

# 기본 통화 녹음 구성 및 문제 해결

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## 소개

이 문서에서는 CUCM(Cisco Unified Communications Manager)에서 통화 녹음을 수행하는 기본 사항에 대해 설명합니다.

## 사전 요구 사항

### 요구 사항

Cisco에서는 서드파티 녹음 서버와 통합된 CUCM에 대해 알고 있는 것이 좋습니다.

### 사용되는 구성 요소

이 문서의 정보는 다음 소프트웨어 및 하드웨어 버전을 기반으로 합니다.

- CUCM

- Cisco IP(인터넷 프로토콜)
- 전화 통화 녹음 서버

이 문서의 정보는 특정 랩 환경의 디바이스를 토대로 작성되었습니다. 이 문서에 사용된 모든 디바이스는 초기화된(기본) 컨피그레이션으로 시작되었습니다. 현재 네트워크가 작동 중인 경우 모든 명령의 잠재적인 영향을 미리 숙지하시기 바랍니다.

## 배경 정보

또한 이 문서에서는 예상 미디어 흐름, SIP(Session Initiation Protocol) 및 SCCP(Skinny Client Control Protocol) 디바이스에 대한 예상 통화 흐름, 일반적인 유형의 통화 녹음 설정 실패의 예에 대해 설명합니다.

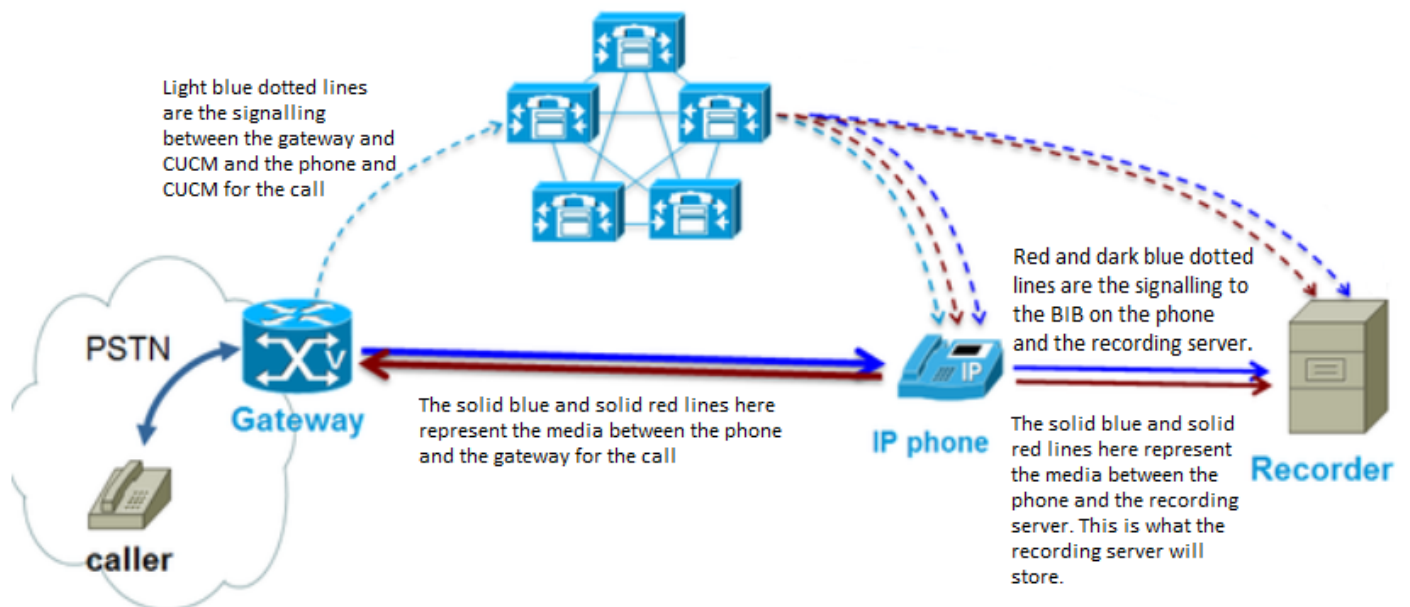
## 통화 녹음 유형

### 자동

자동 통화 녹음의 주요 요소는 다음과 같습니다.

- 오디오를 녹음 대상으로 포크하기 위해 IP 폰의 BIB(Built-In-Bridge)를 사용합니다.
- IP 전화기에서 전화를 걸거나 받을 때마다 시작됨
- CUCM과 녹음 대상 사이에 SIP 트렁크만 있으면 됩니다. 일부 녹음 공급업체에는 CTI(Computer Telephony Integration)가 필요합니다.
- 관리되는 네트워크 외부에 있는 전화기의 녹음을 허용하지 않습니다(녹음 서버에 RTP를 직접 보낼 수 있는 액세스 권한이 있어야 하며 BIB를 할당할 수 있는 Cisco IP 전화기여야 함).

