
TOMORROW
starts here.

Troubleshooting SIP

BRKUCC-3662

Richard Middelberg

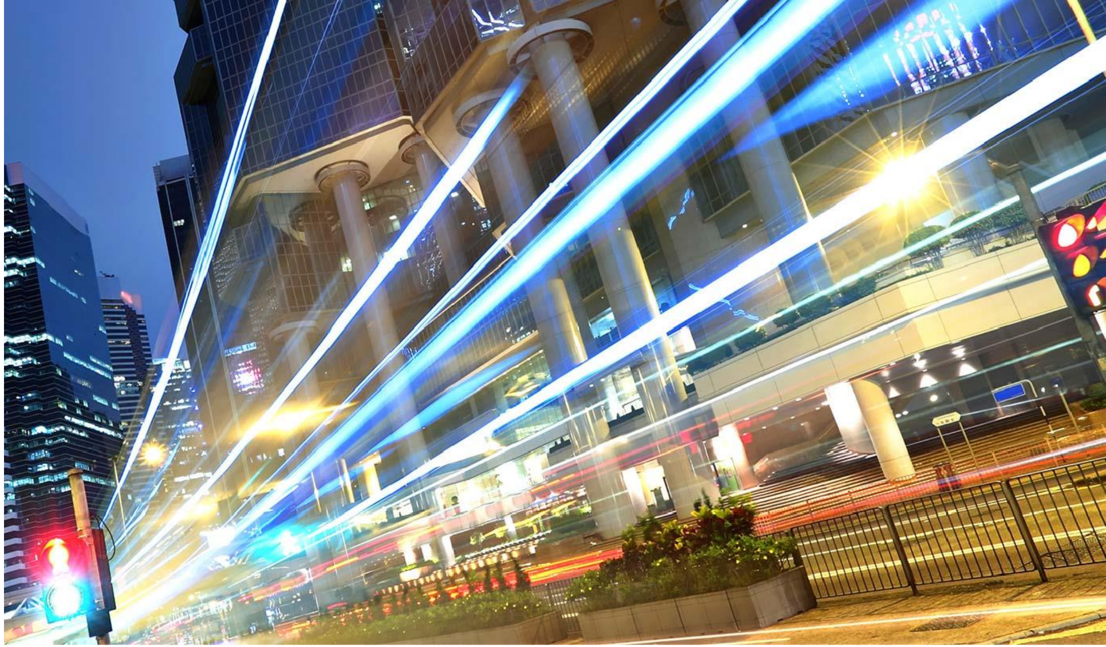
High Touch Engineer

#clmel

Cisco live!

Agenda

- SIP Overview
- Case Studies



SIP Overview

What is SIP?

- Signalling protocol used mainly for Unified Communication session negotiation
- Defined in a number of RFC's, the main one being RFC 3261
- ASCII-based messages
- Application layer protocol – can run on TCP or UDP.
- Endpoints are referred to as User Agents
- Works in conjunction with other protocols for call setup and media transmission – e.g. SDP and RTP

What is SIP?

- Network Elements
  - User Agents
  - Registrar
  - Session Border Controller
  - Gateway
- SIP Messages
  - Requests and Responses
  - Headers

What is SIP?

- Media Negotiation
  - Session Description Protocol
  - Offer/Answer Model
  - Early Offer vs. Delayed Offer
  - Early Media
  - Re-Invite

User Agents (UA)

- Manages SIP sessions
  - Act as a UAS (User Agent Server) – sends SIP requests
  - As well as UAC (User Agent Client) – receives SIP requests
- SIP phone can be hardware or software – anything that can dial, reject, answer etc

SIP Request – RFC 3261

- INVITE: Invite another user agent to join a session
- ACK: Confirms reliable message exchanges.
- CANCEL: Terminates a pending request.
- BYE: Terminates an existing session.
- OPTIONS: Requests information about the capabilities of a caller without the need to set up a session. Often used as keep-alive messages.
- REGISTER: Used by a UA to register to the registrar.

Additional SIP Request Methods

- PRACK (RFC 3262) – Acknowledge a provisional response
- SUBSCRIBE (RFC 3265) – Tell a remote node to look for a certain event
- NOTIFY (RFC 6665) – Respond when a certain event occurs
- PUBLISH (RFC 3903) – Publishes an event to the Server
- INFO (RFC 6086) – Sends more info inside a session
- REFER (RFC 3515) – Receiver will send a SIP request to another UA (transfer)
- MESSAGE (RFC 3428) – Transports instant messages using SIP
- UPDATE (RFC 3311) – Changes parameters of a session set-up

SIP INVITE

INVITE sip:0323456789@stark.winterfell.com SIP/2.0

Via: SIP/2.0/UDP 10.80.90.20:5060;branch=z9hG4bK9cf4db6836d1861bb6841b447b8b4d04.1;rport

Call-ID: 34f5abc5bf476312342baed216f1e2a5

CSeq: 100 INVITE

Contact: <sip:p0234567890@10.80.100.220:5060>

From: <sip:p0234567890@lannister.casterlyrock.com>;tag=2a202c8336b0dd1f

To: <sip:0323456789@stark.winterfell.com>

Max-Forwards: 70

Route: <sip:192.168.0.2;lr>

Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY

User-Agent: TANDBERG/520 (TC7.1.0.48db3d2)

Supported: replaces,100rel,timer,gruu,path,outbound,sdp-anat

Session-Expires: 1800

Content-Type: application/sdp

Content-Length: 2660

SIP Request Line

INVITE sip:0323456789@stark.winterfell.com SIP/2.0

Via: SIP/2.0/UDP 10.80.90.20:5060;branch=z9hG4bK9cf4db6836d1861bb6841b447b8b4d04.1;rport

Call-ID: 34f5abc5bf476312342baed216f1e2a5

CSeq: 100 INVITE

Contact: <sip:p0234567890@10.80.100.220:5060>

From: <sip:p0234567890@lannister.casterlyrock.com>;tag=2a202c8336b0dd1f

To: <sip:0323456789@stark.winterfell.com>

Max-Forwards: 70

Route: <sip:192.168.0.2;lr>

Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY

User-Agent: TANDBERG/520 (TC7.1.0.48db3d2)

Supported: replaces,100rel,timer,gruu,path,outbound,sdp-anat

Session-Expires: 1800

Content-Type: application/sdp

Content-Length: 2660

URI

SIP Version

SIP Request

SIP Headers

INVITE sip:0323456789@stark.winterfell.com SIP/2.0

Via: SIP/2.0/UDP 10.80.90.20:5060;branch=z9hG4bK9cf4db6836d1861bb6841b447b8b4d04.1;rport

Call-ID: 34f5abc5bf476312342baed216f1e2a5

CSeq: 100 INVITE

Contact: <sip:0234567890@10.80.100.220:5060>

From: <sip:0234567890@lannister.casterlyrock.com>;tag=2a202c8336b0dd1f

To: <sip:0323456789@stark.winterfell.com>

Max-Forwards: 70

Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY

User-Agent: TANDBERG/520 (TC7.1.0.48db3d2)

Supported: replaces,100rel,timer,gruu,path,outbound,sdp-anat

Session-Expires: 1800

Content-Type: application/sdp

Content-Length: 2660

SIP Responses

Response Code	Description	Example
1xx	Informational – Request Received and Continuing to Process Request	100 Trying 180 Ringing 183 Session Progress
2xx	Success – Action was successfully received, understood, and accepted	200 OK 202 Acceptable
3xx	Redirection – Another SIP Element needs to be contacted in order to complete the request	300 Multiple Choices 301 Moved Permanently 302 Moved Temporarily
4xx	Client Error – Request contains bad syntax or cannot be fulfilled at this server	401 Unauthorized 404 Not Found 406 Not Acceptable 486 Busy Here 488 Not Acceptable Here
5xx	Server Error – Server failed to fulfill an apparently valid request	503 Service Unavailable
6xx	Global Failure – Request is invalid at any server	600 Busy Everywhere 603 Decline

SIP Response

SIP/2.0 404 Not Found

Free-text Reason

Via: SIP/2.0/UDP 172.18.106.59:5060;branch=z9hG4bKb5291d44b969a4

From: ”Tyrion Lannister” <sip:[illegible]@172.18.106.59>;tag=19210123ca7-45568313

