`application/kpml-request+xml or message/sipfrag;version=2.0`	Unified CM SIP Trunk supports message/sipfrag;version=2.0, application/kpml-request+xml and application/pidf+xml

NOTIFY

Table 1-16 NOTIFY Message Fields

Field	Example	Notes
Request-Line	`NOTIFY sip:subscriber@destIP:destPort SIP/2.0`	destIP = resolved IP address of configured DestAddr/DestAddrIPv6 under SIPTrunk, it could also be FQDN SRV instead of destIP; If the address is an IPv6 address then it will be specified within [ ] square brackets. destPort=configured destPort or resolved port from DNS SRV;

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Table 1-16 NOTIFY Message Fields (continued)

Field	Example	Notes
From	From: "callerName" <sip:cgpn@CCM_IP_addr>	callerName=caller display i.e. IP phone Cgpn = calling party number CCM_IP_addr=CCM IP address (Depending on if this is a IPv4 or an IPv6 call, the appropriate CCM IP Address will be used). If the address is an IPv6 address then it will be specified within [ ] square brackets.
To	To: "calledName"<sip:cdpn@destIP>	calledName is included if available destIP=resolved IP address of configured DestAddr/DestAddrIPv6 under SIPTrunk; it could also be FQDN SRV instead of destIP. If the address is an IPv6 address then it will be specified within [ ] square brackets.
Via	Via:SIP/2.0 IP addr:Port;Branch=number	IP addr=CCM IP (Depending on if this is a IPv4 or an IPv6 call, the appropriate CCM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets. Port=CCM Port Branch=Unique number
Call-ID	Call-ID: number@CCM IP addr	Number is generated internally by Cisco Unified CM.
Contact	Contact: <sip:cgpn@CCM_IP_addr:localPort>	localPort = configured "Incoming port" of the indicated SIPTrunk.
Cseq	Cseq:number method	Number=a traditional sequence number that is incremented for each new request within a dialog Method=NOTIFY
Subscription-State:	Subscription-State:state-value; expires=number	State-value=active|pending|terminated Expires= authoritative subscription duration.
Max-Forwards	Max-Forwards:number	Number= 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 by SIP proxy.
Event	kpml, presence	The Event Type that the Subscribe if for. Cisco Unified CM supports kpml and presence event packages.
Content-Type:	application/kpml-request+xml or message/sipfrag;version=2.0	Cisco Unified CM SIP Trunk supports message/sipfrag;version=2.0, application/kpml-request+xml and application/pidf+xml

PUBLISH

Table 1-17 PUBLISH Message Fields

Field	Example	Notes
Request-Line	`PUBLISH sip:user@destIP:destPort SIP/2.0`	user=end user name associated with a line appearance; destIP = resolved IP address of configured DestAddr/DestAddrIPv6 under SIPTrunk, it could also be FQDN SRV instead of destIP; If the address is an IPv6 address then it will be specified within [ ] square brackets. destPort=configured destPort or resolved port from DNS SRV;
From	`From: <sip:user@CCM_IP_addr>`	user=end user name associated with a line appearance; CCM_IP_addr=Cisco Unified CM IP address (Depending on if this is a IPv4 or an IPv6 call, the appropriate Cisco Unified CM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets.
To	`To: <sip:user@destIP>`	user=end user name associated with a line appearance; destIP=resolved IP address of configured DestAddr/DestAddrIPv6 under SIPTrunk; it could also be FQDN SRV instead of destIP; If the address is an IPv6 address then it will be specified within [ ] square brackets.
Via	`Via:SIP/2.0 IP addr:Port;Branch=number`	IP addr=Cisco Unified CM IP (Depending on if this is a IPv4 or an IPv6 call, the appropriate Cisco Unified CM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets. Port=Cisco Unified CM Port Branch=Unique number
Call-ID	`Call-ID: number@CCM IP addr`	Cisco Unified CM generates the number internally.
Cseq	`Cseq:number method`	Number=a traditional sequence number that is incremented for each new request within a dialog Method=PUBLISH
Max-Forwards	`Max-Forwards:number`	Number= 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 by sip proxy.

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Table 1-17 PUBLISH Message Fields (continued)

Field	Example	Notes
Event	`presence`	Cisco Unified CM supports presence event for PUBLISH.
Expires	`Expires: number`	number=suggested expiration time for PUBLISH in seconds. Default is 3600.
Content-Type:	`application/pidf+xml`	

OPTIONS

Table 1-18 OPTIONS Message Fields

Field	Example	Notes
Request-Line	`OPTIONS sip:destIP:destPort SIP/2.0`	destIP=resolved IP address of configured DestAddr/DestAddrIPv6 under SIPTrunk, it could also be FQDN SRV instead of destIP; If the address is an IPv6 address then it will be specified within [ ] square brackets. destPort=configured destPort or resolved port from DNS SRV;
From	`From: <sip:CCM_IP_addr>;tag=number`	CCM_IP_addr=Cisco Unified CM IP address (Depending on if this is a IPv4 or an IPv6 call, the appropriate Cisco Unified CM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets. Tag=unique number.
To	`To: <sip:destIP>`	destIP=resolved IP address of configured DestAddr/DestAddrIPv6 under SIPTrunk; it could also be FQDN SRV instead of destIP; If the address is an IPv6 address then it will be specified within [ ] square brackets.
Via	`Via:SIP/2.0/Transport_type IP addr:Port;Branch=number`	Transport_type=TCP or UDP. IP addr=Cisco Unified CM IP (Depending on if this is a IPv4 or an IPv6 call, the appropriate Cisco Unified CM IP Address will be used) If the address is an IPv6 address then it will be specified within [ ] square brackets. Port=Cisco Unified CM Port Branch=Unique number
Call-ID	`Call-ID: number@CCM IP addr`	Cisco Unified CM generates the number internally.

Table 1-18 OPTIONS Message Fields (continued)

Field	Example	Notes
Cseq	`Cseq:number method`	Number=a traditional sequence number that is incremented for each new request within a dialog Method=OPTIONS
Max-Forwards	`Max-Forwards:0`	Number= 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 by sip proxy.

Response Messages

18x

Table 1-19 18X Message Fields

Field	Example	Notes

Remote-Party-Id	`IPv4 Example`
	
	`Remote-Party-ID:"Bob Jones" <sip:9728135111@161.44.147.67;user=phone>; party=called;screen=no;privacy=off`
	
	`IPv6 Example`
	
	`Remote-Party-ID:"Bob Jones" <sip:9728135111@[2001:db8:1:101::12];user=phone>; party=called;screen=no;privacy=off`
		Cisco Unified CM uses this SIP extension to convey Connected Name and ID information.

P-Asserted-Identity	`IPv4 Example`
	
	`P-Asserted-Identity: "Alice" <sip:4762424@172.18.199.211>`
	
	`IPv6 Example`
	
	`P-Asserted-Identity: "Alice" <sip:4762424@[2001:db8:1:101::12]>`
		Cisco Unified CM uses this header for conveying the Alerting Name/Number.
		Cisco Unified CM adds the PAI header, if SIP Trunk is configured with Asserted-Type=PAI or Asserted-Type=Default and when the call control in Cisco Unified CM provides the screening indication of UserProvided "VerifiedAndPassed" or NetworkProvided.
		PAI header values are displayed as per the user configuration.

Table 1-19 18X Message Fields

Field	Example	Notes

P-Preferred-Identity	`IPv4 Example`
	
	`P-Preferred-Identity: "Alice" <sip: 4762424@172.18.199.211>`
	
	`IPv6 Example`
	
	`P-Preferred-Identity: "Alice" <sip: 4762424@[2001:db8:1:101::12]>`
		Cisco Unified CM uses this header for conveying the Alerting Name/Number.
		
		Cisco Unified CM adds the PPI header, if SIP Trunk is configured with Asserted-Type=PPI or Asserted-Type=Default and when the call control in Cisco Unified CM provides the screening indication of "Not Screened" or "VerifiedAndFailed" values.
		
		PPI header values are displayed as per the user configuration.

Content-Disposition	`Signal;handling=optional`	Added only when QSIG tunneling is enabled.

SDP	`m=audio 30844 RTP/AVP 0 101`
	`a=rtpmap:0 pcmu/8000`
	`a=rtpmap:101 telephone-event/8000`
	`a=fmtp:101 0-11`
		If a call resulted in early media setup (that is, a SIP to MGCP PRI call), Cisco Unified CM includes SDP answer in 183 message.

QSIG	`Binary content`	If QSIG tunneling is enabled on the SIP tunk, then 183 message will have tunneled PROGRESS, ALERT, or DISCONNECT/RELEASE/COMPLETE message. Cisco Unified CM always tunnels PROGRESS only. Other messages are tunneled when interworking with the IOS SIP gateway.

2xx

Table 1-20 2XX Message Fields

Field	Example	Notes

Remote-Party-Id	`IPv4 Example`
	
	`Remote-Party-ID:"Bob Jones" <sip:9728135111@161.44.147.67;user=phone>; party=called;screen=no;privacy=off`
	
	`IPv6 Example`
	
	`Remote-Party-ID:"Bob Jones" <sip:9728135111@[2001:db8:1:101::12];user=phone>; party=called;screen=no;privacy=off`
		Cisco Unified CM uses this SIP extension to convey Connected Name and ID information.

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Table 1-20 2XX Message Fields (continued)

Field	Example	Notes
P-Asserted-Identity	IPv4 Example P-Asserted-Identity: "Alice" <sip:4762424@172.18.199.211> IPv6 Example P-Asserted-Identity: "Alice" <sip:4762424@[2001:db8:1:101::12]>	Cisco Unified CM uses this header for conveying the Connected Name/Number. Cisco Unified CM adds the PAI header, if SIP Trunk is configured with Asserted-Type=PAI or Asserted-Type=Default and when the call control in Cisco Unified CM provides the screening indication of UserProvided "VerifiedAndPassed" or NetworkProvided. PAI header values are displayed as per the user configuration.
P-Preferred-Identity	IPv4 Example P-Preferred-Identity: "Alice" <sip: 4762424@172.18.199.211> IPv6 Example P-Preferred-Identity: "Alice" <sip: 4762424@[2001:db8:1:101::12]>	Cisco Unified CM uses this header for conveying the Connected Name/Number. Cisco Unified CM adds the PPI header, if SIP Trunk is configured with Asserted-Type=PPI or Asserted-Type=Default and when the call control in Cisco Unified CM provides the screening indication of "Not Screened" or "VerifiedAndFailed" values. PPI header values are displayed as per the user configuration.
Content-Disposition	Signal;handling=optional	Added only when QSIG tunneling is enabled.
QSIG	Binary content	If QSIG tunneling is enabled on the SIP tunk, then 200 OK response will have tunneled CONNECT.

3xx

3xx responses give information about the new user location or about alternative services that might be able to satisfy the call.

Table 1-21 3XX Message Fields

Field	Example	Notes
Status Code	SIP/2.0 302 Moved Temporarily	The requesting client SHOULD retry the request at the new address(es) given by the Contact header field.
From	IPv4 Example From: <sip: 1101@10.89.79.203>;tag=16777234 IPv6 Example From: <sip: 1101@[2001:db8:1:101::12]>;tag=16777234	10.89.79.203 is CCM IPv4 address 2001:db8:1:101::12 is CCM IPv6 Address 16777234 is the Call Id

Table 1-21 3XX Message Fields (continued)

Field	Example	Notes

To	`IPv4 Example`
	
	`To: <sip:30000@10.89.73.75>;tag=0002fd06e9300108228d58c1-614f99be`
	
	`IPv6 Example`
	`To: <sip:30000@[2001:db8:1:2::12]>;tag=0002fd06e9300108228d58c1-614f99be`
		3000 is the Calling Party Number

Via	`IPv4 Example`
	
	`Via: SIP/2.0/TCP 10.89.79.203:5060;received=10.89.79.203;branch=z9hG4bKfe8d27ec`
	
	`IPv6 Example`
	
	`Via: SIP/2.0/TCP [2001:db8:1:101::12]:5060;branch=z9hG4bKfe8d27ec`
		10.89.79.203=CCM IPv4 Address
		2001:db8:1:101::12 is CCM IPv6 Address
		5060=CCM Port
		Branch=Unique number

Contact	`IPv4 Example`
	
	`Contact: <sip:30000@10.8.69.115:5060>`
	
	`IPv6 Example`
	
	`Contact: <sip:30000@[2001:db8:1:101::12]:5060>`
		localPort = configured "Incoming port" of the indicated SIPTrunk

4xx

4xx responses represent definite failure responses from a particular server.

Table 1-22 4XX Message Fields

Field	Example	Notes
Status Code	`SIP/2.0 487 Request Cancelled`	The request was terminated by a BYE or CANCEL request.

From	`IPv4 Example`
	
	`From: <sip: 1101@10.89.79.203>;tag=16777234`
	
	`IPv6 Example`
	
	`From: <sip: 1101@[2001:db8:1:101::12]>;tag=16777234`
		10.89.79.203 is CCM IPv4 address
		2001:db8:1:101::12 is CCM IPv6 Address
		16777234 is the Call Id

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Table 1-22 4XX Message Fields (continued)

Field	Example	Notes
To	IPv4 Example To: <sip:30000@10.89.73.75>;tag=0002fd06e9300108228d58c1-614f99be IPv6 Example To: <sip:30000@[2001:db8:1:2::12]>;tag=0002fd06e9300108228d58c1-614f99be	3000 is the Calling Party Number.
Via	IPv4 Example Via: SIP/2.0/TCP 10.89.79.203:5060;received=10.89.79.203;branch=z9hG4bKfe8d27ec IPv6 Example Via: SIP/2.0/TCP [2001:db8:1:101::12]:5060;branch=z9hG4bKfe8d27ec	10.89.79.203=CCM IPv4 Address 2001:db8:1:101::12 is CCM IPv6 Address 5060=CCM Port Branch=Unique number
Contact	IPv4 Example Contact: <sip:30000@10.8.69.115:5060> IPv6 Example Contact: <sip:30000@[2001:db8:1:101::12]:5060>	localPort = configured "Incoming port" of the indicated SIPTrunk
Content-Disposition	Signal;handling=optional	Added only when QSIG tunneling is enabled.
QSIG	Binary content	If QSIG tunneling is enabled on the SIP tunk, then 4xx response will have tunneled RELEASE/REL COMP.

5xx

The server encountered an unexpected condition that prevented it from fulfilling the request.

Table 1-23 5XX Message Fields

Field	Example	Notes
Status Code	SIP/2.0 501 Not Implemented	This is the appropriate response when a UAS does not recognize the request method
From	IPv4 Example From: <sip: 1101@10.89.79.203>;tag=16777234 IPv6 Example From: <sip: 1101@[2001:db8:1:101::12]>;tag=16777234	10.89.79.203 is CCM IPv4 address 2001:db8:1:101::12 is CCM IPv6 Address 16777234 is the Call Id

Table 1-23 5XX Message Fields (continued)

Field	Example	Notes

To	`IPv4 Example`
	
	`To: <sip:30000@10.89.73.75>;tag=0002fd06e930010 8228d58c1-614f99be`
	
	`IPv6 Example`
	
	`To: <sip:30000@`
	
	`[2001:db8:1:2::12]>;tag=0002fd06e9300108228 d58c1-614f99be`
		3000 is the Calling Party Number

Via	`IPv4 Example`
	
	`Via: SIP/2.0/TCP 10.89.79.203:5060;received=10.89.79.203;bra nch=z9hG4bKfe8d27ec`
	
	`IPv6 Example`
	`Via: SIP/2.0/TCP [2001:db8:1:101::12]:5060;branch=z9hG4bKfe8 d27ec`
		10.89.79.203=CCM IPv4 Address
		2001:db8:1:101::12 is CCM IPv6 Address
		5060=CCM Port
		Branch=Unique number

Contact	`IPv4 Example`
	
	`Contact: <sip:30000@10.8.69.115:5060>`
	
	`IPv6 Example`
	
	`Contact: <sip:30000@[2001:db8:1:101::12]:5060>`
		localPort = configured "Incoming port" of the indicated SIPTrunk

Content-Disposition	`Signal;handling=optional`	Added only when QSIG tunneling is enabled.
QSIG	`Binary content`	If QSIG tunneling is enabled on the SIP tunk, then 5xx response will have tunneled RELEASE/REL COMP.

6xx

6xx indicates that the callee end system was contacted successfully but the callee is busy and does not want to take the call at this time.

Table 1-24 6XX Message Fields

Field	Example	Notes
Status Code	`SIP/2.0 600 Busy Everywhere`	The callee end system was contacted successfully, but the callee is busy and does not want to take the call at this time.

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Table 1-24 6XX Message Fields (continued)

Field	Example	Notes

From	`IPv4 Example`
	
	`From: <sip: 1101@10.89.79.203>;tag=16777234`
	
	`IPv6 Example`
	
	`From: <sip: 1101@[2001:db8:1:101::12]>;tag=16777234`
		10.89.79.203 is CCM IPv4 address
		2001:db8:1:101::12 is CCM IPv6 Address
		16777234 is the Call Id

To	`IPv4 Example`
	
	`To: <sip:30000@10.89.73.75>;tag=0002fd06e9300108228d58c1-614f99be`
	
	`IPv6 Example`
	
	`To: <sip:30000@[2001:db8:1:2::12]>;tag=0002fd06e9300108228d58c1-614f99be`
		3000 is the Calling Party Number

Via	`IPv4 Example`
	
	`Via: SIP/2.0/TCP 10.89.79.203:5060;received=10.89.79.203;branch=z9hG4bKfe8d27ec`
	
	`IPv6 Example`
	`Via: SIP/2.0/TCP [2001:db8:1:101::12]:5060;branch=z9hG4bKfe8d27ec`
		10.89.79.203=CCM IPv4 Address
		2001:db8:1:101::12 is CCM IPv6 Address
		5060=CCM Port
		Branch=Unique number

Contact	`IPv4 Example`
	
	`Contact: <sip:30000@10.8.69.115:5060>`
	
	`IPv6 Example`
	
	`Contact: <sip:30000@[2001:db8:1:101::12]:5060>`
		localPort=configured “Incoming port” of the indicated SIPTrunk

Message Timers

The following timers are service parameters that are configurable in Cisco Unified CM Administration. Cisco Unified CM maintains the following configuration data for the SIP timers.

Table 1-25 Message Timers

Timer	Value(Default/range)	Definition
trying	500 ms/100–1000 ms	The time to wait for a 100 response to an INVITE request.
connect	500 ms / 100–1000 ms	The time to wait for a 200 response to an ACK request.
disconnect	500 ms / 100–1000 ms	The time to wait for a 200 response to a BYE request.
expires	3 min/ 1–5 min	Limits the time duration for which an INVITE is valid.

Table 1-25 Message Timers (continued)

Timer	Value(Default/range)	Definition
rel1xx	500 ms / 100–1000 ms	The time that Cisco Unified CM should wait before retransmitting the reliable 1xx responses.
prack	500 ms / 100–1000 ms	The time that Cisco Unified CM should wait before retransmitting the PRACK request.
notify	500 ms / 100–1000 ms	The time that Cisco Unified CM should wait before retransmitting the Notify message.
publish	500 ms / 100–1000 ms	The time that Cisco Unified CM should wait before retransmitting the Publish message.
options	500 ms / 100–1000 ms	The time that Cisco Unified CM should wait before retransmitting the Options message.

Message Retry Counts

All the following retry counts are service parameters that are configurable in Cisco Unified CM Administration. Cisco Unified CM maintains the following configuration data for the SIP retries. In case of TCP transportation type, the timers will still pop as usual; however, in the event of timeout, Stack does not retransmit; it will instead rely on TCP itself to do the retry.

Table 1-26 Message Retry Counts

Counter	Default Value	Suggested Range	Definition
Invite retry count	5	1 – 10	Number of INVITE retries
Response retry count	6	1 – 10	Number of RESPONSE retries
Bye retry count	10	1 – 10	Number of BYE retries
Cancel retry count	10	1 – 10	Number of CANCEL retries
PRACK retry count	6	1 – 10	Number of PRACK retries
Rel1xx retry count	10	1 – 10	Number of Reliable 1xx response retries
Notify retry count	6	1 - 10	Number of NOTIFY retries
Publish retry count	6	1 - 10	Number of PUBLISH retries
Options retry count	6	1 - 10	Number of OPTIONS retries

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SIP Status Code to Q.850 Cause Code Mapping

Table 1-27 lists the SIP Status Codes and maps them to the Q.850 Release Cause Codes.

Table 1-27 SIP Status Code to Q.850 Cause Code Mapping

SIP Status Code	Q.850 Cause Code	Q.850 Release Cause Description	Scenarios when generated by Cisco Unified CM (due to internal errors)
404 Not Found 485 Ambiguous 604 Does not exist anywhere	1 Unallocated (unassigned) number	Indicates that the destination requested by theuser which the calling cannot be reached because the number is unassigned.	The number is not in the routing table, or it has no path across the ISDN network.
486 Busy here 491 Request pending 493 Undecipherable 600 Busy everywhere	17 User busy	Indicates that the called party cannot accept another call because the user busy condition has been encountered. Either the called user or the network can generate this cause value. In the case of a user-determined user busy, be aware that the user equipment is compatible with the call.	User is already using the telephone.
480 Temporarily unavailable	18 No user responding	Used when the called party does not respond to a call establishment message with either an alerting or connect indication within the time allotted. The number that is being dialed has an active D-channel, but the far end chooses not to answer.	The user does not answer the telephone.
401 Unauthorized 402 Payment Required 403 Forbidden 407 Proxy Authentication Required 600 Decline	21 Call rejected	Indicates that the equipment sending this cause code does not want to accept this call, although it could accept the call because the equipment sending the cause is neither busy nor incompatible. The network might also generate this code to indicate that the call was cleared because of a supplementary service constraint. The diagnostic field might contain additional information about the supplementary service and reason for rejection.	A subscriber has a service constraint that does not accept this call.
410 Gone	22 Number changed	Returned to a calling party when the called number that is indicated by the calling party is no longer assigned. This diagnostic field might optionally contain the new called party number.	A subscriber changed their number.

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Table 1-27 SIP Status Code to Q.850 Cause Code Mapping (continued)

SIP Status Code	Q.850 Cause Code	Q.850 Release Cause Description	Scenarios when generated by Cisco Unified CM (due to internal errors)
482 Loop detected 483 Too many hoops	25 Exchange routing error	Indicates that the destination indicated by the user cannot be reached because an intermediate exchange released the call due to reaching a limit in executing the hop counter procedure.	The network is overloaded.
484 Address incomplete	28 Invalid number format	Indicates that the called party cannot be reached because the called party number is not in a valid format or is not complete.	The caller calls out by using a network type number (enterprise) rather instead of Unknown or National.
487 Request terminated 488 Not acceptable here 606 Not acceptable	31 Normal unspecified	Reports a normal event only when no other cause in the normal class applies.	Normal operation
502 Bad gateway	38 Network out of order	Indicates that the network is not functioning correctly and that the condition is likely to last for an extended time .	Network failure
400 Bad Request 481 Call leg does not exist 500 Server internal error 503 Service Unavailable	41 Temporary failure	Indicates that the network is not functioning correctly and that the condition is likely to be resolved quickly.	Network failure
405 Method not allowed	63 Service or option not available, unspecified	Reports a service or option as a not available event only when no other cause in the service or option not available class applies.	Service not available
406 Not acceptable 415 Unsupported media type 501 Not implemented	79 Service or option not implemented, unspecified	Reports a service or option as a not implemented event only when no other cause in the service or option not implemented class applies.	Service not implemented
408 Request timeout 504 Server timeout	102 Recovery on timer expiry	Indicates that the expiration of a timer in association with error handling procedures initiated the procedure.	•No H.323 call proceeding •No H.323 alerting or connect message received from the terminating gateway •Invite expires timer reached maximum number of retries that are allowed.

Table 1-27 SIP Status Code to Q.850 Cause Code Mapping (continued)

SIP Status Code	Q.850 Cause Code	Q.850 Release Cause Description	Scenarios when generated by Cisco Unified CM (due to internal errors)
411 Length required 413 Request entity too long 414 Request URI too long 416 Unsupported URI scheme 420 Bad extension 421 Extension required 423 Interval too brief 505 SIP version not supported 513 Message too large	127 Internal error, unspecified	CC_CAUSE_INTERWORKING Indicates that interworking occurred with a network that does not provide causes for actions that it takes. The precise cause cannot be ascertained.	Failed to send message to Public Switched Telephone Network (PSTN).

SIP Trunk Supported Features

This section provides details with respect to overall flow and handling of basic SIP trunk features. This includes, but is not limited to, the following features:

- Identification Services
- Basic Call
- Simple Hold/Resume
- Transfer
- Enhanced Click-to-Dial
- Conference
- Call Forwarding
- Message Waiting Indication
- Endpoint 302 Redirect
- Park and Retrieve
- Video
- T38 Fax
- Presence (Busy Lamp Field)
- Out-of-band DTMF using KPML
- SIP Over TLS Connection
- Call Preservation
- Outbound PUBLISH
- Click-to-Call
- Monitoring and Recording
- SIP Trunk Device Identification

- Click-to-conference
- G.729 with MTP
- PAI/PPI
- SRTP
- G.Clear
- MLPP
- V.150.1
- SIP T38 Fax Interoperability with Microsoft Exchange
- MMoH
- Location based CAC
- IPv6
- Calling Party Number Transformations
- Logical Partitioning
- Early Offer G.Clear
- SRTP over non-secure SIP Trunks
- End to End Preconditions
- Connected Party Number Transformation
- Secure Monitoring and Recording
- End to end call tracing
- Q.735 MLPP over SIP Trunk
- SIP OPTIONS Ping
- Static Call Routing
- SIP Header Enhancements for Recordings
- Third Party HD video support
- Early Offer Support for SIP Trunk
- QSIG Tunneling over SIP
- SIP Refer Transparency for CVP
- BFCP support
- Cisco Telepresence MCU integration as Unicast Audio / Video mixer
- VCS integration
- Cisco Unified CM Video - SIP Video Encryption
- Cisco Unified CM G.722.1 codec support
- Cisco Unified CM AAC-LD MP4A-LATM codec support on SIP
- Cisco Unified CM V.150.1 MER

Identification Services

This section describes the SIP Identification Services in Cisco Unified CM. These services include Line Identification Services and Name Identification Services. Line Identification Services include Calling Line and Connected Line Presentation/Restriction. Name Identification Services include Calling Name and Connected Name Presentation/Restriction.

The following sections describe the options that Cisco Unified CM 6.1 and later provide for communication of identity and presentation information. The selection of these options is controllable through a SIP trunk configuration. You can use either or both options. If you select both, Asserted-Identity takes precedence over RPID.

- Using Remote-Party-ID Header, page 1-40
- Using P-Asserted-Identity/P-Preferred-Identity and Privacy Header, page 1-42

Using Remote-Party-ID Header

Cisco Unified CM provides flexible configuration options to provide these services on a call-by-call basis or statically preconfigured for each SIP trunk. This section does not describe those configuration options; it only provides the details on how Cisco Unified CM conveys these ID services in SIP by using the Remote-Party-ID header. Table 1-28 captures the support levels for the various parameters:

Table 1-28 Support Levels for Various Parameters

Parameter	Values	Notes
party	calling called	Ignored if received by Cisco Unified CM. Set to called for outgoing INVITE or UPDATE from Cisco Unified CM. Set to calling for outgoing responses from Cisco Unified CM.
id-type	subscriber user term	Ignored if received by Cisco Unified CM. Set to subscriber for outgoing requests and responses.
privacy	full name uri off	Supported if received by Cisco Unified CM. Cisco Unified CM also supports sending all values in either INVITE or UPDATE requests and responses for the same.
screen	no yes	Ignored if received by Cisco Unified CM. Cisco Unified CM always sends yes when generating an Remote-Party-ID header.

The following sections provide additional details:

- Calling Line and Name Identification Presentation, page 1-41
- Calling Line and Name Identification Restriction, page 1-41
- Connected Line and Name Identification Presentation, page 1-41
- Connected Line and Name Identification Restriction, page 1-41

Cisco Unified CM uses SIP “From” and “Remote-Party-ID” headers to provide ID services as described in the following sections.

Calling Line and Name Identification Presentation

You can configure Number and Name restriction independently on the SIP trunk. Customers can choose to restrict only number and allow name to be presented or vice versa. The default SIP trunk configuration setting specifies "not selected" or "per call setting."

Calling Line and Name Identification Restriction

- **Name**—When name is restricted, the display field (calling Name) in "From" header is set to a configurable string (that is, "Anonymous"). The display field in the "Remote-Party-ID" header still includes the actual name but the privacy field is set to **"name"**. For example:

```
From: "Anonymous" <sip:9728135001@localhost>
Remote-Party-ID: "Bob Jones"<9728135001@localhost; user=phone>;
party=calling;screen=yes;privacy=name
```

- **Number**—When number is restricted, the calling Line is left out in the "From" header; however, it is still included in the "Remote-Party-ID" header with **privacy=*uri***. For example:

```
From: "Bob Jones" <sip: 9728135001@localhost>
Remote-Party-ID: "Bob Jones"<9728135001@localhost; user=phone>;
party=calling;screen=yes;privacy=uri
```

- **Both Name and Number**—When both name and number are restricted, the same principle applies with privacy=full:

```
From: "Anonymous" <sip: 9728135001@localhost>
Remote-Party-ID: "Bob Jones"<9728135001@localhost; user=phone>;
party=calling;screen=yes;privacy=full
```

- **None**—When both name and number are allowed, the following example applies:

```
From: "Bob Jones" <sip: 9728135001@localhost>
Remote-Party-ID: "Bob Jones"<9728135001@localhost; user=phone>;
party=calling;screen=yes;privacy=off
```

Connected Line and Name Identification Presentation

The Connected Number (Line) and Name Identification supplementary service provides the calling user with the called (connected) user number and/or name.

Cisco Unified CM uses the "Remote-Party-ID" headers in 18x, 200 and re-INVITE or UPDATE messages to convey connected information. The "party" field of the "Remote-Party-ID" header is set to "called" (instead of "calling" for calling ID services).

Connected Line and Name Identification Restriction

Similar to Calling ID services, customers have option to restrict connected number and name independently.

- **Name**—When name is restricted, the connected name still gets included with **privacy=name** For example:

```
Remote-Party-ID: "Bob Jones"<9728135001@localhost; user=phone>;
party=called;screen=yes;privacy=name
```

- **Number Restrict only**—When number is restricted, the connected number still gets included with **privacy=*uri***. For example:

```
Remote-Party-ID: "Bob Jones"<9728135001@localhost; user=phone>;
party=called;screen=yes;privacy=uri
```

- **Both Name and Number Restrict**—When both name and number are restricted, both get included with **privacy=*full***. For example:

```
Remote-Party-ID: "Bob Jones"<9728135001@localhost; user=phone>;
party=called;screen=yes;privacy=full
```

- **None**—Both name and number are allowed.

  For example, if Cisco Unified CM receives an INVITE that is destined to extension 9728135001, Cisco Unified CM includes the called party name in 18x and 200 messages as follows:

```
Remote-Party-ID: "Bob Jones"<9728135001@localhost; user=phone>;
party=called;screen=yes;privacy=off
```

Using P-Asserted-Identity/P-Preferred-Identity and Privacy Header

Cisco Unified CM provides a flexible configuration options to provide these services based on a static configuration for each SIP Trunk. This section provides an overview of these configuration options and of how Cisco Unified CM conveys ID services in SIP protocol using the P-Asserted-Identity/P-Preferred-Identity and Privacy headers.

Table 1-29 shows information about the Cisco Unified CM SIP trunk use and relevance of the various values for a "Privacy" header.

Table 1-29 Values for Various Parameters

Value	Notes
Header doesn't exist	The absence of Privacy header in an INVITE, UPDATE, 180, 183 or 200 message means that the presentation is allowed for name and number. In case of initial INVITE, the presentation information applies to calling party information. In the case of reINVITE, UPDATE, 180, 183, or 200 it applies to connected party information.
id	For outgoing INVITE: Used for specifying presentation restriction for Calling Party name/number. For outgoing re-INVITE, UPDATE, 180, 183, or 200: Used for specifying presentation restriction for Connected Party name/number. For incoming INVITE, UPDATE, 180, 183, or 200: Interpreted as presentation restriction for Connected Party name/number.
header	If present in an incoming SIP request (INVITE or UPDATE), interpreted as presentation restriction for Connected Party name/number. If present in an incoming SIP response, ignored (interpreted as presentation allowed).

Table 1-29 Values for Various Parameters (continued)

Value	Notes
user	If present in an coming SIP request (INVITE or UPDATE), interpreted as presentation restriction for Connected Party name/number. If present in an incoming SIP response, ignored (interpreted as presentation allowed).
none	For outgoing INVITE: Used for specifying presentation allowed for Calling Party name/number. For outgoing re-INVITE, UPDATE, 180, 183, or 200: Used for specifying presentation allowed for Connected Party name/number. For incoming INVITE, UPDATE, 180, 183, or 200: Interpreted as presentation allowed for Connected Party's name/number.
critical	For outgoing INVITE/UPDATE, implies that privacy services requested for this message are critical and therefore, if these privacy services cannot be provided by the network, this request should be rejected. Additionally, the request should carry a Proxy-Require header containing the new option-tag “privacy.” Per rfc3323, the “critical” value is not set in outgoing responses. For incoming SIP messages (requests or Responses), Cisco Unified CM performs no specific handling for “critical” other than syntax verification.

Table 1-30 provides information about P-Asserted-Identity/P-Preferred-Identity.

Table 1-30 P-Asserted-Identity/P-Preferred-Identity Headers

Headers with value formats	Notes
P-Asserted-Identity: “name” SIPURI, tel:+DN	Comma separated headers. One SIP URI and one tel URI.
P-Asserted-Identity: “name” SIPSURI, tel:+DN	Comma separated headers. One SIPS URI and one tel URI.
P-Asserted-Identity: “name” SIPURI P-Asserted-Identity: tel:+DN	Multiple PAI headers.

The P-Asserted-Identity/P-Preferred-Identity headers are validated for syntactical correctness. If the format is not expected as per rfc3325, a "400 Bad Request" response is sent to the UAC and the request is dropped.

The format of P-Preferred-Identity header the same as that of P-Asserted-Identity header, so this section does not explicitly cover P-Preferred-Identity.

The identity information is communicated by using P-Asserted-Identity or P-Preferred-Identity headers. The presentation information is communicated by using Privacy header.

Various pages in Cisco Unified CM Administration (such as the route pattern and the translation pattern configuration pages) allow the configuration of calling number or name restrictions. The SIP trunk screen provides configuration options for the type of Identity (either Asserted or RPID). For Asserted-Identity, there is no option for allowing or restricting name/number separately (as RFC 3323 & RFC 3325 does not provide separate option for this configuration). Therefore, the SIP Privacy option applies to name and number (calling and connected). When SIP Privacy selection box set to Default, the presentation information that is received from the upper layer (call control) is used by the SIP trunk for setting the presentation of name/number in outgoing SIP request and response messages.

The following sections provide additional details:

- Calling Line and Name Identification Presentation, page 1-44
- Connected Line and Name Identification Presentation, page 1-45

Calling Line and Name Identification Presentation

- **Both name and number are allowed**—The Cisco Unified CM SIP trunk does not include a Privacy header when privacy for calling name an number is allowed. For example:

```
From: "Bob Jones" <sip:9728135001@ipAddr1>
P-Asserted-Identity: "Bob Jones"<sip:9728135001@ipAddr1>
```

Note The Privacy: none header also results in the same behavior: name and number are allowed.

- **Name is restricted**—When calling name is restricted and Privacy:id header must be sent, the From header with specific values for Display Name and SIP URI must be used. The P-Asserted-Identity header still includes the actual name. For example:

```
From: "Anonymous" <sip:anonymous@anonymous.invalid>
P-Asserted-Identity: "Bob Jones"<sip:9728135001@ipAddr1>
Privacy: id
```

Note When Name is restricted, the inclusion of Privacy: id also results in a restriction for number.

- **Number is restricted**—When calling number is restricted and Privacy:id header must be sent, it the From header with specific values for Display Name and SIP URI must be used. The P-Asserted-Identity header still includes the actual user part in the SIP URI. For example:

```
From: "Anonymous" <sip:anonymous@anonymous.invalid>
P-Asserted-Identity: "Bob Jones"<sip:9728135001@ipAddr1>
Privacy: id
```

Note When Number is restricted, the inclusion of Privacy: id also results in restriction for display name.

- **Both name and number are restricted**—When calling name and number are restricted, the same principle applies. For example:

```
From: "Anonymous" <sip:anonymous@anonymous.invalid>
P-Asserted-Identity: "Bob Jones"<sip:9728135001@ipAddr1>
Privacy: id
```

Connected Line and Name Identification Presentation

- **Both name and number are allowed**—The Cisco Unified CM SIP trunk does not include a Privacy header when privacy for connected name and number is allowed. For example:

```
P-Asserted-Identity: "alice"<sip:9193929001@ipAddr2>
```

Note **Name is restricted**—When connected name is restricted, a Privacy:id header must be sent. The P-Asserted-Identity header still includes the actual connected name. For example:

```
P-Asserted-Identity: "alice"<sip:9193929001@ipAddr2>
Privacy: id
```

- **Number is restricted**—When connected number is restricted, a Privacy:id header must be sent. The P-Asserted-Identity header still includes the actual connected number. For example:

```
P-Asserted-Identity: "alice"<sip:9193929001@ipAddr2>
Privacy: id
```

- **Both name and number are restricted**—When name and number are restricted, the same principle applies: the P-Asserted-Identity header still includes the actual connected name and number. For example:

```
P-Asserted-Identity: "alice"<sip:9193929001@ipAddr2>
Privacy: id
```

Outbound PUBLISH

Cisco Unified CM uses the PUBLISH method as the preferred mechanism to send IP phone presence information over a SIP trunk.. The main reason for using the PUBLISH method is the performance improvement over the SUBSCRIBE/NOTIFY mechanism used in previous releases.

- Cisco Unified CM sends presence information for the phones that it manages over a SIP trunk.
- Release 5.x uses the SUBSCRIBE/NOTIFY framework for the presence communication over a SIP trunk; Release 6.0(1) uses the SIP PUBLISH method for presence communication.
- Presence status in Release 5.x (SUB/NOT) applies on a per-directory-number basis. Presence status in Release 6.0(1) (PUBLISH) applies on a per-line-appearance basis.
- Line appearance maps to one directory number on a specific phone device. Thus, if two phones share the same directory number 1000, two line appearances exist: (phone1, 1000) and (phone2, 1000)
- The PUBLISH message has user association.
- The PUBLISH mechanism works with multiple partitions.
- Only two status possibilities exist for Busy/Idle in PUBLISH: “Busy” or “Idle.” (Release 5.x also supports “Available.”)
- DND Status gets published if it is turned on.

- Mobility support improves because the mobile number is in the PUBLISH message.
- Performance improvement: by using the SUBSCRIBE/NOTIFY framework, a refresh requires four messages with the PIDF body. Using the PUBLISH mechanism, a refresh requires two messages without the PIDF body.
- Release 6.0(1) includes a set of BAT tools to facilitate the upgrade from Release 5.x.

G.729 with MTP

In Cisco Unified CM 7.0, the SIP trunk allows early-offer calls (calls with pre-allocated MTP) to be initiated with low bandwidth codecs such as G.729. In previous releases, the SIP trunk supports only G711.a/ulaw with pre-allocated MTP. This feature is required for endpoints that do not support delayed media calls and that do not want to use the high bandwidth G.711 codec. This feature is turned off by default.

Software MTP on Cisco Unified CM does not support the G.729 codec. Therefore, this feature requires an external MTP/Xcoder device that supports the G29 coded to be configured.

G.729 has four variants. However, because there is no difference from the signaling perspective between G.729 and G.729a and between G.729b and G.729ab, the configuration menu for the preferred originating codec provides only show two options.

The two variants are not compatible with each other on the IOS MTP devices. Interworking with each other requires the presence of a transcoder that supports G.729 (annexb=no) to G.729 (annexb=yes) transcoding.

Cisco Unified CM treats an incoming call with G.729 with annexb=yes as an indication of all the 4 G.729 codec variants.

If MTP required is configured. The initial INVITE only contains an offer, if the trunk is able to reserve MTP/Xcoder resource. Otherwise the call is initiated as a Delayed Media call.

If the terminating side does not support the codec that is offered, the call is torn down and not tried as a delayed media call.

T38 Fax is supported with G.729 codec. However, the call will not switch back to an audio call after the fax transmission completes. This functionality is consistent with T38 Fax support with other codecs.

Mid-call codec switching from the UAS is supported if there is an Xcoder/MTP device that supports both the original codec and the new codec that is offered.

MLPP

The Multilevel Precedence and Preemption (MLPP) feature implementation is based on RFC4411 (Reason Header for Preemption Events) and RFC4412 (Resource Priority for SIP). MLPP supplementary services over SIP trunk are:

- Precedence Call Waiting
- Call Hold
- Call Transfer
- Call Forwarding
- Three-way Calling
- Call Pickup

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- Hunt List

Resource Priority Header Overview

- Resource priority (RP) headers can be used in the following SIP messages: INVITE, UPDATE, REFER (in-dialogue only)
- Messages that contain RP header must mark message with the resource priority required tag:

  Require: resource-priority
- RP headers use either of the following formats:
  - Resource-Priority:<network-domain>.<priority-value>
  - Resource-Priority:<network-domain>-<precedence-domain>.<priority-value>
- RP Header precedence domain originates from the MLPP domain of the call. AMLPP domain of 000000 may or may not be present as it is the default MLPP domain and optional.

Preemption Reason Header Overview

Preemption reason headers are included in SIP messages that terminate existing dialogues.

The following text shows the format of the preemption reason header:

```
Reason: preemption;cause=<cause value>;text=<text string>
```

The following cause values are defined by RFC4411.

1. UA Preemption
2. Reserved Resources Preempted
3. Generic Preemption
4. Non-IP Preemption
5. Network Preemption

V.150.1 MOIP

ITU V.150.1 specifies the transport of modem communications, including modem relay and voice band data, over IP networks. For the SIP, this information is transmitted in the SDP by using the media capability sequence that is specified by RFC3407. The following example shows an SDP with MOIP data:

```
v=0
o=CiscoSystemsCCM-SIP 2000 1 IN IP4 172.18.152.50
s=SIP Call
c=IN IP4 172.18.154.236
t=0 0
m=audio 4000 RTP/AVP 0 8 18 118
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=rtpmap:118 v150fw/8000
a=sqn:0
a=cdsc: 1 audio udpsprt 120
```

```
a=cpar: a=sprtmap:120 v150mr/8000
a=sendonly
```

In this example, the dynamic payload type 118 is mapped to v150 modem relay using sprt channel 120. The MOIP SDP information is internally generated based on device type and is not configurable. The following values are always used and systems that need to interconnect must use these values:

- SPRT channel: 120
- Modem relay payload type: 118
- Voice band data payload type: 100

Note There is no Cisco Unified CM configuration to enable this feature or change the information that is signaled. When a V.150.1 capable device is registered with Cisco Unified CM, the V.150.1 capabilities are automatically signaled in the SDP when communicating over a SIP trunk.

SIP T.38 Interoperability with Microsoft Exchange

The feature provides Cisco Unified CM with an alternate way to signal the call party switching over to T.38 in SIP.

Cisco Unified CM and gateways use a consistent way to initiate T.38 over audio calls in H.323, SIP, MGCP. The current way of representation in protocol signaling indicates to switch the existing audio channel to an image channel for a T.38 facsimile, and the existing ports are normally reused. The signal also implies that the audio transmission is be terminated before establishing image channels for a facsimile.

Some third-party products such as Microsoft Exchange Server prefer switching an established audio call to T.38 by explicitly indicating that the existing voice channel is terminated/not wanted and another new channel is needed to transmit T.38 facsimile. Both ways of representation for T.38 fax relay in SIP conform to the standards suggested by ITU T.38 and RFC3264. However, Cisco Unified CM does not handle the Microsoft negotiation method so that the call fails to switch over to fax.

Cisco Unified CM continues to behave the same so that when an audio call is established and a call party sends out T.38 call request, Cisco Unified CM terminates the existing audio channels and re-establishes an image channel for facsimile. Therefore, the existing functionality/mechanisms for T.38 will not be changed. The change supports incoming SIP signal that is supported by Microsoft and provides a configurable option in Cisco Unified CM to initiate fax request in the new signaling context.

Cisco Unified CM accepts the context of SIP signaling suggested by Microsoft to switch an audio call to a T.38 facsimile. Specifically, the SIP mid-call INVITE signal specifies that the current audio channel will be terminated by setting the audio port number as 0. In the same signal, another SIP media line syntax will be included to indicate detailed capability of T.38 and its receiving channel. Cisco Unified CM interprets this message as request of T.38 request.

If a SIP INVITE signal indicates active audio and image channels, it is continuously treated as an audio call request and Cisco Unified CM ignores the fax capability. Cisco Unified CM does not support Audio + Image call.

However, the limitation does not affect MS interoperability request as MS also intends to terminate audio session before establishing image channel for T.38.

Besides accepting multiple m=line (audio+image line) syntax in SIP signal to switch a call to T.38, Microsoft provides a configurable parameter in the SIP profile to initiate outgoing SIP signal in multiple m=line with audio port as 0 to negotiate T.38 calls. This parameter is offered to the SIP trunk only. Therefore, when an Exchange server (a server from another vendor) has been registered as SIP trunk in Microsoft and the parameter is checked, the server receives SIP T.38 in multiple m=lines.

This interoperability support is in Cisco Unified CM only and Cisco Unified CM should continue to interact with Cisco SIP gateways with a signal image m=line for a T.38 request. Therefore, the configurable parameter should not be checked to any SIP trunk that is associated with Cisco gateways.

Although the new parameter can be checked in a SIP profile that is associated with a SIP ICT, you should avoid doing so as it does not offer any benefit for the call.

This interoperability support for MS Exchange should not change any existing behavior of T.38 calls in any VoIP protocols. In addition, this support should not affect any VoIP protocol interoperability. But the current T.38 limitations stated will remain.

Multicast MOH over H.323/SIP Trunk

Currently, multicast MoH does not work fully as expected over SIP ICT trunk. Due to this, extra bandwidth used for each unicast MoH over the same ICT is wasted. This feature is to make multicast MoH work completely over SIP and H.323 ICT.

If the holding side is configured to use MMoH, multicast address is sent in SIP SDP message. If the other side supports MMoH, only then will it work. The new service parameter "Send Multicast MOH in H.245 OLC Message" does not apply for SIP ICT. MMoH works on tandem SIP/SIP and H.323/H.323 cases but not for SIP/H.323 interop scenarios.

IPv6

Cisco Unified CM has been enhanced to operate in both IPv4 only mode, or Dual Mode, i.e. IPv4_IPv6 mode. If IPv6 is enabled on Cisco Unified CM, then it behaves in Dual Mode, otherwise it behaves in IPv4 only mode.

SIP Trunk was one of the components on Cisco Unified CM that added support for IPv6. SIP Trunk via configuration can behave in IPv4 only mode, IPv6 only mode, or Dual Mode IPv4_IPv6.

The mode a trunk behaves is configured under the Common Device Configuration via setting the IP Addressing Mode. The default mode for a SIP Trunk is Dual Mode.

The IP Addressing Mode has three settings

- IPv4 only – The SIP Trunk uses the Cisco Unified CM IPv4 server address as its source address for SIP signaling and either, an MTP's, or Phone's IPv4 address for media.
- IPv6 only – The SIP Trunk uses the Cisco Unified CM IPv6 server address as its source address for SIP signaling and either, an MTP's, or Phone's IPv6 address for media.
- IPv4 and IPv6 – The SIP Trunk uses either the Cisco Unified CM IPv4 server address, or the Cisco Unified CM server IPv6 address as its source address for signaling and either, an MTP's IPv4 and/or IPv6 address, or the Phone's IPv4 and/or IPv6 address for media. If the trunk is configured in Dual Mode, then Cisco Unified CM will listen for SIP Traffic on both the IPv4 and IPv6 Interface.

For an IPv4 only trunk, the destination address will be populated with the remote peer's IPv4 address.

For an IPv6 only trunk, the destination IPv6 address field will be populated with the remote peer's IPv6 address.

For a dual mode trunk, both the Destination Address and the Destination v6 address fields are populated. If for some reason only one of the fields is populated, then the trunk will behave in only that mode. For a dual mode SIP Trunk, with both IP Addresses configured on the SIP Trunk device page, the enterprise parameter *IP addressing mode Preference Control* will dictate whether to use IPv4 or the IPv6 address for signaling.

For an IPv6 call, all the SIP URI's will use the following IPv6 address format (address enclosed in [ ] square brackets) as defined by RFC 3261

```
SIP-URI          =  "sip:" [ userinfo ] hostport
                              uri-parameters [ headers ]hostport
hostport        =  host [ ":" port ]
host             =  hostname / IPv4address / IPv6reference
IPv6reference  =  "[" IPv6address "]"
```

Example:

```
INVITE sip:25421@[2001:db8:1:2::12]:5083 SIP/2.0
```

Support for Alternative Network Address Types (ANAT) over SIP Trunk

Cisco Unified CM supports ANAT over dual mode (IPv4_IPv6) SIP Trunks. ANAT allows for SIP devices to send both IPv4 and IPv6 addresses in the SDP body of a SIP Offer, and to return in the SDP body of the SIP Answer a preferred IP address (IPv4 or IPv6) with which to establish a media connection. ANAT is only supported over dual mode SIP Trunk and must be supported by both ends of the SIP Trunk.

Example ANAT Offer SDP

```
v=0
o=CiscoSystemsCCM-SIP 2000 1 IN IP4 172.18.199.129
s=SIP Call
t=0 0
a=group:ANAT 1 2
m=audio 18484 RTP/AVP 0 101
c=IN IP6 2001:db8:c18:1:21c:58ff:fe2a:23f8
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=mid:1
m=audio 18282 RTP/AVP 0 101
c=IN IP4 172.18.199.55
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=mid:2
```

ANAT is enabled by checking the "Enable ANAT" checkbox on the SIP Profile associated with the SIP Trunk. ANAT can be used with both Early Offer and Delayed Offer calls. ANAT should only be enabled on SIP Trunks with an IP Addressing Mode setting of IPv4 and IPv6.

The use of ANAT on a Dual Mode SIP Trunk is indicated in the header of the SIP Invite

- The field "Require: sdp-anat" is used by Cisco Unified CM SIP Trunks using Early Offer.
- The field "Supported: sdp-anat" is used by Cisco Unified CM SIP Trunks using Delayed Offer.
- The "require" sdp-anat value indicates to the far end of the SIP Trunk connection that an ANAT based response "must" be supported.

- The "supported" sdp-anat value indicates to the far end of the SIP Trunk connection that that an ANAT based response "should" be supported.

If ANAT is enabled it should be configured on both ends of the SIP Trunk, (If "require: sdp-anat" is sent in the SIP Invite and the receiving SIP Trunk does not support ANAT all calls will be rejected).

Inbound SIP Trunk Call Rules

- If IPv6 is not enabled on Cisco Unified CM and a call comes into Cisco Unified CM on IPv6 signaling, call is dropped with no response being sent to caller. The reason for this is that when IPv6 is disabled, Cisco Unified CM does not have any ports listening for IPv6 connections. The caller hears dead air.
- If IPv6 is enabled on Cisco Unified CM and a call comes into Cisco Unified CM with invalid mode, the call is rejected with an error being sent from Cisco Unified CM to caller.
  For example, in case IPv6_only trunk receives a call w/v4 signaling in INVITE, Cisco Unified CM sends a 503 response for that INVITE request. The caller hears fast busy tone.

Outgoing SIP Trunk Calls Rules

- If IP addressing mode for SIP Trunk is set to IPv4_only and no IPv4 address is specified on the SIP Trunk configuration page, no call can be made from that sip trunk. Same is true for IPv6_only.
- If IP addressing mode for SIP Trunk is set to IPv6_only, IPv6 is chosen for signaling to send out INVITE. Same is true for IPv4_only.
- If IP addressing mode for SIP Trunk is IPv4_v6, signaling mode is chosen based on 'IP addressing mode preference for Signaling' and specified IP address on trunk configuration page. If preferred IP is not configured on trunk page then whatever is configured on SIP trunk is chosen.
- If IP addressing mode of SIP Trunk is set to IPv4_only and MTP Required is checked, IPv4 media is sent out in SDP if MTP is allocated. If no MTP is available, INVITE without SDP is sent.
- If IP addressing mode of SIP Trunk is set to IPv6_only and MTP Required is checked, IPv6 media is sent out in SDP If dual stack MTP is allocated. If no IPv4\IPv6 MTP is available, INVITE without SDP is sent.
- If IP addressing mode of SIP Trunk is set to IPv4_v6, and MTP Required is checked, a dual stack MTP is allocated if media preference is IPv6; Otherwise IPv4 MTP is allocated. If no MTP is available, the call is tried as a delayed media call.
- If IP addressing mode of SIP Trunk is set to IPv4_v6, and MTP Required is checked and a dual stack is allocated, SIP Trunk decides whether to send both IPv4 and IPv6 SDP or only IPv4 or only IPv6 SDP based on "ANAT Configuration" parameter.
- If ANAT is enabled both IPv4 and IPv6 SDP is sent with ANAT parameters set. Peer device can select one and that one is used for media for that call.
- If ANAT is disabled, SDP sent is based on media preference configured under Enterprise parameters.

Calling Party Number Transformations

The current design in Cisco Unified CM for globalizing on ingress and localizing on egress leaves very few holes that remain to be addressed.

The current design allows for prefixing digits to inbound calls based on ISDN number type. There is no way to manipulate the digits other than prefixing or stripping a fixed number of digits.

A problem that has come up in some of the early Alpha testing is that service providers are not consistently delivering numbering plan information correctly, leading to incorrect number manipulation because the calls are presented as unknown when they should be coming in as national or international. Unknown calls need to be transformed conditionally based on the digits presented and the current prefix/strip field does not offer a way to modify digits conditionally.

Also, SIP only has unknown numbering type, making it almost impossible to prefix anything that will apply to all the different numbers that may come in. This will be problematic for customers subscribing to SIP trunking services. Some of this problem can possibly be addressed if there is an SBC in the picture or the call is going to a SIP gateway by doing the manipulation on the SBC/Gateway, but the capabilities of the SBCs are questionable.

As part of this feature the prefix digits field is modified to allow stripping as well with a special notation (prefix:strip digits). However this is not very user friendly for an administrator.

To address these limitations, the following enhancements are being made:

- For all the current Incoming Calling Party Settings configuration sections on Gateway and Trunk configuration pages as well as Device Pool and Service Parameters, adding an option to configure Number of Digits to Strip in addition to the already available Prefix fields. This would replace the colon (:) notation that was added in Cisco Unified CM 7.0.
- Add the ability to configure a Transformation CSS on each of the above mentioned configuration pages as well. There will be a transformation CSS for each calling party number type to allow the ability to conditionally transform the calling party number based on number type.

For Globalization with this feature, three operations are possible to the Incoming Calling Party Number

- Strip - valid entries 0-24
- Prefix - maximum length is 16 digits
- Incoming Calling Party Transformation CSS which uses Calling Party Transformation patterns

It is recommended to use Strip & Prefix together as per the customer requirements. If conditional modifications are required then instead CSS can be used. There are not many cases where all 3 should be used together.

If all three are configured then the order in which these are applied are:

1. Strip
2. Prefix
3. CSS match and transformations applied

After the above feature changes, administrator is able to configure Prefix and Strip Digits at different levels and there was no sync at device, device pool and service parameter levels. In Cisco Unified CM 7.0, this issue was non-existant, as both the Prefix and Strip Digits, were configure in the Prefix field, using (Prefix: Strip digits) notation.

In addition to this, the Strip Digits and Transformation CSS field, by default had the NULL and <None> configuration.

In initial feature implementation, it was assumed that these default configuration for both Strip Digits and CGPN Transformation CSS, indicates that the call processing will use the values configured at the next level (Device Pool). Whereas, the Prefix field has the "Default" option, which implies that the call processing will use Prefix Digits, at the next level setting (Device Pool/Service Parameter).

So, ideally there's no way to indicate the call processing that, the user does not want to apply Strip Digits and CGPN Transformation CSS. Call processing will always go to the next level (Device Pool) and if nothing is configured at the next level, then no Strip Digits and CGPN Transformation CSS, will be applied.

Extra changes for this feature addressed these limitations in Cisco Unified CM 7.1(2) by:

- Adding "Use Device Pool Calling Party Transformation CSS" checkbox, associated with Incoming Calling Party Transformation CSS for National, International, Subscriber and Unknown number types.
- Database shouldn't allow Strip Digits to be configured, when the Prefix field is set to "Default".

Connected Party Number Transformation

The Cisco Unified CM's 6.0 & 7.0 releases added transformations support for the calling party and called party numbers by introducing the concept of "Calling Party Transformation Pattern" and "Called Party Transformation Pattern". Depending upon UCM deployments, routing requirements, number presentation requirements & CDR requirements, these existing transformations allow the calling/called numbers to be transformed to globalized, localized etc formats.

For the incoming SIP Trunk calls on UCM, in the backwards direction signaling for ringing or connect messages, the UCM communicates the display number information of target destination's Directory Number or Route Pattern number. Currently, if required there is no provision on UCM for transforming the number, which is sent in backward direction of signaling.

- In IME (ViPR) UCM deployments, the requirement is for terminating cluster to communicate E164 number of the called or connected party in IME SIP Trunk communication. The internal numbers of the called enterprise may not be useful for communication across the enterprise.
- In Tandem/Leaf UCM cluster solutions which use Softswitch for PSTN Trunking, the requirement is for UCM to communicate DID number of the called or connected party in SIP Trunk communication. This is required for billing purpose etc.

To address these requirements, the proposal is to support mechanism to transform connected party number by performing transformations on the called/connected number before sending the display number in identity headers (PAI or RemotePartyId) of SIP messages on SIP Trunk.

This feature is going be useful for Cisco Unified CM deployments involving SIP Trunk and needing capability to communicate transformed number in identity headers of 180 Ringing, 200 OK and mid-call UPDATE/reINVITE messages. This would allow the calling endpoint in other Cisco Unified CM cluster, Voice Gateway or 3rd Party [IP]PBX to be able to display the alerting or connected number in format such as DID or E164 number.

For connected party transformations support, the SIP Trunk (all Service Types) is enhanced by adding following configurations:-

1. On SIP Trunk device configuration pages:-

```
--Inbound Calls------------------------
--Connected Party Settings------------------------
-Connected Party Transformation CSS  [DropDown]
- [] Use Device Pool Connected Party Transformation CSS
```

2. On Device Pool configuration pages, new configuration for:-

```
--Call Routing Information------------------------
--Connected Party Settings------------------------
- Connected Party Transformation CSS  [DropDown]
```

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Q.735 MLPP over SIP Trunk

The Cisco Unified CM currently provides MLPP capability over the SIP Trunk for DSN/DRSN namespaces as defined by RFC4412; however it is geared to DoD requirements. The German Army uses MLPP based on I.255.3/Q.735.3 and Q.955.3. With this feature we will support the registered namespace Q735. As done for DSN/DSRN, UCM will continue using the “Resource-Priority” header as specified in RFC4412 to signal the call precedence using the values defined in the RFC. This has been covered in MLPP, page 1-46 of this document.

Call preemption will continue to be signaled using the preemption values for the SIP Reason header as specified by RFC4411 and as covered in MLPP, page 1-46. No changes are made in this area.

SIP OPTIONS ping

Currently SIP trunk doesn't track the status of its remote destination peers. SIP trunk always sends (or tries to send) SIP request to the remote destination peer. If the remote destination peer is actually unavailable, the call will not get dropped or failed over to another destination until the timeout happens. By default, it takes one minute to get a timeout. Users get nothing but silence during the time.

With the OPTIONS Ping feature, the Cisco Unified CM periodically sends SIP OPTIONS to every remote destination peer to detect its availability. If the remote destination peer is unavailable (no response or it responds "408 Request Timeout" or "503 Service Unavailable"), the Cisco Unified CM will mark this peer as unavailable. If the remote destination peer is available (any other responses other than "503" or "408"), Cisco Unified CM will mark this peer as available. The Cisco Unified CM will send a new INVITE only to the "available" remote destination peers.

Only a SIP trunk with a default type, "type None(default)", supports this feature. If FQDN or SRV is configured as a remote destination peer, Cisco Unified CM will send OPTIONS to all resolved IP addresses. The FQDN/SRV remote destination peer is marked as available if one or more of its resolved IP addresses are available. The FQDN/SRV remote destination peer is marked as unavailable if ALL of its resolved IP addresses are unavailable. The FQDN/SRV remote destination peer is marked as unavailable if DNS lookup failed.

Static Call Routing

One SIP trunk can have up to 16 destination addresses. The stack will randomly select one address to send out the SIP message. Also, the Cisco Unified CM will be able to handle the inbound messages from all those configured addresses.

SIP Header Enhancements for Recording

This feature enhances the From header in the SIP INVITE and UPDATE for recording. Now, the Cisco Unified CM sends both agent (nearend) and customer (farend) call information to the recorder. New data includes the farend refci, devicename, and address. This is more scalable since the recorder no longer requires a CTI connection to get the farend call information.

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Third Party HD video support

The Cisco Unified CM mainly supports H.264 negotiation with Profile, Level and all the optional parameters specified in the H241 and RFC3984 specifications. The support of these parameters for SCCP protocol is limited to a small subset of these. However, Cisco Unified CM does not currently take notice of the signaled packetization modes from H323 endpoints. Also, RFC3984 is undergoing a revision and the current draft of RFC3984 bis-08 has added additional optional parameters and updated the semantics of the offer/answer negotiation, taking these parameters into account.

The additional parameters are:

- max-smbps
- sprop-level-parameter-sets
- use-level-src-parameter-sets
- in-band-parameter-sets
- sar-understood
- sar-supported
- max-recv-level
- level-asymmetry-allowed

The Cisco Unified CM does not currently support rtcp-fb attributes in SDP. Cisco Telepresence endpoints are being enhanced to use a new proprietary mux attribute in the SDP and Cisco Unified CM should allow this to be passed through in the offer and answer SDPs.

As a part of this feature enhancement we have added the support of the above mentioned 8 H264 attributes, rtcp-fb and the Cisco proprietary x-cisco-mux parameter.

Sample SDP message with the above parameters:

```
m=video 20310 RTP/AVP 99
c=IN IP4 10.13.5.196
b=TIAS:1000000
a=X-Cisco-mux: cisco partner
a=rtpmap:99 H264/90000
a=fmtp:99 profile-level-id=42801E;
max-recv-level=B00E;
max-mbps=2000;
max-br=1550;
max-cpb=15;max-fs=512;
max-smbps=1000;
max-dpb=64;
sprop-parameter-sets=Z0IACpZTBYmI,aMljiA==;
max-rcmd-nalu-size=1024;
sar-supported=16;
sar-understood=124;
packetization-mode=1
a=rtcp-fb: 99 nack pli
```

These are no changes in any call flow, it would just have a few additional attributes in the SDP message as mentioned above.

Early Offer support for SIP Trunk

This feature allows the SIP trunk to support early offer outbound calls without using MTP when the media capabilities and media port information of the calling endpoint is available (for example, SIP IP Phones). For the endpoints where the media port information is not available (for example, H323 slow start calls or delayed offer SIP calls or legacy SCCP phones), Cisco Unified CM will still allocate an MTP in order to provide an offer. This means the Cisco Unified CM will allocate MTP only when necessary. Offer SDP generated by Cisco Unified CM supports sending more than one audio codec.

The goal of this feature is to improve interoperability between Cisco Unified CM SIP trunk and 3rd party SIP PBXes for the initial call setup, while reducing the issues associated with the use of the MTP. This feature is supported on both the Cisco Unified CM and CUCM-SME.

You can enable and disable this feature in the configuration on the SIP trunk page. By default, Cisco Unified CM SIP trunk will continue to send delayed offer INVITEs.

When this feature is enabled, a call originating from a Cisco Unified CM endpoint such as a phone, gateway, or another trunk and routed to an early offer SIP trunk is required to include an offer SDP in the initial INVITE. The offer SDP must be in send-recv mode, so it must have a valid media port and IP address.

The following sections provide additional information:

- When endpoint’s media capabilities and media port is available, page 1-56
- When endpoint’s media capabilities and/or media port is not available, page 1-57
- Server Initiated Call from Cisco Unified CM, page 1-57
- “Send sendrec SDP in mid-call INVITE” for mid call feature, page 1-58
- QSIG Tunneling over SIP, page 1-58

When endpoint’s media capabilities and media port is available

Cisco Unified CM gets the media capabilities and port information of the calling device for the following cases:

- Outgoing call initiated from a SIP phone registered with Cisco Unified CM.
- Outgoing call initiated from a RT-Lite or TNP SCCP phone (SCCP v20) registered with Cisco Unified CM
- When incoming INVITE with offer SDP is received on a SIP trunk
- Incoming fast start call on a H323 trunk
- Incoming call from MGCP GW

In the above cases, the Cisco Unified CM will pass the media capabilities of the calling device (A leg) to the SIP trunk (B leg) with some modifications. The Cisco Unified CM will use the media capabilities of the endpoint and apply the filtering and audio codec priority rules based on the region-pair of the calling device and outgoing SIP trunk to create the offer SDP for the outbound SIP trunk. The offer SDP will have the IP address and port of the endpoint initiating the call. This is assuming that the Cisco Unified CM does not have to insert an MTP for other reasons such as DTMF mismatch, IPv4-IPv6 interworking, TRP requirement or transcoder (no common codec between caller device and SIP trunk).

For example, calling device SIP phone offers codec G.711, G.729, L16 and G.722 in the INVITE to the Cisco Unified CM. The region-pair settings applicable for the calling phone and the SIP trunk may require L16 to be filtered. In this case, SIP trunk's offer SDP contains the G.729, G.711 and G.722 codecs. Thus, the outgoing offer SDP may have equal or less codecs than the calling device's codec capabilities.

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When endpoint's media capabilities and/or media port is not available

The Cisco Unified CM may not have the media capabilities and/or media port of the calling device for the following cases:

- call initiated from an older version of SCCP phone
- call initiated from a Cisco Unified CM controlled slow start H.323 or SCCP gateway
- delayed-offer call coming on a SIP trunk
- slow start call coming on a H.323 trunk

In the all of above cases, Cisco Unified CM allocates an MTP if an MTP was not allocated for other reasons. This is required to generate a send-recv offer with a valid media port and IP address. The allocation of an MTP will take place from the media resources associated with the endpoint rather than from the SIP trunk side media resources. This is required to avoid issues with the media path when the MTP is allocated on the SIP trunk side.

If the media capabilities of the endpoint are available (for example, for SCCP phone or MGCP gateway initiated calls), the Cisco Unified CM creates a superset of the endpoint and MTP codec capabilities and applies the codec filtering and priorities based on the applicable region-pair settings. In this case, it is expected that the MTP supports the getPort capability and also supports codec pass-through.

If the MTP does not support codec passthrough, then only the codec supported by MTP shall be offered, subject to codec preference and filtering rules applicable for the MTP and SIP trunk region.

If the media capabilities of the caller are not available, the media capabilities of the MTP are modified with the codec filtering and priorities based on the region-pair settings, to create the codec list for the outbound SDP. With this change, the Cisco Unified CM advertises multiple codecs when MTP is allocated.

For slow start H323, Cisco Unified CM advertises the superset of codecs since the codec needs tobe re-negotaited after the media cut-thru. Having a superset of codecs can avoid possible call failures due to a codec mismatch.

If the MTP allocated does not support pass thru, the Cisco Unified CM can only advertise MTP's codecs. The IP address and port in the offer SDP is that of the MTP.

If the MTP resource is not available, the call might be sent as a delayed offer call or might be rejected based on the SIP service parameter - **Fail call over SIP trunk if MTP Allocation Fails**.

Server Initiated Call from Cisco Unified CM

The Cisco Unified CM may initiate a call on behalf of the IP phones. This can happen if the user wants to move the call from an IP Phone to their cell phone. Other scenarios include using the recording feature on sip trunk, OOD REFER initiated click to call, or click to conference etc.

The Cisco Unified CM will have to allocate an MTP port to provide a valid IP port and address for the outbound SDP for server initiated calls. This port will be used untill the call is answered and then will be replaced with the caller's media information using a re-INVITE. Since the duration of the MTP usage is small, the same MTP port can be reused for multiple server initiated calls.

If the MTP resource is not available, the call might be sent as a delayed offer call or it might be rejected based on the SIP service parameter **Fail call over SIP trunk if MTP Allocation Fails**.

"Send sendrec SDP in mid-call INVITE" for mid call feature

This is a new configuration added in the **Trunk Specific Configuration** section of the SIP Profile configuration page. If this configuration is enabled, the Cisco Unified CM SIP trunk will not send **a=inactive**, **a=sendonly**, or **a=recvonly** in the outgoing SDP of a midcall INVITE or UPDATE. This usually happens when the media is being disconnected for call hold/resume or for a feature invocation like transfer or conference.

You should enable this configuration only if the "Early Offer for voice call setup" configuration is enabled.

For MOH or TOH insertion when configuration is enabled, the SIP trunk will directly send a sendrcv SDP to the remote-peer directing the media to the MOH or Annunciator source. If the local MOH or Annunicator for TOH is not available, the Cisco Unified CM SIP trunk will send an inactive SDP to break the media stream.

During the media resumption phase, SIP trunk will directly send a delayed-offer mid-call INVITE to resume the media path.

QSIG Tunneling over SIP

This feature adds support for tunneling QSIG messages in SIP on the trunk side. You can enable the feature by configuring tunneling on the sip trunk configuration page.

This feature makes SIP trunk on par with H323 ICT. Cisco Unified CM supports QSIG features like Call Transfer, Call Diversion, Call Completion, Path replacement, ID Services, and Message Waiting Indicator. One of the ways by which these features can be delivered between Cisco Unified CM clusters connected with SIP Intercluster trunk is by tunneling the QSIG content in SIP messages.