이 다이어그램에서 실선은 예상 미디어 흐름을 나타내고 파선은 예상 시그널링 흐름을 나타냅니다.



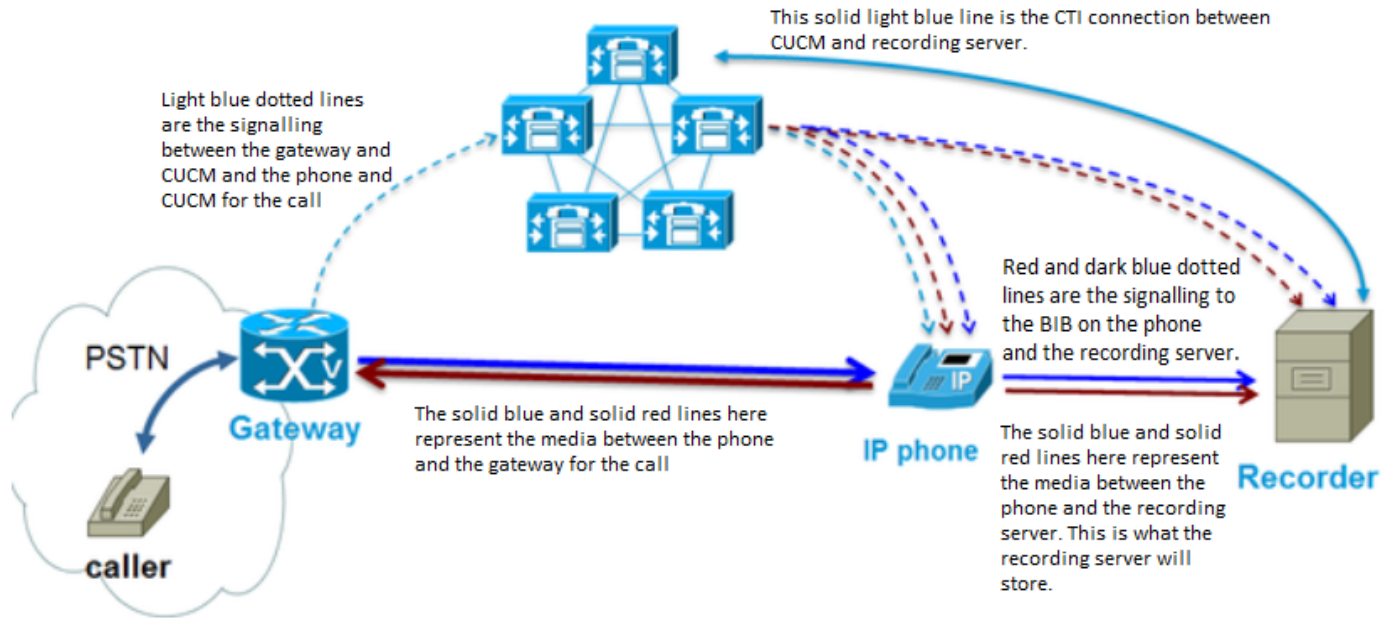
## 애플리케이션 호출됨

애플리케이션 호출 통화 녹음의 주요 요소는 다음과 같습니다.

- 오디오를 녹음 대상으로 포크하기 위해 IP 전화의 BIB를 사용합니다.

- 애플리케이션(레코더)이 시작되어야 한다고 지시할 때 시작됨
- 녹음 애플리케이션이 포함된 SIP 트렁크 및 CTI 필요
- CTI 애플리케이션 사용자는 기록해야 하는 엔드포인트에 대한 액세스 권한이 있어야 합니다
- 관리되는 네트워크 외부에 있는 전화기의 녹음을 허용하지 않습니다(RTP를 녹음 서버로 직접 보내기 위한 액세스 권한이 있어야 함).

여기 다이어그램에서 실선은 예상되는 미디어 흐름을 나타내고 점선은 예상되는 시그널링 흐름을 나타낸다. CUCM과 녹음 서버 사이의 실선은 CUCM과 애플리케이션 사이의 CTI 연결을 나타낸다.