Response code

To: <sip:+19195551212@10.81.2.30>;tag=253488-726

Date: Mon, 16 Jan 2012 04:00:22 GMT

Call-ID: e59bc600-f1319fa5-b1ea4a-3b6a12ac@172.18.106.59

CSeq: 101 INVITE

Allow-Events: telephone-event

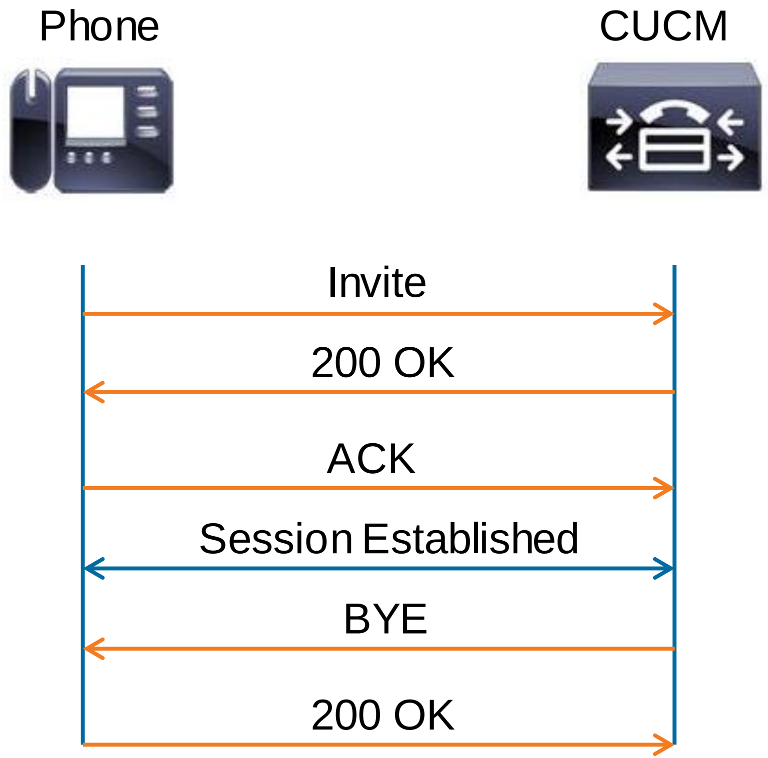
Server: Cisco-SIPGateway/IOS-15.2.2.T

Reason: Q.850;cause=1

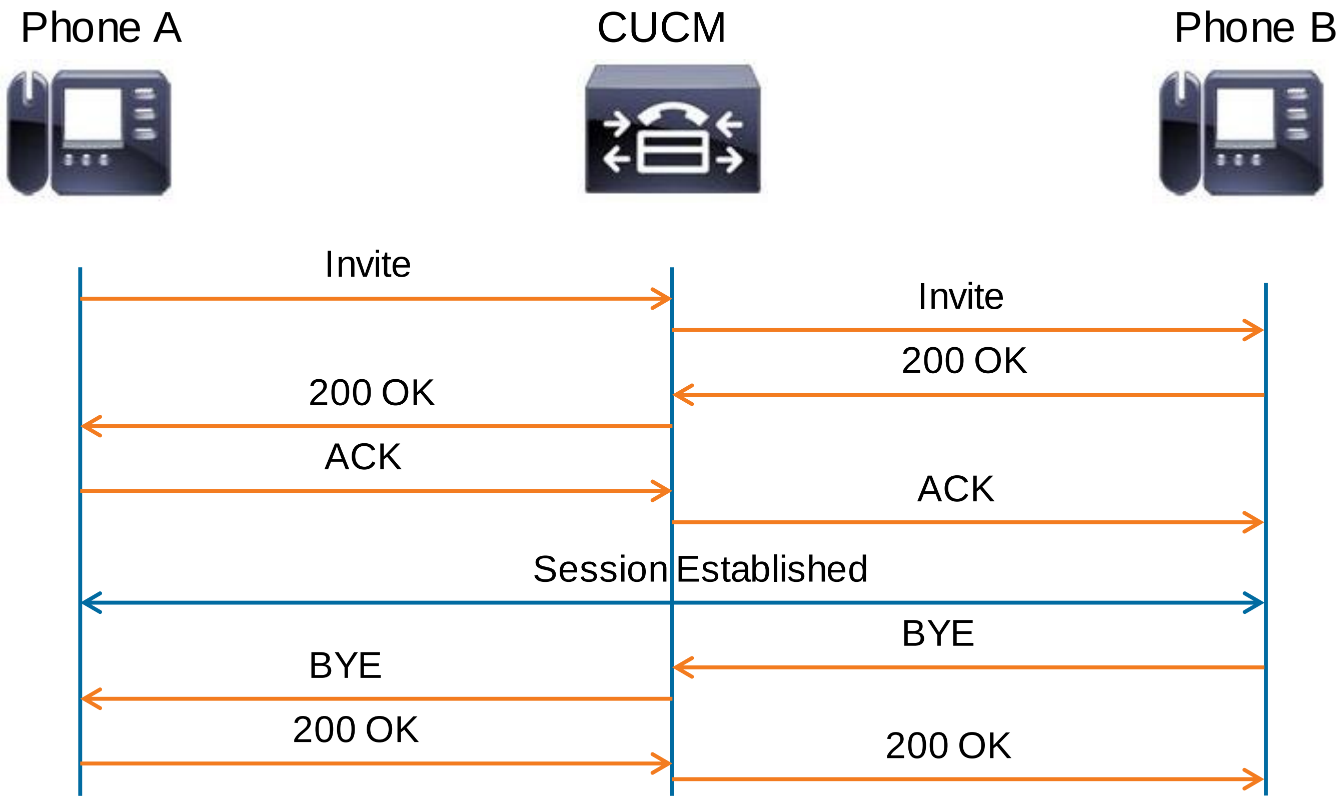
Content-Length: 0

Reason code

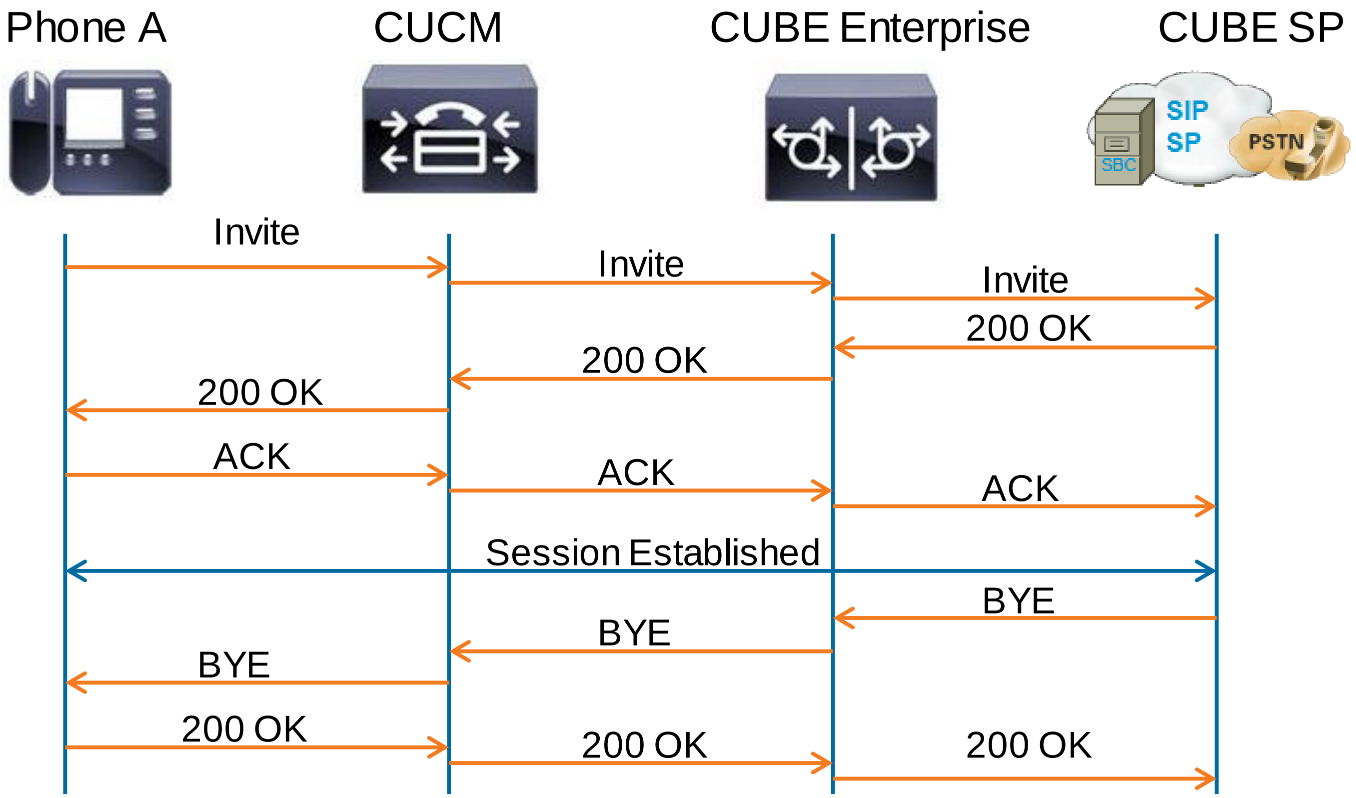
Basic SIP Call Setup



SIP Call Setup with a B2BUA



SIP Call Setup with CUBE



Session Description Protocol (SDP)

- Used for media negotiation
- Session Profile
  - Media type
  - Format
  - Associated properties
- Standard is RFC 4566

Offer/Answer Model (RFC 3264)

- One endpoint sends an offer SDP containing all the capabilities the endpoint wishes to negotiate
- Contains m lines for each media stream being negotiated (i.e. audio, video, content channels, etc.)
- Receiving endpoint sends an answer SDP that contains the same or a subset of capabilities received in the offer.
- For each m= line in the offer, there MUST be a corresponding m= line in the answer. The answer MUST contain exactly the same number of m= lines as the offer.

SDP Fields

- **Session description**
  - v= (protocol version)
  - o= (originator and session identifier)
  - s= (session name)
  - i= (session information)
  - u= (URI of description)
  - e= (email address)
  - p= (phone number)
  - c= (connection information -- not required if included in all media)
  - b= (zero or more bandwidth information lines) One or more time descriptions

SDP Fields

- **Time description**
  - t= (time the session is active)
  - r=* (zero or more repeat times)
- **Media description**
  - m= (media name and transport address)
  - i=* (media title)
  - c=* (connection information -- optional if included at session level)
  - b=* (zero or more bandwidth information lines)
  - k=* (encryption key)
  - a=* (zero or more media attribute lines)

Session Description Protocol (SDP) - Offer

Session Description

```
v=0
o=tandberg 9 3 IN IP4 172.168.0.58
s=SIP Call
b=AS:4064
c=IN IP4 172.168.0.58
t=0 0
```

Media Description

```
m=audio 16454 RTP/AVP 104 9 18 8 0 101
b=TIAS:64000
a=rtpmap:104 G7221/16000
a=fmtp:104 bitrate=32000
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

Media Description

```
m=video 49354 RTP/AVP 97
b=TIAS:4000000
a=label:11
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=428016;max-mbps=108000;max-fs=3600
a=content:main
```

Media Description

```
m=application 58284 RTP/AVP 100
a=rtpmap:100 H224/0
```

Session Description Protocol (SDP) - Answer

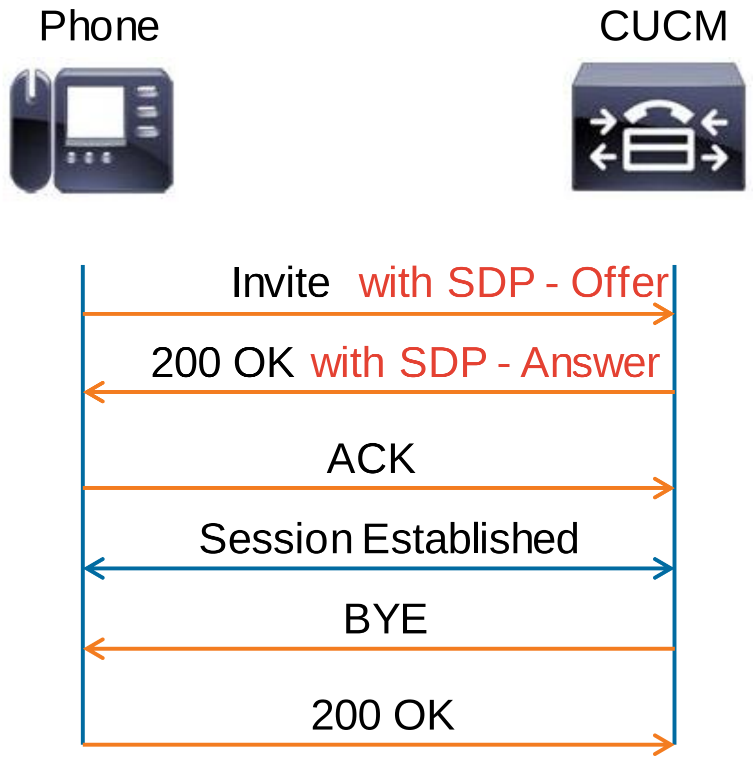
v=0
o=CiscoSystemsCCM-SIP 2000 1 IN IP4 172.168.0.233
s=SIP Call
b=AS:4064
c=IN IP4 172.168.0.233
t=0 0
m=audio 16454 RTP/AVP 104 0 101
b=TIAS:64000
a=rtpmap:104 G7221/16000
a=fmtp:104 bitrate=32000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
m=video 49354 RTP/AVP 97
b=TIAS:4000000
a=label:11
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=428016;max-mbps=108000;max-fs=3600
a=content:main
m=application 58284 RTP/AVP 100
a=rtpmap:100 H224/0

m=audio 16454 RTP/AVP 104 0 101
b=TIAS:64000
a=rtpmap:104 G7221/16000
a=fmtp:104 bitrate=32000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

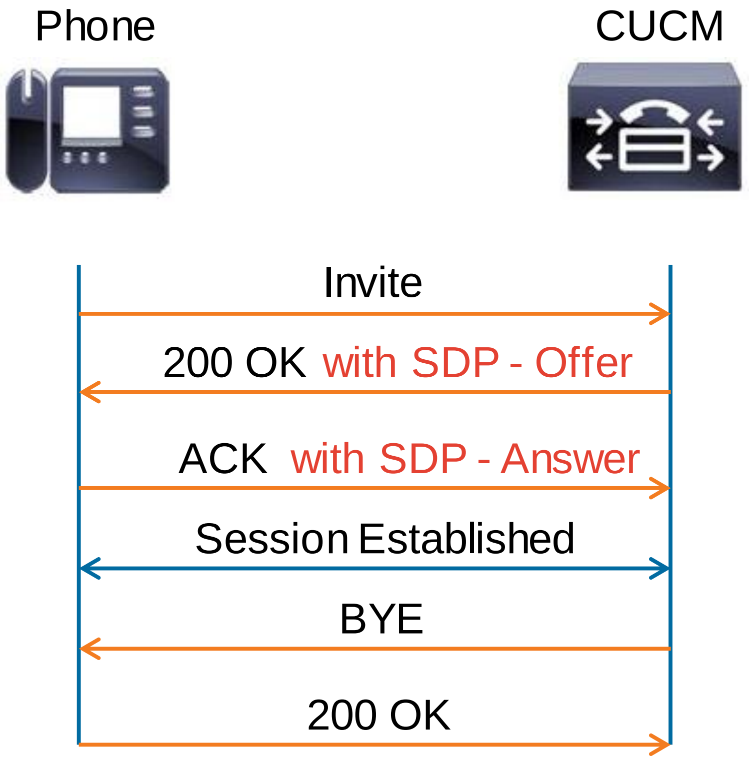
Media Negotiation – Early Offer and Delayed Offer

- Initiator of the call can send SDP offer in the INVITE – this is called an Early Offer (EO)
- Receiving endpoint can send the SDP offer in a response if the INVITE did not contain an offer – this is called a Delayed Offer (DO)
- For Early Offer, the answer is sent in a response (usually 200 OK).
- For Delayed Offer, the answer is typically sent in the ACK.

Early Offer


Phone
CUCM
Invite with SDP - Offer
200 OK with SDP - Answer
ACK
Session Established
BYE
200 OK

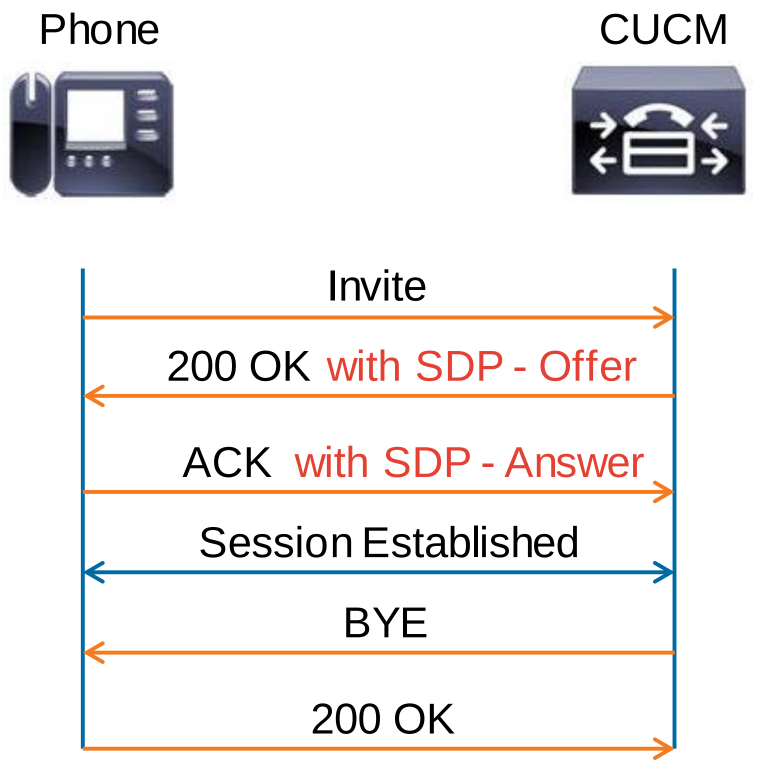
Delayed Offer



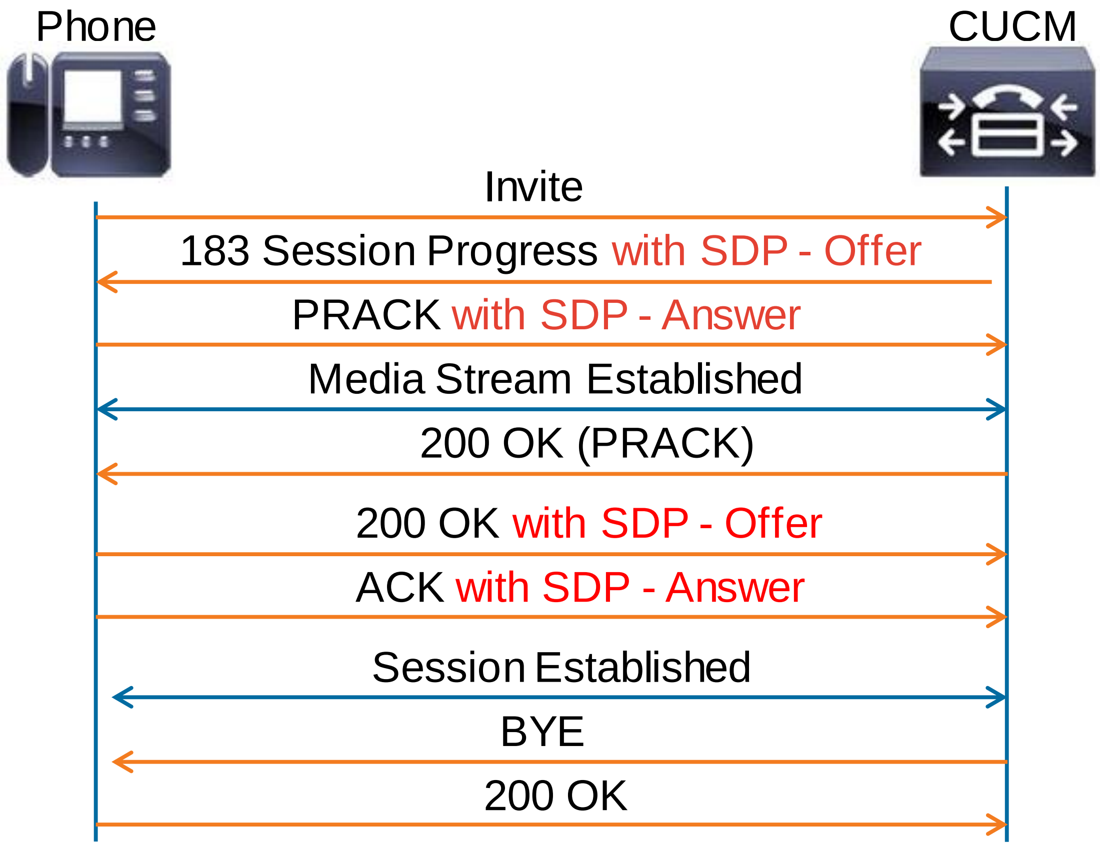
Early Media

- Delayed offer calls do not set up media until the 200 OK (call is answered)
- If media is required prior to the call being connected, SIP has provisions for Early Media
- With Early Media on a Delayed Offer call, the offer comes from the terminating side in a provisional response (e.g. 183 Session Progress)
- Originating side sends SDP Answer in a PRACK message (defined in RFC 3262)

Delayed Offer



Early Media



Phone
CUCM
Invite
183 Session Progress with SDP - Offer
PRACK with SDP - Answer
Media Stream Established
200 OK (PRACK)
200 OK with SDP - Offer
ACK with SDP - Answer
Session Established
BYE
200 OK

Media Re-negotiation – Re-INVITE

- Either UA involved in a call can re-INVITE an existing dialogue to re-negotiate parameters for the call.
- Cannot re-INVITE until any previous INVITE messages have received a final response.
- UPDATE method can also be used to re-negotiate prior to a final

Media Re-negotiation – Re-INVITE

