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# Troubleshooting SIP

BRKUCC-3662

Richard Middelberg  
High Touch Engineer

#clmel

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# Agenda

- SIP Overview
- Case Studies



A long-exposure photograph of a city street at night. The foreground is dominated by vibrant, multi-colored light trails from moving vehicles, creating a sense of motion and energy. In the background, a modern pedestrian bridge with blue lighting spans across the street. Tall buildings with illuminated windows and storefronts line the street, and several flags are visible on the left side. The overall scene is a dynamic urban environment.

# 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

URI

SIP Version

CSeq: 100 INVITE

SIP Request

Contact: sip:0323456789@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 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: "Tyron [redacted]" <[redacted]@172.18.106.59>;tag=19210123ca7-45568313

To: <[redacted]>;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

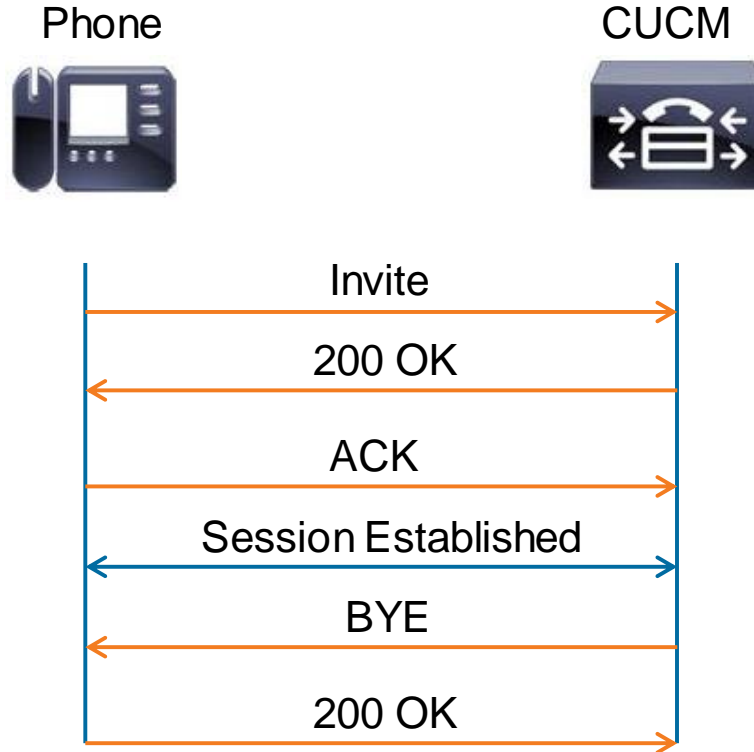
Server: Cisco-SIPGateway/IOS-15.2.2.T

**Reason: Q.850;cause=1**

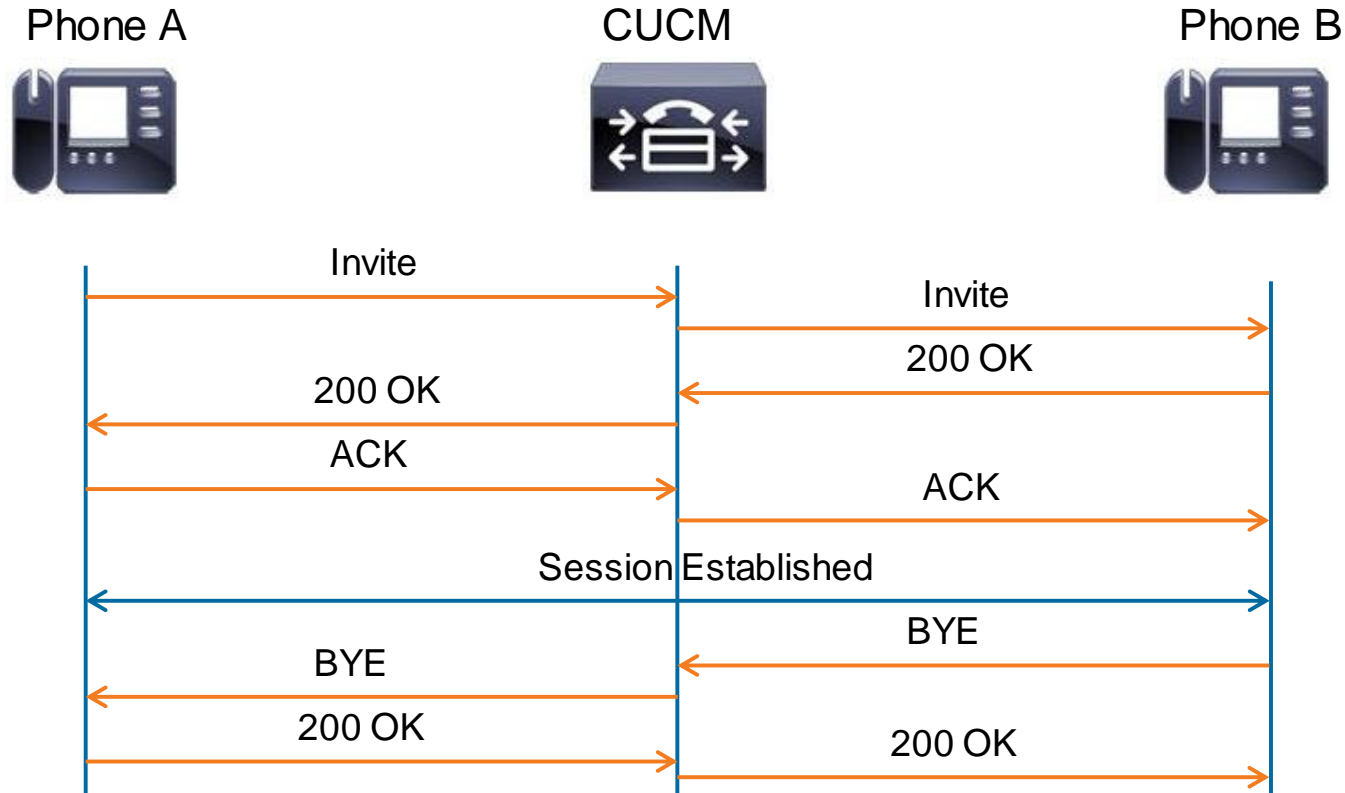
Content-Length: 0

**Reason code**

# Basic SIP Call Setup



# SIP Call Setup with a B2BUA





# SIP Call Setup with CUBE

Phone A



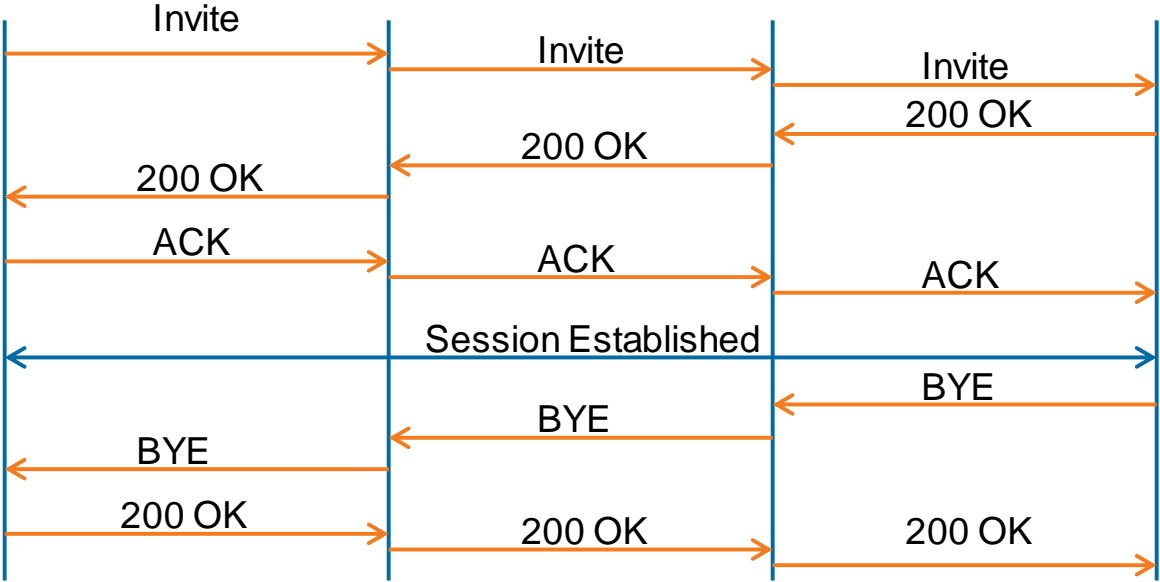
CUCM



CUBE Enterprise