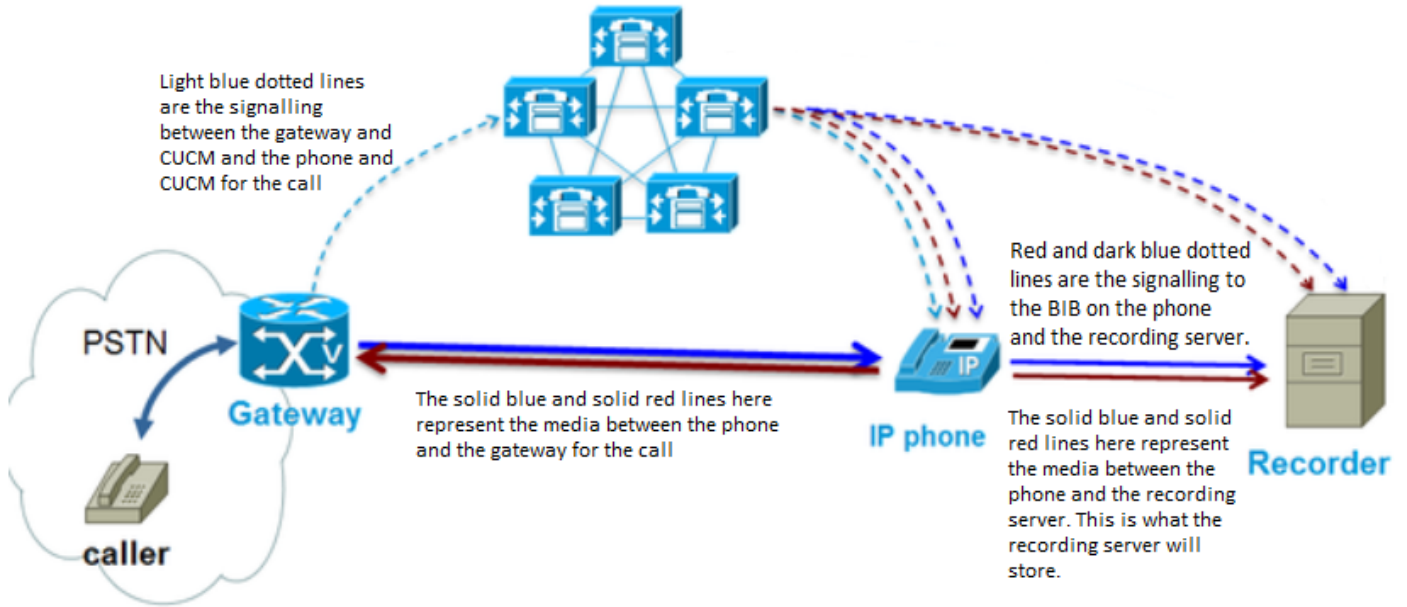


## 선택

선택적 통화 녹음의 주요 요소는 다음과 같습니다.

- 오디오를 녹음 대상으로 포크하기 위해 IP 전화의 BIB를 사용합니다.
- IP Phone 사용자가 IP Phone(CUCM 9.x+) 또는 [이 이미지](#)의 애플리케이션과 같은 애플리케이션에서 녹음 옵션을 선택할 때마다 **시작됩니다**
- 일반적으로 CUCM과 녹음 대상(녹음 애플리케이션 공급업체에 따라 다름) 간의 SIP 트렁크만 필요합니다.
- 관리되는 네트워크 외부에 있는 전화기의 녹음을 허용하지 않습니다(RTP를 녹음 서버로 직접 보내기 위한 액세스 권한이 있어야 함).

이 다이어그램에서 볼 수 있듯이 미디어 및 신호 경로는 자동 통화 녹음과 매우 유사합니다.