Table 1-31 shows mapping of QSIG messages and the corresponding SIP messages.

Table 1-31 QSIG and SIP message correspondance

QSIG message	SIP Message
SETUP	INVITE
ALERTING	180 Ringing
PROGRESS	183 Session Progress
CONNECT	200 OK (INVITE)
CALL PROCEEDING	Not Tunneled
DISCONNECT	INFO
RELEASE	INFO
REL COMP	BYE
FACILITY	INFO

Secure Icon Enhancement over SIP Trunk

This enhancement provides an extension to the existing security configuration of the SIP trunk, which enables a SIP trunk call leg to be considered secure if SRTP is negotiated, irrespective of the signaling transport.

On the SIP Trunk device page, the **Consider Traffic on this Trunk Secure** drop down menu item is introduced, below the **sRTP Allowed** checkbox on the SIP Trunk device page. Drop down menu items consist of the following two selections:

- When using both sRTP and TLS
- When using only sRTP

You can select either of these options only when you enable the **sRTP Allowed** checkbox (that is present above the new drop-down box) on the sip trunk device configuration page.

Migration Rules

- New trunks would default to "When using both sRTP and TLS"
- Existing trunks would become "When using both sRTP and TLS"

Feature behavior using a migrated SIP trunk would not change.

When user configures the SIP Trunk device with "When using both sRTP and TLS" option, then there is no behaviour change noticed on this SIP Trunk calls. The system displays the secure lock icon on the phone only when both sRTP and TLS gets selected as negotiated media and negotiated transport respectively.

When you configure the SIP Trunk device with "When using only sRTP" option, then you will notice a behaviour change. SIP Trunk device layer code would not take the device security mode into account when the "When using only sRTP" is utilized. This would achieve H.323 parity. The system displays the secure lock icon on the phone only when sRTP media is negotiated regardless of what transport protocol (TCP/UDP/TLS) is negotiated, for example:

- If your call has TCP as negotiated transport and uses sRTP as negotiated media, then you will see the Secure Lock Icon display on your phone.
- If your call has TCP as negotiated transport and uses RTP as negotiated media, then you will NOT see the Secure Lock Icon display on your phone.

Overall call security must still depend on the SIP Trunk and other parties in the call. Incoming call-Info header (over SIP Trunk) can still influence the overall secure status of the call.

Support for Image Attribute in SDP

This feature adds support for the SDP attribute, image attribute. Image attribute is a video media-level attribute to specify different image properties, such as the supported image size for sending and receiving video stream. For early offer, delayed offer, or split/join scenarios, Cisco Unified CM's answer will contain image attribute only if both party offer image attributes for the negotiated video payload. If both payload-specific (for example, imageattr:97) and all-payload (for example, imageattr:*) image attributes are offered in the same SDP for the negotiated codec, payload-specific image attribute is chosen. For early offer to early offer or mid-call reINVITES, image attribute transparently passes through to the second call leg. Same rules apply when MTP is part of the calling path.

Support for Initial Invite Requri Parameter Passthrough

Request URI header parameters received in an initial INVITE are automatically relayed (transparently passed-through) in the corresponding outbound initial INVITE associated with the call.

Support for Blind Transfer Refer Parameter Passthrough

During a blind transfer operation, if any header parameters are included in the Refer-To header of the blind transfer REFER, those header parameters are automatically relayed in the corresponding outbound initial INVITE's requri header associated with the call.

Support for Contact Header Parameter Passthrough

Contact header parameters embedded in any of the below SIP messages received by Cisco Unified CM are automatically relayed (transparently passed-through) in subsequent applicable outgoing messages to the other endpoint involved in the two party call:

- INVITE (initial)
- 180 Ringing
- 183 Session Progress
- 200 for INVITE/reINVITE/UPDATE
- Mid-Call reINVITE
- UPDATE

The only exception to the above is with respect to the ";video" parameter. Cisco Unified CM will only relay this parameter if it believes the call to have negotiated the video.

MAX-FPS attribute support for H264 Codec

Max-fps is another H.264 optional parameter, and it is treated as a declarative parameter. This parameter will be passed transparently from one side to the other. This is the same behavior as all other H.264 optional parameters we supported in 8.5.

Sample SDP:

```
b=TIAS:128000
a=rtpmap:97 H264/90000
a=fmtp:97
profile-level-id=42800D;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-smbps=40500;m
ax-fps=5440
a=rtpmap:98 H264/90000
a=fmtp:98
profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=
925;max-smbps=40500;max-fps=5440
```

AMR/AMR-WB Codec Support

GSM mobile uses the AMR or AMR-WB codecs for voice call. Generally, when a mobile user calls another mobile phone, the call would be established using the AMR codec end-to-end without requiring a transcoder. With the HEC solution, all HEC mobile calls are now routed via Cisco Unified CM first. However, UCM does not support AMR/AMR-WB codecs. Cisco Unified CM would ignore any codecs it does not understand. So for a HEC mobile call another mobile thru Cisco Unified CM, it would select the G711 codec instead of AMR. As a result, a transcoder is needed in the mobile network to convert from AMR to G711 and back out to AMR.

The AMR/AMR-WB codec support would eliminate the need for a transcoder for mobile to mobile calls in a HEC deployment.

Sample SDP Content of an offer with AMR & AMR-WB codec:

```
v=0
o=user1 53655765 2353687637 IN IP4 10.77.21.34
s=-
c=IN IP4 10.77.21.34
t=0 0
m=audio 49194 RTP/AVP 97 98 99
a=rtpmap:97 AMR/8000
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2; mode-change-neighbor=1
a=maxptime:20
a=rtpmap:98 AMR/8000
a=fmtp:98 mode-set=0,2,5,7; mode-change-period=2; mode-change-neighbor=1
a=maxptime:30
a=rtpmap:99 AMR-WB/16000/2
a=fmtp:99 mode-change-period=2; mode-change-neighbor=1
a=sendrecv
```

BFCP Support

The purpose of the Binary Floor Control Protocol (BFCP) is to enable presentation sharing between BFCP capable endpoints. BFCP support in Cisco Unified CM 8.6 strictly applies to SIP devices. Both SIP Trunk and SIP Line interfaces are fully supported. BFCP is supported on Early Offer (EO) and Delayed Offer (DO) SIP trunks. BFCP capabilities on a SIP Trunk or SIP Line can be turned on or off via SIP Profile Configuration. Cisco Unified CM's main task is to negotiate BFCP between endpoints via SIP SDP. Cisco Unified CM never terminates the BFCP protocol. BFCP protocol exchanges are between endpoints only.

In a normal call scenario with successful BFCP negotiation, the following media are negotiated via the SIP Protocol:

- Audio.
- Video (main).
- Video (presentation).
- BFCP Application Line

Audio is the audio stream.

Video (main) is the standard video stream (i.e. person to person).

Video (presentation) is the video from a presentation source such as a laptop connected to a BFCP capable phone.

BFCP Application Line is the control channel and is the media line used to perform the majority of the BFCP negotiation.

Note Cisco Unified CM and the BFCP capable SIP Phones negotiate the actual media via the SIP Protocol. However, in the BFCP call, all four media types travel from SIP Phone to SIP Phone.

In order for successful BFCP Presentation Sharing during a call, all SIP Interfaces for the call must have BFCP enabled. BFCP feature activation for a particular SIP Line or SIP Trunk interface is controlled by the **Allow Presentation Sharing using BFCP** checkbox within the SIP Profile. It is very important to note that BFCP Presentation sharing is disabled by default in the SIP Profile.

If the **Allow Presentation Sharing using BFCP** checkbox is unchecked (i.e. reject BFCP), then:

- Cisco Unified CM will not send an offer to the line or trunk endpoint containing the BFCP application media line or Presentation video line.
  - Note - a second video line is allowed if it isn't in the context of BFCP. This is for future use when non-BFCP applications utilize multiple video lines. Cisco Unified CM will know it's not in the context of BFCP if the BFCP Application line isn't present.
- Offers received by Cisco Unified CM with BFCP and Presentation Media Line will have those media lines rejected via media port if the media line set to zero.

As long as this checkbox is checked, Cisco Unified CM does not reject BFCP capabilities for that particular interface. However, BFCP must be enabled on all SIP interfaces in the call path for BFCP Presentation to work.

The new BFCP control line is an SDP Application line with the below example markings:

```
m=application 5070 UDP/BFCP *
```

UDP/BFCP indicates that the application being negotiated is BFCP using the UDP protocol. TCP/BFCP is not supported by Cisco Unified CM.

The following sections provide additional information:

- Security Icon Support and BFCP, page 1-62
- MTP, TRP, Transcoder and RSVP Agents, page 1-62

Security Icon Support and BFCP

Secure media is supported for Audio, Video (main), and Video (presentation). Security is not supported for the BFCP media line. Cisco Unified CM 8.6 introduces Cisco CallManager service parameter, **Ignore BFCP Application Stream Encryption Status When Designating Call Security Status**, which indicates whether Cisco Unified CM considers the secure status of BFCP application stream when determining whether a call is designated secure. When this parameter is set to "True" (default value), calls utilizing a non-secured BFCP application stream will be treated as secure, provided that the remaining media streams in the call are secured.

MTP, TRP, Transcoder and RSVP Agents

In CUCM 8.6, when any of the following devices are inserted in any part of the BFCP call path, BFCP presentation sharing will not function:

- Media Termination Point (MTP)
- Trusted Relay Point (TRP)
- RSVP Agent
- Transcoder

If any of these devices are present in a BFCP call, the BFCP application media line will be zeroed as well as the secondary video line.

CUCM G.722.1 Codec Support

G.722.1 provides high quality, wideband audio codec at moderate bit rates of 24 or 32 kb/s. G.722.1 is supported for calls between two SIP devices or between two H.323 devices, and calls between SIP and H.323 devices.

Cisco Unified CM supports the following call scenarios:

- Calls between SIP devices with audio codec G.722.1 (Symmetric Dynamic Payload Number).
- Calls between SIP devices with audio codec G.722.1 (Asymmetric Dynamic Payload Number)
- MTP pass-through calls with audio codec G.722.1.
- E2E RSVP calls with audio codec G.722.1.
- Local RSVP calls with audio codec G.722.1.
- Calls across SIP trunk (configured as DO or EO) with audio codec G.722.1.
- Calls between SIP and H.323 devices with G.722.1 codec Dynamic Payload Number mismatching.

Note G.722.1Annex C is not supported.

Sample SDP Content of an offer with G.722.1 code:

```
m=audio 2366 RTP/AVP 100 101 9 8 0
b=TIAS:32000
a=rtpmap:100 G7221/16000
a=fmtp:100 bitrate=32000
a=rtpmap:101 G7221/16000
a=fmtp:101 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=sendrecv
```

CUCM AAC-LD MP4A-LATM Codec Support on SIP

AAC-LD stands for Advanced Audio Coding-Low Delay, which is a super-wideband audio codec that provides superior sound quality for voice and music. This codec provides equal or improved sound quality over older codecs, even when using relatively lower bit rates.

There are 2 different RTP payload formats for this codec:

- **mpeg4-generic** (RFC 3640), used for MPEG-4 audio, video, and systems, has been supported for SIP in CUCM since version 6.0. Cisco TelePresence uses this variant.
- **MP4A-LATM** (RFC 3016). used for MPEG-4 audio, is supported for SIP starting from 8.5(1). Tandberg and most third parties use this variant.

Cisco Unified CM supports MP4A-LATM of the following bit rates: 128 kbps, 64 kbps, 56 kbps, 32 kbps, 24 kbps.

Cisco Unified CM supports the following call scenarios using the MP4A-LATM codec:

- Calls between SIP devices with audio codec MP4A-LATM (Symmetric Dynamic Payload Number).
- Calls between SIP devices with audio codec G.722.1 (Asymmetric Dynamic Payload number).

- MTP pass-through calls (including E2E RSVP and Local RSVP calls) with audio codec MP4A-LATM.
- Calls across SIP trunk (configured as DO or EO) with audio codec MP4A-LATM.

Note CUCM supports AAC-LD MP4A-LATM codec for SIP only.

The following is a sample SDP with MP4A-LATM codec parameters:

```
m=audio 2358 RTP/AVP 100 101 102 103 104 105 9 8 0
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 profile-level-id=25;object=23;bitrate=128000
a=rtpmap:101 MP4A-LATM/90000
a=fmtp:101 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:102 MP4A-LATM/90000
a=fmtp:102 profile-level-id=24;object=23;bitrate=56000
a=rtpmap:103 MP4A-LATM/90000
a=fmtp:103 profile-level-id=24;object=23;bitrate=48000
a=rtpmap:104 G7221/16000
a=fmtp:104 bitrate=32000
a=rtpmap:105 G7221/16000
a=fmtp:105 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=sendrecv
```

SIP REFER Transparency

In earlier SME release, the way SME (or Cisco Unified CM) handles an incoming feature transfer request (via SIP Refer request) that is received on an active call leg is by acting on the transfer request and initiaing a new call to the transferred leg. (i.e SME/Cisco Unified CM processes the incoming SIP Refer request and processes the Refer-To header information and creates a brand new call to the number given in the Refer-To header.) Due to this behaviour, SME will always be involved in the call arc even if the endpoints are not geographically present near SME.

This SIP Refer Transparancy feature causes SME to transparently pass the transfer request across the call arc, instead of processing it locally, thereby allowing the local SME to be dropped completely from the triggered call being originated from the transferring party.

This behavior does not apply to transfer requests generated by line-side devices.

This feature applies only to SIP REFER requests received with the following characteristics

- in a SIP tandem trunk deployment. i.e both sides of call arc must be a SIPTrunk. One of the call leg cannot be a line or non-SIP signaling protocol.
- the REFER request is received within an existing dialog. Specifically, a 'blind' transfer feature invocation where the transfer-to party is 1st contacted by transferee.
- does not contain a Replaces tag, which has relevance only within the local user-agent (in this case SME/CUCM).
- the peer endpoint has advertised support for receiving REFER requests in the Allow: header received during call setup in either the INVITE request, or the 200 OK response to INVITE.
- Finally, a provided LUA script (refer-passthrough ua script) must be provisioned on the trunk on which the REFER request is received.

When SME/Cisco Unified CM receives an in-dialog incoming REFER (with no Replaces tag) and this feature is invoked by hitting the LUA script, and above said conditions are met, you will notice that SME/Cisco Unified CM will forward the incoming REFER message to the other side of the call arc. Any NOTIFY messages that are sent by the endpoint (in response to the forwarded REFER message indicated the progress of this transfer) will be forwarded by SME/Cisco Unified CM to the originating side (that initially sent the REFER request).

CUCM Video - SIP Video Encryption

Cisco Unified CM supports the following SIP Video Encryption features:

1. Support of new Crypto Suites and Session Parameters in the SDP
   a. Crypto suites - AES_CM_128_HMAC_SHA1_32,AES_CM_128_HMAC_SHA1_80, F8_128_HMAC_SHA1_80
   b. Crypto session parameters:
   – Negotiated parameters - UNENCRYPTED_SRTP, UNENCRYPTED_SRTCP, UNAUTHENTICATED_SRTP
   – Declarative parameters - KDR, FEC_ORDER, FEC_KEY, WSH
2. For SIP to Non-SIP calls, support AES_CM_128_HMAC_SHA1_32 only and no session parameters.
3. CUCM shall not support crypto lines with UNENCRYPTED_SRTP. Hence crypto lines containing UNENCRYPTED_SRTP (session negotiated parameter), will be removed from the offer before the crypto matching is done.
4. New Crypto Policy is supported for Audio, Video (main and 2nd) and FECC Lines. It is not supported for BFCP and T.38 fax lines.
5. For MTP pass thru and RSVP scenarios 2nd Video M Line and BFCP crypto is not supported.
6. Cisco Unified CM supports only first encryption key in the crypto line.
7. Added support for preferential selection of encryption algorithms for SIP to SIP calls
   a. Both Crypto Suite and Session parameters should match.
   b. Select the crypto algorithm based on lower sum of matched crypto positional indexes from the offered Crypto lines from both endpoints.
   c. Prefer 80 bit encryption algorithm in case of multiple matches with the same sum.

Sample SDP

```
v=0
o=tandberg 77 1 IN IP4 10.29.6.85
s=-
c=IN IP4 10.29.6.85
m=audio 16888 RTP/SAVP 100 101 102 9 18 11 8 0 103
a=crypto:0 AES_CM_128_HMAC_SHA1_80 inline:g4bG4IwinJEgmkefTeR0rnueTcFF7UAQfhoSqCHd|2^48
UNENCRYPTED_SRTCP
a=sendrecv
m=video 16890 RTP/SAVP 97 98 99 34 31
a=crypto:0 AES_CM_128_HMAC_SHA1_80 inline:2iYWKcuIrKp+IDyd1yQg3Le0J1SF7wPF5aCxl/uA|2^48
UNENCRYPTED_SRTCP
a=sendrecv
a=content:main
m=application 5070 UDP/BFCP *
a=floorctrl:c-s
m=video 16892 RTP/SAVP 97 98 99 34 31
b=TIAS:1152000
```

```
a=crypto:0 AES_CM_128_HMAC_SHA1_80 inline:kGQ0bpXhtnH6Uwz48LUwZSXnOiata5pbYw+fsTpP|2^48
UNENCRYPTED_SRTCP
a=sendrecv
m=application 16894 RTP/SAVP 104
a=rtpmap:104 H224/4800
a=crypto:0 AES_CM_128_HMAC_SHA1_80 inline:n+95fbtBmH+FzgZekfNbhRih2Ky9NQ4fiaP10xOe|2^48
UNENCRYPTED_SRTCP
a=sendrecv

Current Encryption Selection Policy in CUCM -
Offer from A:
a=crypto:1 AES_CM_128_HMAC_SHA1_32 inline :...............
a=crypto:2 AES_CM_128_HMAC_SHA1_32 inline :...............

Offer from B
a=crypto:1 AES_CM_128_HMAC_SHA1_32 inline :...............
a=crypto:2 AES_CM_128_HMAC_SHA1_32 inline:...............
```

Result - CUCM will not consider any Algorithm other than AES_CM_128_HMAC_SHA1_32 inline for crypto line match. Therefore Cisco Unified CM picks the first crypto line match which is highlighted in green.

```
New Encryption Selection policy in CUCM -
Offer from A:
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:............... UNENCRYPTED_SRTCP
a=crypto:2 AES_CM_128_HMAC_SHA1_80 inline:...............
a=crypto:3 AES_CM_128_HMAC_SHA1_32 inline:...............

Offer from B:
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:...............
a=crypto:2 AES_CM_128_HMAC_SHA1_32 inline:............... FEC_ORDER=FEC_SRTP
a=crypto:3 AES_CM_128_HMAC_SHA1_80 inline:............... UNENCRYPTED_SRTCP
```

In this example, there are two matches. When there is more than one match, the following rule shall be applied in selecting the crypto pair:

- Both Crypto Suite and Session parameters should match.
- Select the crypto algorithm based on lower sum of matched crypto indexes.
- Prefer 80 bit encryption algorithm in case of multiple matches with the same sum.

Result - second match has a lower sum value so Cisco Unified CM picks that match. The selected crypto lines shall be responded in the respective answer SDP.

```
1.crypto-lineIndex =1 (A) and crypto-lineIndex=3(B) ? sum of crypto line index values = 4
2.crypto-lineIndex=2 (A) and crypto-lineIndex=1(B) ? sum of crypto line Index values = 3
```

V.150.1 MER

Modem relay as implemented according to standard SCIP-215/216 is referred to as V150 MER modem relay, MER is short for "Minimal Essential Requirements". SCIP is developed to provide end to end encrypted voice and data communication between terminals operating on heterogeneous networks.

V.150.1 MER enables the use of non-secure Modem over IP (MOIP) via modem-relay and audio passthrough as well as Fax over IP (FOIP) via T.38 v3. Note that although we say this is non-secure many cases may involve the endpoints themselves setting up their own encryption, such as with STEs. This implementation replaces legacy Griphus/Semper Fi V.150 and is backwards compatable with it.

Sample SDP content:

```
v=0
o=CiscoSystemsCCM-SIP 2000 1 IN IP4 172.18.154.43
s=SIP Call
c=IN IP4 172.18.155.77
t=0 0
m=audio 4000 RTP/AVP 0 8 18 101 100 118 126
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=sendonly
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15,32-35
a=rtpmap:100 X-NSE/8000 Optional Line: Indicates legacy VBD support
a=rtpmap:118 v150fw/8000
a=rtpmap:126 NoAudio/8000 Optional Line: Will be omitted in responses w/ negotiated audio
a=sprtparm:190 200 132 220 50 8  Optional Line: Various SPRT Window sizes and max payloads
a=sqn:0
a=cdsc: 1 audio RTP/AVP 0 8 18 101 100 118 126
a=cdsc: 8 audio udpsprt 120
a=cpar: a=sprtmap:120 v150mr/8000
a=cpar: a=fmtp:120 mr=1;mg=0;CDSCselect=1;jmdelay=no;Versn=1.1;mrmods=1-5,7-8,10
a=vndpar:2 9 2 15 Optional Line: Keeps track of legacy capabilities
```

T.38 support is signalled as normal except with an a=cdsc line rather than an m line. Event 4 also needs to be listed on the fmtp line associated with the v150fw payload type.

```
…(abridged)…
a=rtpmap:118 v150fw/8000
a=fmtp:118 1,3-4
a=rtpmap:126 NoAudio/8000
a=sqn:0
a=cdsc: 1 audio RTP/AVP 0 8 18 101 118 126
a=cdsc: 7 audio udpsprt 120
a=cpar: a=sprtmap:120 v150mr/8000
a=cpar: a=fmtp:120 mr=1;mg=0;CDSCselect=1;jmdelay=no;Versn=1.1;mrmods=1,3
a=cdsc: 8 image udptl t38
a=cpar: a=T38FaxVersion:3
a=cpar: a=T38MaxBitRate:33600
a=cpar: a=T38FaxFillBitRemoval:0
a=cpar: a=T38FaxTranscodingMMR:0
a=cpar: a=T38FaxTranscodingJBIG:0
a=cpar: a=T38FaxRateManagement:transferredTCF
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
a=cpar: a=T38FaxMaxBuffer:200
a=cpar: a=T38FaxMaxDatagram:320
```

User-Agent/Server header and Identity Header hostname pass-through

In prior releases, the User-Agent header added by Cisco Unified CM to outgoing SIP requests specified Cisco Unified CM and the software release (e.g., Cisco-CUCM8.5). In no case was a SIP Server header included in outgoing responses.

There are 3 new configuration settings available within the SIP Profile to control the pass-through of the User-Agent and Server SIP headers. These are under the configuration: **User-Agent and Server header information**

- **Send Unified CM Version Information as User-Agent** - This option maintains the currect behavior of Cisco Unified CM to provide the current release as the User-Agent in requests and not provide any Server header in responses.
- **Pass Through Received Information as Contact Header parameter** - This option is intended only for intercluster SIP trunks, and passes the information through as a parameter to the Contact parameter
- **Pass Through Received Information as User-Agent Header** - This configuration option is intended to be enabled on all trunks pointing to 3rd party equipment, where it is useful to pass-through the User-Agent and Server headers of the peer device.

The SIP identity headers (From:, Remote-Party-ID:, and P-Asserted-Identity:) provide the identity of the party generating the messages. Because existing Cisco Unified CM behavior is geared toward the use of numeric user parts (for example, telephone numbers), the incoming hostname info provided in the incoming SIP message was suppressed, and replaced with the IP address of the local Cisco Unified CM when those messages were delivered to the endpoints or sent out a SIP trunk. However, some devices are geared toward the use of full SIP URIs. It thus became useful to provide the full SIP URL (user@host) that was received in the incoming SIP messages. This allows these endpoints to acquire the full username and host of the other party, and enable their native capability of making calls directly to those SIP URLs rather than relying on that information being provisioned by the Cisco Unified CM.

On the SIP Profile configuration page, the **Use Fully Qualified Domain Name in SIP Requests** checkbox was added. When disabled (the default), the previous behavior of Cisco Unified CM is maintained: only the user part of the identity header is passed through, the hostpart is replaced with the Cisco Unified CM IP address. When enabled, the hostname received by the Cisco Unified CM in the incoming SIP message will be passed through, and will appear in the outgoing messages.

Troubleshooting

Table 1-32 highlights some of the common problems that might be encountered when you are configuring a SIP trunk.

Table 1-32 Troubleshooting SIP Trunk Configuration

<table>
<tr><th>Symptom</th><th>Possible Cause</th><th>Recommended Action</th></tr>
<tr><td>Cannot receive or make calls through a SIP trunk.</td><td>The SIP trunks are not initialized because the Cisco Unified CM process is not part of a Cisco Unified CM Group.</td><td>Associate the Cisco Unified CM to a Cisco Unified CM Group.
1. In Cisco Unified CM Administration, choose System > Cisco Unified CM Group.
2. Click Find and choose a group name (such as Default).
3. Make sure that the Cisco Unified CM appears under the “Selected Cisco Unified CMs” section.
The following example SDL logs illustrate how a SIP trunk fails initialization because the Cisco Unified CM node is not a member of a Cisco Unified CM Group:
<code>|SIPD(1,100,76,1)|SIPInit(1,100,73,1)||NumOfCurrentInstances: 1 000000561|2005/07/20 12:54:17.598| 001|SdlSig|Start|start| SIPD(1,100,76,1)|SIPD(1,100,76,1)|(1,100,76,1).1-(*:*) |[R:HP - HP: 0, NP: 0, LP: 0, VLP: 0, LZP: 0 DBP: 0] 000000562| 2005/07/20 12:54:17.598| 001|SdlSig|DbSIPTspReq|initializing| Db(1,100,160,1)|SIPD(1,100,76,1)|(1,100,76,1).1-(*:*)|[NP-PQ: 0]</code>

<code>000000563|2005/07/20 12:54:17.620|001|SdlSig|DbSIPTspRes tsp_discovery|SIPD(1,100,76,1)|Db(1,100,160,1)|(1,100,76,1).1-(*:*) |[R:NP - HP: 0, NP: 0, LP: 0, VLP: 0, LZP: 0 DBP: 0]</code>

<code>000000564|2005/07/20 12:54:17.621|001|SdlSig| DbSimpleDeviceServerReq|initializing|Db(1,100,160,1) |SIPD(1,100,76,1) |(1,100,76,1).1-(*:*)|[NP - PQ: 0]</code>

<code>000000565|2005/07/20 12:54:17.636|001|SdlSig |DbSimpleDeviceServerRes|device_server_discovery |SIPD(1,100,76,1)|Db(1,100,160,1)|(1,100,76,1).1-(*:*) |[R:NP - HP: 0, NP: 0, LP: 0, VLP: 0, LZP: 0 DBP: 0]</code>

<code>000000566| 2005/07/20 12:54:17.636|001|Stopping|| |SIPD(1,100,76,1)|SIPD(1,100,76,1)||NumOfCurrentInstances: 1</code>

<code>000000567| 2005/07/20 12:54:17.637|001|Stopped|| |SIPD(1,100,76,1)|SIPD(1,100,76,1)||NumOfCurrentInstances: 0</code>

<code>000000568| 2005/07/20 12:54:17.637| 001| SdlSig|DeviceStop</code></td></tr>
<tr><td>The UAS may reject all outbound calls associated with a SIP trunk with a 4xx response, or signaling may go through, but no audio path gets detected.</td><td>Unusual third-party SIP UA might not support delayed media INVITE.</td><td>The SIP trunk default configuration results in sending INVITE (without SDP). This definition prevents the use of an MTP resource. Cisco recommends that you leave the “MTP Required” checkbox on the SIP trunk configuration page unchecked; however, if third-party devices do not support delayed media INVITE requests, you can check this box.
You must reset the SIP trunk for the change to take effect.</td></tr>
</table>

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Table 1-32 Troubleshooting SIP Trunk Configuration (continued)

Symptom	Possible Cause	Recommended Action
DTMF digits do not get sent to the remote SIP device such as a Unity Voice Messaging server.	This could represent an interoperability issue. The SIP trunk supports RFC2833 and Out-of-Band methods when sending DTMF tones across the network. - Most SIP devices support RFC2833. - The supported OOB methods are KPML and Unsolicited Notify. KPML is not widely used in the marketplace at this time. Currently, the only known products supporting KPML are the Cisco TNP phones, Cisco Unified CM, and Cisco IOS Gateway (12.4 and later). Unsolicited Notify is a Cisco proprietary method that is used only on Cisco IOS Gateways (12.2 and later?). Unity does not support either one at this time but might support KPML in the future. If you are connecting Unity to Cisco Unified CM via SCCP, OOB is assumed.	Check whether an MTP resource is allocated for the call: • If both parties have at least one common DTMF method, an MTP is *not* required. • If one party only supports the Out-of-Band method, but the other party only supports RFC2833, an MTP is required. For RFC23833 DTMF events, verify that the media stream (via Sniffer) contains packets with the DTMF Payload Type value. For OOB, incoming or outgoing NOTIFY messages get captured in the Cisco Unified CM trace file. the following example shows Subscribing for DTMF-KPML: `SUBSCRIBE sip:172.18.199.61:5060 SIP/2.0` `Via: SIP/2.0/UDP 172.18.199.62:5060;branch=z9hG4bK1BD` `From: <sip:3601@172.18.199.62>;tag=169AEB4-93D` `To: "sccp_3000"` `<sip:3000@172.18.199.61>;tag=520767e3-a20b-488e-9ca2-3b1506ab9e94-24577005` `Call-ID: 47b5f280-2de1b302-3fc-3dc712ac@172.18.199.61` `CSeq: 101 SUBSCRIBE` `Max-Forwards: 70` `Date: Wed, 20 Jul 2005 20:24:36 GMT` `User-Agent: Cisco-SIPGateway/IOS-12.x` `Event: kpml` `Expires: 7200` `Contact: <sip:3601@172.18.199.62:5060>` `Content-Type: application/kpml-request+xml` `Content-Length: 327` `<?xml version="1.0" encoding="UTF-8"?><kpml-request xmlns="urn:ietf:params:xml:ns:kpml-request" xmlns:xsi="http://www.w3.org/2001 /XMLSchema-instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-request kpml-request.xsd" version="1.0"><pattern persist="persist"><regex tag="dtmf">[x*#ABCD]</regex></pattern></kpml-request>`  `SIP/2.0 200 OK` `Via: SIP/2.0/UDP 172.18.199.62:5060;branch=z9hG4bK1BD` `From: <sip:3601@172.18.199.62>;tag=169AEB4-93D` `To: "sccp_3000"` `<sip:3000@172.18.199.61>;tag=520767e3-a20b-488e-9ca2-3b1506ab9e94-24577005` `Call-ID: 47b5f280-2de1b302-3fc-3dc712ac@172.18.199.61` `CSeq: 101 SUBSCRIBE` `Content-Length: 0` `Contact: <sip:172.18.199.61:5060>` `Expires: 3600`

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Table 1-32 Troubleshooting SIP Trunk Configuration (continued)

Symptom	Possible Cause	Recommended Action
DTMF digits do not get sent to the remote SIP device (continued)		The following example shows an outbound KPML NOTIFY message: NOTIFY sip:3010@172.18.199.92:5060 SIP/2.0 Via: SIP/2.0/UDP 172.18.199.61:5060;branch=z9hG4bK48b28f9b From: <sip:3010@172.18.199.61>;tag=520767e3-a20b-488e-9ca2-3b1506ab9e 94-26499709 To: <sip:3501@172.18.199.92>;tag=1A60AE98-324 Call-ID: 4724DD80-FC6211D9-8190EC13-60F39CA2@172.18.199.92 CSeq: 103 NOTIFY Max-Forwards: 70 Date: Mon, 25 Jul 2005 16:45:29 GMT User-Agent: Cisco-CCM5.0 Event: kpml Subscription-State: active;expires=3600 Contact: <sip:172.18.199.61:5060> Content-Type: application/kpml-response+xml Content-Length: 177 <?xml version="1.0" encoding="UTF-8" standalone="no" ?> <kpml-response code="200" digits="1" forced_flush="false" suppressed="false" tag="dtmf" text="Success" version="1.0"/> The following example shows negotiating Unsolicited NOTIFY request: INVITE sip:3501@172.18.199.92:5060 SIP/2.0 Call-Info: <sip:172.18.199.61:5060>;method="NOTIFY;Event=telephone-event;D uration=500" Response: SIP/2.0 200 OK Call-Info: <sip:172.18.199.92:5060>;method="NOTIFY;Event=telephone-event;D uration=500" The following example shows an outbound Unsolicited NOTIFY message: 07/26/2005 14:15:18.658 CCM|Outgoing UDP SIP message to 172.18.199.92:[57475]: NOTIFY sip:172.18.199.92:57475 SIP/2.0 Via: SIP/2.0/UDP 172.18.199.61:5060;branch=z9hG4bK66516ae9 From: "sccp_3010" <sip:3010@172.18.199.61>;tag=520767e3-a20b-488e-9ca2-3b1506ab9e 94-26499723 To: <sip:3501@172.18.199.92>;tag=1FD98A34-1DC0 Call-ID: 314ae380-2e617dac-325-3dc712ac@172.18.199.61 CSeq: 102 NOTIFY Max-Forwards: 70 Date: Tue, 26 Jul 2005 18:15:18 GMT User-Agent: Cisco-CCM5.0 Event: telephone-event;rate=1000 Subscription-State: active;expires=-1281397684 Contact: <sip:172.18.199.61:5060> Content-Type: audio/telephone-event Content-Length: 4

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Table 1-32 Troubleshooting SIP Trunk Configuration (continued)

Symptom	Possible Cause	Recommended Action

UAS responds to an INVITE request with a 401 (Unauthorized) message.	Authentication or Authorization is failing	If authentication or authorization are not needed, make sure the appropriate check boxes on the SIP Trunk Security Profile window are unchecked.
		or
		Configure the application user on Cisco Unified CM with proper authorization.
		Configure the UAC to match credentials on Cisco Unified CM.
		`INVITE sip:5000@172.18.193.222 SIP/2.0`
		`Via: SIP/2.0/UDP 172.18.194.208:5060;branch=z9hG4bK5ba60c17`
		`From: "4900"`
		`<sip:4900@172.18.193.222>;tag=000f24c6a16f000d171bb590-611445d9`
		`To: <sip:5000@172.18.193.222>`
		`Call-ID: 000f24c6-a16f000c-245cb98c-57a622e0@172.18.194.208`
		……
		
		`SIP/2.0 401 Unauthorized`
		`From: "4900"`
		`<sip:4900@172.18.193.222>;tag=000f24c6a16f000d171bb590-611445d9`
		`To: <sip:5000@172.18.193.222>;tag=466540560`
		`Call-ID: 000f24c6-a16f000c-245cb98c-57a622e0@172.18.194.208`
		`CSeq: 101 INVITE`
		`WWW-Authenticate: DIGEST realm="siptrunk41",`
		`nonce="YoK5FiEuXpeIlp52EnUWFLIU1m24t5gV", algorithm=MD5`
		`Content-Length: 0`
		……

UAS responds to a SIP request with a 403 (Forbidden) message.	In the System->Security Profile->SIP Security Profile window, and per the default settings, the following features require authorization:
	- Presence Subscription
	- OOD REFER
	- Unsolicited NOTIFY
	- INVITE and REFER w/ Replaces header
		If authentication or authorization are not needed, make sure the appropriate check boxes on the SIP Trunk Security Profile window are unchecked.
		or
		Configure the application user on Cisco Unified CM with proper authorization.
		Configure the UAC to match credentials on Cisco Unified CM.
		`INVITE sip:5000@172.18.193.222:5060 SIP/2.0`
		`Via: SIP/2.0/UDP 172.18.195.83:5060`
		`From: sipp <sip:sipp@172.18.195.83:5060>;tag=1`
		`To: sut <sip:5000@172.18.193.222:5060>`
		`Call-ID: 1-21473@172.18.195.83`
		`Cseq: 1 INVITE`
		`Replaces: 425928@bobster.example.org;to-tag=7743;from-tag=6472`
		……
		
		`SIP/2.0 403 Forbidden`
		`Via: SIP/2.0/UDP 172.18.195.83:5060`
		`From: sipp <sip:sipp@172.18.195.83:5060>;tag=1`
		`To: sut <sip:5000@172.18.193.222:5060>;tag=673966968`
		`Date: Tue, 5 Jul 2005 18:25:02 GMT`
		`Call-ID: 1-21473@172.18.195.83`
		`CSeq: 1 INVITE`
		`Content-Length: 0`

Table 1-32 Troubleshooting SIP Trunk Configuration (continued)

Symptom	Possible Cause	Recommended Action
UAS responds to a SIP request with a 404 Not Found message.	Digit analysis failed to find the device or a route for the called number.	This could happen for many reasons. If the phone with the dialed number is not registered, or if the host in the request does not match the Cisco Unified CM host or IP address and there is no SIP route pattern to route this number or DN, Digit Analysis will fail.
Potential problems with forking exist. Downstream SIP endpoints do not receive ACK to responses.	Not actually a problem. The SIP trunk does not support downstream forking for delayed media INVITE (w/o SDP).	If downstream forking support is required, the SIP trunk must use an MTP. Make sure the “MTP Required” check box is checked on the SIP trunk configuration window.
TLS connection for SIP trunk fails with “HandleSSLError - TLS protocol error…” in SDI log (ccmtrace).	TLS SIP trunk peer X509 certificate has not been imported into the local Cisco Unified CM trust store or is the incorrect version.	Ensure that the peer X509 certificate is version 3 and has been properly imported into the local Cisco Unified CM trust store.
TLS connection for SIP trunk fails with a ‘ConnectionFailure’ alarm on the TLS SIP trunk.	X509 Certificate validation failed for the TLS SIP trunk peer.	Possible problems include X509 Subject Name mismatch or cipher string mismatch. Check the SDL log for more detailed logs under ‘validTLSConnection’ log. Reason code 1 – Got X509 certificate for neither Authenticate or Encrypted trunk. (Should not happen.) Reason code 2 – X509 Subject Common Name (CN=) mismatch with the trunk security profile settings. Reason code 3 – TLS cipher string mismatch. (The trunk security profile ‘Device Security Mode’ settings determine this.)
TLS connection from CSPS to Cisco Unified CM via SIP fails with ‘416 Unsupported URI Scheme’ or fails following a ‘301 Redirect’ message.	Cisco Unified CM does not support SIPS URI handling, and CSPS will not complete a TLS connection without SIPS URI support.	Do not use SIP TLS connections between CSPS and Cisco Unified CM.
No Diversion header exists in the outgoing INVITE message when the call is forwarded	The SIP trunk configuration option "Redirecting Diversion Header Delivery - outbound" does not get checked.	Make sure that the "Redirecting Diversion Header Delivery - outbound" is checked.
Although an end user is associated with primary extension and phone, no PUBLISH comes out.	The preceding configuration applies for SUB/NOT based presence. It does not get used for PUBLISH.	Associate the user to the line appearance in the phone number configuration window.

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Table 1-32 Troubleshooting SIP Trunk Configuration (continued)

Symptom	Possible Cause	Recommended Action
Although an end user is associated to a line appearance, no PUBLISH comes out.	PUBLISH will not get sent for the line appearance if its associated user does not have a Cisco Unified Presence license.	Assign a Cisco Unified Presence license to that user.
Application Server for click2call wants to get the call status until the call terminates, but the final NOTIFY was sent when the call is connected.	Cisco Unified CM's default behavior is to terminate the implicit subscriptions after a call connects.	Ensure the Refer-To header has a "x-cisco-monitor-call" tag.
REFER (method=BYE) does not terminate the call in a multinode Cisco Unified CM cluster.	Refer Manager cannot find the dialog to terminate.	Make sure REFER(method=BYE) comes into the same node as the node that receives REFER(method=INVITE), for the same call.

Troubleshooting Calling Party Number Transformations feature	Troubleshooting Calling Party Number Transformations feature	There are a few detailed level SDI trace line that will help us find the number Prefixed and the number of digits striped, and the CSS applied.
		Example Stripping/Prefixing Digits, CSS:
		`12/17/2008 00:43:18.826 CCM|//SIP/SIPCdpc(0,0,0)/ci=0/ccbId=0/scbId=0/globalize: Performing stripAndPrependDigits --- Prefix data = +1, Strip Data = 1|<CLID::StandAloneCluster><NID::rtp-galaxy><LVL::Detailed><MASK::20000>`
		`12/17/2008 00:43:18.826 CCM|SPROC :: stripAndPrependDigits- The number 4089023019 is prepended with prefix +1, updated number=+14089023019|<CLID::StandAloneCluster><NID::rtp-galaxy><LVL::Detailed><MASK::ffffff>`
		`12/17/2008 00:43:18.826 CCM|//SIP/SIPCdpc(0,0,0)/ci=0/ccbId=0/scbId=0/globalize: CallingNumber after stripAndPrependDigits +14089023019`
		`12/17/2008 00:58:54.923 CCM|//SIP/SIPCdpc(0,0,0)/ci=0/ccbId=0/scbId=0/globalize: Using Calling CSS 57803b7e-1bcc-b0d7-da08-f023c78dd179|<CLID::StandAloneCluster><NID::rtp-galaxy><LVL::Detailed><MASK::20000>`

Calls using SRTP over non secure trunk don't behave correctly.	Cisco Unified CM down grades to RTP (i.e. AVP)	Check if endpoint on the far end of the trunk indicates support for SAVP in the SDP.
Calls using SRTP over non secure trunk don't behave correctly.	Cisco Unified CM down grades to RTP (i.e. AVP)	Check if other call leg (e.g. perhaps a line side endpoint) on Cisco Unified CM supports SRTP.
Calls using SRTP over non secure trunk don't behave correctly.	Cisco Unified CM down grades to RTP (i.e. AVP)	Check if Cisco Unified CM had to insert an MTP (e.g. due to codec mismatch for example).

Table 1-32 Troubleshooting SIP Trunk Configuration (continued)

Symptom	Possible Cause	Recommended Action
Calls using SRTP over non secure trunk don't behave correctly.	Cisco Unified CM disconnects call during setup.	Check if endpoint on the other side of the trunk indicates support for X-cisco-srtp-fallback but the call requires RTP for other reasons (see problems above).
G.Clear calls not working for SIP line/trunk to MGCP GW T1/E1 PRI	Call drops due to media mismatch.	Check if G.Clear codec was offered by SIP leg.
G.Clear calls not working for SIP line/trunk to MGCP GW T1/E1 PRI	Call drops due to media mismatch.	Check if "Enable G.Clear" checkbox is checked on the MGCP PRI.
G.Clear calls not working for SIP line/trunk to MGCP GW T1/E1 PRI	Call rejected by PSTN/PBX with incompatible destination (88).	Check if ISDN setup from MGCP GW to PBX/PSTN has bearer cap set to 0x8890.
G.Clear calls not working for SIP line to SIP line/trunk	INVITE without G.Clear SDP going out.	Check if G.Clear codec was offered by SIP leg.
G.Clear calls not working for SIP line to SIP line/trunk	INVITE without G.Clear SDP going out	Check if SIP Profile for both the legs have "Enable Early Offer for G.Clear" enabled.
G.Clear calls not working for SIP line to SIP line/trunk	G.Clear call rejected with 488 Incompatible Media	Check if the terminating device supports G.Clear codec / data calls. Cisco IP Phones do not support G.Clear codec.

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Table 1-32 Troubleshooting SIP Trunk Configuration (continued)

Symptom	Possible Cause	Recommended Action
Logical Partitioning feature not working	Logical Partitioning feature not working	• Check if Enterprise Param "Enable Logical Partitioning" is True. • Check that the device is associated with a valid Geolocation at device or device pool level. • Check that the device is associated with a valid Geolocation filter, having selection of some of the Geolocation fields, at device or device pool level. • When Enterprise Param "Logical Partitioning Default Policy" is DENY, check if ALLOW LP policies between GeolocationPolicy of a Gateway & VoIP site configured. • Make sure case is correct for fields of the LP GeolocationPolicy records and match with that configured for Geolocation records. • There is no LP Policy check for VoIP to VoIP device call or feature with only VoIP participants. • The Cisco Unified CM Admin will allow configuring policies between Interior:geolocpolicyX to Interior:geolocpolicyY but it will not be used during LP checks. • Hierarchy is important for fields of Geolocation 1. Say searching for policy between Border:IN:KA and Interior:IN:KA 2. The following possible policies will match in order • The possible policies that lack some of the fields in hierarchy, such as following will not match: – Border:KA – Interior:KA – Border:BLR – Interior:BLR – Border:KA:BLR – Interior:KA:BLR – Note: Country=IN is missing
IPv6 calls not working	IPv6_only phones are not able to register with Cisco Unified CM	• Verify that IPv6 has been enabled on the Cisco Unified CM server using platform CLI. • Verify that "Enable IPv6" enterprise parameter is set to True. • Verify Server IPv6 Name has been configured with either the hostname or the IPv6 address. If configured for hostname, verify that phone has been configured with a DNS address to resolve hostname to an IPv6 address. • Verify that Cisco Unified CM host only has 1 non link-local IPv6 address. • If the phone will get an IPv6 address via RA, verify that "Allow Auto Configuration for Phone" enterprise parameter is set to ON. • Verify Cisco Unified CM and Cisco TFTP services are running.

Table 1-32 Troubleshooting SIP Trunk Configuration (continued)

Symptom	Possible Cause	Recommended Action
IPv6 calls not working	Incoming calls on V6_only SIP Trunk are not answered	• Verify that IPv6 has been enabled on the Cisco Unified CM server using CLI. • Verify that "Enable IPv6" enterprise parameter is set to True. • Verify that the INVITE does not contains IPv4 signaling.
IPv6 calls not working	Outgoing calls on V6_only SIP Trunk are not answered	• Verify that IPv6 has been enabled on the Cisco Unified CM server using CLI. • Verify that "Enable IPv6" enterprise parameter is set to True. • Verify that an IPv6 address is configured on the SIP Trunk configuration page.
IPv6 calls not working	Calls between two endpoint fail	Verify the addressing modes of the endpoints. If one is configured for IPv4_only and the other is configured for IPv6_only, ensure that a dual stack MTP is available to do the media translation.
IPv6 call no MOH is heard	No MOH is heard	• Verify the addressing modes of the endpoint to which MOH is played. If the addressing mode is IPv6_only and MOH is configured for unicast, ensure that a dual stack MTP is available. • If MOH is configured for multicast, the expected behavior is that no MOH will be heard on the IPv6_only phones.
Fax works but no midcall INVITE with T.38 SDP is seen.	T38fax configuration issue.	T.38 fax relay is not being used for fax transmission. • For SIP and H.323 gateways, verify that the T.38 fax is enabled under VOIP dial-peer. • For MGCP gateway, verify fxr-package is enabled. Verify that “mgcp fax t38 inhibit” is not enabled.
Fax call fails as the initial INVITE transaction is rejected with 580 or 488.	T38fax configuration issue.	Check the locations page on both clusters and verify that the qos is enabled between the SIP trunk and endpoint. Verify that the media resources like MTP(enabled for RSVP) are configured on the endpoint’s media resource group. Verify that the devices are registered with CUCM.
Initial INVITE is rejected with 420 Bad extension	End to end RSVP is not configured correctly.	If the unsupported: preconditions is seen in 420, verify that the SIP trunk is configured for E2E RSVP and PRACK is enabled on the cluster that is rejecting the call attempt.
Mid call INVITE with T.38 SDP is rejected with 488.	T38fax problem	Verify that the endpoint supports fax i.e. not an IP Phone. Verify that the endpoint is configured to support fax. See the CLI commands for the gateways.
After tandem/remote transfer, the final call is no longer E2E RSVP call.	End to end RSVP is not configured correctly.	In the transferring node, make sure the RSVP policy is activated for locations the inbound and outbound SIP Trunks assigned to.
When the call is put on hold, no E2E RSVP between the MOH server and held party.	End to end RSVP is not configured correctly.	In the holding cluster, make sure the MOH’s device pool has MRGL that has the RSVP agents assigned. Also make sure RSVP policy is activated for locations the MOH server and SIP Trunks assigned to.

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Table 1-32 Troubleshooting SIP Trunk Configuration (continued)

Symptom	Possible Cause	Recommended Action
When the device in campus (Hub_none location) is making a call, no E2E RSVP.	End to end RSVP is not configured correctly.	Make sure RSVP policy is activated between Hub_none location and location that the SIP Trunk assigned to.
When the conference call is invoked, no E2E RSVP between the Conference Bridge and remote conference participants.	End to end RSVP is not configured correctly.	In the Conference invoked cluster, make sure the Conference Bridge's device pool has MRGL that has the RSVP agents assigned. Also make sure RSVP policy is activated for locations the Conference Bridge and SIP Trunks assigned to.
When a call is blind transferred to a remote system, no E2E RSVP between the Announciator and calling phone.	End to end RSVP is not configured correctly.	In the Transferring cluster, make sure the Announciator's device pool has MRGL that has the RSVP agents assigned. Also make sure RSVP policy is activated for locations the Announciator and SIP Trunks assigned to.
The original call between agent and customer is non secure. However, the non secure supervisor received reorder when trying to monitor the agent	Agent security is not configured correctly.	Check the secure capability of the agent. If the agent is encrypted, this scenario works as designed. The supervisor needs to meet or exceed the agent's secure capability in order for the monitoring call to be successful.
When conference is started during secure monitoring, the secure icon display is incorrect	Hardware problem	Check if using hardware conference bridge or default (SW) conference bridge. SW conference bridge does not support security.
While using 7970/71 phones, if auto recording is enabled on these, we are unable to start a conference call	Codec negotiation problem	7970/71 phones offer/negotiate G.722 codec by default unless it is disabled. During recording, the codec is locked to be the same as the original call between the customer and agent. If using SW conference bridge with limited codec support, a transcoder maybe needed to complete the conference call. As alternative options, G.722 codec can be disabled on the agent phone or HW conference bridge which supports G.722 codec can be used.
The agent is secure and the recorder is secure. Auto recording is enabled on the agent. However, the CUCM does not send invite to the recorder	Network topology issue.	The SIP Trunk connected to the recorder needs to be secure as well for CUCM to send INVITE to the recorder. No MTP should be inserted on the SIP Trunk between agent and recorder (since MTP does not support security).
The agent is authenticated. Recording does not start on this agent	Configuration issue.	CUCM does not support recording on authenticated phones.

Table 1-32 Troubleshooting SIP Trunk Configuration (continued)

Symptom	Possible Cause	Recommended Action
The farend call info is missing the remote address/directory number	This could be due to feature interaction such as Call Park/Call Retrieve	
No remote farend call information update	Configuration Issue	Use a SIP Trunk, H323 Trunk, PRI DMS100, PRI DMS250, or PRI ISO Qsig T1 between the two clusters.
The b-number is missing for a remote conference	Configuration Issue	Use a SIP Trunk between the clusters and enable 'Deliver Conference Bridge Identifier' on the remote SIP Trunk's SIP Profile.
"isfocus" is missing for a remote conference	Configuration Issue	Use a SIP Trunk or H323 trunk between the two clusters.
OPTIONS message is not sent out	TCP is used and cannot create socket	If TCP is used and socket cannot be created you won't see OPTIONS going out. However you should still see the alarm in RTMT or logs.
	By default the OPTIONS is sent out every 60 or 120 seconds and you didn't wait long enough	Make sure you wait enough time or you can shorten the timer value in the SIP Profile.
	SIP trunk is not default type	Only the default type SIP trunk supports this feature. IME, SAF and EMCC SIP trunks do not. You must use a default type SIP trunk.
OPTIONS message is sent to a destination I didn't configure	If FQDN or SRV is configured as SIP trunk's destination OPTIONS is sent to ALL resolved IP addresses	Not a real issue, you can configure the IP address as the destination.
Configured SIP trunk between two UCM. One side sent OPTIONS but the other side responded with "503 Service Unavailable"	Receiving side Cisco Unified CM didn't recognize the sending side Cisco Unified CM	On the receiving side Cisco Unified CM, configure a SIP trunk that points back to the sending side Cisco Unified CM. Make sure the destination address, port number and transport type match.

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Table 1-32 Troubleshooting SIP Trunk Configuration (continued)

Symptom	Possible Cause	Recommended Action
Initial outgoing INVITE for Early Offer SIP trunk call does not have SDP	Early Offer configuration is disabled	Verify that the associated SIP Profile has "Enable Early Offer for voice and video calls" enabled.
	SIP Trunk is in Ipv6 only mode or dual mode with media preference as Ipv6	Verify that the SIP trunk is not in Ipv6 only mode or dual-mode trunk with ANAT or media preference set to Ipv6.
	Calling device is an Ipv6 only device	Verify that the Calling device is not an Ipv6 only device.
	Calling device is a pre-SCCP v20 device or H323 device and no MTP is available that supports GetPort capability	If initiating calls from a pre SCCP v20 device or H323 Slowstart device or delayed offer incoming call, verify that the MTP allocation is taking place. Ensure that the caller or SIP trunk MRGL has an MTP available. Verify that the MTP firmware supports getPort capability. If the MTP image does not support getPort capability, upgrade to a newer image with the fix for CSCtb19331 (IOS release: 15.1(2)T).
Outgoing call for Early Offer SIP trunk fails. No INVITE is sent out	Calling device is a pre-SCCP v20 device or H323 device	If initiating calls from a pre SCCP v20 device or H323 slowstart device or delayed offer incoming trunk, verify that the MTP allocation is taking place. Ensureo that MTP supports SCCP v20. If MTP allocation fails, check the configuration for "Fail Call Over SIP trunk if MTP Allocation Fails" and set it to FALSE.
	No MTP is available that supports GetPort capability	If the MTP allocation fails, reconfigure to have an MTP in MRGL associated with SIP trunk or default pool.
	The "Fail Call Over SIP trunk if MTP Allocation Fails" configuration is set to FALSE	If the MTP image does not support getPort, upgrade to newer image with the fix for CSCtb19331 (IOS release: 15.1(2)T).
Outgoing call for Early Offer SIP trunk always has SDP with one codec and MTP's IP & Port.	The "MTP Required" configuration is enabled.	Verify that "MTP Required" is not selected on the SIP Trunk page. If "MTP Required" is not selected on the SIP trunk page, check if a media resource is being allocated. Media resource can get allocated for local RSVP, TRP enabled on trunk, Early offer, DTMF mismatch or Codec mismatch.
	MTP is getting allocated and is not configured for media passthru	Verify that the media resource is configured for pass-through codec.

Table 1-32 Troubleshooting SIP Trunk Configuration (continued)

Symptom	Possible Cause	Recommended Action
Caller's QSIG/SIP Call is disconnected	On a static SIP trunk, if the originating side trunk is QSIG tunneling enabled but the terminating side trunk is not, then the call will be disconnected when originating cluster does not get first provisional response with QSIG content	On the terminating side SIP trunk, enable QSIG tunneling if possible. If the terminating cluster is based on pre-8.5 version, then disable QSIG tunneling on originating side trunk.
Cisco Telepresence MCU status shows unregistered on CUCM	Most likely a configuration issue	• Verify that the MCU address is correct • Verify that the SIP port incoming, and outgoing match what is configured on the MCU • Verify that the MCU is online and alive. • Verify that the MCU is configured for SIP. • Verify the UCM traces, and see the results for the OPTIONS ping to the MCU.
MCU shows registered but can not make conference calls using that MCU	Most likely a configuration issue	Verify that the MCU http information is configured correctly, and match what is configured on the UCM, mainly: 1. Admin user name 2. Admin password 3. HTTP port. 4. Verify in the UCM traces that http create Requests are being sent to the MCU, and that the MCU is sending an HTTP response back. 5. Verify that the MCU http information is configured correctly with the port reservation setting enabled.
It is not possible to add a 4th participant to the MCU Ad-Hoc conference	Most likely not enough ports available	Verify that the MCU has enough ports. Verify in the UCM traces that the HTTP modify Request is sent and that the MCU sends a Response back to UCM
V.150.1 (MER or Legacy) capabilities are lost over SIP trunk	Most likely a configuration issue	Ensure that appropriate V.150.1 SDP Filtering options are set for the trunk. Filtering options are set via the trunks associated SIP Trunk Security Profile and the "SIP V.150 Outbound Offer SDP filtering" service parameter.
MP4A-LATM codec is not selected for the call, even though both EPs supports the codec and the Region configuration limits the codec bit rate to that of MP4A-LATM	The EPs does not have matching MP4A-LATM codec parameters, even though both are MP4A-LATM-capable For example: Tandberg MXP1700 vs. Tandberg E20	For MP4A-LATM codec to be selected, not only the two End must both support the codec, they must also match on the following MP4A-LATM-specific parameters: • clock rate • profile-level-id • object • bitrate

Table 1-32 Troubleshooting SIP Trunk Configuration (continued)

Symptom	Possible Cause	Recommended Action
G.722.1codec is not selected for the call, even though both EPs supports the codec and the Region configuration limits audio maximum bandwidth as 32K or 24 K	Verify that both ends provide clock rate,	Cisco Unified CM only supports the G.722.1 codec with a clock rate of 16000. Both ends must provide the same clock rate.
One way audio	Verify the SDL trace.	Ensure that the dynamic Payload number is propagated between endpoints correctly if both ends are SIP devices. If a call has SIP and H.323 devices, the SIP device must honor H.323 side's dynamic PT number; this negotiation via reInvite is issued by Cisco Unified CM and sent to the SIP side. The SIP device must send back the same dynamic PT number via 200 OK. Ensure that reInvite has a=x-cisco-symm-pt.
Presentation sharing via BFCP does not work or Cisco Unified CM rejects theBFCP stream	Configuration issue or endpoint issue	There are several scenarios in which CUCM rejects the BFCP and Presentation stream. Here are a few common scenarios: • The “Allow Presentation Sharing using BFCP” SIP Profile checkbox on the SIP Trunk or SIP Line is not enabled. • One Party offers BFCP and the other Party does not offer BFCP. • SIP endpoint to non-SIP endpoint. • Both sides offer BFCP but the floor control attributes within BFCP application line are in conflict. • Unsupported BFCP transport type is offered (e.g. "TCP/BFCP") • MTP, TRP, Transcoder, or RSVP Agent is inserted in the call.

CHAPTER 2

SIP Trunk Call Flows

This chapter describes the new and modified SIP trunk call flows for OL-22509-01 Release 8.6(1). It shows the interfaces and interactions between OL-22509-01 and SIP networks for basic calls and supplementary services. The new features and enhancements that are introduced in this release do not impose any backward compatibility implications on previous versions of the SIP trunk.

Note This chapter includes the new callflows added to Unfied CM 8.6(1). It is recommended that you view the complete list of exisitng SIP basic call flows from SIP Line Messaging Guide (Standard) for Release 8.0(1)from:
http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_programming_reference_guides_list.html

Tip A few SIP devices are not SIP compliant, and therefore, they do not support delayed media (INVITE without SDP). To interoperate with these types of devices, the SIP trunk that is associated with these devices must have MTP enabled.

This document captures the callflows introduced/modified in 8.6(1) involving the SIP Trunk.

This document contains a number of SIP call flows. Each one has the same format:

1. A description for the scenario that describes what is happening from the various phone user's perspective.
2. The Message Sequence Charts (MSCs) including phones and Unified CM nodes involved in the various SIP transactions. Each MSC uses the following conventions:
   a. Each message is preceded by a dialog identifier and message number relative to the scenario. For example "(d2) [3] INVITE..." indicates that the INVITE is the third message in the sequence and is in the second dialog. Other messages in the same dialog will also have (d2) at the beginning. Also, the message link color is the same for all messages within a dialog.
   b. Phones are named with letters. For example, Phone A.
   c. Nodes in the MSC presenting phones will be labled with the phone letter and the line number. For example, A 1100 represents line 1100 on phone A.
3. The detailed SIP messages associated with the MSC.

Each message arrow is clickable. Clicking a message arrow will cause the document to jump to the detailed message. Clicking [diagram] above the message will cause the document to jump back to the page with MSC that contains that message.

Note All the callflows captured in this document are just for understanding purposes. Message flows might be different in real deployment due to other configuration changes. These message callflows cannot be considered AS IS for interop with 3rd party servers.

List of Scenarios:

1. AMR/AMR-WB Codec Support

1.1 Basic Call between 3rd party SIP device via an Inter Cluster SIP Trunk

Title: Basic Call between 3rd party SIP device via an Inter Cluster SIP Trunk

Description:

The following call flow illustrates the SIP messaging that takes place between two 3rd part SIP devices CMs via an inter cluster SIP trunk.

We need to verify if CUCM passes AMR/AMR-WB codec
Client 1 sent out the initial INVITE.

Configuration:

Node = Unified CM1, IP = 10.89.79.115

Phone = A, Line = 9728135282, IP = 10.77.31.171, Model = 3rd party SIP
Phone = B, Line = 2145359087, IP = 10.77.31.133, Model = 3rd party SIP

Unified CM1 is a Tandem Node and Connected to the client with a SIP Trunk

Scenario:

Phone A DN 9728135282 calls Phone B DN 2145359087 over the SIP Trunk
Phone B DN 2145359087 Answers
Phone A DN 9728135282 goes onHook

End of Scenario

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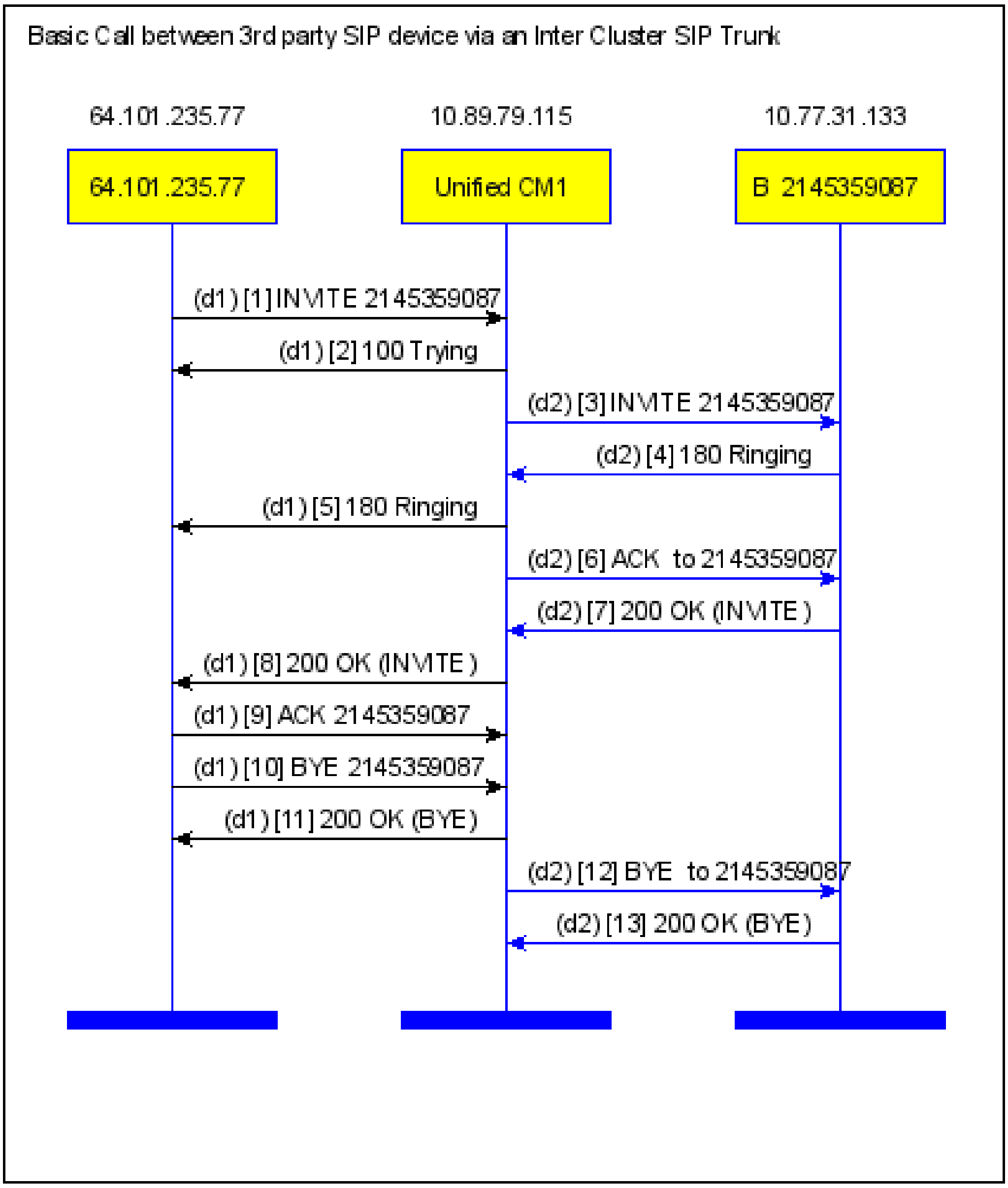

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```
[diagram]   Call-ID:[prev][next]
[1] INVITE sip:2145359087@10.89.79.115:5060 SIP/2.0

Via: SIP/2.0/UDP 64.101.235.77:5060

From: <sip:9728135282@64.101.235.77:5060>;tag=1

To: <sip:2145359087@10.89.79.115:5060>

Call-ID: 1-6152@64.101.235.77

CSeq: 1 INVITE

Contact: sip:9728135282@64.101.235.77:5060

P-Asserted-Identity:<sip:3489900888@SIPI.VODAFONE.IT;user=phone>,<tel:3489900888;phone-context=+39>

P-Charging-Vector:icid-vlaue=kp0maaefrlewaaaabozav;icid-generated-at=19.77.31.190;orig-ioi=19.77.31.190

Max-Forwards: 70

Subject: Performance Test

Content-Type: application/sdp

Content-Length:   410


v=0

o=user1 53655765 2353687637 IN IP4 10.77.31.171

s=-

c=IN IP4 10.77.31.171

t=0 0

m=audio 49194 RTP/AVP 8 102 104 101

a=rtpmap:8 PCMA/8000

a=rtpmap:102 AMR/8000/1

a=fmtp:102 mode-set=0,1,2,3,4,5,6,7; mode-change-period=1; mode-change-capability=2;

a=rtpmap:104 AMR-WB/16000/1

a=fmtp:104 mode-change-period=1; mode-change-capability=2;

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15

a=sendrecv

[diagram]   Call-ID:[prev][next]
[2] SIP/2.0 100 Trying

Via: SIP/2.0/UDP 64.101.235.77:5060

From: <sip:9728135282@64.101.235.77:5060>;tag=1

To: <sip:2145359087@10.89.79.115:5060>

Date: Tue, 30 Aug 2011 19:13:31 GMT

Call-ID: 1-6152@64.101.235.77

CSeq: 1 INVITE

Allow-Events: presence

Content-Length: 0

[diagram]   Call-ID:[prev][next]
[3] INVITE sip:2145359087@10.77.31.133:5060 SIP/2.0

Via: SIP/2.0/UDP 10.89.79.115:5060;branch=z9hG4bKd6ec50197
```

```
From: <sip:3489900888@10.89.79.115>;tag=61~072801bb-dbba-4423-9b2c-2040b9a2e5bb-24431038
To: <sip:2145359087@10.77.31.133>
Date: Tue, 30 Aug 2011 19:13:31 GMT
Call-ID: 24c5c500-e5d1365b-6-734f590a@10.89.79.115
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM8.6
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Cisco-Guid: 0616940800-0000065536-0000000010-1934579978
Session-Expires:  1800
P-Asserted-Identity: <sip:3489900888@10.89.79.115>
Remote-Party-ID: <sip:3489900888@10.89.79.115>;party=calling;screen=yes;privacy=off
Contact: <sip:3489900888@10.89.79.115:5060>
Max-Forwards: 69
Content-Type: application/sdp
Content-Length: 432

v=0
o=CiscoSystemsCCM-SIP 61 1 IN IP4 10.89.79.115
s=SIP Call
c=IN IP4 10.77.31.171
b=TIAS:64000
b=AS:64
t=0 0
m=audio 49194 RTP/AVP 104 102 8 101
a=rtpmap:104 AMR-WB/16000
a=fmtp:104 mode-change-period=1;mode-change-capability=2
a=rtpmap:102 AMR/8000
a=fmtp:102 mode-set=0,1,2,3,4,5,6,7;mode-change-period=1;mode-change-capability=2
a=rtpmap:8 PCMA/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

[diagram] Call-ID:[prev][next]

[4] SIP/2.0 180 Ringing

```
Via: SIP/2.0/UDP 10.89.79.115:5060;branch=z9hG4bKd6ec50197
From: <sip:3489900888@10.89.79.115>;tag=61~072801bb-dbba-4423-9b2c-2040b9a2e5bb-24431038
To: <sip:2145359087@10.77.31.133>;tag=1
Call-ID: 24c5c500-e5d1365b-6-734f590a@10.89.79.115
CSeq: 101 INVITE
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[5] SIP/2.0 180 Ringing

```
Via: SIP/2.0/UDP 64.101.235.77:5060
From: <sip:9728135282@64.101.235.77:5060>;tag=1
To: <sip:2145359087@10.89.79.115:5060>;tag=60~072801bb-dbba-4423-9b2c-2040b9a2e5bb-24431037
Date: Tue, 30 Aug 2011 19:13:31 GMT
Call-ID: 1-6152@64.101.235.77
CSeq: 1 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:2145359087@10.89.79.115>
Remote-Party-ID: <sip:2145359087@10.89.79.115>;party=called;screen=yes;privacy=off
Contact: <sip:2145359087@10.89.79.115:5060>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[6] ACK sip:10.77.31.133:5060 SIP/2.0

```
Via: SIP/2.0/UDP 10.89.79.115:5060;branch=z9hG4bKe52e27db8
From: <sip:3489900888@10.89.79.115>;tag=61~072801bb-dbba-4423-9b2c-2040b9a2e5bb-24431038
To: <sip:2145359087@10.77.31.133>;tag=1
Date: Tue, 30 Aug 2011 19:13:31 GMT
Call-ID: 24c5c500-e5d1365b-6-734f590a@10.89.79.115
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[7] SIP/2.0 200 OK

```
Via: SIP/2.0/UDP 10.89.79.115:5060;branch=z9hG4bKd6ec50197
From: <sip:3489900888@10.89.79.115>;tag=61~072801bb-dbba-4423-9b2c-2040b9a2e5bb-24431038
To: <sip:2145359087@10.77.31.133>;tag=1
Call-ID: 24c5c500-e5d1365b-6-734f590a@10.89.79.115
CSeq: 101 INVITE
Contact: <sip:10.77.31.133:5060>
Call-Info: 0003CBB8-7DB7-185A-AE76-0A7802D2AA77-1299%4010.120.2.210;gen-rt=ds-54de-a473120f;gen-lt=16795319
```

```
Session-Expires: 1800;refresher=uas
Min-SE: 90
Supported: timer
Content-Type: application/sdp
Content-Length:   299

v=0
o=Genesys 3141163340 3141163341 IN IP4 10.77.31.133
s=Genesys SIP Call
c=IN IP4 10.77.31.133
t=0 0
m=audio 2438 RTP/AVP 102 101
a=rtpmap:102 AMR/8000/1
a=fmtp:102 mode-set=0,1,2,3,4,5,6,7; mode-change-period=1; mode-change-capability=2;
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

[diagram]   Call-ID:[prev][next]
[8] SIP/2.0 200 OK

```
Via: SIP/2.0/UDP 64.101.235.77:5060
From: <sip:9728135282@64.101.235.77:5060>;tag=1
To: <sip:2145359087@10.89.79.115:5060>;tag=60~072801bb-dbba-4423-9b2c-2040b9a2e5bb-24431037
Date: Tue, 30 Aug 2011 19:13:31 GMT
Call-ID: 1-6152@64.101.235.77
CSeq: 1 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:2145359087@10.89.79.115>
Remote-Party-ID: <sip:2145359087@10.89.79.115>;party=called;screen=yes;privacy=off
Contact: <sip:2145359087@10.89.79.115:5060>
Content-Type: application/sdp
Content-Length: 227

v=0
o=CiscoSystemsCCM-SIP 60 1 IN IP4 10.89.79.115
s=SIP Call
c=IN IP4 10.77.31.133
t=0 0
```

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```
m=audio 2438 RTP/AVP 102

a=rtpmap:102 AMR/8000

a=fmtp:102 mode-set=0,1,2,3,4,5,6,7;mode-change-period=1;mode-change-capability=2
```

[diagram] Call-ID:[prev][next]
[9] ACK sip:2145359087@10.89.79.115:5060 SIP/2.0

```
Via: SIP/2.0/UDP 64.101.235.77:5060

From: <sip:9728135282@64.101.235.77:5060>;tag=1

To: <sip:2145359087@10.89.79.115:5060>;tag=60~072801bb-dbba-4423-9b2c-2040b9a2e5bb-24431037

Call-ID: 1-6152@64.101.235.77

CSeq: 1 ACK

Contact: sip:9728135282@64.101.235.77:5060

Max-Forwards: 70

Subject: Performance Test

Content-Length:     0
```

[diagram] Call-ID:[prev][next]
[10] BYE sip:2145359087@10.89.79.115:5060 SIP/2.0

```
Via: SIP/2.0/UDP 64.101.235.77:5060

From: <sip:9728135282@64.101.235.77:5060>;tag=1

To: <sip:2145359087@10.89.79.115:5060>;tag=60~072801bb-dbba-4423-9b2c-2040b9a2e5bb-24431037

Call-ID: 1-6152@64.101.235.77

CSeq: 2 BYE

Contact: sip:9728135282@64.101.235.77:5060

Max-Forwards: 70

Subject: Performance Test

Content-Length:     0
```

[diagram] Call-ID:[prev][next]
[11] SIP/2.0 200 OK

```
Via: SIP/2.0/UDP 64.101.235.77:5060

From: <sip:9728135282@64.101.235.77:5060>;tag=1

To: <sip:2145359087@10.89.79.115:5060>;tag=60~072801bb-dbba-4423-9b2c-2040b9a2e5bb-24431037

Date: Tue, 30 Aug 2011 19:13:37 GMT

Call-ID: 1-6152@64.101.235.77

CSeq: 2 BYE

Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[12] BYE sip:10.77.31.133:5060 SIP/2.0

```
Via: SIP/2.0/UDP 10.89.79.115:5060;branch=z9hG4bKf117a91

From: <sip:3489900888@10.89.79.115>;tag=61~072801bb-dbba-4423-9b2c-2040b9a2e5bb-24431038

To: <sip:2145359087@10.77.31.133>;tag=1

Date: Tue, 30 Aug 2011 19:13:31 GMT

Call-ID: 24c5c500-e5d1365b-6-734f590a@10.89.79.115

User-Agent: Cisco-CUCM8.6
```

```
Max-Forwards: 70

P-Asserted-Identity: <sip:3489900888@10.89.79.115>

CSeq: 102 BYE

Reason: Q.850;cause=16

Content-Length: 0
```

[diagram]   Call-ID:[prev][next]
[13] SIP/2.0 200 OK

```
Via: SIP/2.0/UDP 10.89.79.115:5060;branch=z9hG4bKf117a91

From: <sip:3489900888@10.89.79.115>;tag=61~072801bb-dbba-4423-9b2c-2040b9a2e5bb-24431038

To: <sip:2145359087@10.77.31.133>;tag=1

Call-ID: 24c5c500-e5d1365b-6-734f590a@10.89.79.115

CSeq: 102 BYE

Content-Length: 0
```

2. BFCP Support

2.1 Basic call with Presentation between Cisco Telepresence EX90s via a SIP ICT

Title: Basic call with Presentation between Cisco Telepresence EX90s via a SIP ICT

Description:
The following call flow illustrates the SIP messaging that takes place between two Cisco Unified CMs via an inter cluster SIP trunk for a presentation call between two EX90.