CUBE SP



# 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

```
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
```

Session 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
```

Media Description



# 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

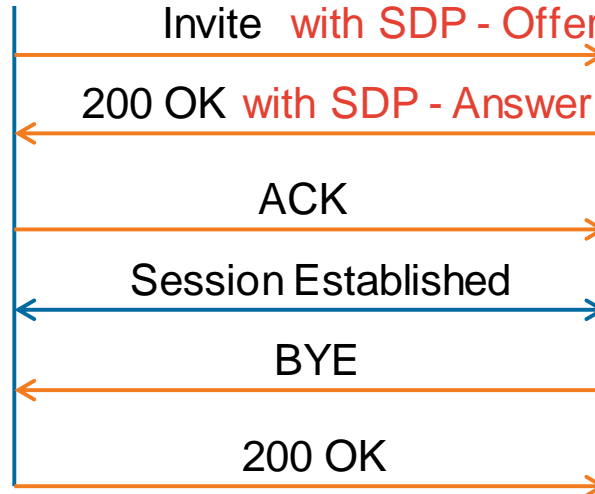
- Initiator of the call can send SDP offer in the INVITE – this is called an Early Offer (EO)
- Receiving endpoint can send the SDP offer in a response if the INVITE did not contain an offer – this is called a Delayed Offer (DO)
- For Early Offer, the answer is sent in a response (usually 200 OK).
-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

# Early Offer

Phone



CUCM

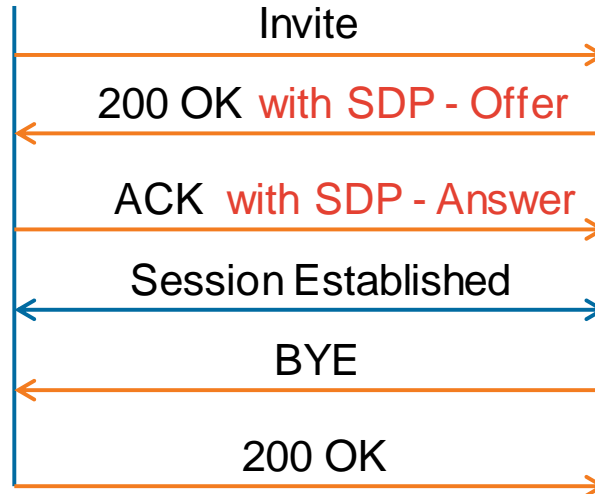


# Delayed Offer

Phone



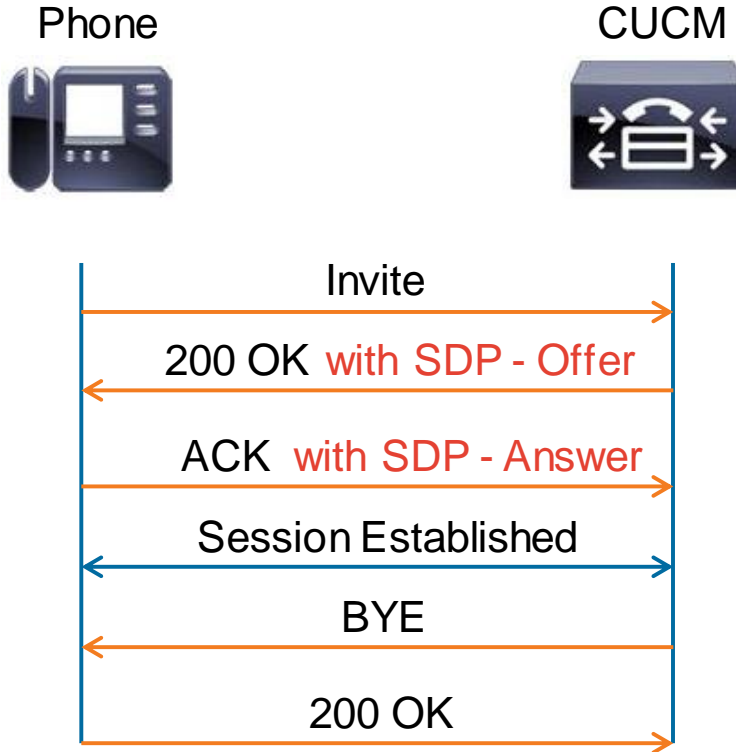
CUCM



# Early Media

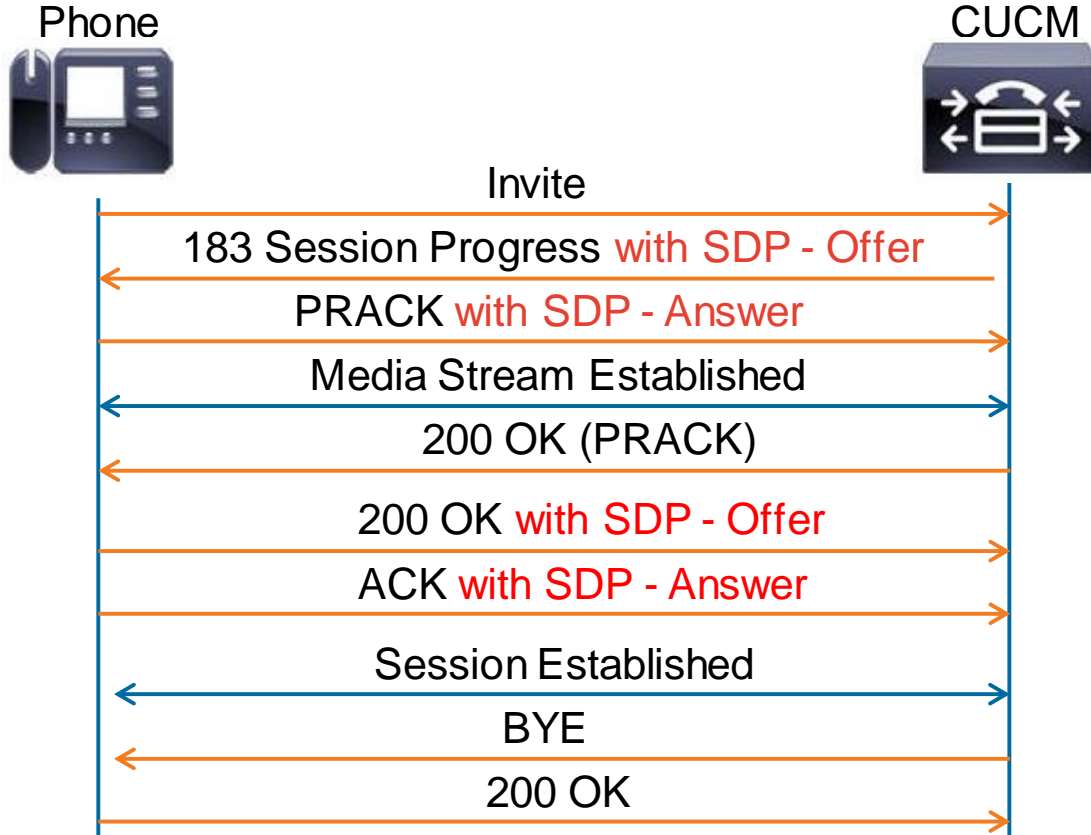
- Delayed offer calls do not set up media until the 200 OK (call is answered)
- If media is required prior to the call being connected, SIP has provisions for Early Media
- With Early Media on a Delayed Offer call, the offer comes from the terminating side in a provisional response (e.g. 183 Session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

# Delayed Offer





# Early Media



# 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:db40e44-0dfe-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
```