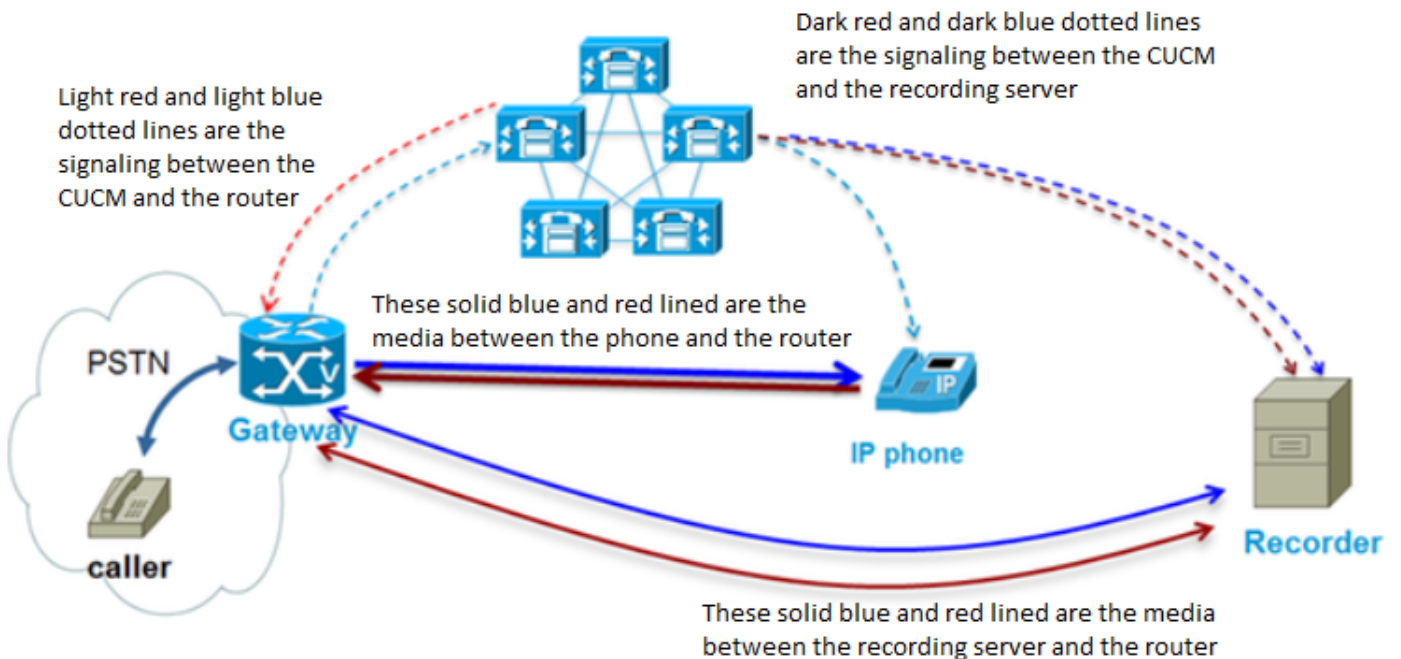


## 게이트웨이 기반

게이트웨이 기반 통화 녹음의 주요 요소는 다음과 같습니다.

- 음성 게이트웨이가 미디어를 녹음 대상으로 분기합니다.
- CUCM이 게이트웨이에 애플리케이션으로 등록됨
- CUCM은 HTTP를 사용하여 미디어를 녹음 대상으로 스트리밍하도록 게이트웨이(GW)에 지시합니다.
- CUCM은 SIP 트렁크를 통해 녹음 대상과 통합
- 관리되는 네트워크(예: 모바일 사용자)를 통과하는 통화 또는 BIB를 지원하지 않는 전화기의 녹음을 허용합니다.

여기 다이어그램에서 볼 수 있듯이 미디어 흐름은 다른 유형의 통화 녹음 방식과 상당히 다릅니다.



## SIP 전용 통합을 위한 자동 통화 녹음 컨피그레이션

이 섹션에서는 녹음 서버의 SIP 통합을 설정하는 방법에 대해 설명합니다.

## 녹음 대상에 SIP 트렁크 만들기

- Device(디바이스) > Trunk(트렁크)로 이동하고 Add New(새로 추가)를 선택합니다.
- 이미지에 표시된 대로 설정을 사용하여 SIP 트렁크를 생성합니다.

The screenshot shows the 'Trunk Configuration' interface. At the top, there is a 'Next' button with a green arrow. Below that is a 'Status' section with an information icon and the text 'Status: Ready'. The main section is 'Trunk Information', which contains three dropdown menus: 'Trunk Type\*' set to 'SIP Trunk', 'Device Protocol\*' set to 'SIP', and 'Trunk Service Type\*' set to 'None(Default)'. At the bottom, there is another 'Next' button.

- 적절한 디바이스 이름, 디바이스 풀, MRGL, SIP 트렁크 보안 프로파일 및 SIP 프로파일을 입력합니다
- 구성된 대상 주소는 녹음 애플리케이션 서버의 주소입니다.

## 녹음 프로필 만들기

- Device(디바이스) > Device Settings(디바이스 설정) > Recording Profile(녹음 프로필)로 이동합니다.
- 녹음 대상 주소는 그림과 같이 녹음 통화가 전송되는 곳입니다.

### Recording Profile Configuration

Save
 Delete
 Copy
 Add New

---

**Status**

Status: Ready

---

**Recording Profile Information**

Name\*

Recording Calling Search Space

Recording Destination Address\*

---

## 녹음 통화를 라우팅하기 위한 경로 패턴 만들기

- 이전 단계에서 구성한 녹음 대상 주소와 일치하는 경로 패턴을 생성합니다
- 이중화 SIP 트렁크를 구성하려는 경우 SIP 트렁크에서 직접 이동하지 않고 경로 목록을 가리킬 수 있습니다

**참고:** 이 경로 패턴에 할당된 파티션은 이미지에 표시된 대로 RecordingCallingSearch 공간과 연결되어야 합니다.

### Pattern Definition

Route Pattern\*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence\*

Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

Route Class\*

Gateway/Route List\*  [\(Edit\)](#)

Route Option  Route this pattern

## 전화 회선에 녹음 프로파일 할당

- 기존 내선 번호가 있는 이미 생성된 전화기에서 생성된 녹음 프로필을 할당합니다
- 이 위치에 통화 녹음 유형도 할당합니다.
- 이 예에서는 이미지에 표시된 대로 자동 녹음을 보여 줍니다.