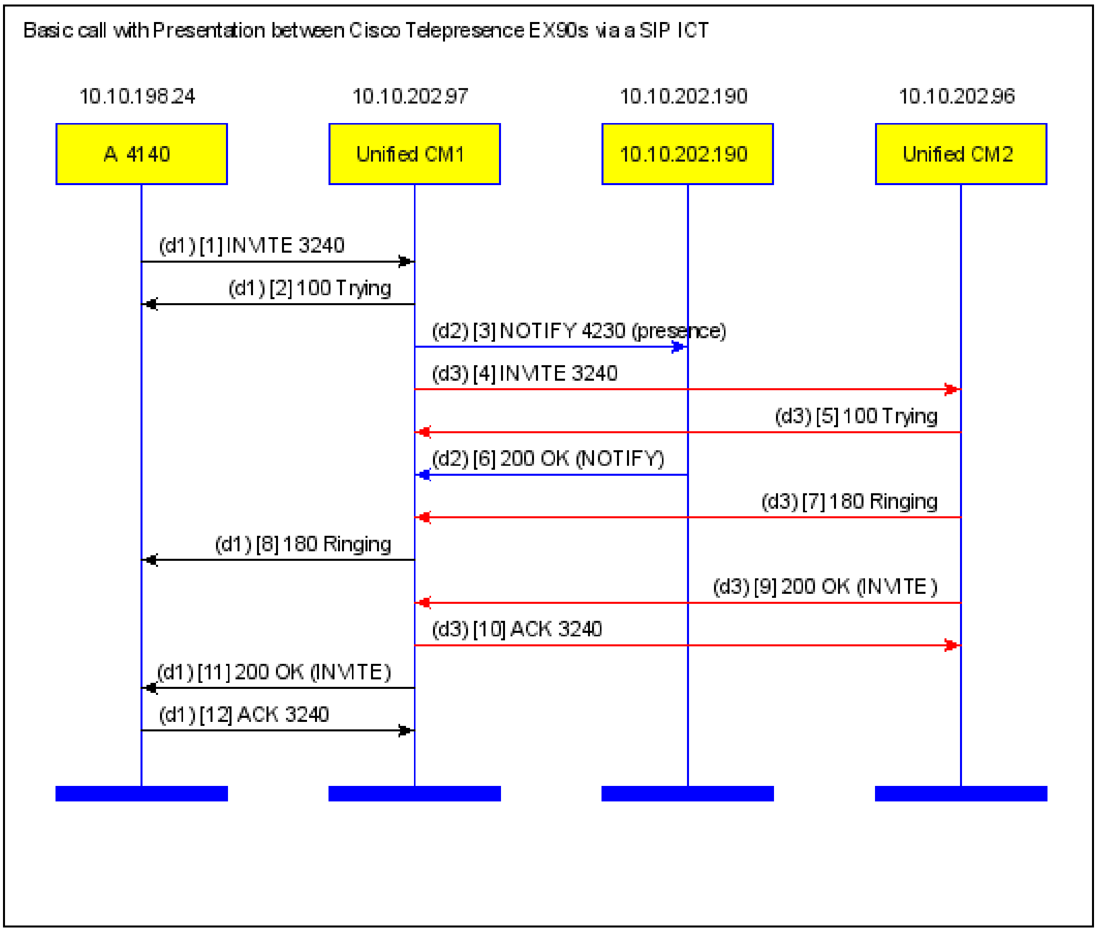
Cisco Unified CM1 sent out the initial INVITE.

Configuration:
Node = Unified CM1, IP = 10.10.202.97
Node = Unified CM2, IP = 10.10.202.96

Phone = A, Line = 4140, IP = 10.10.198.24, Model = Cisco Telepresence EX90
Phone = B, Line = 3240, IP = 10.10.202.251, Model = Cisco Telepresence EX90

A and B each have laptop connected to it via HDMI.
SIP Trunk between Unified CM1 and Unified CM2 has a route pattern 3XXX.

Scenario:
Phone A DN 4140 calls Phone B DN 3240
Phone B DN 3240 answers the call
Phone A DN 4140 presses Present and chooses HDMI as input source for presentation
Main video and Presentation video are seen on Phone B DN 3240
End of Scenario


Basic call with Presentation between Cisco Telepresence EX90s via a SIP ICT
10.10.198.24
10.10.202.97
10.10.202.190
10.10.202.96
A 4140
Unified CM1
10.10.202.190
Unified CM2
(d1) [1] INVITE 3240
(d1) [2] 100 Trying
(d2) [3] NOTIFY 4230 (presence)
(d3) [4] INVITE 3240
(d3) [5] 100 Trying
(d2) [6] 200 OK (NOTIFY)
(d3) [7] 180 Ringing
(d1) [8] 180 Ringing
(d3) [9] 200 OK (INVITE)
(d3) [10] ACK 3240
(d1) [11] 200 OK (INVITE)
(d1) [12] ACK 3240

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[diagram]   Call-ID:[prev][next]
[1] INVITE sip:3240@10.10.202.97 SIP/2.0
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK851fe812245a3c3ac51855d4352beed4.1;rport
Call-ID: 73dd5aec16b7b5ac@10.10.198.24
CSeq: 100 INVITE
Contact: <sip:4140@10.10.198.24:44773;transport=tcp>
From: <sip:4140@10.10.202.97>;tag=57b50d06ceecce60
To: <sip:3240@10.10.202.97>
Max-Forwards: 70
Route: <sip:10.10.202.97;lr>
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-service-
control,norefersub,extended-refer
Session-Expires: 1800
Remote-Party-ID: <sip:4140@10.10.202.97>;privacy=off;id-type=subscriber;screen=yes;party=calling
Content-Type: application/sdp
Content-Length: 1575

v=0
o=tandberg 7 1 IN IP4 10.10.198.24
s=-
c=IN IP4 10.10.198.24
b=AS:6000
t=0 0
m=audio 16424 RTP/AVP 100 102 103 104 105 106 9 8 0 101
b=TIAS:128000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 profile-level-id=25;object=23;bitrate=128000
a=rtpmap:102 MP4A-LATM/90000
a=fmtp:102 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:103 MP4A-LATM/90000
a=fmtp:103 profile-level-id=24;object=23;bitrate=56000
a=rtpmap:104 MP4A-LATM/90000
a=fmtp:104 profile-level-id=24;object=23;bitrate=48000
a=rtpmap:105 G7221/16000
a=fmtp:105 bitrate=32000
a=rtpmap:106 G7221/16000
a=fmtp:106 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
m=video 16426 RTP/AVP 97 98 99 34 31
b=TIAS:6000000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=428016;max-br=5000;max-mbps=108000;max-fs=3600;max-smbps=108000;max-fps=6000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=428016;max-br=5000;max-mbps=108000;max-fs=3600;max-smbps=108000;packetization-
mode=1;max-fps=6000
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1280,768,3;custom=1280,720,3;custom=1024,768,1;custom=1024,576,2;custom=800,600,1;cif4=1;custom=720,4
80,1;custom=640,480,1;custom=512,288,1;cif=1;custom=352,240,1;qcif=1;maxbr=20000
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=1;cif=1;qcif=1;maxbr=20000
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=20000
a=rtcp-fb:* nack pli
a=sendrecv
a=content:main
a=label:11
a=answer:full
m=application 16430 RTP/AVP 107
a=rtpmap:107 H224/4800
a=sendrecv

[diagram]   Call-ID:[prev][next]
[2] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK851fe812245a3c3ac51855d4352beed4.1;rport
From: <sip:4140@10.10.202.97>;tag=57b50d06ceecce60
To: <sip:3240@10.10.202.97>
Date: Wed, 22 Jun 2011 15:18:36 GMT
Call-ID: 73dd5aec16b7b5ac@10.10.198.24
CSeq: 100 INVITE
Allow-Events: presence
Content-Length: 0
```

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[diagram]   Call-ID:[prev][next]
[3] NOTIFY sip:4230@10.10.202.190:38139;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12033bdb96e
From: <sip:4140@10.10.202.97>;tag=1669962485
To: <sip:4230@10.10.202.97>;tag=0653a9395eb8cc0d
Call-ID: 14004e146e462686@10.10.202.190
CSeq: 134 NOTIFY
Max-Forwards: 70
Date: Wed, 22 Jun 2011 15:18:36 GMT
User-Agent: Cisco-CUCM8.6
Event: presence
Subscription-State: active;expires=2551
Contact: <sip:4140@10.10.202.97:5060;transport=tcp>
Content-Type: application/pidf+xml
Content-Length: 826

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<presence xmlns="urn:ietf:params:xml:ns:pidf" entity="sip:4140@10.10.202.97"
xmlns:e="urn:ietf:params:xml:ns:pidf:status:rpid" xmlns:dm="urn:ietf:params:xml:ns:pidf:data-model"
xmlns:ce="urn:cisco:params:xml:ns:pidf:rpid" xmlns:sc="urn:ietf:params:xml:ns:pidf:servcaps">
  <dm:person>
    <status>
    <basic>open</basic>
    </status>
    <e:activities>
      <e:on-the-phone/>
    </e:activities>
  </dm:person>
  <tuple id="cmp-1-351">
    <status>
      <basic>open</basic>
      <e:activities>
        <e:on-the-phone/>
      </e:activities>
    </status>
    <sc:servcaps>
      <sc:audio>true</sc:audio>
    </sc:servcaps>
    <contact priority="0.8">sip:4140@10.10.202.97:5060</contact>
    <timestamp>2011-06-22T15:18:36Z</timestamp>
  </tuple>
</presence>

[diagram]   Call-ID:[prev][next]
[4] INVITE sip:3240@10.10.202.96:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK1211ea452d4
From: <sip:4140@vcs.domain>;tag=9833~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037388
To: <sip:3240@10.10.202.96>
Date: Wed, 22 Jun 2011 15:18:36 GMT
Call-ID: e4feb380-e02107cc-21-61ca12ac@10.10.202.97
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Cisco-Guid: 3841897344-0000065536-0000000007-1640633004
Session-Expires:  1800
P-Asserted-Identity: <sip:4140@vcs.domain>
Remote-Party-ID: <sip:4140@vcs.domain>;party=calling;screen=yes;privacy=off
Contact: <sip:4140@10.10.202.97:5060;transport=tcp>;video;audio
Max-Forwards: 69
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[5] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK1211ea452d4
From: <sip:4140@vcs.domain>;tag=9833~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037388
To: <sip:3240@10.10.202.96>
Date: Wed, 22 Jun 2011 15:18:36 GMT
Call-ID: e4feb380-e02107cc-21-61ca12ac@10.10.202.97
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[6] SIP/2.0 200 OK
```

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12033bdb96e;received=10.10.202.97
Call-ID: 14004e146e462686@10.10.202.190
CSeq: 134 NOTIFY
From: <sip:4140@10.10.202.97>;tag=1669962485
To: <sip:4230@10.10.202.97>;tag=0653a9395eb8cc0d
Server: TANDBERG/257 (TE4.1.0.253886Alpha4)
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[7] SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK1211ea452d4
From: <sip:4140@vcs.domain>;tag=9833~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037388
To: <sip:3240@10.10.202.96>;tag=15911~b43ea526-a700-4a22-8605-9a5b66389cb6-17037404
Date: Wed, 22 Jun 2011 15:18:36 GMT
Call-ID: e4feb380-e02107cc-21-61ca12ac@10.10.202.97
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:3240@10.10.202.96>
Remote-Party-ID: <sip:3240@10.10.202.96>;party=called;screen=yes;privacy=off
Contact: <sip:3240@10.10.202.96:5060;transport=tcp>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[8] SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK851fe812245a3c3ac51855d4352beed4.1;rport
From: <sip:4140@10.10.202.97>;tag=57b50d06ceecce60
To: <sip:3240@10.10.202.97>;tag=9832~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037387
Date: Wed, 22 Jun 2011 15:18:36 GMT
Call-ID: 73dd5aec16b7b5ac@10.10.198.24
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; ui-state= ringout;
gci= 1-162007; call-instance= 1
Send-Info: conference, x-cisco-conference
Remote-Party-ID: <sip:3240@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:3240@10.10.202.97:5060;transport=tcp>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[9] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK1211ea452d4
From: <sip:4140@vcs.domain>;tag=9833~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037388
To: <sip:3240@10.10.202.96>;tag=15911~b43ea526-a700-4a22-8605-9a5b66389cb6-17037404
Date: Wed, 22 Jun 2011 15:18:36 GMT
Call-ID: e4feb380-e02107cc-21-61ca12ac@10.10.202.97
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Session-Expires:  1800;refresher=uas
Require:  timer
P-Asserted-Identity: <sip:3240@10.10.202.96>
Remote-Party-ID: <sip:3240@10.10.202.96>;party=called;screen=yes;privacy=off
Contact: <sip:3240@10.10.202.96:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 2228

v=0
o=CiscoSystemsCCM-SIP 15911 1 IN IP4 10.10.202.96
s=SIP Call
b=TIAS:6000000
b=AS:6000
t=0 0
m=audio 16520 RTP/AVP 100 102 103 104 9 105 106 0 8 101
c=IN IP4 10.10.202.251
b=TIAS:128000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 bitrate=128000;profile-level-id=25;object=23
a=rtpmap:102 MP4A-LATM/90000
a=fmtp:102 bitrate=64000;profile-level-id=24;object=23
a=rtpmap:103 MP4A-LATM/90000
a=fmtp:103 bitrate=56000;profile-level-id=24;object=23
a=rtpmap:104 MP4A-LATM/90000
a=fmtp:104 bitrate=48000;profile-level-id=24;object=23
```

```
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:105 G7221/16000
a=fmtp:105 bitrate=32000
a=rtpmap:106 G7221/16000
a=fmtp:106 bitrate=24000
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:8 PCMA/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 16522 RTP/AVP 97 98 99 34 31
c=IN IP4 10.10.202.251
b=TIAS:5872000
a=label:11
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=428016;max-mbps=108000;max-fs=3600;max-cpb=200;max-br=5000;max-smbps=108000;max-
fps=6000
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-smbps=108000;max-fps=6000
a=rtpmap:99 H263-1998/90000
a=fmtp:99 QCIF=1;CIF=1;CIF4=1;CUSTOM=352,240,1
a=rtpmap:34 H263/90000
a=fmtp:34 QCIF=1;CIF=1;CIF4=1
a=rtpmap:31 H261/90000
a=fmtp:31 CIF=1;QCIF=1
a=content:main
a=rtcp-fb:* nack pli
m=video 16524 RTP/AVP 97 98 99 34 31
c=IN IP4 10.10.202.251
b=TIAS:5872000
a=label:12
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=428016;max-mbps=108000;max-fs=3840;max-cpb=200;max-br=5000;max-smbps=108000;max-
fps=6000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=428016;packetization-mode=1;max-mbps=108000;max-fs=3840;max-cpb=200;max-
br=5000;max-smbps=108000;max-fps=6000
a=rtpmap:99 H263-1998/90000
a=fmtp:99 QCIF=1;CIF=1;CIF4=1;CUSTOM=352,240,1
a=rtpmap:34 H263/90000
a=fmtp:34 QCIF=1;CIF=1;CIF4=1
a=rtpmap:31 H261/90000
a=fmtp:31 CIF=1;QCIF=1
a=content:slides
a=rtcp-fb:* nack pli
m=application 5070 UDP/BFCP *
c=IN IP4 10.10.202.251
a=floorctrl:c-s
a=floorid:2 mstrm:12
a=confid:1
a=userid:19
m=application 16526 RTP/AVP 107
c=IN IP4 10.10.202.251
a=rtpmap:107 H224/0
```

[diagram]   Call-ID:[prev][next]
[10] ACK sip:3240@10.10.202.96:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12241f97ea8
From: <sip:4140@vcs.domain>;tag=9833~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037388
To: <sip:3240@10.10.202.96>;tag=15911~b43ea526-a700-4a22-8605-9a5b66389cb6-17037404
Date: Wed, 22 Jun 2011 15:18:36 GMT
Call-ID: e4feb380-e02107cc-21-61ca12ac@10.10.202.97
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 1392

v=0
o=CiscoSystemsCCM-SIP 9833 1 IN IP4 10.10.202.97
s=SIP Call
b=TIAS:6000000
b=AS:6000
t=0 0
m=audio 16424 RTP/AVP 102 101
c=IN IP4 10.10.198.24
b=TIAS:64000
```

```
a=rtpmap:102 MP4A-LATM/90000
a=fmtp:102 bitrate=64000;profile-level-id=24;object=23
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 16426 RTP/AVP 98
c=IN IP4 10.10.198.24
b=TIAS:5936000
a=label:11
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-smbps=108000;max-fps=6000
a=content:main
a=rtcp-fb:* nack pli
m=video 0 RTP/AVP 97 98 99 34 31
c=IN IP4 10.10.202.251
b=TIAS:5872000
a=label:12
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=428016;max-mbps=108000;max-fs=3840;max-cpb=200;max-br=5000;max-smbps=108000;max-
fps=6000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=428016;packetization-mode=1;max-mbps=108000;max-fs=3840;max-cpb=200;max-
br=5000;max-smbps=108000;max-fps=6000
a=rtpmap:99 H263-1998/90000
a=fmtp:99 QCIF=1;CIF=1;CIF4=1;CUSTOM=352,240,1
a=rtpmap:34 H263/90000
a=fmtp:34 QCIF=1;CIF=1;CIF4=1
a=rtpmap:31 H261/90000
a=fmtp:31 CIF=1;QCIF=1
a=content:slides
a=rtcp-fb:* nack pli
m=application 0 UDP/BFCP *
c=IN IP4 10.10.202.251
a=floorctrl:c-s
a=floorid:2 mstrm:12
a=confid:1
a=userid:19
m=application 16430 RTP/AVP 107
c=IN IP4 10.10.198.24
a=rtpmap:107 H224/0
```

[diagram]   Call-ID:[prev][next]

[11] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK851fe812245a3c3ac51855d4352beed4.1;rport
From: <sip:4140@10.10.202.97>;tag=57b50d06ceecce60
To: <sip:3240@10.10.202.97>;tag=9832~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037387
Date: Wed, 22 Jun 2011 15:18:36 GMT
Call-ID: 73dd5aec16b7b5ac@10.10.198.24
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; gci= 1-162007;
call-instance= 1
Send-Info: conference, x-cisco-conference
Session-Expires:  1800;refresher=uas
Require:  timer
Remote-Party-ID: <sip:3240@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:3240@10.10.202.97:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 614

v=0
o=CiscoSystemsCCM-SIP 9832 1 IN IP4 10.10.202.97
s=SIP Call
c=IN IP4 10.10.202.251
b=AS:6000
t=0 0
m=audio 16520 RTP/AVP 102 101
b=TIAS:64000
a=rtpmap:102 MP4A-LATM/90000
a=fmtp:102 bitrate=64000;profile-level-id=24;object=23
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 16522 RTP/AVP 98
b=TIAS:5936000
a=label:11
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-smbps=108000;max-fps=6000
a=content:main
a=rtcp-fb:* nack pli
m=application 16526 RTP/AVP 107
a=rtpmap:107 H224/0
```

[diagram]   Call-ID:[prev][next]

```
[12] ACK sip:3240@10.10.202.97:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK00bb6c622fc1b4749fe6dd600578d3fb.1;rport
Call-ID: 73dd5aec16b7b5ac@10.10.198.24
CSeq: 100 ACK
From: <sip:4140@10.10.202.97>;tag=57b50d06ceecce60
To: <sip:3240@10.10.202.97>;tag=9832~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037387
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-service-
control,norefersub,extended-refer
Content-Length: 0
```

2.2 Basic call with Presentation between EX90 and E20 via a SIP ICT

Title: Basic call with Presentation between EX90 and E20 via a SIP ICT

Description:

The following call flow illustrates the SIP messaging that takes place between two Cisco Unified CMs via an inter cluster SIP trunk for a presentation call between EX90 and E20.

Cisco Unified CM1 sent out the initial INVITE.

Configuration:

Node = Unified CM1, IP = 10.10.202.97
Node = Unified CM2, IP = 10.10.202.96

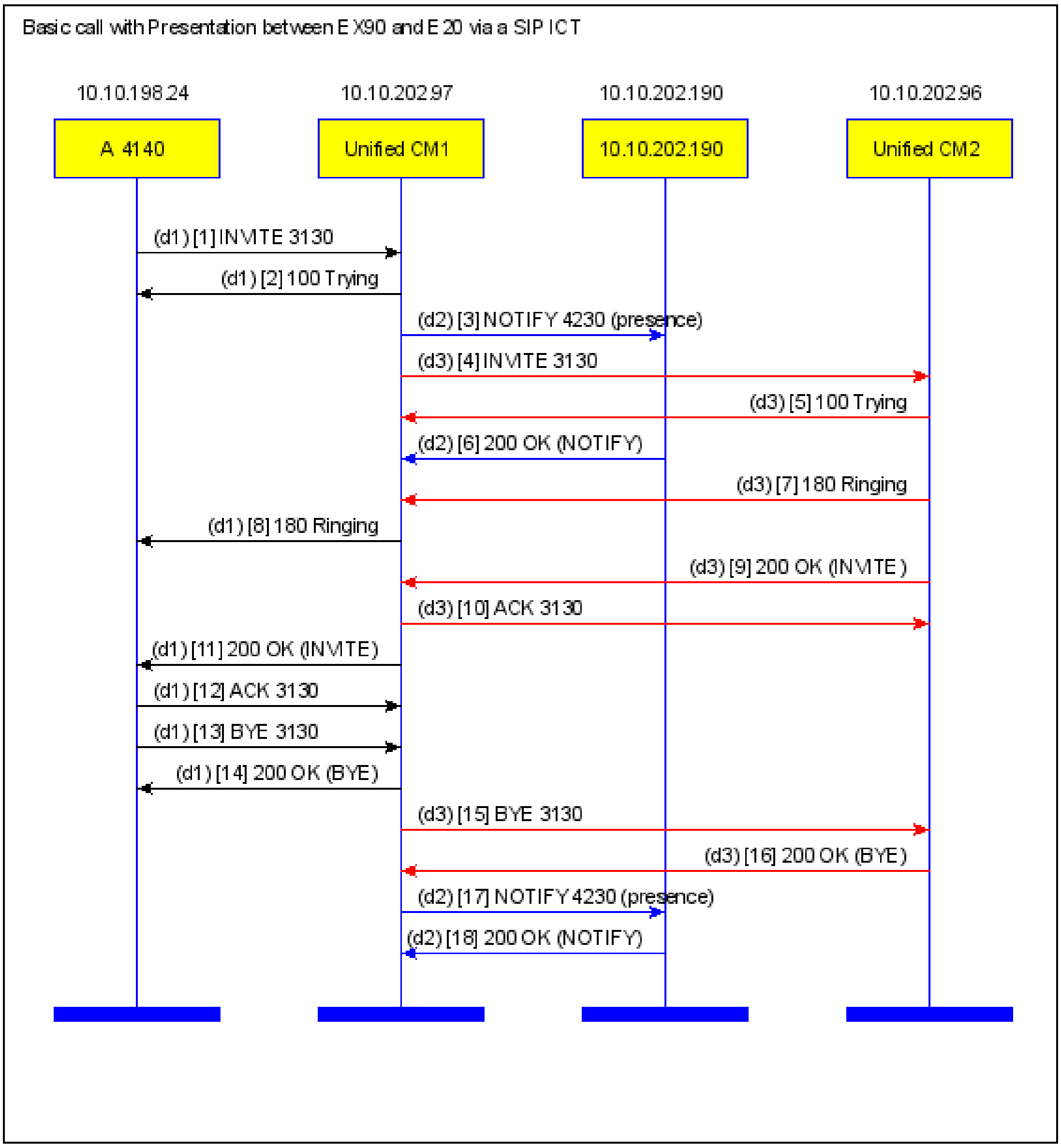
Phone = A, Line = 4140, IP = 10.10.198.24, Model = Cisco Telepresence EX90
Phone = B, Line = 3130, IP = 10.10.202.27, Model = Cisco E20

A has laptop connected to it via HDMI.
SIP Trunk between Unified CM1 and Unified CM2 has a route pattern 3XXX.

Scenario:

Phone A DN 4140 calls Phone B DN 3130
Phone B DN 3130 answers the call
Phone A DN 4140 presses Present and chooses HDMI as input source for presentation
Main video and Presentation video are available on Phone B DN 3130
Phone A DN 4140 ends Presentation
Phone A DN 4140 goes onhook

End of Scenario


Basic call with Presentation between EX90 and E20 via a SIP ICT
10.10.198.24
10.10.202.97
10.10.202.190
10.10.202.96
A 4140
Unified CM1
10.10.202.190
Unified CM2
(d1) [1] INVITE 3130
(d1) [2] 100 Trying
(d2) [3] NOTIFY 4230 (presence)
(d3) [4] INVITE 3130
(d3) [5] 100 Trying
(d2) [6] 200 OK (NOTIFY)
(d3) [7] 180 Ringing
(d1) [8] 180 Ringing
(d3) [9] 200 OK (INVITE)
(d3) [10] ACK 3130
(d1) [11] 200 OK (INVITE)
(d1) [12] ACK 3130
(d1) [13] BYE 3130
(d1) [14] 200 OK (BYE)
(d3) [15] BYE 3130
(d3) [16] 200 OK (BYE)
(d2) [17] NOTIFY 4230 (presence)
(d2) [18] 200 OK (NOTIFY)

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```
[diagram]   Call-ID:[prev][next]
[1] INVITE sip:3130@10.10.202.97 SIP/2.0
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK70ad83e1bdbbf75e2056c4162067d774.1;rport
Call-ID: 93b464f5c09c2cfb@10.10.198.24
CSeq: 100 INVITE
Contact: <sip:4140@10.10.198.24:44773;transport=tcp>
From: <sip:4140@10.10.202.97>;tag=b2fb5a11f4081777
To: <sip:3130@10.10.202.97>
Max-Forwards: 70
Route: <sip:10.10.202.97;lr>
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-service-
control,norefersub,extended-refer
Session-Expires: 1800
Remote-Party-ID: <sip:4140@10.10.202.97>;privacy=off;id-type=subscriber;screen=yes;party=calling
Content-Type: application/sdp
Content-Length: 1575

v=0
o=tandberg 5 1 IN IP4 10.10.198.24
s=-
c=IN IP4 10.10.198.24
b=AS:6000
t=0 0
m=audio 16408 RTP/AVP 100 102 103 104 105 106 9 8 0 101
b=TIAS:128000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 profile-level-id=25;object=23;bitrate=128000
a=rtpmap:102 MP4A-LATM/90000
a=fmtp:102 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:103 MP4A-LATM/90000
a=fmtp:103 profile-level-id=24;object=23;bitrate=56000
a=rtpmap:104 MP4A-LATM/90000
a=fmtp:104 profile-level-id=24;object=23;bitrate=48000
a=rtpmap:105 G7221/16000
a=fmtp:105 bitrate=32000
a=rtpmap:106 G7221/16000
a=fmtp:106 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
m=video 16410 RTP/AVP 97 98 99 34 31
b=TIAS:6000000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=428016;max-br=5000;max-mbps=108000;max-fs=3600;max-smbps=108000;max-fps=6000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=428016;max-br=5000;max-mbps=108000;max-fs=3600;max-smbps=108000;packetization-
mode=1;max-fps=6000
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1280,768,3;custom=1280,720,3;custom=1024,768,1;custom=1024,576,2;custom=800,600,1;cif4=1;custom=720,4
80,1;custom=640,480,1;custom=512,288,1;cif=1;custom=352,240,1;qcif=1;maxbr=20000
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=1;cif=1;qcif=1;maxbr=20000
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=20000
a=rtcp-fb:* nack pli
a=sendrecv
a=content:main
a=label:11
a=answer:full
m=application 16414 RTP/AVP 107
a=rtpmap:107 H224/4800
a=sendrecv

[diagram]   Call-ID:[prev][next]
[2] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK70ad83e1bdbbf75e2056c4162067d774.1;rport
From: <sip:4140@10.10.202.97>;tag=b2fb5a11f4081777
To: <sip:3130@10.10.202.97>
Date: Wed, 22 Jun 2011 15:12:28 GMT
Call-ID: 93b464f5c09c2cfb@10.10.198.24
CSeq: 100 INVITE
Allow-Events: presence
Content-Length: 0
```

```
[diagram]   Call-ID:[prev][next]
[3] NOTIFY sip:4230@10.10.202.190:38139;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK11654b177f7
From: <sip:4140@10.10.202.97>;tag=1669962485
To: <sip:4230@10.10.202.97>;tag=0653a9395eb8cc0d
Call-ID: 14004e146e462686@10.10.202.190
CSeq: 130 NOTIFY
Max-Forwards: 70
Date: Wed, 22 Jun 2011 15:12:28 GMT
User-Agent: Cisco-CUCM8.6
Event: presence
Subscription-State: active;expires=2919
Contact: <sip:4140@10.10.202.97:5060;transport=tcp>
Content-Type: application/pidf+xml
Content-Length: 826

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<presence xmlns="urn:ietf:params:xml:ns:pidf" entity="sip:4140@10.10.202.97"
xmlns:e="urn:ietf:params:xml:ns:pidf:status:rpid" xmlns:dm="urn:ietf:params:xml:ns:pidf:data-model"
xmlns:ce="urn:cisco:params:xml:ns:pidf:rpid" xmlns:sc="urn:ietf:params:xml:ns:pidf:servcaps">
  <dm:person>
    <status>
    <basic>open</basic>
    </status>
    <e:activities>
      <e:on-the-phone/>
    </e:activities>
  </dm:person>
  <tuple id="cmp-1-351">
    <status>
      <basic>open</basic>
      <e:activities>
        <e:on-the-phone/>
      </e:activities>
    </status>
    <sc:servcaps>
      <sc:audio>true</sc:audio>
    </sc:servcaps>
    <contact priority="0.8">sip:4140@10.10.202.97:5060</contact>
    <timestamp>2011-06-22T15:12:28Z</timestamp>
  </tuple>
</presence>

[diagram]   Call-ID:[prev][next]
[4] INVITE sip:3130@10.10.202.96:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK117671fe8ee
From: <sip:4140@vcs.domain>;tag=9773~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037384
To: <sip:3130@10.10.202.96>
Date: Wed, 22 Jun 2011 15:12:28 GMT
Call-ID: 9a65b80-e021065c-1f-61ca12ac@10.10.202.97
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Cisco-Guid: 0161897344-0000065536-0000000005-1640633004
Session-Expires:  1800
P-Asserted-Identity: <sip:4140@vcs.domain>
Remote-Party-ID: <sip:4140@vcs.domain>;party=calling;screen=yes;privacy=off
Contact: <sip:4140@10.10.202.97:5060;transport=tcp>;video;audio
Max-Forwards: 69
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[5] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK117671fe8ee
From: <sip:4140@vcs.domain>;tag=9773~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037384
To: <sip:3130@10.10.202.96>
Date: Wed, 22 Jun 2011 15:12:28 GMT
Call-ID: 9a65b80-e021065c-1f-61ca12ac@10.10.202.97
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[6] SIP/2.0 200 OK
```

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```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK11654b177f7;received=10.10.202.97
Call-ID: 14004e146e462686@10.10.202.190
CSeq: 130 NOTIFY
From: <sip:4140@10.10.202.97>;tag=1669962485
To: <sip:4230@10.10.202.97>;tag=0653a9395eb8cc0d
Server: TANDBERG/257 (TE4.1.0.253886Alpha4)
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[7] SIP/2.0 180 Ringing

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK117671fe8ee
From: <sip:4140@vcs.domain>;tag=9773~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037384
To: <sip:3130@10.10.202.96>;tag=15822~b43ea526-a700-4a22-8605-9a5b66389cb6-17037400
Date: Wed, 22 Jun 2011 15:12:28 GMT
Call-ID: 9a65b80-e021065c-1f-61ca12ac@10.10.202.97
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:3130@10.10.202.96>
Remote-Party-ID: <sip:3130@10.10.202.96>;party=called;screen=yes;privacy=off
Contact: <sip:3130@10.10.202.96:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[8] SIP/2.0 180 Ringing

```
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK70ad83e1bdbbf75e2056c4162067d774.1;rport
From: <sip:4140@10.10.202.97>;tag=b2fb5a11f4081777
To: <sip:3130@10.10.202.97>;tag=9772~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037383
Date: Wed, 22 Jun 2011 15:12:28 GMT
Call-ID: 93b464f5c09c2cfb@10.10.198.24
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; ui-state= ringout;
gci= 1-162005; call-instance= 1
Send-Info: conference, x-cisco-conference
Remote-Party-ID: <sip:3130@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:3130@10.10.202.97:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[9] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK117671fe8ee
From: <sip:4140@vcs.domain>;tag=9773~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037384
To: <sip:3130@10.10.202.96>;tag=15822~b43ea526-a700-4a22-8605-9a5b66389cb6-17037400
Date: Wed, 22 Jun 2011 15:12:28 GMT
Call-ID: 9a65b80-e021065c-1f-61ca12ac@10.10.202.97
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Session-Expires:  1800;refresher=uas
Require:  timer
P-Asserted-Identity: <sip:3130@10.10.202.96>
Remote-Party-ID: <sip:3130@10.10.202.96>;party=called;screen=yes;privacy=off
Contact: <sip:3130@10.10.202.96:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 1988

v=0
o=CiscoSystemsCCM-SIP 15822 1 IN IP4 10.10.202.96
s=SIP Call
b=TIAS:1152000
b=AS:1152
t=0 0
m=audio 16434 RTP/AVP 100 9 102 103 0 8 18 101
c=IN IP4 10.10.202.27
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 bitrate=64000;profile-level-id=24;object=23
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:102 G7221/16000
a=fmtp:102 bitrate=32000
a=rtpmap:103 G7221/16000
a=fmtp:103 bitrate=24000
```

```
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=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 16436 RTP/AVP 97 98 99 34 31
c=IN IP4 10.10.202.27
b=TIAS:1088000
a=label:11
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800D;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-smbps=40500;max-
fps=3000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-
smbps=40500;max-fps=3000
a=rtpmap:99 H263-1998/90000
a=fmtp:99 QCIF=1;CIF=1;CIF4=2;MAXBR=10880;CUSTOM=352,240,1
a=rtpmap:34 H263/90000
a=fmtp:34 QCIF=1;CIF=1;CIF4=2;MAXBR=10880
a=rtpmap:31 H261/90000
a=fmtp:31 CIF=1;QCIF=1;MAXBR=10880
a=content:main
a=rtcp-fb:* nack pli
m=video 16438 RTP/AVP 97 98 99 34 31
c=IN IP4 10.10.202.27
b=TIAS:1088000
a=label:12
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800D;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-smbps=40500;max-
fps=3000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-
smbps=40500;max-fps=3000
a=rtpmap:99 H263-1998/90000
a=fmtp:99 QCIF=1;CIF=1;CIF4=2;MAXBR=10880;CUSTOM=352,240,1
a=rtpmap:34 H263/90000
a=fmtp:34 QCIF=1;CIF=1;CIF4=2;MAXBR=10880
a=rtpmap:31 H261/90000
a=fmtp:31 CIF=1;QCIF=1;MAXBR=10880
a=content:slides
a=rtcp-fb:* nack pli
m=application 5070 UDP/BFCP *
c=IN IP4 10.10.202.27
a=floorctrl:c-s
a=floorid:2 mstrm:12
a=confid:1
a=userid:10
```

[diagram] Call-ID:[prev][next]

[10] ACK sip:3130@10.10.202.96:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK118599d9b16
From: <sip:4140@vcs.domain>;tag=9773~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037384
To: <sip:3130@10.10.202.96>;tag=15822~b43ea526-a700-4a22-8605-9a5b66389cb6-17037400
Date: Wed, 22 Jun 2011 15:12:28 GMT
Call-ID: 9a65b80-e021065c-1f-61ca12ac@10.10.202.97
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 1340

v=0
o=CiscoSystemsCCM-SIP 9773 1 IN IP4 10.10.202.97
s=SIP Call
b=TIAS:6000000
b=AS:6000
t=0 0
m=audio 16408 RTP/AVP 102 101
c=IN IP4 10.10.198.24
b=TIAS:64000
a=rtpmap:102 MP4A-LATM/90000
a=fmtp:102 bitrate=64000;profile-level-id=24;object=23
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 16410 RTP/AVP 98
c=IN IP4 10.10.198.24
```

```
b=TIAS:5936000
a=label:11
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-smbps=108000;max-fps=6000
a=content:main
a=rtcp-fb:* nack pli
m=video 0 RTP/AVP 97 98 99 34 31
c=IN IP4 10.10.202.27
b=TIAS:1088000
a=label:12
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800D;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-smbps=40500;max-
fps=3000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-
smbps=40500;max-fps=3000
a=rtpmap:99 H263-1998/90000
a=fmtp:99 QCIF=1;CIF=1;CIF4=2;MAXBR=10880;CUSTOM=352,240,1
a=rtpmap:34 H263/90000
a=fmtp:34 QCIF=1;CIF=1;CIF4=2;MAXBR=10880
a=rtpmap:31 H261/90000
a=fmtp:31 CIF=1;QCIF=1;MAXBR=10880
a=content:slides
a=rtcp-fb:* nack pli
m=application 0 UDP/BFCP *
c=IN IP4 10.10.202.27
a=floorctrl:c-s
a=floorid:2 mstrm:12
a=confid:1
a=userid:10
```

[diagram]   Call-ID:[prev][next]

[11] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK70ad83e1bdbbf75e2056c4162067d774.1;rport
From: <sip:4140@10.10.202.97>;tag=b2fb5a11f4081777
To: <sip:3130@10.10.202.97>;tag=9772~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037383
Date: Wed, 22 Jun 2011 15:12:28 GMT
Call-ID: 93b464f5c09c2cfb@10.10.198.24
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; gci= 1-162005;
call-instance= 1
Send-Info: conference, x-cisco-conference
Session-Expires:  1800;refresher=uas
Require:  timer
Remote-Party-ID: <sip:3130@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:3130@10.10.202.97:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 653

v=0
o=CiscoSystemsCCM-SIP 9772 1 IN IP4 10.10.202.97
s=SIP Call
b=AS:1152
t=0 0
m=audio 16434 RTP/AVP 100 101
c=IN IP4 10.10.202.27
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 bitrate=64000;profile-level-id=24;object=23
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 16436 RTP/AVP 98
c=IN IP4 10.10.202.27
b=TIAS:1088000
a=label:11
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-
smbps=40500;max-fps=3000
a=content:main
a=rtcp-fb:* nack pli
m=application 0 RTP/AVP 107
c=IN IP4 10.10.198.24
a=rtpmap:107 H224/0
```

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```
[diagram]   Call-ID:[prev][next]
[12] ACK sip:3130@10.10.202.97:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK78b1ada6c0e883022080870d2ff38a30.1;rport
Call-ID: 93b464f5c09c2cfb@10.10.198.24
CSeq: 100 ACK
From: <sip:4140@10.10.202.97>;tag=b2fb5a11f4081777
To: <sip:3130@10.10.202.97>;tag=9772~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037383
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-service-
control,norefersub,extended-refer
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[13] BYE sip:3130@10.10.202.97:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK6e9a4a1b32066e8c295390c455ef8ab4.1;rport
Call-ID: 93b464f5c09c2cfb@10.10.198.24
CSeq: 101 BYE
Contact: <sip:4140@10.10.198.24:44773;transport=tcp>;+sip.instance="<urn:uuid:00000000-0000-0000-0000-
0050600491d7>";+u.sip!model.ccm.cisco.com="584";audio=TRUE;video=TRUE;mobility="fixed";duplex="full";descrip
tion="TANDBERG-SIP"
From: <sip:4140@10.10.202.97>;tag=b2fb5a11f4081777
To: <sip:3130@10.10.202.97>;tag=9772~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037383
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-service-
control,norefersub,extended-refer
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[14] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK6e9a4a1b32066e8c295390c455ef8ab4.1;rport
From: <sip:4140@10.10.202.97>;tag=b2fb5a11f4081777
To: <sip:3130@10.10.202.97>;tag=9772~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037383
Date: Wed, 22 Jun 2011 15:12:55 GMT
Call-ID: 93b464f5c09c2cfb@10.10.198.24
CSeq: 101 BYE
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[15] BYE sip:3130@10.10.202.96:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK119923df1a
From: <sip:4140@vcs.domain>;tag=9773~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037384
To: <sip:3130@10.10.202.96>;tag=15822~b43ea526-a700-4a22-8605-9a5b66389cb6-17037400
Date: Wed, 22 Jun 2011 15:12:28 GMT
Call-ID: 9a65b80-e021065c-1f-61ca12ac@10.10.202.97
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Max-Forwards: 70
P-Asserted-Identity: <sip:4140@vcs.domain>
CSeq: 102 BYE
Reason: Q.850;cause=16
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[16] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK119923df1a
From: <sip:4140@vcs.domain>;tag=9773~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037384
To: <sip:3130@10.10.202.96>;tag=15822~b43ea526-a700-4a22-8605-9a5b66389cb6-17037400
Date: Wed, 22 Jun 2011 15:12:55 GMT
Call-ID: 9a65b80-e021065c-1f-61ca12ac@10.10.202.97
CSeq: 102 BYE
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[17] NOTIFY sip:4230@10.10.202.190:38139;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK11a1b771ec0
From: <sip:4140@10.10.202.97>;tag=1669962485
To: <sip:4230@10.10.202.97>;tag=0653a9395eb8cc0d
Call-ID: 14004e146e462686@10.10.202.190
CSeq: 131 NOTIFY
Max-Forwards: 70
Date: Wed, 22 Jun 2011 15:12:55 GMT
User-Agent: Cisco-CUCM8.6
Event: presence
Subscription-State: active;expires=2892
Contact: <sip:4140@10.10.202.97:5060;transport=tcp>
Content-Type: application/pidf+xml
Content-Length: 733
```

```
<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<presence xmlns="urn:ietf:params:xml:ns:pidf" entity="sip:4140@10.10.202.97"
xmlns:e="urn:ietf:params:xml:ns:pidf:status:rpid" xmlns:dm="urn:ietf:params:xml:ns:pidf:data-model"
xmlns:ce="urn:cisco:params:xml:ns:pidf:rpid" xmlns:sc="urn:ietf:params:xml:ns:pidf:servcaps">
  <dm:person>
    <status>
    <basic>open</basic>
    </status>
    <e:activities>
    </e:activities>
  </dm:person>
  <tuple id="cmp-1-351">
    <status>
      <basic>open</basic>
    </status>
    <sc:servcaps>
      <sc:audio>true</sc:audio>
    </sc:servcaps>
    <contact priority="0.8">sip:4140@10.10.202.97:5060</contact>
    <timestamp>2011-06-22T15:12:55Z</timestamp>
  </tuple>
</presence>
```


[diagram]   Call-ID:[prev][next]

```
[18] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK11a1b771ec0;received=10.10.202.97
Call-ID: 14004e146e462686@10.10.202.190
CSeq: 131 NOTIFY
From: <sip:4140@10.10.202.97>;tag=1669962485
To: <sip:4230@10.10.202.97>;tag=0653a9395eb8cc0d
Server: TANDBERG/257 (TE4.1.0.253886Alpha4)
Content-Length: 0
```

2.3 Basic audio call between EX90 and 7970 SIP via a SIP ICT

Title: Basic audio call between EX90 and 7970 SIP via a SIP ICT

Description:

The following call flow illustrates the SIP messaging that takes place between two Cisco Unified CMs via an inter cluster SIP trunk for an audio call between EX90 and 7970 SIP.

Cisco Unified CM1 sent out the initial INVITE.

Configuration:

Node = Unified CM1, IP = 10.10.202.97
Node = Unified CM2, IP = 10.10.202.96

Phone = A, Line = 4140, IP = 10.10.198.24, Model = Cisco Telepresence EX90
Phone = B, Line = 3150, IP = 10.10.202.219, Model = SIP 7970

A has laptop connected to it via HDMI.
SIP Trunk between Unified CM1 and Unified CM2 has a route pattern 3XXX.

Scenario:

Phone A DN 4140 calls Phone B DN 3150
Phone B DN 3150 answers the call

End of Scenario

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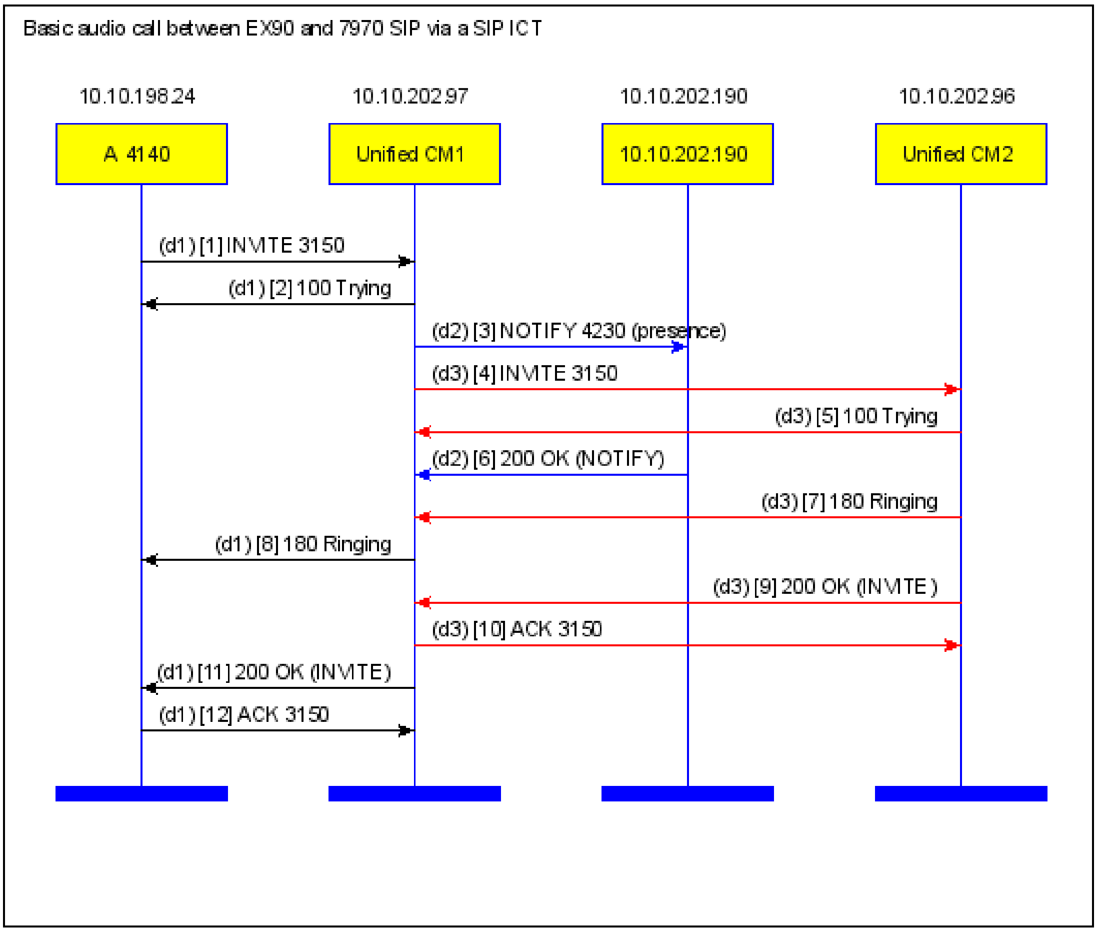


```
[diagram]   Call-ID:[prev][next]
[1] INVITE sip:3150@10.10.202.97 SIP/2.0
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK781cca08c4261d5835888b42acc74b6d.1;rport
Call-ID: 338175e0103c3f3d@10.10.198.24
CSeq: 100 INVITE
Contact: <sip:4140@10.10.198.24:44773;transport=tcp>
From: <sip:4140@10.10.202.97>;tag=a1390c2edced1ee8
To: <sip:3150@10.10.202.97>
Max-Forwards: 70
Route: <sip:10.10.202.97;lr>
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-service-
control,norefersub,extended-refer
Session-Expires: 1800
Remote-Party-ID: <sip:4140@10.10.202.97>;privacy=off;id-type=subscriber;screen=yes;party=calling
Content-Type: application/sdp
Content-Length: 1576

v=0
o=tandberg 10 1 IN IP4 10.10.198.24
s=-
c=IN IP4 10.10.198.24
b=AS:6000
t=0 0
m=audio 16448 RTP/AVP 100 102 103 104 105 106 9 8 0 101
b=TIAS:128000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 profile-level-id=25;object=23;bitrate=128000
a=rtpmap:102 MP4A-LATM/90000
a=fmtp:102 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:103 MP4A-LATM/90000
a=fmtp:103 profile-level-id=24;object=23;bitrate=56000
a=rtpmap:104 MP4A-LATM/90000
a=fmtp:104 profile-level-id=24;object=23;bitrate=48000
a=rtpmap:105 G7221/16000
a=fmtp:105 bitrate=32000
a=rtpmap:106 G7221/16000
a=fmtp:106 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
m=video 16450 RTP/AVP 97 98 99 34 31
b=TIAS:6000000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=428016;max-br=5000;max-mbps=108000;max-fs=3600;max-smbps=108000;max-fps=6000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=428016;max-br=5000;max-mbps=108000;max-fs=3600;max-smbps=108000;packetization-
mode=1;max-fps=6000
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1280,768,3;custom=1280,720,3;custom=1024,768,1;custom=1024,576,2;custom=800,600,1;cif4=1;custom=720,4
80,1;custom=640,480,1;custom=512,288,1;cif=1;custom=352,240,1;qcif=1;maxbr=20000
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=1;cif=1;qcif=1;maxbr=20000
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=20000
a=rtcp-fb:* nack pli
a=sendrecv
a=content:main
a=label:11
a=answer:full
m=application 16454 RTP/AVP 107
a=rtpmap:107 H224/4800
a=sendrecv

[diagram]   Call-ID:[prev][next]
[2] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK781cca08c4261d5835888b42acc74b6d.1;rport
From: <sip:4140@10.10.202.97>;tag=a1390c2edced1ee8
To: <sip:3150@10.10.202.97>
Date: Wed, 22 Jun 2011 15:29:58 GMT
Call-ID: 338175e0103c3f3d@10.10.198.24
CSeq: 100 INVITE
Allow-Events: presence
Content-Length: 0
```

```
[diagram]   Call-ID:[prev][next]
[3] NOTIFY sip:4230@10.10.202.190:38139;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK1306795a3ca
From: <sip:4140@10.10.202.97>;tag=1669962485
To: <sip:4230@10.10.202.97>;tag=0653a9395eb8cc0d
Call-ID: 14004e146e462686@10.10.202.190
CSeq: 140 NOTIFY
Max-Forwards: 70
Date: Wed, 22 Jun 2011 15:29:58 GMT
User-Agent: Cisco-CUCM8.6
Event: presence
Subscription-State: active;expires=1869
Contact: <sip:4140@10.10.202.97:5060;transport=tcp>
Content-Type: application/pidf+xml
Content-Length: 826

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<presence xmlns="urn:ietf:params:xml:ns:pidf" entity="sip:4140@10.10.202.97"
xmlns:e="urn:ietf:params:xml:ns:pidf:status:rpid" xmlns:dm="urn:ietf:params:xml:ns:pidf:data-model"
xmlns:ce="urn:cisco:params:xml:ns:pidf:rpid" xmlns:sc="urn:ietf:params:xml:ns:pidf:servcaps">
  <dm:person>
    <status>
    <basic>open</basic>
    </status>
    <e:activities>
      <e:on-the-phone/>
    </e:activities>
  </dm:person>
  <tuple id="cmp-1-351">
    <status>
      <basic>open</basic>
      <e:activities>
        <e:on-the-phone/>
      </e:activities>
    </status>
    <sc:servcaps>
      <sc:audio>true</sc:audio>
    </sc:servcaps>
    <contact priority="0.8">sip:4140@10.10.202.97:5060</contact>
    <timestamp>2011-06-22T15:29:58Z</timestamp>
  </tuple>
</presence>

[diagram]   Call-ID:[prev][next]
[4] INVITE sip:3150@10.10.202.96:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK1314db2d181
From: <sip:4140@vcs.domain>;tag=9943~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037394
To: <sip:3150@10.10.202.96>
Date: Wed, 22 Jun 2011 15:29:58 GMT
Call-ID: 7b7fa480-e0210a76-24-61ca12ac@10.10.202.97
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Cisco-Guid: 2071962752-0000065536-0000000010-1640633004
Session-Expires:  1800
P-Asserted-Identity: <sip:4140@vcs.domain>
Remote-Party-ID: <sip:4140@vcs.domain>;party=calling;screen=yes;privacy=off
Contact: <sip:4140@10.10.202.97:5060;transport=tcp>;video;audio
Max-Forwards: 69
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[5] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK1314db2d181
From: <sip:4140@vcs.domain>;tag=9943~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037394
To: <sip:3150@10.10.202.96>
Date: Wed, 22 Jun 2011 15:29:58 GMT
Call-ID: 7b7fa480-e0210a76-24-61ca12ac@10.10.202.97
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[6] SIP/2.0 200 OK
```

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK1306795a3ca;received=10.10.202.97
Call-ID: 14004e146e462686@10.10.202.190
CSeq: 140 NOTIFY
From: <sip:4140@10.10.202.97>;tag=1669962485
To: <sip:4230@10.10.202.97>;tag=0653a9395eb8cc0d
Server: TANDBERG/257 (TE4.1.0.253886Alpha4)
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[7] SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK1314db2d181
From: <sip:4140@vcs.domain>;tag=9943~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037394
To: <sip:3150@10.10.202.96>;tag=16075~b43ea526-a700-4a22-8605-9a5b66389cb6-17037411
Date: Wed, 22 Jun 2011 15:29:58 GMT
Call-ID: 7b7fa480-e0210a76-24-61ca12ac@10.10.202.97
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:3150@10.10.202.96>
Remote-Party-ID: <sip:3150@10.10.202.96>;party=called;screen=yes;privacy=off
Contact: <sip:3150@10.10.202.96:5060;transport=tcp>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[8] SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK781cca08c4261d5835888b42acc74b6d.1;rport
From: <sip:4140@10.10.202.97>;tag=a1390c2edced1ee8
To: <sip:3150@10.10.202.97>;tag=9942~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037393
Date: Wed, 22 Jun 2011 15:29:58 GMT
Call-ID: 338175e0103c3f3d@10.10.198.24
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; ui-state= ringout;
gci= 1-162010; call-instance= 1
Send-Info: conference, x-cisco-conference
Remote-Party-ID: <sip:3150@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:3150@10.10.202.97:5060;transport=tcp>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[9] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK1314db2d181
From: <sip:4140@vcs.domain>;tag=9943~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037394
To: <sip:3150@10.10.202.96>;tag=16075~b43ea526-a700-4a22-8605-9a5b66389cb6-17037411
Date: Wed, 22 Jun 2011 15:29:58 GMT
Call-ID: 7b7fa480-e0210a76-24-61ca12ac@10.10.202.97
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence, kpml
Supported: replaces
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Session-Expires:  1800;refresher=uas
Require:  timer
P-Asserted-Identity: <sip:3150@10.10.202.96>
Remote-Party-ID: <sip:3150@10.10.202.96>;party=called;screen=yes;privacy=off
Contact: <sip:3150@10.10.202.96:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 372

v=0
o=CiscoSystemsCCM-SIP 16075 1 IN IP4 10.10.202.96
s=SIP Call
c=IN IP4 10.10.202.219
b=TIAS:256000
b=AS:256
t=0 0
m=audio 20832 RTP/AVP 9 0 8 18 101
a=rtpmap:9 G722/8000
a=ptime:20
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=fmtp:18 annexb=no
```

```
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

[diagram]   Call-ID:[prev][next]
[10] ACK sip:3150@10.10.202.96:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK1325cf2f9
From: <sip:4140@vcs.domain>;tag=9943~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037394
To: <sip:3150@10.10.202.96>;tag=16075~b43ea526-a700-4a22-8605-9a5b66389cb6-17037411
Date: Wed, 22 Jun 2011 15:29:58 GMT
Call-ID: 7b7fa480-e0210a76-24-61ca12ac@10.10.202.97
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 251

v=0
o=CiscoSystemsCCM-SIP 9943 1 IN IP4 10.10.202.97
s=SIP Call
c=IN IP4 10.10.198.24
b=TIAS:64000
b=AS:64
t=0 0
m=audio 16448 RTP/AVP 9 101
b=TIAS:64000
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

[diagram]   Call-ID:[prev][next]
[11] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK781cca08c4261d5835888b42acc74b6d.1;rport
From: <sip:4140@10.10.202.97>;tag=a1390c2edced1ee8
To: <sip:3150@10.10.202.97>;tag=9942~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037393
Date: Wed, 22 Jun 2011 15:29:58 GMT
Call-ID: 338175e0103c3f3d@10.10.198.24
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; gci= 1-162010;
call-instance= 1
Send-Info: conference, x-cisco-conference
Session-Expires:  1800;refresher=uas
Require:  timer
Remote-Party-ID: <sip:3150@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:3150@10.10.202.97:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 919

v=0
o=CiscoSystemsCCM-SIP 9942 1 IN IP4 10.10.202.97
s=SIP Call
b=AS:64
t=0 0
m=audio 20832 RTP/AVP 9 101
c=IN IP4 10.10.202.219
b=TIAS:64000
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 0 RTP/AVP 97 98 99 34 31
c=IN IP4 10.10.198.24
b=TIAS:6000000
a=label:11
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=428016;max-mbps=108000;max-fs=3600;max-cpb=200;max-br=5000;max-smbps=108000;max-
fps=6000
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-smbps=108000;max-fps=6000
a=rtpmap:99 H263-1998/90000
a=fmtp:99 QCIF=1;CIF=1;CIF4=1;CUSTOM=352,240,1
a=rtpmap:34 H263/90000
a=fmtp:34 QCIF=1;CIF=1;CIF4=1
a=rtpmap:31 H261/90000
a=fmtp:31 CIF=1;QCIF=1
a=content:main
```

```
a=rtcp-fb:* nack pli
m=application 0 RTP/AVP 107
c=IN IP4 10.10.198.24
a=rtpmap:107 H224/0
```

[diagram] Call-ID:**[prev]**[next]

[12] ACK sip:3150@10.10.202.97:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bKc7cf5e9743b81d615d831c01ab9f39b6.1;rport
Call-ID: 338175e0103c3f3d@10.10.198.24
CSeq: 100 ACK
From: <sip:4140@10.10.202.97>;tag=a1390c2edced1ee8
To: <sip:3150@10.10.202.97>;tag=9942~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037393
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-service-
control,norefersub,extended-refer
Content-Length: 0
```

2.4 Basic video call between EX90 and 7985 via a SIP ICT

Title: Basic video call between EX90 and 7985 via a SIP ICT

Description:
The following call flow illustrates the SIP messaging that takes place between two Cisco Unified CMs via an inter cluster SIP trunk for a video call between EX90 and 7985.

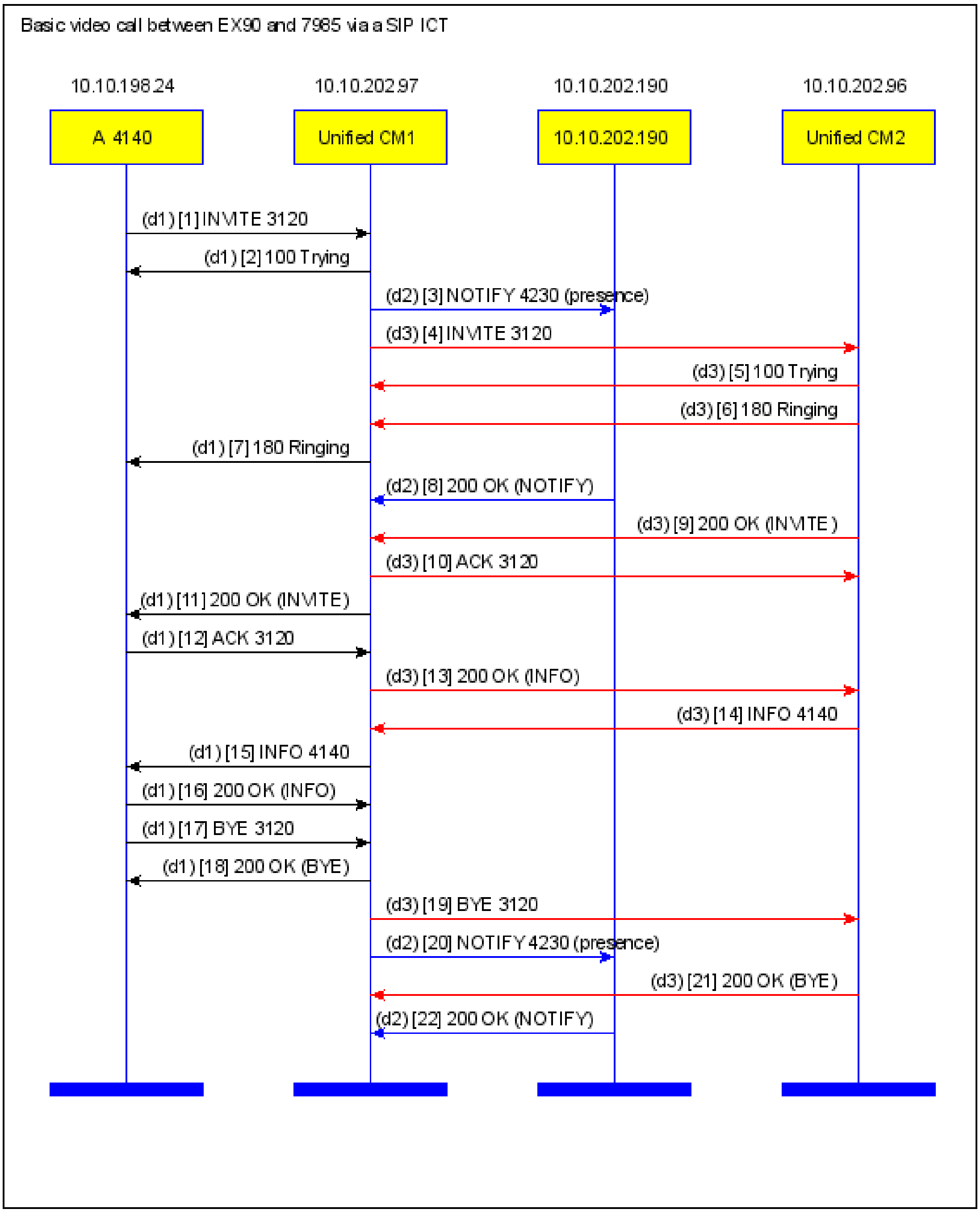
Cisco Unified CM1 sent out the initial INVITE.

Configuration:
Node = Unified CM1, IP = 10.10.202.97
Node = Unified CM2, IP = 10.10.202.96

Phone = A, Line = 4140, IP = 10.10.198.24, Model = Cisco Telepresence EX90
Phone = B, Line = 3120, IP = 10.10.202.242, Model = SCCP 7985

A has laptop connected to it via HDMI.
SIP Trunk between Unified CM1 and Unified CM2 has a route pattern 3XXX.

Scenario:
Phone A DN 4140 calls Phone B DN 3120
Phone B DN 3120 answers the call
Phone A DN 4140 goes onhook
End of Scenario



Basic video call between EX90 and 7985 via a SIP ICT
10.10.198.24
10.10.202.97
10.10.202.190
10.10.202.96
A 4140
Unified CM1
10.10.202.190
Unified CM2
(d1) [1] INVITE 3120
(d1) [2] 100 Trying
(d2) [3] NOTIFY 4230 (presence)
(d3) [4] INVITE 3120
(d3) [5] 100 Trying
(d3) [6] 180 Ringing
(d1) [7] 180 Ringing
(d2) [8] 200 OK (NOTIFY)
(d3) [9] 200 OK (INVITE)
(d3) [10] ACK 3120
(d1) [11] 200 OK (INVITE)
(d1) [12] ACK 3120
(d3) [13] 200 OK (INFO)
(d3) [14] INFO 4140
(d1) [15] INFO 4140
(d1) [16] 200 OK (INFO)
(d1) [17] BYE 3120
(d1) [18] 200 OK (BYE)
(d3) [19] BYE 3120
(d2) [20] NOTIFY 4230 (presence)
(d3) [21] 200 OK (BYE)
(d2) [22] 200 OK (NOTIFY)