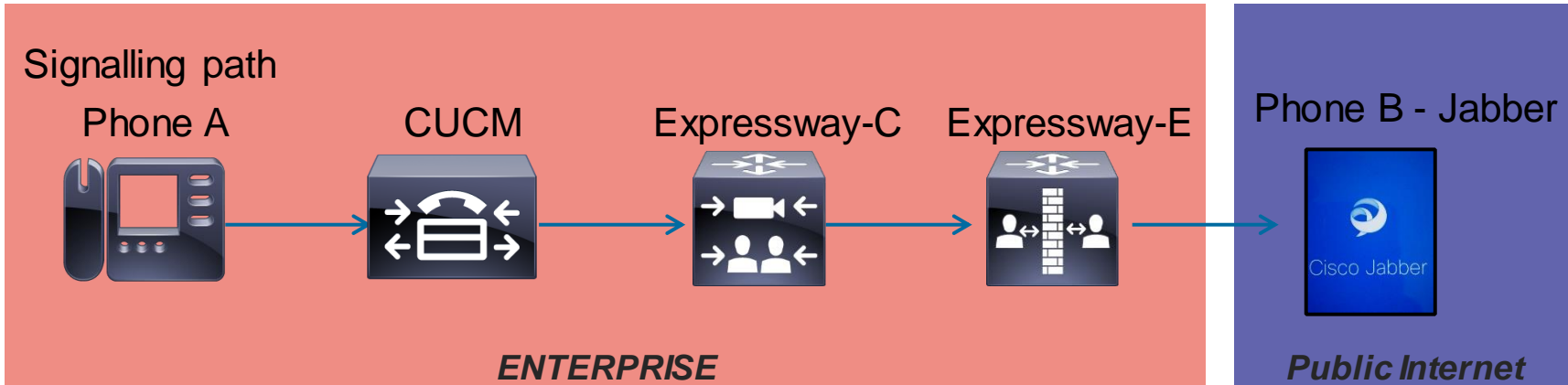


# Troubleshoot Expressway

# Troubleshoot Expressway

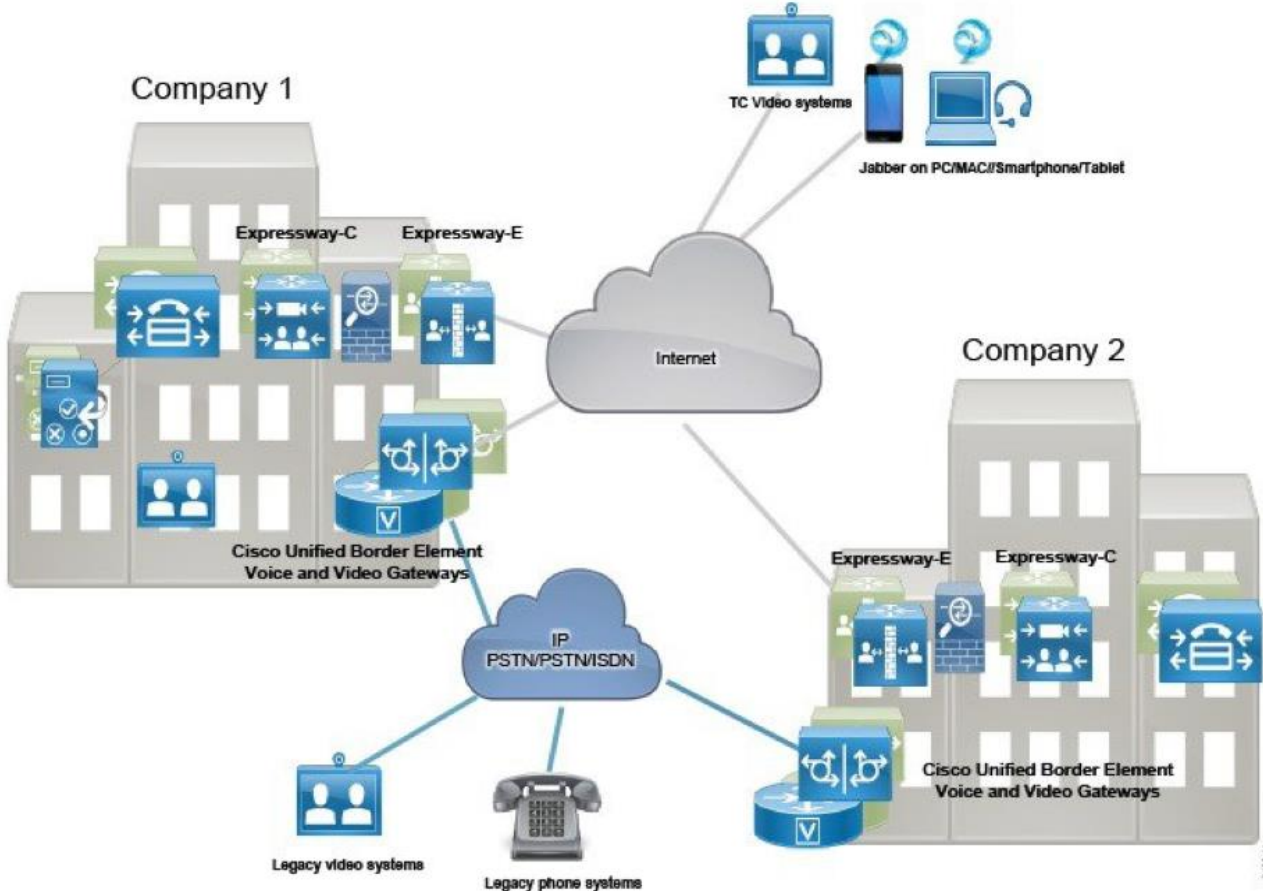
## Problem Description

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





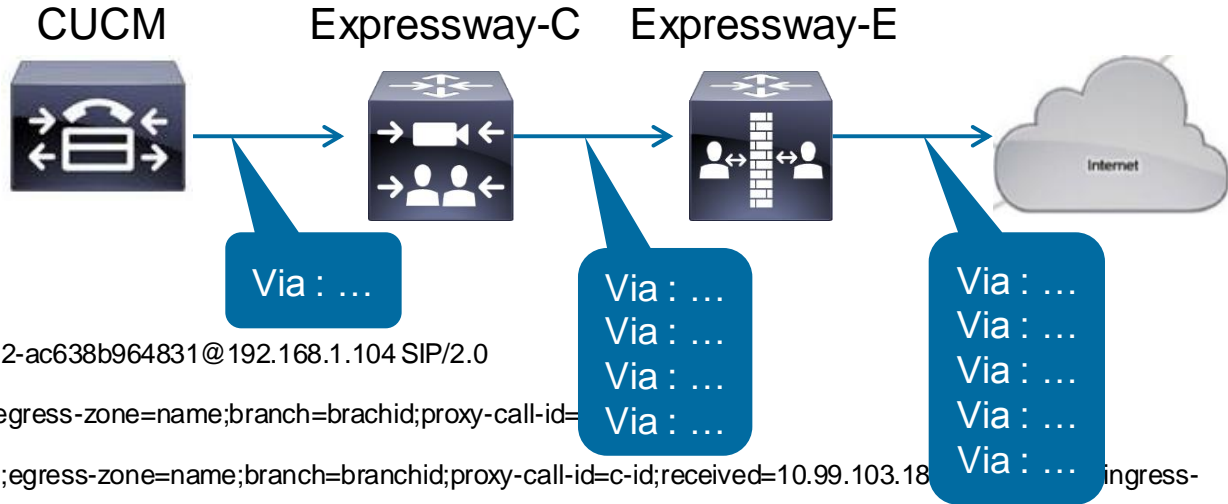
# Expressway MRA Overview





# 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=branchid;proxy-call-id=

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

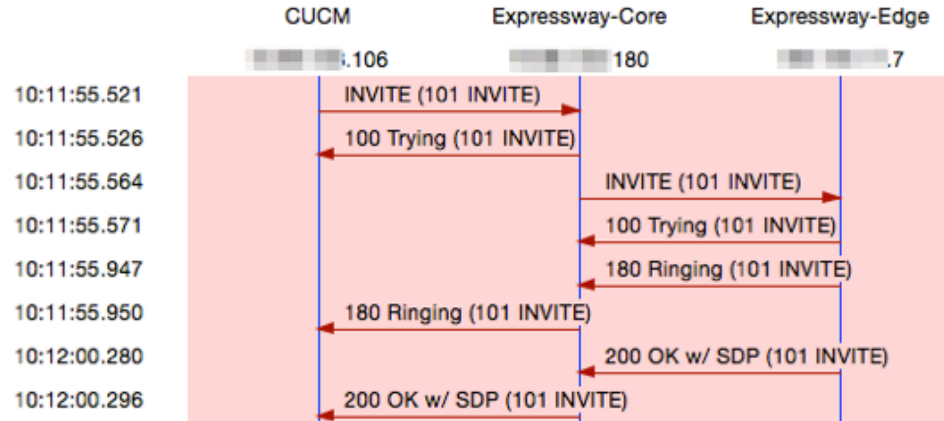
**Via:** SIP/2.0/TLS 10.99.103.180:5061;egress-zone=DefaultZone;branch=branchid;proxy-call-id=c-id;received=10.99.103.180;rport=25216