Recording Option*	Automatic Call Recording Enabled
Recording Profile	Test Recording Profile
Recording Media Source*	Phone Preferred
Monitoring Calling Search Space	< None >

## 전화기 컨피그레이션 페이지에서 BIB를 On으로 설정하고 프라이버시를 Off로 설정

디바이스 컨피그레이션 페이지에서 Device Information(디바이스 정보)이라는 제목의 섹션으로 이동합니다. 이미지에 표시된 대로 Built In Bridge(기본 브리지)를 On(켜기)으로 설정하고 Privacy(프라이버시)를 Off(끄기)로 설정합니다.

Built In Bridge*	On
Privacy*	Off

## 다음을 확인합니다.

구성이 올바르게 작동하는지 확인하려면 이 섹션을 활용하십시오.

지정된 컨피그레이션을 사용하는 SCCP 및 SIP 전화에 대한 Call Manager 추적의 예상 동작은 다음과 같습니다. 이러한 예는 전화기 중 하나가 통화 녹음을 위해 설정된 동안 동일한 클러스터의 다른 전화기에 전화를 거는 전화기에 대한 것입니다.

**참고:** CUCM에서 수집할 로그는 CTIManager, CallManager, Event Viewer App/Sys이며, 일부 시나리오에서는 pcap가 필요할 수 있습니다.

**참고:** 전화기에서 수집할 로그는 콘솔 로그와 pcap입니다. 전화기에서 pcap를 받음과 동시에 녹음 서버에서 pcap를 받을 수 있습니다.

## SCCP

```

~~~~~
Normal CCM Traces for SCCP phone to SCCP phone with SIP Integrated Call Recording
~~~~~

### Calling phone places call

03796977.001 |20:21:08.055 |AppInfo |StationInit: (0000109) SoftKeyEvent softKeyEvent=1(Redial)
lineInstance=0 callReference=0.

### CUCM performs digit analysis against the dialed digits (dd="9110001")

03797017.001 |20:21:08.057 |AppInfo |Digit Analysis: star_DaReq: daReq.partitionSearchSpace(),
filteredPartitionSearchSpaceString(), partitionSearchSpaceString()
03797017.002 |20:21:08.057 |AppInfo |Digit Analysis: star_DaReq: Matching Legacy Numeric,
digits=9110001
03797017.003 |20:21:08.057 |AppInfo |Digit Analysis: getDaRes data&colon; daRes.ssType=[0]
Intercept DAMR.sstype=[0], TPcount=[0], DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]

```

```
03797017.004 |20:21:08.057 |AppInfo |Digit Analysis: getDaRes - Remote Destination [] isURI[1]
03797017.005 |20:21:08.057 |AppInfo |Digit analysis: patternUsage=2
03797017.006 |20:21:08.057 |AppInfo |Digit analysis: match(pi="2", fqcn="9110006",
cn="9110006",plv="5", pss="", TodFilteredPss="", dd="9110001",dac="0")
03797017.007 |20:21:08.057 |AppInfo |Digit analysis: analysis results
03797017.008 |20:21:08.057 |AppInfo ||PretransformCallingPartyNumber=9110006
|CallingPartyNumber=9110006
|DialingPartition=
|DialingPattern=9110001
|FullyQualifiedCalledPartyNumber=9110001
|DialingPatternRegularExpression=(9110001)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=9110001
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=9110001
|CollectedDigits=9110001
```

```
### CUCM determines call must stay on same node; go to LineControl
(PID=LineControl(2,100,174,137))
```

```
03797019.001 |20:21:08.058 |AppInfo |Digit analysis: wait_DmPidRes- Partition=[]
Pattern=[9110001] Where=[],cmDeviceType=[UserDevice], OutsideDialtone =[0], DeviceOverride=[0],
PID=LineControl(2,100,174,137),CI=[38960749],Sender=Cdcc(2,100,219,29)
```

```
### CUCM extends call to phone
```

```
03797036.003 |20:21:08.058 |AppInfo |StationD: (0000114) DEBUG whatToDo: line=1 calls=0
limit=4, busy=2. GCI=(2, 5033), cm_PL=(5, 0).
03797036.004 |20:21:08.058 |AppInfo |StationD: (0000114) DEBUG whatToDo: busy trigger not
hit... send to open appearance
03797036.005 |20:21:08.058 |AppInfo |preFilterCapCount =[11], preFilterCaps :: (Cap)= (25) (6)
(4) (2) (7) (8) (15) (16) (11) (12) (257) Filtering Caps due to Service Parameter Configuration
postFilterCapCount =[8], postFilterCaps :: (Cap)= (25) (4) (2) (15) (16) (11) (12) (257)
03797036.006 |20:21:08.058 |AppInfo |preFilterCapCount =[0], preFilterCaps :: (Cap)= Filtering
Caps due to Service Parameter Configuration postFilterCapCount =[0], postFilterCaps :: (Cap)=
03797036.007 |20:21:08.058 |Created | |
|StationCdpc(2,100,64,22) |StationD(2,100,63,114) |
|NumOfCurrentInstances: 2
03797036.008 |20:21:08.058 |AppInfo |StationD: (0000114) DEBUG- getLineRingSetting:
retVal=4.
03797036.009 |20:21:08.058 |AppInfo |StationD: (0000114) DEBUG- saveRinger for: ci=38960750,
line=1, mode=2, cm_precedence=5, callPhase=5.
03797036.010 |20:21:08.058 |AppInfo |StationD: (0000114) DEBUG- saveRinger: ci=38960750,
line=1, mode=2, cm_precedence=5, callPhase=5, modifier=0
03797036.011 |20:21:08.058 |AppInfo |StationD: (0000114) INFO sendCallAcceptReq: Try to
send StationLineCallAccept to cdpc=22 .
03797036.012 |20:21:08.058 |AppInfo |StationD: (0000114) playRinger for: ci=38960750.
03797036.013 |20:21:08.058 |AppInfo |StationD: (0000114) DEBUG- getLineRingSetting:
retVal=4.
03797036.014 |20:21:08.058 |AppInfo |StationD: (0000114) DEBUG- getLineRingSetting:
retVal=4.
03797036.015 |20:21:08.058 |AppInfo |StationD: (0000114) DEBUG- getLineRingSetting:
retVal=4.
```

```
### Called (recorded) phone goes off hook
```

```
03797089.001 |20:21:09.335 |AppInfo |StationD: (0000114) restart0_StationOffHook - INFO:
CI=38960750 on line=1, SPKMode=0, alwaysPrimeLine=0, alwaysUsePrimeLineForVM=0, fid=0,
offHookTrigger=0.
```