```
INVITE sip:dbe40e44-0dfe-45f1-bd7f-e652098ca344@10.116.101.41:49833;transport=tls SIP/2.0
Via: SIP/2.0/TLS 172.18.106.59:5061;branch=z9hG4bK901f9c72c19221
From: "Paul Giralt" <sip:89915644@172.18.106.59>;tag=15462272~0d0d25d7-4931-4a07-83c6-b82e2c213ca7-45545776
To: <sip:89915644@172.18.106.59>;tag=0022bdd6843100702aae8e5b-4be253be
Date: Wed, 11 Jan 2012 03:08:51 GMT
Call-ID: 8c045780-f0c1fd34-8d838f-3b6a12ac@172.18.106.59
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: 104 INVITE
Max-Forwards: 70
Expires: 180
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= Authenticated; orientation= from; gci= 2-231448; call-instance= 2
Remote-Party-ID: "Paul Giralt" <sip:89915644@172.18.106.59>;party=calling;screen=yes;privacy=off
Contact: <sip:89915644@172.18.106.59:5061;transport=tls>
Content-Type: application/sdp
Content-Length: 489
```


Media Re-negotiation

Re-INVITE – Stopping a Media Session

```
v=0
o=CiscoSystemsCCM-SIP 15462272 2 IN IP4 172.18.106.59
s=SIP Call
c=IN IP4 0.0.0.0
t=0 0
m=audio 19594 RTP/AVP 9 101
a=rtpmap:9 G722/8000
a=ptime:20
a=inactive
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=video 19444 RTP/AVP 126
b=TIAS:1000000
a=rtpmap:126 H264/90000
a=fmtp:126 profile-level-id=42801E;packetization-mode=1;level-asymmetry-allowed=1
a=inactive
a=mid:227796888
```

Media Re-negotiation

Re-INVITE – Delayed Offer to Re-establish Media Stream

```
INVITE sip:dbe40e44-0dfe-45f1-bd7f-e652098ca344@10.116.101.41:49833;transport=tls SIP/2.0
Via: SIP/2.0/TLS 172.18.106.59:5061;branch=z9hG4bK901fac34c0fb1b
From: "Paul Giralt" <sip:89915644@172.18.106.59>;tag=15462272~0d0d25d7-4931-4a07-83c6-b82e2c213ca7-45545776
To: <sip:89915644@172.18.106.59>;tag=0022bdd6843100702aae8e5b-4be253be
Date: Wed, 11 Jan 2012 03:08:52 GMT
Call-ID: 8c045780-f0c1fd34-8d838f-3b6a12ac@172.18.106.59
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: 106 INVITE
Max-Forwards: 70
Expires: 180
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= from; gci= 2-231448; call-instance= 2
Remote-Party-ID: "Paul Giralt" <sip:89915644@172.18.106.59>;party=calling;screen=yes;privacy=off
Contact: <sip:89915644@172.18.106.59:5061;transport=tls>
Content-Length: 0
```

Media Re-negotiation

Re-INVITE – Offer in 200 OK

SIP/2.0 200 OK
Via: SIP/2.0/TLS 172.18.106.59:5061;branch=z9hG4bK901fac34c0fb1b
From: "Paul Giralt" <sip:89915644@172.18.106.59>;tag=15462272~0d0d25d7-4931-4a07-83c6-b82e2c213ca7-45545776
To: <sip:89915644@172.18.106.59>;tag=0022bdd6843100702aae8e5b-4be253be
Call-ID: 8c045780-f0c1fd34-8d838f-3b6a12ac@172.18.106.59
Date: Wed, 11 Jan 2012 03:08:52 GMT
CSeq: 106 INVITE
Server: Cisco-CPCIUS/9.2.1
Contact: <sip:dbe40e44-0dfe-45f1-bd7f-e652098ca344@10.116.101.41:49833;transport=tls>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "Paul Giralt" <sip:89915644@172.18.106.59>;party=called;id-type=subscriber;privacy=off;screen=yes
Supported: replaces,join,sdp-anat,norefersub,extended-refer,X-cisco-callinfo,X-cisco-serviceuri,
Allow-Events: kpml,dialog
Recv-Info: conference
Recv-Info: x-cisco-conference
Content-Length: 788
Content-Type: application/sdp
Content-Disposition: session;handling=optional

Media Re-negotiation

Re-INVITE – Offer in 200 OK