**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

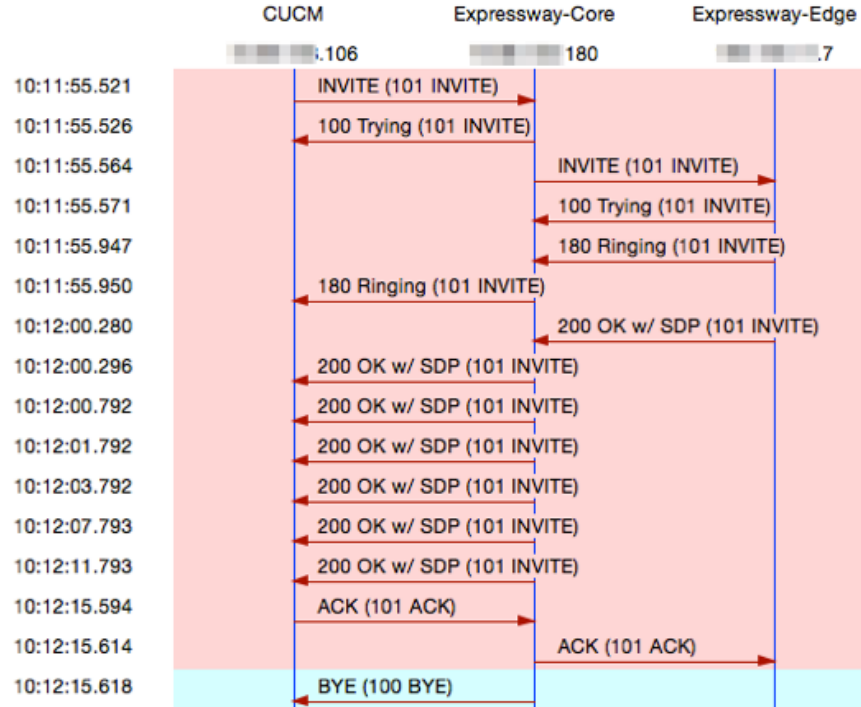
- Call setup



# 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

A long-exposure photograph of a city street at night. The foreground is dominated by vibrant, multi-colored light trails from moving vehicles, creating a sense of motion. In the middle ground, a modern pedestrian bridge with blue lighting spans across the street. The background features several tall buildings with lit windows and streetlights, creating a bright urban atmosphere.

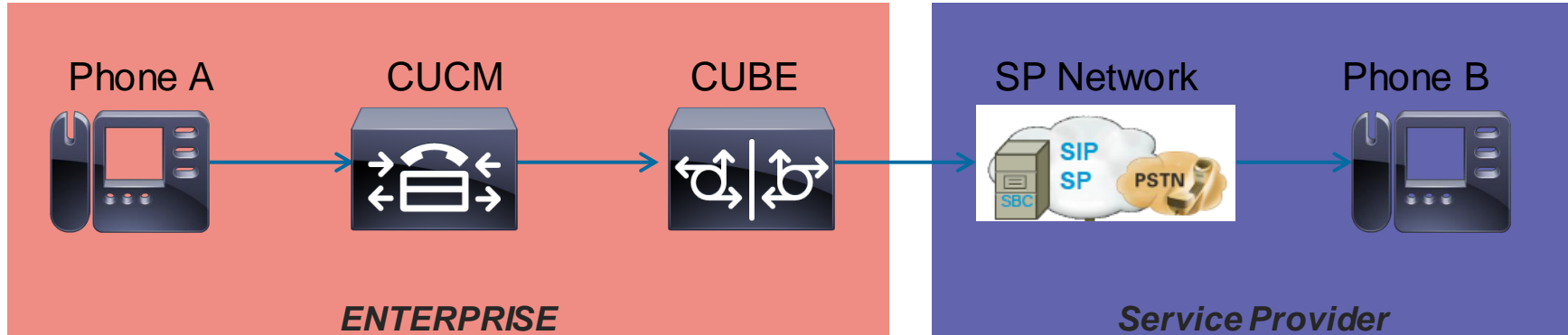
# Troubleshoot One Way Audio



# Troubleshoot One Way Audio

## Problem Description

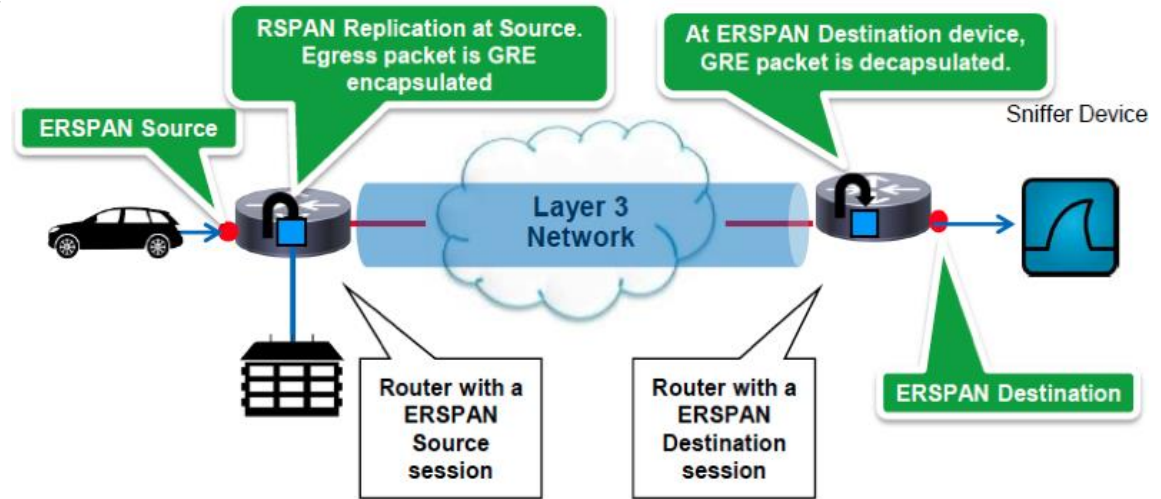
- Calls from some endpoints experience one way audio
- Call flow:



# 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

The screenshot displays the Wireshark interface with the following components:

- Filter:** sip
- Packet List:**

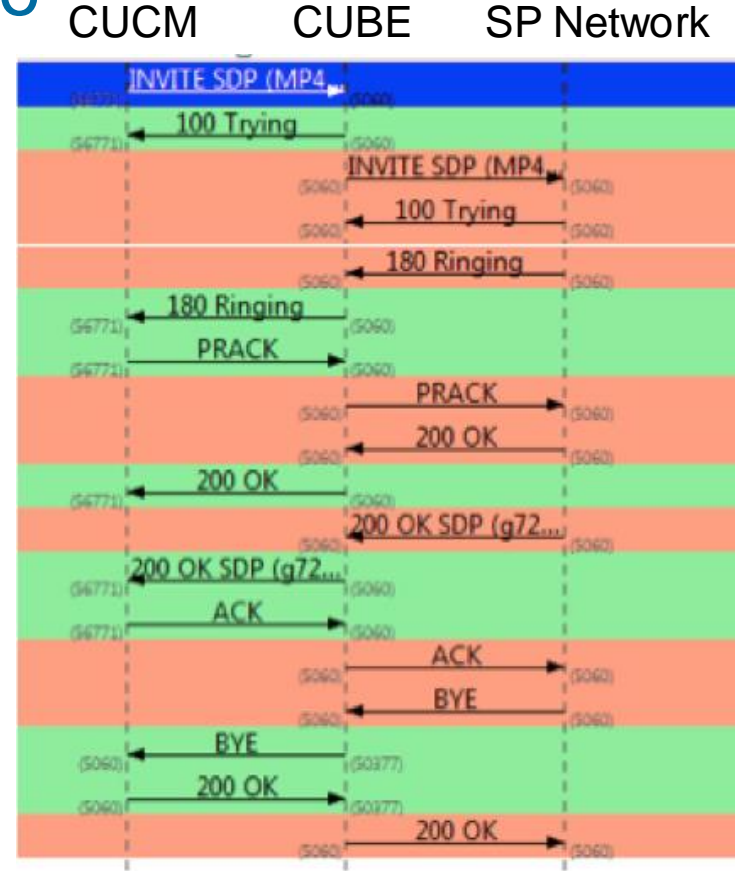
No.	Time	Source	Destination	Protocol	Length	Info
6	2014-06-12 16:09:11.5572900	58.	58.	SIP/SDF	436	Request: INVITE sip
7	2014-06-12 16:09:11.5590230	58.	58.	SIP	494	Status: 100 Trying
8	2014-06-12 16:09:11.5611250	58.	58.	SIP/SDF	1450	Request: INVITE sip
9	2014-06-12 16:09:11.5814690	203	203	SIP	428	Status: 100 Trying
10	2014-06-12 16:09:11.6161230	203	203	SIP	598	Status: 401 Unauthor
11	2014-06-12 16:09:11.6169920	58.	58.	SIP	568	Request: ACK sip:+6:
12	2014-06-12 16:09:11.6180100	58.	58.	SIP/SDF	1450	Request: INVITE sip
13	2014-06-12 16:09:11.6439640	203	203	SIP	428	Status: 100 Trying
14	2014-06-12 16:09:11.7672710	203	203	SIP	645	Status: 180 Ringing
15	2014-06-12 16:09:11.7680280	58.	58.	SIP	640	Status: 180 Ringing
16	2014-06-12 16:09:11.7869010	58.	58.	SIP	585	Request: PRACK sip:!
17	2014-06-12 16:09:11.7881050	58.	58.	SIP	869	Request: PRACK sip:-
18	2014-06-12 16:09:11.8390120	203	203	SIP	468	Status: 200 OK