### CUCM Tells the calling phone to open the logical channel

03797153.001 |20:21:09.337 |AppInfo |StationD: (0000109) SEP0018195AA209 ,  
star\_MediaExchangeAgenaOpenLogicalChannel packetSize=20, codec=4, ci=38960749

### CUCM Tells the called (recorded party) phone to open the logical channel

03797156.001 |20:21:09.337 |AppInfo |StationD: (0000114) SEP001795BDD16B ,  
star\_MediaExchangeAgenaOpenLogicalChannel packetSize=20, codec=4, ci=38960750

### CUCM Tells the calling phone to open the receive channel

03797164.002 |20:21:09.337 |AppInfo |StationD: (0000109) OpenReceiveChannel  
conferenceID=38960749 passThruPartyID=33554450 millisecondPacketSize=20  
compressionType=4(Media\_Payload\_G711UlAW64k) RFC2833PayloadType=0 qualifierIn=?  
sourceIpAddr=IpAddr.type:0 ipAddr:0x0e302021000000000000000000000000(10.48.32.33). myIP:  
IpAddr.type:0 ipv4Addr:0x0e30201c(10.48.32.28)

### CUCM Tells the called (recorded party) phone to open the receive channel

03797168.002 |20:21:09.337 |AppInfo |StationD: (0000114) OpenReceiveChannel  
conferenceID=38960750 passThruPartyID=33554451 millisecondPacketSize=20  
compressionType=4(Media\_Payload\_G711UlAW64k) RFC2833PayloadType=0 qualifierIn=?  
sourceIpAddr=IpAddr.type:0 ipAddr:0x0e30201c000000000000000000000000(10.48.32.28). myIP:  
IpAddr.type:0 ipv4Addr:0x0e302021(10.48.32.33)

### CUCM allocates BIB on called (recorded) phone

03797210.000 |20:21:09.338 |SdlSig |MrmAllocateUcbResourceReq |waiting  
|MediaResourceManager(2,100,138,1) |Cc(2,100,220,1)  
|2,100,14,8384.91^10.48.32.33^SEP001795BDD16B |[R:N-H:0,N:1,L:0,V:0,Z:0,D:0] CI=38960751  
SsType=33554461 SsKey=9 BridgeType=0 MRGLPkid= NumStream=1 Bib=89cdb152-4ef2-4d60-9e6b-  
ab8c77c22618 BibTgCi=38960750 FeatId=159 PL=5 PLDmn=0 DeviceCapability=0 NumVideoCapable=0  
requestDeviceType=0 requestDeviceLocale=64 forkingDevicePosition=2 playToneDir=3

### BiB places first call to recording destination address (cn is calling party which is the BiB  
cn="b00223908001" and it is dialing the recordingdestination dd="8675309")