```
[diagram]   Call-ID:[prev][next]
[1] INVITE sip:3120@10.10.202.97 SIP/2.0
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK3bc8d2772686dedb89d125aa22472bf3.1;rport
Call-ID: 3232ea9eb98063c0@10.10.198.24
CSeq: 100 INVITE
Contact: <sip:4140@10.10.198.24:44773;transport=tcp>
From: <sip:4140@10.10.202.97>;tag=8dfa692fd5736da6
To: <sip:3120@10.10.202.97>
Max-Forwards: 70
Route: <sip:10.10.202.97;lr>
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-service-
control,norefersub,extended-refer
Session-Expires: 1800
Remote-Party-ID: <sip:4140@10.10.202.97>;privacy=off;id-type=subscriber;screen=yes;party=calling
Content-Type: application/sdp
Content-Length: 1575

v=0
o=tandberg 8 1 IN IP4 10.10.198.24
s=-
c=IN IP4 10.10.198.24
b=AS:6000
t=0 0
m=audio 16432 RTP/AVP 100 102 103 104 105 106 9 8 0 101
b=TIAS:128000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 profile-level-id=25;object=23;bitrate=128000
a=rtpmap:102 MP4A-LATM/90000
a=fmtp:102 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:103 MP4A-LATM/90000
a=fmtp:103 profile-level-id=24;object=23;bitrate=56000
a=rtpmap:104 MP4A-LATM/90000
a=fmtp:104 profile-level-id=24;object=23;bitrate=48000
a=rtpmap:105 G7221/16000
a=fmtp:105 bitrate=32000
a=rtpmap:106 G7221/16000
a=fmtp:106 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
m=video 16434 RTP/AVP 97 98 99 34 31
b=TIAS:6000000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=428016;max-br=5000;max-mbps=108000;max-fs=3600;max-smbps=108000;max-fps=6000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=428016;max-br=5000;max-mbps=108000;max-fs=3600;max-smbps=108000;packetization-
mode=1;max-fps=6000
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1280,768,3;custom=1280,720,3;custom=1024,768,1;custom=1024,576,2;custom=800,600,1;cif4=1;custom=720,4
80,1;custom=640,480,1;custom=512,288,1;cif=1;custom=352,240,1;qcif=1;maxbr=20000
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=1;cif=1;qcif=1;maxbr=20000
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=20000
a=rtcp-fb:* nack pli
a=sendrecv
a=content:main
a=label:11
a=answer:full
m=application 16438 RTP/AVP 107
a=rtpmap:107 H224/4800
a=sendrecv
```

```
[diagram]   Call-ID:[prev][next]
[2] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK3bc8d2772686dedb89d125aa22472bf3.1;rport
From: <sip:4140@10.10.202.97>;tag=8dfa692fd5736da6
To: <sip:3120@10.10.202.97>
Date: Wed, 22 Jun 2011 15:22:52 GMT
Call-ID: 3232ea9eb98063c0@10.10.198.24
CSeq: 100 INVITE
Allow-Events: presence
Content-Length: 0
```

```
[diagram]   Call-ID:[prev][next]
[3] NOTIFY sip:4230@10.10.202.190:38139;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12532ababe2
From: <sip:4140@10.10.202.97>;tag=1669962485
To: <sip:4230@10.10.202.97>;tag=0653a9395eb8cc0d
Call-ID: 14004e146e462686@10.10.202.190
CSeq: 136 NOTIFY
Max-Forwards: 70
Date: Wed, 22 Jun 2011 15:22:52 GMT
User-Agent: Cisco-CUCM8.6
Event: presence
Subscription-State: active;expires=2295
Contact: <sip:4140@10.10.202.97:5060;transport=tcp>
Content-Type: application/pidf+xml
Content-Length: 826
```


```
<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<presence xmlns="urn:ietf:params:xml:ns:pidf" entity="sip:4140@10.10.202.97"
xmlns:e="urn:ietf:params:xml:ns:pidf:status:rpid" xmlns:dm="urn:ietf:params:xml:ns:pidf:data-model"
xmlns:ce="urn:cisco:params:xml:ns:pidf:rpid" xmlns:sc="urn:ietf:params:xml:ns:pidf:servcaps">
  <dm:person>
    <status>
    <basic>open</basic>
    </status>
    <e:activities>
      <e:on-the-phone/>
    </e:activities>
  </dm:person>
  <tuple id="cmp-1-351">
    <status>
      <basic>open</basic>
      <e:activities>
        <e:on-the-phone/>
      </e:activities>
    </status>
    <sc:servcaps>
      <sc:audio>true</sc:audio>
    </sc:servcaps>
    <contact priority="0.8">sip:4140@10.10.202.97:5060</contact>
    <timestamp>2011-06-22T15:22:52Z</timestamp>
  </tuple>
</presence>
```


```
[diagram]   Call-ID:[prev][next]
[4] INVITE sip:3120@10.10.202.96:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK1267087983a
From: <sip:4140@vcs.domain>;tag=9874~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037390
To: <sip:3120@10.10.202.96>
Date: Wed, 22 Jun 2011 15:22:52 GMT
Call-ID: 7d953380-e02108cc-22-61ca12ac@10.10.202.97
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Cisco-Guid: 2106930048-0000065536-0000000008-1640633004
Session-Expires:  1800
P-Asserted-Identity: <sip:4140@vcs.domain>
Remote-Party-ID: <sip:4140@vcs.domain>;party=calling;screen=yes;privacy=off
Contact: <sip:4140@10.10.202.97:5060;transport=tcp>;video;audio
Max-Forwards: 69
Content-Length: 0
```

```
[diagram]   Call-ID:[prev][next]
[5] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK1267087983a
From: <sip:4140@vcs.domain>;tag=9874~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037390
To: <sip:3120@10.10.202.96>
Date: Wed, 22 Jun 2011 15:22:52 GMT
Call-ID: 7d953380-e02108cc-22-61ca12ac@10.10.202.97
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0
```

```
[diagram]   Call-ID:[prev][next]
[6] SIP/2.0 180 Ringing
```

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK1267087983a
From: <sip:4140@vcs.domain>;tag=9874~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037390
To: <sip:3120@10.10.202.96>;tag=15969~b43ea526-a700-4a22-8605-9a5b66389cb6-17037406
Date: Wed, 22 Jun 2011 15:22:52 GMT
Call-ID: 7d953380-e02108cc-22-61ca12ac@10.10.202.97
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:3120@10.10.202.96>
Remote-Party-ID: <sip:3120@10.10.202.96>;party=called;screen=yes;privacy=off
Contact: <sip:3120@10.10.202.96:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[7] SIP/2.0 180 Ringing

```
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK3bc8d2772686dedb89d125aa22472bf3.1;rport
From: <sip:4140@10.10.202.97>;tag=8dfa692fd5736da6
To: <sip:3120@10.10.202.97>;tag=9873~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037389
Date: Wed, 22 Jun 2011 15:22:52 GMT
Call-ID: 3232ea9eb98063c0@10.10.198.24
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; ui-state= ringout;
gci= 1-162008; call-instance= 1
Send-Info: conference, x-cisco-conference
Remote-Party-ID: <sip:3120@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:3120@10.10.202.97:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[8] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12532ababe2;received=10.10.202.97
Call-ID: 14004e146e462686@10.10.202.190
CSeq: 136 NOTIFY
From: <sip:4140@10.10.202.97>;tag=1669962485
To: <sip:4230@10.10.202.97>;tag=0653a9395eb8cc0d
Server: TANDBERG/257 (TE4.1.0.253886Alpha4)
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[9] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK1267087983a
From: <sip:4140@vcs.domain>;tag=9874~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037390
To: <sip:3120@10.10.202.96>;tag=15969~b43ea526-a700-4a22-8605-9a5b66389cb6-17037406
Date: Wed, 22 Jun 2011 15:22:52 GMT
Call-ID: 7d953380-e02108cc-22-61ca12ac@10.10.202.97
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence, kpml
Supported: replaces
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Session-Expires:  1800;refresher=uas
Require:  timer
P-Asserted-Identity: <sip:3120@10.10.202.96>
Remote-Party-ID: <sip:3120@10.10.202.96>;party=called;screen=yes;privacy=off
Contact: <sip:3120@10.10.202.96:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 697

v=0
o=CiscoSystemsCCM-SIP 15969 1 IN IP4 10.10.202.96
s=SIP Call
c=IN IP4 10.10.202.112
b=TIAS:960000
b=AS:960
t=0 0
m=audio 17856 RTP/AVP 9 0 8 18 101
a=rtpmap:9 G722/8000
a=ptime:20
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=rtpmap:101 telephone-event/8000
```

```
a=fmtp:101 0-15
m=video 16676 RTP/AVP 97 96 34 31
b=TIAS:704000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42000D;packetization-mode=0;level-asymmetry-allowed=1
a=rtpmap:96 H263-1998/90000
a=fmtp:96 QCIF=1;CIF=1;MAXBR=7040;CUSTOM=352,240,1
a=rtpmap:34 H263/90000
a=fmtp:34 MAXBR=7040
a=rtpmap:31 H261/90000
a=fmtp:31 CIF=1;QCIF=1;MAXBR=7040;D=1
```

[diagram] Call-ID:[prev][next]
[10] ACK sip:3120@10.10.202.96:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12710193174
From: <sip:4140@vcs.domain>;tag=9874~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037390
To: <sip:3120@10.10.202.96>;tag=15969~b43ea526-a700-4a22-8605-9a5b66389cb6-17037406
Date: Wed, 22 Jun 2011 15:22:52 GMT
Call-ID: 7d953380-e02108cc-22-61ca12ac@10.10.202.97
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 488

v=0
o=CiscoSystemsCCM-SIP 9874 1 IN IP4 10.10.202.97
s=SIP Call
c=IN IP4 10.10.198.24
b=TIAS:6000000
b=AS:6000
t=0 0
m=audio 16432 RTP/AVP 9 101
b=TIAS:64000
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 16434 RTP/AVP 97
b=TIAS:5936000
a=label:11
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=428016;max-mbps=108000;max-fs=3600;max-cpb=200;max-br=5000;max-smbps=108000;max-
fps=6000
a=content:main
a=rtcp-fb:* nack pli
```

[diagram] Call-ID:[prev][next]
[11] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK3bc8d2772686dedb89d125aa22472bf3.1;rport
From: <sip:4140@10.10.202.97>;tag=8dfa692fd5736da6
To: <sip:3120@10.10.202.97>;tag=9873~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037389
Date: Wed, 22 Jun 2011 15:22:52 GMT
Call-ID: 3232ea9eb98063c0@10.10.198.24
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; gci= 1-162008;
call-instance= 1
Send-Info: conference, x-cisco-conference
Session-Expires:  1800;refresher=uas
Require:  timer
Remote-Party-ID: <sip:3120@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:3120@10.10.202.97:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 485

v=0
o=CiscoSystemsCCM-SIP 9873 1 IN IP4 10.10.202.97
s=SIP Call
b=AS:960
t=0 0
m=audio 17856 RTP/AVP 9 101
c=IN IP4 10.10.202.112
b=TIAS:64000
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
m=video 16676 RTP/AVP 97
c=IN IP4 10.10.202.112
b=TIAS:896000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42000D;packetization-mode=0;level-asymmetry-allowed=1
m=application 0 RTP/AVP 107
c=IN IP4 10.10.198.24
a=rtpmap:107 H224/0
```

[diagram] Call-ID:[prev][next]
[12] ACK sip:3120@10.10.202.97:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bKfb17066b036d342a7ae7fe4e6bb80bb4.1;rport
Call-ID: 3232ea9eb98063c0@10.10.198.24
CSeq: 100 ACK
From: <sip:4140@10.10.202.97>;tag=8dfa692fd5736da6
To: <sip:3120@10.10.202.97>;tag=9873~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037389
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-service-
control,norefersub,extended-refer
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[13] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.202.96:5060;branch=z9hG4bK6b52c008c1f
From: <sip:3120@10.10.202.96>;tag=15969~b43ea526-a700-4a22-8605-9a5b66389cb6-17037406
To: <sip:4140@vcs.domain>;tag=9874~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037390
Date: Wed, 22 Jun 2011 15:22:54 GMT
Call-ID: 7d953380-e02108cc-22-61ca12ac@10.10.202.97
Server: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
CSeq: 101 INFO
Contact: <sip:4140@10.10.202.97:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[14] INFO sip:4140@10.10.202.97:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.202.96:5060;branch=z9hG4bK6b52c008c1f
From: <sip:3120@10.10.202.96>;tag=15969~b43ea526-a700-4a22-8605-9a5b66389cb6-17037406
To: <sip:4140@vcs.domain>;tag=9874~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037390
Date: Wed, 22 Jun 2011 15:22:53 GMT
Call-ID: 7d953380-e02108cc-22-61ca12ac@10.10.202.97
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 101 INFO
Contact: <sip:3120@10.10.202.96:5060;transport=tcp>
Content-Type: application/media_control+xml
Content-Length: 190

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<media_control>

  <vc_primitive>
    <to_encoder>
      <picture_fast_update/>
    </to_encoder>
  </vc_primitive>

</media_control>
```

[diagram] Call-ID:[prev][next]
[15] INFO sip:4140@10.10.198.24:44773;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12821e14c0e
From: <sip:3120@10.10.202.97>;tag=9873~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037389
To: <sip:4140@10.10.202.97>;tag=8dfa692fd5736da6
Date: Wed, 22 Jun 2011 15:22:53 GMT
Call-ID: 3232ea9eb98063c0@10.10.198.24
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 101 INFO
Contact: <sip:3120@10.10.202.97:5060;transport=tcp>
Content-Type: application/media_control+xml
Content-Length: 190

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<media_control>

  <vc_primitive>
    <to_encoder>
      <picture_fast_update/>
```

```
    </to_encoder>
  </vc_primitive>

</media_control>
```


```
[diagram]   Call-ID:[prev][next]
[16] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12821e14c0e;received=10.10.202.97
Call-ID: 3232ea9eb98063c0@10.10.198.24
CSeq: 101 INFO
From: <sip:3120@10.10.202.97>;tag=9873~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037389
To: <sip:4140@10.10.202.97>;tag=8dfa692fd5736da6
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-service-
control,norefersub,extended-refer
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[17] BYE sip:3120@10.10.202.97:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bKb43f459f9801ac5adc11c6e16ca62fe3.1;rport
Call-ID: 3232ea9eb98063c0@10.10.198.24
CSeq: 101 BYE
Contact: <sip:4140@10.10.198.24:44773;transport=tcp>;+sip.instance="<urn:uuid:00000000-0000-0000-0000-
0050600491d7>";+u.sip!model.ccm.cisco.com="584";audio=TRUE;video=TRUE;mobility="fixed";duplex="full";descrip
tion="TANDBERG-SIP"
From: <sip:4140@10.10.202.97>;tag=8dfa692fd5736da6
To: <sip:3120@10.10.202.97>;tag=9873~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037389
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-service-
control,norefersub,extended-refer
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[18] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bKb43f459f9801ac5adc11c6e16ca62fe3.1;rport
From: <sip:4140@10.10.202.97>;tag=8dfa692fd5736da6
To: <sip:3120@10.10.202.97>;tag=9873~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037389
Date: Wed, 22 Jun 2011 15:23:13 GMT
Call-ID: 3232ea9eb98063c0@10.10.198.24
CSeq: 101 BYE
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[19] BYE sip:3120@10.10.202.96:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK1292e5db3b
From: <sip:4140@vcs.domain>;tag=9874~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037390
To: <sip:3120@10.10.202.96>;tag=15969~b43ea526-a700-4a22-8605-9a5b66389cb6-17037406
Date: Wed, 22 Jun 2011 15:22:54 GMT
Call-ID: 7d953380-e02108cc-22-61ca12ac@10.10.202.97
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Max-Forwards: 70
P-Asserted-Identity: <sip:4140@vcs.domain>
CSeq: 102 BYE
Reason: Q.850;cause=16
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[20] NOTIFY sip:4230@10.10.202.190:38139;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12a619a233c
From: <sip:4140@10.10.202.97>;tag=1669962485
To: <sip:4230@10.10.202.97>;tag=0653a9395eb8cc0d
Call-ID: 14004e146e462686@10.10.202.190
CSeq: 137 NOTIFY
Max-Forwards: 70
Date: Wed, 22 Jun 2011 15:23:13 GMT
User-Agent: Cisco-CUCM8.6
Event: presence
Subscription-State: active;expires=2274
Contact: <sip:4140@10.10.202.97:5060;transport=tcp>
Content-Type: application/pidf+xml
Content-Length: 733
```


```
<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<presence xmlns="urn:ietf:params:xml:ns:pidf" entity="sip:4140@10.10.202.97"
xmlns:e="urn:ietf:params:xml:ns:pidf:status:rpid" xmlns:dm="urn:ietf:params:xml:ns:pidf:data-model"
xmlns:ce="urn:cisco:params:xml:ns:pidf:rpid" xmlns:sc="urn:ietf:params:xml:ns:pidf:servcaps">
  <dm:person>
```

```
    <status>
    <basic>open</basic>
    </status>
    <e:activities>
    </e:activities>
  </dm:person>
  <tuple id="cmp-1-351">
    <status>
      <basic>open</basic>
    </status>
    <sc:servcaps>
      <sc:audio>true</sc:audio>
    </sc:servcaps>
    <contact priority="0.8">sip:4140@10.10.202.97:5060</contact>
    <timestamp>2011-06-22T15:23:13Z</timestamp>
  </tuple>
</presence>
```

[diagram] Call-ID:**[prev]**[next]

```
[21] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK1292e5db3b
From: <sip:4140@vcs.domain>;tag=9874~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037390
To: <sip:3120@10.10.202.96>;tag=15969~b43ea526-a700-4a22-8605-9a5b66389cb6-17037406
Date: Wed, 22 Jun 2011 15:23:13 GMT
Call-ID: 7d953380-e02108cc-22-61ca12ac@10.10.202.97
CSeq: 102 BYE
Content-Length: 0
```

[diagram] Call-ID:**[prev]**[next]

```
[22] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12a619a233c;received=10.10.202.97
Call-ID: 14004e146e462686@10.10.202.190
CSeq: 137 NOTIFY
From: <sip:4140@10.10.202.97>;tag=1669962485
To: <sip:4230@10.10.202.97>;tag=0653a9395eb8cc0d
Server: TANDBERG/257 (TE4.1.0.253886Alpha4)
Content-Length: 0
```

2.5 Basic video call between EX90 and 9971 via a SIP ICT

Title: Basic video call between EX90 and 9971 via a SIP ICT

Description:

The following call flow illustrates the SIP messaging that takes place between two Cisco Unified CMs via an inter cluster SIP trunk for a video call between EX90 and 9971 SIP.

Cisco Unified CM1 sent out the initial INVITE.

Configuration:

Node = Unified CM1, IP = 10.10.202.97
Node = Unified CM2, IP = 10.10.202.96

Phone = A, Line = 4140, IP = 10.10.198.24, Model = Cisco Telepresence EX90
Phone = B, Line = 3210, IP = 10.10.202.63, Model = SIP 9971

A has laptop connected to it via HDMI.
SIP Trunk between Unified CM1 and Unified CM2 has a route pattern 3XXX.

Scenario:

Phone A DN 4140 calls Phone B DN 3210
Phone B DN 3210 answers the call
Phone A DN 4140 goes onhook

End of Scenario

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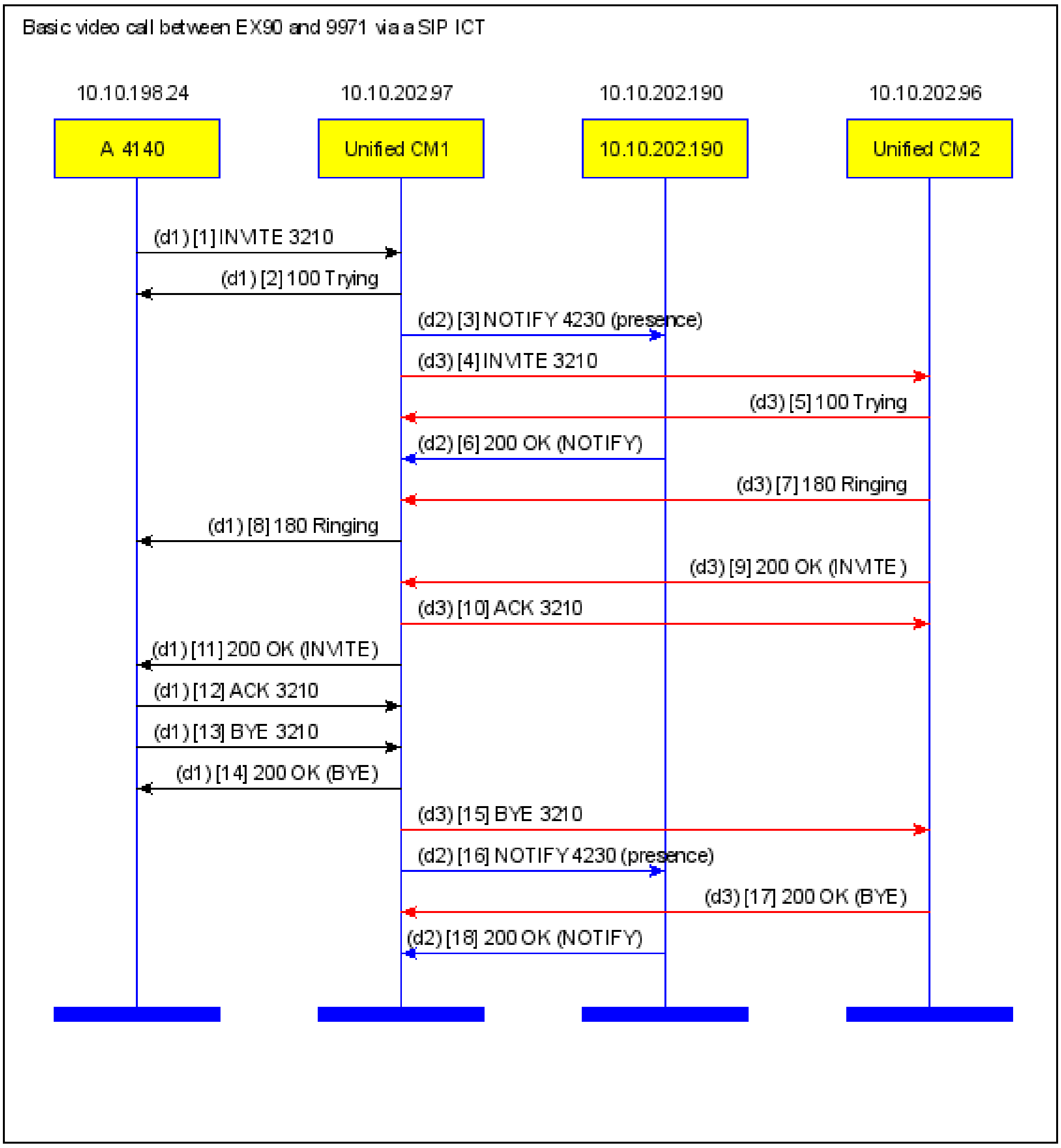

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```
[diagram]   Call-ID:[prev][next]
[1] INVITE sip:3210@10.10.202.97 SIP/2.0
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bKb2db5f34b707a3d0f77162dc37696ba4.1;rport
Call-ID: 4d7774bb727f1730@10.10.198.24
CSeq: 100 INVITE
Contact: <sip:4140@10.10.198.24:44773;transport=tcp>
From: <sip:4140@10.10.202.97>;tag=b72982061e809d0b
To: <sip:3210@10.10.202.97>
Max-Forwards: 70
Route: <sip:10.10.202.97;lr>
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-service-
control,norefersub,extended-refer
Session-Expires: 1800
Remote-Party-ID: <sip:4140@10.10.202.97>;privacy=off;id-type=subscriber;screen=yes;party=calling
Content-Type: application/sdp
Content-Length: 1575

v=0
o=tandberg 9 1 IN IP4 10.10.198.24
s=-
c=IN IP4 10.10.198.24
b=AS:6000
t=0 0
m=audio 16440 RTP/AVP 100 102 103 104 105 106 9 8 0 101
b=TIAS:128000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 profile-level-id=25;object=23;bitrate=128000
a=rtpmap:102 MP4A-LATM/90000
a=fmtp:102 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:103 MP4A-LATM/90000
a=fmtp:103 profile-level-id=24;object=23;bitrate=56000
a=rtpmap:104 MP4A-LATM/90000
a=fmtp:104 profile-level-id=24;object=23;bitrate=48000
a=rtpmap:105 G7221/16000
a=fmtp:105 bitrate=32000
a=rtpmap:106 G7221/16000
a=fmtp:106 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
m=video 16442 RTP/AVP 97 98 99 34 31
b=TIAS:6000000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=428016;max-br=5000;max-mbps=108000;max-fs=3600;max-smbps=108000;max-fps=6000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=428016;max-br=5000;max-mbps=108000;max-fs=3600;max-smbps=108000;packetization-
mode=1;max-fps=6000
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1280,768,3;custom=1280,720,3;custom=1024,768,1;custom=1024,576,2;custom=800,600,1;cif4=1;custom=720,4
80,1;custom=640,480,1;custom=512,288,1;cif=1;custom=352,240,1;qcif=1;maxbr=20000
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=1;cif=1;qcif=1;maxbr=20000
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=20000
a=rtcp-fb:* nack pli
a=sendrecv
a=content:main
a=label:11
a=answer:full
m=application 16446 RTP/AVP 107
a=rtpmap:107 H224/4800
a=sendrecv
```

```
[diagram]   Call-ID:[prev][next]
[2] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bKb2db5f34b707a3d0f77162dc37696ba4.1;rport
From: <sip:4140@10.10.202.97>;tag=b72982061e809d0b
To: <sip:3210@10.10.202.97>
Date: Wed, 22 Jun 2011 15:25:59 GMT
Call-ID: 4d7774bb727f1730@10.10.198.24
CSeq: 100 INVITE
Allow-Events: presence
Content-Length: 0
```

```
[diagram]   Call-ID:[prev][next]
[3] NOTIFY sip:4230@10.10.202.190:38139;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12b21ea14f3
From: <sip:4140@10.10.202.97>;tag=1669962485
To: <sip:4230@10.10.202.97>;tag=0653a9395eb8cc0d
Call-ID: 14004e146e462686@10.10.202.190
CSeq: 138 NOTIFY
Max-Forwards: 70
Date: Wed, 22 Jun 2011 15:25:59 GMT
User-Agent: Cisco-CUCM8.6
Event: presence
Subscription-State: active;expires=2108
Contact: <sip:4140@10.10.202.97:5060;transport=tcp>
Content-Type: application/pidf+xml
Content-Length: 826
```


```
<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<presence xmlns="urn:ietf:params:xml:ns:pidf" entity="sip:4140@10.10.202.97"
xmlns:e="urn:ietf:params:xml:ns:pidf:status:rpid" xmlns:dm="urn:ietf:params:xml:ns:pidf:data-model"
xmlns:ce="urn:cisco:params:xml:ns:pidf:rpid" xmlns:sc="urn:ietf:params:xml:ns:pidf:servcaps">
  <dm:person>
    <status>
    <basic>open</basic>
    </status>
    <e:activities>
      <e:on-the-phone/>
    </e:activities>
  </dm:person>
  <tuple id="cmp-1-351">
    <status>
      <basic>open</basic>
      <e:activities>
        <e:on-the-phone/>
      </e:activities>
    </status>
    <sc:servcaps>
      <sc:audio>true</sc:audio>
    </sc:servcaps>
    <contact priority="0.8">sip:4140@10.10.202.97:5060</contact>
    <timestamp>2011-06-22T15:25:59Z</timestamp>
  </tuple>
</presence>
```


```
[diagram]   Call-ID:[prev][next]
[4] INVITE sip:3210@10.10.202.96:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12c69b52412
From: <sip:4140@vcs.domain>;tag=9905~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037392
To: <sip:3210@10.10.202.96>
Date: Wed, 22 Jun 2011 15:25:59 GMT
Call-ID: ed0b2300-e0210987-23-61ca12ac@10.10.202.97
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Cisco-Guid: 3976930048-0000065536-0000000009-1640633004
Session-Expires:  1800
P-Asserted-Identity: <sip:4140@vcs.domain>
Remote-Party-ID: <sip:4140@vcs.domain>;party=calling;screen=yes;privacy=off
Contact: <sip:4140@10.10.202.97:5060;transport=tcp>;video;audio
Max-Forwards: 69
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[5] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12c69b52412
From: <sip:4140@vcs.domain>;tag=9905~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037392
To: <sip:3210@10.10.202.96>
Date: Wed, 22 Jun 2011 15:25:59 GMT
Call-ID: ed0b2300-e0210987-23-61ca12ac@10.10.202.97
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[6] SIP/2.0 200 OK
```

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12b21ea14f3;received=10.10.202.97
Call-ID: 14004e146e462686@10.10.202.190
CSeq: 138 NOTIFY
From: <sip:4140@10.10.202.97>;tag=1669962485
To: <sip:4230@10.10.202.97>;tag=0653a9395eb8cc0d
Server: TANDBERG/257 (TE4.1.0.253886Alpha4)
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[7] SIP/2.0 180 Ringing

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12c69b52412
From: <sip:4140@vcs.domain>;tag=9905~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037392
To: <sip:3210@10.10.202.96>;tag=16018~b43ea526-a700-4a22-8605-9a5b66389cb6-17037409
Date: Wed, 22 Jun 2011 15:25:59 GMT
Call-ID: ed0b2300-e0210987-23-61ca12ac@10.10.202.97
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:3210@10.10.202.96>
Remote-Party-ID: <sip:3210@10.10.202.96>;party=called;screen=yes;privacy=off
Contact: <sip:3210@10.10.202.96:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[8] SIP/2.0 180 Ringing

```
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bKb2db5f34b707a3d0f77162dc37696ba4.1;rport
From: <sip:4140@10.10.202.97>;tag=b72982061e809d0b
To: <sip:3210@10.10.202.97>;tag=9904~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037391
Date: Wed, 22 Jun 2011 15:25:59 GMT
Call-ID: 4d7774bb727f1730@10.10.198.24
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; ui-state= ringout;
gci= 1-162009; call-instance= 1
Send-Info: conference, x-cisco-conference
Remote-Party-ID: <sip:3210@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:3210@10.10.202.97:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[9] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12c69b52412
From: <sip:4140@vcs.domain>;tag=9905~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037392
To: <sip:3210@10.10.202.96>;tag=16018~b43ea526-a700-4a22-8605-9a5b66389cb6-17037409
Date: Wed, 22 Jun 2011 15:25:59 GMT
Call-ID: ed0b2300-e0210987-23-61ca12ac@10.10.202.97
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence, kpml
Supported: replaces
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Session-Expires:  1800;refresher=uas
Require:  timer
P-Asserted-Identity: <sip:3210@10.10.202.96>
Remote-Party-ID: <sip:3210@10.10.202.96>;party=called;screen=yes;privacy=off
Contact: <sip:3210@10.10.202.96:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 778

v=0
o=CiscoSystemsCCM-SIP 16018 1 IN IP4 10.10.202.96
s=SIP Call
c=IN IP4 10.10.202.63
b=TIAS:1256000
b=AS:1256
t=0 0
m=audio 22422 RTP/AVP 9 124 0 8 116 18 101
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:124 iSAC/16000
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:8 PCMA/8000
a=ptime:20
a=rtpmap:116 iLBC/8000
a=ptime:20
```

```
a=maxptime:20
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=ptime:20
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 29658 RTP/AVP 126 97
b=TIAS:1000000
a=rtpmap:126 H264/90000
a=fmtp:126 profile-level-id=42801E;packetization-mode=1;level-asymmetry-allowed=1
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42801E;packetization-mode=0;level-asymmetry-allowed=1
a=imageattr:* recv [x=640,y=480,q=0.50]
```

[diagram] Call-ID:[prev][next]
[10] ACK sip:3210@10.10.202.96:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12deed6117
From: <sip:4140@vcs.domain>;tag=9905~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037392
To: <sip:3210@10.10.202.96>;tag=16018~b43ea526-a700-4a22-8605-9a5b66389cb6-17037409
Date: Wed, 22 Jun 2011 15:25:59 GMT
Call-ID: ed0b2300-e0210987-23-61ca12ac@10.10.202.97
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 509

v=0
o=CiscoSystemsCCM-SIP 9905 1 IN IP4 10.10.202.97
s=SIP Call
c=IN IP4 10.10.198.24
b=TIAS:6000000
b=AS:6000
t=0 0
m=audio 16440 RTP/AVP 9 101
b=TIAS:64000
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 16442 RTP/AVP 98
b=TIAS:5936000
a=label:11
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-smbps=108000;max-fps=6000
a=content:main
a=rtcp-fb:* nack pli
```

[diagram] Call-ID:[prev][next]
[11] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bKb2db5f34b707a3d0f77162dc37696ba4.1;rport
From: <sip:4140@10.10.202.97>;tag=b72982061e809d0b
To: <sip:3210@10.10.202.97>;tag=9904~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037391
Date: Wed, 22 Jun 2011 15:25:59 GMT
Call-ID: 4d7774bb727f1730@10.10.198.24
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; gci= 1-162009;
call-instance= 1
Send-Info: conference, x-cisco-conference
Session-Expires:  1800;refresher=uas
Require:  timer
Remote-Party-ID: <sip:3210@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:3210@10.10.202.97:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 488

v=0
o=CiscoSystemsCCM-SIP 9904 1 IN IP4 10.10.202.97
s=SIP Call
b=AS:1256
t=0 0
m=audio 22422 RTP/AVP 9 101
c=IN IP4 10.10.202.63
b=TIAS:64000
a=rtpmap:9 G722/8000
```

```
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 29658 RTP/AVP 126
c=IN IP4 10.10.202.63
b=TIAS:1192000
a=rtpmap:126 H264/90000
a=fmtp:126 profile-level-id=42801E;packetization-mode=1;level-asymmetry-allowed=1
m=application 0 RTP/AVP 107
c=IN IP4 10.10.198.24
a=rtpmap:107 H224/0

[diagram]  Call-ID:[prev][next]
[12] ACK sip:3210@10.10.202.97:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK488f8f836e4da36c94c0a9e54a5f8bdd.1;rport
Call-ID: 4d7774bb727f1730@10.10.198.24
CSeq: 100 ACK
From: <sip:4140@10.10.202.97>;tag=b72982061e809d0b
To: <sip:3210@10.10.202.97>;tag=9904~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037391
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-service-
control,norefersub,extended-refer
Content-Length: 0

[diagram]  Call-ID:[prev][next]
[13] BYE sip:3210@10.10.202.97:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK7281fea4727c2283503311170eee5e32.1;rport
Call-ID: 4d7774bb727f1730@10.10.198.24
CSeq: 101 BYE
Contact: <sip:4140@10.10.198.24:44773;transport=tcp>;+sip.instance="<urn:uuid:00000000-0000-0000-0000-
0050600491d7>";+u.sip!model.ccm.cisco.com="584";audio=TRUE;video=TRUE;mobility="fixed";duplex="full";descrip
tion="TANDBERG-SIP"
From: <sip:4140@10.10.202.97>;tag=b72982061e809d0b
To: <sip:3210@10.10.202.97>;tag=9904~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037391
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-service-
control,norefersub,extended-refer
Content-Length: 0

[diagram]  Call-ID:[prev][next]
[14] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.198.24:5060;branch=z9hG4bK7281fea4727c2283503311170eee5e32.1;rport
From: <sip:4140@10.10.202.97>;tag=b72982061e809d0b
To: <sip:3210@10.10.202.97>;tag=9904~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037391
Date: Wed, 22 Jun 2011 15:26:23 GMT
Call-ID: 4d7774bb727f1730@10.10.198.24
CSeq: 101 BYE
Content-Length: 0

[diagram]  Call-ID:[prev][next]
[15] BYE sip:3210@10.10.202.96:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12e5a6ca4e0
From: <sip:4140@vcs.domain>;tag=9905~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037392
To: <sip:3210@10.10.202.96>;tag=16018~b43ea526-a700-4a22-8605-9a5b66389cb6-17037409
Date: Wed, 22 Jun 2011 15:25:59 GMT
Call-ID: ed0b2300-e0210987-23-61ca12ac@10.10.202.97
User-Agent: TANDBERG/516 (TC5.0.0.-PreAlpha (TEST SW))
Max-Forwards: 70
P-Asserted-Identity: <sip:4140@vcs.domain>
CSeq: 102 BYE
Reason: Q.850;cause=16
Content-Length: 0

[diagram]  Call-ID:[prev][next]
[16] NOTIFY sip:4230@10.10.202.190:38139;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12f1cba4cc9
From: <sip:4140@10.10.202.97>;tag=1669962485
To: <sip:4230@10.10.202.97>;tag=0653a9395eb8cc0d
Call-ID: 14004e146e462686@10.10.202.190
CSeq: 139 NOTIFY
Max-Forwards: 70
Date: Wed, 22 Jun 2011 15:26:23 GMT
User-Agent: Cisco-CUCM8.6
Event: presence
Subscription-State: active;expires=2084
Contact: <sip:4140@10.10.202.97:5060;transport=tcp>
```

```
Content-Type: application/pidf+xml
Content-Length: 733

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<presence xmlns="urn:ietf:params:xml:ns:pidf" entity="sip:4140@10.10.202.97"
xmlns:e="urn:ietf:params:xml:ns:pidf:status:rpid" xmlns:dm="urn:ietf:params:xml:ns:pidf:data-model"
xmlns:ce="urn:cisco:params:xml:ns:pidf:rpid" xmlns:sc="urn:ietf:params:xml:ns:pidf:servcaps">
  <dm:person>
    <status>
    <basic>open</basic>
    </status>
    <e:activities>
    </e:activities>
  </dm:person>
  <tuple id="cmp-1-351">
    <status>
      <basic>open</basic>
    </status>
    <sc:servcaps>
      <sc:audio>true</sc:audio>
    </sc:servcaps>
    <contact priority="0.8">sip:4140@10.10.202.97:5060</contact>
    <timestamp>2011-06-22T15:26:23Z</timestamp>
  </tuple>
</presence>

[diagram]   Call-ID:[prev][next]
[17] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12e5a6ca4e0
From: <sip:4140@vcs.domain>;tag=9905~b5a88942-7acc-4cc9-9d65-67021cfecfed-17037392
To: <sip:3210@10.10.202.96>;tag=16018~b43ea526-a700-4a22-8605-9a5b66389cb6-17037409
Date: Wed, 22 Jun 2011 15:26:23 GMT
Call-ID: ed0b2300-e0210987-23-61ca12ac@10.10.202.97
CSeq: 102 BYE
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[18] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK12f1cba4cc9;received=10.10.202.97
Call-ID: 14004e146e462686@10.10.202.190
CSeq: 139 NOTIFY
From: <sip:4140@10.10.202.97>;tag=1669962485
To: <sip:4230@10.10.202.97>;tag=0653a9395eb8cc0d
Server: TANDBERG/257 (TE4.1.0.253886Alpha4)
Content-Length: 0
```

3. CUCM G.722.1 Codec Support

3.1 Basic G.722.1 call between two Tandberg SIP Devices via a Inter Cluster SIP Trunk

Title: Basic G.722.1 call between two Tandberg SIP Devices via a Inter Cluster SIP Trunk

Description:
The following call flow illustrates the SIP messaging that takes place between
two Cisco Unified CMs via an inter cluster SIP trunk.

Cisco Unified CM1 sent out the initial INVITE.

Configuration:
Node = Unified CM1, IP = 10.10.79.23
Node = Unified CM2, IP = 10.10.63.47

Phone = A, Line = 3003, IP = 10.10.81.77, Model = SIP
Phone = B, Line = 202001, IP = 10.10.58.75, Model = SIP

SIP Trunk has route pattern 99.XXXXXX

Scenario:
Phone A DN 3003 calls Phone B DN 202001 over the SIP Trunk by dialing 99202001
Phone B DN 202001 Answers
Phone A DN 3003 goes onHook
End of Scenario

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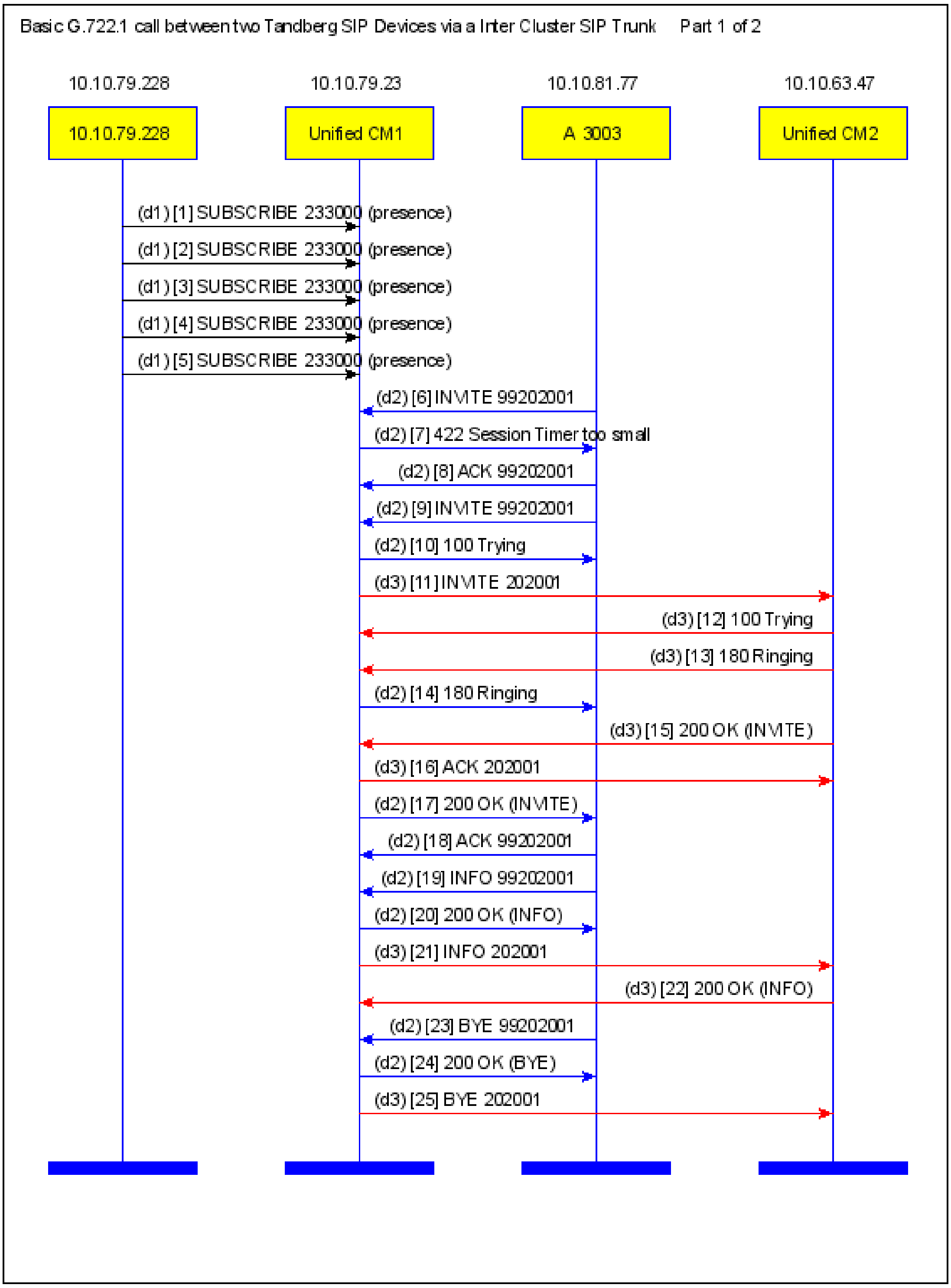
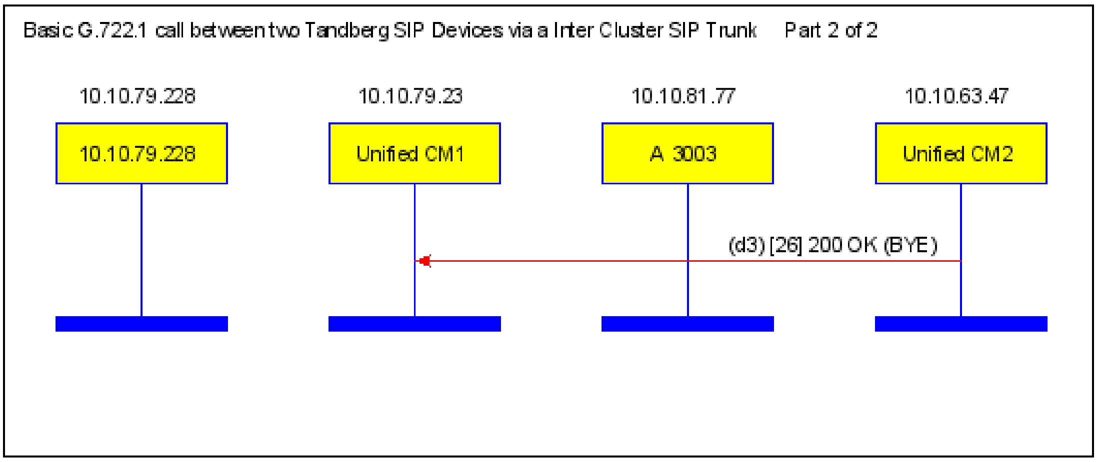


CISCO CONFIDENTIAL



```
[diagram]   Call-ID:[prev][next]
[1] SUBSCRIBE sip:233000@10.10.79.23 SIP/2.0
Via: SIP/2.0/UDP 10.10.79.228:5060;branch=z9hG4bKc7b180f916621a99246bed81177cc853.1;rport
Call-ID: a31874fadcf765bf@10.10.79.228
CSeq: 7947 SUBSCRIBE
Contact: <sip:4500@10.10.79.228:5060>;+sip.instance="<urn:uuid:00000000-0000-0000-0000-
005060041f52>";+u.sip!model.ccm.cisco.com="580";audio=TRUE;video=TRUE;mobility="fixed";duplex="full";descrip
tion="TANDBERG-SIP"
From: <sip:4500@10.10.79.23>;tag=f4259d355dca4cdf
To: <sip:233000@10.10.79.23>;tag=1493690881
Max-Forwards: 70
Route: <sip:10.10.79.23;lr>
User-Agent: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Expires: 3600
Event: presence
Accept: application/pidf+xml
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[2] SUBSCRIBE sip:233000@10.10.79.23 SIP/2.0
Via: SIP/2.0/UDP 10.10.79.228:5060;branch=z9hG4bKc7b180f916621a99246bed81177cc853.1;rport
Call-ID: a31874fadcf765bf@10.10.79.228
CSeq: 7947 SUBSCRIBE
Contact: <sip:4500@10.10.79.228:5060>;+sip.instance="<urn:uuid:00000000-0000-0000-0000-
005060041f52>";+u.sip!model.ccm.cisco.com="580";audio=TRUE;video=TRUE;mobility="fixed";duplex="full";descrip
tion="TANDBERG-SIP"
From: <sip:4500@10.10.79.23>;tag=f4259d355dca4cdf
To: <sip:233000@10.10.79.23>;tag=1493690881
Max-Forwards: 70
Route: <sip:10.10.79.23;lr>
User-Agent: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Expires: 3600
Event: presence
Accept: application/pidf+xml
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[3] SUBSCRIBE sip:233000@10.10.79.23 SIP/2.0
Via: SIP/2.0/UDP 10.10.79.228:5060;branch=z9hG4bKc7b180f916621a99246bed81177cc853.1;rport
Call-ID: a31874fadcf765bf@10.10.79.228
CSeq: 7947 SUBSCRIBE
Contact: <sip:4500@10.10.79.228:5060>;+sip.instance="<urn:uuid:00000000-0000-0000-0000-
005060041f52>";+u.sip!model.ccm.cisco.com="580";audio=TRUE;video=TRUE;mobility="fixed";duplex="full";descrip
tion="TANDBERG-SIP"
From: <sip:4500@10.10.79.23>;tag=f4259d355dca4cdf
To: <sip:233000@10.10.79.23>;tag=1493690881
Max-Forwards: 70
Route: <sip:10.10.79.23;lr>
User-Agent: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Expires: 3600
Event: presence
Accept: application/pidf+xml
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[4] SUBSCRIBE sip:233000@10.10.79.23 SIP/2.0
Via: SIP/2.0/UDP 10.10.79.228:5060;branch=z9hG4bKc7b180f916621a99246bed81177cc853.1;rport
Call-ID: a31874fadcf765bf@10.10.79.228
CSeq: 7947 SUBSCRIBE
Contact: <sip:4500@10.10.79.228:5060>;+sip.instance="<urn:uuid:00000000-0000-0000-0000-
005060041f52>";+u.sip!model.ccm.cisco.com="580";audio=TRUE;video=TRUE;mobility="fixed";duplex="full";descrip
tion="TANDBERG-SIP"
From: <sip:4500@10.10.79.23>;tag=f4259d355dca4cdf
To: <sip:233000@10.10.79.23>;tag=1493690881
Max-Forwards: 70
Route: <sip:10.10.79.23;lr>
User-Agent: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Expires: 3600
Event: presence
Accept: application/pidf+xml
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[5] SUBSCRIBE sip:233000@10.10.79.23 SIP/2.0
Via: SIP/2.0/UDP 10.10.79.228:5060;branch=z9hG4bKc7b180f916621a99246bed81177cc853.1;rport
Call-ID: a31874fadcf765bf@10.10.79.228
CSeq: 7947 SUBSCRIBE
Contact: <sip:4500@10.10.79.228:5060>;+sip.instance="<urn:uuid:00000000-0000-0000-0000-
005060041f52>";+u.sip!model.ccm.cisco.com="580";audio=TRUE;video=TRUE;mobility="fixed";duplex="full";descrip
tion="TANDBERG-SIP"
```

```
From: <sip:4500@10.10.79.23>;tag=f4259d355dca4cdf
To: <sip:233000@10.10.79.23>;tag=1493690881
Max-Forwards: 70
Route: <sip:10.10.79.23;lr>
User-Agent: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Expires: 3600
Event: presence
Accept: application/pidf+xml
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[6] INVITE sip:99202001@10.10.79.23 SIP/2.0
Via: SIP/2.0/TCP 10.10.81.77:5060;branch=z9hG4bK163c50031f6cf3d9fd7b94507ee26e6e.1;rport
Call-ID: 9f44b7b39074676c@10.10.81.77
CSeq: 100 INVITE
Contact: <sip:3003@10.10.81.77:5060;transport=tcp>
From: "3003" <sip:3003@10.10.79.23>;tag=14889e64f918a627
To: <sip:99202001@10.10.79.23>
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/73 (F8.2 NTSC)
Supported: replaces,100rel,timer,com.tandberg.sdp.extensions.v1
Session-Expires: 500
Content-Type: application/sdp
Content-Length: 1419

v=0
o=tandberg 1 1 IN IP4 10.10.81.77
s=-
c=IN IP4 10.10.81.77
b=CT:768
t=0 0
m=audio 46532 RTP/AVP 100 101 102 103 104 9 15 0 8 105
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:101 MP4A-LATM/90000
a=fmtp:101 profile-level-id=24;object=23;bitrate=56000
a=rtpmap:102 MP4A-LATM/90000
a=fmtp:102 profile-level-id=24;object=23;bitrate=48000
a=rtpmap:103 G7221/16000
a=fmtp:103 bitrate=32000
a=rtpmap:104 G7221/16000
a=fmtp:104 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:15 G728/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:105 telephone-event/8000
a=fmtp:105 0-15
a=sendrecv
m=video 46534 RTP/AVP 97 98 99 34 31
c=IN IP4 10.10.81.77
b=TIAS:768000
a=rtpmap:97 H264-RCDO/90000
a=fmtp:97 profile-level-id=008016;max-mbps=42000;max-fs=3600;max-smbps=323500
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=428016;max-mbps=35000;max-fs=3600;max-smbps=323500
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1280,800,0;custom=1280,768,0;custom=1280,720,3;custom=1024,768,4;custom=1024,576,2;custom=800,600,3;c
if4=2;custom=720,480,2;custom=640,480,2;custom=512,288,1;cif=1;custom=352,240,1;qcif=1;sqcif=1;maxbr=7680
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=2;cif=1;qcif=1;sqcif=1;maxbr=7680
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=7680
a=rtcp-fb:* nack pli
a=sendrecv
a=content:main
a=label:11
a=answer:full

[diagram]   Call-ID:[prev][next]
[7] SIP/2.0 422 Session Timer too small
Via: SIP/2.0/TCP 10.10.81.77:5060;branch=z9hG4bK163c50031f6cf3d9fd7b94507ee26e6e.1;rport
From: "3003" <sip:3003@10.10.79.23>;tag=14889e64f918a627
To: <sip:99202001@10.10.79.23>;tag=1998222394
Date: Tue, 14 Jun 2011 18:21:06 GMT
Call-ID: 9f44b7b39074676c@10.10.81.77
CSeq: 100 INVITE
```

```
Allow-Events: presence
Min-SE:  1800
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]

[8] ACK sip:99202001@10.10.79.23 SIP/2.0

```
Via: SIP/2.0/TCP 10.10.81.77:5060;branch=z9hG4bK163c50031f6cf3d9fd7b94507ee26e6e.1;rport
Call-ID: 9f44b7b39074676c@10.10.81.77
CSeq: 100 ACK
From: "3003" <sip:3003@10.10.79.23>;tag=14889e64f918a627
To: <sip:99202001@10.10.79.23>;tag=1998222394
User-Agent: TANDBERG/73 (F8.2 NTSC)
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]

[9] INVITE sip:99202001@10.10.79.23 SIP/2.0

```
Via: SIP/2.0/TCP 10.10.81.77:5060;branch=z9hG4bK072a410049dad26c1474b949837ffc1a.1;rport
Call-ID: 9f44b7b39074676c@10.10.81.77
CSeq: 101 INVITE
Contact: <sip:3003@10.10.81.77:5060;transport=tcp>
From: "3003" <sip:3003@10.10.79.23>;tag=14889e64f918a627
To: <sip:99202001@10.10.79.23>
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/73 (F8.2 NTSC)
Supported: replaces,100rel,timer,com.tandberg.sdp.extensions.v1
Session-Expires: 1800
Min-SE: 1800
Content-Type: application/sdp
Content-Length: 1419

v=0
o=tandberg 1 1 IN IP4 10.10.81.77
s=-
c=IN IP4 10.10.81.77
b=CT:768
t=0 0
m=audio 46532 RTP/AVP 100 101 102 103 104 9 15 0 8 105
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:101 MP4A-LATM/90000
a=fmtp:101 profile-level-id=24;object=23;bitrate=56000
a=rtpmap:102 MP4A-LATM/90000
a=fmtp:102 profile-level-id=24;object=23;bitrate=48000
a=rtpmap:103 G7221/16000
a=fmtp:103 bitrate=32000
a=rtpmap:104 G7221/16000
a=fmtp:104 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:15 G728/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:105 telephone-event/8000
a=fmtp:105 0-15
a=sendrecv
m=video 46534 RTP/AVP 97 98 99 34 31
c=IN IP4 10.10.81.77
b=TIAS:768000
a=rtpmap:97 H264-RCDO/90000
a=fmtp:97 profile-level-id=008016;max-mbps=42000;max-fs=3600;max-smbps=323500
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=428016;max-mbps=35000;max-fs=3600;max-smbps=323500
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1280,800,0;custom=1280,768,0;custom=1280,720,3;custom=1024,768,4;custom=1024,576,2;custom=800,600,3;c
if4=2;custom=720,480,2;custom=640,480,2;custom=512,288,1;cif=1;custom=352,240,1;qcif=1;sqcif=1;maxbr=7680
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=2;cif=1;qcif=1;sqcif=1;maxbr=7680
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=7680
a=rtcp-fb:* nack pli
a=sendrecv
a=content:main
a=label:11
a=answer:full
```

[diagram]   Call-ID:[prev][next]

[10] SIP/2.0 100 Trying

```
Via: SIP/2.0/TCP 10.10.81.77:5060;branch=z9hG4bK072a410049dad26c1474b949837ffc1a.1;rport
```

```
From: "3003" <sip:3003@10.10.79.23>;tag=14889e64f918a627
To: <sip:99202001@10.10.79.23>
Date: Tue, 14 Jun 2011 18:21:06 GMT
Call-ID: 9f44b7b39074676c@10.10.81.77
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[11] INVITE sip:202001@10.10.63.47:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.79.23:5060;branch=z9hG4bK66f64ec516b
From: <sip:3003@10.10.79.23>;tag=24393~530a8eda-d660-4e45-b30d-f5ddee5ca5b7-29248076
To: <sip:202001@10.10.63.47>
Date: Tue, 14 Jun 2011 18:21:06 GMT
Call-ID: 10661a80-df71a692-1c-174f590a@10.10.79.23
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM8.6
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Cisco-Guid: 0275126912-0000065536-0000000011-0391076106
Session-Expires:  1800
P-Asserted-Identity: <sip:3003@10.10.79.23>
Remote-Party-ID: <sip:3003@10.10.79.23>;party=calling;screen=yes;privacy=off
Contact: <sip:3003@10.10.79.23:5060;transport=tcp>;video;audio
Max-Forwards: 69
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[12] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.79.23:5060;branch=z9hG4bK66f64ec516b
From: <sip:3003@10.10.79.23>;tag=24393~530a8eda-d660-4e45-b30d-f5ddee5ca5b7-29248076
To: <sip:202001@10.10.63.47>
Date: Tue, 14 Jun 2011 18:39:34 GMT
Call-ID: 10661a80-df71a692-1c-174f590a@10.10.79.23
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[13] SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.79.23:5060;branch=z9hG4bK66f64ec516b
From: <sip:3003@10.10.79.23>;tag=24393~530a8eda-d660-4e45-b30d-f5ddee5ca5b7-29248076
To: <sip:202001@10.10.63.47>;tag=240~d5bb0abe-b85e-4fd7-b416-19fe417cc76d-31346327
Date: Tue, 14 Jun 2011 18:39:34 GMT
Call-ID: 10661a80-df71a692-1c-174f590a@10.10.79.23
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:202001@10.10.63.47>
Remote-Party-ID: <sip:202001@10.10.63.47>;party=called;screen=yes;privacy=off
Contact: <sip:202001@10.10.63.47:5060;transport=tcp>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[14] SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.81.77:5060;branch=z9hG4bK072a410049dad26c1474b949837ffc1a.1;rport
From: "3003" <sip:3003@10.10.79.23>;tag=14889e64f918a627
To: <sip:99202001@10.10.79.23>;tag=24392~530a8eda-d660-4e45-b30d-f5ddee5ca5b7-29248075
Date: Tue, 14 Jun 2011 18:21:06 GMT
Call-ID: 9f44b7b39074676c@10.10.81.77
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Send-Info: conference, x-cisco-conference
Remote-Party-ID: <sip:202001@10.10.79.23>;party=called;screen=yes;privacy=off
Contact: <sip:99202001@10.10.79.23:5060;transport=tcp>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[15] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.79.23:5060;branch=z9hG4bK66f64ec516b
From: <sip:3003@10.10.79.23>;tag=24393~530a8eda-d660-4e45-b30d-f5ddee5ca5b7-29248076
To: <sip:202001@10.10.63.47>;tag=240~d5bb0abe-b85e-4fd7-b416-19fe417cc76d-31346327
```

```
Date: Tue, 14 Jun 2011 18:39:34 GMT
Call-ID: 10661a80-df71a692-1c-174f590a@10.10.79.23
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Session-Expires:  1800;refresher=uas
Require:  timer
P-Asserted-Identity: <sip:202001@10.10.63.47>
Remote-Party-ID: <sip:202001@10.10.63.47>;party=called;screen=yes;privacy=off
Contact: <sip:202001@10.10.63.47:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 1199

v=0
o=CiscoSystemsCCM-SIP 240 1 IN IP4 10.10.63.47
s=SIP Call
c=IN IP4 10.10.58.75
b=TIAS:384000
b=AS:384
t=0 0
m=audio 16408 RTP/AVP 100 9 101 102 0 8 18 103
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 bitrate=64000;profile-level-id=24;object=23
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:101 G7221/16000
a=fmtp:101 bitrate=32000
a=rtpmap:102 G7221/16000
a=fmtp:102 bitrate=24000
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=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
m=video 16410 RTP/AVP 97 98 99 34 31
b=TIAS:320000
a=label:11
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800D;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-smbps=40500
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-
smbps=40500
a=rtpmap:99 H263-1998/90000
a=fmtp:99 QCIF=1;CIF=1;CIF4=2;MAXBR=3840;CUSTOM=352,240,1
a=rtpmap:34 H263/90000
a=fmtp:34 QCIF=1;CIF=1;CIF4=2;MAXBR=3840
a=rtpmap:31 H261/90000
a=fmtp:31 CIF=1;QCIF=1;MAXBR=3840
a=content:main
a=rtcp-fb:* nack pli
m=application 16412 RTP/AVP 104
a=rtpmap:104 H224/0
```

[diagram]   Call-ID:[prev][next]

[16] ACK sip:202001@10.10.63.47:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.79.23:5060;branch=z9hG4bK67065848b42
From: <sip:3003@10.10.79.23>;tag=24393~530a8eda-d660-4e45-b30d-f5ddee5ca5b7-29248076
To: <sip:202001@10.10.63.47>;tag=240~d5bb0abe-b85e-4fd7-b416-19fe417cc76d-31346327
Date: Tue, 14 Jun 2011 18:21:06 GMT
Call-ID: 10661a80-df71a692-1c-174f590a@10.10.79.23
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 558

v=0
o=CiscoSystemsCCM-SIP 24393 1 IN IP4 10.10.79.23
s=SIP Call
b=TIAS:384000
b=AS:384
t=0 0
m=audio 46532 RTP/AVP 103 105
```

```
c=IN IP4 10.10.81.77
b=TIAS:32000
a=rtpmap:103 G7221/16000
a=fmtp:103 bitrate=32000
a=rtpmap:105 telephone-event/8000
a=fmtp:105 0-15
m=video 46534 RTP/AVP 98
c=IN IP4 10.10.81.77
b=TIAS:352000
a=label:11
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=428016;max-mbps=35000;max-fs=3600;max-smbps=323500
a=content:main
a=rtcp-fb:* nack pli
m=application 0 RTP/AVP 104
c=IN IP4 10.10.58.75
a=rtpmap:104 H224/0
```

[diagram]   Call-ID:[prev][next]
[17] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.81.77:5060;branch=z9hG4bK072a410049dad26c1474b949837ffc1a.1;rport
From: "3003" <sip:3003@10.10.79.23>;tag=14889e64f918a627
To: <sip:99202001@10.10.79.23>;tag=24392~530a8eda-d660-4e45-b30d-f5ddee5ca5b7-29248075
Date: Tue, 14 Jun 2011 18:21:06 GMT
Call-ID: 9f44b7b39074676c@10.10.81.77
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Send-Info: conference, x-cisco-conference
Session-Expires:  1800;refresher=uas
Require:  timer
Remote-Party-ID: <sip:202001@10.10.79.23>;party=called;screen=yes;privacy=off
Contact: <sip:99202001@10.10.79.23:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 482

v=0
o=CiscoSystemsCCM-SIP 24392 1 IN IP4 10.10.79.23
s=SIP Call
c=IN IP4 10.10.58.75
b=CT:384
t=0 0
m=audio 16408 RTP/AVP 101 103
b=TIAS:32000
a=rtpmap:101 G7221/16000
a=fmtp:101 bitrate=32000
a=ptime:20
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
m=video 16410 RTP/AVP 97
b=TIAS:352000
a=label:11
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800D;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-smbps=40500
a=content:main
a=rtcp-fb:* nack pli
```

[diagram]   Call-ID:[prev][next]
[18] ACK sip:99202001@10.10.79.23:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.81.77:5060;branch=z9hG4bK1c5571fd773b979a9b40efa1571621a1.1;rport
Call-ID: 9f44b7b39074676c@10.10.81.77
CSeq: 101 ACK
From: "3003" <sip:3003@10.10.79.23>;tag=14889e64f918a627
To: <sip:99202001@10.10.79.23>;tag=24392~530a8eda-d660-4e45-b30d-f5ddee5ca5b7-29248075
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/73 (F8.2 NTSC)
Supported: replaces,100rel,timer,com.tandberg.sdp.extensions.v1
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]
[19] INFO sip:99202001@10.10.79.23:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.81.77:5060;branch=z9hG4bKba3dbf000ba3a784ac3bf5f6ebb63402.1;rport
Call-ID: 9f44b7b39074676c@10.10.81.77
CSeq: 102 INFO
Contact: <sip:3003@10.10.81.77:5060>
From: "3003" <sip:3003@10.10.79.23>;tag=14889e64f918a627
To: <sip:99202001@10.10.79.23>;tag=24392~530a8eda-d660-4e45-b30d-f5ddee5ca5b7-29248075
Max-Forwards: 70
```

```
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/73 (F8.2 NTSC)
Supported: replaces,100rel,timer,com.tandberg.sdp.extensions.v1
Content-Type: application/media_control+xml
Content-Length: 168

<?xml version="1.0" encoding="utf-
8"?><media_control><vc_primitive><to_encoder><picture_fast_update></picture_fast_update></to_encoder></vc_pr
imitive></media_control>
```

[diagram]   Call-ID:[prev][next]

[20] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.81.77:5060;branch=z9hG4bKba3dbf000ba3a784ac3bf5f6ebb63402.1;rport
From: "3003" <sip:3003@10.10.79.23>;tag=14889e64f918a627
To: <sip:99202001@10.10.79.23>;tag=24392~530a8eda-d660-4e45-b30d-f5ddee5ca5b7-29248075
Date: Tue, 14 Jun 2011 18:21:08 GMT
Call-ID: 9f44b7b39074676c@10.10.81.77
CSeq: 102 INFO
Contact: <sip:99202001@10.10.79.23:5060;transport=tcp>
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]

[21] INFO sip:202001@10.10.63.47:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.79.23:5060;branch=z9hG4bK6717ad80f0e
From: <sip:3003@10.10.79.23>;tag=24393~530a8eda-d660-4e45-b30d-f5ddee5ca5b7-29248076
To: <sip:202001@10.10.63.47>;tag=240~d5bb0abe-b85e-4fd7-b416-19fe417cc76d-31346327
Date: Tue, 14 Jun 2011 18:21:06 GMT
Call-ID: 10661a80-df71a692-1c-174f590a@10.10.79.23
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 102 INFO
Contact: <sip:3003@10.10.79.23:5060;transport=tcp>
Remote-Party-ID: <sip:3003@10.10.79.23>;party=calling;screen=yes;privacy=off
Content-Type: application/media_control+xml
Content-Length: 190

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<media_control>

  <vc_primitive>
    <to_encoder>
      <picture_fast_update/>
    </to_encoder>
  </vc_primitive>

</media_control>
```

[diagram]   Call-ID:[prev][next]

[22] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.79.23:5060;branch=z9hG4bK6717ad80f0e
From: <sip:3003@10.10.79.23>;tag=24393~530a8eda-d660-4e45-b30d-f5ddee5ca5b7-29248076
To: <sip:202001@10.10.63.47>;tag=240~d5bb0abe-b85e-4fd7-b416-19fe417cc76d-31346327
Date: Tue, 14 Jun 2011 18:39:36 GMT
Call-ID: 10661a80-df71a692-1c-174f590a@10.10.79.23
CSeq: 102 INFO
Contact: <sip:202001@10.10.63.47:5060;transport=tcp>
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]

[23] BYE sip:99202001@10.10.79.23:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.81.77:5060;branch=z9hG4bK34abd7cc798f22251d850641b2218eb3.1;rport
Call-ID: 9f44b7b39074676c@10.10.81.77
CSeq: 103 BYE
Contact: <sip:3003@10.10.81.77:5060;transport=tcp>
From: "3003" <sip:3003@10.10.79.23>;tag=14889e64f918a627
To: <sip:99202001@10.10.79.23>;tag=24392~530a8eda-d660-4e45-b30d-f5ddee5ca5b7-29248075
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/73 (F8.2 NTSC)
Supported: replaces,100rel,timer,com.tandberg.sdp.extensions.v1
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]

[24] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.81.77:5060;branch=z9hG4bK34abd7cc798f22251d850641b2218eb3.1;rport
From: "3003" <sip:3003@10.10.79.23>;tag=14889e64f918a627
To: <sip:99202001@10.10.79.23>;tag=24392~530a8eda-d660-4e45-b30d-f5ddee5ca5b7-29248075
Date: Tue, 14 Jun 2011 18:21:17 GMT
Call-ID: 9f44b7b39074676c@10.10.81.77
CSeq: 103 BYE
```