```
v=0
o=Cisco-SIPUA 26259 2 IN IP4 10.116.101.41
s=SIP Call
t=0 0
m=audio 32518 RTP/AVP 0 8 18 102 9 116 124 101
c=IN IP4 10.116.101.41
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:102 L16/16000
a=rtpmap:9 G722/8000
a=rtpmap:116 iLBC/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
m=video 17614 RTP/AVP 126 97
c=IN IP4 10.116.101.41
b=TIAS:2500000
a=rtpmap:126 H264/90000
a=fmtp:126 profile-level-id=42801F;packetization-mode=1;level-asymmetry-allowed=1
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42801F;packetization-mode=0;level-asymmetry-allowed=1
a=sendrecv
```

Media Re-negotiation

Re-INVITE – Answer in ACK

ACK sip:dbe40e44-0dfe-45f1-bd7f-e652098ca344@10.116.101.41:49833;transport=tls SIP/2.0
Via: SIP/2.0/TLS 172.18.106.59:5061;branch=z9hG4bK901fb064465a06
From: "Paul Giralt" <sip:89915644@172.18.106.59>;tag=15462272~0d0d25d7-4931-4a07-83c6-b82e2c213ca7-45545776
To: <sip:89915644@172.18.106.59>;tag=0022bdd6843100702aae8e5b-4be253be
Date: Wed, 11 Jan 2012 03:08:52 GMT
Call-ID: 8c045780-f0c1fd34-8d838f-3b6a12ac@172.18.106.59
Max-Forwards: 70
CSeq: 106 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 446

Media Re-negotiation

Re-INVITE – Answer in ACK – Decline Video Support

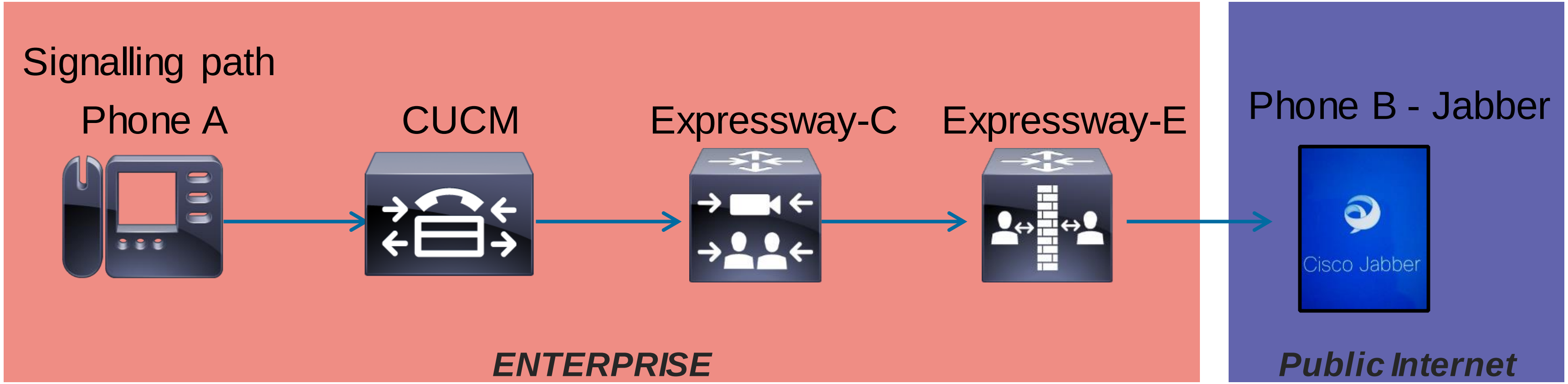
```
v=0
o=CiscoSystemsCCM-SIP 15462272 3 IN IP4 172.18.106.59
s=SIP Call
t=0 0
m=audio 4000 RTP/AVP 0
c=IN IP4 172.18.106.58
a=rtpmap:0 PCMU/8000
a=ptime:20
a=sendonly
m=video 0 RTP/AVP 126
c=IN IP4 10.116.101.50
b=TIAS:1000000
a=rtpmap:126 H264/90000
a=fmtp:126 profile-level-id=42801E;packetization-mode=1;level-asymmetry-allowed=1
a=mid:227796888
```

Troubleshoot Expressway

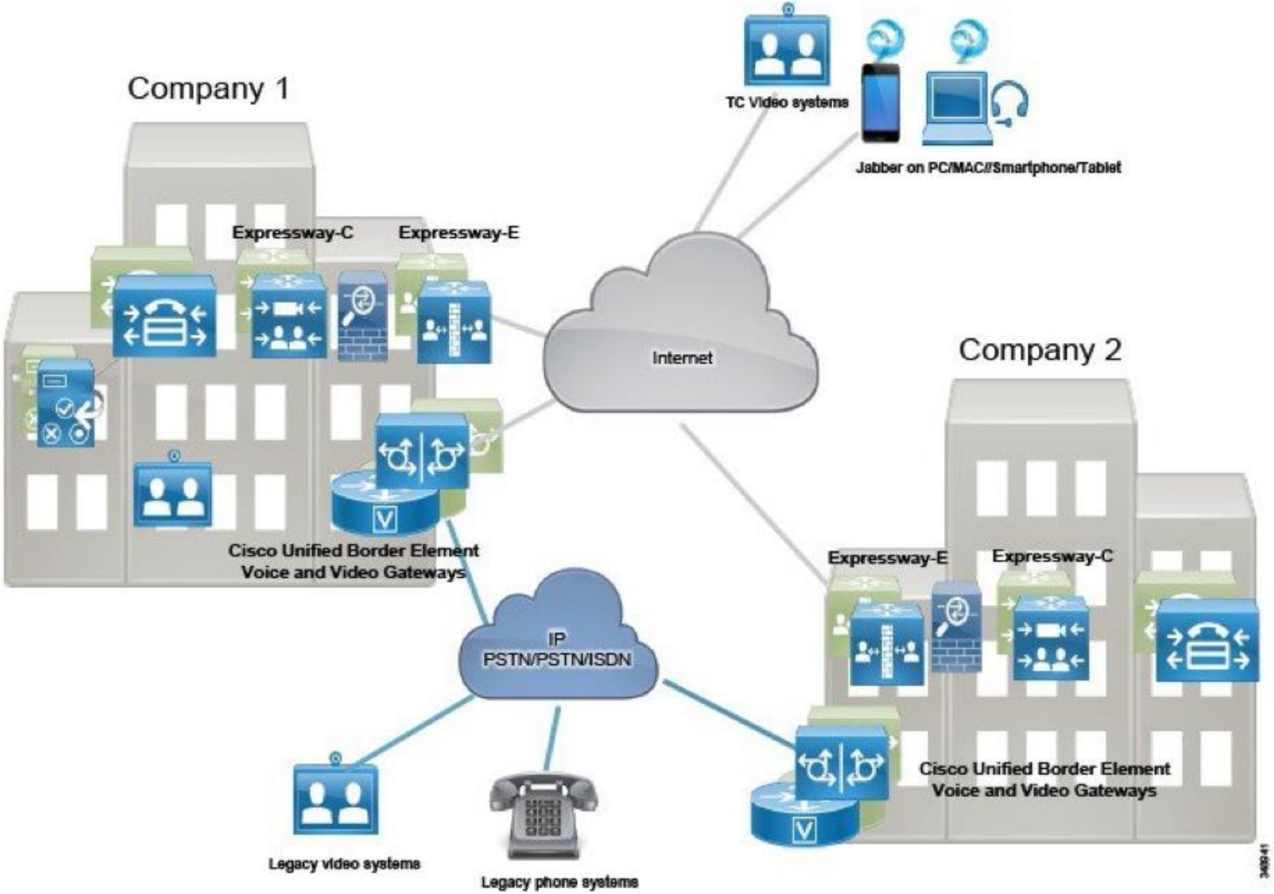
Troubleshoot Expressway

Problem Description

- Calls from an endpoint inside the enterprise to a mobile Jabber client on IOS/Android fails when pushing answer.

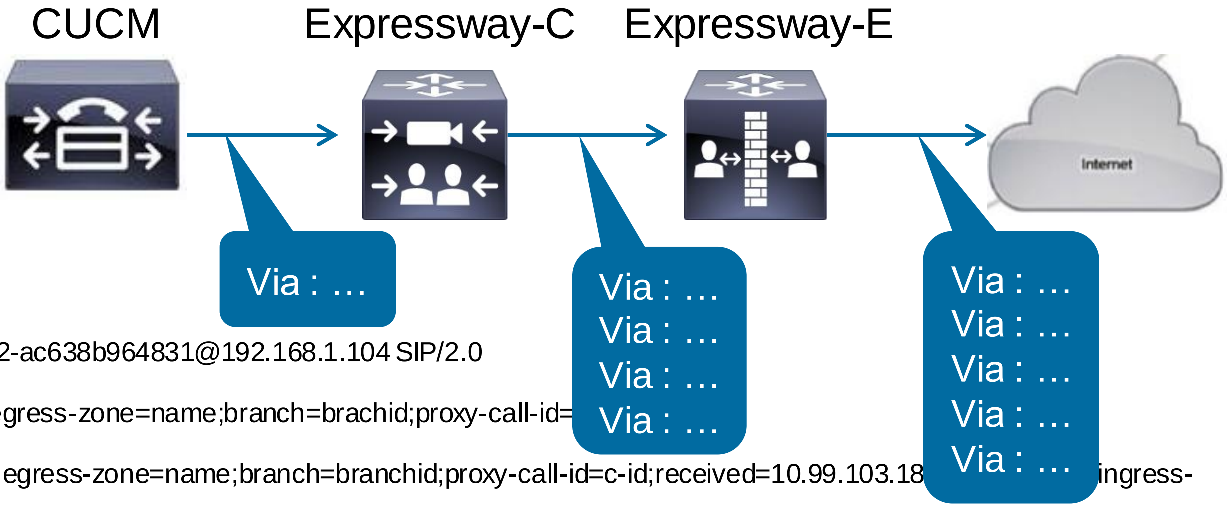


Expressway MRA Overview



Expressway Case Study

Expressway-Edge to Firewall



INVITE sip:e5a44831-0d3c-a090-b092-ac638b964831@192.168.1.104 SIP/2.0

Via: SIP/2.0/TLS 136.153.17.7:5061;egress-zone=name;branch=brachid;proxy-call-id=

Via: SIP/2.0/TLS 10.99.103.180:5061;egress-zone=name;branch=branchid;proxy-call-id=c-id;received=10.99.103.18 ingress-zone=CollaborationEdge

Via: SIP/2.0/TLS 10.99.103.180:5073;branch=branchid;x-cisco-local-service=nettle;received=10.99.103.180;rport=33268;ingress-zone=DefaultZone

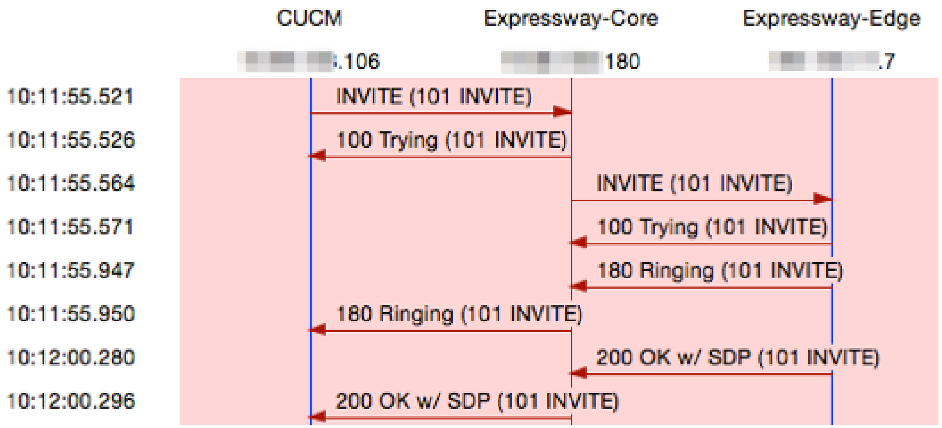
Via: SIP/2.0/TLS 10.99.103.180:5061;egress-zone=DefaultZone;branch=branchid;proxy-call-id=c-id;received=10.99.103.180;rport=25216

Via: SIP/2.0/TCP 10.99.103.106:5060;branch=z9hG4bK117141557c6be;received=10.99.103.106;ingress-zone=CEtcp1099103106

Expressway Case Study

Ladder Diagram

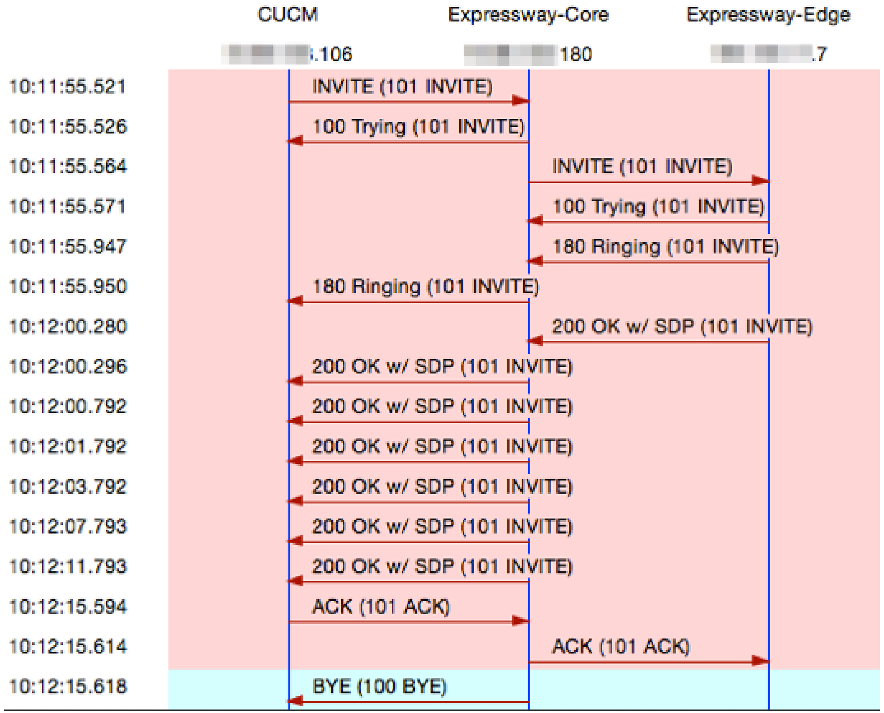
- Call setup



Expressway Case Study

Ladder Diagram

- 15 seconds to send an ACK



Expressway Case Study

Solution

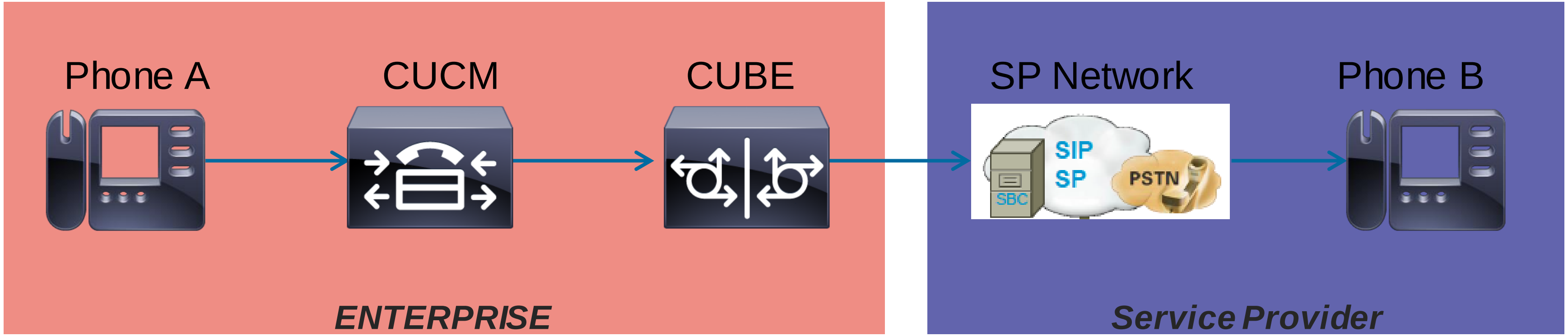
- From the MRA deployment guide we found that:
  - MRA feature wasn't supported by the version of CUCM
  - Upgrade to a CUCM version to 9.1(2)SU1

Troubleshoot One Way Audio

Troubleshoot One Way Audio

Problem Description

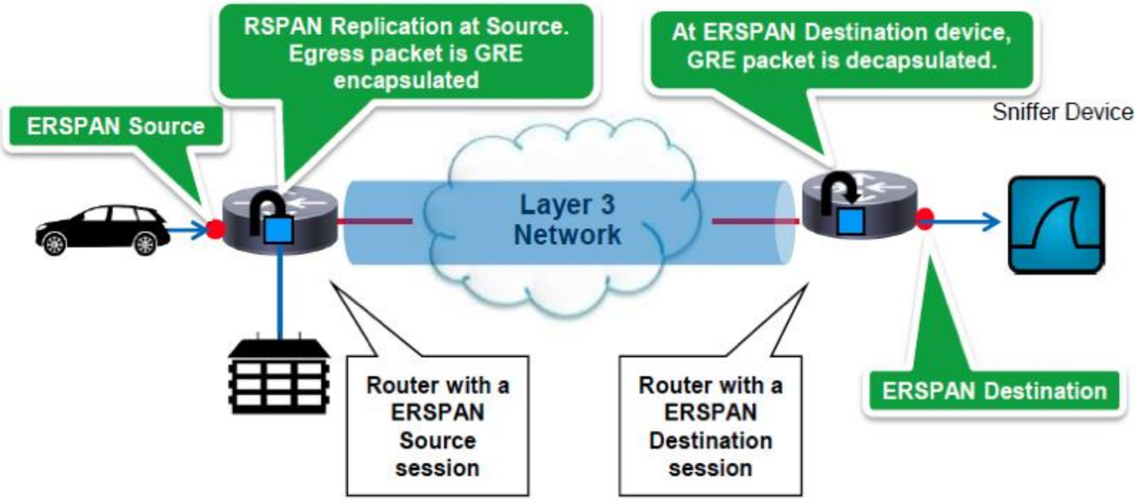
- Calls from some endpoints experience one way audio
- Call flow:



Encapsulated Remote SPAN (ERSPAN)

Overview

- ERSPAN supports source ports, source VLANs, and destination on different switches
- It Uses a GRE tunnel to carry traffic
- ERSPAN consists of an ERSPAN source session, routable ERSPAN GRE-encapsulated traffic, and an ERSPAN destination session

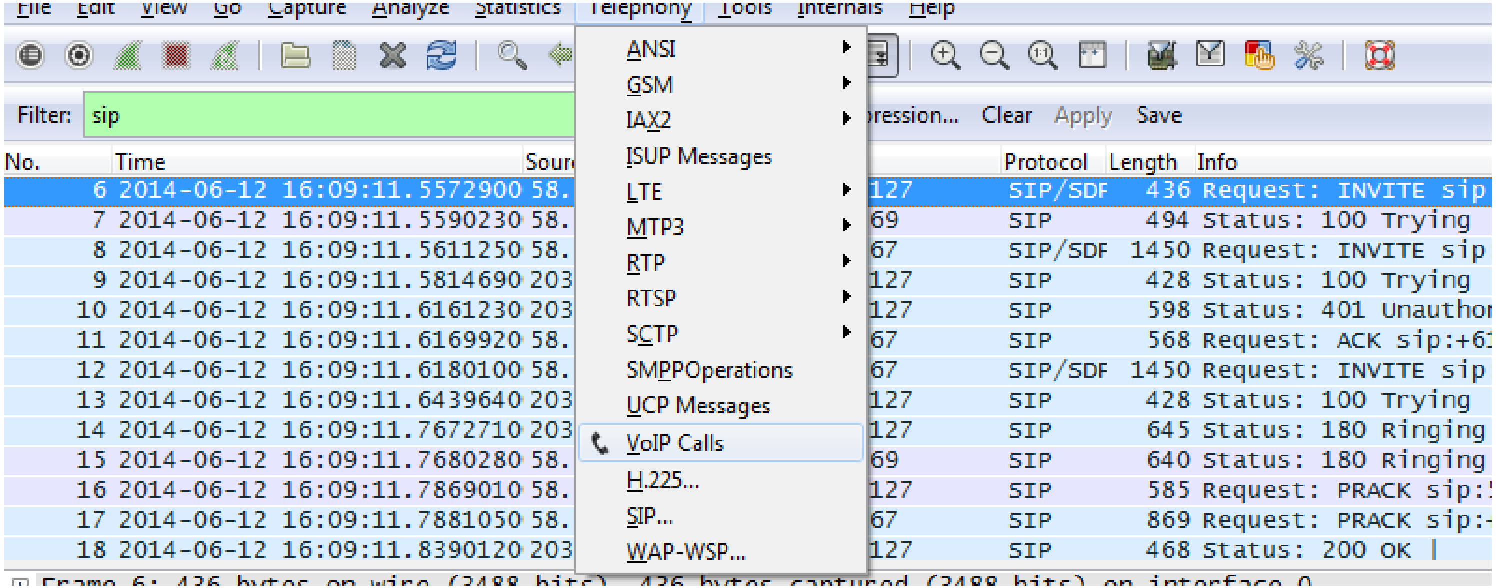


Wireshark

- Open Source network packet capture and analysis tool
- Available at http://www.wireshark.org
- Available for Windows, Mac OS X, and UNIX/Linux
- Provides VoIP Call and SIP analysis

Wireshark

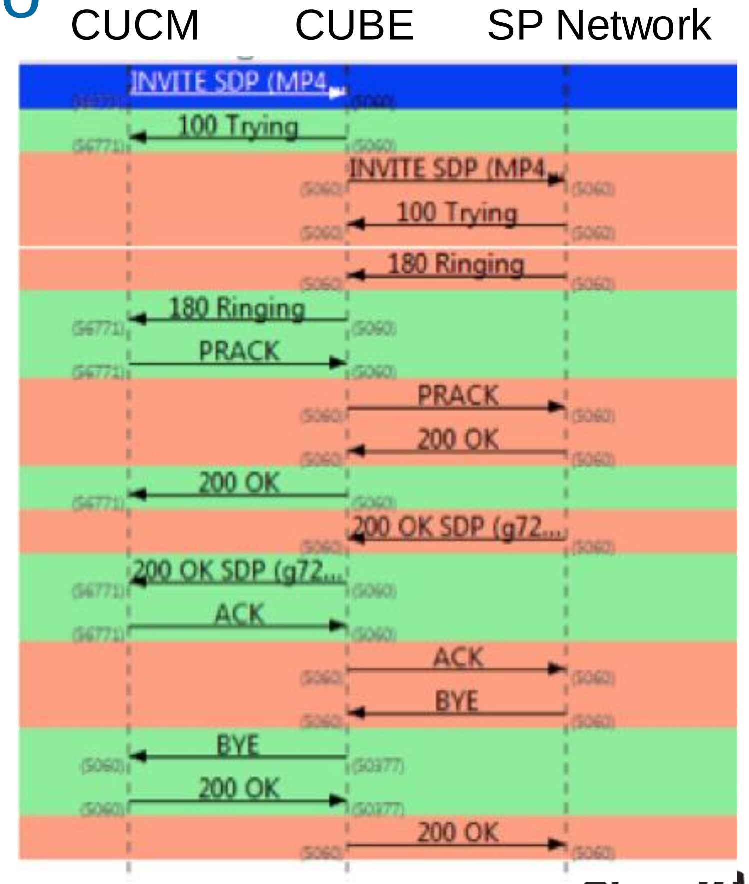
VoIP call analysis



Troubleshoot One Way Audio

VoIP Ladder diagram

- Follow Call Setup
- Confirm interactions are correct
- Does the call setup look normal?



Troubleshoot One Way Audio

RTP packets

- After call establishment
- RTP flows in both directions

297	14.7145190	.69	6.57	SIP	444	Status: 200 OK |
298	14.7147980	6.57	.69	TCP	60	49692→5060 [ACK] Seq=10042 Ack=8857 Win=65535 Len=0
299	14.7176740	6.57	167	RTP	214	PT=ITU-T G.722, SSRC=0x4DEA34C9, Seq=7312, Time=280010303
300	14.7280950	27	6.57	RTP	214	PT=ITU-T G.722, SSRC=0x21670082, Seq=45361, Time=3244919900
301	14.7364500	6.57	167	RTP	214	PT=ITU-T G.722, SSRC=0x4DEA34C9, Seq=7313, Time=280010463
302	14.7479750	.2		HSRP	62	Hello (state Active)
303	14.7483590	27	6.57	RTP	214	PT=ITU-T G.722, SSRC=0x21670082, Seq=45362, Time=3244920060
304	14.7569430	6.57	167	RTP	214	PT=ITU-T G.722, SSRC=0x4DEA34C9, Seq=7314, Time=280010623
305	14.7680750	27	6.57	RTP	214	PT=ITU-T G.722, SSRC=0x21670082, Seq=45363, Time=3244920220

Troubleshoot One Way Audio

SDP

- Find destination IP in SDP
- Compare with working call