03797269.001 |20:21:09.340 |AppInfo |Digit Analysis: star\_DaReq: daReq.partitionSearchSpace(),  
filteredPartitionSearchSpaceString(), partitionSearchSpaceString()  
03797269.002 |20:21:09.340 |AppInfo |Digit Analysis: star\_DaReq: Matching Legacy Numeric,  
digits=8675309  
03797269.003 |20:21:09.340 |AppInfo |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept  
DAMR.ssType=[0], TPcount=[0], DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]  
03797269.004 |20:21:09.340 |AppInfo |Digit Analysis: getDaRes - Remote Destination [8675309]  
isURI[0]  
03797269.005 |20:21:09.340 |AppInfo |CMUtility routeCallThroughCTIRD: no matching  
RemDestDynamic record exists for remdest [8675309]  
03797269.006 |20:21:09.340 |AppInfo |DbMobility: getMatchedRemDest starts: cnumber = 8675309  
03797269.007 |20:21:09.340 |AppInfo |DbMobility: getMatchedRemDest: full match case  
03797269.008 |20:21:09.340 |AppInfo |DbMobility SelectByDestination: no matching RemDestDynamic  
record exists for remdest [8675309]  
03797269.009 |20:21:09.340 |AppInfo |DbMobility: can't find remdest 8675309 in map  
03797269.010 |20:21:09.340 |AppInfo |Digit analysis: patternUsage=5  
03797269.011 |20:21:09.340 |AppInfo |Digit analysis: match(pi="1", fqcn="",  
cn="b00223908001", plv="5", pss="E911\_PT:Phones\_PT:EMERGENCY\_PT:INTERNAL\_PT:INFORMACAST\_PT",  
TodFilteredPss="E911\_PT:Phones\_PT:EMERGENCY\_PT:INTERNAL\_PT:INFORMACAST\_PT",  
dd="8675309", dac="0")  
03797269.012 |20:21:09.340 |AppInfo |Digit analysis: analysis results  
03797269.013 |20:21:09.340 |AppInfo ||PretransformCallingPartyNumber=b00223908001  
|CallingPartyNumber=b00223908001  
|DialingPartition=  
|DialingPattern=8675309

|FullyQualifiedCalledPartyNumber=8675309  
|DialingPatternRegularExpression=(8675309)  
|DialingWhere=  
|PatternType=Enterprise  
|PotentialMatches=NoPotentialMatchesExist  
|DialingSdlProcessId=(0,0,0)  
|PretransformDigitString=8675309  
|PretransformTagsList=SUBSCRIBER  
|PretransformPositionalMatchList=8675309  
|CollectedDigits=8675309

### CUCM sends INVITE #1 to configured recording server (10.48.32.170)

03797320.001 |20:21:09.343 |AppInfo |//SIP/SIPUdp/wait\_SdlSPISignal: Outgoing SIP UDP message to 10.48.32.170:[5060]:

[212231,NET]

INVITE sip:8675309@10.48.32.170:5060 SIP/2.0

Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK204d520fedb3

From: <sip:9110001@10.48.32.90;x-nearend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=38960749;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=73601~713e2333-4032-45f1-blf5-e33cf471acec-38960754

To: <sip:8675309@10.48.32.170>

Date: Tue, 30 Sep 2014 00:21:09 GMT

Call-ID: abbb8e00-4291f775-204c-5a20300e@10.48.32.90

Supported: timer,resource-priority,replaces

Min-SE: 1800

User-Agent: Cisco-CUCM10.5

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: ;method="NOTIFY;Event=telephone-event;Duration=500"

Cisco-Guid: 2881195520-0000065536-0000000011-1512058894

Session-Expires: 1800

P-Asserted-Identity: <sip:9110001@10.48.32.90>

Remote-Party-ID: <sip:9110001@10.48.32.90>;party=calling;screen=yes;privacy=off

Contact: <sip:9110001@10.48.32.90:5060>;isFocus

Max-Forwards: 70

Content-Length: 0

### BiB places second call to recording destination address (cn is calling party which is the BiB cn="b00223908001" and it is dialing the recordingdestination dd="8675309")

Note that the BiB number stayed the same (b00223908001) and so did the recordingdestination number

03797367.010 |20:21:09.344 |AppInfo |Digit analysis: patternUsage=5

03797367.011 |20:21:09.344 |AppInfo |Digit analysis: match(pi="1", fqcn="",

cn="b00223908001",plv="5", pss="E911\_PT:Phones\_PT:EMERGENCY\_PT:INTERNAL\_PT:INFORMACAST\_PT",

TodFilteredPss="E911\_PT:Phones\_PT:EMERGENCY\_PT:INTERNAL\_PT:INFORMACAST\_PT",

dd="8675309",dac="0")

03797367.012 |20:21:09.344 |AppInfo |Digit analysis: analysis results

03797367.013 |20:21:09.344 |AppInfo ||PretransformCallingPartyNumber=b00223908001

|CallingPartyNumber=b00223908001

|DialingPartition=

|DialingPattern=8675309

|FullyQualifiedCalledPartyNumber=8675309

|DialingPatternRegularExpression=(8675309)

|DialingWhere=

|PatternType=Enterprise

|PotentialMatches=NoPotentialMatchesExist

|DialingSdlProcessId=(0,0,0)  
|PretransformDigitString=8675309  
|PretransformTagsList=SUBSCRIBER  
|PretransformPositionalMatchList=8675309  
|CollectedDigits=8675309