```
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[25] BYE sip:202001@10.10.63.47:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.79.23:5060;branch=z9hG4bK67210ffdccd
From: <sip:3003@10.10.79.23>;tag=24393~530a8eda-d660-4e45-b30d-f5ddee5ca5b7-29248076
To: <sip:202001@10.10.63.47>;tag=240~d5bb0abe-b85e-4fd7-b416-19fe417cc76d-31346327
Date: Tue, 14 Jun 2011 18:21:06 GMT
Call-ID: 10661a80-df71a692-1c-174f590a@10.10.79.23
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
P-Asserted-Identity: <sip:3003@10.10.79.23>
CSeq: 103 BYE
Reason: Q.850;cause=16
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[26] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.79.23:5060;branch=z9hG4bK67210ffdccd
From: <sip:3003@10.10.79.23>;tag=24393~530a8eda-d660-4e45-b30d-f5ddee5ca5b7-29248076
To: <sip:202001@10.10.63.47>;tag=240~d5bb0abe-b85e-4fd7-b416-19fe417cc76d-31346327
Date: Tue, 14 Jun 2011 18:39:44 GMT
Call-ID: 10661a80-df71a692-1c-174f590a@10.10.79.23
CSeq: 103 BYE
Content-Length: 0
```

4. CUCM AAC-LD MP4A-LATM Codec Support on SIP

4.1 Basic MP4A-LATM Call between 2 SIP EndPoints over SIP Trunk

Title: Basic MP4A-LATM Call between 2 SIP EndPoints over SIP Trunk

Description:
The following call flow illustrates the SIP messaging that takes place between two Cisco Unified CMs via an inter cluster SIP trunk.

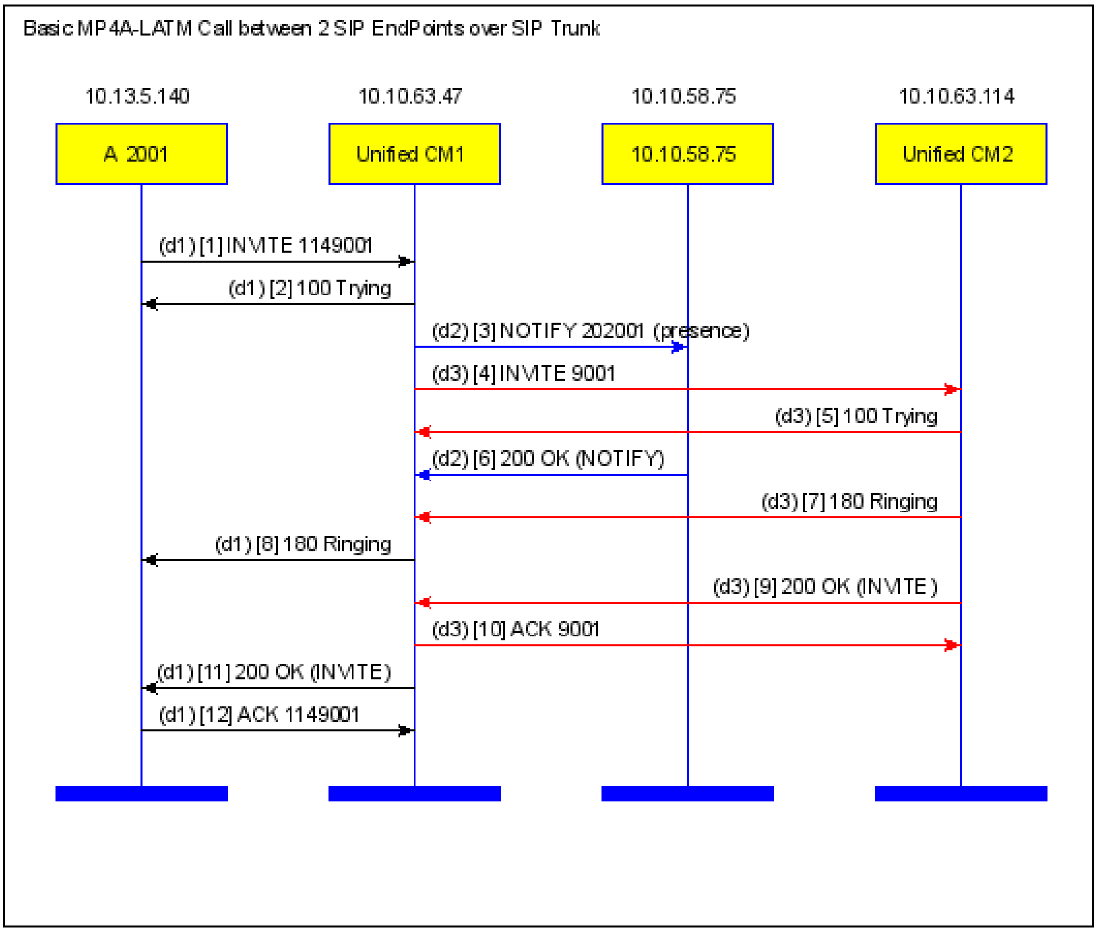
Cisco Unified CM1 sent out the initial INVITE.

Configuration:
Node = Unified CM1, IP = 10.10.63.47
Node = Unified CM2, IP = 10.10.63.114

Phone = A, Line = 2001, IP = 10.13.5.140, Model = SIP
Phone = B, Line = 9001, IP = 10.13.5.136, Model = SIP

SIP Trunk between Unified CM1 and Unified CM2 has route pattern 114.XXXX

Scenario:
Phone A DN 2001 calls Phone B DN 9001 over the SIP Trunk by dialing 1149001
Phone B DN 9001 Answers
End of Scenario


Basic MP4A-LATM Call between 2 SIP EndPoints over SIP Trunk
10.13.5.140
10.10.63.47
10.10.58.75
10.10.63.114
A 2001
Unified CM1
10.10.58.75
Unified CM2
(d1) [1] INVITE 1149001
(d1) [2] 100 Trying
(d2) [3] NOTIFY 202001 (presence)
(d3) [4] INVITE 9001
(d3) [5] 100 Trying
(d2) [6] 200 OK (NOTIFY)
(d3) [7] 180 Ringing
(d1) [8] 180 Ringing
(d3) [9] 200 OK (INVITE)
(d3) [10] ACK 9001
(d1) [11] 200 OK (INVITE)
(d1) [12] ACK 1149001

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```
[diagram]   Call-ID:[prev][next]
[1] INVITE sip:1149001@10.10.63.47 SIP/2.0
Via: SIP/2.0/TCP 10.13.5.140:5060;branch=z9hG4bK16012fc5301949b30066ddc72716770e.1;rport
Call-ID: a019034f2b8b96b3@10.13.5.140
CSeq: 100 INVITE
Contact: <sip:2001@10.13.5.140:5060;transport=tcp>
From: <sip:2001@10.10.63.47>;tag=648d982735c69df5
To: <sip:1149001@10.10.63.47>
Max-Forwards: 70
Route: <sip:10.10.63.47;lr>
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.246968)
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-service-control
Session-Expires: 1800
Remote-Party-ID: <sip:2001@10.10.63.47>;privacy=off;id-type=subscriber;screen=yes;party=calling
Content-Type: application/sdp
Content-Length: 1305

v=0
o=tandberg 5 1 IN IP4 10.13.5.140
s=-
c=IN IP4 10.13.5.140
b=AS:1152
t=0 0
m=audio 16402 RTP/AVP 100 101 102 9 18 11 8 0 103
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:101 G7221/16000
a=fmtp:101 bitrate=32000
a=rtpmap:102 G7221/16000
a=fmtp:102 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:11 L16/16000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
a=sendrecv
m=video 16404 RTP/AVP 97 98 99 34 31
b=TIAS:1152000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;packetization-mode=1
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1024,768,4;custom=1024,576,4;custom=800,600,4;cif4=2;custom=720,480,2;custom=640,480,2;custom=512,288
,1;cif=1;custom=352,240,1;qcif=1;maxbr=10880
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=2;cif=1;qcif=1;maxbr=10880
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=10880
a=rtcp-fb:* nack pli
a=sendrecv
a=content:main
a=label:11
a=answer:full
m=application 16406 RTP/AVP 104
a=rtpmap:104 H224/4800
a=sendrecv

[diagram]   Call-ID:[prev][next]
[2] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.13.5.140:5060;branch=z9hG4bK16012fc5301949b30066ddc72716770e.1;rport
From: <sip:2001@10.10.63.47>;tag=648d982735c69df5
To: <sip:1149001@10.10.63.47>
Date: Wed, 08 Jun 2011 21:45:58 GMT
Call-ID: a019034f2b8b96b3@10.13.5.140
CSeq: 100 INVITE
Allow-Events: presence
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[3] NOTIFY sip:202001@10.10.58.75:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.63.47:5060;branch=z9hG4bKb7d32fde5c1
From: <sip:2001@10.10.63.47>;tag=1314722347
To: <sip:202001@10.10.63.47>;tag=fceff5e45928265d
```

```
Call-ID: 6726bb791469779f@10.10.58.75
CSeq: 117 NOTIFY
Max-Forwards: 70
Date: Wed, 08 Jun 2011 21:45:58 GMT
User-Agent: Cisco-CUCM8.6
Event: presence
Subscription-State: active;expires=1426
Contact: <sip:2001@10.10.63.47:5060;transport=tcp>
Content-Type: application/pidf+xml
Content-Length: 824

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<presence xmlns="urn:ietf:params:xml:ns:pidf" entity="sip:2001@10.10.63.47"
xmlns:e="urn:ietf:params:xml:ns:pidf:status:rpid" xmlns:dm="urn:ietf:params:xml:ns:pidf:data-model"
xmlns:ce="urn:cisco:params:xml:ns:pidf:rpid" xmlns:sc="urn:ietf:params:xml:ns:pidf:servcaps">
  <dm:person>
    <status>
    <basic>open</basic>
    </status>
    <e:activities>
      <e:on-the-phone/>
    </e:activities>
  </dm:person>
  <tuple id="cmp-1-31010">
    <status>
      <basic>open</basic>
      <e:activities>
        <e:on-the-phone/>
      </e:activities>
    </status>
    <sc:servcaps>
      <sc:audio>true</sc:audio>
    </sc:servcaps>
    <contact priority="0.8">sip:2001@10.10.63.47:5060</contact>
    <timestamp>2011-06-08T21:45:58Z</timestamp>
  </tuple>
</presence>
```

```
[diagram]   Call-ID:[prev][next]
[4] INVITE sip:9001@10.10.63.114:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.63.47:5060;branch=z9hG4bKb7e59966e9f
From: <sip:2001@10.10.63.47>;tag=31168~d5bb0abe-b85e-4fd7-b416-19fe417cc76d-24799382
To: <sip:9001@10.10.63.114>
Date: Wed, 08 Jun 2011 21:45:58 GMT
Call-ID: b085f480-def1ed96-7c-2f3f590a@10.10.63.47
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM8.6
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Cisco-Guid: 2961568896-0000065536-0000000026-0792680714
Session-Expires:  1800
P-Asserted-Identity: <sip:2001@10.10.63.47>
Remote-Party-ID: <sip:2001@10.10.63.47>;party=calling;screen=yes;privacy=off
Contact: <sip:2001@10.10.63.47:5060;transport=tcp>;video;audio
Max-Forwards: 69
Content-Length: 0
```

```
[diagram]   Call-ID:[prev][next]
[5] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.63.47:5060;branch=z9hG4bKb7e59966e9f
From: <sip:2001@10.10.63.47>;tag=31168~d5bb0abe-b85e-4fd7-b416-19fe417cc76d-24799382
To: <sip:9001@10.10.63.114>
Date: Wed, 08 Jun 2011 21:45:58 GMT
Call-ID: b085f480-def1ed96-7c-2f3f590a@10.10.63.47
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0
```

```
[diagram]   Call-ID:[prev][next]
[6] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.63.47:5060;branch=z9hG4bKb7d32fde5c1;received=10.10.63.47
Call-ID: 6726bb791469779f@10.10.58.75
CSeq: 117 NOTIFY
From: <sip:2001@10.10.63.47>;tag=1314722347
To: <sip:202001@10.10.63.47>;tag=fceff5e45928265d
```

```
Server: TANDBERG/257 (TE4.0.0.240633)
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[7] SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.63.47:5060;branch=z9hG4bKb7e59966e9f
From: <sip:2001@10.10.63.47>;tag=31168~d5bb0abe-b85e-4fd7-b416-19fe417cc76d-24799382
To: <sip:9001@10.10.63.114>;tag=438~bb53db68-d044-4bff-9b76-e50c9a99459a-29267198
Date: Wed, 08 Jun 2011 21:45:58 GMT
Call-ID: b085f480-def1ed96-7c-2f3f590a@10.10.63.47
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:9001@10.10.63.114>
Remote-Party-ID: <sip:9001@10.10.63.114>;party=called;screen=yes;privacy=off
Contact: <sip:9001@10.10.63.114:5060;transport=tcp>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[8] SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.13.5.140:5060;branch=z9hG4bK16012fc5301949b30066ddc72716770e.1;rport
From: <sip:2001@10.10.63.47>;tag=648d982735c69df5
To: <sip:1149001@10.10.63.47>;tag=31167~d5bb0abe-b85e-4fd7-b416-19fe417cc76d-24799381
Date: Wed, 08 Jun 2011 21:45:58 GMT
Call-ID: a019034f2b8b96b3@10.13.5.140
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; ui-state= ringout;
gci= 1-9076; call-instance= 1
Send-Info: conference, x-cisco-conference
Remote-Party-ID: <sip:9001@10.10.63.47>;party=called;screen=yes;privacy=off
Contact: <sip:1149001@10.10.63.47:5060;transport=tcp>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[9] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.63.47:5060;branch=z9hG4bKb7e59966e9f
From: <sip:2001@10.10.63.47>;tag=31168~d5bb0abe-b85e-4fd7-b416-19fe417cc76d-24799382
To: <sip:9001@10.10.63.114>;tag=438~bb53db68-d044-4bff-9b76-e50c9a99459a-29267198
Date: Wed, 08 Jun 2011 21:45:58 GMT
Call-ID: b085f480-def1ed96-7c-2f3f590a@10.10.63.47
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Session-Expires:  1800;refresher=uas
Require:  timer
P-Asserted-Identity: <sip:9001@10.10.63.114>
Remote-Party-ID: <sip:9001@10.10.63.114>;party=called;screen=yes;privacy=off
Contact: <sip:9001@10.10.63.114:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 1376

v=0
o=CiscoSystemsCCM-SIP 438 1 IN IP4 10.10.63.114
s=SIP Call
c=IN IP4 10.13.5.136
b=TIAS:384000
b=AS:384
t=0 0
m=audio 16392 RTP/AVP 101 102 103 9 104 105 0 8 106
b=TIAS:64000
a=rtpmap:101 MP4A-LATM/90000
a=fmtp:101 bitrate=64000;profile-level-id=24;object=23
a=rtpmap:102 MP4A-LATM/90000
a=fmtp:102 bitrate=56000;profile-level-id=24;object=23
a=rtpmap:103 MP4A-LATM/90000
a=fmtp:103 bitrate=48000;profile-level-id=24;object=23
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:104 G7221/16000
a=fmtp:104 bitrate=32000
a=rtpmap:105 G7221/16000
a=fmtp:105 bitrate=24000
a=rtpmap:0 PCMU/8000
```

```
a=ptime:20
a=rtpmap:8 PCMA/8000
a=ptime:20
a=rtpmap:106 telephone-event/8000
a=fmtp:106 0-15
m=video 16394 RTP/AVP 97 98 99 34 31
b=TIAS:320000
a=label:11
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=428016;max-mbps=108000;max-fs=3600;max-cpb=200;max-br=5000;max-smbps=108000;max-fps=6000
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-smbps=108000;max-fps=6000
a=rtpmap:99 H263-1998/90000
a=fmtp:99 QCIF=1;CIF=1;CIF4=1;MAXBR=3840;CUSTOM=352,240,1
a=rtpmap:34 H263/90000
a=fmtp:34 QCIF=1;CIF=1;CIF4=1;MAXBR=3840
a=rtpmap:31 H261/90000
a=fmtp:31 CIF=1;QCIF=1;MAXBR=3840
a=content:main
a=rtcp-fb:* nack pli
m=application 16398 RTP/AVP 107
a=rtpmap:107 H224/0
```

[diagram]   Call-ID:[prev][next]

[10] ACK sip:9001@10.10.63.114:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.63.47:5060;branch=z9hG4bKb7f26bfa37
From: <sip:2001@10.10.63.47>;tag=31168~d5bb0abe-b85e-4fd7-b416-19fe417cc76d-24799382
To: <sip:9001@10.10.63.114>;tag=438~bb53db68-d044-4bff-9b76-e50c9a99459a-29267198
Date: Wed, 08 Jun 2011 21:45:58 GMT
Call-ID: b085f480-def1ed96-7c-2f3f590a@10.10.63.47
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 594

v=0
o=CiscoSystemsCCM-SIP 31168 1 IN IP4 10.10.63.47
s=SIP Call
c=IN IP4 10.13.5.140
b=TIAS:384000
b=AS:384
t=0 0
m=audio 16402 RTP/AVP 100 103
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 bitrate=64000;profile-level-id=24;object=23
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
m=video 16404 RTP/AVP 98
b=TIAS:320000
a=label:11
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-smbps=40500
a=content:main
a=rtcp-fb:* nack pli
m=application 16406 RTP/AVP 104
a=rtpmap:104 H224/0
```

[diagram]   Call-ID:[prev][next]

[11] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.13.5.140:5060;branch=z9hG4bK16012fc5301949b30066ddc72716770e.1;rport
From: <sip:2001@10.10.63.47>;tag=648d982735c69df5
To: <sip:1149001@10.10.63.47>;tag=31167~d5bb0abe-b85e-4fd7-b416-19fe417cc76d-24799381
Date: Wed, 08 Jun 2011 21:45:58 GMT
Call-ID: a019034f2b8b96b3@10.13.5.140
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; gci= 1-9076; call-instance= 1
Send-Info: conference, x-cisco-conference
Session-Expires:  1800;refresher=uas
Require:  timer
Remote-Party-ID: <sip:9001@10.10.63.47>;party=called;screen=yes;privacy=off
Contact: <sip:1149001@10.10.63.47:5060;transport=tcp>
```

```
Content-Type: application/sdp
Content-Length: 608

v=0
o=CiscoSystemsCCM-SIP 31167 1 IN IP4 10.10.63.47
s=SIP Call
c=IN IP4 10.13.5.136
b=AS:384
t=0 0
m=audio 16392 RTP/AVP 101 106
b=TIAS:64000
a=rtpmap:101 MP4A-LATM/90000
a=fmtp:101 bitrate=64000;profile-level-id=24;object=23
a=ptime:20
a=rtpmap:106 telephone-event/8000
a=fmtp:106 0-15
m=video 16394 RTP/AVP 98
b=TIAS:320000
a=label:11
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-smbps=108000;max-fps=6000
a=content:main
a=rtcp-fb:* nack pli
m=application 16398 RTP/AVP 107
a=rtpmap:107 H224/0
```

[diagram] Call-ID:[prev][next]

[12] ACK sip:1149001@10.10.63.47:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.13.5.140:5060;branch=z9hG4bK7868d0c5a9b0494d3d98724ff27fdfe1.1;rport
Call-ID: a019034f2b8b96b3@10.13.5.140
CSeq: 100 ACK
From: <sip:2001@10.10.63.47>;tag=648d982735c69df5
To: <sip:1149001@10.10.63.47>;tag=31167~d5bb0abe-b85e-4fd7-b416-19fe417cc76d-24799381
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.246968)
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-service-control
Content-Length: 0
```

5. SIP REFER Transparency

5.1 Basic SIP Trunk tandem call thru CUCM SME

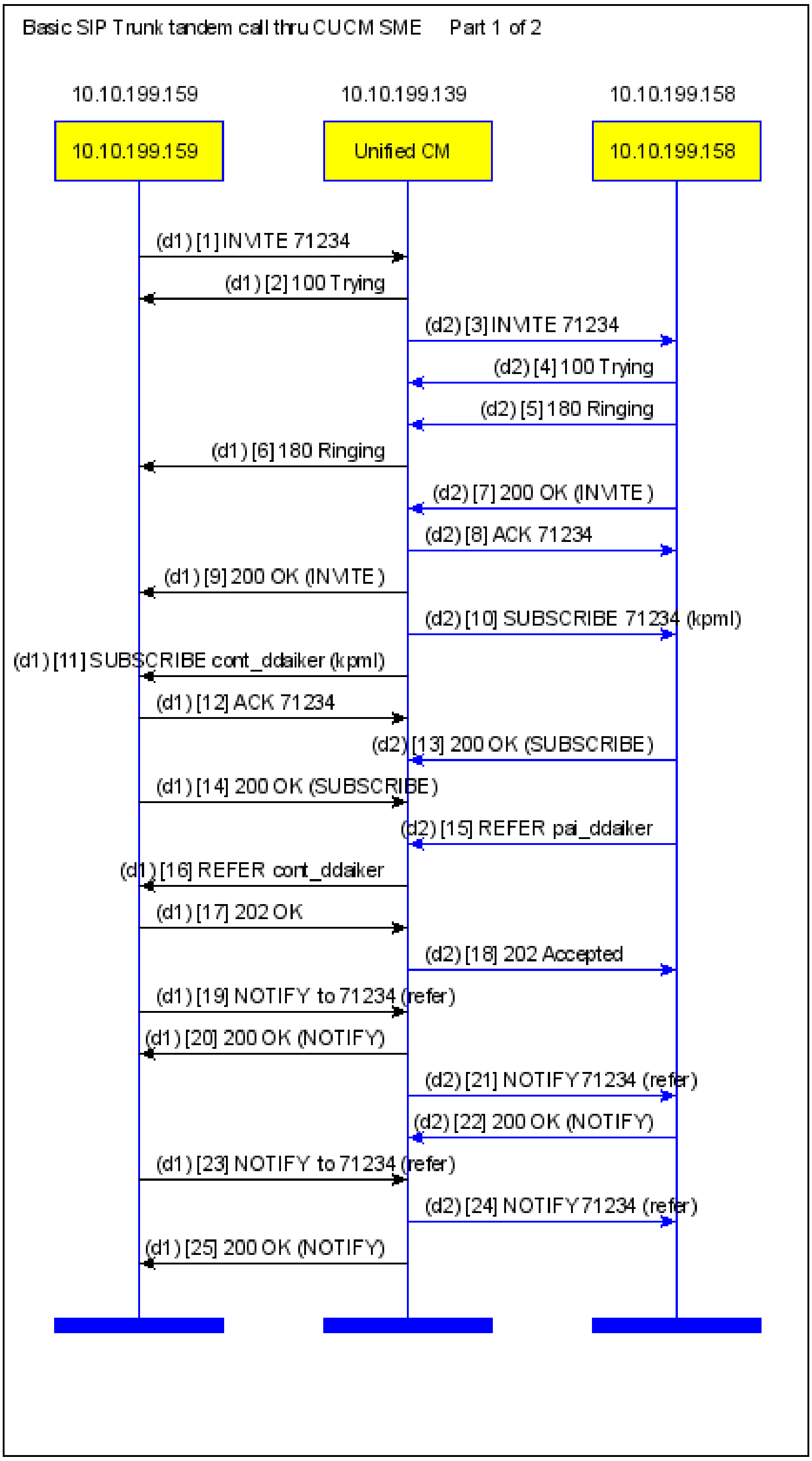
Title: Basic SIP Trunk tandem call thru CUCM SME

Description:
The following call flow shows the sip messaging that takes place thru a CUCM SME when refer pass-thru is enabled

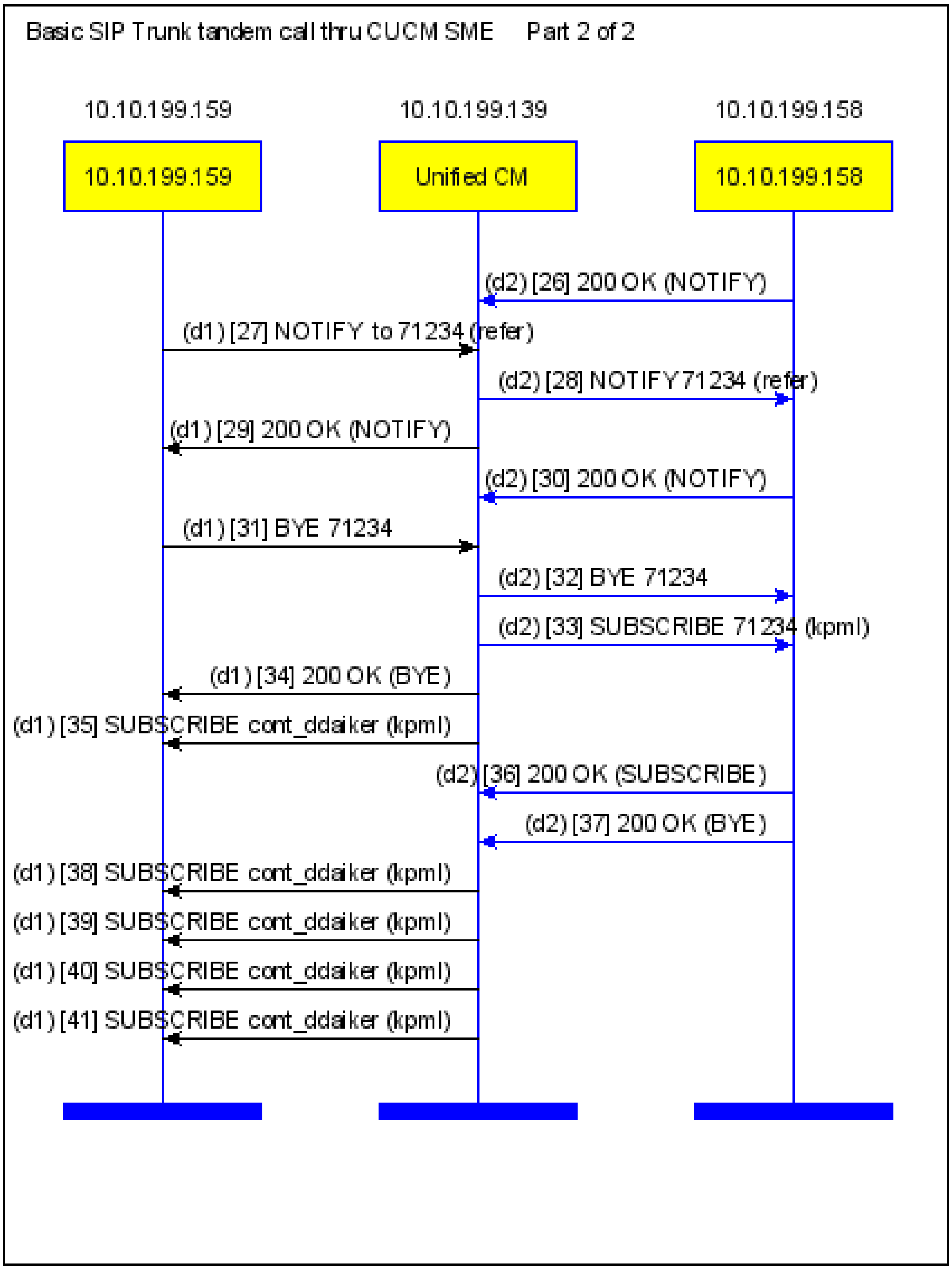
Configuration:
Node = 3rd party endpointA, (DN ddaiker) IP = 10.10.199.159
Node = Unified CM, IP = 10.10.199.139
Node = 3rd party endpointB, (DN 71234) IP = 10.10.199.158

SIP Trunk between 3rd party endpointA and CUCM
SIP Trunk between CUCM and 3rd party endpointB with route pattern 7.xxxx

Scenario:
3rd party endpointA (ddaiker) calls 3rd party endpointB (71234) via sme
3rd party endpointB answers
3rd party endpointB sends REFER to 3rd party endpointA
3rd party endpointA initiates new call
3rd party endpointA sends notify updates to 3rd party endpointB
3rd party endpointA disconnects original call
End of Scenario



Basic SIP Trunk tandem call thru CUCM SME Part 1 of 2
10.10.199.159
10.10.199.139
10.10.199.158
10.10.199.159
Unified CM
10.10.199.158
(d1) [1] INVITE 71234
(d1) [2] 100 Trying
(d2) [3] INVITE 71234
(d2) [4] 100 Trying
(d2) [5] 180 Ringing
(d1) [6] 180 Ringing
(d2) [7] 200 OK (INVITE)
(d2) [8] ACK 71234
(d1) [9] 200 OK (INVITE)
(d2) [10] SUBSCRIBE 71234 (kpml)
(d1) [11] SUBSCRIBE cont_ddaiker (kpml)
(d1) [12] ACK 71234
(d2) [13] 200 OK (SUBSCRIBE)
(d1) [14] 200 OK (SUBSCRIBE)
(d2) [15] REFER pai_ddaiker
(d1) [16] REFER cont_ddaiker
(d1) [17] 202 OK
(d2) [18] 202 Accepted
(d1) [19] NOTIFY to 71234 (refer)
(d1) [20] 200 OK (NOTIFY)
(d2) [21] NOTIFY 71234 (refer)
(d2) [22] 200 OK (NOTIFY)
(d1) [23] NOTIFY to 71234 (refer)
(d2) [24] NOTIFY 71234 (refer)
(d1) [25] 200 OK (NOTIFY)


Basic SIP Trunk tandem call thru CUCM SME Part 2 of 2
10.10.199.159
10.10.199.139
10.10.199.158
10.10.199.159
Unified CM
10.10.199.158
(d2) [26] 200 OK (NOTIFY)
(d1) [27] NOTIFY to 71234 (refer)
(d2) [28] NOTIFY 71234 (refer)
(d1) [29] 200 OK (NOTIFY)
(d2) [30] 200 OK (NOTIFY)
(d1) [31] BYE 71234
(d2) [32] BYE 71234
(d2) [33] SUBSCRIBE 71234 (kpml)
(d1) [34] 200 OK (BYE)
(d1) [35] SUBSCRIBE cont_ddaiker (kpml)
(d2) [36] 200 OK (SUBSCRIBE)
(d2) [37] 200 OK (BYE)
(d1) [38] SUBSCRIBE cont_ddaiker (kpml)
(d1) [39] SUBSCRIBE cont_ddaiker (kpml)
(d1) [40] SUBSCRIBE cont_ddaiker (kpml)
(d1) [41] SUBSCRIBE cont_ddaiker (kpml)

```
[diagram]   Call-ID:[prev][next]
[1] INVITE sip:71234@10.10.199.139 SIP/2.0
Via: SIP/2.0/Tcp 10.10.199.159:5060;branch=z9hG4bK2af-mytcltrunk--680267448-766
From: "from_ddaiker string" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
To: <sip:71234@10.10.199.139>
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 101 INVITE
User-Agent: Cisco-SIPTcl-Trunk
P-Asserted-Identity: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>
Contact: <sip:cont_ddaiker@10.10.199.159:5060;transport=Tcp>
Expires: 1800
Accept: application/sdp
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,SUBSCRIBE,INFO,UPDATE
Allow-Events: kpml
Supported: replaces, X-cisco-srtp-fallback, X-kpml
Content-Length: 294
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 12345 12345 IN IP4 10.10.199.159
s=SIPTcl Call-ddd
c=IN IP4 10.10.199.159
t=0 0
m=audio 16007 RTP/AVP 0 8 18 9 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv

[diagram]   Call-ID:[prev][next]
[2] SIP/2.0 100 Trying
Via: SIP/2.0/Tcp 10.10.199.159:5060;branch=z9hG4bK2af-mytcltrunk--680267448-766
From: "from_ddaiker string" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
To: <sip:71234@10.10.199.139>
Date: Tue, 28 Jun 2011 18:06:42 GMT
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[3] INVITE sip:71234@10.10.199.158:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK225b15ecd2
From: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
To: <sip:71234@10.10.199.158>
Date: Tue, 28 Jun 2011 18:06:42 GMT
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM8.6
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: <sip:10.10.199.139:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 1597155968-0000065536-0000000022-2345079468
Session-Expires:  1800
P-Asserted-Identity: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>
Remote-Party-ID: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;party=calling;screen=yes;privacy=off
Contact: <sip:pai_ddaiker@10.10.199.139:5060;transport=tcp>
Max-Forwards: 69
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[4] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK225b15ecd2
From: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
To: <sip:71234@10.10.199.158>
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
CSeq: 101 INVITE
Server: Cisco-SIPTcl-Trunk
Content-Length: 0
```

```
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE
```

[diagram] Call-ID:[prev][next]
[5] SIP/2.0 180 Ringing

```
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK225b15ecd2
From: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
To: <sip:71234@10.10.199.158>
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
CSeq: 101 INVITE
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE
P-Asserted-Identity: <sip:called11@siptool11.cisco.com>
Remote-Party-ID:: <sip:called11@siptool11.cisco.com>;party=called;screen=yes;privacy=off
Contact: <sip:71234@10.10.199.158:5060;transport=tcp>
Allow-Events: kpml,dialog
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[6] SIP/2.0 180 Ringing

```
Via: SIP/2.0/Tcp 10.10.199.159:5060;branch=z9hG4bK2af-mytcltrunk--680267448-766
From: "from_ddaiker string" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
To: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
Date: Tue, 28 Jun 2011 18:06:42 GMT
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:called11@siptool11.cisco.com>
Remote-Party-ID: <sip:called11@siptool11.cisco.com>;party=called;screen=yes;privacy=off
Contact: <sip:71234@10.10.199.139:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[7] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK225b15ecd2
From: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
To: <sip:71234@10.10.199.158>;tag=mytcltrunk-680267402
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
CSeq: 101 INVITE
P-Asserted-Identity: <sip:called22@siptool22.cisco.com>
Server: Cisco-SIPTcl-Trunk
Contact: <sip:71234@10.10.199.158:5060;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Allow-Events: kpml
Content-Length: 294
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 12345 12345 IN IP4 10.10.199.158
s=SIPTcl Call-ddd
c=IN IP4 10.10.199.158
t=0 0
m=audio 16007 RTP/AVP 0 8 18 9 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:9 G722/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

[diagram] Call-ID:[prev][next]
[8] ACK sip:71234@10.10.199.158:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK23498c4008
From: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
To: <sip:71234@10.10.199.158>;tag=mytcltrunk-680267402
Date: Tue, 28 Jun 2011 18:06:42 GMT
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Type: application/sdp
Content-Length: 215

v=0
```

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```
o=CiscoSystemsCCM-SIP 139 1 IN IP4 10.10.199.139
s=SIP Call
c=IN IP4 10.10.199.159
t=0 0
m=audio 16007 RTP/AVP 9 101
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

[diagram] Call-ID:[prev][next]

[9] SIP/2.0 200 OK

```
Via: SIP/2.0/Tcp 10.10.199.159:5060;branch=z9hG4bK2af-mytcltrunk--680267448-766
From: "from_ddaiker string" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
To: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
Date: Tue, 28 Jun 2011 18:06:42 GMT
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence, kpml
Supported: replaces
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:called22@siptool22.cisco.com>
Remote-Party-ID: <sip:called22@siptool22.cisco.com>;party=called;screen=yes;privacy=off
Contact: <sip:71234@10.10.199.139:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 215

v=0
o=CiscoSystemsCCM-SIP 138 1 IN IP4 10.10.199.139
s=SIP Call
c=IN IP4 10.10.199.158
t=0 0
m=audio 16007 RTP/AVP 9 101
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

[diagram] Call-ID:[prev][next]

[10] SUBSCRIBE sip:71234@10.10.199.158:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK244aacaf8a
From: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
To: <sip:71234@10.10.199.158>;tag=mytcltrunk-680267402
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
CSeq: 102 SUBSCRIBE
Date: Tue, 28 Jun 2011 18:06:47 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 7200
Contact: <sip:10.10.199.139:5060;transport=tcp>
Accept: application/kpml-response+xml
Max-Forwards: 70
Content-Type: application/kpml-request+xml
Content-Length: 370

<?xml version="1.0" encoding="UTF-8" ?>
<kpml-request xmlns="urn:ietf:params:xml:ns:kpml-request" xmlns:xsi="http://www.w3.org/2001/XMLSchema-
instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-request kpml-request.xsd" version="1.0">

  <pattern interdigittimer="7260000" persist="persist">
    <regex tag="dtmf">[x*#ABCD]</regex>
  </pattern>

</kpml-request>
```

[diagram] Call-ID:[prev][next]

[11] SUBSCRIBE sip:cont_ddaiker@10.10.199.159:5060;transport=Tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK257dd1b08e
From: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
To: "from_ddaiker string" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 101 SUBSCRIBE
Date: Tue, 28 Jun 2011 18:06:47 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 7200
Contact: <sip:10.10.199.139:5060;transport=tcp>
```

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```
Accept: application/kpml-response+xml
Max-Forwards: 70
Content-Type: application/kpml-request+xml
Content-Length: 370

<?xml version="1.0" encoding="UTF-8" ?>
<kpml-request xmlns="urn:ietf:params:xml:ns:kpml-request" xmlns:xsi="http://www.w3.org/2001/XMLSchema-
instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-request kpml-request.xsd" version="1.0">

  <pattern interdigittimer="7260000" persist="persist">
    <regex tag="dtmf">[x*#ABCD]</regex>
  </pattern>

</kpml-request>

[diagram]   Call-ID:[prev][next]
[12] ACK sip:71234@10.10.199.139 SIP/2.0
Via: SIP/2.0/Tcp 10.10.199.159:5060;branch=z9hG4bK2af-mytcltrunk--680262369-161
From: "from_ddaiker string" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
To: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
Max-Forwards: 70
Cseq: 101 ACK
User-Agent: Cisco-SIPTcl-Trunk
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[13] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK244aacaf8a
From: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
To: <sip:71234@10.10.199.158>;tag=mytcltrunk-680267402
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
CSeq: 102 SUBSCRIBE
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Allow-Events: kpml
Supported: replaces
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[14] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK257dd1b08e
From: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
To: "from_ddaiker string" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 101 SUBSCRIBE
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Allow-Events: kpml
Supported: replaces
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[15] REFER sip:pai_ddaiker@10.10.199.139:5060;transport=tcp SIP/2.0
Via: SIP/2.0/Tcp 10.10.199.158:5060;branch=z9hG4bK2af-mytcltrunk--680257380-559
From: <sip:71234@10.10.199.158>;tag=mytcltrunk-680267402
To: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
CSeq: 501 REFER
Contact: <sip:71234@10.10.199.158:5060;transport=tcp>
Refer-To: <sip:siptool2.cisco.com:5050>
Referred-By: sip:71234@10.10.199.158:5060;transport=tcp
Max-Forwards: 70
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[16] REFER sip:cont_ddaiker@10.10.199.159:5060;transport=Tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK2664b59e6d
From: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
To: "from_ddaiker string" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 102 REFER
Max-Forwards: 70
Contact: <sip:71234@10.10.199.139:5060;transport=tcp>
User-Agent: Cisco-CUCM8.6
Referred-By: sip:71234@10.10.199.158:5060;transport=tcp
Refer-To: <sip:siptool2.cisco.com:5050>
Content-Length: 0
```

```
[diagram]   Call-ID:[prev][next]
[17] SIP/2.0 202 OK
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK2664b59e6d
From: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
To: "from_ddaiker string" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
Contact: <sip:emcc-rsvp@10.10.199.159:5060;transport=tcp;type=visiting;trunk=321781a6-30a8-dba5-fa79-
0bcba9cb8c40;node=0;instance=0>;
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 102 REFER
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[18] SIP/2.0 202 Accepted
Via: SIP/2.0/Tcp 10.10.199.158:5060;branch=z9hG4bK2af-mytcltrunk--680257380-559
From: <sip:71234@10.10.199.158>;tag=mytcltrunk-680267402
To: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
Date: Tue, 28 Jun 2011 18:06:52 GMT
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
CSeq: 501 REFER
Contact: <sip:pai_ddaiker@10.10.199.139:5060;transport=tcp>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[19] NOTIFY sip:10.10.199.139:5060;transport=tcp SIP/2.0
Via: SIP/2.0/tcp 10.10.199.159;branch=z9hG4bK2af-mytcltrunk--680252362-421
From: "display name" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
To: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 102 NOTIFY
Max-Forwards: 70
User-Agent: Cisco-SIPTcl-Trunk
Event: refer
Expires: 0
Contact: sip:cont_ddaiker@10.10.199.159:5060;transport=Tcp
Content-Type: message/sipfrag;version=2.0
Content-Length: 20

SIP/2.0 100 Trying

[diagram]   Call-ID:[prev][next]
[20] SIP/2.0 200 OK
Via: SIP/2.0/tcp 10.10.199.159;branch=z9hG4bK2af-mytcltrunk--680252362-421
From: "from_ddaiker string" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
To: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
Date: Tue, 28 Jun 2011 18:06:57 GMT
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 102 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[21] NOTIFY sip:71234@10.10.199.158:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK2744eb03ad
From: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
To: <sip:71234@10.10.199.158>;tag=mytcltrunk-680267402
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
CSeq: 103 NOTIFY
Max-Forwards: 70
Date: Tue, 28 Jun 2011 18:06:57 GMT
User-Agent: Cisco-CUCM8.6
Event: refer
Subscription-State: active
Contact: <sip:10.10.199.139:5060;transport=tcp>
Content-Type: message/sipfrag
Content-Length: 20

SIP/2.0 100 Trying

[diagram]   Call-ID:[prev][next]
[22] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK2744eb03ad
From: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
To: <sip:71234@10.10.199.158>;tag=mytcltrunk-680267402
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
CSeq: 103 NOTIFY
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Allow-Events: kpml
Supported: replaces
```

```
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[23] NOTIFY sip:10.10.199.139:5060;transport=tcp SIP/2.0
Via: SIP/2.0/tcp 10.10.199.159;branch=z9hG4bK2af-mytcltrunk--680247355-787
From: "display name" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
To: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 103 NOTIFY
Max-Forwards: 70
User-Agent: Cisco-SIPTcl-Trunk
Event: refer
Expires: 0
Contact: sip:cont_ddaiker@10.10.199.159:5060;transport=Tcp
Content-Type: message/sipfrag;version=2.0
Content-Length: 22

SIP/2.0 180 Alerting

[diagram]   Call-ID:[prev][next]
[24] NOTIFY sip:71234@10.10.199.158:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK28598e426d
From: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
To: <sip:71234@10.10.199.158>;tag=mytcltrunk-680267402
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
CSeq: 104 NOTIFY
Max-Forwards: 70
Date: Tue, 28 Jun 2011 18:07:02 GMT
User-Agent: Cisco-CUCM8.6
Event: refer
Subscription-State: active
Contact: <sip:10.10.199.139:5060;transport=tcp>
Content-Type: message/sipfrag
Content-Length: 22

SIP/2.0 180 Alerting

[diagram]   Call-ID:[prev][next]
[25] SIP/2.0 200 OK
Via: SIP/2.0/tcp 10.10.199.159;branch=z9hG4bK2af-mytcltrunk--680247355-787
From: "from_ddaiker string" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
To: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
Date: Tue, 28 Jun 2011 18:07:02 GMT
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 103 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[26] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK28598e426d
From: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
To: <sip:71234@10.10.199.158>;tag=mytcltrunk-680267402
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
CSeq: 104 NOTIFY
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Allow-Events: kpml
Supported: replaces
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[27] NOTIFY sip:10.10.199.139:5060;transport=tcp SIP/2.0
Via: SIP/2.0/tcp 10.10.199.159;branch=z9hG4bK2af-mytcltrunk--680242349-208
From: "display name" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
To: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 104 NOTIFY
Max-Forwards: 70
User-Agent: Cisco-SIPTcl-Trunk
Event: refer
Expires: 0
Contact: sip:cont_ddaiker@10.10.199.159:5060;transport=Tcp
Content-Type: message/sipfrag;version=2.0
Content-Length: 16

SIP/2.0 200 OK

[diagram]   Call-ID:[prev][next]
[28] NOTIFY sip:71234@10.10.199.158:5060;transport=tcp SIP/2.0
```

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```
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK2927288ecd
From: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
To: <sip:71234@10.10.199.158>;tag=mytcltrunk-680267402
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
CSeq: 105 NOTIFY
Max-Forwards: 70
Date: Tue, 28 Jun 2011 18:07:07 GMT
User-Agent: Cisco-CUCM8.6
Event: refer
Subscription-State: active
Contact: <sip:10.10.199.139:5060;transport=tcp>
Content-Type: message/sipfrag
Content-Length: 16

SIP/2.0 200 OK

[diagram]   Call-ID:[prev][next]
[29] SIP/2.0 200 OK
Via: SIP/2.0/tcp 10.10.199.159;branch=z9hG4bK2af-mytcltrunk--680242349-208
From: "from_ddaiker string" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
To: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
Date: Tue, 28 Jun 2011 18:07:07 GMT
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 104 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[30] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK2927288ecd
From: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
To: <sip:71234@10.10.199.158>;tag=mytcltrunk-680267402
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
CSeq: 105 NOTIFY
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Allow-Events: kpml
Supported: replaces
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[31] BYE sip:71234@10.10.199.139 SIP/2.0
Via: SIP/2.0/Tcp 10.10.199.159:5060;branch=z9hG4bK2af-mytcltrunk--680217362-939
From: "display name" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
To: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
Max-Forwards: 70
CSeq: 105 BYE
User-Agent: Cisco-SIPTcl-Trunk
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[32] BYE sip:71234@10.10.199.158:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK2b34dcb2a1
From: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
To: <sip:71234@10.10.199.158>;tag=mytcltrunk-680267402
Date: Tue, 28 Jun 2011 18:06:52 GMT
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 107 BYE
Reason: Q.850;cause=16
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[33] SUBSCRIBE sip:71234@10.10.199.158:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK2a60514b7c
From: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
To: <sip:71234@10.10.199.158>;tag=mytcltrunk-680267402
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
CSeq: 106 SUBSCRIBE
Date: Tue, 28 Jun 2011 18:07:32 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 0
Contact: <sip:10.10.199.139:5060;transport=tcp>
Max-Forwards: 70
Content-Length: 0
```

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```
[diagram]   Call-ID:[prev][next]
[34] SIP/2.0 200 OK
Via: SIP/2.0/Tcp 10.10.199.159:5060;branch=z9hG4bK2af-mytcltrunk--680217362-939
From: "from_ddaiker string" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
To: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
Date: Tue, 28 Jun 2011 18:07:32 GMT
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 105 BYE
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[35] SUBSCRIBE sip:cont_ddaiker@10.10.199.159:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.199.139:5060;branch=z9hG4bK2c4f219706
From: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
To: "from_ddaiker string" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 103 SUBSCRIBE
Date: Tue, 28 Jun 2011 18:07:32 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 0
Contact: <sip:10.10.199.139:5060>
Max-Forwards: 70
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[36] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK2a60514b7c
From: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
To: <sip:71234@10.10.199.158>;tag=mytcltrunk-680267402
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
CSeq: 106 SUBSCRIBE
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Allow-Events: kpml
Supported: replaces
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[37] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.199.139:5060;branch=z9hG4bK2b34dcb2a1
From: "pai_ddaiker string" <sip:pai_ddaiker@10.10.199.159>;tag=139~27b43f2e-fe79-4e39-bfa4-36ed154df44f-
19409918
To: <sip:71234@10.10.199.158>;tag=mytcltrunk-680267402
Call-ID: 5f32aa80-e0a11832-f-8bc712ac@10.10.199.139
CSeq: 107 BYE
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[38] SUBSCRIBE sip:cont_ddaiker@10.10.199.159:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.199.139:5060;branch=z9hG4bK2c4f219706
From: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
To: "from_ddaiker string" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 103 SUBSCRIBE
Date: Tue, 28 Jun 2011 18:07:32 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 0
Contact: <sip:10.10.199.139:5060>
Max-Forwards: 70
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[39] SUBSCRIBE sip:cont_ddaiker@10.10.199.159:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.199.139:5060;branch=z9hG4bK2c4f219706
From: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
To: "from_ddaiker string" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 103 SUBSCRIBE
Date: Tue, 28 Jun 2011 18:07:32 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 0
Contact: <sip:10.10.199.139:5060>
Max-Forwards: 70
Content-Length: 0
```

```
[diagram]   Call-ID:[prev][next]
[40] SUBSCRIBE sip:cont_ddaiker@10.10.199.159:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.199.139:5060;branch=z9hG4bK2c4f219706
From: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
To: "from_ddaiker string" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 103 SUBSCRIBE
Date: Tue, 28 Jun 2011 18:07:32 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 0
Contact: <sip:10.10.199.139:5060>
Max-Forwards: 70
Content-Length: 0
```

```
[diagram]   Call-ID:[prev][next]
[41] SUBSCRIBE sip:cont_ddaiker@10.10.199.159:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.199.139:5060;branch=z9hG4bK2c4f219706
From: <sip:71234@10.10.199.139>;tag=138~27b43f2e-fe79-4e39-bfa4-36ed154df44f-19409917
To: "from_ddaiker string" <sip:ddaiker@10.10.199.159>;tag=mytcltrunk-680267449
Call-ID: mytcltru-nk-ddaiker@10.10.199.159--680267454
CSeq: 103 SUBSCRIBE
Date: Tue, 28 Jun 2011 18:07:32 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 0
Contact: <sip:10.10.199.139:5060>
Max-Forwards: 70
Content-Length: 0
```

6. CUCM Video - SIP Video Encryption

6.1 Basic Call from Encrypted E20 to another Encrypted E20

Title: Basic Call from Encrypted E20 to another Encrypted E20

Description:

The following call flow illustrates the SIP messaging that takes place between two E20 in encrypted mode with Audio , Video and FECC line encrypted. The endpoints are across SIP trunks and not in native mode.

Configuration:

Node = Unified CM1, IP = 10.10.13.50
Node = Unified CM2, IP = 10.10.31.104

Phone = A, IP = 10.10.31.107, Model = SIP
Phone = B, Line = 500124, IP = 10.10.31.104, Model = SIP

SIP Trunk between to reach Phone B from Phone A is via 0.5

Scenario:

Phone A calls Phone B DN 500124 over the SIP Trunk by dialing 0500124
Phone B DN 500124 Answers
Phone B DN 500124 goes onHook

End of Scenario

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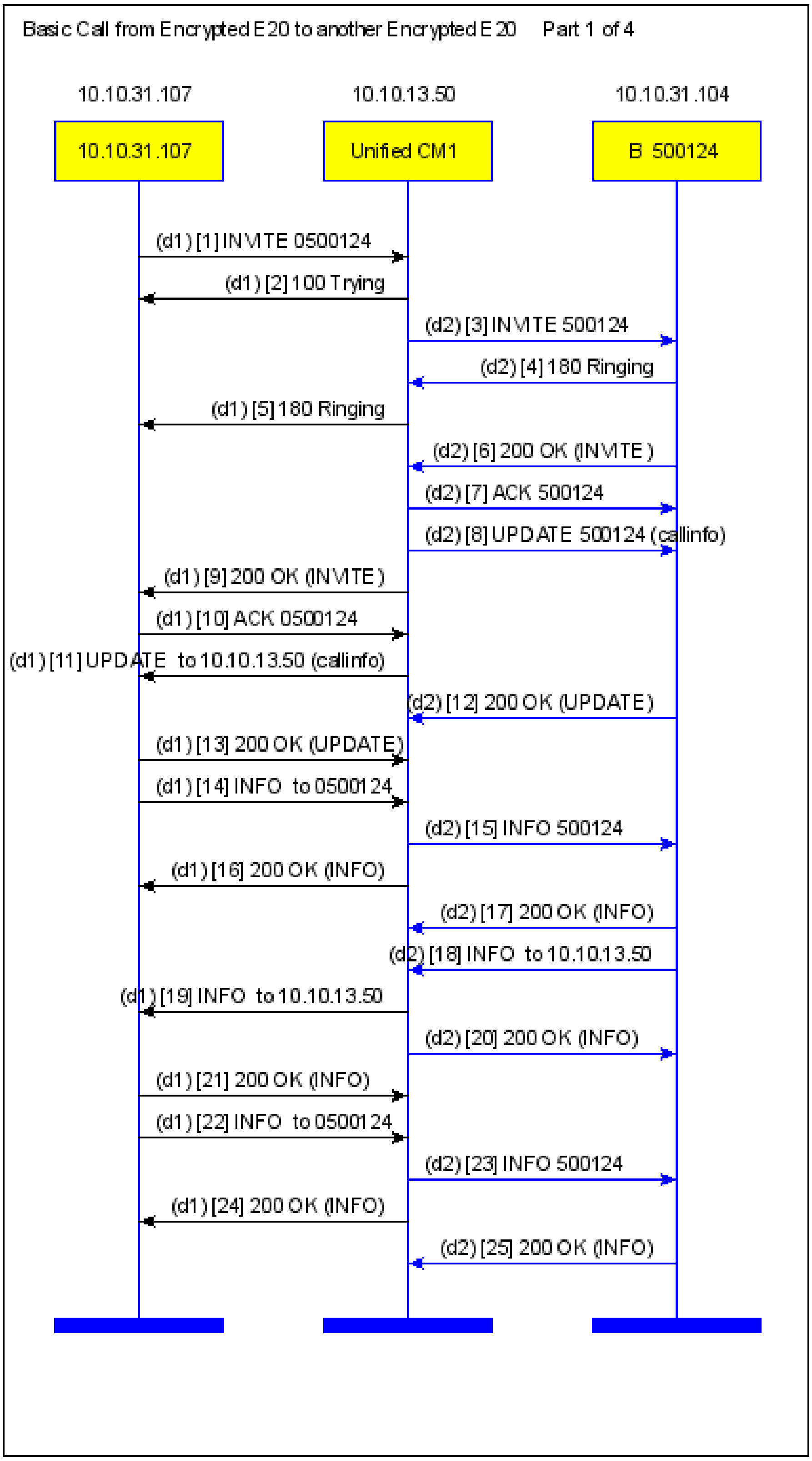

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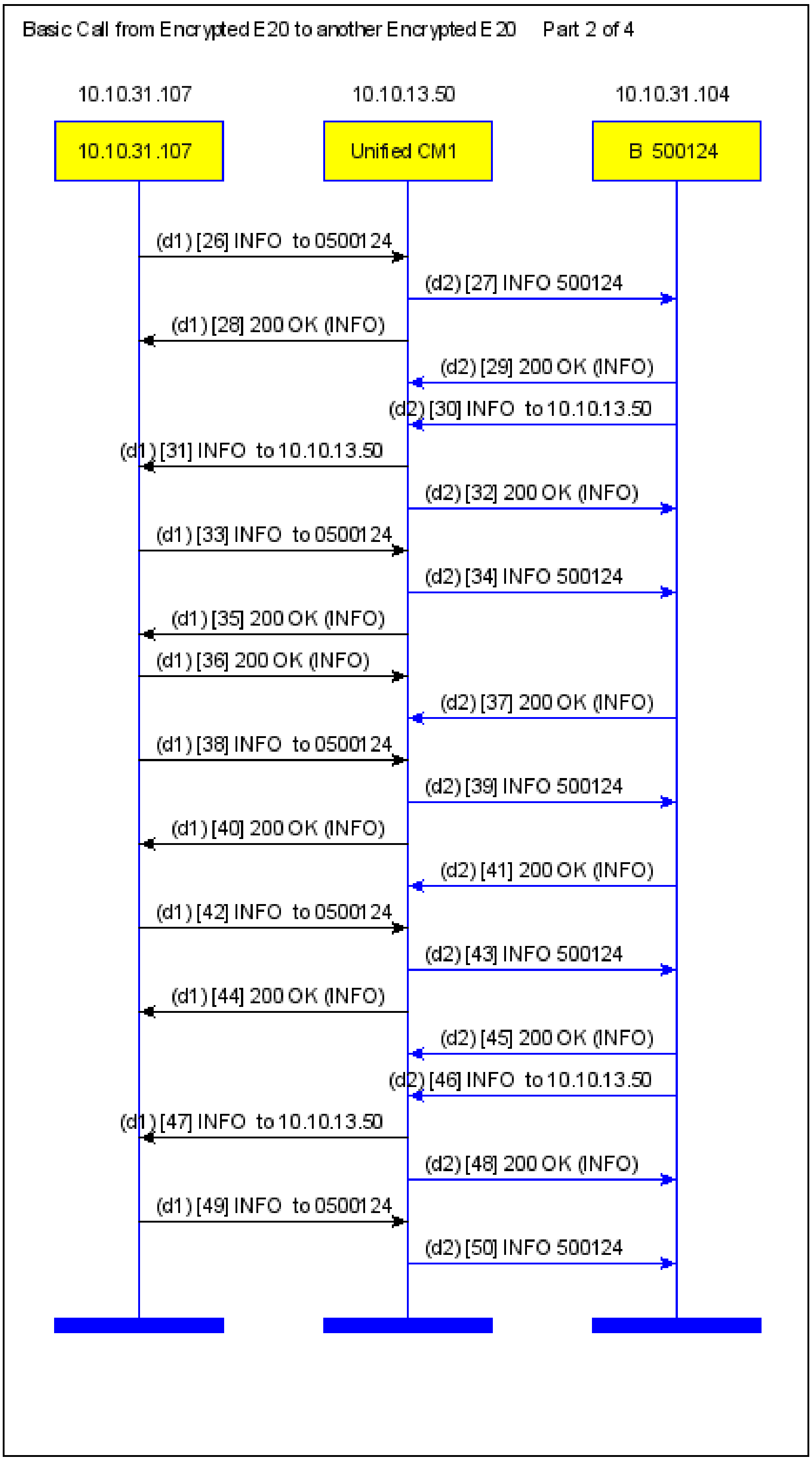
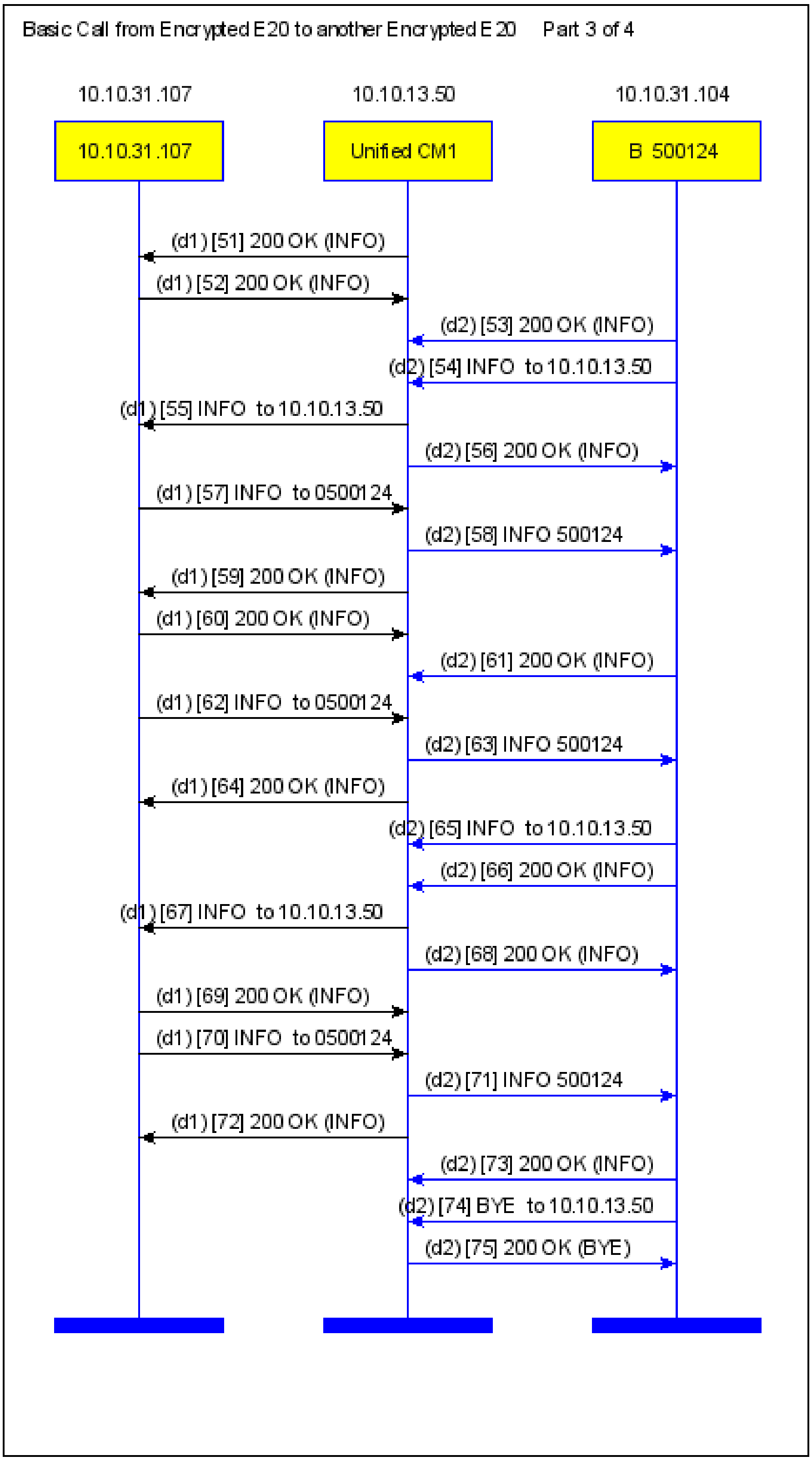
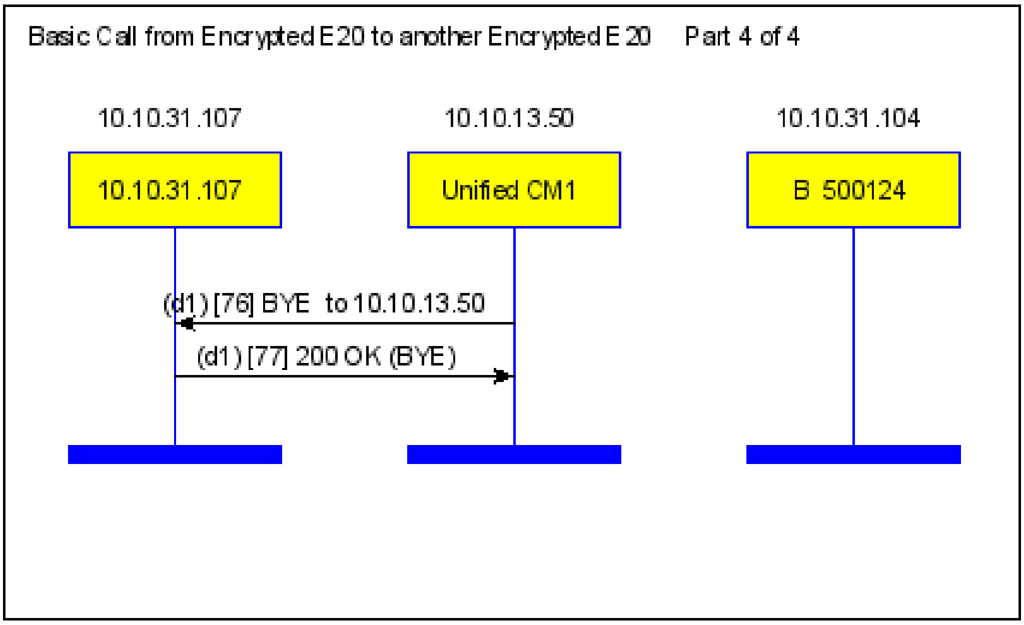



Basic Call from Encrypted E20 to another Encrypted E20 Part 3 of 4
10.10.31.107
10.10.13.50
10.10.31.104
10.10.31.107
Unified CM1
B 500124
(d1) [51] 200 OK (INFO)
(d1) [52] 200 OK (INFO)
(d2) [53] 200 OK (INFO)
(d2) [54] INFO to 10.10.13.50
(d1) [55] INFO to 10.10.13.50
(d2) [56] 200 OK (INFO)
(d1) [57] INFO to 0500124
(d2) [58] INFO 500124
(d1) [59] 200 OK (INFO)
(d1) [60] 200 OK (INFO)
(d2) [61] 200 OK (INFO)
(d1) [62] INFO to 0500124
(d2) [63] INFO 500124
(d1) [64] 200 OK (INFO)
(d2) [65] INFO to 10.10.13.50
(d2) [66] 200 OK (INFO)
(d1) [67] INFO to 10.10.13.50
(d2) [68] 200 OK (INFO)
(d1) [69] 200 OK (INFO)
(d1) [70] INFO to 0500124
(d2) [71] INFO 500124
(d1) [72] 200 OK (INFO)
(d2) [73] 200 OK (INFO)
(d2) [74] BYE to 10.10.13.50
(d2) [75] 200 OK (BYE)


Basic Call from Encrypted E20 to another Encrypted E20 Part 4 of 4
10.10.31.107
10.10.13.50
10.10.31.104
10.10.31.107
Unified CM1
B 500124
(d1) [76] BYE to 10.10.13.50
(d1) [77] 200 OK (BYE)

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```
[diagram]   Call-ID:[prev][next]
[1] INVITE sip:0500124@10.10.13.50 SIP/2.0
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bK105e878ca7202ccd7225a7ba91ebc1e1.1;rport
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 100 INVITE
Contact: <sip:10.10.31.107:5061;transport=tls>
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>
Max-Forwards: 70
Route: <sip:10.10.13.50;lr>
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Session-Expires: 1800
Content-Type: application/sdp
Content-Length: 2555

v=0
o=tandberg 8 1 IN IP4 10.10.31.107
s=-
c=IN IP4 10.10.31.107
b=AS:1152
t=0 0
m=audio 16420 RTP/SAVP 100 101 102 9 18 11 8 0 103
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:101 G7221/16000
a=fmtp:101 bitrate=32000
a=rtpmap:102 G7221/16000
a=fmtp:102 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:11 L16/16000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=sendrecv
m=video 16422 RTP/SAVP 97 98 99 34 31
b=TIAS:1152000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;packetization-mode=1
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1024,768,4;custom=1024,576,4;custom=800,600,4;cif4=2;custom=720,480,2;custom=640,480,2;custom=512,288
,1;cif=1;custom=352,240,1;qcif=1;maxbr=10880
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=2;cif=1;qcif=1;maxbr=10880
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=10880
a=rtcp-fb:* nack pli
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=sendrecv
a=content:main
a=label:11
a=answer:full
m=application 5070 UDP/BFCP *
a=floorctrl:c-s
a=confid:1
a=floorid:2 mstrm:12
a=userid:8
a=setup:passive
a=connection:new
m=video 16424 RTP/SAVP 97 98 99 34 31
b=TIAS:1152000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;packetization-mode=1
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1024,768,4;custom=1024,576,4;custom=800,600,4;cif4=2;custom=720,480,2;custom=640,480,2;custom=512,288
,1;cif=1;custom=352,240,1;qcif=1;maxbr=10880
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=2;cif=1;qcif=1;maxbr=10880
```

```
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=10880
a=rtcp-fb:* nack pli
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=sendrecv
a=content:slides
a=label:12
m=application 16426 RTP/SAVP 104
a=rtpmap:104 H224/4800
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=sendrecv
```

[diagram] Call-ID:[prev][next]
[2] SIP/2.0 100 Trying

```
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bK105e878ca7202ccd7225a7ba91ebc1e1.1;rport
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>
Date: Tue, 12 Apr 2011 22:55:57 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 100 INVITE
Allow-Events: presence
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[3] INVITE sip:500124@10.10.31.104:5061 SIP/2.0

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bKcc879773
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>
Date: Tue, 12 Apr 2011 22:55:57 GMT
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM8.6
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Cisco-Guid: 0096911872-0000065536-0000000004-0839719852
Session-Expires:  1800
P-Asserted-Identity: <sip:10.10.13.50>
Remote-Party-ID: <sip:10.10.13.50>;party=calling;screen=yes;privacy=off
Contact: <sip:10.10.13.50:5061;transport=tls>;video;audio
Max-Forwards: 69
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[4] SIP/2.0 180 Ringing

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bKcc879773;received=10.10.13.50
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 101 INVITE
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[5] SIP/2.0 180 Ringing

```
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bK105e878ca7202ccd7225a7ba91ebc1e1.1;rport
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Date: Tue, 12 Apr 2011 22:55:57 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:500124@10.10.13.50>
Remote-Party-ID: <sip:500124@10.10.13.50>;party=called;screen=yes;privacy=off
Contact: <sip:0500124@10.10.13.50:5061;transport=tls>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[6] SIP/2.0 200 OK

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bKcc879773;received=10.10.13.50
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 101 INVITE
Contact: <sip:500124@10.10.31.104:5061;transport=tls>
```

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```
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Require: timer
Session-Expires: 1800;refresher=uas
Min-SE: 1800
Content-Type: application/sdp
Content-Length: 2555

v=0
o=tandberg 6 1 IN IP4 10.10.31.104
s=-
c=IN IP4 10.10.31.104
b=AS:1152
t=0 0
m=audio 16412 RTP/SAVP 100 101 102 9 18 11 8 0 103
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:101 G7221/16000
a=fmtp:101 bitrate=32000
a=rtpmap:102 G7221/16000
a=fmtp:102 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:11 L16/16000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=sendrecv
m=video 16414 RTP/SAVP 97 98 99 34 31
b=TIAS:1152000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;packetization-mode=1
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1024,768,4;custom=1024,576,4;custom=800,600,4;cif4=2;custom=720,480,2;custom=640,480,2;custom=512,288
,1;cif=1;custom=352,240,1;qcif=1;maxbr=10880
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=2;cif=1;qcif=1;maxbr=10880
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=10880
a=rtcp-fb:* nack pli
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=sendrecv
a=content:main
a=label:11
a=answer:full
m=application 5070 UDP/BFCP *
a=floorctrl:c-s
a=confid:1
a=floorid:2 mstrm:12
a=userid:6
a=setup:passive
a=connection:new
m=video 16416 RTP/SAVP 97 98 99 34 31
b=TIAS:1152000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;packetization-mode=1
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1024,768,4;custom=1024,576,4;custom=800,600,4;cif4=2;custom=720,480,2;custom=640,480,2;custom=512,288
,1;cif=1;custom=352,240,1;qcif=1;maxbr=10880
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=2;cif=1;qcif=1;maxbr=10880
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=10880
a=rtcp-fb:* nack pli
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=sendrecv
a=content:slides
```

```
a=label:12
m=application 16418 RTP/SAVP 104
a=rtpmap:104 H224/4800
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=sendrecv
```

[diagram] Call-ID:[prev][next]

[7] ACK sip:500124@10.10.31.104:5061;transport=tls SIP/2.0

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bKd7438e282
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Date: Tue, 12 Apr 2011 22:55:57 GMT
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 1423

v=0
o=CiscoSystemsCCM-SIP 519 1 IN IP4 10.10.13.50
s=SIP Call
b=AS:384
t=0 0
m=audio 16420 RTP/SAVP 100 103
c=IN IP4 10.10.31.107
b=TIAS:64000
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 bitrate=64000;profile-level-id=24;object=23
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
m=video 16422 RTP/SAVP 98
c=IN IP4 10.10.31.107
b=TIAS:320000
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=label:11
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-
smbps=40500
a=content:main
a=rtcp-fb:* nack pli
m=application 5070 UDP/BFCP *
c=IN IP4 10.10.31.107
a=floorctrl:s-only
a=floorid:2 mstrm:12
a=confid:1
a=userid:8
m=video 16424 RTP/SAVP 98
c=IN IP4 10.10.31.107
b=TIAS:320000
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=label:12
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-
smbps=40500
a=content:slides
a=rtcp-fb:* nack pli
m=application 16426 RTP/SAVP 104
c=IN IP4 10.10.31.107
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:104 H224/0
```