The 'VoIP Calls' menu is open, showing options like ANSI, GSM, IAX2, ISUP Messages, LTE, MTP3, RTP, RTSP, SCTP, SMPP Operations, UCP Messages, VoIP Calls, H.225..., SIP..., and WAP-WSP... The 'VoIP Calls' option is highlighted.

# 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 [REDACTED].167
  Session Name (s): -
  Connection Information (c): IN IP4 [REDACTED].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 [REDACTED].167
  Media Attribute (a): label:11
  Media Attribute (a): rtpmap:100 H264/90000
  Media Attribute (a): fmp:100 profile-level-id=640028;packetization-mode=1
```

# Solution

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



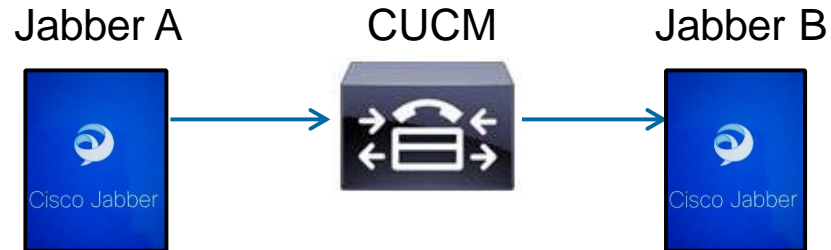
A long-exposure photograph of a city street at night. The foreground is dominated by vibrant, multi-colored light trails from moving vehicles, creating a sense of motion and energy. In the background, a modern pedestrian bridge with blue lighting spans across the street. Tall buildings with illuminated windows and signs are visible, along with several flags on poles to the left. The overall scene is a dynamic urban environment.

# 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;level-asymmetry=allowed
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/ccblId=0/scblId=0/handleSDPOfferInd:ipAddrMode IpAddrMode_v4
|3,100,68,804909.1^*^*
12:37:19.664 //SIP/SIPCdpc(3,74,1103501)/ci=62953946/ccblId=12275384/scblId=0/sendSIPMediaToUpdateCc:
Secure status=1, mSrtpPresent=03,100,68,804909.1^*^*
12:37:19.664 //SIP/SIPCdpc(3,74,1103501)/ci=62953946/ccblId=12275384/scblId=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/ccblId=12275384/scblId=0/handleSDPOfferInd: Sending offer to
12:37:19.664 |DET-SIPInterface-(804908)::BFCP is disabled 4908.1^*^*
12:37:19.664 |DET-SDPMsg-getBWFromSession: 1 0,4000
12:37:19.664 |DET-SDPMsg-getBWFromSession: session
12:37:19.664 |DET-SDPMsg-()::setTIASforDO, sessionBV
12:37:19.664 |DET-SDPMsg-()::setTIASforVideo, session
12:37:19.664 |DET-SDPMsg-()::setSessionBandwidthModifiers(bw=, action=0) - session(bitmask=0x1,tias=0,as=*
12:37:19.664 |DET-SIPInterface-(804908)::waitSDPResponse_SDPOfferInd, 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

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

### SIP Profile Configuration

 Save  Delete  Copy  Reset  Apply Config  Add New

Calling Line Identification Presentation\*  ▾

- Deliver Conference Bridge Identifier
- Early Offer support for voice and video calls (insert MTP if needed)
- Send send receive SDP in mid-call INVITE
- Allow Presentation Sharing using BFCP

A nighttime city street scene with a pedestrian bridge in the background. The foreground is dominated by long, colorful light trails from moving vehicles, creating a sense of motion and energy. The background shows modern buildings and streetlights.

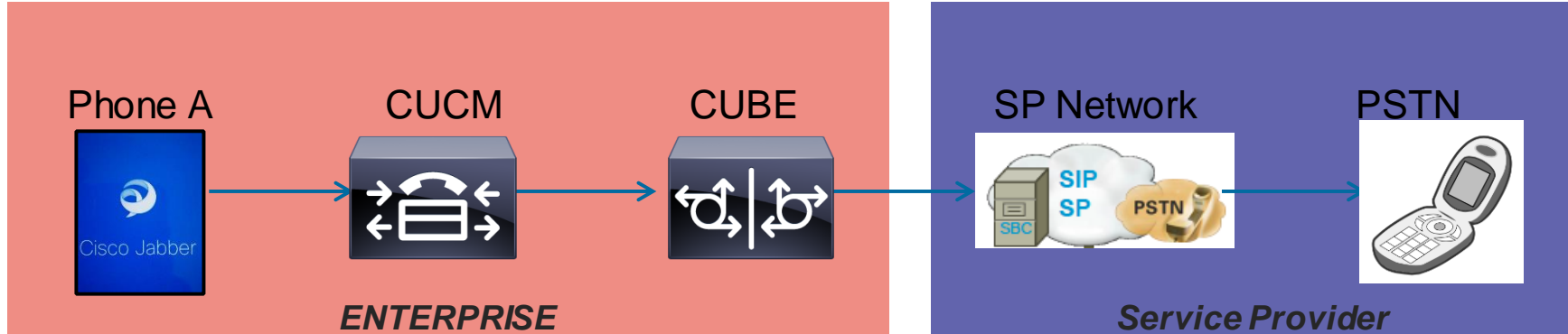
# Troubleshoot a UC Application



# 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

**Cisco Unified Serviceability**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified Serviceability Go

pgiralt About Logout

Alarm Trace Tools Snmp Help

**Trace Configuration**  
Configuration  
Troubleshooting Trace Settings

Related Links: SDL Configuration Go

**Status**  
Status : Ready

**Select Server, Service Group and Service**

Server\* 172.18.106.59 Go

Service Group\* CM Services Go

Service\* Cisco CallManager (Active) Go

Apply to All Nodes

Trace On

**Trace Filter Settings**

Select the Server

Select Service Group

Select the Service on Which Trace Needs to Be Enabled

# Unified CM Trace Configuration

The screenshot shows the 'Trace Configuration' page in the Unified CM administration console. At the top, there are 'Save' and 'Set Default' buttons. A callout box points to the 'Set Default' button with the text '1. Press Set Default'. Below this is the 'Status' section, which shows 'Ready'. The 'Select Server, Service Group and Service' section contains three dropdown menus: 'Server\*' (vnt-cm1b.cisco.com), 'Service Group\*' (CM Services), and 'Service\*' (Cisco CallMan...), each with a 'Go' button. A callout box points to these dropdowns with the text 'Updates All Servers in This Cluster with These Settings'. Below this is a 'Trace On' checkbox, which is checked. The 'Trace Filter Settings' section includes a 'Debug Trace Level' dropdown set to 'Detailed', with a callout box pointing to it that says '2. Set to Detailed'. Below the dropdown are four checked checkboxes: 'Enable H245 Message Trace', 'Enable DT-24+/DE-30+ Trace', 'Enable CDR Trace', and 'Enable Analog Trunk Trace'.