```
⊟ Session Description Protocol
    Session Description Protocol Version (v): 0
  ⊞ Owner/Creator, Session Id (o): BroadWorks 115649443 1 IN IP4 [illegible].167
    Session Name (s): -
  ⊞ Connection Information (c): IN IP4 [illegible]167
  ⊞ Time Description, active time (t): 0 0
  ⊞ Media Description, name and address (m): audio 19424 RTP/AVP 9 101
  ⊞ Media Attribute (a): rtpmap:9 G722/8000
  ⊞ Media Attribute (a): rtpmap:101 telephone-event/8000
  ⊞ Media Description, name and address (m): video 0 RTP/AVP 100 126 97
  ⊞ Connection Information (c): IN IP4 [illegible]167
  ⊞ Media Attribute (a): label:11
  ⊞ Media Attribute (a): rtpmap:100 H264/90000
  ⊞ Media Attribute (a): fmtp:100 profile-level-id=640028;packetization-mode=1
```


Solution

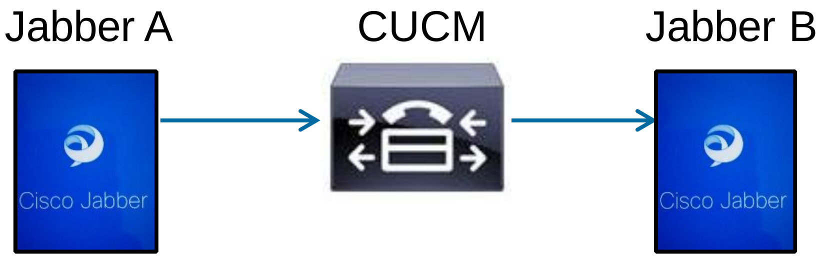
- Remote side was sending an incorrect IP
- Contact the service provider to resolve.

Troubleshoot Content Sharing

Troubleshoot Content Sharing

Problem Description

- Jabber Desktop sharing fails
- Call flow:



Troubleshoot Content Sharing

SDP - Invite

```
v=0
o=Cisco-SIPUA 9180 0 IN IP4 172.16.0.154
s=SIP Call
c=IN IP4 172.16.0.154
b=AS:4000
t=0 0
m=audio 17924 RTP/AVP 0 8 18 105 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:105 G7221/16000
a=fmtp:105 bitrate=24000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
m=video 31886 RTP/AVP 97
b=TIAS:4000000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42E01E;packetization-mode=0;lev
a=content:main
a=label:11
a=sendrecv
m=application 5070 UDP/BFCP *
a=floorctrl:c-s
a=confid:1
a=floorid:2 mstrm:12
a=userid:1
a=setup:actpass
a=connection:new
a=sendrecv
```


m=application 5070 UDP/BFCP *
a=floorctrl:c-s
a=confid:1
a=floorid:2 mstrm:12
a=userid:1
a=setup:actpass
a=connection:new
a=sendrecv

Troubleshoot Content Sharing

SDP – Invite