[diagram] Call-ID:[prev][next]

[8] UPDATE sip:500124@10.10.31.104:5061;transport=tls SIP/2.0

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bKe2e713bd9
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Date: Tue, 12 Apr 2011 22:55:57 GMT
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
Supported: timer,resource-priority,replaces
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 102 UPDATE
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= Encrypted; gci= 1-277002
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:10.10.13.50>
Remote-Party-ID: <sip:10.10.13.50>;party=calling;screen=yes;privacy=off
```

```
Contact: <sip:10.10.13.50:5061;transport=tls>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[9] SIP/2.0 200 OK

```
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bK105e878ca7202ccd7225a7ba91ebc1e1.1;rport
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Date: Tue, 12 Apr 2011 22:55:57 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Session-Expires:  1800;refresher=uas
Require:  timer
P-Asserted-Identity: <sip:500124@10.10.13.50>
Remote-Party-ID: <sip:500124@10.10.13.50>;party=called;screen=yes;privacy=off
Contact: <sip:0500124@10.10.13.50:5061;transport=tls>
Content-Type: application/sdp
Content-Length: 1423

v=0
o=CiscoSystemsCCM-SIP 518 1 IN IP4 10.10.13.50
s=SIP Call
b=AS:384
t=0 0
m=audio 16412 RTP/SAVP 100 103
c=IN IP4 10.10.31.104
b=TIAS:64000
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 bitrate=64000;profile-level-id=24;object=23
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
m=video 16414 RTP/SAVP 98
c=IN IP4 10.10.31.104
b=TIAS:320000
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=label:11
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-
smbps=40500
a=content:main
a=rtcp-fb:* nack pli
m=application 5070 UDP/BFCP *
c=IN IP4 10.10.31.104
a=floorctrl:c-only
a=floorid:2 mstrm:12
a=confid:1
a=userid:6
m=video 16416 RTP/SAVP 98
c=IN IP4 10.10.31.104
b=TIAS:320000
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=label:12
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-
smbps=40500
a=content:slides
a=rtcp-fb:* nack pli
m=application 16418 RTP/SAVP 104
c=IN IP4 10.10.31.104
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:104 H224/0
```

[diagram] Call-ID:[prev][next]

[10] ACK sip:0500124@10.10.13.50:5061;transport=tls SIP/2.0

```
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bK5db97c2c8873c3e3bdbce662947b1e34.1;rport
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 100 ACK
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Length: 0
```

```
[diagram]   Call-ID:[prev][next]
[11] UPDATE sip:10.10.31.107:5061;transport=tls SIP/2.0
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bKf3eebaf9a
From: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
To: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
Date: Tue, 12 Apr 2011 22:55:58 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
Supported: timer,resource-priority,replaces
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 UPDATE
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= Encrypted; gci= 1-277002
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:500124@10.10.13.50>
Remote-Party-ID: <sip:500124@10.10.13.50>;party=calling;screen=yes;privacy=off
Contact: <sip:10.10.13.50:5061;transport=tls>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[12] SIP/2.0 200 OK
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bKe2e713bd9;received=10.10.13.50
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 102 UPDATE
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Require: timer
Session-Expires: 1800;refresher=uas
Min-SE: 1800
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[13] SIP/2.0 200 OK
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bKf3eebaf9a;received=10.10.13.50
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 101 UPDATE
From: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
To: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Session-Expires: 1800;refresher=uas
Min-SE: 90
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[14] INFO sip:10.10.13.50:5061;transport=tls SIP/2.0
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bKfb3e637975717f0d43e6a157c3fb54de.1;rport
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 101 INFO
Contact: <sip:10.10.31.107:5061>
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Type: application/media_control+xml
Content-Length: 193

<?xml version="1.0" encoding="utf-
8"?><media_control><vc_primitive><to_encoder><picture_fast_update></picture_fast_update></to_encoder><stream
_id>11</stream_id></vc_primitive></media_control>

[diagram]   Call-ID:[prev][next]
[15] INFO sip:500124@10.10.31.104:5061;transport=tls SIP/2.0
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK104659ebf0
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Date: Tue, 12 Apr 2011 22:55:57 GMT
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 103 INFO
Contact: <sip:10.10.13.50:5061;transport=tls>
```

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```
Content-Type: application/media_control+xml
Content-Length: 190

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<media_control>

  <vc_primitive>
    <to_encoder>
      <picture_fast_update/>
    </to_encoder>
  </vc_primitive>

</media_control>

[diagram]   Call-ID:[prev][next]
[16] SIP/2.0 200 OK
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bKfb3e637975717f0d43e6a157c3fb54de.1;rport
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Date: Tue, 12 Apr 2011 22:56:01 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 101 INFO
Contact: <sip:0500124@10.10.13.50:5061;transport=tls>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[17] SIP/2.0 200 OK
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK104659ebf0;received=10.10.13.50
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 103 INFO
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[18] INFO sip:10.10.13.50:5061;transport=tls SIP/2.0
Via: SIP/2.0/TLS 10.10.31.104:5061;branch=z9hG4bK9ed2d2b47a6c3324e606bda1e2a3c376.1;rport
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 501 INFO
Contact: <sip:500124@10.10.31.104:5061>
From: "" <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
To: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Type: application/media_control+xml
Content-Length: 193

<?xml version="1.0" encoding="utf-
8"?><media_control><vc_primitive><to_encoder><picture_fast_update></picture_fast_update></to_encoder><stream
_id>11</stream_id></vc_primitive></media_control>

[diagram]   Call-ID:[prev][next]
[19] INFO sip:10.10.31.107:5061;transport=tls SIP/2.0
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK112262a13c
From: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
To: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
Date: Tue, 12 Apr 2011 22:56:01 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 102 INFO
Contact: <sip:0500124@10.10.13.50:5061;transport=tls>
Content-Type: application/media_control+xml
Content-Length: 190

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<media_control>

  <vc_primitive>
    <to_encoder>
      <picture_fast_update/>
    </to_encoder>
  </vc_primitive>

</media_control>
```

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[diagram] Call-ID:[prev][next]
[20] SIP/2.0 200 OK
```
Via: SIP/2.0/TLS 10.10.31.104:5061;branch=z9hG4bK9ed2d2b47a6c3324e606bda1e2a3c376.1;rport
From: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
To: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
Date: Tue, 12 Apr 2011 22:56:01 GMT
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 501 INFO
Contact: <sip:10.10.13.50:5061;transport=tls>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[21] SIP/2.0 200 OK
```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK112262a13c;received=10.10.13.50
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 102 INFO
From: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
To: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[22] INFO sip:10.10.13.50:5061;transport=tls SIP/2.0
```
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bKa511fa50d23a28bf72983e1f59b2fdf0.1;rport
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 102 INFO
Contact: <sip:10.10.31.107:5061>
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Type: application/media_control+xml
Content-Length: 193

<?xml version="1.0" encoding="utf-
8"?><media_control><vc_primitive><to_encoder><picture_fast_update></picture_fast_update></to_encoder><stream
_id>11</stream_id></vc_primitive></media_control>
```

[diagram] Call-ID:[prev][next]
[23] INFO sip:500124@10.10.31.104:5061;transport=tls SIP/2.0
```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK125f4989a6
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Date: Tue, 12 Apr 2011 22:56:01 GMT
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 104 INFO
Contact: <sip:10.10.13.50:5061;transport=tls>
Content-Type: application/media_control+xml
Content-Length: 190

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<media_control>

  <vc_primitive>
    <to_encoder>
      <picture_fast_update/>
    </to_encoder>
  </vc_primitive>

</media_control>
```

[diagram] Call-ID:[prev][next]
[24] SIP/2.0 200 OK
```
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bKa511fa50d23a28bf72983e1f59b2fdf0.1;rport
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Date: Tue, 12 Apr 2011 22:56:01 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 102 INFO
Contact: <sip:0500124@10.10.13.50:5061;transport=tls>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[25] SIP/2.0 200 OK

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK125f4989a6;received=10.10.13.50
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 104 INFO
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[26] INFO sip:10.10.13.50:5061;transport=tls SIP/2.0

```
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bKa7aa7d146c7f6d476cfd9ccc5a51ae1f.1;rport
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 103 INFO
Contact: <sip:10.10.31.107:5061>
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Type: application/media_control+xml
Content-Length: 193

<?xml version="1.0" encoding="utf-
8"?><media_control><vc_primitive><to_encoder><picture_fast_update></picture_fast_update></to_encoder><stream
_id>11</stream_id></vc_primitive></media_control>
```

[diagram] Call-ID:[prev][next]

[27] INFO sip:500124@10.10.31.104:5061;transport=tls SIP/2.0

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK136d223d46
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Date: Tue, 12 Apr 2011 22:56:01 GMT
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 105 INFO
Contact: <sip:10.10.13.50:5061;transport=tls>
Content-Type: application/media_control+xml
Content-Length: 190

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<media_control>

  <vc_primitive>
    <to_encoder>
      <picture_fast_update/>
    </to_encoder>
  </vc_primitive>

</media_control>
```

[diagram] Call-ID:[prev][next]

[28] SIP/2.0 200 OK

```
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bKa7aa7d146c7f6d476cfd9ccc5a51ae1f.1;rport
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Date: Tue, 12 Apr 2011 22:56:02 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 103 INFO
Contact: <sip:0500124@10.10.13.50:5061;transport=tls>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[29] SIP/2.0 200 OK

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK136d223d46;received=10.10.13.50
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 105 INFO
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[30] INFO sip:10.10.13.50:5061;transport=tls SIP/2.0
```
Via: SIP/2.0/TLS 10.10.31.104:5061;branch=z9hG4bK3bb3befe2e490147827b44e4bb8bc939.1;rport
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 502 INFO
Contact: <sip:500124@10.10.31.104:5061>
From: "" <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
To: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Type: application/media_control+xml
Content-Length: 193

<?xml version="1.0" encoding="utf-
8"?><media_control><vc_primitive><to_encoder><picture_fast_update></picture_fast_update></to_encoder><stream
_id>11</stream_id></vc_primitive></media_control>
```

[diagram] Call-ID:[prev][next]
[31] INFO sip:10.10.31.107:5061;transport=tls SIP/2.0
```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK1418b75bbb
From: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
To: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
Date: Tue, 12 Apr 2011 22:56:02 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 103 INFO
Contact: <sip:0500124@10.10.13.50:5061;transport=tls>
Content-Type: application/media_control+xml
Content-Length: 190

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<media_control>

  <vc_primitive>
    <to_encoder>
      <picture_fast_update/>
    </to_encoder>
  </vc_primitive>

</media_control>
```

[diagram] Call-ID:[prev][next]
[32] SIP/2.0 200 OK
```
Via: SIP/2.0/TLS 10.10.31.104:5061;branch=z9hG4bK3bb3befe2e490147827b44e4bb8bc939.1;rport
From: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
To: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
Date: Tue, 12 Apr 2011 22:56:03 GMT
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 502 INFO
Contact: <sip:10.10.13.50:5061;transport=tls>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[33] INFO sip:10.10.13.50:5061;transport=tls SIP/2.0
```
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bKf0f3e1e6f296294299d37b5bb575bbe7.1;rport
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 104 INFO
Contact: <sip:10.10.31.107:5061>
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Type: application/media_control+xml
Content-Length: 193

<?xml version="1.0" encoding="utf-
8"?><media_control><vc_primitive><to_encoder><picture_fast_update></picture_fast_update></to_encoder><stream
_id>11</stream_id></vc_primitive></media_control>
```

[diagram] Call-ID:[prev][next]
[34] INFO sip:500124@10.10.31.104:5061;transport=tls SIP/2.0
```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK1512ed5d20
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Date: Tue, 12 Apr 2011 22:56:03 GMT
```

```
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 106 INFO
Contact: <sip:10.10.13.50:5061;transport=tls>
Content-Type: application/media_control+xml
Content-Length: 190

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<media_control>

  <vc_primitive>
    <to_encoder>
      <picture_fast_update/>
    </to_encoder>
  </vc_primitive>

</media_control>
```

[diagram] Call-ID:[prev][next]
[35] SIP/2.0 200 OK

```
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bKf0f3e1e6f296294299d37b5bb575bbe7.1;rport
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Date: Tue, 12 Apr 2011 22:56:03 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 104 INFO
Contact: <sip:0500124@10.10.13.50:5061;transport=tls>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[36] SIP/2.0 200 OK

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK1418b75bbb;received=10.10.13.50
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 103 INFO
From: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
To: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[37] SIP/2.0 200 OK

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK1512ed5d20;received=10.10.13.50
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 106 INFO
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[38] INFO sip:10.10.13.50:5061;transport=tls SIP/2.0

```
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bKec17f873edfe3ad8d14cc04993c51d7c.1;rport
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 105 INFO
Contact: <sip:10.10.31.107:5061>
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Type: application/media_control+xml
Content-Length: 193

<?xml version="1.0" encoding="utf-
8"?><media_control><vc_primitive><to_encoder><picture_fast_update></picture_fast_update></to_encoder><stream
_id>11</stream_id></vc_primitive></media_control>
```

[diagram] Call-ID:[prev][next]
[39] INFO sip:500124@10.10.31.104:5061;transport=tls SIP/2.0

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK161669f146
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Date: Tue, 12 Apr 2011 22:56:03 GMT
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
```

```
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 107 INFO
Contact: <sip:10.10.13.50:5061;transport=tls>
Content-Type: application/media_control+xml
Content-Length: 190

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<media_control>

  <vc_primitive>
    <to_encoder>
      <picture_fast_update/>
    </to_encoder>
  </vc_primitive>

</media_control>
```

[diagram]   Call-ID:[prev][next]
[40] SIP/2.0 200 OK

```
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bKec17f873edfe3ad8d14cc04993c51d7c.1;rport
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Date: Tue, 12 Apr 2011 22:56:03 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 105 INFO
Contact: <sip:0500124@10.10.13.50:5061;transport=tls>
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]
[41] SIP/2.0 200 OK

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK161669f146;received=10.10.13.50
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 107 INFO
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]
[42] INFO sip:10.10.13.50:5061;transport=tls SIP/2.0

```
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bK727ade50e485bfbc19382286f4057b4e.1;rport
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 106 INFO
Contact: <sip:10.10.31.107:5061>
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Type: application/media_control+xml
Content-Length: 193

<?xml version="1.0" encoding="utf-
8"?><media_control><vc_primitive><to_encoder><picture_fast_update></picture_fast_update></to_encoder><stream
_id>11</stream_id></vc_primitive></media_control>
```

[diagram]   Call-ID:[prev][next]
[43] INFO sip:500124@10.10.31.104:5061;transport=tls SIP/2.0

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK173bfee5ec
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Date: Tue, 12 Apr 2011 22:56:03 GMT
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 108 INFO
Contact: <sip:10.10.13.50:5061;transport=tls>
Content-Type: application/media_control+xml
Content-Length: 190

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<media_control>

  <vc_primitive>
    <to_encoder>
      <picture_fast_update/>
```

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```
    </to_encoder>
  </vc_primitive>

</media_control>
```

[diagram] Call-ID:[prev][next]

[44] SIP/2.0 200 OK

```
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bK727ade50e485bfbc19382286f4057b4e.1;rport
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Date: Tue, 12 Apr 2011 22:56:07 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 106 INFO
Contact: <sip:0500124@10.10.13.50:5061;transport=tls>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[45] SIP/2.0 200 OK

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK173bfee5ec;received=10.10.13.50
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 108 INFO
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[46] INFO sip:10.10.13.50:5061;transport=tls SIP/2.0

```
Via: SIP/2.0/TLS 10.10.31.104:5061;branch=z9hG4bKbb22af198e4a46db3b9570011c564b12.1;rport
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 503 INFO
Contact: <sip:500124@10.10.31.104:5061>
From: "" <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
To: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Type: application/media_control+xml
Content-Length: 193

<?xml version="1.0" encoding="utf-
8"?><media_control><vc_primitive><to_encoder><picture_fast_update></picture_fast_update></to_encoder><stream
_id>11</stream_id></vc_primitive></media_control>
```

[diagram] Call-ID:[prev][next]

[47] INFO sip:10.10.31.107:5061;transport=tls SIP/2.0

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK186002d026
From: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
To: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
Date: Tue, 12 Apr 2011 22:56:07 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 104 INFO
Contact: <sip:0500124@10.10.13.50:5061;transport=tls>
Content-Type: application/media_control+xml
Content-Length: 190

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<media_control>

  <vc_primitive>
    <to_encoder>
      <picture_fast_update/>
    </to_encoder>
  </vc_primitive>

</media_control>
```

[diagram] Call-ID:[prev][next]

[48] SIP/2.0 200 OK

```
Via: SIP/2.0/TLS 10.10.31.104:5061;branch=z9hG4bKbb22af198e4a46db3b9570011c564b12.1;rport
From: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
To: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
Date: Tue, 12 Apr 2011 22:56:08 GMT
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 503 INFO
```

```
Contact: <sip:10.10.13.50:5061;transport=tls>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[49] INFO sip:10.10.13.50:5061;transport=tls SIP/2.0
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bKb51ae5dbfeaee5a4d7ad51667b6dbaca.1;rport
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 107 INFO
Contact: <sip:10.10.31.107:5061>
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Type: application/media_control+xml
Content-Length: 193

<?xml version="1.0" encoding="utf-
8"?><media_control><vc_primitive><to_encoder><picture_fast_update></picture_fast_update></to_encoder><stream
_id>11</stream_id></vc_primitive></media_control>

[diagram]   Call-ID:[prev][next]
[50] INFO sip:500124@10.10.31.104:5061;transport=tls SIP/2.0
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK19b521674
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Date: Tue, 12 Apr 2011 22:56:08 GMT
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 109 INFO
Contact: <sip:10.10.13.50:5061;transport=tls>
Content-Type: application/media_control+xml
Content-Length: 190

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<media_control>

  <vc_primitive>
    <to_encoder>
      <picture_fast_update/>
    </to_encoder>
  </vc_primitive>

</media_control>

[diagram]   Call-ID:[prev][next]
[51] SIP/2.0 200 OK
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bKb51ae5dbfeaee5a4d7ad51667b6dbaca.1;rport
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Date: Tue, 12 Apr 2011 22:56:08 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 107 INFO
Contact: <sip:0500124@10.10.13.50:5061;transport=tls>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[52] SIP/2.0 200 OK
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK186002d026;received=10.10.13.50
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 104 INFO
From: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
To: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[53] SIP/2.0 200 OK
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK19b521674;received=10.10.13.50
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 109 INFO
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
```

```
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[54] INFO sip:10.10.13.50:5061;transport=tls SIP/2.0
Via: SIP/2.0/TLS 10.10.31.104:5061;branch=z9hG4bKfd0fd9df96431639eb2b384d82e64d3b.1;rport
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 504 INFO
Contact: <sip:500124@10.10.31.104:5061>
From: "" <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
To: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Type: application/media_control+xml
Content-Length: 193

<?xml version="1.0" encoding="utf-
8"?><media_control><vc_primitive><to_encoder><picture_fast_update></picture_fast_update></to_encoder><stream
_id>11</stream_id></vc_primitive></media_control>

[diagram]   Call-ID:[prev][next]
[55] INFO sip:10.10.31.107:5061;transport=tls SIP/2.0
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK1a334996d6
From: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
To: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
Date: Tue, 12 Apr 2011 22:56:08 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 105 INFO
Contact: <sip:0500124@10.10.13.50:5061;transport=tls>
Content-Type: application/media_control+xml
Content-Length: 190

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<media_control>

  <vc_primitive>
    <to_encoder>
      <picture_fast_update/>
    </to_encoder>
  </vc_primitive>

</media_control>

[diagram]   Call-ID:[prev][next]
[56] SIP/2.0 200 OK
Via: SIP/2.0/TLS 10.10.31.104:5061;branch=z9hG4bKfd0fd9df96431639eb2b384d82e64d3b.1;rport
From: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
To: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
Date: Tue, 12 Apr 2011 22:56:09 GMT
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 504 INFO
Contact: <sip:10.10.13.50:5061;transport=tls>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[57] INFO sip:10.10.13.50:5061;transport=tls SIP/2.0
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bKc5fea3af96c42037f2ed3bfdfaa6f2b9.1;rport
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 108 INFO
Contact: <sip:10.10.31.107:5061>
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Type: application/media_control+xml
Content-Length: 193

<?xml version="1.0" encoding="utf-
8"?><media_control><vc_primitive><to_encoder><picture_fast_update></picture_fast_update></to_encoder><stream
_id>11</stream_id></vc_primitive></media_control>

[diagram]   Call-ID:[prev][next]
[58] INFO sip:500124@10.10.31.104:5061;transport=tls SIP/2.0
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK1b61c4287a
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
```

```
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Date: Tue, 12 Apr 2011 22:56:09 GMT
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 110 INFO
Contact: <sip:10.10.13.50:5061;transport=tls>
Content-Type: application/media_control+xml
Content-Length: 190

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<media_control>

  <vc_primitive>
    <to_encoder>
      <picture_fast_update/>
    </to_encoder>
  </vc_primitive>

</media_control>

[diagram]   Call-ID:[prev][next]
[59] SIP/2.0 200 OK
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bKc5fea3af96c42037f2ed3bfdfaa6f2b9.1;rport
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Date: Tue, 12 Apr 2011 22:56:09 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 108 INFO
Contact: <sip:0500124@10.10.13.50:5061;transport=tls>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[60] SIP/2.0 200 OK
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK1a334996d6;received=10.10.13.50
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 105 INFO
From: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
To: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[61] SIP/2.0 200 OK
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK1b61c4287a;received=10.10.13.50
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 110 INFO
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[62] INFO sip:10.10.13.50:5061;transport=tls SIP/2.0
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bKfe2dc799716052b84d321de529d2dd38.1;rport
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 109 INFO
Contact: <sip:10.10.31.107:5061>
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Type: application/media_control+xml
Content-Length: 193

<?xml version="1.0" encoding="utf-
8"?><media_control><vc_primitive><to_encoder><picture_fast_update></picture_fast_update></to_encoder><stream
_id>11</stream_id></vc_primitive></media_control>

[diagram]   Call-ID:[prev][next]
[63] INFO sip:500124@10.10.31.104:5061;transport=tls SIP/2.0
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK1c3649cabd
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
```

```
Date: Tue, 12 Apr 2011 22:56:09 GMT
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 111 INFO
Contact: <sip:10.10.13.50:5061;transport=tls>
Content-Type: application/media_control+xml
Content-Length: 190

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<media_control>

  <vc_primitive>
    <to_encoder>
      <picture_fast_update/>
    </to_encoder>
  </vc_primitive>

</media_control>
```

[diagram] Call-ID:[prev][next]
[64] SIP/2.0 200 OK

```
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bKfe2dc799716052b84d321de529d2dd38.1;rport
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Date: Tue, 12 Apr 2011 22:56:10 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 109 INFO
Contact: <sip:0500124@10.10.13.50:5061;transport=tls>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[65] INFO sip:10.10.13.50:5061;transport=tls SIP/2.0

```
Via: SIP/2.0/TLS 10.10.31.104:5061;branch=z9hG4bK1d99e0a534ae3d2dff93d81933d4b8a4.1;rport
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 505 INFO
Contact: <sip:500124@10.10.31.104:5061>
From: "" <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
To: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Type: application/media_control+xml
Content-Length: 193

<?xml version="1.0" encoding="utf-
8"?><media_control><vc_primitive><to_encoder><picture_fast_update></picture_fast_update></to_encoder><stream
_id>11</stream_id></vc_primitive></media_control>
```

[diagram] Call-ID:[prev][next]
[66] SIP/2.0 200 OK

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK1c3649cabd;received=10.10.13.50
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 111 INFO
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[67] INFO sip:10.10.31.107:5061;transport=tls SIP/2.0

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK1d3bf4eaa0
From: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
To: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
Date: Tue, 12 Apr 2011 22:56:10 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 106 INFO
Contact: <sip:0500124@10.10.13.50:5061;transport=tls>
Content-Type: application/media_control+xml
Content-Length: 190

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<media_control>

  <vc_primitive>
```

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```
    <to_encoder>
      <picture_fast_update/>
    </to_encoder>
  </vc_primitive>

</media_control>

[diagram]   Call-ID:[prev][next]
[68] SIP/2.0 200 OK
Via: SIP/2.0/TLS 10.10.31.104:5061;branch=z9hG4bK1d99e0a534ae3d2dff93d81933d4b8a4.1;rport
From: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
To: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
Date: Tue, 12 Apr 2011 22:56:10 GMT
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 505 INFO
Contact: <sip:10.10.13.50:5061;transport=tls>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[69] SIP/2.0 200 OK
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK1d3bf4eaa0;received=10.10.13.50
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 106 INFO
From: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
To: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[70] INFO sip:10.10.13.50:5061;transport=tls SIP/2.0
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bKb048a8447ebf5bbdde47959c852be497.1;rport
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 110 INFO
Contact: <sip:10.10.31.107:5061>
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Type: application/media_control+xml
Content-Length: 193

<?xml version="1.0" encoding="utf-
8"?><media_control><vc_primitive><to_encoder><picture_fast_update></picture_fast_update></to_encoder><stream
_id>11</stream_id></vc_primitive></media_control>

[diagram]   Call-ID:[prev][next]
[71] INFO sip:500124@10.10.31.104:5061;transport=tls SIP/2.0
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK1e59db6e9
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Date: Tue, 12 Apr 2011 22:56:10 GMT
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 112 INFO
Contact: <sip:10.10.13.50:5061;transport=tls>
Content-Type: application/media_control+xml
Content-Length: 190

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<media_control>

  <vc_primitive>
    <to_encoder>
      <picture_fast_update/>
    </to_encoder>
  </vc_primitive>

</media_control>

[diagram]   Call-ID:[prev][next]
[72] SIP/2.0 200 OK
Via: SIP/2.0/TLS 10.10.31.107:5061;branch=z9hG4bKb048a8447ebf5bbdde47959c852be497.1;rport
From: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
To: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
Date: Tue, 12 Apr 2011 22:56:15 GMT
```

```
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 110 INFO
Contact: <sip:0500124@10.10.13.50:5061;transport=tls>
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]
[73] SIP/2.0 200 OK

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK1e59db6e9;received=10.10.13.50
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 112 INFO
From: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
To: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]
[74] BYE sip:10.10.13.50:5061;transport=tls SIP/2.0

```
Via: SIP/2.0/TLS 10.10.31.104:5061;branch=z9hG4bKb901136cdbed130360dc66e48c114b5e.1;rport
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 506 BYE
Contact: <sip:500124@10.10.31.104:5061;transport=tls>
From: "" <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
To: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]
[75] SIP/2.0 200 OK

```
Via: SIP/2.0/TLS 10.10.31.104:5061;branch=z9hG4bKb901136cdbed130360dc66e48c114b5e.1;rport
From: <sip:500124@10.10.31.104>;tag=af7d2fac4d7aa9fa
To: <sip:10.10.13.50>;tag=519~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365300
Date: Tue, 12 Apr 2011 22:56:16 GMT
Call-ID: 5c6c200-da41d87d-5-320d1bac@10.10.13.50
CSeq: 506 BYE
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]
[76] BYE sip:10.10.31.107:5061;transport=tls SIP/2.0

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK1f4afec5fd
From: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
To: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
Date: Tue, 12 Apr 2011 22:56:15 GMT
Call-ID: 3d9865c2dda0dc33@10.10.31.107
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 107 BYE
Reason: Q.850;cause=16
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]
[77] SIP/2.0 200 OK

```
Via: SIP/2.0/TLS 10.10.13.50:5061;branch=z9hG4bK1f4afec5fd;received=10.10.13.50
Call-ID: 3d9865c2dda0dc33@10.10.31.107
CSeq: 107 BYE
From: <sip:0500124@10.10.13.50>;tag=518~e9c104bd-6f9c-4ff9-a684-33f6b4afdab6-26365299
To: <sip:10.10.13.50>;tag=1dd3182ddbb34bd7
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.0.0.-Beta4 (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Content-Length: 0
```

7. V.150.1 MER

7.1 Basic SCCP-SCCP V.150 Call Device via an Inter Cluster SIP Trunk

Title: Basic SCCP-SCCP V.150 Call Device via an Inter Cluster SIP Trunk

Description:

The following call flow illustrates the SIP messaging that takes place between two Cisco Unified CMs via an inter cluster SIP trunk for a V.150.1 call with standard options.

Cisco Unified CM1 sent out the initial INVITE. Note that due to equipment limitations, V.150.1 endpoints are only attached to a gateway on CM1, so there are really two trunks from CM1 to CM2, one in each direction so that we can get a roundtrip back to the other V.150.1 endpoint attached to CM1.

Configuration:

Node = Unified CM1, IP = 10.10.154.62
Node = Unified CM2, IP = 10.10.154.61

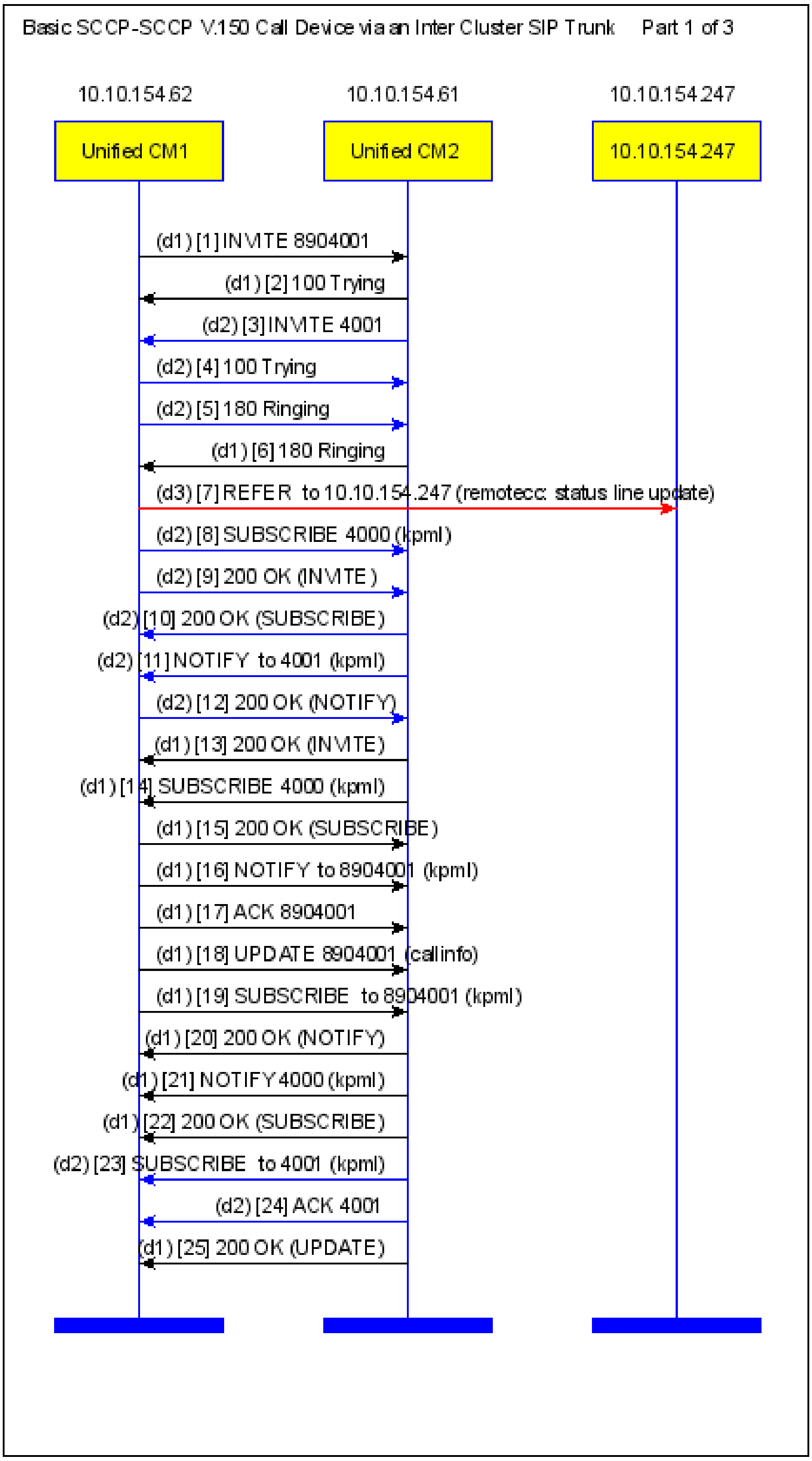
Phone = A, Line = 4000, IP = 10.10.153.132, Model = SCCP via gateway FXS
Phone = B, Line = 4001, IP = 10.10.153.132, Model = SCCP via gateway FXS

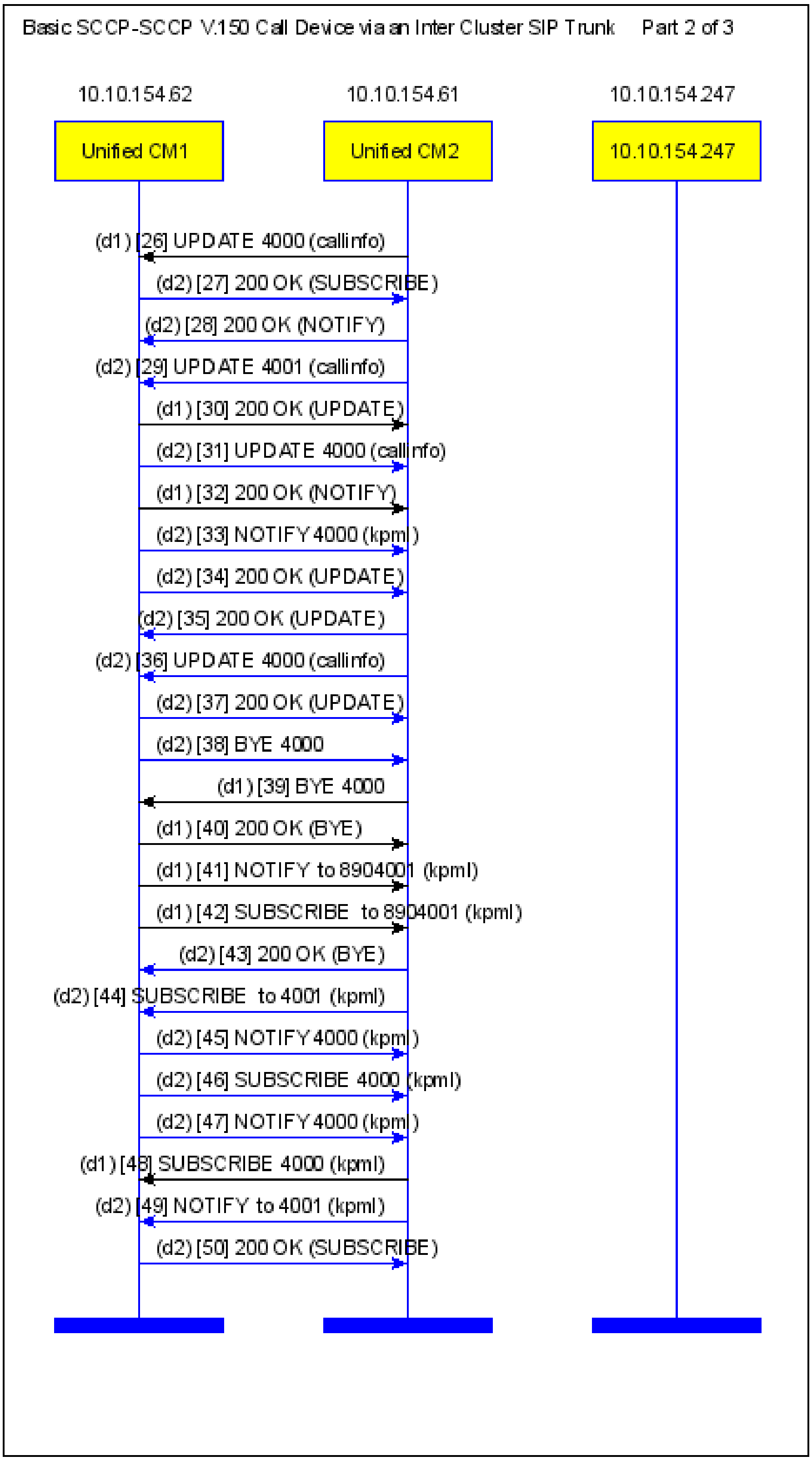
SIP Trunk between Unified CM1 & Unified CM2 has route pattern 615.XXXX that transforms to 890.XXXX which causes the call to be routed back to CM1 over a SIP Trunk in the other direction.

Scenario:

Phone A DN 4000 calls Phone B DN 4001 over the SIP Trunk by dialing 6154001
Phone B DN 4001 Answers
Phone B DN 4001 goes onHook

End of Scenario


Basic SCCP-SCCP V.150 Call Device via an Inter Cluster SIP Trunk Part 1 of 3
10.10.154.62
10.10.154.61
10.10.154.247
Unified CM1
Unified CM2
10.10.154.247
(d1) [1] INVITE 8904001
(d1) [2] 100 Trying
(d2) [3] INVITE 4001
(d2) [4] 100 Trying
(d2) [5] 180 Ringing
(d1) [6] 180 Ringing
(d3) [7] REFER to 10.10.154.247 (remotecc: status line update)
(d2) [8] SUBSCRIBE 4000 (kpml)
(d2) [9] 200 OK (INVITE)
(d2) [10] 200 OK (SUBSCRIBE)
(d2) [11] NOTIFY to 4001 (kpml)
(d2) [12] 200 OK (NOTIFY)
(d1) [13] 200 OK (INVITE)
(d1) [14] SUBSCRIBE 4000 (kpml)
(d1) [15] 200 OK (SUBSCRIBE)
(d1) [16] NOTIFY to 8904001 (kpml)
(d1) [17] ACK 8904001
(d1) [18] UPDATE 8904001 (callinfo)
(d1) [19] SUBSCRIBE to 8904001 (kpml)
(d1) [20] 200 OK (NOTIFY)
(d1) [21] NOTIFY 4000 (kpml)
(d1) [22] 200 OK (SUBSCRIBE)
(d2) [23] SUBSCRIBE to 4001 (kpml)
(d2) [24] ACK 4001
(d1) [25] 200 OK (UPDATE)


Basic SCCP-SCCP V.150 Call Device via an Inter Cluster SIP Trunk Part 2 of 3
10.10.154.62
10.10.154.61
10.10.154.247
Unified CM1
Unified CM2
10.10.154.247
(d1) [26] UPDATE 4000 (callinfo)
(d2) [27] 200 OK (SUBSCRIBE)
(d2) [28] 200 OK (NOTIFY)
(d2) [29] UPDATE 4001 (callinfo)
(d1) [30] 200 OK (UPDATE)
(d2) [31] UPDATE 4000 (callinfo)
(d1) [32] 200 OK (NOTIFY)
(d2) [33] NOTIFY 4000 (kpml)
(d2) [34] 200 OK (UPDATE)
(d2) [35] 200 OK (UPDATE)
(d2) [36] UPDATE 4000 (callinfo)
(d2) [37] 200 OK (UPDATE)
(d2) [38] BYE 4000
(d1) [39] BYE 4000
(d1) [40] 200 OK (BYE)
(d1) [41] NOTIFY to 8904001 (kpml)
(d1) [42] SUBSCRIBE to 8904001 (kpml)
(d2) [43] 200 OK (BYE)
(d2) [44] SUBSCRIBE to 4001 (kpml)
(d2) [45] NOTIFY 4000 (kpml)
(d2) [46] SUBSCRIBE 4000 (kpml)
(d2) [47] NOTIFY 4000 (kpml)
(d1) [48] SUBSCRIBE 4000 (kpml)
(d2) [49] NOTIFY to 4001 (kpml)
(d2) [50] 200 OK (SUBSCRIBE)

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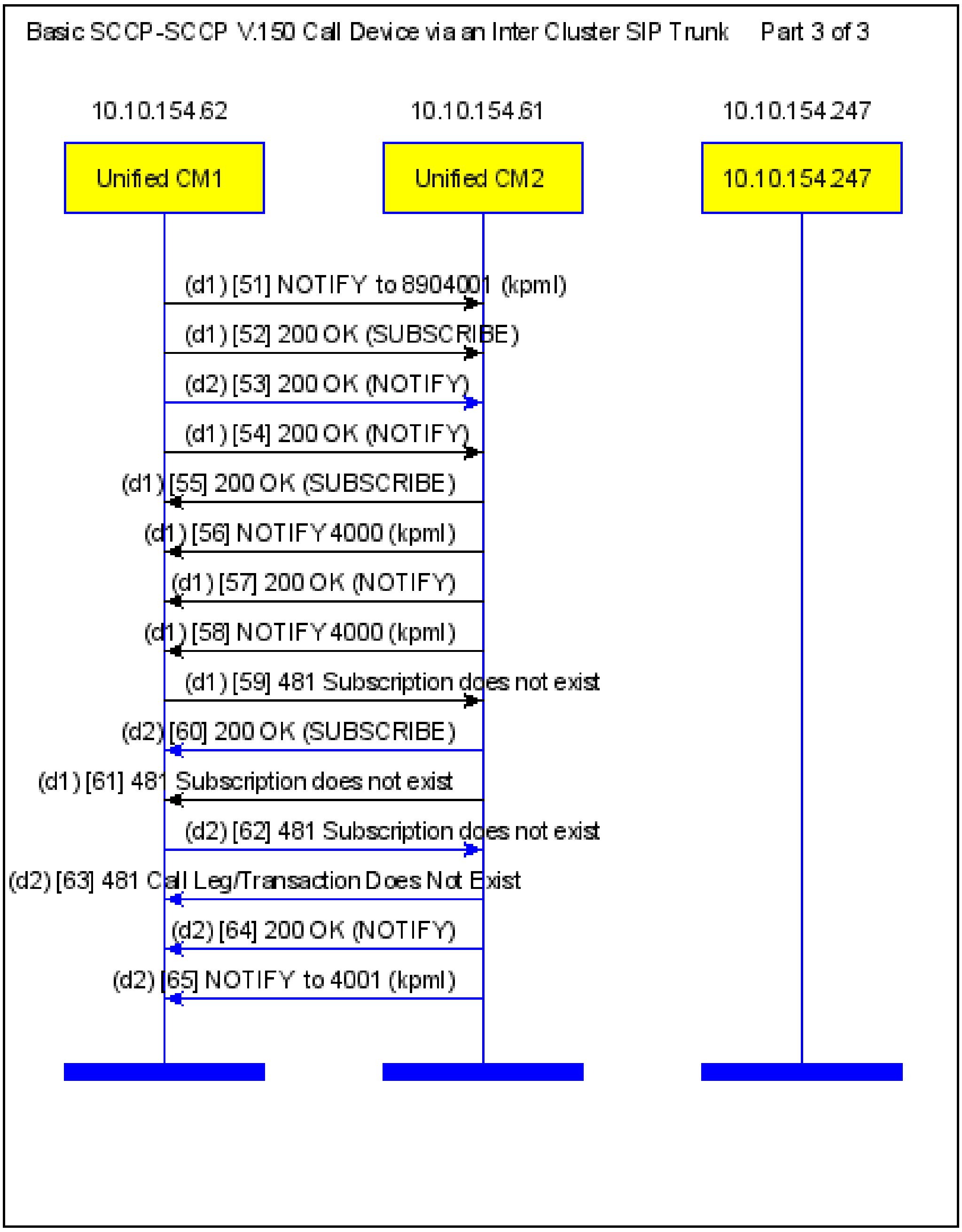

```
[diagram]   Call-ID:[prev][next]
[1] INVITE sip:8904001@10.10.154.61:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK28d2bbb3be7
From: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
To: <sip:8904001@10.10.154.61>
Date: Thu, 21 Apr 2011 00:27:04 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM8.6
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: <sip:10.10.154.62:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 0329957760-0000065536-0000000001-1050284716
Session-Expires:  1800
P-Asserted-Identity: <sip:4000@10.10.154.62>
Remote-Party-ID: <sip:4000@10.10.154.62>;party=calling;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
Max-Forwards: 70
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[2] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK28d2bbb3be7
From: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
To: <sip:8904001@10.10.154.61>
Date: Thu, 21 Apr 2011 01:02:21 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[3] INVITE sip:4001@10.10.154.62:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK1a48c248af
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>
Date: Thu, 21 Apr 2011 01:02:22 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM8.6
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: <sip:10.10.154.61:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 0035121280-0000065536-0000000004-1033507500
Session-Expires:  1800
P-Asserted-Identity: <sip:4000@10.10.154.61>
Remote-Party-ID: <sip:4000@10.10.154.61>;party=calling;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
Max-Forwards: 69
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[4] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK1a48c248af
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>
Date: Thu, 21 Apr 2011 00:27:06 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[5] SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK1a48c248af
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
Date: Thu, 21 Apr 2011 00:27:06 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 101 INVITE
```

```
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4001@10.10.154.62>
Remote-Party-ID: <sip:4001@10.10.154.62>;party=called;screen=yes;privacy=off
Contact: <sip:4001@10.10.154.62:5060;transport=tcp>
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]
[6] SIP/2.0 180 Ringing

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK28d2bbb3be7
From: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
To: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
Date: Thu, 21 Apr 2011 01:02:21 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4001@10.10.154.61>
Remote-Party-ID: <sip:4001@10.10.154.61>;party=called;screen=yes;privacy=off
Contact: <sip:8904001@10.10.154.61:5060;transport=tcp>
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]
[7] REFER sip:10.10.154.247:51937 SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK28e75a24238
From: <sip:10.10.154.62>;tag=440375864
To: <sip:10.10.154.247>
Call-ID: 14dbee80-daf179da-28f-3e9a12ac@10.10.154.62
CSeq: 101 REFER
Max-Forwards: 70
Contact: <sip:10.10.154.62:5060;transport=tcp>
User-Agent: Cisco-CUCM8.6
Expires: 30
Refer-To: cid:1234567890@10.10.154.62
Content-Id: <1234567890@10.10.154.62>
Content-Type: application/x-cisco-remotecc-request+xml
Referred-By: <sip:10.10.154.62>
Content-Length: 265

<?xml version="1.0" encoding="iso-8859-1"?>
<x-cisco-remotecc-request>
<statuslineupdatereq>
<action>notify_display</action>
<statustext>U</statustext>
<displaytimeout>10</displaytimeout>
<priority>1</priority>
</statuslineupdatereq>
</x-cisco-remotecc-request>
```

[diagram]   Call-ID:[prev][next]
[8] SUBSCRIBE sip:4000@10.10.154.61:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK28f60b90bc
From: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
To: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 101 SUBSCRIBE
Date: Thu, 21 Apr 2011 00:27:08 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 7200
Contact: <sip:10.10.154.62:5060;transport=tcp>
Accept: application/kpml-response+xml
Max-Forwards: 70
Content-Type: application/kpml-request+xml
Content-Length: 370

<?xml version="1.0" encoding="UTF-8" ?>
<kpml-request xmlns="urn:ietf:params:xml:ns:kpml-request" xmlns:xsi="http://www.w3.org/2001/XMLSchema-
instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-request kpml-request.xsd" version="1.0">

  <pattern interdigittimer="7260000" persist="persist">
    <regex tag="dtmf">[x*#ABCD]</regex>
  </pattern>

</kpml-request>
```

[diagram] Call-ID:[prev][next]

```
[9] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK1a48c248af
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
Date: Thu, 21 Apr 2011 00:27:06 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence, kpml
Supported: replaces
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Session-Expires:  1800;refresher=uas
Require:  timer
P-Asserted-Identity: <sip:4001@10.10.154.62>
Remote-Party-ID: <sip:4001@10.10.154.62>;party=called;screen=yes;privacy=off
Contact: <sip:4001@10.10.154.62:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 803

v=0
o=CiscoSystemsCCM-SIP 1309 1 IN IP4 10.10.154.62
s=SIP Call
c=IN IP4 10.10.153.132
b=TIAS:64000
b=AS:64
t=0 0
m=audio 25972 RTP/SAVP 0 8 116 18 101 100 118 126
a=crypto:1 AES_CM_128_HMAC_SHA1_32 inline:6rbm93LqlWIMY5Df16LUUmWJRypknkBD+QG3rg4P
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:8 PCMA/8000
a=ptime:20
a=rtpmap:116 iLBC/8000
a=ptime:20
a=maxptime:120
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15,32-35
a=rtpmap:100 X-NSE/8000
a=rtpmap:118 v150fw/8000
a=rtpmap:126 NoAudio/8000
a=sqn:0
a=cdsc: 1 audio RTP/SAVP 0 8 116 18 101 100 118 126
a=cdsc: 9 audio udpsprt 120
a=cpar: a=sprtmap:120 v150mr/8000
a=cpar: a=fmtp:120 mr=1;mg=2;CDSCselect=1;jmdelay=yes;Versn=1.1;mrmods=1-5,7-8,10-11,13
a=vndpar:2 9 2 15
```

[diagram] Call-ID:[prev][next]

```
[10] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK28f60b90bc
From: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
To: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
Date: Thu, 21 Apr 2011 01:02:25 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 101 SUBSCRIBE
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
Expires: 7200
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

```
[11] NOTIFY sip:10.10.154.62:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK1b2bbf8498
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 102 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 01:02:25 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: active;expires=7200
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.61>
Content-Length: 0
```

```
[diagram]   Call-ID:[prev][next]
[12] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK1b2bbf8498
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
Date: Thu, 21 Apr 2011 00:27:09 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 102 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[13] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK28d2bbb3be7
From: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
To: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
Date: Thu, 21 Apr 2011 01:02:21 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence, kpml
Supported: replaces
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Session-Expires:  1800;refresher=uas
Require:  timer
P-Asserted-Identity: <sip:4001@10.10.154.61>
Remote-Party-ID: <sip:4001@10.10.154.61>;party=called;screen=yes;privacy=off
Contact: <sip:8904001@10.10.154.61:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 801

v=0
o=CiscoSystemsCCM-SIP 12 1 IN IP4 10.10.154.61
s=SIP Call
c=IN IP4 10.10.153.132
b=TIAS:64000
b=AS:64
t=0 0
m=audio 25972 RTP/SAVP 0 8 116 18 101 100 118 126
a=crypto:1 AES_CM_128_HMAC_SHA1_32 inline:6rbm93LqlWIMY5Df16LUUmWJRypknkBD+QG3rg4P
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:8 PCMA/8000
a=ptime:20
a=rtpmap:116 iLBC/8000
a=ptime:20
a=maxptime:120
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15,32-35
a=rtpmap:100 X-NSE/8000
a=rtpmap:118 v150fw/8000
a=rtpmap:126 NoAudio/8000
a=sqn:0
a=cdsc: 1 audio RTP/SAVP 0 8 116 18 101 100 118 126
a=cdsc: 9 audio udpsprt 120
a=cpar: a=sprtmap:120 v150mr/8000
a=cpar: a=fmtp:120 mr=1;mg=2;CDSCselect=1;jmdelay=yes;Versn=1.1;mrmods=1-5,7-8,10-11,13
a=vndpar:2 9 2 15

[diagram]   Call-ID:[prev][next]
[14] SUBSCRIBE sip:4000@10.10.154.62:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK1c51a256f
From: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
To: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 101 SUBSCRIBE
Date: Thu, 21 Apr 2011 01:02:25 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 7200
Contact: <sip:10.10.154.61:5060;transport=tcp>
Accept: application/kpml-response+xml
Max-Forwards: 70
Content-Type: application/kpml-request+xml
Content-Length: 370

<?xml version="1.0" encoding="UTF-8" ?>
```

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```
<kpml-request xmlns="urn:ietf:params:xml:ns:kpml-request" xmlns:xsi="http://www.w3.org/2001/XMLSchema-
instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-request kpml-request.xsd" version="1.0">

  <pattern interdigittimer="7260000" persist="persist">
    <regex tag="dtmf">[x*#ABCD]</regex>
  </pattern>

</kpml-request>
```


```
[diagram]   Call-ID:[prev][next]
[15] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK1c51a256f
From: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
To: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
Date: Thu, 21 Apr 2011 00:27:09 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 101 SUBSCRIBE
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
Expires: 7200
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[16] NOTIFY sip:10.10.154.61:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2909b0b98b
From: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
To: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 102 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 00:27:09 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: active;expires=7200
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.62>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[17] ACK sip:8904001@10.10.154.61:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK291e5c2eb4
From: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
To: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
Date: Thu, 21 Apr 2011 00:27:04 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Type: application/sdp
Content-Length: 553

v=0
o=CiscoSystemsCCM-SIP 1308 1 IN IP4 10.10.154.62
s=SIP Call
c=IN IP4 10.10.153.132
b=TIAS:64000
b=AS:64
t=0 0
m=audio 29236 RTP/SAVP 0 101 118
a=crypto:1 AES_CM_128_HMAC_SHA1_32 inline:eC3+6uklpSSiJNqTbNX5dQffA8/fTVkatkktyaYU
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15,32-35
a=rtpmap:118 v150fw/8000
a=sqn:0
a=cdsc: 1 audio RTP/SAVP 0 101 118
a=cdsc: 4 audio udpsprt 120
a=cpar: a=sprtmap:120 v150mr/8000
a=cpar: a=fmtp:120 mr=1;mg=0;CDSCselect=1;jmdelay=no;Versn=1.1;mrmods=1-5,7-8,10-11

[diagram]   Call-ID:[prev][next]
[18] UPDATE sip:8904001@10.10.154.61:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2935ddb0e7d
From: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
To: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
Date: Thu, 21 Apr 2011 00:27:04 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
Supported: timer,resource-priority,replaces
```

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```
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 104 UPDATE
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; gci= 1-51001
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4000@10.10.154.62>
Remote-Party-ID: <sip:4000@10.10.154.62>;party=calling;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]

```
[19] SUBSCRIBE sip:10.10.154.61:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK292341d1b49
From: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
To: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 103 SUBSCRIBE
Date: Thu, 21 Apr 2011 00:27:09 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 7200
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.62>
Accept: application/kpml-response+xml
Max-Forwards: 70
Content-Type: application/kpml-request+xml
Content-Length: 370

<?xml version="1.0" encoding="UTF-8" ?>
<kpml-request xmlns="urn:ietf:params:xml:ns:kpml-request" xmlns:xsi="http://www.w3.org/2001/XMLSchema-
instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-request kpml-request.xsd" version="1.0">

  <pattern interdigittimer="7260000" persist="persist">
    <regex tag="dtmf">[x*#ABCD]</regex>
  </pattern>

</kpml-request>
```

[diagram]   Call-ID:[prev][next]

```
[20] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2909b0b98b
From: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
To: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
Date: Thu, 21 Apr 2011 01:02:25 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 102 NOTIFY
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]

```
[21] NOTIFY sip:4000@10.10.154.62:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK1f48862773
From: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
To: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 102 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 01:02:25 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: active;expires=7200
Contact: <sip:10.10.154.61:5060;transport=tcp>
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]

```
[22] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK292341d1b49
From: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
To: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
Date: Thu, 21 Apr 2011 01:02:25 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 103 SUBSCRIBE
Contact: <sip:10.10.154.61:5060;transport=tcp>
Expires: 7200
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]

```
[23] SUBSCRIBE sip:10.10.154.62:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK1e6d2f51e4
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
```

```
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 103 SUBSCRIBE
Date: Thu, 21 Apr 2011 01:02:25 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 7200
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.61>
Accept: application/kpml-response+xml
Max-Forwards: 70
Content-Type: application/kpml-request+xml
Content-Length: 370
```


```
<?xml version="1.0" encoding="UTF-8" ?>
<kpml-request xmlns="urn:ietf:params:xml:ns:kpml-request" xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-request kpml-request.xsd" version="1.0">

  <pattern interdigittimer="7260000" persist="persist">
    <regex tag="dtmf">[x*#ABCD]</regex>
  </pattern>

</kpml-request>
```


[diagram] Call-ID:[prev][next]

[24] ACK sip:4001@10.10.154.62:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK1d6e23d98d
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
Date: Thu, 21 Apr 2011 01:02:22 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Type: application/sdp
Content-Length: 551

v=0
o=CiscoSystemsCCM-SIP 13 1 IN IP4 10.10.154.61
s=SIP Call
c=IN IP4 10.10.153.132
b=TIAS:64000
b=AS:64
t=0 0
m=audio 29236 RTP/SAVP 0 101 118
a=crypto:1 AES_CM_128_HMAC_SHA1_32 inline:eC3+6uklpSSiJNqTbNX5dQffA8/fTVkatkktyaYU
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15,32-35
a=rtpmap:118 v150fw/8000
a=sqn:0
a=cdsc: 1 audio RTP/SAVP 0 101 118
a=cdsc: 4 audio udpsprt 120
a=cpar: a=sprtmap:120 v150mr/8000
a=cpar: a=fmtp:120 mr=1;mg=0;CDSCselect=1;jmdelay=no;Versn=1.1;mrmods=1-5,7-8,10-11
```

[diagram] Call-ID:[prev][next]

[25] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2935ddb0e7d
From: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
To: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
Date: Thu, 21 Apr 2011 01:02:25 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 104 UPDATE
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4001@10.10.154.61>
Remote-Party-ID: <sip:4001@10.10.154.61>;party=called;screen=yes;privacy=off
Contact: <sip:8904001@10.10.154.61:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[26] UPDATE sip:4000@10.10.154.62:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK20377f6f4d
From: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
To: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
Date: Thu, 21 Apr 2011 01:02:25 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
User-Agent: Cisco-CUCM8.6
```

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```
Max-Forwards: 70
Supported: timer,resource-priority,replaces
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 103 UPDATE
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= Encrypted; gci= 1-33004
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4001@10.10.154.61>
Remote-Party-ID: <sip:4001@10.10.154.61>;party=calling;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[27] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK1e6d2f51e4
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
Date: Thu, 21 Apr 2011 00:27:09 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 103 SUBSCRIBE
Contact: <sip:10.10.154.62:5060;transport=tcp>
Expires: 7200
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[28] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2941544e873
From: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
To: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
Date: Thu, 21 Apr 2011 01:02:25 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 102 NOTIFY
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[29] UPDATE sip:4001@10.10.154.62:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK2148b3848d
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
Date: Thu, 21 Apr 2011 01:02:22 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
Supported: timer,resource-priority,replaces
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 104 UPDATE
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= Encrypted; gci= 1-33004
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4000@10.10.154.61>
Remote-Party-ID: <sip:4000@10.10.154.61>;party=calling;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[30] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK20377f6f4d
From: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
To: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
Date: Thu, 21 Apr 2011 00:27:09 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 103 UPDATE
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4000@10.10.154.62>
Remote-Party-ID: <sip:4000@10.10.154.62>;party=called;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[31] UPDATE sip:4000@10.10.154.61:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK29513de9b8b
From: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
To: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
Date: Thu, 21 Apr 2011 00:27:09 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
Supported: timer,resource-priority,replaces
```

```
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 103 UPDATE
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; gci= 1-51002
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4001@10.10.154.62>
Remote-Party-ID: <sip:4001@10.10.154.62>;party=calling;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[32] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK1f48862773
From: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
To: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
Date: Thu, 21 Apr 2011 00:27:09 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 102 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[33] NOTIFY sip:4000@10.10.154.61:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2941544e873
From: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
To: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 102 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 00:27:09 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: active;expires=7200
Contact: <sip:10.10.154.62:5060;transport=tcp>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[34] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK2148b3848d
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
Date: Thu, 21 Apr 2011 00:27:09 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 104 UPDATE
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4001@10.10.154.62>
Remote-Party-ID: <sip:4001@10.10.154.62>;party=called;screen=yes;privacy=off
Contact: <sip:4001@10.10.154.62:5060;transport=tcp>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[35] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK29513de9b8b
From: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
To: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
Date: Thu, 21 Apr 2011 01:02:25 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 103 UPDATE
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4000@10.10.154.61>
Remote-Party-ID: <sip:4000@10.10.154.61>;party=called;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[36] UPDATE sip:4000@10.10.154.62:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK2235a89a1
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
Date: Thu, 21 Apr 2011 01:02:25 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
Supported: timer,resource-priority,replaces
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 105 UPDATE
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; gci= 1-33004
```

```
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4000@10.10.154.61>
Remote-Party-ID: <sip:4000@10.10.154.61>;party=calling;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[37] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK2235a89a1
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
Date: Thu, 21 Apr 2011 00:27:09 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 105 UPDATE
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4001@10.10.154.62>
Remote-Party-ID: <sip:4001@10.10.154.62>;party=called;screen=yes;privacy=off
Contact: <sip:4001@10.10.154.62:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[38] BYE sip:4000@10.10.154.61:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2966f42e334
From: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
To: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
Date: Thu, 21 Apr 2011 00:27:09 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 104 BYE
Reason: Q.850;cause=16
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[39] BYE sip:4000@10.10.154.62:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK2361ba2e9f
From: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
To: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
Date: Thu, 21 Apr 2011 01:02:25 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 104 BYE
Reason: Q.850;cause=16
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[40] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK2361ba2e9f
From: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
To: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
Date: Thu, 21 Apr 2011 00:27:14 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 104 BYE
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[41] NOTIFY sip:10.10.154.61:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK298722a1c29
From: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
To: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 106 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 00:27:14 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: terminated
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.62>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[42] SUBSCRIBE sip:10.10.154.61:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK29750d621c9
From: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
To: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
```

```
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 105 SUBSCRIBE
Date: Thu, 21 Apr 2011 00:27:14 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 0
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.62>
Max-Forwards: 70
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[43] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2966f42e334
From: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
To: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
Date: Thu, 21 Apr 2011 01:02:30 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 104 BYE
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[44] SUBSCRIBE sip:10.10.154.62:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK242c440c6c
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 106 SUBSCRIBE
Date: Thu, 21 Apr 2011 01:02:30 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 0
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.61>
Max-Forwards: 70
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[45] NOTIFY sip:4000@10.10.154.61:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK29a5683006b
From: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
To: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 106 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 00:27:14 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: terminated
Contact: <sip:10.10.154.62:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[46] SUBSCRIBE sip:4000@10.10.154.61:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2994e512912
From: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
To: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 105 SUBSCRIBE
Date: Thu, 21 Apr 2011 00:27:14 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 0
Contact: <sip:10.10.154.62:5060;transport=tcp>
Max-Forwards: 70
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[47] NOTIFY sip:4000@10.10.154.61:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK29b2ab827be
From: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
To: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 107 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 00:27:14 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: terminated;reason=timeout
Contact: <sip:10.10.154.62:5060;transport=tcp>
```

```
Content-Type: application/kpml-response+xml
Content-Length: 348

<?xml version="1.0" encoding="UTF-8" ?>
<kpml-response xmlns="urn:ietf:params:xml:ns:kpml-response" xmlns:xsi="http://www.w3.org/2001/XMLSchema-
instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-response kpml-response.xsd" code="487" digits=""
forced_flush="false" suppressed="false" tag="dtmf" text="Subscription Expired" version="1.0"/>
```

[diagram] Call-ID:[prev][next]
[48] SUBSCRIBE sip:4000@10.10.154.62:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK2632188fc8
From: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
To: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 105 SUBSCRIBE
Date: Thu, 21 Apr 2011 01:02:30 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 0
Contact: <sip:10.10.154.61:5060;transport=tcp>
Max-Forwards: 70
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[49] NOTIFY sip:10.10.154.62:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK2523af7e0f
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 107 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 01:02:30 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: terminated
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.61>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[50] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK242c440c6c
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
Date: Thu, 21 Apr 2011 00:27:14 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 106 SUBSCRIBE
Contact: <sip:10.10.154.62:5060;transport=tcp>
Expires: 0
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[51] NOTIFY sip:10.10.154.61:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK29c531dea5c
From: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
To: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 107 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 00:27:14 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: terminated;reason=timeout
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.62>
Content-Type: application/kpml-response+xml
Content-Length: 348

<?xml version="1.0" encoding="UTF-8" ?>
<kpml-response xmlns="urn:ietf:params:xml:ns:kpml-response" xmlns:xsi="http://www.w3.org/2001/XMLSchema-
instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-response kpml-response.xsd" code="487" digits=""
forced_flush="false" suppressed="false" tag="dtmf" text="Subscription Expired" version="1.0"/>
```

[diagram] Call-ID:[prev][next]
[52] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK2632188fc8
From: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
To: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
Date: Thu, 21 Apr 2011 00:27:14 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
```

```
CSeq: 105 SUBSCRIBE
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
Expires: 0
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[53] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK2523af7e0f
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
Date: Thu, 21 Apr 2011 00:27:14 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 107 NOTIFY
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[54] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK27262102f2
From: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
To: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
Date: Thu, 21 Apr 2011 00:27:14 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 106 NOTIFY
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[55] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK29750d621c9
From: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
To: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
Date: Thu, 21 Apr 2011 01:02:30 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 105 SUBSCRIBE
Contact: <sip:10.10.154.61:5060;transport=tcp>
Expires: 0
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[56] NOTIFY sip:4000@10.10.154.62:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK27262102f2
From: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
To: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 106 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 01:02:30 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: terminated
Contact: <sip:10.10.154.61:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[57] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK298722a1c29
From: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
To: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
Date: Thu, 21 Apr 2011 01:02:30 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 106 NOTIFY
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[58] NOTIFY sip:4000@10.10.154.62:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK2811711082
From: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
To: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 107 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 01:02:30 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: terminated;reason=timeout
Contact: <sip:10.10.154.61:5060;transport=tcp>
Content-Type: application/kpml-response+xml
Content-Length: 348
```


```
<?xml version="1.0" encoding="UTF-8" ?>
<kpml-response xmlns="urn:ietf:params:xml:ns:kpml-response" xmlns:xsi="http://www.w3.org/2001/XMLSchema-
```

```
instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-response kpml-response.xsd" code="487" digits=""
forced_flush="false" suppressed="false" tag="dtmf" text="Subscription Expired" version="1.0"/>

[diagram]   Call-ID:[prev][next]
[59] SIP/2.0 481 Subscription does not exist
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK2811711082
From: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
To: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
Date: Thu, 21 Apr 2011 00:27:14 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 107 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[60] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2994e512912
From: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
To: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
Date: Thu, 21 Apr 2011 01:02:30 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 105 SUBSCRIBE
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
Expires: 0
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[61] SIP/2.0 481 Subscription does not exist
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK29c531dea5c
From: <sip:4000@10.10.154.62>;tag=1308~90227f56-6e59-4152-9c5d-6d54ef528a66-28780366
To: <sip:8904001@10.10.154.61>;tag=12~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895547
Date: Thu, 21 Apr 2011 01:02:30 GMT
Call-ID: 13aac180-daf179d8-28e-3e9a12ac@10.10.154.62
CSeq: 107 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[62] SIP/2.0 481 Subscription does not exist
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK29d197cad
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
Date: Thu, 21 Apr 2011 00:27:14 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 108 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[63] SIP/2.0 481 Call Leg/Transaction Does Not Exist
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK29b2ab827be
From: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
To: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
Date: Thu, 21 Apr 2011 01:02:30 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 107 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[64] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK29a5683006b
From: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
To: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
Date: Thu, 21 Apr 2011 01:02:30 GMT
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 106 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[65] NOTIFY sip:10.10.154.62:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK29d197cad
From: <sip:4000@10.10.154.61>;tag=13~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895548
To: <sip:4001@10.10.154.62>;tag=1309~90227f56-6e59-4152-9c5d-6d54ef528a66-28780367
Call-ID: 217e880-daf1821e-4-3d9a12ac@10.10.154.61
CSeq: 108 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 01:02:30 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: terminated;reason=timeout
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.61>
Content-Type: application/kpml-response+xml
```