**Trace Configuration**

Save Set Default

**1. Press Set Default**

**Status:**  
Ready

**Select Server, Service Group and Service**

Server\* vnt-cm1b.cisco.com Go

Service Group\* CM Services

Service\* Cisco CallMan... Go

Apply to All Nodes

Trace On

**Trace Filter Settings**

Debug Trace Level Detailed

Enable H245 Message Trace

Enable DT-24+/DE-30+ Trace

Enable CDR Trace

Enable Analog Trunk Trace

**2. Set to Detailed**

**Updates All Servers in This Cluster with These Settings**



# Unified CM Trace Configuration

**Trace Filter Settings**

Debug Trace Level

<input checked="" type="checkbox"/> Enable H245 Message Trace	<input checked="" type="checkbox"/> Enable CDR Trace
<input checked="" type="checkbox"/> Enable DT-24+/DE-30+ Trace	<input checked="" type="checkbox"/> Enable Analog Trunk Trace
<input checked="" type="checkbox"/> Enable PRI Trace	<input checked="" type="checkbox"/> Enable All Phone Device Trace
<input checked="" type="checkbox"/> Enable ISDN Translation Trace	<input checked="" type="checkbox"/> Enable MTP Trace
<input checked="" type="checkbox"/> Enable H225 & Gatekeeper Trace	<input type="checkbox"/> Enable All GateWay Trace
<input type="checkbox"/> Enable Miscellaneous Trace	<input checked="" type="checkbox"/> Enable Forward & Miscellaneous Trace
<input checked="" type="checkbox"/> Enable Conference Bridge Trace	<input checked="" type="checkbox"/> Enable MGCP Trace
<input checked="" type="checkbox"/> Enable Music On Hold Trace	<input checked="" type="checkbox"/> Enable Media Resource Manager Trace
<input checked="" type="checkbox"/> Enable CM Real-Time Information Server Trace	<input checked="" type="checkbox"/> Enable SIP Call Processing Trace
<input checked="" type="checkbox"/> Enable SIP Stack Trace	<input type="checkbox"/> Enable SCCP Keep Alive Trace
<input checked="" type="checkbox"/> Enable Annunciator Trace	<input type="checkbox"/> Enable SpeedDial Trace
<input type="checkbox"/> Enable SoftKey Trace	<input type="checkbox"/> Enable SIP Keep Alive (REGISTER Refresh)
<input type="checkbox"/> Enable Route or Hunt List Trace	

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

Cisco Unified Communications Trace Translator

Filters Enabled    New Filter    Filters...    Clear Filters    0 Filters Configured    Call List...    Search    Clear

Timestamp	Node/Interface	Device IP	Directi...	Protocol	Message Name	TCP Handle/From Tag	Call Ref / ID
05/18/2014 19:05:19.692	172.18.106.59	not found	Out	SCCP	Register/Reject	(0017684)	
05/18/2014 19:05:19.696	172.18.106.59	not found	In	SCCP	Socket Broken	(0017684)	
05/18/2014 19:05:19.700	172.18.106.60	161.44.248.21	In	MGCP	Notify (NTFY)		
05/18/2014 19:05:19.700	172.18.106.60	161.44.248.21	Out	MGCP	200		
05/18/2014 19:05:19.704	172.18.106.59	10.116.101.41	In	SIP	NOTIFY	d0c78914113b42697...	e05e2700-37913caf-3...
05/18/2014 19:05:19.704	172.18.106.59	10.116.101.41	Out	SIP	200 OK	d0c78914113b42697...	e05e2700-37913caf-3...
05/18/2014 19:05:20.620	172.18.106.59	172.18.107.119	Out	SIP	PUBLISH	265344905	e0f6bd80-37913cb0-...
05/18/2014 19:05:20.626	172.18.106.59	172.18.107.119	In	SIP	200 OK	265344905	e0f6bd80-37913cb0-...
05/18/2014 19:05:22.119	172.18.106.59	172.18.106.41	In	MGCP	Notify (NTFY)		
05/18/2014 19:05:22.120	172.18.106.59	172.18.106.41	Out	MGCP	200		
05/18/2014 19:05:22.283	172.18.106.59	172.18.159.80	Out	SIP	REFER	1865199433	e227ea80-37913cb2-....
05/18/2014 19:05:22.307	172.18.106.59	172.18.159.80	In	SIP	202 Accepted	1865199433	e227ea80-37913cb2-....
05/18/2014 19:05:22.307	172.18.106.59	172.18.159.80	In	SIP	NOTIFY	d0c78914101642145...	e227ea80-37913cb2-....
05/18/2014 19:05:22.308	172.18.106.59	172.18.159.80	Out	SIP	200 OK	d0c78914101642145...	e227ea80-37913cb2-....
05/18/2014 19:05:23.763	172.18.106.59	172.18.107.119	Out	SIP	PUBLISH	1049368256	e2c08100-37913cb3-....
05/18/2014 19:05:23.769	172.18.106.59	172.18.107.119	In	SIP	200 OK	1049368256	e2c08100-37913cb3-....
05/18/2014 19:05:25.266	172.18.106.59	10.98.46.99	Out	SIP	REFER	1101253481	e3f1ae00-37913cb5-3...
05/18/2014 19:05:25.344	172.18.106.59	10.98.46.99	In	SIP	202 Accepted	1101253481	e3f1ae00-37913cb5-3...
05/18/2014 19:05:25.410	172.18.106.59	10.98.46.99	In	SIP	NOTIFY	d0c7891414060dbf32...	e3f1ae00-37913cb5-3...