```
v=0
o=Cisco-SIPUA 9180 0 IN IP4 172.16.0.155
s=SIP Call
c=IN IP4 172.16.0.155
b=AS:4000
t=0 0
m=audio 17924 RTP/AVP 0 8 18 105 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:105 G7221/16000
a=fmtp:105 bitrate=24000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
m=video 31886 RTP/AVP 97
b=TIAS:4000000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42E01E;packetization-mode=0;level-asymmetry-allowed=1;max-fs=1621
a=content:main
a=label:11
a=sendrecv
m=application 0 UDP/BFCP *
a=floorctrl:c-s
a=confid:1
a=floorid:2 mstrm:12
a=userid:1
a=setup:actpass
a=connection:new
a=sendrecv
```

m=application 0 UDP/BFCP *

Troubleshoot Content Sharing

CUCM Log

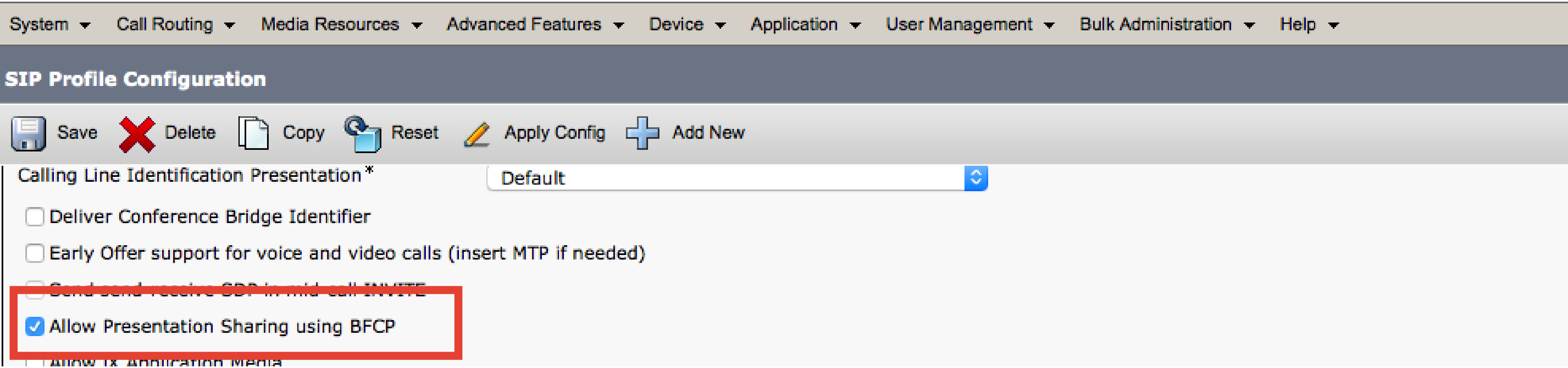
12:37:19.664 |//SIP/SIPCdpc(3,74,1103501)/ci=0/ccbId=0/scbId=0/handleSDPOfferInd:ipAddrMode IpAddrMode_v4 |3,100,68,804909.1^*^*
12:37:19.664 |//SIP/SIPCdpc(3,74,1103501)/ci=62953946/ccbId=12275384/scbId=0/sendSIPMediaToUpdateCc: Secure status=1, mSrtpPresent=03,100,68,804909.1^*^*
12:37:19.664 |//SIP/SIPCdpc(3,74,1103501)/ci=62953946/ccbId=12275384/scbId=0/getDefCcRegister: Secure status=1,
12:37:19.664 |SIPSdp::session level AS found. bandwidth.enabledMask=1 bandwidth.as=4000|*^*^*
12:37:19.664 |//SIP/SIPCdpc(3,74,1103501)/ci=62953946/ccbId=12275384/scbId=0/handleSDPOfferInd: Sending offer to
12:37:19.664 |DET-SIPInterface-(804908)::BFCP is disab 4908.1^*^*
12:37:19.664 |DET-SDPMsg-getBWFromSession: 1 0,400
12:37:19.664 |DET-SDPMsg-getBWFromSession: sessio
12:37:19.664 |DET-SDPMsg-()::setTIASforDO, sessionB tedAudio
12:37:19.664 |DET-SDPMsg-()::setTIASforVideo, session bps=0|*^*^*
12:37:19.664 |DET-SDPMsg-()::setSessionBandwidthModifiers(bw= action=0) - session(bitmask=0x1,tias=0,as=*
12:37:19.664 |DET-SIPInterface-(804908)::waitSDPResponse_SDPO erInd, updated SDPMode(1st mline)=aud 0, vid
12:37:19.664 |DET-SIPInterface-(804908)::**BFCP is disabled:Filter out Presentation Video and BFCP**|3,100,68,804908.1^*^*
12:37:19.664 |SIG-MediaManager-(562647)::wait_AuConnectReply|3,100,68,804909.1^*^*
12:37:19.664 |SIG-MediaManager-(562647)::wait_AuConnectReply, received 1 resps, sent AuConnecReply for party
12:37:19.664 |DET-SIPInterface-(804909)::BFCP is disabled:Filter out Presentation Video and BFCP|3,100,68,804909.1^*^*
12:37:19.664 |DET-SDPMsg-getBWFromSession: 1 0,4000,0|*^*^*

BFCP is disabled: Filter out Presentation Video and BFCP

Troubleshoot Content Sharing

Solution

- Enable BFCP on CUCM
- Device -> Device Settings -> SIP Profile

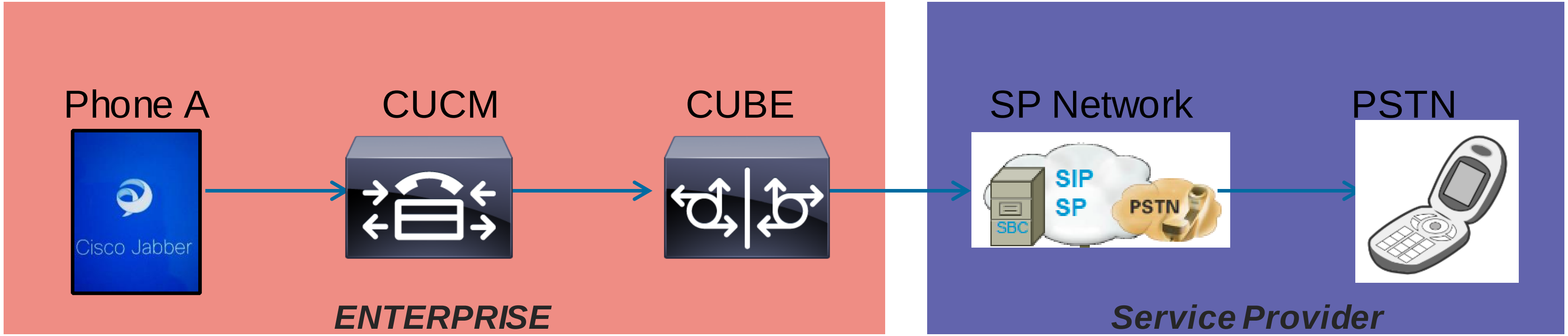


Troubleshoot a UC Application

Troubleshoot a UC Application

Problem Description

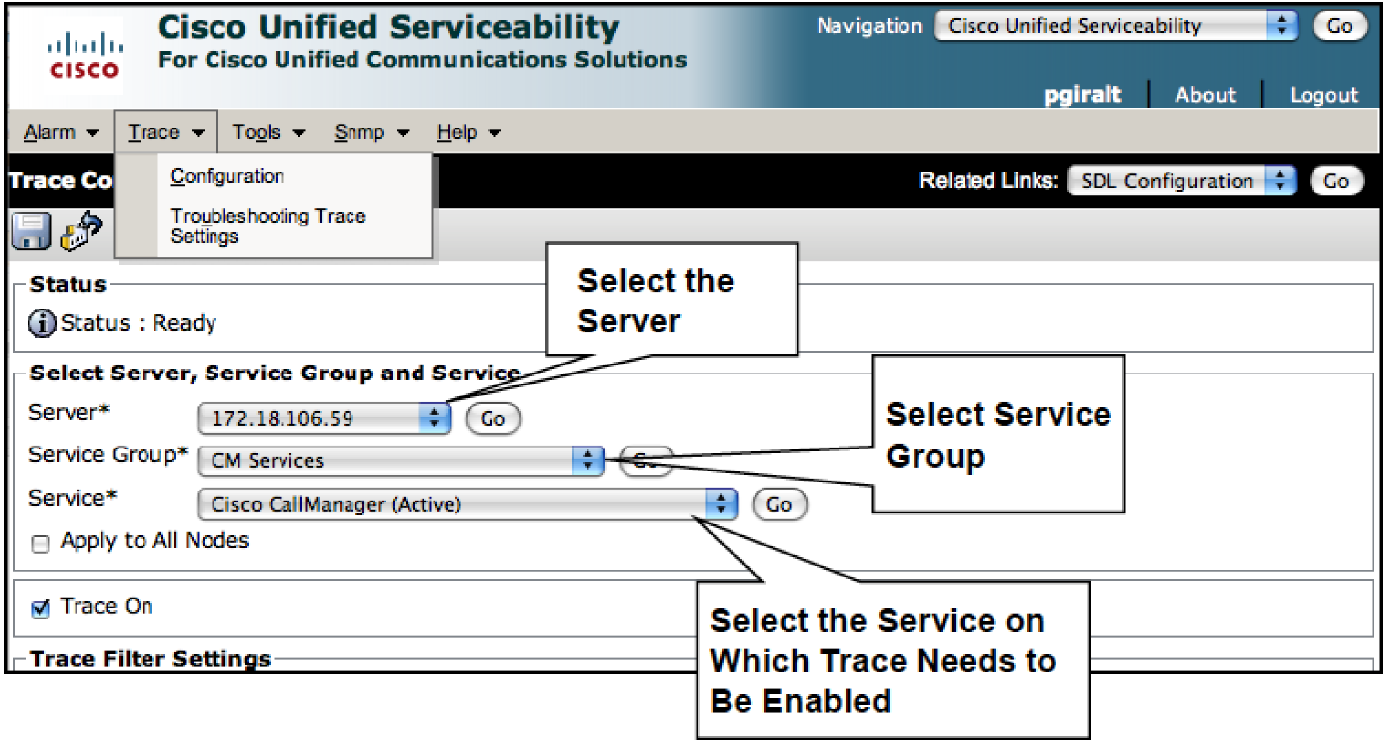
- Calls fail after hold/resume for calls from Jabber endpoints



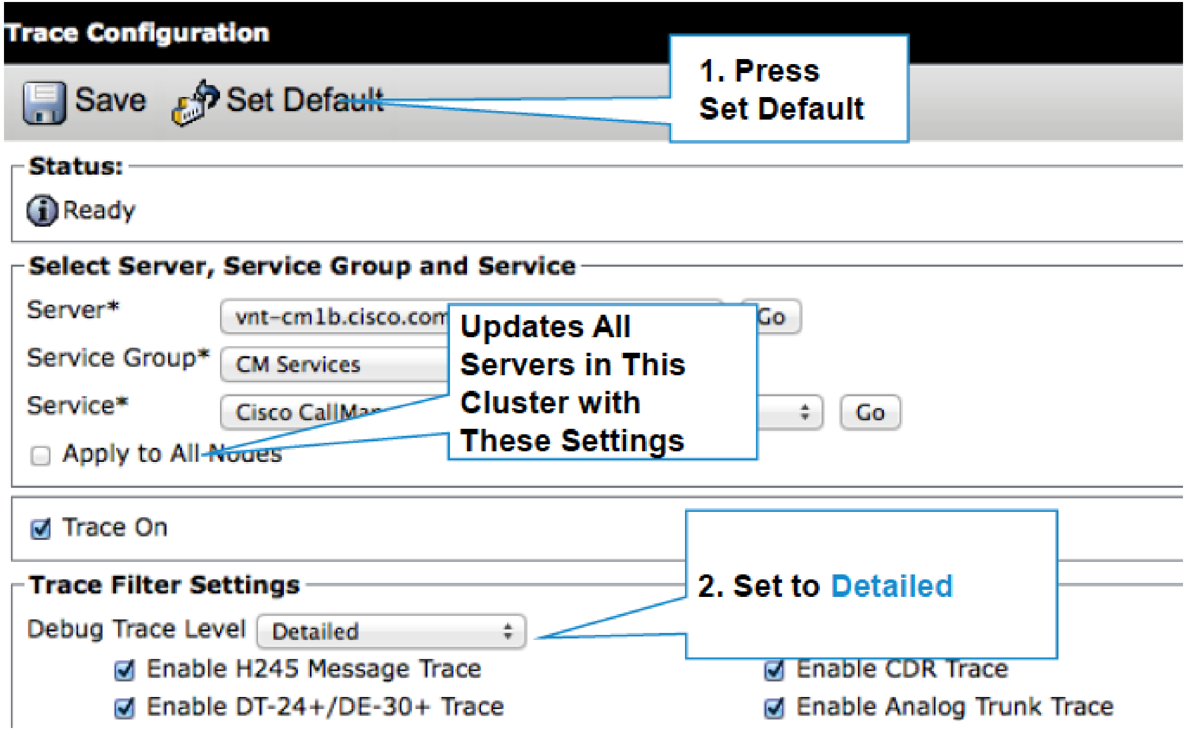
Unified CM Trace Configuration

- SIP messaging in CUCM is written to the CCM/SDI trace file when appropriate trace levels are set (SDL trace in 9.0+)
- Configured from **Cisco Unified Serviceability** > **Trace** > **Configuration** or by using AnalysisManager
- Unified CM 9.0 combines SDI and SDL traces into the SDL traces
- Unified CM 9.0 and later default to detailed tracing – no need to configure traces.

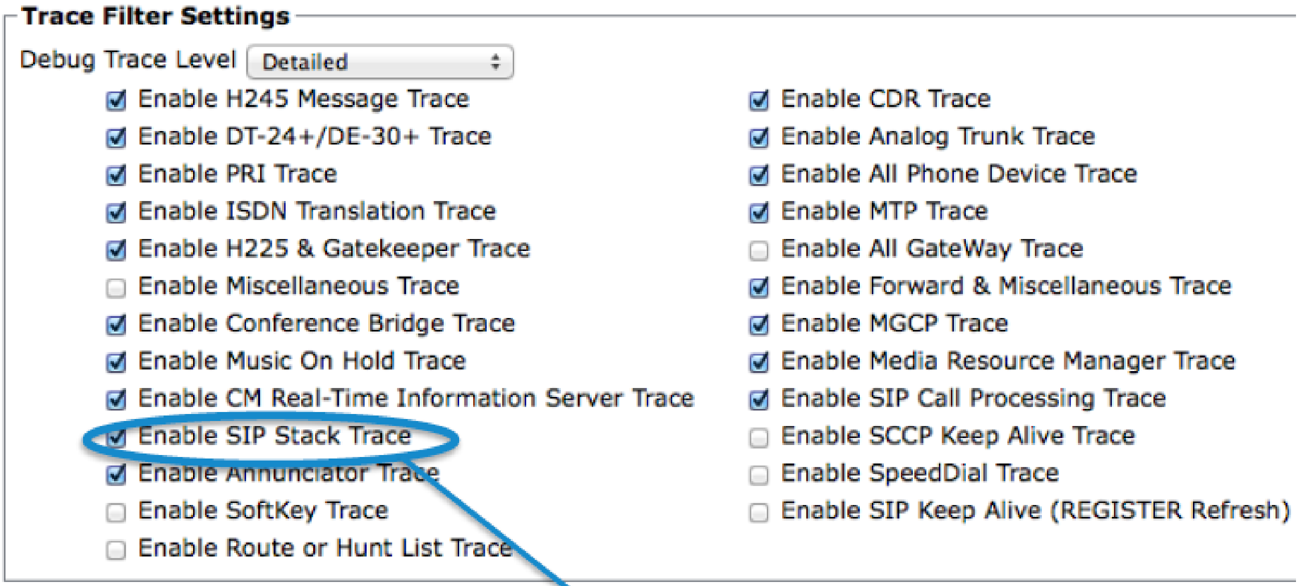
Unified CM Trace Configuration



Unified CM Trace Configuration