```
Content-Length: 348

<?xml version="1.0" encoding="UTF-8" ?>
<kpml-response xmlns="urn:ietf:params:xml:ns:kpml-response" xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-response kpml-response.xsd" code="487" digits="" forced_flush="false" suppressed="false" tag="dtmf" text="Subscription Expired" version="1.0"/>
```

7.2 SCCP-SCCP V.150 Fax Call via an Inter Cluster SIP Trunk

Title: SCCP-SCCP V.150 Fax Call via an Inter Cluster SIP Trunk

Description:
The following call flow illustrates the SIP messaging that takes place between two Cisco Unified CMs via an inter cluster SIP trunk for a V.150.1 call with T.38 fax.

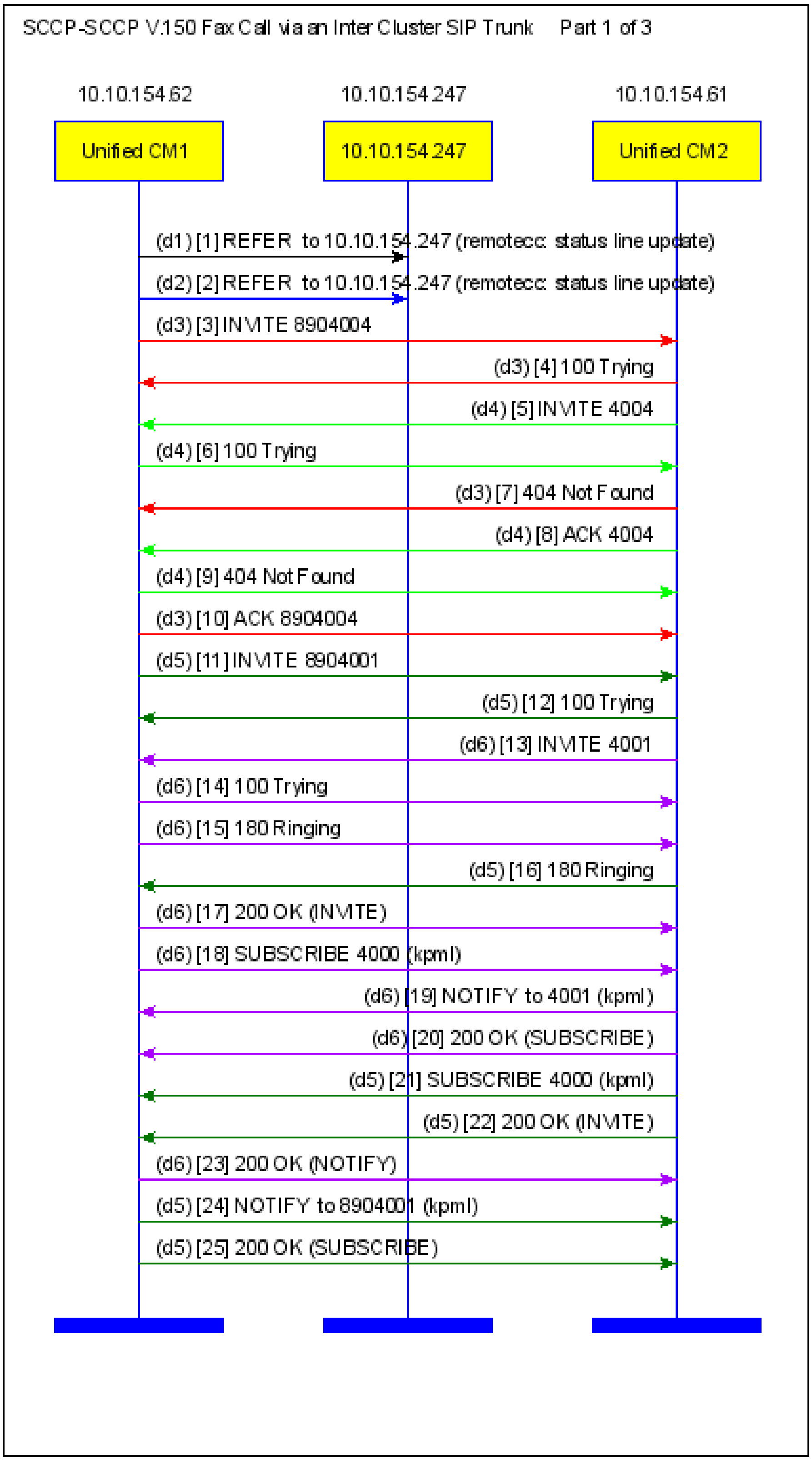
Cisco Unified CM1 sent out the initial INVITE. Note that due to equipment limitations, V.150.1 endpoints are only attached to a gateway on CM1, so there are really two trunks from CM1 to CM2, one in each direction so that we can get a roundtrip back to the other V.150.1 endpoint attached to CM1.

Configuration:
Node = Unified CM1, IP = 10.10.154.62
Node = Unified CM2, IP = 10.10.154.61

Phone = A, Line = 4000, IP = 10.10.153.132, Model = SCCP via gateway FXS
Phone = B, Line = 4001, IP = 10.10.153.132, Model = SCCP via gateway FXS

SIP Trunk between Unified CM1 & Unified CM2 has route pattern 615.XXXX that transforms to 890.XXXX which causes the call to be routed back to CM1 over a SIP Trunk in the other direction.

Scenario:
Phone A DN 4000 calls Phone B DN 4001 over the SIP Trunk by dialing 6154001
Phone B DN 4001 Answers
Phone B DN 4001 goes onHook
End of Scenario


SCCP-SCCP V.150 Fax Call via an Inter Cluster SIP Trunk Part 1 of 3
10.10.154.62
10.10.154.247
10.10.154.61
Unified CM1
10.10.154.247
Unified CM2
(d1) [1] REFER to 10.10.154.247 (remotecc: status line update)
(d2) [2] REFER to 10.10.154.247 (remotecc: status line update)
(d3) [3] INVITE 8904004
(d3) [4] 100 Trying
(d4) [5] INVITE 4004
(d4) [6] 100 Trying
(d3) [7] 404 Not Found
(d4) [8] ACK 4004
(d4) [9] 404 Not Found
(d3) [10] ACK 8904004
(d5) [11] INVITE 8904001
(d5) [12] 100 Trying
(d6) [13] INVITE 4001
(d6) [14] 100 Trying
(d6) [15] 180 Ringing
(d5) [16] 180 Ringing
(d6) [17] 200 OK (INVITE)
(d6) [18] SUBSCRIBE 4000 (kpml)
(d6) [19] NOTIFY to 4001 (kpml)
(d6) [20] 200 OK (SUBSCRIBE)
(d5) [21] SUBSCRIBE 4000 (kpml)
(d5) [22] 200 OK (INVITE)
(d6) [23] 200 OK (NOTIFY)
(d5) [24] NOTIFY to 8904001 (kpml)
(d5) [25] 200 OK (SUBSCRIBE)

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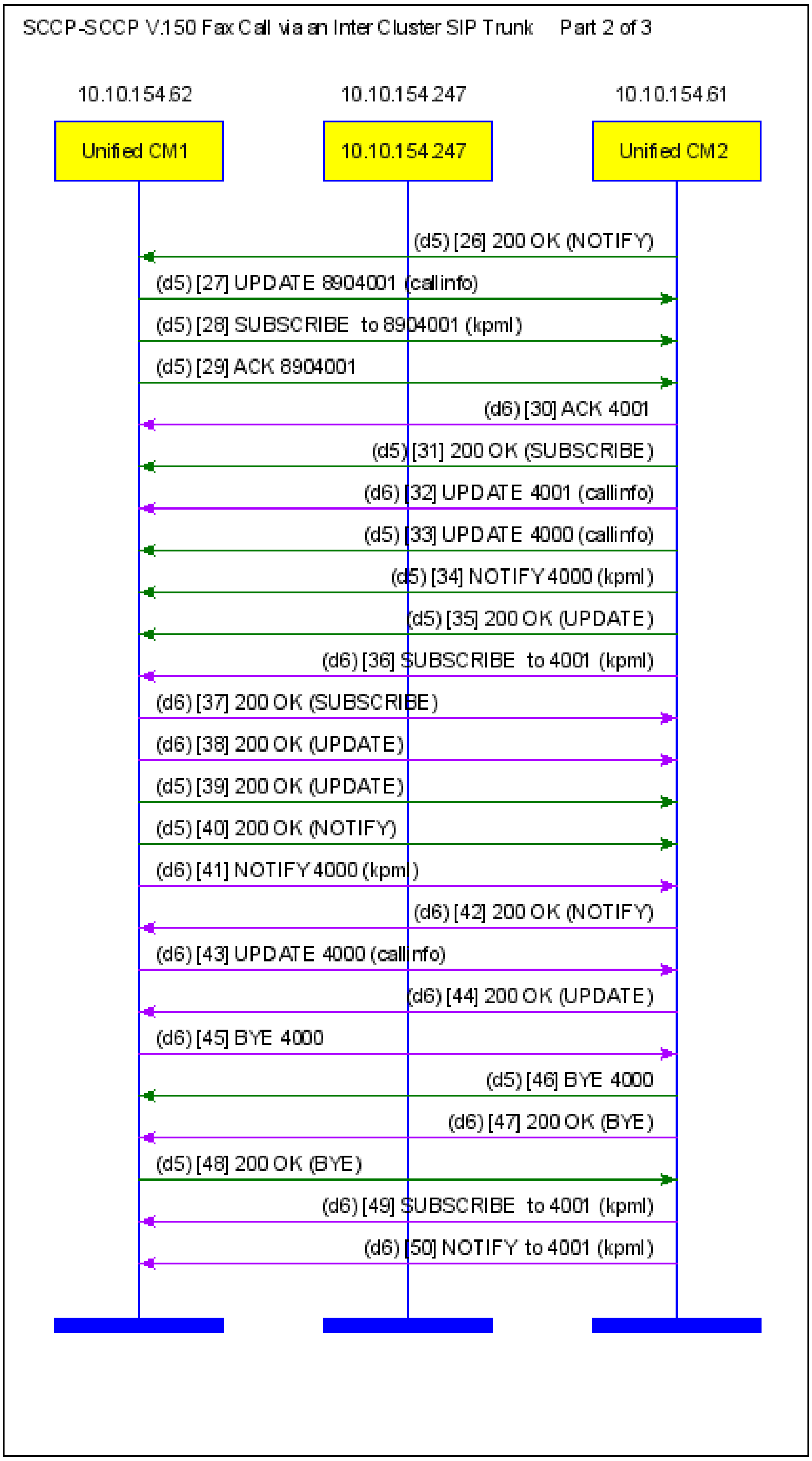

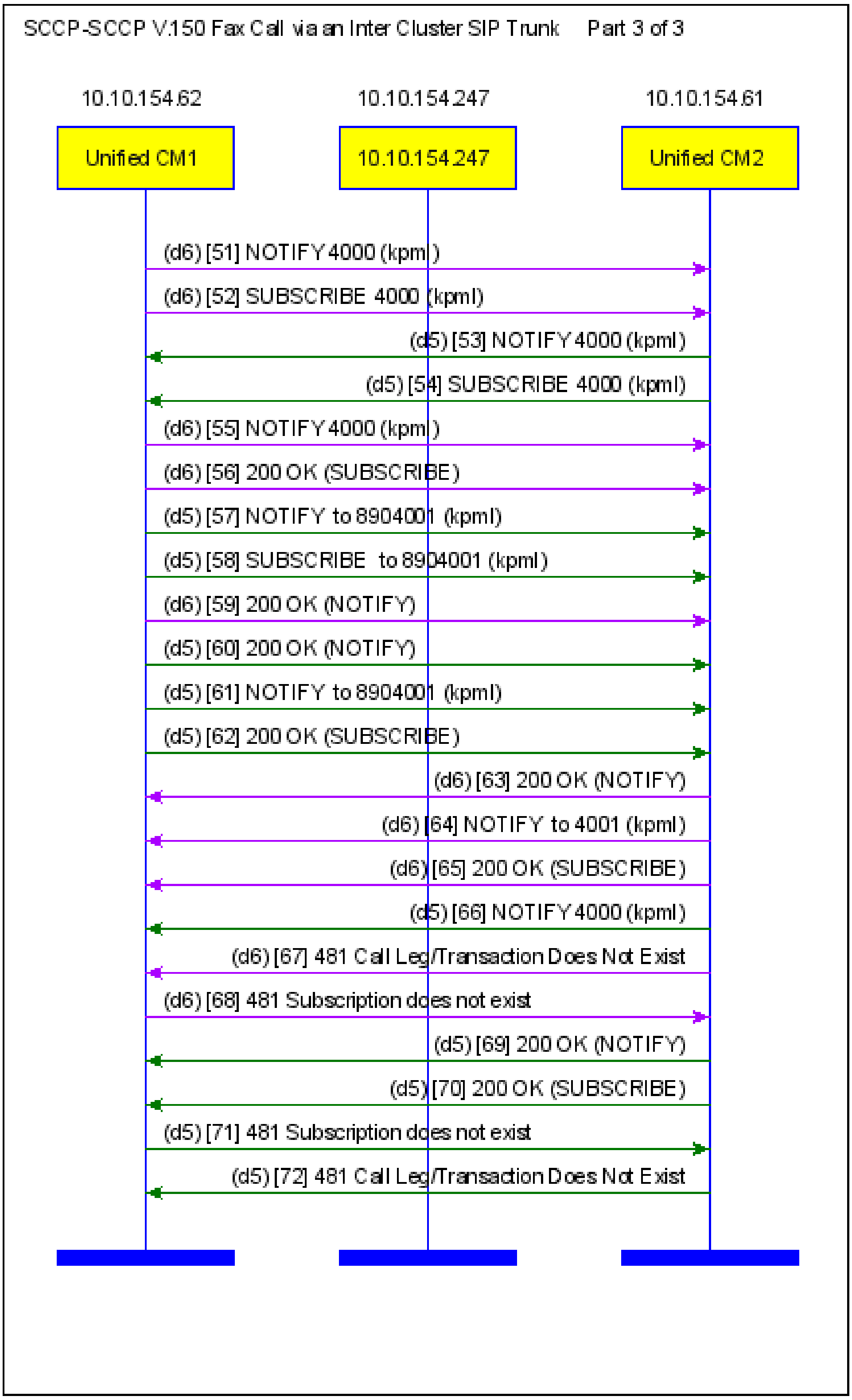
SCCP-SCCP V.150 Fax Call via an Inter Cluster SIP Trunk Part 3 of 3
10.10.154.62
10.10.154.247
10.10.154.61
Unified CM1
10.10.154.247
Unified CM2
(d6) [51] NOTIFY 4000 (kpml)
(d6) [52] SUBSCRIBE 4000 (kpml)
(d5) [53] NOTIFY 4000 (kpml)
(d5) [54] SUBSCRIBE 4000 (kpml)
(d6) [55] NOTIFY 4000 (kpml)
(d6) [56] 200 OK (SUBSCRIBE)
(d5) [57] NOTIFY to 8904001 (kpml)
(d5) [58] SUBSCRIBE to 8904001 (kpml)
(d6) [59] 200 OK (NOTIFY)
(d5) [60] 200 OK (NOTIFY)
(d5) [61] NOTIFY to 8904001 (kpml)
(d5) [62] 200 OK (SUBSCRIBE)
(d6) [63] 200 OK (NOTIFY)
(d6) [64] NOTIFY to 4001 (kpml)
(d6) [65] 200 OK (SUBSCRIBE)
(d5) [66] NOTIFY 4000 (kpml)
(d6) [67] 481 Call Leg/Transaction Does Not Exist
(d6) [68] 481 Subscription does not exist
(d5) [69] 200 OK (NOTIFY)
(d5) [70] 200 OK (SUBSCRIBE)
(d5) [71] 481 Subscription does not exist
(d5) [72] 481 Call Leg/Transaction Does Not Exist

```
[diagram]   Call-ID:[prev][next]
[1] REFER sip:10.10.154.247:50414 SIP/2.0
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2b61d29706c
From: <sip:10.10.154.62>;tag=60729825
To: <sip:10.10.154.247>
Call-ID: 68e8d200-daf1811d-29b-3e9a12ac@10.10.154.62
CSeq: 101 REFER
Max-Forwards: 70
Contact: <sip:10.10.154.62:5060;transport=tcp>
User-Agent: Cisco-CUCM8.6
Expires: 30
Refer-To: cid:1234567890@10.10.154.62
Content-Id: <1234567890@10.10.154.62>
Content-Type: application/x-cisco-remotecc-request+xml
Referred-By: <sip:10.10.154.62>
Content-Length: 265

<?xml version="1.0" encoding="iso-8859-1"?>
<x-cisco-remotecc-request>
<statuslineupdatereq>
<action>notify_display</action>
<statustext>U</statustext>
<displaytimeout>10</displaytimeout>
<priority>1</priority>
</statuslineupdatereq>
</x-cisco-remotecc-request>
```

```
[diagram]   Call-ID:[prev][next]
[2] REFER sip:10.10.154.247:52839 SIP/2.0
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2b75e397455
From: <sip:10.10.154.62>;tag=199975237
To: <sip:10.10.154.247>
Call-ID: bc5b2000-daf181a9-29c-3e9a12ac@10.10.154.62
CSeq: 101 REFER
Max-Forwards: 70
Contact: <sip:10.10.154.62:5060;transport=tcp>
User-Agent: Cisco-CUCM8.6
Expires: 30
Refer-To: cid:1234567890@10.10.154.62
Content-Id: <1234567890@10.10.154.62>
Content-Type: application/x-cisco-remotecc-request+xml
Referred-By: <sip:10.10.154.62>
Content-Length: 265

<?xml version="1.0" encoding="iso-8859-1"?>
<x-cisco-remotecc-request>
<statuslineupdatereq>
<action>notify_display</action>
<statustext>U</statustext>
<displaytimeout>10</displaytimeout>
<priority>1</priority>
</statuslineupdatereq>
</x-cisco-remotecc-request>
```

```
[diagram]   Call-ID:[prev][next]
[3] INVITE sip:8904004@10.10.154.61:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2b857618b59
From: <sip:4000@10.10.154.62>;tag=1342~90227f56-6e59-4152-9c5d-6d54ef528a66-28780378
To: <sip:8904004@10.10.154.61>
Date: Thu, 21 Apr 2011 01:01:18 GMT
Call-ID: dbf24880-daf181de-29d-3e9a12ac@10.10.154.62
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM8.6
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: <sip:10.10.154.62:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 3690088576-0000065536-0000000003-1050284716
Session-Expires:  1800
P-Asserted-Identity: <sip:4000@10.10.154.62>
Remote-Party-ID: <sip:4000@10.10.154.62>;party=calling;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
Max-Forwards: 70
Content-Length: 0
```

```
[diagram]   Call-ID:[prev][next]
[4] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2b857618b59
From: <sip:4000@10.10.154.62>;tag=1342~90227f56-6e59-4152-9c5d-6d54ef528a66-28780378
To: <sip:8904004@10.10.154.61>
Date: Thu, 21 Apr 2011 01:36:34 GMT
Call-ID: dbf24880-daf181de-29d-3e9a12ac@10.10.154.62
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[5] INVITE sip:4004@10.10.154.62:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK3a1041681f
From: <sip:4000@10.10.154.61>;tag=21~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895552
To: <sip:4004@10.10.154.62>
Date: Thu, 21 Apr 2011 01:36:34 GMT
Call-ID: c92e4280-daf18a22-6-3d9a12ac@10.10.154.61
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM8.6
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: <sip:10.10.154.61:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 3375252096-0000065536-0000000006-1033507500
Session-Expires:  1800
P-Asserted-Identity: <sip:4000@10.10.154.61>
Remote-Party-ID: <sip:4000@10.10.154.61>;party=calling;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
Max-Forwards: 69
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[6] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK3a1041681f
From: <sip:4000@10.10.154.61>;tag=21~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895552
To: <sip:4004@10.10.154.62>
Date: Thu, 21 Apr 2011 01:01:18 GMT
Call-ID: c92e4280-daf18a22-6-3d9a12ac@10.10.154.61
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[7] SIP/2.0 404 Not Found
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2b857618b59
From: <sip:4000@10.10.154.62>;tag=1342~90227f56-6e59-4152-9c5d-6d54ef528a66-28780378
To: <sip:8904004@10.10.154.61>;tag=20~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895551
Date: Thu, 21 Apr 2011 01:36:34 GMT
Call-ID: dbf24880-daf181de-29d-3e9a12ac@10.10.154.62
CSeq: 101 INVITE
Allow-Events: presence
Reason: Q.850;cause=1
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[8] ACK sip:4004@10.10.154.62:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK3a1041681f
From: <sip:4000@10.10.154.61>;tag=21~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895552
To: <sip:4004@10.10.154.62>;tag=1343~90227f56-6e59-4152-9c5d-6d54ef528a66-28780379
Date: Thu, 21 Apr 2011 01:36:34 GMT
Call-ID: c92e4280-daf18a22-6-3d9a12ac@10.10.154.61
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[9] SIP/2.0 404 Not Found
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK3a1041681f
From: <sip:4000@10.10.154.61>;tag=21~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895552
To: <sip:4004@10.10.154.62>;tag=1343~90227f56-6e59-4152-9c5d-6d54ef528a66-28780379
Date: Thu, 21 Apr 2011 01:01:18 GMT
Call-ID: c92e4280-daf18a22-6-3d9a12ac@10.10.154.61
CSeq: 101 INVITE
Allow-Events: presence
```

```
Reason: Q.850;cause=1
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[10] ACK sip:8904004@10.10.154.61:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2b857618b59
From: <sip:4000@10.10.154.62>;tag=1342~90227f56-6e59-4152-9c5d-6d54ef528a66-28780378
To: <sip:8904004@10.10.154.61>;tag=20~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895551
Date: Thu, 21 Apr 2011 01:01:18 GMT
Call-ID: dbf24880-daf181de-29d-3e9a12ac@10.10.154.62
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[11] INVITE sip:8904001@10.10.154.61:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2b9cf65eb9
From: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
To: <sip:8904001@10.10.154.61>
Date: Thu, 21 Apr 2011 01:01:26 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM8.6
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: <sip:10.10.154.62:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 3770088576-0000065536-0000000004-1050284716
Session-Expires:  1800
P-Asserted-Identity: <sip:4000@10.10.154.62>
Remote-Party-ID: <sip:4000@10.10.154.62>;party=calling;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
Max-Forwards: 70
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[12] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2b9cf65eb9
From: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
To: <sip:8904001@10.10.154.61>
Date: Thu, 21 Apr 2011 01:36:42 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[13] INVITE sip:4001@10.10.154.62:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK3b40529f2f
From: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
To: <sip:4001@10.10.154.62>
Date: Thu, 21 Apr 2011 01:36:42 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM8.6
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: <sip:10.10.154.61:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 3455252096-0000065536-0000000007-1033507500
Session-Expires:  1800
P-Asserted-Identity: <sip:4000@10.10.154.61>
Remote-Party-ID: <sip:4000@10.10.154.61>;party=calling;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
Max-Forwards: 69
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[14] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK3b40529f2f
From: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
```

```
To: <sip:4001@10.10.154.62>
Date: Thu, 21 Apr 2011 01:01:27 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[15] SIP/2.0 180 Ringing

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK3b40529f2f
From: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
To: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
Date: Thu, 21 Apr 2011 01:01:27 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4001@10.10.154.62>
Remote-Party-ID: <sip:4001@10.10.154.62>;party=called;screen=yes;privacy=off
Contact: <sip:4001@10.10.154.62:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[16] SIP/2.0 180 Ringing

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2b9cf65eb9
From: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
To: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
Date: Thu, 21 Apr 2011 01:36:42 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4001@10.10.154.61>
Remote-Party-ID: <sip:4001@10.10.154.61>;party=called;screen=yes;privacy=off
Contact: <sip:8904001@10.10.154.61:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[17] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK3b40529f2f
From: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
To: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
Date: Thu, 21 Apr 2011 01:01:27 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence, kpml
Supported: replaces
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Session-Expires:  1800;refresher=uas
Require:  timer
P-Asserted-Identity: <sip:4001@10.10.154.62>
Remote-Party-ID: <sip:4001@10.10.154.62>;party=called;screen=yes;privacy=off
Contact: <sip:4001@10.10.154.62:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 1161

v=0
o=CiscoSystemsCCM-SIP 1345 1 IN IP4 10.10.154.62
s=SIP Call
c=IN IP4 10.10.153.132
b=TIAS:64000
b=AS:64
t=0 0
m=audio 16572 RTP/SAVP 0 8 116 18 101 100 118 126
a=crypto:1 AES_CM_128_HMAC_SHA1_32 inline:7/AYPmkOMCiYfeSmw/zaBiDpQ/s0xVbnRf9yF3/E
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:8 PCMA/8000
a=ptime:20
a=rtpmap:116 iLBC/8000
a=ptime:20
a=maxptime:120
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
```

```
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15,32-35
a=rtpmap:100 X-NSE/8000
a=rtpmap:118 v150fw/8000
a=fmtp:118 1,3-4
a=rtpmap:126 NoAudio/8000
a=sqn:0
a=cdsc: 1 audio RTP/SAVP 0 8 116 18 101 100 118 126
a=cdsc: 9 audio udpsprt 120
a=cpar: a=sprtmap:120 v150mr/8000
a=cpar: a=fmtp:120 mr=1;mg=2;CDSCselect=1;jmdelay=yes;Versn=1.1;mrmods=1-5,7-8,10-11,13
a=cdsc: 10 image udptl t38
a=cpar: a=T38FaxVersion:3
a=cpar: a=T38MaxBitRate:33600
a=cpar: a=T38FaxFillBitRemoval:0
a=cpar: a=T38FaxTranscodingMMR:0
a=cpar: a=T38FaxTranscodingJBIG:0
a=cpar: a=T38FaxRateManagement:transferredTCF
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
a=cpar: a=T38FaxMaxBuffer:200
a=cpar: a=T38FaxMaxDatagram:320
a=vndpar:2 9 2 15
```

[diagram] Call-ID:[prev][next]
[18] SUBSCRIBE sip:4000@10.10.154.61:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2ba5a2de84
From: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
To: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 101 SUBSCRIBE
Date: Thu, 21 Apr 2011 01:01:29 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 7200
Contact: <sip:10.10.154.62:5060;transport=tcp>
Accept: application/kpml-response+xml
Max-Forwards: 70
Content-Type: application/kpml-request+xml
Content-Length: 370

<?xml version="1.0" encoding="UTF-8" ?>
<kpml-request xmlns="urn:ietf:params:xml:ns:kpml-request" xmlns:xsi="http://www.w3.org/2001/XMLSchema-
instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-request kpml-request.xsd" version="1.0">

  <pattern interdigittimer="7260000" persist="persist">
    <regex tag="dtmf">[x*#ABCD]</regex>
  </pattern>

</kpml-request>
```

[diagram] Call-ID:[prev][next]
[19] NOTIFY sip:10.10.154.62:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK3cd72f2d8
From: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
To: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 102 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 01:36:45 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: active;expires=7200
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.61>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[20] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2ba5a2de84
From: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
To: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
Date: Thu, 21 Apr 2011 01:36:45 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 101 SUBSCRIBE
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
Expires: 7200
Content-Length: 0
```

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[diagram] Call-ID:[prev][next]


```
[21] SUBSCRIBE sip:4000@10.10.154.62:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK3d6864a22b
From: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
To: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 101 SUBSCRIBE
Date: Thu, 21 Apr 2011 01:36:45 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 7200
Contact: <sip:10.10.154.61:5060;transport=tcp>
Accept: application/kpml-response+xml
Max-Forwards: 70
Content-Type: application/kpml-request+xml
Content-Length: 370

<?xml version="1.0" encoding="UTF-8" ?>
<kpml-request xmlns="urn:ietf:params:xml:ns:kpml-request" xmlns:xsi="http://www.w3.org/2001/XMLSchema-
instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-request kpml-request.xsd" version="1.0">

  <pattern interdigittimer="7260000" persist="persist">
    <regex tag="dtmf">[x*#ABCD]</regex>
  </pattern>

</kpml-request>
```


[diagram] Call-ID:[prev][next]


```
[22] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2b9cf65eb9
From: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
To: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
Date: Thu, 21 Apr 2011 01:36:42 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence, kpml
Supported: replaces
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Session-Expires:  1800;refresher=uas
Require:  timer
P-Asserted-Identity: <sip:4001@10.10.154.61>
Remote-Party-ID: <sip:4001@10.10.154.61>;party=called;screen=yes;privacy=off
Contact: <sip:8904001@10.10.154.61:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 1159

v=0
o=CiscoSystemsCCM-SIP 22 1 IN IP4 10.10.154.61
s=SIP Call
c=IN IP4 10.10.153.132
b=TIAS:64000
b=AS:64
t=0 0
m=audio 16572 RTP/SAVP 0 8 116 18 101 100 118 126
a=crypto:1 AES_CM_128_HMAC_SHA1_32 inline:7/AYPmkOMCiYfeSmw/zaBiDpQ/s0xVbnRf9yF3/E
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:8 PCMA/8000
a=ptime:20
a=rtpmap:116 iLBC/8000
a=ptime:20
a=maxptime:120
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15,32-35
a=rtpmap:100 X-NSE/8000
a=rtpmap:118 v150fw/8000
a=fmtp:118 1,3-4
a=rtpmap:126 NoAudio/8000
a=sqn:0
a=cdsc: 1 audio RTP/SAVP 0 8 116 18 101 100 118 126
a=cdsc: 9 audio udpsprt 120
a=cpar: a=sprtmap:120 v150mr/8000
a=cpar: a=fmtp:120 mr=1;mg=2;CDSCselect=1;jmdelay=yes;Versn=1.1;mrmods=1-5,7-8,10-11,13
a=cdsc: 10 image udptl t38
a=cpar: a=T38FaxVersion:3
```

```
a=cpar: a=T38MaxBitRate:33600
a=cpar: a=T38FaxFillBitRemoval:0
a=cpar: a=T38FaxTranscodingMMR:0
a=cpar: a=T38FaxTranscodingJBIG:0
a=cpar: a=T38FaxRateManagement:transferredTCF
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
a=cpar: a=T38FaxMaxBuffer:200
a=cpar: a=T38FaxMaxDatagram:320
a=vndpar:2 9 2 15

[diagram]   Call-ID:[prev][next]
[23] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK3cd72f2d8
From: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
To: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
Date: Thu, 21 Apr 2011 01:01:29 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 102 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[24] NOTIFY sip:10.10.154.61:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2bb442e51f3
From: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
To: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 102 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 01:01:29 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: active;expires=7200
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.62>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[25] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK3d6864a22b
From: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
To: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
Date: Thu, 21 Apr 2011 01:01:29 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 101 SUBSCRIBE
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
Expires: 7200
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[26] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2bb442e51f3
From: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
To: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
Date: Thu, 21 Apr 2011 01:36:45 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 102 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[27] UPDATE sip:8904001@10.10.154.61:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2be5446be3c
From: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
To: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
Date: Thu, 21 Apr 2011 01:01:26 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
Supported: timer,resource-priority,replaces
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 104 UPDATE
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; gci= 1-51009
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4000@10.10.154.62>
Remote-Party-ID: <sip:4000@10.10.154.62>;party=calling;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[28] SUBSCRIBE sip:10.10.154.61:5060;transport=tcp SIP/2.0
```

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2bd712b6329
From: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
To: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 103 SUBSCRIBE
Date: Thu, 21 Apr 2011 01:01:29 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 7200
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.62>
Accept: application/kpml-response+xml
Max-Forwards: 70
Content-Type: application/kpml-request+xml
Content-Length: 370

<?xml version="1.0" encoding="UTF-8" ?>
<kpml-request xmlns="urn:ietf:params:xml:ns:kpml-request" xmlns:xsi="http://www.w3.org/2001/XMLSchema-
instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-request kpml-request.xsd" version="1.0">

  <pattern interdigittimer="7260000" persist="persist">
    <regex tag="dtmf">[x*#ABCD]</regex>
  </pattern>

</kpml-request>
```

[diagram] Call-ID:[prev][next]
[29] ACK sip:8904001@10.10.154.61:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2bc7f925d2
From: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
To: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
Date: Thu, 21 Apr 2011 01:01:26 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Type: application/sdp
Content-Length: 557

v=0
o=CiscoSystemsCCM-SIP 1344 1 IN IP4 10.10.154.62
s=SIP Call
c=IN IP4 10.10.153.132
b=TIAS:64000
b=AS:64
t=0 0
m=audio 21076 RTP/SAVP 0 101 118
a=crypto:1 AES_CM_128_HMAC_SHA1_32 inline:/7rs1Mct/Na7Oa6kXC/yKDm/hXg0wJX9i7+CIOIy
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15,32-35
a=rtpmap:118 v150fw/8000
a=sqn:0
a=cdsc: 1 audio RTP/SAVP 0 101 118
a=cdsc: 4 audio udpsprt 120
a=cpar: a=sprtmap:120 v150mr/8000
a=cpar: a=fmtp:120 mr=1;mg=2;CDSCselect=1;jmdelay=yes;Versn=1.1;mrmods=1-5,7-8,10-11,13
```

[diagram] Call-ID:[prev][next]
[30] ACK sip:4001@10.10.154.62:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK3e380f260
From: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
To: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
Date: Thu, 21 Apr 2011 01:36:42 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Type: application/sdp
Content-Length: 555

v=0
o=CiscoSystemsCCM-SIP 23 1 IN IP4 10.10.154.61
s=SIP Call
c=IN IP4 10.10.153.132
b=TIAS:64000
b=AS:64
t=0 0
m=audio 21076 RTP/SAVP 0 101 118
```

```
a=crypto:1 AES_CM_128_HMAC_SHA1_32 inline:/7rs1Mct/Na7Oa6kXC/yKDm/hXg0wJX9i7+CIOIy
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15,32-35
a=rtpmap:118 v150fw/8000
a=sqn:0
a=cdsc: 1 audio RTP/SAVP 0 101 118
a=cdsc: 4 audio udpsprt 120
a=cpar: a=sprtmap:120 v150mr/8000
a=cpar: a=fmtp:120 mr=1;mg=2;CDSCselect=1;jmdelay=yes;Versn=1.1;mrmods=1-5,7-8,10-11,13
```

[diagram] Call-ID:[prev][next]
[31] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2bd712b6329
From: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
To: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
Date: Thu, 21 Apr 2011 01:36:45 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 103 SUBSCRIBE
Contact: <sip:10.10.154.61:5060;transport=tcp>
Expires: 7200
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[32] UPDATE sip:4001@10.10.154.62:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK42381509c5
From: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
To: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
Date: Thu, 21 Apr 2011 01:36:42 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
Supported: timer,resource-priority,replaces
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 104 UPDATE
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; gci= 1-33007
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4000@10.10.154.61>
Remote-Party-ID: <sip:4000@10.10.154.61>;party=calling;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[33] UPDATE sip:4000@10.10.154.62:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK411733f6cc
From: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
To: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
Date: Thu, 21 Apr 2011 01:36:45 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
Supported: timer,resource-priority,replaces
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 103 UPDATE
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= Encrypted; gci= 1-33007
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4001@10.10.154.61>
Remote-Party-ID: <sip:4001@10.10.154.61>;party=calling;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[34] NOTIFY sip:4000@10.10.154.62:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK407e27b4da
From: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
To: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 102 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 01:36:45 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: active;expires=7200
Contact: <sip:10.10.154.61:5060;transport=tcp>
Content-Length: 0
```

```
[diagram]   Call-ID:[prev][next]
[35] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2be5446be3c
From: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
To: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
Date: Thu, 21 Apr 2011 01:36:45 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 104 UPDATE
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4001@10.10.154.61>
Remote-Party-ID: <sip:4001@10.10.154.61>;party=called;screen=yes;privacy=off
Contact: <sip:8904001@10.10.154.61:5060;transport=tcp>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[36] SUBSCRIBE sip:10.10.154.62:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK3f5fa98b5f
From: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
To: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 103 SUBSCRIBE
Date: Thu, 21 Apr 2011 01:36:45 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 7200
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.61>
Accept: application/kpml-response+xml
Max-Forwards: 70
Content-Type: application/kpml-request+xml
Content-Length: 370

<?xml version="1.0" encoding="UTF-8" ?>
<kpml-request xmlns="urn:ietf:params:xml:ns:kpml-request" xmlns:xsi="http://www.w3.org/2001/XMLSchema-
instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-request kpml-request.xsd" version="1.0">

  <pattern interdigittimer="7260000" persist="persist">
    <regex tag="dtmf">[x*#ABCD]</regex>
  </pattern>

</kpml-request>

[diagram]   Call-ID:[prev][next]
[37] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK3f5fa98b5f
From: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
To: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
Date: Thu, 21 Apr 2011 01:01:29 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 103 SUBSCRIBE
Contact: <sip:10.10.154.62:5060;transport=tcp>
Expires: 7200
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[38] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK42381509c5
From: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
To: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
Date: Thu, 21 Apr 2011 01:01:29 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 104 UPDATE
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4001@10.10.154.62>
Remote-Party-ID: <sip:4001@10.10.154.62>;party=called;screen=yes;privacy=off
Contact: <sip:4001@10.10.154.62:5060;transport=tcp>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[39] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK411733f6cc
From: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
To: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
Date: Thu, 21 Apr 2011 01:01:29 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 103 UPDATE
```

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```
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4000@10.10.154.62>
Remote-Party-ID: <sip:4000@10.10.154.62>;party=called;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]

[40] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK407e27b4da
From: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
To: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
Date: Thu, 21 Apr 2011 01:01:29 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 102 NOTIFY
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]

[41] NOTIFY sip:4000@10.10.154.61:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2bfb35680b
From: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
To: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 102 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 01:01:29 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: active;expires=7200
Contact: <sip:10.10.154.62:5060;transport=tcp>
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]

[42] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2bfb35680b
From: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
To: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
Date: Thu, 21 Apr 2011 01:36:45 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 102 NOTIFY
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]

[43] UPDATE sip:4000@10.10.154.61:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2c07fdf26ff
From: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
To: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
Date: Thu, 21 Apr 2011 01:01:29 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
Supported: timer,resource-priority,replaces
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 103 UPDATE
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; gci= 1-51010
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4001@10.10.154.62>
Remote-Party-ID: <sip:4001@10.10.154.62>;party=calling;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]

[44] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2c07fdf26ff
From: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
To: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
Date: Thu, 21 Apr 2011 01:36:45 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 103 UPDATE
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4000@10.10.154.61>
Remote-Party-ID: <sip:4000@10.10.154.61>;party=called;screen=yes;privacy=off
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
Content-Length: 0
```

```
[diagram]   Call-ID:[prev][next]
[45] BYE sip:4000@10.10.154.61:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2c1695abacd
From: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
To: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
Date: Thu, 21 Apr 2011 01:01:29 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 104 BYE
Reason: Q.850;cause=16
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[46] BYE sip:4000@10.10.154.62:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK4361635266
From: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
To: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
Date: Thu, 21 Apr 2011 01:36:45 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 104 BYE
Reason: Q.850;cause=16
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[47] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2c1695abacd
From: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
To: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
Date: Thu, 21 Apr 2011 01:36:49 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 104 BYE
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[48] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK4361635266
From: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
To: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
Date: Thu, 21 Apr 2011 01:01:33 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 104 BYE
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[49] SUBSCRIBE sip:10.10.154.62:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK445af993e8
From: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
To: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 105 SUBSCRIBE
Date: Thu, 21 Apr 2011 01:36:49 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 0
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.61>
Max-Forwards: 70
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[50] NOTIFY sip:10.10.154.62:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK45197fbe9e
From: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
To: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 106 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 01:36:49 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: terminated
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.61>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[51] NOTIFY sip:4000@10.10.154.61:5060;transport=tcp SIP/2.0
```

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2c36cb6d775
From: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
To: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 106 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 01:01:33 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: terminated
Contact: <sip:10.10.154.62:5060;transport=tcp>
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]
[52] SUBSCRIBE sip:4000@10.10.154.61:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2c2968f080
From: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
To: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 105 SUBSCRIBE
Date: Thu, 21 Apr 2011 01:01:33 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 0
Contact: <sip:10.10.154.62:5060;transport=tcp>
Max-Forwards: 70
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]
[53] NOTIFY sip:4000@10.10.154.62:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK477e131994
From: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
To: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 106 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 01:36:49 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: terminated
Contact: <sip:10.10.154.61:5060;transport=tcp>
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]
[54] SUBSCRIBE sip:4000@10.10.154.62:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK466266d592
From: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
To: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 105 SUBSCRIBE
Date: Thu, 21 Apr 2011 01:36:49 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 0
Contact: <sip:10.10.154.61:5060;transport=tcp>
Max-Forwards: 70
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]
[55] NOTIFY sip:4000@10.10.154.61:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2c63946ab79
From: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
To: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 107 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 01:01:33 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: terminated;reason=timeout
Contact: <sip:10.10.154.62:5060;transport=tcp>
Content-Type: application/kpml-response+xml
Content-Length: 348

<?xml version="1.0" encoding="UTF-8" ?>
<kpml-response xmlns="urn:ietf:params:xml:ns:kpml-response" xmlns:xsi="http://www.w3.org/2001/XMLSchema-
instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-response kpml-response.xsd" code="487" digits=""
forced_flush="false" suppressed="false" tag="dtmf" text="Subscription Expired" version="1.0"/>
```

```
[diagram]   Call-ID:[prev][next]
[56] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK445af993e8
From: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
To: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
Date: Thu, 21 Apr 2011 01:01:33 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 105 SUBSCRIBE
Contact: <sip:10.10.154.62:5060;transport=tcp>
Expires: 0
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[57] NOTIFY sip:10.10.154.61:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2c520bb3183
From: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
To: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 106 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 01:01:33 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: terminated
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.62>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[58] SUBSCRIBE sip:10.10.154.61:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2c475fd9aa4
From: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
To: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 105 SUBSCRIBE
Date: Thu, 21 Apr 2011 01:01:33 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 0
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.62>
Max-Forwards: 70
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[59] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK45197fbe9e
From: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
To: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
Date: Thu, 21 Apr 2011 01:01:33 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 106 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[60] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK477e131994
From: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
To: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
Date: Thu, 21 Apr 2011 01:01:33 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 106 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[61] NOTIFY sip:10.10.154.61:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2c74b7e7f29
From: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
To: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 107 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 01:01:33 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: terminated;reason=timeout
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.62>
Content-Type: application/kpml-response+xml
Content-Length: 348
```

```
<?xml version="1.0" encoding="UTF-8" ?>
<kpml-response xmlns="urn:ietf:params:xml:ns:kpml-response" xmlns:xsi="http://www.w3.org/2001/XMLSchema-
instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-response kpml-response.xsd" code="487" digits=""
forced_flush="false" suppressed="false" tag="dtmf" text="Subscription Expired" version="1.0"/>
```


```
[diagram]   Call-ID:[prev][next]
[62] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK466266d592
From: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
To: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
Date: Thu, 21 Apr 2011 01:01:33 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 105 SUBSCRIBE
Contact: <sip:4000@10.10.154.62:5060;transport=tcp>
Expires: 0
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[63] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2c36cb6d775
From: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
To: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
Date: Thu, 21 Apr 2011 01:36:49 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 106 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[64] NOTIFY sip:10.10.154.62:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK4815a8b20d
From: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
To: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 107 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 01:36:49 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: terminated;reason=timeout
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
P-Asserted-Identity: <sip:4000@10.10.154.61>
Content-Type: application/kpml-response+xml
Content-Length: 348
```


```
<?xml version="1.0" encoding="UTF-8" ?>
<kpml-response xmlns="urn:ietf:params:xml:ns:kpml-response" xmlns:xsi="http://www.w3.org/2001/XMLSchema-
instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-response kpml-response.xsd" code="487" digits=""
forced_flush="false" suppressed="false" tag="dtmf" text="Subscription Expired" version="1.0"/>
```


```
[diagram]   Call-ID:[prev][next]
[65] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2c2968f080
From: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
To: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
Date: Thu, 21 Apr 2011 01:36:49 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 105 SUBSCRIBE
Contact: <sip:4000@10.10.154.61:5060;transport=tcp>
Expires: 0
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[66] NOTIFY sip:4000@10.10.154.62:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK4928b577cd
From: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
To: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 107 NOTIFY
Max-Forwards: 70
Date: Thu, 21 Apr 2011 01:36:49 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Subscription-State: terminated;reason=timeout
Contact: <sip:10.10.154.61:5060;transport=tcp>
Content-Type: application/kpml-response+xml
Content-Length: 348
```


```
<?xml version="1.0" encoding="UTF-8" ?>
<kpml-response xmlns="urn:ietf:params:xml:ns:kpml-response" xmlns:xsi="http://www.w3.org/2001/XMLSchema-
```

```
instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-response kpml-response.xsd" code="487" digits=""
forced_flush="false" suppressed="false" tag="dtmf" text="Subscription Expired" version="1.0"/>

[diagram]   Call-ID:[prev][next]
[67] SIP/2.0 481 Call Leg/Transaction Does Not Exist
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2c63946ab79
From: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
To: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
Date: Thu, 21 Apr 2011 01:36:49 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 107 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[68] SIP/2.0 481 Subscription does not exist
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK4815a8b20d
From: <sip:4000@10.10.154.61>;tag=23~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895554
To: <sip:4001@10.10.154.62>;tag=1345~90227f56-6e59-4152-9c5d-6d54ef528a66-28780384
Date: Thu, 21 Apr 2011 01:01:33 GMT
Call-ID: cdf2f680-daf18a2a-7-3d9a12ac@10.10.154.61
CSeq: 107 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[69] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2c520bb3183
From: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
To: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
Date: Thu, 21 Apr 2011 01:36:49 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 106 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[70] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2c475fd9aa4
From: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
To: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
Date: Thu, 21 Apr 2011 01:36:49 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 105 SUBSCRIBE
Contact: <sip:10.10.154.61:5060;transport=tcp>
Expires: 0
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[71] SIP/2.0 481 Subscription does not exist
Via: SIP/2.0/TCP 10.10.154.61:5060;branch=z9hG4bK4928b577cd
From: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
To: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
Date: Thu, 21 Apr 2011 01:01:33 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 107 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[72] SIP/2.0 481 Call Leg/Transaction Does Not Exist
Via: SIP/2.0/TCP 10.10.154.62:5060;branch=z9hG4bK2c74b7e7f29
From: <sip:4000@10.10.154.62>;tag=1344~90227f56-6e59-4152-9c5d-6d54ef528a66-28780383
To: <sip:8904001@10.10.154.61>;tag=22~4a8aee53-4967-4e5d-9507-09cb5b296e46-23895553
Date: Thu, 21 Apr 2011 01:36:49 GMT
Call-ID: e0b6fc80-daf181e6-29e-3e9a12ac@10.10.154.62
CSeq: 107 NOTIFY
Content-Length: 0
```

8. User-Agent/Server Header and Identity Header hostname pass-through

8.1 Basic incoming call from VCS E20 to CUCM E20

Title: Basic incoming call from VCS E20 to CUCM E20

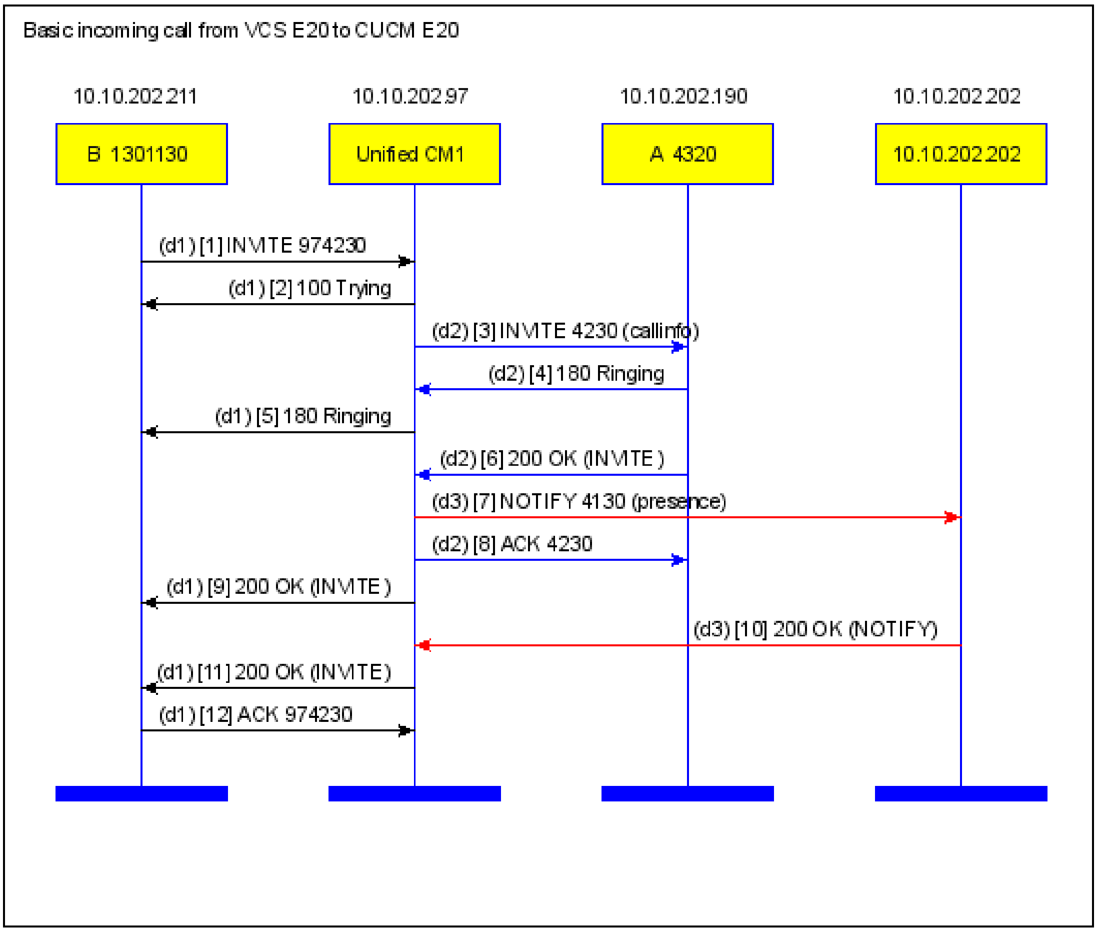
Description:
The following basic incoming call shows the pass-thru of the User-Agent/Server and calling party hostname received from VCS to the CUCM line device.

Configuration:
Node = Unified CM1, IP = 10.10.202.97
Node = VCS, IP = 10.10.202.130

Phone = A, Line = 4320, IP = 10.10.202.190, Model = SIP
Phone = B, Line = 1301130, IP = 10.10.202.211, Model = SIP

SIP Trunk between VCS and Unified CM with route pattern 130.xxxx

Scenario:
Phone B DN 1301130 behind VCS calls Phone A DN 4320 over a SIP Trunk
Phone A DN 4320 answers
End of Scenario


Basic incoming call from VCS E20 to CUCM E20
10.10.202.211
10.10.202.97
10.10.202.190
10.10.202.202
B 1301130
Unified CM1
A 4320
10.10.202.202
(d1) [1] INVITE 974230
(d1) [2] 100 Trying
(d2) [3] INVITE 4230 (callinfo)
(d2) [4] 180 Ringing
(d1) [5] 180 Ringing
(d2) [6] 200 OK (INVITE)
(d3) [7] NOTIFY 4130 (presence)
(d2) [8] ACK 4230
(d1) [9] 200 OK (INVITE)
(d3) [10] 200 OK (NOTIFY)
(d1) [11] 200 OK (INVITE)
(d1) [12] ACK 974230

```
[diagram]   Call-ID:[prev][next]
[1] INVITE sip:974230@10.10.202.97 SIP/2.0
Via: SIP/2.0/TCP 10.10.202.130:5060;egress-
zone=sipsigccm97;branch=z9hG4bK94faa003a1519a6bb12f601f7268d44a32893.38ad819390ee0c56f31e6454fa1b3263;proxy-
call-id=983bbd04-69ce-11e0-a3f3-0010f31d16ee;rport
Via: SIP/2.0/TCP
10.10.202.211:5060;branch=z9hG4bK9595504c2b4d92ca64a81add54806f0d.1;received=10.10.202.211;rport=53189;ingre
ss-zone=DefaultSubZone
Call-ID: 289990c6a567e373@10.10.202.211
CSeq: 100 INVITE
Contact: <sip:1301130@10.10.202.211:5060;transport=tcp>
From: <sip:1301130@vcs.domain>;tag=9d5cd8bbea2379e1
To: <sip:974230@vcs.domain>
Max-Forwards: 15
Record-Route: <sip:proxy-call-id=983bbd04-69ce-11e0-a3f3-0010f31d16ee@10.10.202.130:5060;transport=tcp;lr>
Record-Route: <sip:proxy-call-id=983bbd04-69ce-11e0-a3f3-0010f31d16ee@10.10.202.130:5060;transport=tcp;lr>
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Supported: replaces,timer,gruu,path,outbound
Session-Expires: 1800
P-Asserted-Identity: <sip:1301130@vcs.domain>
X-TAATag: 983bbdea-69ce-11e0-915a-0010f31d16ee
Content-Type: application/sdp
Content-Length: 2109

v=0
o=tandberg 27 1 IN IP4 10.10.202.211
s=-
c=IN IP4 10.10.202.211
b=AS:1152
t=0 0
m=audio 2326 RTP/AVP 100 101 102 9 18 11 8 0 103
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:101 G7221/16000
a=fmtp:101 bitrate=32000
a=rtpmap:102 G7221/16000
a=fmtp:102 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:11 L16/16000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
a=sendrecv
m=video 2328 RTP/AVP 97 98 99 34 31
b=TIAS:1152000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;max-fps=3000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;packetization-
mode=1;max-fps=3000
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1024,768,4;custom=1024,576,4;custom=800,600,4;cif4=2;custom=720,480,2;custom=640,480,2;custom=512,288
,1;cif=1;cu
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=2;cif=1;qcif=1;maxbr=10880
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=10880
a=rtcp-fb:* nack pli
a=sendrecv
a=content:main
a=label:11
a=answer:full
m=application 5070 UDP/BFCP *
a=floorctrl:c-s
a=confid:1
a=floorid:2 mstrm:12
a=userid:27
a=setup:actpass
a=connection:new
m=video 2330 RTP/AVP 97 98 99 34 31
b=TIAS:1152000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;max-fps=3000
a=rtpmap:98 H264/90000
```

```
a=fmtp:98 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;packetization-
mode=1;max-fps=3000
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1024,768,4;custom=1024,576,4;custom=800,600,4;cif4=2;custom=720,480,2;custom=640,480,2;custom=512,288
,1;cif=1;cu
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=2;cif=1;qcif=1;maxbr=10880
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=10880
a=rtcp-fb:* nack pli
a=sendrecv
a=content:slides
a=label:12
m=application 2332 RTP/AVP 104
a=rtpmap:104 H224/4800
a=sendrecv
```

[diagram]   Call-ID:[prev][next]
[2] SIP/2.0 100 Trying

```
Via: SIP/2.0/TCP 10.10.202.130:5060;egress-
zone=sipsigccm97;branch=z9hG4bK94faa003a1519a6bb12f601f7268d44a32893.38ad819390ee0c56f31e6454fa1b3263;proxy-
call-id=983bbd04-69ce-11e0-a3f3-0010f31d16ee;rport,SIP/2.0/TCP
10.10.202.211:5060;branch=z9hG4bK9595504c2b4d92ca64a81add54806f0d.1;received=10.10.202.211;rport=53189;ingre
ss-zone=DefaultSubZone
From: <sip:1301130@vcs.domain>;tag=9d5cd8bbea2379e1
To: <sip:974230@vcs.domain>
Date: Mon, 18 Apr 2011 15:14:45 GMT
Call-ID: 289990c6a567e373@10.10.202.211
CSeq: 100 INVITE
Allow-Events: presence
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]
[3] INVITE sip:4230@10.10.202.190:56641;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK58205ba413
From: <sip:1301130@vcs.domain>;tag=620~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379751
To: <sip:4230@10.10.202.97>
Date: Mon, 18 Apr 2011 15:14:45 GMT
Call-ID: 96752600-dac15565-22-61ca12ac@10.10.202.97
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= from; gci= 1-158005;
call-instance= 1
Send-Info: conference, x-cisco-conference
Alert-Info: <file://Bellcore-dr2/>
Remote-Party-ID: <sip:1301130@vcs.domain;x-cisco-callback-
number=1301130>;party=calling;screen=yes;privacy=off
Contact: <sip:1301130@10.10.202.97:5060;transport=tcp>;video;audio
Max-Forwards: 14
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]
[4] SIP/2.0 180 Ringing

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK58205ba413;received=10.10.202.97
Call-ID: 96752600-dac15565-22-61ca12ac@10.10.202.97
CSeq: 101 INVITE
From: <sip:1301130@vcs.domain>;tag=620~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379751
To: <sip:4230@10.10.202.97>;tag=b182382462867a1c
Server: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Remote-Party-ID: <sip:4230@10.10.202.97>;privacy=off;id-type=subscriber;screen=yes;party=calling
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]
[5] SIP/2.0 180 Ringing

```
Via: SIP/2.0/TCP 10.10.202.130:5060;egress-
zone=sipsigccm97;branch=z9hG4bK94faa003a1519a6bb12f601f7268d44a32893.38ad819390ee0c56f31e6454fa1b3263;proxy-
call-id=983bbd04-69ce-11e0-a3f3-0010f31d16ee;rport,SIP/2.0/TCP
10.10.202.211:5060;branch=z9hG4bK9595504c2b4d92ca64a81add54806f0d.1;received=10.10.202.211;rport=53189;ingre
ss-zone=DefaultSubZone
From: <sip:1301130@vcs.domain>;tag=9d5cd8bbea2379e1
To: <sip:974230@vcs.domain>;tag=619~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379750
Date: Mon, 18 Apr 2011 15:14:45 GMT
Call-ID: 289990c6a567e373@10.10.202.211
CSeq: 100 INVITE
```

```
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Record-Route: <sip:proxy-call-id=983bbd04-69ce-11e0-a3f3-
0010f31d16ee@10.10.202.130:5060;transport=tcp;lr>,<sip:proxy-call-id=983bbd04-69ce-11e0-a3f3-
0010f31d16ee@10.10.202.130:5060;transport=tcp;lr>
Server: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:4230@10.10.202.97>
Remote-Party-ID: <sip:4230@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:974230@10.10.202.97:5060;transport=tcp>
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]

[6] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK58205ba413;received=10.10.202.97
Call-ID: 96752600-dac15565-22-61ca12ac@10.10.202.97
CSeq: 101 INVITE
Contact: <sip:4230@10.10.202.190:56641;transport=tcp>
From: <sip:1301130@vcs.domain>;tag=620~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379751
To: <sip:4230@10.10.202.97>;tag=b182382462867a1c
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-service-
control,norefersub,extended-refer
Require: timer
Session-Expires: 1800;refresher=uas
Min-SE: 1800
Remote-Party-ID: <sip:4230@10.10.202.97>;privacy=off;id-type=subscriber;screen=yes;party=calling
Content-Type: application/sdp
Content-Length: 2176

v=0
o=tandberg 3 1 IN IP4 10.10.202.190
s=-
c=IN IP4 10.10.202.190
b=AS:1152
t=0 0
m=audio 16392 RTP/AVP 100 101 102 9 18 11 8 0 103
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:101 G7221/16000
a=fmtp:101 bitrate=32000
a=rtpmap:102 G7221/16000
a=fmtp:102 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:11 L16/16000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
a=sendrecv
m=video 16394 RTP/AVP 97 98 99 34 31
b=TIAS:1152000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;max-fps=3000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;packetization-
mode=1;max-fps=3000
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1024,768,4;custom=1024,576,4;custom=800,600,4;cif4=2;custom=720,480,2;custom=640,480,2;custom=512,288
,1;cif=1;custom=352,240,1;qcif=1;maxbr=10880
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=2;cif=1;qcif=1;maxbr=10880
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=10880
a=rtcp-fb:* nack pli
a=sendrecv
a=content:main
a=label:11
a=answer:full
m=application 5070 UDP/BFCP *
a=floorctrl:c-s
a=confid:1
a=floorid:2 mstrm:12
a=userid:3
```

```
a=setup:actpass
a=connection:new
m=video 16396 RTP/AVP 97 98 99 34 31
b=TIAS:1152000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;max-fps=3000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;packetization-
mode=1;max-fps=3000
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1024,768,4;custom=1024,576,4;custom=800,600,4;cif4=2;custom=720,480,2;custom=640,480,2;custom=512,288
,1;cif=1;custom=352,240,1;qcif=1;maxbr=10880
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=2;cif=1;qcif=1;maxbr=10880
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=10880
a=rtcp-fb:* nack pli
a=sendrecv
a=content:slides
a=label:12
m=application 16398 RTP/AVP 104
a=rtpmap:104 H224/4800
a=sendrecv
```

[diagram] Call-ID:[prev][next]

[7] NOTIFY sip:4130@10.10.202.202:36986;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK59360cd3d9
From: <sip:4230@10.10.202.97>;tag=1049761642
To: <sip:4130@10.10.202.97>;tag=2531604e22b4a804
Call-ID: 8573f8c53bb85a25@10.10.202.202
CSeq: 111 NOTIFY
Max-Forwards: 70
Date: Mon, 18 Apr 2011 15:14:46 GMT
User-Agent: Cisco-CUCM8.6
Event: presence
Subscription-State: active;expires=1049
Contact: <sip:4230@10.10.202.97:5060;transport=tcp>
Content-Type: application/pidf+xml
Content-Length: 826
```


```
<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<presence xmlns="urn:ietf:params:xml:ns:pidf" entity="sip:4230@10.10.202.97"
xmlns:e="urn:ietf:params:xml:ns:pidf:status:rpid" xmlns:dm="urn:ietf:params:xml:ns:pidf:data-model"
xmlns:ce="urn:cisco:params:xml:ns:pidf:rpid" xmlns:sc="urn:ietf:params:xml:ns:pidf:servcaps">
  <dm:person>
    <status>
    <basic>open</basic>
    </status>
    <e:activities>
      <e:on-the-phone/>
    </e:activities>
  </dm:person>
  <tuple id="cmp-1-288">
    <status>
      <basic>open</basic>
      <e:activities>
        <e:on-the-phone/>
      </e:activities>
    </status>
    <sc:servcaps>
      <sc:audio>true</sc:audio>
    </sc:servcaps>
    <contact priority="0.8">sip:4230@10.10.202.97:5060</contact>
    <timestamp>2011-04-18T15:14:46Z</timestamp>
  </tuple>
</presence>
```


[diagram] Call-ID:[prev][next]

[8] ACK sip:4230@10.10.202.190:56641;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK5a6c1aa506
From: <sip:1301130@vcs.domain>;tag=620~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379751
To: <sip:4230@10.10.202.97>;tag=b182382462867a1c
Date: Mon, 18 Apr 2011 15:14:45 GMT
Call-ID: 96752600-dac15565-22-61ca12ac@10.10.202.97
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 1043
```

```
v=0
o=CiscoSystemsCCM-SIP 620 1 IN IP4 10.10.202.97
s=SIP Call
b=AS:1152
t=0 0
m=audio 2326 RTP/AVP 100 103
c=IN IP4 10.10.202.211
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 bitrate=64000;profile-level-id=24;object=23
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
m=video 2328 RTP/AVP 98
c=IN IP4 10.10.202.211
b=TIAS:1088000
a=label:11
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-
smbps=40500;max-fps=3000
a=content:main
a=rtcp-fb:* nack pli
m=application 5070 UDP/BFCP *
c=IN IP4 10.10.202.211
a=floorctrl:s-only
a=floorid:2 mstrm:12
a=confid:1
a=userid:27
m=video 2330 RTP/AVP 98
c=IN IP4 10.10.202.211
b=TIAS:1088000
a=label:12
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-
smbps=40500;max-fps=3000
a=content:slides
a=rtcp-fb:* nack pli
m=application 2332 RTP/AVP 104
c=IN IP4 10.10.202.211
a=rtpmap:104 H224/0
```

[diagram]   Call-ID:[prev][next]
[9] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.202.130:5060;egress-
zone=sipsigccm97;branch=z9hG4bK94faa003a1519a6bb12f601f7268d44a32893.38ad819390ee0c56f31e6454fa1b3263;proxy-
call-id=983bbd04-69ce-11e0-a3f3-0010f31d16ee;rport,SIP/2.0/TCP
10.10.202.211:5060;branch=z9hG4bK9595504c2b4d92ca64a81add54806f0d.1;received=10.10.202.211;rport=53189;ingre
ss-zone=DefaultSubZone
From: <sip:1301130@vcs.domain>;tag=9d5cd8bbea2379e1
To: <sip:974230@vcs.domain>;tag=619~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379750
Date: Mon, 18 Apr 2011 15:14:45 GMT
Call-ID: 289990c6a567e373@10.10.202.211
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Record-Route: <sip:proxy-call-id=983bbd04-69ce-11e0-a3f3-
0010f31d16ee@10.10.202.130:5060;transport=tcp;lr>,<sip:proxy-call-id=983bbd04-69ce-11e0-a3f3-
0010f31d16ee@10.10.202.130:5060;transport=tcp;lr>
Supported: replaces
Server: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Session-Expires:  1800;refresher=uas
Require:  timer
P-Asserted-Identity: <sip:4230@10.10.202.97>
Remote-Party-ID: <sip:4230@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:974230@10.10.202.97:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 1046

v=0
o=CiscoSystemsCCM-SIP 619 1 IN IP4 10.10.202.97
s=SIP Call
b=AS:1152
t=0 0
m=audio 16392 RTP/AVP 100 103
c=IN IP4 10.10.202.190
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 bitrate=64000;profile-level-id=24;object=23
```

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```
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
m=video 16394 RTP/AVP 98
c=IN IP4 10.10.202.190
b=TIAS:1088000
a=label:11
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-smbps=40500;max-fps=3000
a=content:main
a=rtcp-fb:* nack pli
m=application 5070 UDP/BFCP *
c=IN IP4 10.10.202.190
a=floorctrl:c-only
a=floorid:2 mstrm:12
a=confid:1
a=userid:3
m=video 16396 RTP/AVP 98
c=IN IP4 10.10.202.190
b=TIAS:1088000
a=label:12
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-smbps=40500;max-fps=3000
a=content:slides
a=rtcp-fb:* nack pli
m=application 16398 RTP/AVP 104
c=IN IP4 10.10.202.190
a=rtpmap:104 H224/0
```

[diagram] Call-ID:[prev][next]

[10] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK59360cd3d9;received=10.10.202.97
Call-ID: 8573f8c53bb85a25@10.10.202.202
CSeq: 111 NOTIFY
From: <sip:4230@10.10.202.97>;tag=1049761642
To: <sip:4130@10.10.202.97>;tag=2531604e22b4a804
Server: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[11] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.202.130:5060;egress-zone=sipsigccm97;branch=z9hG4bK94faa003a1519a6bb12f601f7268d44a32893.38ad819390ee0c56f31e6454fa1b3263;proxy-call-id=983bbd04-69ce-11e0-a3f3-0010f31d16ee;rport,SIP/2.0/TCP 10.10.202.211:5060;branch=z9hG4bK9595504c2b4d92ca64a81add54806f0d.1;received=10.10.202.211;rport=53189;ingress-zone=DefaultSubZone
From: <sip:1301130@vcs.domain>;tag=9d5cd8bbea2379e1
To: <sip:974230@vcs.domain>;tag=619~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379750
Date: Mon, 18 Apr 2011 15:14:45 GMT
Call-ID: 289990c6a567e373@10.10.202.211
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Record-Route: <sip:proxy-call-id=983bbd04-69ce-11e0-a3f3-0010f31d16ee@10.10.202.130:5060;transport=tcp;lr>,<sip:proxy-call-id=983bbd04-69ce-11e0-a3f3-0010f31d16ee@10.10.202.130:5060;transport=tcp;lr>
Supported: replaces
Server: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Session-Expires:  1800;refresher=uas
Require:  timer
P-Asserted-Identity: <sip:4230@10.10.202.97>
Remote-Party-ID: <sip:4230@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:974230@10.10.202.97:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 1046

v=0
o=CiscoSystemsCCM-SIP 619 1 IN IP4 10.10.202.97
s=SIP Call
b=AS:1152
t=0 0
m=audio 16392 RTP/AVP 100 103
c=IN IP4 10.10.202.190
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 bitrate=64000;profile-level-id=24;object=23
a=rtpmap:103 telephone-event/8000
```

```
a=fmtp:103 0-15
m=video 16394 RTP/AVP 98
c=IN IP4 10.10.202.190
b=TIAS:1088000
a=label:11
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-smbps=40500;max-fps=3000
a=content:main
a=rtcp-fb:* nack pli
m=application 5070 UDP/BFCP *
c=IN IP4 10.10.202.190
a=floorctrl:c-only
a=floorid:2 mstrm:12
a=confid:1
a=userid:3
m=video 16396 RTP/AVP 98
c=IN IP4 10.10.202.190
b=TIAS:1088000
a=label:12
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-smbps=40500;max-fps=3000
a=content:slides
a=rtcp-fb:* nack pli
m=application 16398 RTP/AVP 104
c=IN IP4 10.10.202.190
a=rtpmap:104 H224/0
```

[diagram]   Call-ID:[prev][next]

[12] ACK sip:974230@10.10.202.97:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.202.130:5060;egress-zone=sipsigccm97;branch=z9hG4bK7b54a94a7c18bf069bc3fe6ae6676c4732894.38ad819390ee0c56f31e6454fa1b3263;proxy-call-id=983bbd04-69ce-11e0-a3f3-0010f31d16ee;rport
Via: SIP/2.0/TCP 10.10.202.211:5060;branch=z9hG4bK811b7581afe371e1e5a3fb1adb539592.1;received=10.10.202.211;rport=53189;ingress-zone=DefaultSubZone
Call-ID: 289990c6a567e373@10.10.202.211
CSeq: 100 ACK
From: <sip:1301130@vcs.domain>;tag=9d5cd8bbea2379e1
To: <sip:974230@vcs.domain>;tag=619~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379750
Max-Forwards: 69
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
X-TAATag: 983bbdea-69ce-11e0-915a-0010f31d16ee
Content-Length: 0
```

8.2 Basic outgoing call from CUCM E20 to VCS E20

Title: Basic outgoing call from CUCM E20 to VCS E20

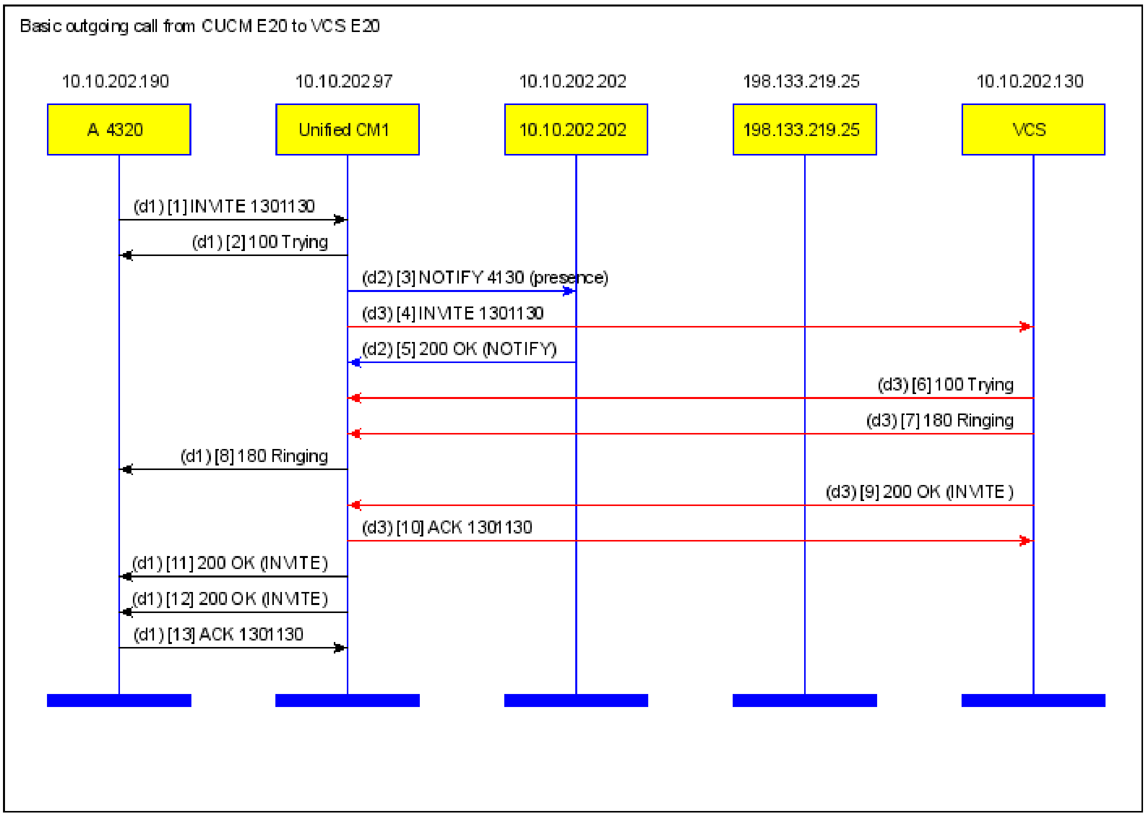
Description:
The following basic outgoing call shows the pass-thru of the User-Agent/Server and calling party hostname received from the CUCM E20 across the sip trunk to the VCS