01467685.001 | 19:05:22.283 | AppInfo | SIP - wait\_sd|SIP:signal: outgoing SIP TCP message to 172.18.159.80 on port 51424 index 2554 [24406001,NET]

REFER sip:3c7278d8-023b-3043-e66d-bc171809185b@172.18.159.80;51424;transport=tls SIP/2.0

Via: SIP/2.0/TLS 172.18.106.59;5061;branch=a9hg4bK3e430e2353745b

From: <sip:89915660@172.18.106.59>;tag=1865199433

To: <sip:89915660@172.18.159.80>

Call-ID: e227ea80-37913cb2-3a1199-3b6a12ac@172.18.106.59

CSeq: 101 REFER

Max-Forwards: 70

Contact: <sip:89915660@172.18.106.59;5061;transport=tls>

User-Agent: Cisco-CDM10.5

Expires: 30

Refer-To: cid:1234567890@172.18.106.59

Content-Id: <1234567890@172.18.106.59>

Content-Type: application/x-cisco-remotecc-request+xml

Referred-By: <sip:89915660@172.18.106.59>

Content-Length: 550

Lines Processed: 1231185     SCCP     H.245     Exclude SCCP and MGCP Keepalives   

Msgs Processed: 40220     SIP     MGCP     Exclude SIP REGISTER     Exclude SIP OPTIONS   

Msgs Displayed: 4051     Q.931 / H.225     MGCP BH     Exclude SIP SUBSCRIBE / NOTIFY / PUBLISH

# Translatorx

## Call List Window

Call List

Search

Calling Number  Called Number  All Calls

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...	8062AAF1620811...	
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	89944262	(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		158E7180000100...
12/9/10 1:06:50 PM	+13102314939	89916498	89916498	(16) Normal Call CL...	(0) No error	809EEF168A1A11...	16EF9E80000100...
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...	006000000088C...
12/9/10 1:08:04 PM	89915659	+19194762612	+19194762612	(16) Normal Call CL...	(0) No error		469E6A680000100...
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...	

Calls     14

Show Call List...   C T

# Troubleshoot a UC Application

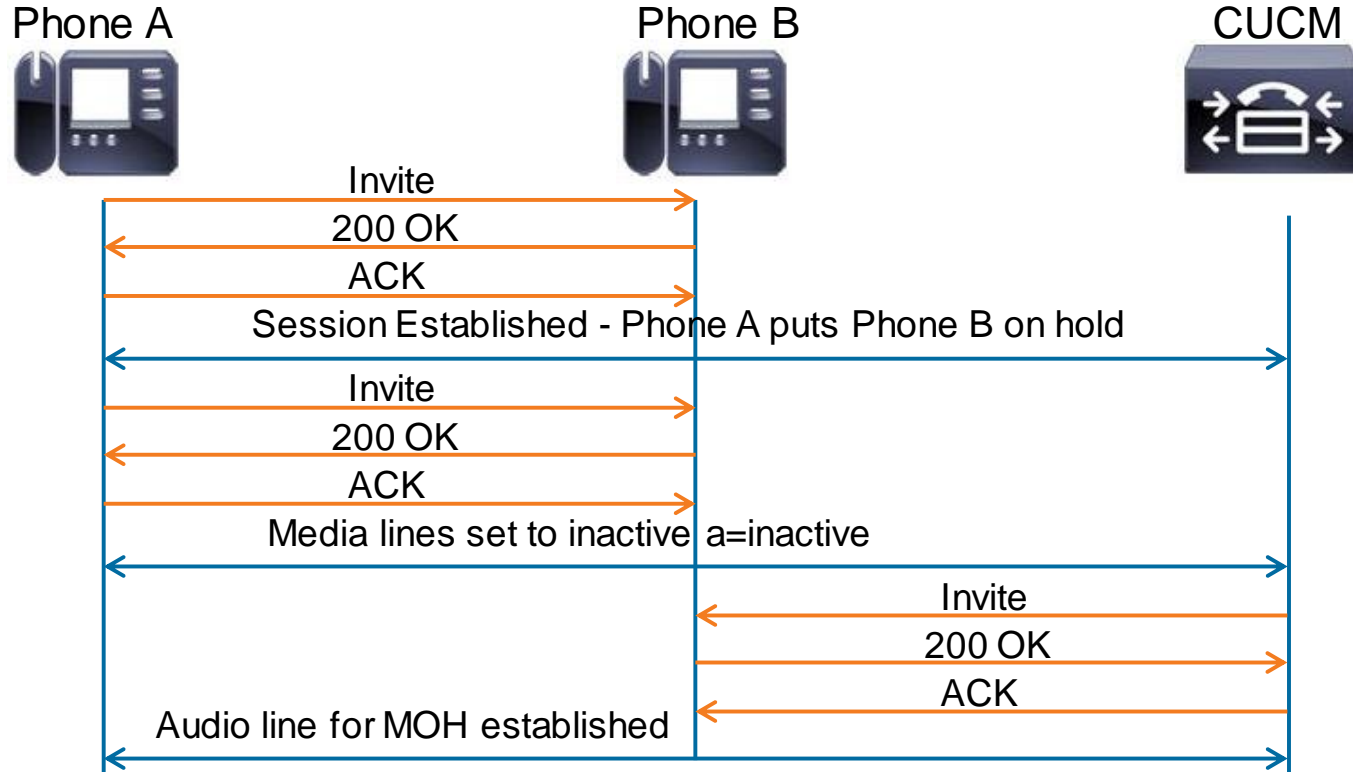
## TranslatorX

- Create Filter

The screenshot displays the 'Cisco Unified Communications Trace Translator' interface. The top toolbar includes a 'Filters Enabled' checkbox, a 'New Filter' button (highlighted with a red arrow), a 'Filters...' button, a 'Clear Filters' button, a '0 Filters Configured' status indicator, and a 'Call List...' button. Below this is the 'Message Filters' configuration window, which contains various filter options such as Device IP, TCP Handle, From Tag, Correlation Tag, Timestamp, Search, Protocol, Message, Direction, Call ID, Call Ref, and Node ID. The 'Call ID' filter is checked and its value 'fcfbfb11-1ed70006-6bbc926c-...' is highlighted with a red circle. A red arrow points from this circle to the 'Add Filter' button. At the bottom of the window, there are 'Update Filter' and 'Add Filter' buttons. Below the configuration area is an 'Active Filters' table with columns for Device IP, Node/Interface, Direct..., Message, TCP Handle, Call Ref, From Tag, SIP Call ID, and Text. The table shows two active filters, with the second one having the same SIP Call ID as the selected filter. At the bottom right of the window are 'Clear All' and 'Remove Selected' buttons.

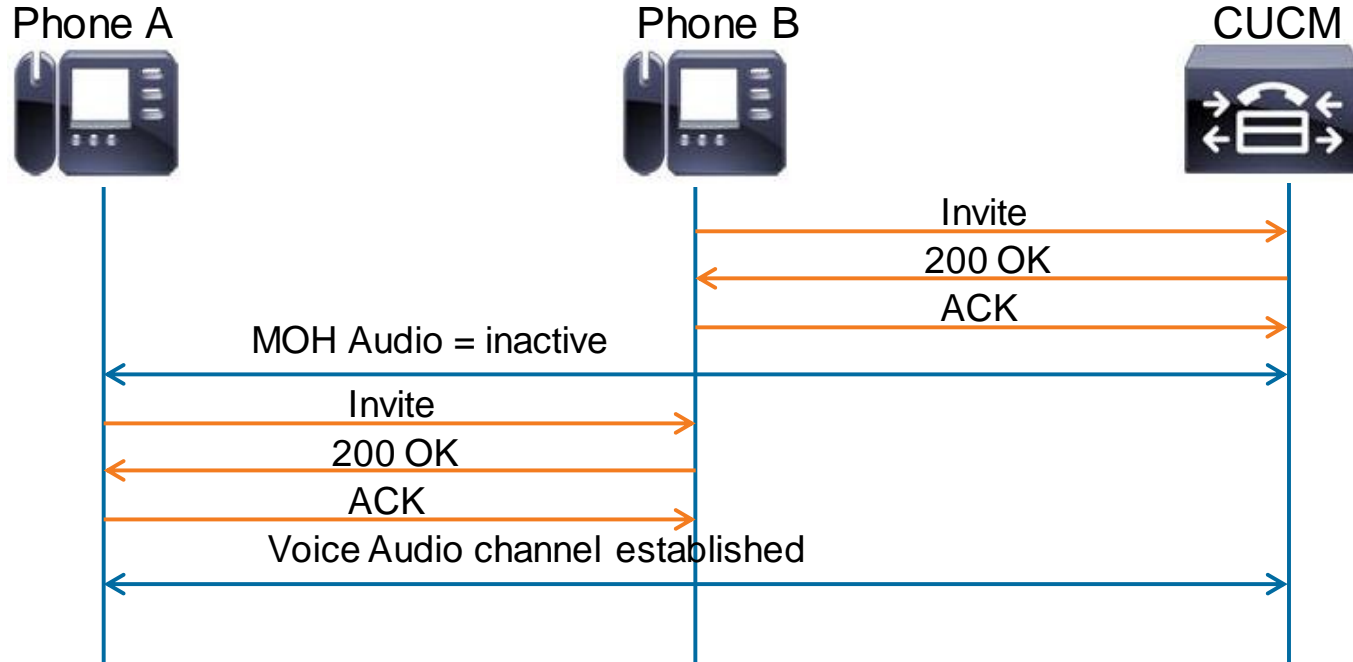
# Troubleshoot a UC Application

## MOH 1/2



# Troubleshoot a UC Application

## MOH 2/2

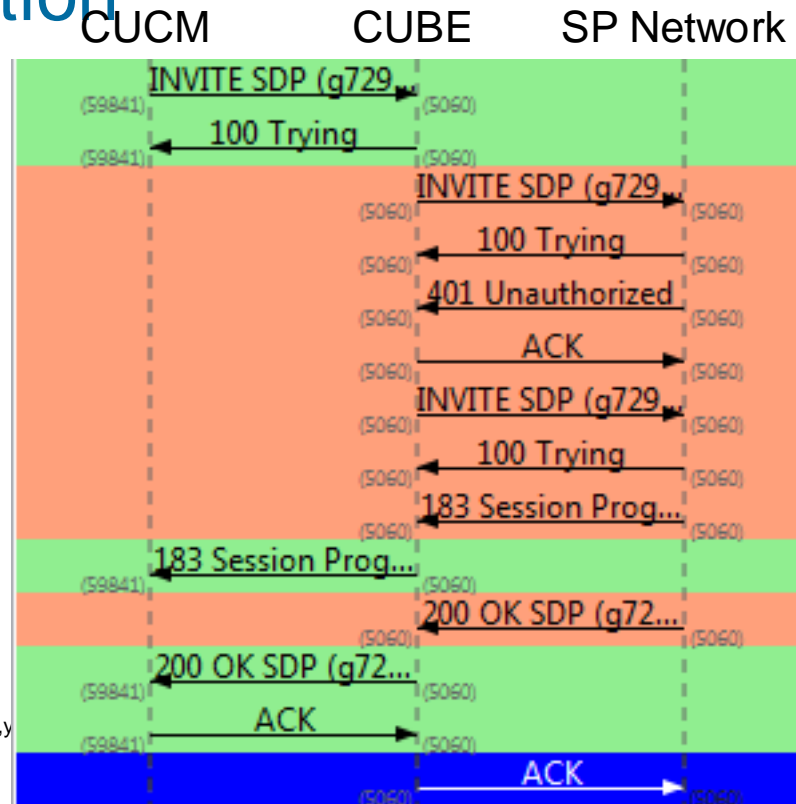




# Troubleshoot a UC Application

## Call Setup - Invite

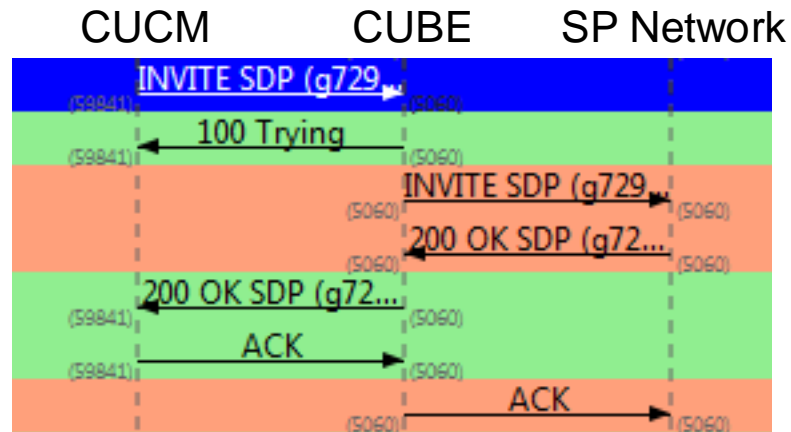
```
INVITE sip:+61427092553@172.168.0.255:5060 SIP/2.0
v=0
o=CiscoSystemsCCM-SIP 150270 1 IN IP4 172.168.0.181
s=SIP Call
c=IN IP4 192.168.2.237
b=TIAS:1984000
b=AS:1984
t=0 0
m=audio 16576 RTP/AVP 18 101
a=rtpmap:18 G729/8000
a=ptime:20
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 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
```



# Troubleshoot a UC Application

## Re-Invite for Hold

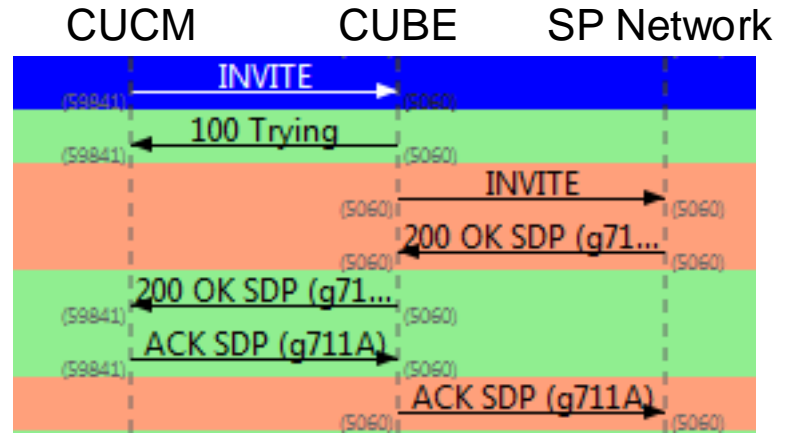
v=0  
o=CiscoSystemsCCM-SIP 150270 2 IN IP4 172.168.0.181  
s=SIP Call  
**c=IN IP4 0.0.0.0**  
b=TIAS:1984000  
b=AS:1984  
t=0 0  
m=audio 16576 RTP/AVP 18 101  
a=rtpmap:18 G729/8000  
a=ptime:20  
a=fmtp:18 annexb=no  
**a=inactive**  
a=rtpmap:101 telephone-event/8000  
a=fmtp:101 0-15  
m=video 16578 RTP/AVP 126 97  
b=TIAS:1976000  
a=rtpmap:126 H264/90000  
a=rtpmap:97 H264/90000  
a=content:main  
**a=inactive**



# 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-mps=36000;max-fs=1200;level-asymmetry-allowed=1
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=428014;packetization-mode=0;max-mps=36000;max-fs=1200;level-asymmetry-allowed=1
a=content:main
a=inactive
a=mid:2
```