Unified CM Trace Configuration

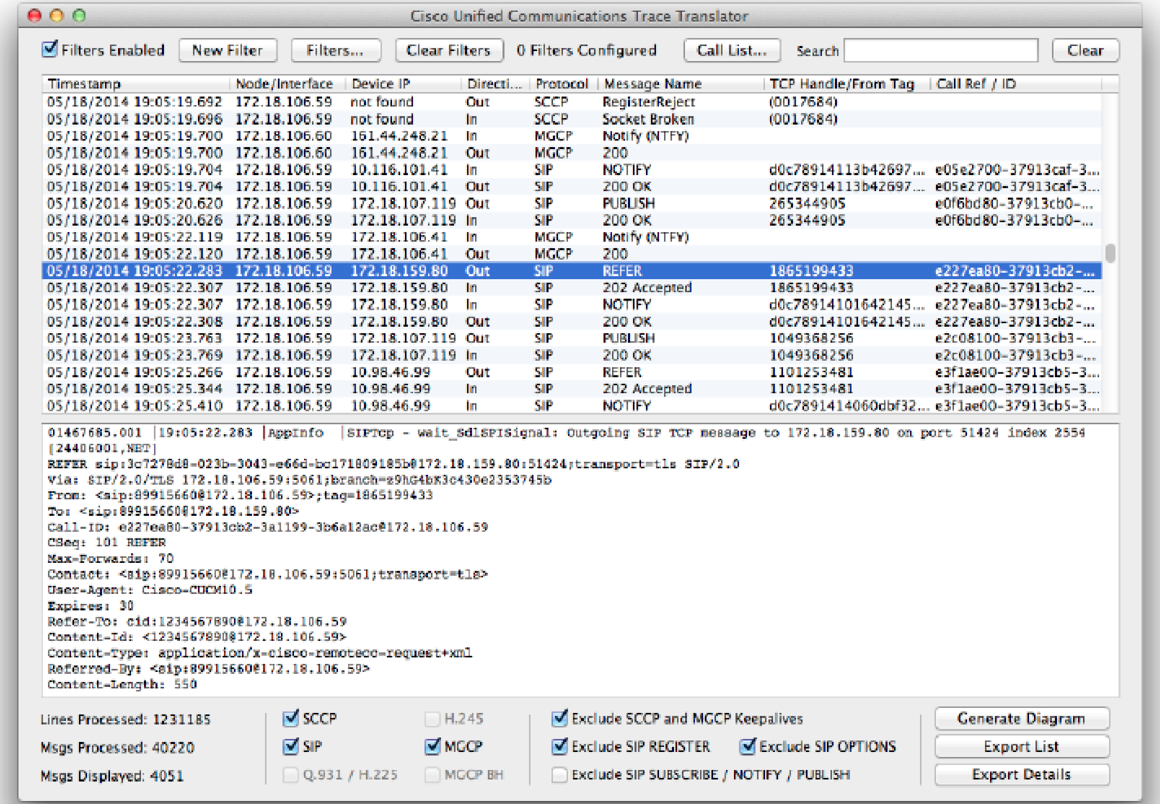


Enable SIP Stack Trace is NOT needed to see SIP Messages.
Do not enable SIP Stack Trace prior to 9.0 unless directed by TAC.
Okay to have enabled in 9.x and later

TranslatorX Tool

- Download from http://translatorx.cisco.com/
- Developed and maintained by Paul Giralt
- Supports Q.931, H.225, SCCP (Skinny), MGCP, or SIP messages.

TranslatorX Tool



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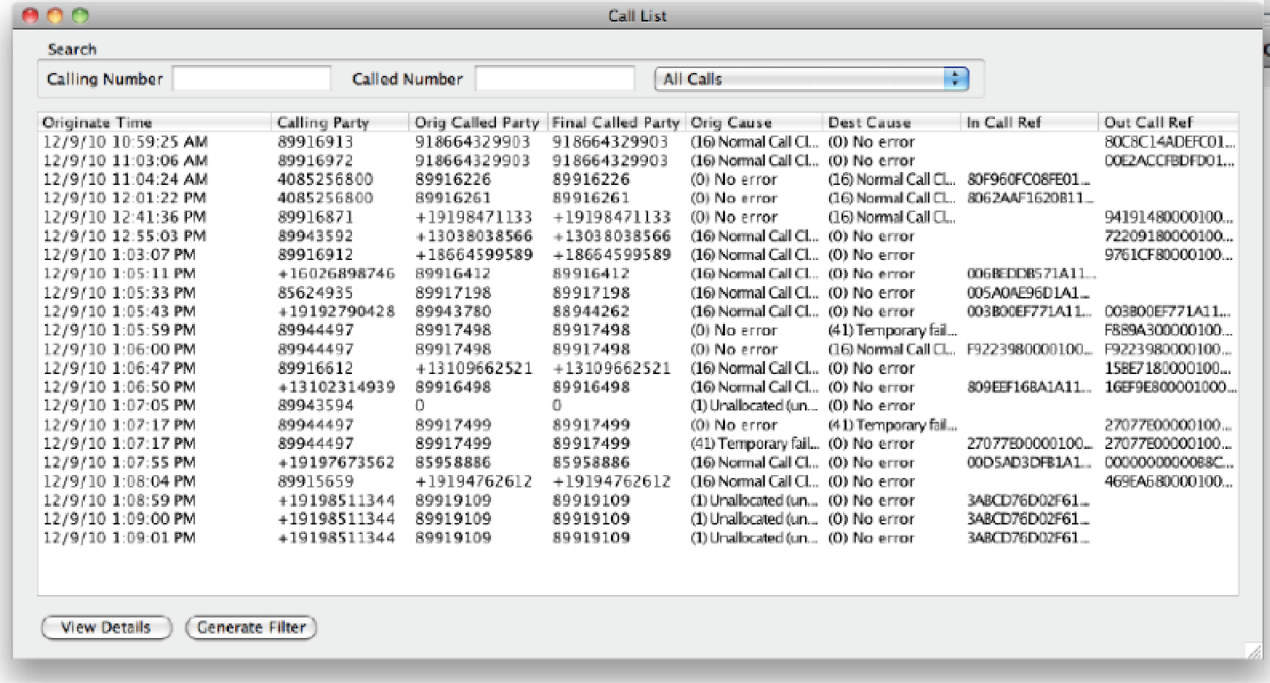
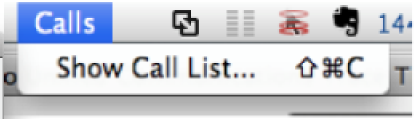
Translatorx

Call List Window



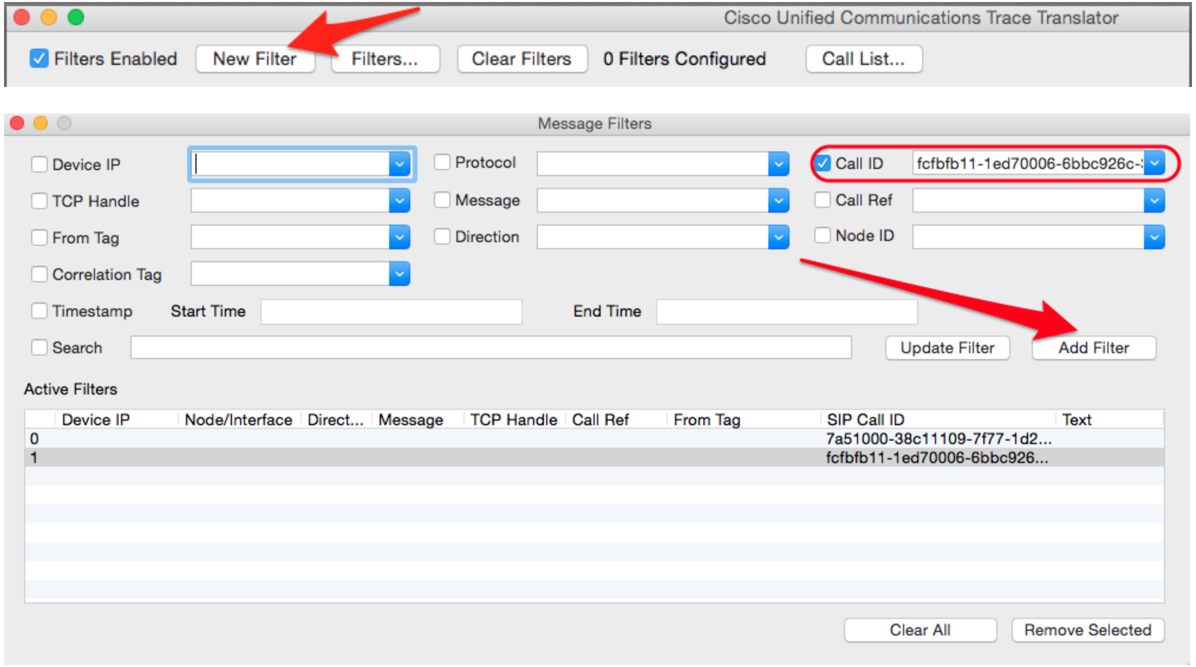
Originate Time	Calling Party	Orig Called Party	Final Called Party	Orig Cause	Dest Cause	In Call Ref	Out Call Ref
12/9/10 10:59:25 AM	89916913	918664329903	918664329903	(16) Normal Call Cl...	(0) No error		80C8C14ADEFC01...
12/9/10 11:03:06 AM	89916972	918664329903	918664329903	(16) Normal Call Cl...	(0) No error		00E2ACCFBDFD01...
12/9/10 11:04:24 AM	4085256800	89916226	89916226	(0) No error	(16) Normal Call Cl...	80F960FC08FE01...	
12/9/10 12:01:22 PM	4085256800	89916261	89916261	(0) No error	(16) Normal Call Cl...	8062AAF1620B11...	
12/9/10 12:41:36 PM	89916871	+19198471133	+19198471133	(0) No error	(16) Normal Call Cl...		94191480000100...
12/9/10 12:55:03 PM	89943592	+13038038566	+13038038566	(16) Normal Call Cl...	(0) No error		72209180000100...
12/9/10 1:03:07 PM	89916912	+18664599589	+18664599589	(16) Normal Call Cl...	(0) No error		9761CF80000100...
12/9/10 1:05:11 PM	+16026898746	89916412	89916412	(16) Normal Call Cl...	(0) No error	006BEDDB571A11...	
12/9/10 1:05:33 PM	85624935	89917198	89917198	(16) Normal Call Cl...	(0) No error	005A0AE96D1A1...	
12/9/10 1:05:43 PM	+19192790428	89943780	88944262	(16) Normal Call Cl...	(0) No error	003B00EF771A11...	003B00EF771A11...
12/9/10 1:05:59 PM	89944497	89917498	89917498	(0) No error	(41) Temporary fail...		F889A300000100...
12/9/10 1:06:00 PM	89944497	89917498	89917498	(0) No error	(16) Normal Call Cl...	F9223980000100...	F9223980000100...
12/9/10 1:06:47 PM	89916612	+13109662521	+13109662521	(16) Normal Call Cl...	(0) No error		15BE7180000100...
12/9/10 1:06:50 PM	+13102314939	89916498	89916498	(16) Normal Call Cl...	(0) No error	809EEF168A1A11...	16EF9E800001000...
12/9/10 1:07:05 PM	89943594	0	0	(1) Unallocated (un...	(0) No error		
12/9/10 1:07:17 PM	89944497	89917499	89917499	(0) No error	(41) Temporary fail...		27077E00000100...
12/9/10 1:07:17 PM	89944497	89917499	89917499	(41) Temporary fail...	(0) No error	27077E00000100...	27077E00000100...
12/9/10 1:07:55 PM	+19197673562	85958886	85958886	(16) Normal Call Cl...	(0) No error	00D5AD3DFB1A1...	0000000000088C...
12/9/10 1:08:04 PM	89915659	+19194762612	+19194762612	(16) Normal Call Cl...	(0) No error		469EA680000100...
12/9/10 1:08:59 PM	+19198511344	89919109	89919109	(1) Unallocated (un...	(0) No error	3ABCD76D02F61...	
12/9/10 1:09:00 PM	+19198511344	89919109	89919109	(1) Unallocated (un...	(0) No error	3ABCD76D02F61...	
12/9/10 1:09:01 PM	+19198511344	89919109	89919109	(1) Unallocated (un...	(0) No error	3ABCD76D02F61...	



Troubleshoot a UC Application

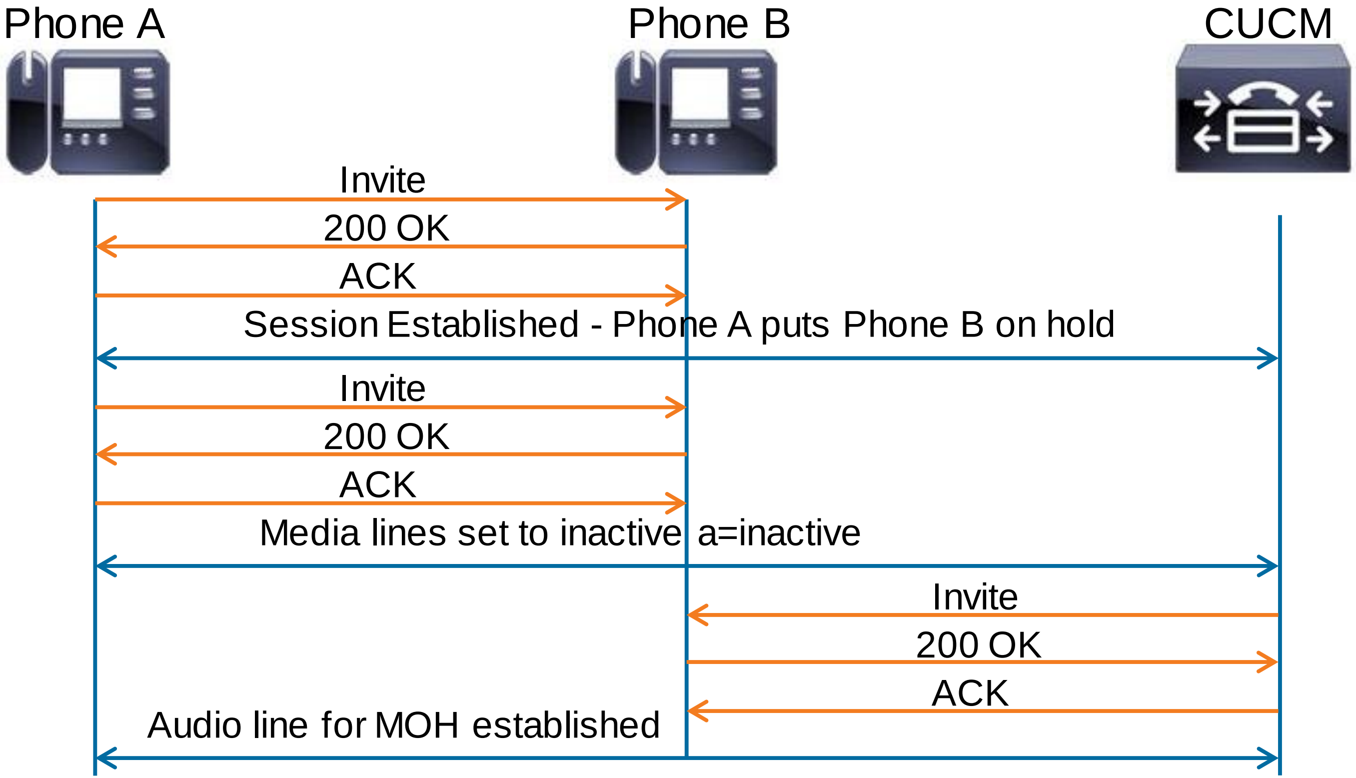
TranslatorX

- Create Filter



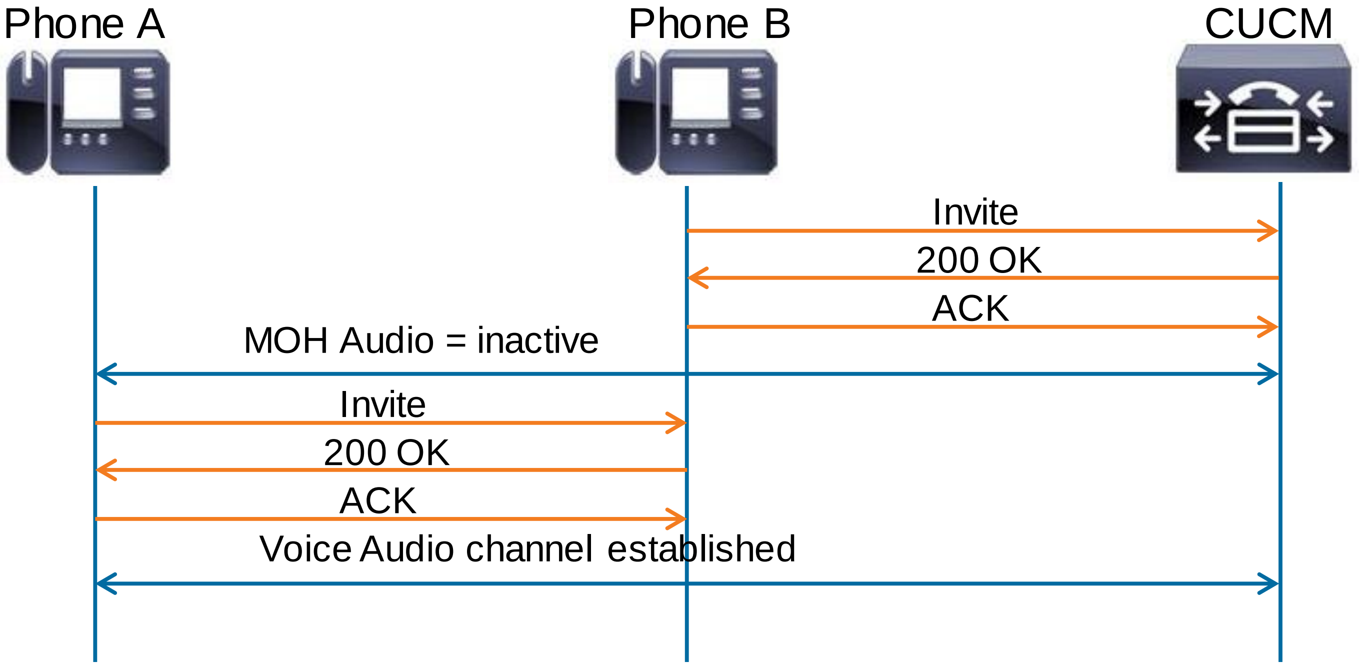
Troubleshoot a UC Application

MOH 1/2