Configuration:
Node = Unified CM1, IP = 10.10.202.97
Node = VCS, IP = 10.10.202.130

Phone = A, Line = 4320, IP = 10.10.202.190, Model = SIP
Phone = B, Line = 1301130, IP = 10.10.202.211, Model = SIP

SIP Trunk between VCS and Unified CM with route pattern 130.xxxx

Scenario:
Phone A DN 4320 calls Phone B DN 1301120 over sip trunk by dialing 1301130
Phone B DN 1301120 answers
End of Scenario


Basic outgoing call from CUCM E20 to VCS E20
10.10.202.190
10.10.202.97
10.10.202.202
198.133.219.25
10.10.202.130
A 4320
Unified CM1
10.10.202.202
198.133.219.25
VCS
(d1) [1] INVITE 1301130
(d1) [2] 100 Trying
(d2) [3] NOTIFY 4130 (presence)
(d3) [4] INVITE 1301130
(d2) [5] 200 OK (NOTIFY)
(d3) [6] 100 Trying
(d3) [7] 180 Ringing
(d1) [8] 180 Ringing
(d3) [9] 200 OK (INVITE)
(d3) [10] ACK 1301130
(d1) [11] 200 OK (INVITE)
(d1) [12] 200 OK (INVITE)
(d1) [13] ACK 1301130

```
[diagram]   Call-ID:[prev][next]
[1] INVITE sip:1301130@10.10.202.97 SIP/2.0
Via: SIP/2.0/TCP 10.10.202.190:5060;branch=z9hG4bK1954642a273b4d638aa6754e8851b4a0.1;rport
Call-ID: 0da9eecc75154a73@10.10.202.190
CSeq: 100 INVITE
Contact: <sip:4230@10.10.202.190:56641;transport=tcp>
From: <sip:4230@10.10.202.97>;tag=c6e5fb441e2ef339
To: <sip:1301130@10.10.202.97>
Max-Forwards: 70
Route: <sip:10.10.202.97;lr>
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-service-
control,norefersub,extended-refer
Session-Expires: 1800
Remote-Party-ID: <sip:4230@10.10.202.97>;privacy=off;id-type=subscriber;screen=yes;party=calling
Content-Type: application/sdp
Content-Length: 2176

v=0
o=tandberg 2 1 IN IP4 10.10.202.190
s=-
c=IN IP4 10.10.202.190
b=AS:1152
t=0 0
m=audio 16384 RTP/AVP 100 101 102 9 18 11 8 0 103
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:101 G7221/16000
a=fmtp:101 bitrate=32000
a=rtpmap:102 G7221/16000
a=fmtp:102 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:11 L16/16000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
a=sendrecv
m=video 16386 RTP/AVP 97 98 99 34 31
b=TIAS:1152000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;max-fps=3000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;packetization-
mode=1;max-fps=3000
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1024,768,4;custom=1024,576,4;custom=800,600,4;cif4=2;custom=720,480,2;custom=640,480,2;custom=512,288
,1;cif=1;custom=352,240,1;qcif=1;maxbr=10880
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=2;cif=1;qcif=1;maxbr=10880
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=10880
a=rtcp-fb:* nack pli
a=sendrecv
a=content:main
a=label:11
a=answer:full
m=application 5070 UDP/BFCP *
a=floorctrl:c-s
a=confid:1
a=floorid:2 mstrm:12
a=userid:2
a=setup:actpass
a=connection:new
m=video 16388 RTP/AVP 97 98 99 34 31
b=TIAS:1152000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;max-fps=3000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;packetization-
mode=1;max-fps=3000
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1024,768,4;custom=1024,576,4;custom=800,600,4;cif4=2;custom=720,480,2;custom=640,480,2;custom=512,288
,1;cif=1;custom=352,240,1;qcif=1;maxbr=10880
```

```
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=2;cif=1;qcif=1;maxbr=10880
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=10880
a=rtcp-fb:* nack pli
a=sendrecv
a=content:slides
a=label:12
m=application 16390 RTP/AVP 104
a=rtpmap:104 H224/4800
a=sendrecv
```

[diagram] Call-ID:[prev][next]
[2] SIP/2.0 100 Trying

```
Via: SIP/2.0/TCP 10.10.202.190:5060;branch=z9hG4bK1954642a273b4d638aa6754e8851b4a0.1;rport
From: <sip:4230@10.10.202.97>;tag=c6e5fb441e2ef339
To: <sip:1301130@10.10.202.97>
Date: Mon, 18 Apr 2011 15:07:48 GMT
Call-ID: 0da9eecc75154a73@10.10.202.190
CSeq: 100 INVITE
Allow-Events: presence
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[3] NOTIFY sip:4130@10.10.202.202:36986;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK5356ec4a31
From: <sip:4230@10.10.202.97>;tag=1049761642
To: <sip:4130@10.10.202.97>;tag=2531604e22b4a804
Call-ID: 8573f8c53bb85a25@10.10.202.202
CSeq: 109 NOTIFY
Max-Forwards: 70
Date: Mon, 18 Apr 2011 15:07:48 GMT
User-Agent: Cisco-CUCM8.6
Event: presence
Subscription-State: active;expires=1467
Contact: <sip:4230@10.10.202.97:5060;transport=tcp>
Content-Type: application/pidf+xml
Content-Length: 826
```



```
<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<presence xmlns="urn:ietf:params:xml:ns:pidf" entity="sip:4230@10.10.202.97"
xmlns:e="urn:ietf:params:xml:ns:pidf:status:rpid" xmlns:dm="urn:ietf:params:xml:ns:pidf:data-model"
xmlns:ce="urn:cisco:params:xml:ns:pidf:rpid" xmlns:sc="urn:ietf:params:xml:ns:pidf:servcaps">
  <dm:person>
    <status>
    <basic>open</basic>
    </status>
    <e:activities>
      <e:on-the-phone/>
    </e:activities>
  </dm:person>
  <tuple id="cmp-1-288">
    <status>
      <basic>open</basic>
      <e:activities>
        <e:on-the-phone/>
      </e:activities>
    </status>
    <sc:servcaps>
      <sc:audio>true</sc:audio>
    </sc:servcaps>
    <contact priority="0.8">sip:4230@10.10.202.97:5060</contact>
    <timestamp>2011-04-18T15:07:48Z</timestamp>
  </tuple>
</presence>
```


[diagram] Call-ID:[prev][next]
[4] INVITE sip:1301130@10.10.202.130:5060 SIP/2.0

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK546232e9
From: <sip:4230@cisco.com>;tag=569~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379749
To: <sip:1301130@10.10.202.130>
Date: Mon, 18 Apr 2011 15:07:48 GMT
Call-ID: 9de7ff80-dac153c4-21-61ca12ac@10.10.202.97
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
```

```
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Cisco-Guid: 2649227136-0000065536-0000000004-1640633004
Session-Expires:  1800
P-Asserted-Identity: <sip:4230@cisco.com>
Remote-Party-ID: <sip:4230@cisco.com>;party=calling;screen=yes;privacy=off
Contact: <sip:4230@10.10.202.97:5060;transport=tcp>;video;audio
Max-Forwards: 69
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[5] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK5356ec4a31;received=10.10.202.97
Call-ID: 8573f8c53bb85a25@10.10.202.202
CSeq: 109 NOTIFY
From: <sip:4230@10.10.202.97>;tag=1049761642
To: <sip:4130@10.10.202.97>;tag=2531604e22b4a804
Server: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[6] SIP/2.0 100 Trying

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK546232e9;received=10.10.202.97;ingress-zone=sipsigccm97
Call-ID: 9de7ff80-dac153c4-21-61ca12ac@10.10.202.97
CSeq: 101 INVITE
From: <sip:4230@cisco.com>;tag=569~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379749
To: <sip:1301130@10.10.202.130>
Server: TANDBERG/4100 (X6.1)
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[7] SIP/2.0 180 Ringing

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK546232e9;received=10.10.202.97;ingress-zone=sipsigccm97
Call-ID: 9de7ff80-dac153c4-21-61ca12ac@10.10.202.97
CSeq: 101 INVITE
From: <sip:4230@cisco.com>;tag=569~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379749
To: <sip:1301130@10.10.202.130>;tag=7525f8157159188b
Server: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[8] SIP/2.0 180 Ringing

```
Via: SIP/2.0/TCP 10.10.202.190:5060;branch=z9hG4bK1954642a273b4d638aa6754e8851b4a0.1;rport
From: <sip:4230@10.10.202.97>;tag=c6e5fb441e2ef339
To: <sip:1301130@10.10.202.97>;tag=568~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379748
Date: Mon, 18 Apr 2011 15:07:48 GMT
Call-ID: 0da9eecc75154a73@10.10.202.190
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Server: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; ui-state= ringout;
gci= 1-158004; call-instance= 1
Send-Info: conference, x-cisco-conference
Remote-Party-ID: <sip:1301130@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:1301130@10.10.202.97:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]

[9] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK546232e9;received=10.10.202.97;ingress-zone=sipsigccm97
Call-ID: 9de7ff80-dac153c4-21-61ca12ac@10.10.202.97
CSeq: 101 INVITE
Contact: <sip:1301130@10.10.202.211:5060;transport=tcp>
From: <sip:4230@cisco.com>;tag=569~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379749
To: <sip:1301130@10.10.202.130>;tag=7525f8157159188b
Record-Route: <sip:proxy-call-id=9f61b490-69cd-11e0-88e4-0010f31d16ee@10.10.202.130:5060;transport=tcp;lr>
Record-Route: <sip:proxy-call-id=9f61b490-69cd-11e0-88e4-0010f31d16ee@10.10.202.130:5060;transport=tcp;lr>
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Require: timer
Session-Expires: 1800;refresher=uas
Min-SE: 1800
Content-Type: application/sdp
Content-Length: 2109

v=0
o=tandberg 26 1 IN IP4 10.10.202.211
s=-
```

```
c=IN IP4 10.10.202.211
b=AS:1152
t=0 0
m=audio 2350 RTP/AVP 100 101 102 9 18 11 8 0 103
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:101 G7221/16000
a=fmtp:101 bitrate=32000
a=rtpmap:102 G7221/16000
a=fmtp:102 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:11 L16/16000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
a=sendrecv
m=video 2352 RTP/AVP 97 98 99 34 31
b=TIAS:1152000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;max-fps=3000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;packetization-
mode=1;max-fps=3000
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1024,768,4;custom=1024,576,4;custom=800,600,4;cif4=2;custom=720,480,2;custom=640,480,2;custom=512,288
,1;cif=1;cu
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=2;cif=1;qcif=1;maxbr=10880
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=10880
a=rtcp-fb:* nack pli
a=sendrecv
a=content:main
a=label:11
a=answer:full
m=application 5070 UDP/BFCP *
a=floorctrl:c-s
a=confid:1
a=floorid:2 mstrm:12
a=userid:26
a=setup:actpass
a=connection:new
m=video 2354 RTP/AVP 97 98 99 34 31
b=TIAS:1152000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;max-fps=3000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;packetization-
mode=1;max-fps=3000
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1024,768,4;custom=1024,576,4;custom=800,600,4;cif4=2;custom=720,480,2;custom=640,480,2;custom=512,288
,1;cif=1;cu
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=2;cif=1;qcif=1;maxbr=10880
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=10880
a=rtcp-fb:* nack pli
a=sendrecv
a=content:slides
a=label:12
m=application 2356 RTP/AVP 104
a=rtpmap:104 H224/4800
a=sendrecv
```

[diagram]   Call-ID:[prev][next]

[10] ACK sip:1301130@10.10.202.211:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bK554f2e75c5
From: <sip:4230@cisco.com>;tag=569~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379749
To: <sip:1301130@10.10.202.130>;tag=7525f8157159188b
Date: Mon, 18 Apr 2011 15:07:48 GMT
Call-ID: 9de7ff80-dac153c4-21-61ca12ac@10.10.202.97
Route: <sip:proxy-call-id=9f61b490-69cd-11e0-88e4-
0010f31d16ee@10.10.202.130:5060;transport=tcp;lr>,<sip:proxy-call-id=9f61b490-69cd-11e0-88e4-
0010f31d16ee@10.10.202.130:5060;transport=tcp;lr>
```

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```
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 1046

v=0
o=CiscoSystemsCCM-SIP 569 1 IN IP4 10.10.202.97
s=SIP Call
b=AS:1152
t=0 0
m=audio 16384 RTP/AVP 100 103
c=IN IP4 10.10.202.190
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 bitrate=64000;profile-level-id=24;object=23
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
m=video 16386 RTP/AVP 98
c=IN IP4 10.10.202.190
b=TIAS:1088000
a=label:11
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-
smbps=40500;max-fps=3000
a=content:main
a=rtcp-fb:* nack pli
m=application 5070 UDP/BFCP *
c=IN IP4 10.10.202.190
a=floorctrl:s-only
a=floorid:2 mstrm:12
a=confid:1
a=userid:2
m=video 16388 RTP/AVP 98
c=IN IP4 10.10.202.190
b=TIAS:1088000
a=label:12
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-
smbps=40500;max-fps=3000
a=content:slides
a=rtcp-fb:* nack pli
m=application 16390 RTP/AVP 104
c=IN IP4 10.10.202.190
a=rtpmap:104 H224/0
```

[diagram] Call-ID:[prev][next]

[11] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.202.190:5060;branch=z9hG4bK1954642a273b4d638aa6754e8851b4a0.1;rport
From: <sip:4230@10.10.202.97>;tag=c6e5fb441e2ef339
To: <sip:1301130@10.10.202.97>;tag=568~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379748
Date: Mon, 18 Apr 2011 15:07:48 GMT
Call-ID: 0da9eecc75154a73@10.10.202.190
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Server: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; gci= 1-158004;
call-instance= 1
Send-Info: conference, x-cisco-conference
Session-Expires:  1800;refresher=uas
Require:  timer
Remote-Party-ID: <sip:1301130@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:1301130@10.10.202.97:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 1043

v=0
o=CiscoSystemsCCM-SIP 568 1 IN IP4 10.10.202.97
s=SIP Call
b=AS:1152
t=0 0
m=audio 2350 RTP/AVP 100 103
c=IN IP4 10.10.202.211
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 bitrate=64000;profile-level-id=24;object=23
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
```

```
m=video 2352 RTP/AVP 98
c=IN IP4 10.10.202.211
b=TIAS:1088000
a=label:11
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-
smbps=40500;max-fps=3000
a=content:main
a=rtcp-fb:* nack pli
m=application 5070 UDP/BFCP *
c=IN IP4 10.10.202.211
a=floorctrl:c-only
a=floorid:2 mstrm:12
a=confid:1
a=userid:26
m=video 2354 RTP/AVP 98
c=IN IP4 10.10.202.211
b=TIAS:1088000
a=label:12
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-
smbps=40500;max-fps=3000
a=content:slides
a=rtcp-fb:* nack pli
m=application 2356 RTP/AVP 104
c=IN IP4 10.10.202.211
a=rtpmap:104 H224/0
```

[diagram]   Call-ID:[prev][next]

[12] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.202.190:5060;branch=z9hG4bK1954642a273b4d638aa6754e8851b4a0.1;rport
From: <sip:4230@10.10.202.97>;tag=c6e5fb441e2ef339
To: <sip:1301130@10.10.202.97>;tag=568~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379748
Date: Mon, 18 Apr 2011 15:07:48 GMT
Call-ID: 0da9eecc75154a73@10.10.202.190
CSeq: 100 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Server: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; gci= 1-158004;
call-instance= 1
Send-Info: conference, x-cisco-conference
Session-Expires:  1800;refresher=uas
Require:  timer
Remote-Party-ID: <sip:1301130@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:1301130@10.10.202.97:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 1043

v=0
o=CiscoSystemsCCM-SIP 568 1 IN IP4 10.10.202.97
s=SIP Call
b=AS:1152
t=0 0
m=audio 2350 RTP/AVP 100 103
c=IN IP4 10.10.202.211
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 bitrate=64000;profile-level-id=24;object=23
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
m=video 2352 RTP/AVP 98
c=IN IP4 10.10.202.211
b=TIAS:1088000
a=label:11
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-
smbps=40500;max-fps=3000
a=content:main
a=rtcp-fb:* nack pli
m=application 5070 UDP/BFCP *
c=IN IP4 10.10.202.211
a=floorctrl:c-only
a=floorid:2 mstrm:12
a=confid:1
a=userid:26
m=video 2354 RTP/AVP 98
c=IN IP4 10.10.202.211
b=TIAS:1088000
```

```
a=label:12
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-smbps=40500;max-fps=3000
a=content:slides
a=rtcp-fb:* nack pli
m=application 2356 RTP/AVP 104
c=IN IP4 10.10.202.211
a=rtpmap:104 H224/0
```

[diagram]   Call-ID:[prev][next]

```
[13] ACK sip:1301130@10.10.202.97:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.202.190:5060;branch=z9hG4bKafbc6c936b93305840aaf306403fb746.1;rport
Call-ID: 0da9eecc75154a73@10.10.202.190
CSeq: 100 ACK
From: <sip:4230@10.10.202.97>;tag=c6e5fb441e2ef339
To: <sip:1301130@10.10.202.97>;tag=568~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379748
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
User-Agent: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-service-control,norefersub,extended-refer
Content-Length: 0
```

8.3 Basic outgoing call from CUCM RT SIP phone to VCS E20

Title: Basic outgoing call from CUCM RT SIP phone to VCS E20

Description:
The following basic outgoing call shows the Identity headers used by CUCM being built using the organizational top-level domain configuration 'cisco.com'

Configuration:
Node = Unified CM1, IP = 10.10.202.97
Node = VCS, IP = 10.10.202.130

Phone = A, Line = 4210, IP = 10.10.202.137, Model=SIP
Phone = B, Line = 1301130, IP = 10.10.202.211, Model = SIP

SIP Trunk between VCS and Unified CM with route pattern 130.xxxx

Scenario:
Phone A DN 4210 calls Phone B DN 1301130 over a SIP Trunk
Phone B DN 1301130 answers
End of Scenario

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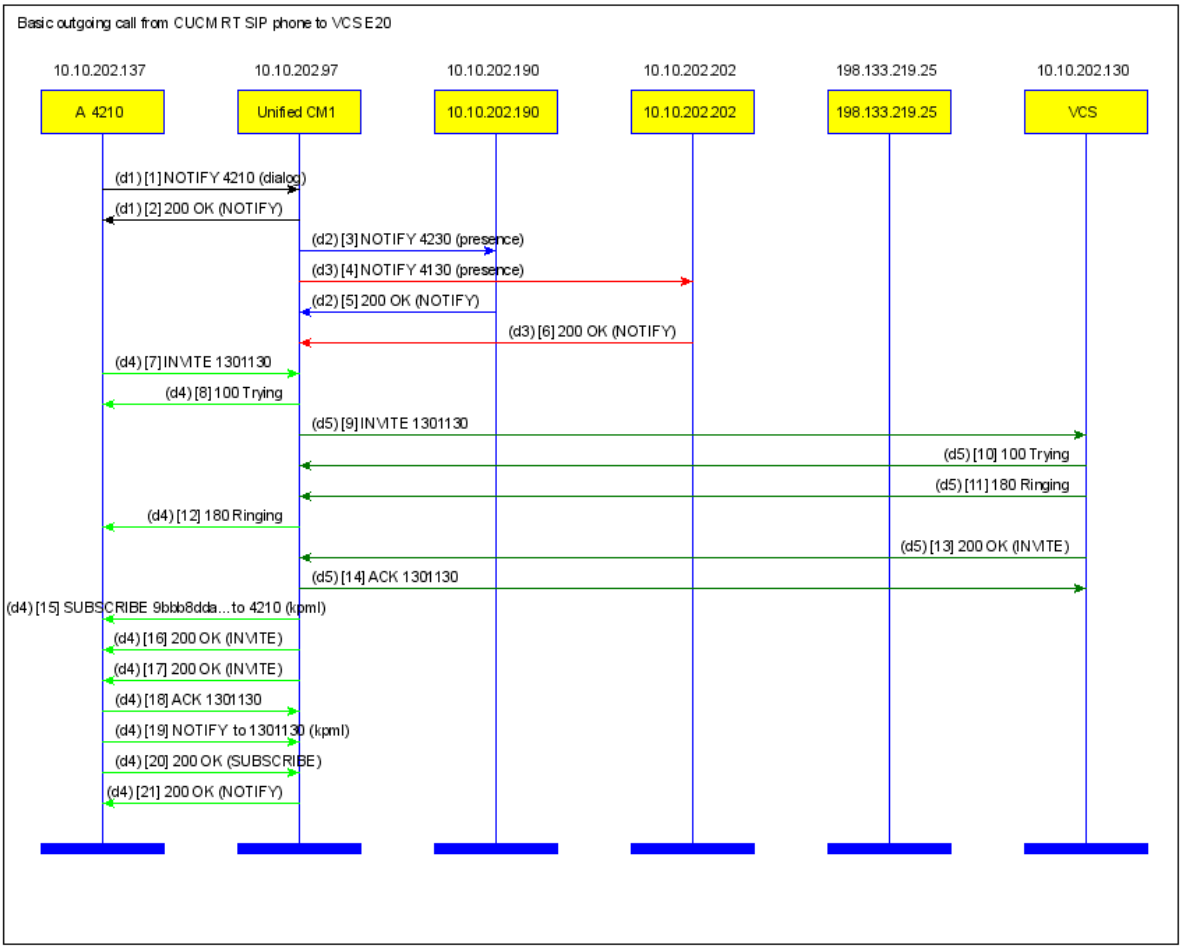

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```
[diagram]   Call-ID:[prev][next]
[1] NOTIFY sip:4210@10.10.202.97 SIP/2.0
Via: SIP/2.0/TCP 10.10.202.137:51547;branch=z9hG4bK54d24580
To: <sip:4210@10.10.202.97>
From: <sip:4210@10.10.202.97>;tag=b4a4e3289b94346a1b6c3789-443ec00e
Call-ID: 5bad7dc0-20eed55b@10.10.202.137
Date: Mon, 18 Apr 2011 17:31:13 GMT
CSeq: 15 NOTIFY
Event: dialog
Subscription-State: active
Max-Forwards: 70
Contact: <sip:9bbb8dda-00ee-701f-1b73-470a40f63088@10.10.202.137:51547;transport=TCP>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE
Content-Length: 358
Content-Type: application/dialog-info+xml
Content-Disposition: session;handling=required

<?xml version="1.0" encoding="UTF-8" ?>
<dialog-info xmlns:call="urn:x-cisco:parmams:xml:ns:dialog-info:dialog:callinfo-dialog" version="0"
state="partial" entity="sip:4210@10.10.202.137">
<dialog id="27" call-id="b4a4e328-9b94001d-4439f603-2fcb5039@10.10.202.137" local-
tag="b4a4e3289b9434697e410cf2-6a191674"><state>trying</state></dialog>
</dialog-info>

[diagram]   Call-ID:[prev][next]
[2] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.202.137:51547;branch=z9hG4bK54d24580
From: <sip:4210@10.10.202.97>;tag=b4a4e3289b94346a1b6c3789-443ec00e
To: <sip:4210@10.10.202.97>;tag=1475921306
Date: Mon, 18 Apr 2011 17:31:13 GMT
Call-ID: 5bad7dc0-20eed55b@10.10.202.137
CSeq: 15 NOTIFY
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[3] NOTIFY sip:4230@10.10.202.190:56641;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bKd446823233
From: <sip:4210@10.10.202.97>;tag=1253272411
To: <sip:4230@10.10.202.97>;tag=7a601a28ee9ab87c
Call-ID: 0199fd076e122f5c@10.10.202.190
CSeq: 102 NOTIFY
Max-Forwards: 70
Date: Mon, 18 Apr 2011 17:31:13 GMT
User-Agent: Cisco-CUCM8.6
Event: presence
Subscription-State: active;expires=2883
Contact: <sip:4210@10.10.202.97:5060;transport=tcp>
Content-Type: application/pidf+xml
Content-Length: 827

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<presence xmlns="urn:ietf:params:xml:ns:pidf" entity="sip:4210@10.10.202.97"
xmlns:e="urn:ietf:params:xml:ns:pidf:status:rpid" xmlns:dm="urn:ietf:params:xml:ns:pidf:data-model"
xmlns:ce="urn:cisco:params:xml:ns:pidf:rpid" xmlns:sc="urn:ietf:params:xml:ns:pidf:servcaps">
  <dm:person>
    <status>
    <basic>open</basic>
    </status>
    <e:activities>
      <e:on-the-phone/>
    </e:activities>
  </dm:person>
  <tuple id="cmp-1-1503">
    <status>
      <basic>open</basic>
      <e:activities>
        <e:on-the-phone/>
      </e:activities>
    </status>
    <sc:servcaps>
      <sc:audio>true</sc:audio>
    </sc:servcaps>
    <contact priority="0.8">sip:4210@10.10.202.97:5060</contact>
    <timestamp>2011-04-18T17:31:13Z</timestamp>
  </tuple>
</presence>

[diagram]   Call-ID:[prev][next]
[4] NOTIFY sip:4130@10.10.202.202:36986;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bKd360156950
```

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```
From: <sip:4210@10.10.202.97>;tag=1067922494
To: <sip:4130@10.10.202.97>;tag=10c3b780ba54b1ce
Call-ID: b588857ce43b544b@10.10.202.202
CSeq: 103 NOTIFY
Max-Forwards: 70
Date: Mon, 18 Apr 2011 17:31:13 GMT
User-Agent: Cisco-CUCM8.6
Event: presence
Subscription-State: active;expires=1196
Contact: <sip:4210@10.10.202.97:5060;transport=tcp>
Content-Type: application/pidf+xml
Content-Length: 827
```


```
<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<presence xmlns="urn:ietf:params:xml:ns:pidf" entity="sip:4210@10.10.202.97"
xmlns:e="urn:ietf:params:xml:ns:pidf:status:rpid" xmlns:dm="urn:ietf:params:xml:ns:pidf:data-model"
xmlns:ce="urn:cisco:params:xml:ns:pidf:rpid" xmlns:sc="urn:ietf:params:xml:ns:pidf:servcaps">
  <dm:person>
    <status>
    <basic>open</basic>
    </status>
    <e:activities>
      <e:on-the-phone/>
    </e:activities>
  </dm:person>
  <tuple id="cmp-1-1308">
    <status>
      <basic>open</basic>
      <e:activities>
        <e:on-the-phone/>
      </e:activities>
    </status>
    <sc:servcaps>
      <sc:audio>true</sc:audio>
    </sc:servcaps>
    <contact priority="0.8">sip:4210@10.10.202.97:5060</contact>
    <timestamp>2011-04-18T17:31:13Z</timestamp>
  </tuple>
</presence>
```


[diagram]   Call-ID:[prev][next]

[5] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bKd446823233;received=10.10.202.97
Call-ID: 0199fd076e122f5c@10.10.202.190
CSeq: 102 NOTIFY
From: <sip:4210@10.10.202.97>;tag=1253272411
To: <sip:4230@10.10.202.97>;tag=7a601a28ee9ab87c
Server: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]

[6] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bKd360156950;received=10.10.202.97
Call-ID: b588857ce43b544b@10.10.202.202
CSeq: 103 NOTIFY
From: <sip:4210@10.10.202.97>;tag=1067922494
To: <sip:4130@10.10.202.97>;tag=10c3b780ba54b1ce
Server: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Content-Length: 0
```

[diagram]   Call-ID:[prev][next]

[7] INVITE sip:1301130@10.10.202.97 SIP/2.0

```
Via: SIP/2.0/TCP 10.10.202.137:51547;branch=z9hG4bK15418c29
From: "4210" <sip:4210@10.10.202.97>;tag=b4a4e3289b9434697e410cf2-6a191674
To: <sip:1301130@10.10.202.97>
Call-ID: b4a4e328-9b94001d-4439f603-2fcb5039@10.10.202.137
Max-Forwards: 70
Date: Mon, 18 Apr 2011 17:31:13 GMT
CSeq: 101 INVITE
User-Agent: Cisco-CP9971/9.2.1
Contact: <sip:9bbb8dda-00ee-701f-1b73-470a40f63088@10.10.202.137:51547;transport=tcp>
Expires: 180
Accept: application/sdp
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "4210" <sip:4210@10.10.202.97>;party=calling;id-type=subscriber;privacy=off;screen=yes
Supported: replaces,join,sdp-anat,norefersub,extended-refer,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-
escapecodes,X-cisco-service-control,X-cisco-srtp-fallback,X-cisco-monrec,X-cisco-config,X-cisco-sis-5.1.0,X-
cisco-xsi-8.0.1
Allow-Events: kpml,dialog
Content-Length: 788
```

```
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 16345 0 IN IP4 10.10.202.137
s=SIP Call
t=0 0
m=audio 26640 RTP/AVP 0 8 18 102 9 116 124 101
c=IN IP4 10.10.202.137
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:102 L16/16000
a=rtpmap:9 G722/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:124 ISAC/16000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
m=video 29806 RTP/AVP 126 97
c=IN IP4 10.10.202.137
b=TIAS:1000000
a=rtpmap:126 H264/90000
a=fmtp:126 profile-level-id=42801E;packetization-mode=1;level-asymmetry-allowed=1
a=imageattr:* recv [x=640,y=480,q=0.50]
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42801E;packetization-mode=0;level-asymmetry-allowed=1
a=imageattr:* recv [x=640,y=480,q=0.50]
a=sendrecv

[diagram]   Call-ID:[prev][next]
[8] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.202.137:51547;branch=z9hG4bK15418c29
From: "4210" <sip:4210@10.10.202.97>;tag=b4a4e3289b9434697e410cf2-6a191674
To: <sip:1301130@10.10.202.97>
Date: Mon, 18 Apr 2011 17:31:13 GMT
Call-ID: b4a4e328-9b94001d-4439f603-2fcb5039@10.10.202.137
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[9] INVITE sip:1301130@10.10.202.130:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bKd5769587a
From: <sip:4210@cisco.com>;tag=1590~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379789
To: <sip:1301130@10.10.202.130>
Date: Mon, 18 Apr 2011 17:31:13 GMT
Call-ID: a6e2cc00-dac17561-23-61ca12ac@10.10.202.97
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CP9971/9.2.1
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: <sip:10.10.202.97:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 2799881216-0000065536-0000000006-1640633004
Session-Expires:  1800
P-Asserted-Identity: <sip:4210@cisco.com>
Remote-Party-ID: <sip:4210@cisco.com>;party=calling;screen=yes;privacy=off
Contact: <sip:4210@10.10.202.97:5060;transport=tcp>;video;audio
Max-Forwards: 69
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[10] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bKd5769587a;received=10.10.202.97;ingress-zone=sipsigccm97
Call-ID: a6e2cc00-dac17561-23-61ca12ac@10.10.202.97
CSeq: 101 INVITE
From: <sip:4210@cisco.com>;tag=1590~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379789
To: <sip:1301130@10.10.202.130>
Server: TANDBERG/4100 (X6.1)
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[11] SIP/2.0 180 Ringing
```

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bKd5769587a;received=10.10.202.97;ingress-zone=sipsigccm97
Call-ID: a6e2cc00-dac17561-23-61ca12ac@10.10.202.97
CSeq: 101 INVITE
From: <sip:4210@cisco.com>;tag=1590~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379789
To: <sip:1301130@10.10.202.130>;tag=48318a43bb8c7875
Server: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[12] SIP/2.0 180 Ringing

```
Via: SIP/2.0/TCP 10.10.202.137:51547;branch=z9hG4bK15418c29
From: "4210" <sip:4210@10.10.202.97>;tag=b4a4e3289b9434697e410cf2-6a191674
To: <sip:1301130@10.10.202.97>;tag=1589~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379788
Date: Mon, 18 Apr 2011 17:31:13 GMT
Call-ID: b4a4e328-9b94001d-4439f603-2fcb5039@10.10.202.137
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; ui-state= ringout;
gci= 1-158006; call-instance= 1
Send-Info: conference, x-cisco-conference
Remote-Party-ID: <sip:1301130@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:1301130@10.10.202.97:5060;transport=tcp>
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[13] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bKd5769587a;received=10.10.202.97;ingress-zone=sipsigccm97
Call-ID: a6e2cc00-dac17561-23-61ca12ac@10.10.202.97
CSeq: 101 INVITE
Contact: <sip:1301130@10.10.202.211:5060;transport=tcp>
From: <sip:4210@cisco.com>;tag=1590~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379789
To: <sip:1301130@10.10.202.130>;tag=48318a43bb8c7875
Record-Route: <sip:proxy-call-id=a89f9590-69e1-11e0-8ec9-0010f31d16ee@10.10.202.130:5060;transport=tcp;lr>
Record-Route: <sip:proxy-call-id=a89f9590-69e1-11e0-8ec9-0010f31d16ee@10.10.202.130:5060;transport=tcp;lr>
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY
Server: TANDBERG/257 (TE4.1.0.-PreAlpha (TEST SW))
Supported: replaces,100rel,timer,gruu,path,outbound
Require: timer
Session-Expires: 1800;refresher=uas
Min-SE: 1800
Content-Type: application/sdp
Content-Length: 2109

v=0
o=tandberg 28 1 IN IP4 10.10.202.211
s=-
c=IN IP4 10.10.202.211
b=AS:1152
t=0 0
m=audio 2334 RTP/AVP 100 101 102 9 18 11 8 0 103
b=TIAS:64000
a=rtpmap:100 MP4A-LATM/90000
a=fmtp:100 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:101 G7221/16000
a=fmtp:101 bitrate=32000
a=rtpmap:102 G7221/16000
a=fmtp:102 bitrate=24000
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:11 L16/16000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
a=sendrecv
m=video 2336 RTP/AVP 97 98 99 34 31
b=TIAS:1152000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;max-fps=3000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;packetization-
mode=1;max-fps=3000
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1024,768,4;custom=1024,576,4;custom=800,600,4;cif4=2;custom=720,480,2;custom=640,480,2;custom=512,288
,1;cif=1;cu
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=2;cif=1;qcif=1;maxbr=10880
```

```
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=10880
a=rtcp-fb:* nack pli
a=sendrecv
a=content:main
a=label:11
a=answer:full
m=application 5070 UDP/BFCP *
a=floorctrl:c-s
a=confid:1
a=floorid:2 mstrm:12
a=userid:28
a=setup:actpass
a=connection:new
m=video 2338 RTP/AVP 97 98 99 34 31
b=TIAS:1152000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;max-fps=3000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800d;max-br=906;max-mbps=40500;max-fs=1344;max-smbps=40500;packetization-
mode=1;max-fps=3000
a=rtpmap:99 H263-1998/90000
a=fmtp:99
custom=1024,768,4;custom=1024,576,4;custom=800,600,4;cif4=2;custom=720,480,2;custom=640,480,2;custom=512,288
,1;cif=1;cu
a=rtpmap:34 H263/90000
a=fmtp:34 cif4=2;cif=1;qcif=1;maxbr=10880
a=rtpmap:31 H261/90000
a=fmtp:31 cif=1;qcif=1;maxbr=10880
a=rtcp-fb:* nack pli
a=sendrecv
a=content:slides
a=label:12
m=application 2340 RTP/AVP 104
a=rtpmap:104 H224/4800
a=sendrecv
```

[diagram] Call-ID:[prev][next]
[14] ACK sip:1301130@10.10.202.211:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bKd633b2b883
From: <sip:4210@cisco.com>;tag=1590~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379789
To: <sip:1301130@10.10.202.130>;tag=48318a43bb8c7875
Date: Mon, 18 Apr 2011 17:31:13 GMT
Call-ID: a6e2cc00-dac17561-23-61ca12ac@10.10.202.97
Route: <sip:proxy-call-id=a89f9590-69e1-11e0-8ec9-
0010f31d16ee@10.10.202.130:5060;transport=tcp;lr>,<sip:proxy-call-id=a89f9590-69e1-11e0-8ec9-
0010f31d16ee@10.10.202.130:5060;transport=tcp;lr>
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Type: application/sdp
Content-Length: 1227

v=0
o=CiscoSystemsCCM-SIP 1590 1 IN IP4 10.10.202.97
s=SIP Call
b=AS:1064
t=0 0
m=audio 26640 RTP/AVP 9 101
c=IN IP4 10.10.202.137
b=TIAS:64000
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 29806 RTP/AVP 126
c=IN IP4 10.10.202.137
b=TIAS:1000000
a=rtpmap:126 H264/90000
a=fmtp:126 profile-level-id=42801E;packetization-mode=1;level-asymmetry-allowed=1
m=application 0 UDP/BFCP *
c=IN IP4 10.10.202.211
a=floorctrl:c-s
a=floorid:2 mstrm:12
a=confid:1
a=userid:28
m=video 0 RTP/AVP 97 98 99 34 31
c=IN IP4 10.10.202.211
b=TIAS:1152000
a=label:12
```

```
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800D;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-smbps=40500;max-fps=3000
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-smbps=40500;max-fps=3000
a=rtpmap:99 H263-1998/90000
a=fmtp:99 CIF=1;CIF4=2;CUSTOM=512,288,1
a=rtpmap:34 H263/90000
a=fmtp:34 QCIF=1;CIF=1;CIF4=2;MAXBR=10880
a=rtpmap:31 H261/90000
a=fmtp:31 CIF=1;QCIF=1;MAXBR=10880
a=content:slides
a=rtcp-fb:* nack pli
m=application 0 RTP/AVP 104
c=IN IP4 10.10.202.211
a=rtpmap:104 H224/0
```

[diagram] Call-ID:[prev][next]

[15] SUBSCRIBE sip:9bbb8dda-00ee-701f-1b73-470a40f63088@10.10.202.137:51547;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bKd76c31933f
From: <sip:1301130@10.10.202.97>;tag=1589~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379788
To: "4210" <sip:4210@10.10.202.97>;tag=b4a4e3289b9434697e410cf2-6a191674
Call-ID: b4a4e328-9b94001d-4439f603-2fcb5039@10.10.202.137
CSeq: 101 SUBSCRIBE
Date: Mon, 18 Apr 2011 17:31:17 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 7200
Contact: <sip:10.10.202.97:5060;transport=tcp>
Accept: application/kpml-response+xml
Max-Forwards: 70
Content-Type: application/kpml-request+xml
Content-Length: 370

<?xml version="1.0" encoding="UTF-8" ?>
<kpml-request xmlns="urn:ietf:params:xml:ns:kpml-request" xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-request kpml-request.xsd" version="1.0">

  <pattern interdigittimer="7260000" persist="persist">
    <regex tag="dtmf">[x*#ABCD]</regex>
  </pattern>

</kpml-request>
```

[diagram] Call-ID:[prev][next]

[16] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.10.202.137:51547;branch=z9hG4bK15418c29
From: "4210" <sip:4210@10.10.202.97>;tag=b4a4e3289b9434697e410cf2-6a191674
To: <sip:1301130@10.10.202.97>;tag=1589~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379788
Date: Mon, 18 Apr 2011 17:31:13 GMT
Call-ID: b4a4e328-9b94001d-4439f603-2fcb5039@10.10.202.137
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; gci= 1-158006;
call-instance= 1
Send-Info: conference, x-cisco-conference
Remote-Party-ID: <sip:1301130@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:1301130@10.10.202.97:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 477

v=0
o=CiscoSystemsCCM-SIP 1589 1 IN IP4 10.10.202.97
s=SIP Call
c=IN IP4 10.10.202.211
t=0 0
m=audio 2334 RTP/AVP 9 103
b=TIAS:64000
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
m=video 2336 RTP/AVP 98
b=TIAS:1088000
a=label:11
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-
```

```
smbps=40500;max-fps=3000
a=content:main
a=rtcp-fb:* nack pli

[diagram]   Call-ID:[prev][next]
[17] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.202.137:51547;branch=z9hG4bK15418c29
From: "4210" <sip:4210@10.10.202.97>;tag=b4a4e3289b9434697e410cf2-6a191674
To: <sip:1301130@10.10.202.97>;tag=1589~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379788
Date: Mon, 18 Apr 2011 17:31:13 GMT
Call-ID: b4a4e328-9b94001d-4439f603-2fcb5039@10.10.202.137
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; gci= 1-158006;
call-instance= 1
Send-Info: conference, x-cisco-conference
Remote-Party-ID: <sip:1301130@10.10.202.97>;party=called;screen=yes;privacy=off
Contact: <sip:1301130@10.10.202.97:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 477

v=0
o=CiscoSystemsCCM-SIP 1589 1 IN IP4 10.10.202.97
s=SIP Call
c=IN IP4 10.10.202.211
t=0 0
m=audio 2334 RTP/AVP 9 103
b=TIAS:64000
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:103 telephone-event/8000
a=fmtp:103 0-15
m=video 2336 RTP/AVP 98
b=TIAS:1088000
a=label:11
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42800D;packetization-mode=1;max-mbps=40500;max-fs=1344;max-cpb=37;max-br=925;max-
smbps=40500;max-fps=3000
a=content:main
a=rtcp-fb:* nack pli

[diagram]   Call-ID:[prev][next]
[18] ACK sip:1301130@10.10.202.97:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.202.137:51547;branch=z9hG4bK7fdf0442
From: "4210" <sip:4210@10.10.202.97>;tag=b4a4e3289b9434697e410cf2-6a191674
To: <sip:1301130@10.10.202.97>;tag=1589~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379788
Call-ID: b4a4e328-9b94001d-4439f603-2fcb5039@10.10.202.137
Max-Forwards: 70
Date: Mon, 18 Apr 2011 17:31:17 GMT
CSeq: 101 ACK
User-Agent: Cisco-CP9971/9.2.1
Remote-Party-ID: "4210" <sip:4210@10.10.202.97>;party=calling;id-type=subscriber;privacy=off;screen=yes
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[19] NOTIFY sip:10.10.202.97:5060 SIP/2.0
Via: SIP/2.0/TCP 10.10.202.137:51547;branch=z9hG4bK2765b592
To: <sip:1301130@10.10.202.97>;tag=1589~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379788
From: "4210" <sip:4210@10.10.202.97>;tag=b4a4e3289b9434697e410cf2-6a191674
Call-ID: b4a4e328-9b94001d-4439f603-2fcb5039@10.10.202.137
Date: Mon, 18 Apr 2011 17:31:17 GMT
CSeq: 102 NOTIFY
Event: kpml
Subscription-State: active; expires=7200
Max-Forwards: 70
Contact: <sip:9bbb8dda-00ee-701f-1b73-470a40f63088@10.10.202.137:51547;transport=TCP>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[20] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.202.97:5060;branch=z9hG4bKd76c31933f
From: <sip:1301130@10.10.202.97>;tag=1589~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379788
To: "4210" <sip:4210@10.10.202.97>;tag=b4a4e3289b9434697e410cf2-6a191674
Call-ID: b4a4e328-9b94001d-4439f603-2fcb5039@10.10.202.137
Date: Mon, 18 Apr 2011 17:31:17 GMT
CSeq: 101 SUBSCRIBE
Server: Cisco-CP9971/9.2.1
```

```
Contact: <sip:9bbb8dda-00ee-701f-1b73-470a40f63088@10.10.202.137:51547;transport=TCP>
Expires: 7200
Content-Length: 0
```

[diagram] Call-ID:**[prev]**[next]

```
[21] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.202.137:51547;branch=z9hG4bK2765b592
From: "4210" <sip:4210@10.10.202.97>;tag=b4a4e3289b9434697e410cf2-6a191674
To: <sip:1301130@10.10.202.97>;tag=1589~b5a88942-7acc-4cc9-9d65-67021cfecfed-28379788
Date: Mon, 18 Apr 2011 17:31:18 GMT
Call-ID: b4a4e328-9b94001d-4439f603-2fcb5039@10.10.202.137
CSeq: 102 NOTIFY
Content-Length: 0
```

8.4 Basic Call between two Video enabled SIP device via an Inter Cluster SIP Trunk

Title: Basic Call between two Video enabled SIP device via an Inter Cluster SIP Trunk

Description:

The following call flow illustrates the SIP messaging that takes place between two Cisco Unified CMs via an inter cluster SIP trunk. The call flow illustrates that the max-fps attribute is processed by CUCM for a Video call. This attribute is used by the VCS for video calls.

Cisco Unified CM1 sent out the initial INVITE.

Configuration:

Node = Unified CM1, IP = 10.77.31.176
Node = Unified CM2, IP = 10.77.31.191

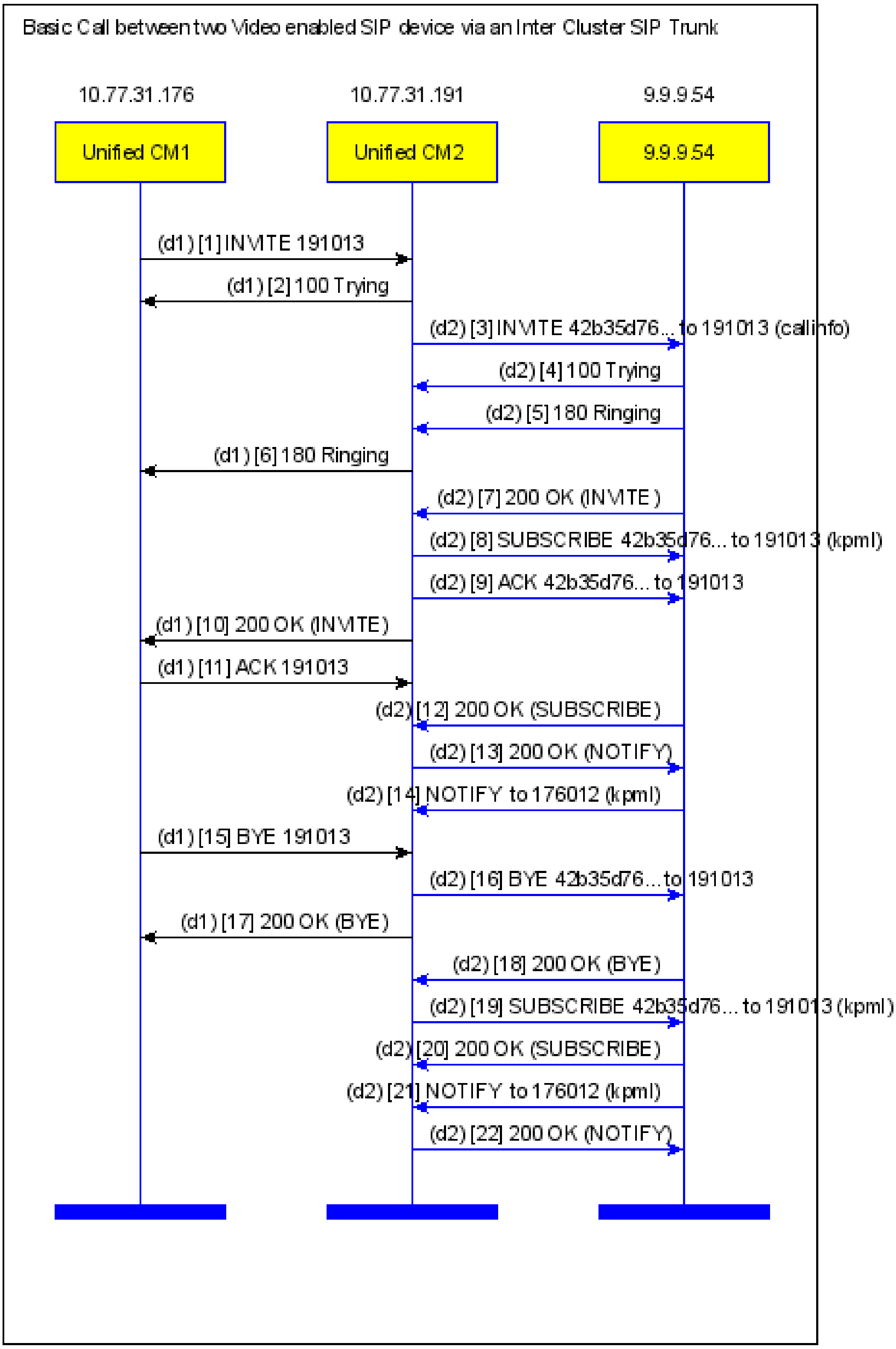
Phone = A, Line = 176012, IP = 9.9.9.15, Model = SIP, Video Enabled
Phone = B, Line = 191013, IP = 9.9.9.54, Model = SIP, Video Enabled

SIP Trunk between Unified CM1 & Unified CM2

Scenario:

Phone A DN 176012 calls Phone B DN 191013 over the SIP Trunk
Phone B DN 191013 Answers
Phone A DN 176012 goes onHook

End of Scenario


Basic Call between two Video enabled SIP device via an Inter Cluster SIP Trunk
10.77.31.176
10.77.31.191
9.9.9.54
Unified CM1
Unified CM2
9.9.9.54
(d1) [1] INVITE 191013
(d1) [2] 100 Trying
(d2) [3] INVITE 42b35d76... to 191013 (callinfo)
(d2) [4] 100 Trying
(d2) [5] 180 Ringing
(d1) [6] 180 Ringing
(d2) [7] 200 OK (INVITE)
(d2) [8] SUBSCRIBE 42b35d76... to 191013 (kpml)
(d2) [9] ACK 42b35d76... to 191013
(d1) [10] 200 OK (INVITE)
(d1) [11] ACK 191013
(d2) [12] 200 OK (SUBSCRIBE)
(d2) [13] 200 OK (NOTIFY)
(d2) [14] NOTIFY to 176012 (kpml)
(d1) [15] BYE 191013
(d2) [16] BYE 42b35d76... to 191013
(d1) [17] 200 OK (BYE)
(d2) [18] 200 OK (BYE)
(d2) [19] SUBSCRIBE 42b35d76... to 191013 (kpml)
(d2) [20] 200 OK (SUBSCRIBE)
(d2) [21] NOTIFY to 176012 (kpml)
(d2) [22] 200 OK (NOTIFY)

```
[diagram]   Call-ID:[prev][next]
[1] INVITE sip:191013@10.77.31.191:5060 SIP/2.0
Via: SIP/2.0/TCP 10.77.31.176:5060;branch=z9hG4bK3f5bd4927d
From: <sip:176012@10.77.31.176>;tag=207~e0c63701-a458-47df-9054-62b959feca5a-30519138
To: <sip:191013@10.77.31.191>
Date: Mon, 07 Mar 2011 06:40:32 GMT
Call-ID: cb497580-d7417de0-18-b01f4d0a@10.77.31.176
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM8.6
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: <sip:10.77.31.176:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 3410589056-0000065536-0000000011-2954841354
Session-Expires:  1800
P-Asserted-Identity: <sip:176012@10.77.31.176>
Remote-Party-ID: <sip:176012@10.77.31.176>;party=calling;screen=yes;privacy=off
Contact: <sip:176012@10.77.31.176:5060;transport=tcp>;video;audio
Max-Forwards: 69
Content-Length: 780
Content-Type: application/sdp

v=0
o=CiscoSystemsCCM-SIP 207 1 IN IP4 10.77.31.176
s=SIP Call
c=IN IP4 9.9.9.15
b=TIAS:384000
b=AS:384
t=0 0
m=audio 20116 RTP/AVP 9 124 0 8 116 18 101
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:124 iSAC/16000
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:8 PCMA/8000
a=ptime:20
a=rtpmap:116 iLBC/8000
a=ptime:20
a=maxptime:20
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=ptime:20
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 20354 RTP/AVP 126 97
b=TIAS:320000
a=rtpmap:126 H264/90000
a=fmtp:126 max-fps=5994;profile-level-id=42801E;packetization-mode=1;level-asymmetry-allowed=1
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42801E;packetization-mode=0;level-asymmetry-allowed=1
a=imageattr:* recv [x=640,y=480,q=0.50]

[diagram]   Call-ID:[prev][next]
[2] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.77.31.176:5060;branch=z9hG4bK3f5bd4927d
From: <sip:176012@10.77.31.176>;tag=207~e0c63701-a458-47df-9054-62b959feca5a-30519138
To: <sip:191013@10.77.31.191>
Date: Mon, 07 Mar 2011 06:40:31 GMT
Call-ID: cb497580-d7417de0-18-b01f4d0a@10.77.31.176
CSeq: 101 INVITE
Allow-Events: presence
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[3] INVITE sip:42b35d76-0edb-4671-ba24-1b7e32c2e0db@9.9.9.54:53026;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.77.31.191:5060;branch=z9hG4bK44569fe292
From: <sip:176012@10.77.31.191>;tag=246~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511574
To: <sip:191013@10.77.31.191>
Date: Mon, 07 Mar 2011 06:40:31 GMT
Call-ID: cab0df00-d7417ddf-1c-bf1f4d0a@10.77.31.191
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM8.6
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
```

```
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= from; gci= 1-119011;
call-instance= 1
Send-Info: conference, x-cisco-conference
Alert-Info: <file://Bellcore-dr2/>
Remote-Party-ID: <sip:176012@10.77.31.191;x-cisco-callback-
number=176012>;party=calling;screen=yes;privacy=off
Contact: <sip:176012@10.77.31.191:5060;transport=tcp>;video;audio
Max-Forwards: 68
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[4] SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.77.31.191:5060;branch=z9hG4bK44569fe292
From: <sip:176012@10.77.31.191>;tag=246~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511574
To: <sip:191013@10.77.31.191>
Call-ID: cab0df00-d7417ddf-1c-bf1f4d0a@10.77.31.191
Date: Mon, 07 Mar 2011 06:40:30 GMT
CSeq: 101 INVITE
Server: Cisco-CP9971/9.2.1
Contact: <sip:42b35d76-0edb-4671-ba24-1b7e32c2e0db@9.9.9.54:53026;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Supported: replaces,join,sdp-anat,norefersub,extended-refer,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-
escapecodes,X-cisco-service-control,X-cisco-srtp-fallback,X-cisco-monrec,X-cisco-config,X-cisco-sis-5.1.0,X-
cisco-xsi-8.0.1
Allow-Events: kpml,dialog
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[5] SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.77.31.191:5060;branch=z9hG4bK44569fe292
From: <sip:176012@10.77.31.191>;tag=246~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511574
To: <sip:191013@10.77.31.191>;tag=1c17d3418f8a003657bc2642-35d633f4
Call-ID: cab0df00-d7417ddf-1c-bf1f4d0a@10.77.31.191
Date: Mon, 07 Mar 2011 06:40:30 GMT
CSeq: 101 INVITE
Server: Cisco-CP9971/9.2.1
Contact: <sip:42b35d76-0edb-4671-ba24-1b7e32c2e0db@9.9.9.54:53026;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "191013" <sip:191013@10.77.31.191>;party=called;id-type=subscriber;privacy=off;screen=yes
Supported: replaces,join,sdp-anat,norefersub,extended-refer,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-
escapecodes,X-cisco-service-control,X-cisco-srtp-fallback,X-cisco-monrec,X-cisco-config,X-cisco-sis-5.1.0,X-
cisco-xsi-8.0.1
Allow-Events: kpml,dialog
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[6] SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.77.31.176:5060;branch=z9hG4bK3f5bd4927d
From: <sip:176012@10.77.31.176>;tag=207~e0c63701-a458-47df-9054-62b959feca5a-30519138
To: <sip:191013@10.77.31.191>;tag=245~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511573
Date: Mon, 07 Mar 2011 06:40:31 GMT
Call-ID: cb497580-d7417de0-18-b01f4d0a@10.77.31.176
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:191013@10.77.31.191>
Remote-Party-ID: <sip:191013@10.77.31.191>;party=called;screen=yes;privacy=off
Contact: <sip:191013@10.77.31.191:5060;transport=tcp>
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[7] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.77.31.191:5060;branch=z9hG4bK44569fe292
From: <sip:176012@10.77.31.191>;tag=246~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511574
To: <sip:191013@10.77.31.191>;tag=1c17d3418f8a003657bc2642-35d633f4
Call-ID: cab0df00-d7417ddf-1c-bf1f4d0a@10.77.31.191
Date: Mon, 07 Mar 2011 06:40:33 GMT
CSeq: 101 INVITE
Server: Cisco-CP9971/9.2.1
Contact: <sip:42b35d76-0edb-4671-ba24-1b7e32c2e0db@9.9.9.54:53026;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "191013" <sip:191013@10.77.31.191>;party=called;id-type=subscriber;privacy=off;screen=yes
Supported: replaces,join,sdp-anat,norefersub,extended-refer,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-
escapecodes,X-cisco-service-control,X-cisco-srtp-fallback,X-cisco-monrec,X-cisco-config,X-cisco-sis-5.1.0,X-
cisco-xsi-8.0.1
```

```
Allow-Events: kpml,dialog
Content-Length: 672
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 19686 0 IN IP4 9.9.9.54
s=SIP Call
t=0 0
m=audio 21712 RTP/AVP 0 8 18 102 9 116 124 101
c=IN IP4 9.9.9.54
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:102 L16/16000
a=rtpmap:9 G722/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:124 ISAC/16000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
m=video 29180 RTP/AVP 126 97
c=IN IP4 9.9.9.54
a=rtpmap:126 H264/90000
a=fmtp:126 profile-level-id=42800A;packetization-mode=1;level-asymmetry-allowed=1
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42800A;packetization-mode=0;level-asymmetry-allowed=1
a=sendrecv
```

[diagram]   Call-ID:[prev][next]
[8] SUBSCRIBE sip:42b35d76-0edb-4671-ba24-1b7e32c2e0db@9.9.9.54:53026;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.77.31.191:5060;branch=z9hG4bK464c337e5b
From: <sip:176012@10.77.31.191>;tag=246~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511574
To: <sip:191013@10.77.31.191>;tag=1c17d3418f8a003657bc2642-35d633f4
Call-ID: cab0df00-d7417ddf-1c-bf1f4d0a@10.77.31.191
CSeq: 102 SUBSCRIBE
Date: Mon, 07 Mar 2011 06:40:34 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 7200
Contact: <sip:10.77.31.191:5060;transport=tcp>
Accept: application/kpml-response+xml
Max-Forwards: 70
Content-Type: application/kpml-request+xml
Content-Length: 370

<?xml version="1.0" encoding="UTF-8" ?>
<kpml-request xmlns="urn:ietf:params:xml:ns:kpml-request" xmlns:xsi="http://www.w3.org/2001/XMLSchema-
instance" xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-request kpml-request.xsd" version="1.0">

  <pattern interdigittimer="7260000" persist="persist">
    <regex tag="dtmf">[x*#ABCD]</regex>
  </pattern>

</kpml-request>
```

[diagram]   Call-ID:[prev][next]
[9] ACK sip:42b35d76-0edb-4671-ba24-1b7e32c2e0db@9.9.9.54:53026;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.77.31.191:5060;branch=z9hG4bK455dfe2d55
From: <sip:176012@10.77.31.191>;tag=246~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511574
To: <sip:191013@10.77.31.191>;tag=1c17d3418f8a003657bc2642-35d633f4
Date: Mon, 07 Mar 2011 06:40:31 GMT
Call-ID: cab0df00-d7417ddf-1c-bf1f4d0a@10.77.31.191
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 370

v=0
o=CiscoSystemsCCM-SIP 246 1 IN IP4 10.77.31.191
s=SIP Call
c=IN IP4 9.9.9.15
t=0 0
m=audio 20116 RTP/AVP 9 101
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
```

```
a=fmtp:101 0-15
m=video 20354 RTP/AVP 126
b=TIAS:320000
a=rtpmap:126 H264/90000
a=fmtp:126 profile-level-id=42801E;packetization-mode=1;level-asymmetry-allowed=1;max-fps=5994
```

[diagram] Call-ID:[prev][next]
[10] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.77.31.176:5060;branch=z9hG4bK3f5bd4927d
From: <sip:176012@10.77.31.176>;tag=207~e0c63701-a458-47df-9054-62b959feca5a-30519138
To: <sip:191013@10.77.31.191>;tag=245~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511573
Date: Mon, 07 Mar 2011 06:40:31 GMT
Call-ID: cb497580-d7417de0-18-b01f4d0a@10.77.31.176
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence, kpml
Supported: replaces
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Session-Expires:  1800;refresher=uas
Require:  timer
P-Asserted-Identity: <sip:191013@10.77.31.191>
Remote-Party-ID: <sip:191013@10.77.31.191>;party=called;screen=yes;privacy=off
Contact: <sip:191013@10.77.31.191:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 382

v=0
o=CiscoSystemsCCM-SIP 245 1 IN IP4 10.77.31.191
s=SIP Call
c=IN IP4 9.9.9.54
b=TIAS:384000
b=AS:384
t=0 0
m=audio 21712 RTP/AVP 9 101
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 29180 RTP/AVP 126
b=TIAS:320000
a=rtpmap:126 H264/90000
a=fmtp:126 profile-level-id=42800A;packetization-mode=1;level-asymmetry-allowed=1
```

[diagram] Call-ID:[prev][next]
[11] ACK sip:191013@10.77.31.191:5060;transport=tcp SIP/2.0

```
Via: SIP/2.0/TCP 10.77.31.176:5060;branch=z9hG4bK40459d5a73
From: <sip:176012@10.77.31.176>;tag=207~e0c63701-a458-47df-9054-62b959feca5a-30519138
To: <sip:191013@10.77.31.191>;tag=245~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511573
Date: Mon, 07 Mar 2011 06:40:32 GMT
Call-ID: cb497580-d7417de0-18-b01f4d0a@10.77.31.176
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[12] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 10.77.31.191:5060;branch=z9hG4bK464c337e5b
From: <sip:176012@10.77.31.191>;tag=246~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511574
To: <sip:191013@10.77.31.191>;tag=1c17d3418f8a003657bc2642-35d633f4
Call-ID: cab0df00-d7417ddf-1c-bf1f4d0a@10.77.31.191
Date: Mon, 07 Mar 2011 06:40:34 GMT
CSeq: 102 SUBSCRIBE
Server: Cisco-CP9971/9.2.1
Contact: <sip:42b35d76-0edb-4671-ba24-1b7e32c2e0db@9.9.9.54:53026;transport=TCP>
Expires: 7200
Content-Length: 0
```

[diagram] Call-ID:[prev][next]
[13] SIP/2.0 200 OK

```
Via: SIP/2.0/TCP 9.9.9.54:53026;branch=z9hG4bK48ddef70
From: <sip:191013@10.77.31.191>;tag=1c17d3418f8a003657bc2642-35d633f4
To: <sip:176012@10.77.31.191>;tag=246~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511574
Date: Mon, 07 Mar 2011 06:40:34 GMT
Call-ID: cab0df00-d7417ddf-1c-bf1f4d0a@10.77.31.191
CSeq: 101 NOTIFY
Content-Length: 0
```

```
[diagram]   Call-ID:[prev][next]
[14] NOTIFY sip:10.77.31.191:5060 SIP/2.0
Via: SIP/2.0/TCP 9.9.9.54:53026;branch=z9hG4bK48ddef70
To: <sip:176012@10.77.31.191>;tag=246~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511574
From: <sip:191013@10.77.31.191>;tag=1c17d3418f8a003657bc2642-35d633f4
Call-ID: cab0df00-d7417ddf-1c-bf1f4d0a@10.77.31.191
Date: Mon, 07 Mar 2011 06:40:34 GMT
CSeq: 101 NOTIFY
Event: kpml
Subscription-State: active; expires=7200
Max-Forwards: 70
Contact: <sip:42b35d76-0edb-4671-ba24-1b7e32c2e0db@9.9.9.54:53026;transport=TCP>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[15] BYE sip:191013@10.77.31.191:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.77.31.176:5060;branch=z9hG4bK424967f683
From: <sip:176012@10.77.31.176>;tag=207~e0c63701-a458-47df-9054-62b959feca5a-30519138
To: <sip:191013@10.77.31.191>;tag=245~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511573
Date: Mon, 07 Mar 2011 06:40:32 GMT
Call-ID: cb497580-d7417de0-18-b01f4d0a@10.77.31.176
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
P-Asserted-Identity: <sip:176012@10.77.31.176>
CSeq: 102 BYE
Reason: Q.850;cause=16
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[16] BYE sip:42b35d76-0edb-4671-ba24-1b7e32c2e0db@9.9.9.54:53026;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.77.31.191:5060;branch=z9hG4bK47771993d
From: <sip:176012@10.77.31.191>;tag=246~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511574
To: <sip:191013@10.77.31.191>;tag=1c17d3418f8a003657bc2642-35d633f4
Date: Mon, 07 Mar 2011 06:40:31 GMT
Call-ID: cab0df00-d7417ddf-1c-bf1f4d0a@10.77.31.191
User-Agent: Cisco-CUCM8.6
Max-Forwards: 70
CSeq: 103 BYE
Reason: Q.850;cause=16
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[17] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.77.31.176:5060;branch=z9hG4bK424967f683
From: <sip:176012@10.77.31.176>;tag=207~e0c63701-a458-47df-9054-62b959feca5a-30519138
To: <sip:191013@10.77.31.191>;tag=245~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511573
Date: Mon, 07 Mar 2011 06:40:43 GMT
Call-ID: cb497580-d7417de0-18-b01f4d0a@10.77.31.176
CSeq: 102 BYE
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[18] SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.77.31.191:5060;branch=z9hG4bK47771993d
From: <sip:176012@10.77.31.191>;tag=246~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511574
To: <sip:191013@10.77.31.191>;tag=1c17d3418f8a003657bc2642-35d633f4
Call-ID: cab0df00-d7417ddf-1c-bf1f4d0a@10.77.31.191
Date: Mon, 07 Mar 2011 06:40:43 GMT
CSeq: 103 BYE
Server: Cisco-CP9971/9.2.1
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[19] SUBSCRIBE sip:42b35d76-0edb-4671-ba24-1b7e32c2e0db@9.9.9.54:53026;transport=TCP SIP/2.0
Via: SIP/2.0/TCP 10.77.31.191:5060;branch=z9hG4bK4822d21ca2
From: <sip:176012@10.77.31.191>;tag=246~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511574
To: <sip:191013@10.77.31.191>;tag=1c17d3418f8a003657bc2642-35d633f4
Call-ID: cab0df00-d7417ddf-1c-bf1f4d0a@10.77.31.191
CSeq: 104 SUBSCRIBE
Date: Mon, 07 Mar 2011 06:40:43 GMT
User-Agent: Cisco-CUCM8.6
Event: kpml
Expires: 0
Contact: <sip:10.77.31.191:5060;transport=tcp>
Max-Forwards: 70
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[20] SIP/2.0 200 OK
```

```
Via: SIP/2.0/TCP 10.77.31.191:5060;branch=z9hG4bK4822d21ca2
From: <sip:176012@10.77.31.191>;tag=246~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511574
To: <sip:191013@10.77.31.191>;tag=1c17d3418f8a003657bc2642-35d633f4
Call-ID: cab0df00-d7417ddf-1c-bf1f4d0a@10.77.31.191
Date: Mon, 07 Mar 2011 06:40:43 GMT
CSeq: 104 SUBSCRIBE
Server: Cisco-CP9971/9.2.1
Contact: <sip:42b35d76-0edb-4671-ba24-1b7e32c2e0db@9.9.9.54:53026;transport=TCP>
Expires: 0
Content-Length: 0

[diagram]   Call-ID:[prev][next]
[21] NOTIFY sip:10.77.31.191:5060 SIP/2.0
Via: SIP/2.0/TCP 9.9.9.54:53026;branch=z9hG4bK51a68fb0
To: <sip:176012@10.77.31.191>;tag=246~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511574
From: <sip:191013@10.77.31.191>;tag=1c17d3418f8a003657bc2642-35d633f4
Call-ID: cab0df00-d7417ddf-1c-bf1f4d0a@10.77.31.191
Date: Mon, 07 Mar 2011 06:40:43 GMT
CSeq: 102 NOTIFY
Event: kpml
Subscription-State: terminated; reason=timeout
Max-Forwards: 70
Contact: <sip:42b35d76-0edb-4671-ba24-1b7e32c2e0db@9.9.9.54:53026;transport=TCP>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE
Content-Length: 214
Content-Type: application/kpml-response+xml
Content-Disposition: session;handling=required

<?xml version="1.0" encoding="UTF-8"?>
<kpml-response xmlns="urn:ietf:params:xml:ns:kpml-response" version="1.0" code="487" text="Subscription Exp"
suppressed="false" forced_flush="false" digits="" tag="dtmf"/>

[diagram]   Call-ID:[prev][next]
[22] SIP/2.0 200 OK
Via: SIP/2.0/TCP 9.9.9.54:53026;branch=z9hG4bK51a68fb0
From: <sip:191013@10.77.31.191>;tag=1c17d3418f8a003657bc2642-35d633f4
To: <sip:176012@10.77.31.191>;tag=246~e1c5c5ca-9bc9-4669-8c3d-a693914b4397-30511574
Date: Mon, 07 Mar 2011 06:40:43 GMT
Call-ID: cab0df00-d7417ddf-1c-bf1f4d0a@10.77.31.191
CSeq: 102 NOTIFY
Content-Length: 0
```