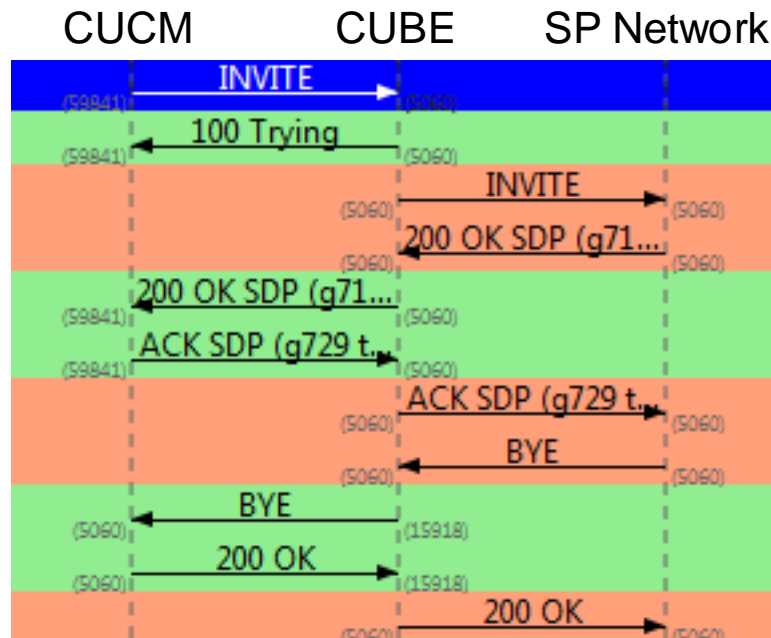


# Troubleshoot a UC Application

## 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:hngl5rp9pujvivh05pvumib1joofrovglm23n3mavrbau76622cr8r9gh0uqjiv
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/8000
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**



# 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
```

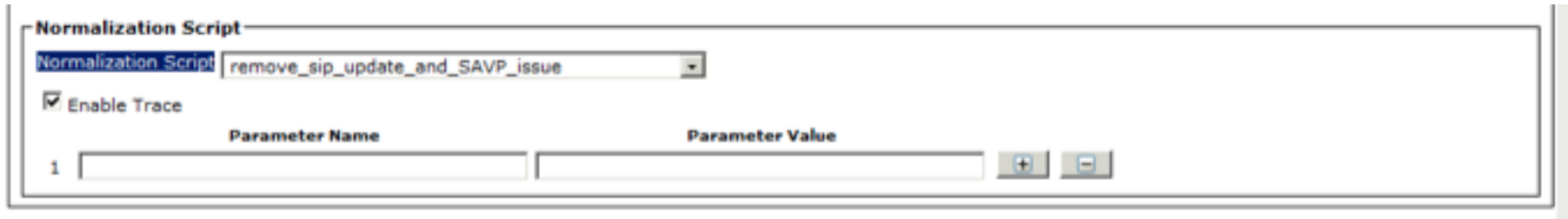
# Troubleshoot a UC Application

## 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



The screenshot shows the 'Normalization Script' configuration page in CUCM. At the top, there is a dropdown menu labeled 'Normalization Script' with the value 'remove\_sip\_update\_and\_SAVP\_issue' selected. Below this is a checkbox labeled 'Enable Trace' which is checked. At the bottom, there is a table with two columns: 'Parameter Name' and 'Parameter Value'. The first row of the table has the number '1' in the first column, followed by two empty input fields for the parameter name and value. To the right of these input fields are two buttons: a plus sign (+) and a minus sign (-).

	Parameter Name	Parameter Value	
1	<input type="text"/>	<input type="text"/>	<input type="button" value="+"/> <input type="button" value="-"/>





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