Troubleshoot a UC Application

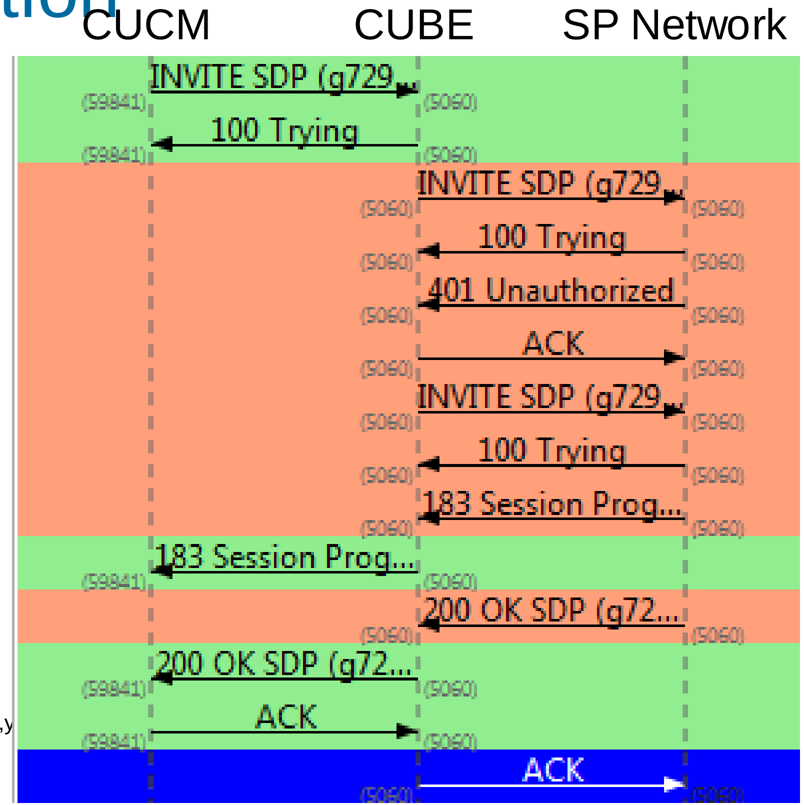
MOH 2/2



Troubleshoot a UC Application

Call Setup - Invite

INVITE sip:+61427092553@172.168.0.255:5060 SIP/2.0
v=0
o=CiscoSystemsCCM-SIP 150270 1 IN IP4 172.168.0.181
s=SIP Call
c=IN IP4 192.168.2.237
b=TIAS:1984000
b=AS:1984
t=0 0
m=audio 16576 RTP/AVP 18 101
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 16578 RTP/AVP 126 97
b=TIAS:1976000
a=rtpmap:126 H264/90000
a=imageattr:126 send [x=640,y=480] [x=640,y=360] [x=352,y=288] [x=176,y=144] recv [x=640,y
a=rtpmap:97 H264/90000
a=content:main
a=mid:2



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Troubleshoot a UC Application

Re-Invite for Hold

v=0

o=CiscoSystemsCCM-SIP 150270 2 IN IP4 172.168.0.181

s=SIP Call

c=IN IP4 0.0.0.0

b=TIAS:1984000

b=AS:1984

t=0 0

m=audio 16576 RTP/AVP 18 101

a=rtpmap:18 G729/8000

a=ptime:20

a=fmtp:18 annexb=no

a=inactive

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15

m=video 16578 RTP/AVP 126 97

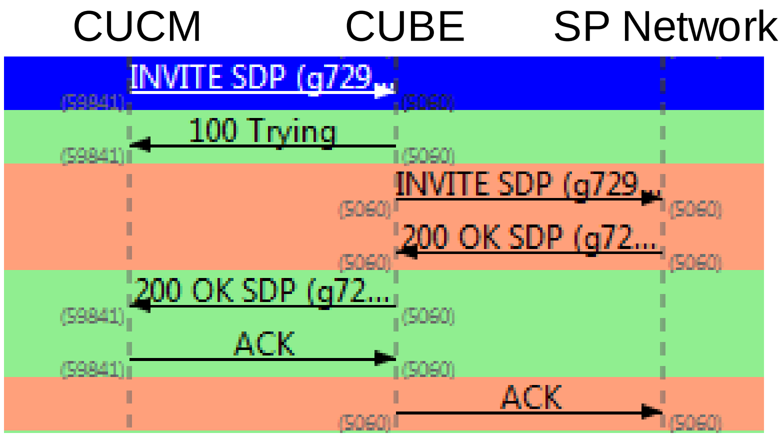
b=TIAS:1976000

a=rtpmap:126 H264/90000

a=rtpmap:97 H264/90000

a=content:main

a=inactive



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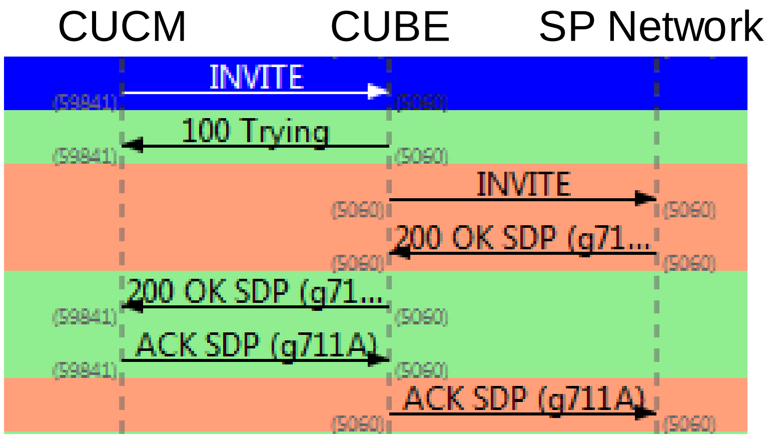
Troubleshoot a UC Application

MOH



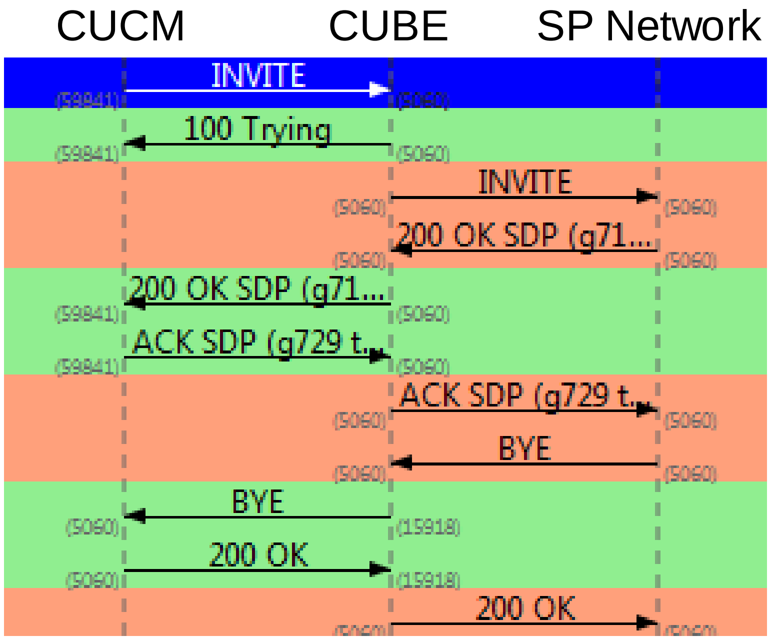
v=0
o=CiscoSystemsCCM-SIP 150270 3 IN IP4 172.168.0.181
s=SIP Call
t=0 0
m=audio 24846 RTP/AVP 8
c=IN IP4 172.168.0.180
a=X-cisco-media:umoh
a=rtpmap:8 PCMA/8000
a=ptime:20
m=video 0 RTP/AVP 126 97
c=IN IP4 192.168.2.237
b=TIAS:1976000
a=rtpmap:126 H264/90000
a=fmtp:126 profile-level-id=428014;packetization-mode=1;max-mbps=36000;max-fs=1200;max-rcmd-nalu-size=1300;level-asymmetry-allowed=1
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=428014;packetization-mode=0;max-mbps=36000;max-fs=1200;level-asymmetry-allowed=1
a=content:main
a=inactive
a=mid:2

Troubleshoot a UC Application

Resume – Late offer – 200OK

v=0
o=BroadWorks 26760612 5 IN IP4 203.52.0.164
s=-
c=IN IP4 203.52.0.164
t=0 0
a=media-release:hngl5rp9pujvivh05pvumib1joofrovgml23n3mavrbaeu76622cr8r9gh0uqijv
a=media-release-con-addr:d6v1k1hvmgvh1081o9j0
m=audio 18310 RTP/AVP 8 18 0 101
a=ptime:20
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/800
a=fmtp:101 0-15
a=fmtp:18 annexb=no
a=silenceSupp:off
m=video 0 RTP/AVP 0

m=audio 18310 RTP/AVP 8 18 0 101
m=video 0 RTP/AVP 0



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Troubleshoot a UC Application

ACK

v=0
o=CiscoSystemsCCM-SIP 150270 5 IN IP4 172.168.0.181
s=SIP Call
c=IN IP4 192.168.2.237
b=TIAS:8000
b=AS:8
t=0 0
m=audio 16576 RTP/AVP 18 101
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 0 RTP/SAVP 126 97
b=TIAS:1976000
a=rtpmap:126 H264/90000
a=rtpmap:97 H264/90000
a=content:main
a=inactive
a=mid:2

m=audio 18310 RTP/AVP 8 18 0 101
m=video 0 RTP/SAVP 126 97

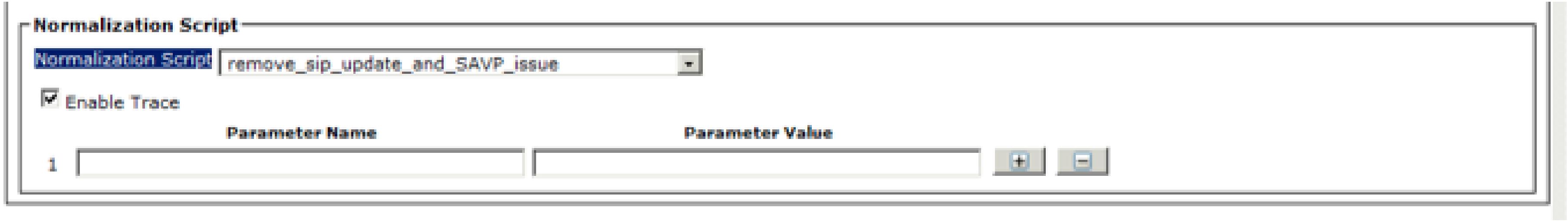
Troubleshoot a UC Application

Solution

- SIP Normalisation script that
  - Check if audio channel exist using AVP
  - If that his true, check if video channel exists with a 0 port using SAVP
  - If true, replace SAVP with AVP

Steps to Implement SIP Normalisation Script on CUCM

- Create new script
- Apply the script to the applicable trunk
- Reset the trunk




Q & A
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