### CUCM receives 200 OK in response to INVITE #1

03797390.001 |20:21:09.345 |AppInfo |//SIP/SIPUdp/wait\_SdlDataInd: Incoming SIP UDP message size 737 from 10.48.32.170:[5060]:  
[212232,NET]  
SIP/2.0 200 OK  
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK204d520fedb3  
From: <sip:9110001@10.48.32.90;x-nearend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=38960749;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=73601~713e2333-4032-45f1-blf5-e33cf471lacec-38960754  
To: <sip:8675309@10.48.32.170>;tag=1  
Call-ID: abbb8e00-4291f775-204c-5a20300e@10.48.32.90  
CSeq: 101 INVITE  
Contact: <sip:10.48.32.170:5060;transport=udp>  
Content-Type: application/sdp  
Content-Length: 135

v=0  
o=user1 53655765 2353687637 IN IP4 10.48.32.170  
s=-  
c=IN IP4 10.48.32.170  
t=0 0  
m=audio 6000 RTP/AVP 0  
a=rtpmap:0 PCMU/8000

### CUCM sends INVITE #2 to recording server (10.48.32.170)

03797445.001 |20:21:09.348 |AppInfo |//SIP/SIPUdp/wait\_SdlSPISignal: Outgoing SIP UDP message to 10.48.32.170:[5060]:  
[212233,NET]  
INVITE sip:8675309@10.48.32.170:5060 SIP/2.0  
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK204e754eaeae  
From: <sip:9110001@10.48.32.90;x-farend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=38960749;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=73602~713e2333-4032-45f1-blf5-e33cf471lacec-38960757  
To: <sip:8675309@10.48.32.170>  
Date: Tue, 30 Sep 2014 00:21:09 GMT  
Call-ID: abbb8e00-4291f775-204d-5a20300e@10.48.32.90  
Supported: timer,resource-priority,replaces  
Min-SE: 1800  
User-Agent: Cisco-CUCM10.5  
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: ;method="NOTIFY;Event=telephone-event;Duration=500"  
Cisco-Guid: 2881195520-0000065536-0000000012-1512058894  
Session-Expires: 1800  
P-Asserted-Identity: <sip:9110001@10.48.32.90>  
Remote-Party-ID: <sip:9110001@10.48.32.90>;party-calling;screen=yes;privacy=off  
Contact: <sip:9110001@10.48.32.90:5060>;isFocus  
Max-Forwards: 70  
Content-Length: 0

### CUCM receives 200 OK in response to INVITE #2

03797498.001 |20:21:09.350 |AppInfo |//SIP/SIPUdp/wait\_SdlDataInd: Incoming SIP UDP message size 736 from 10.48.32.170:[5060]:

[212235,NET]

SIP/2.0 200 OK

Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK204e754eaeae

From: <sip:9110001@10.48.32.90;x-farend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=38960749;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=73602~713e2333-4032-45f1-blf5-e33cf471acec-38960757

To: <sip:8675309@10.48.32.170>;tag=2

Call-ID: abbb8e00-4291f775-204d-5a20300e@10.48.32.90

CSeq: 101 INVITE

Contact: <sip:10.48.32.170:5060;transport=udp>

Content-Type: application/sdp

Content-Length: 135

v=0

o=user1 53655765 2353687637 IN IP4 10.48.32.170

s=-

c=IN IP4 10.48.32.170

t=0 0

m=audio 6000 RTP/AVP 0

a=rtpmap:0 PCMU/8000

### CUCM sends outbound ACK in response to 200 OK #1

03797500.001 |20:21:09.351 |AppInfo |//SIP/SIPUdp/wait\_SdlSPISignal: Outgoing SIP UDP message to 10.48.32.170:[5060]:

[212236,NET]

ACK sip:10.48.32.170:5060;transport=UDP SIP/2.0

Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK204f50bef815

From: <sip:9110001@10.48.32.90;x-nearend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=38960749;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=73601~713e2333-4032-45f1-blf5-e33cf471acec-38960754

To: <sip:8675309@10.48.32.170>;tag=1

Date: Tue, 30 Sep 2014 00:21:09 GMT

Call-ID: abbb8e00-4291f775-204c-5a20300e@10.48.32.90

User-Agent: Cisco-CUCM10.5

Max-Forwards: 70

CSeq: 101 ACK

Allow-Events: presence, kpml

Content-Type: application/sdp

Content-Length: 254

v=0

o=CiscoSystemsCCM-SIP 73601 1 IN IP4 10.48.32.90

s=SIP Call

c=IN IP4 10.48.32.33

b=TIAS:64000

b=CT:64

b=AS:64

t=0 0

m=audio 4000 RTP/AVP 0 101

a=ptime:20

a=rtpmap:0 PCMU/8000

a=sendonly

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15

### CUCM sends startMediaTransmission to the called (recorded) phone telling the phone to send RTP to recording server (10.48.32.170)

03797479.001 |20:21:09.350 |AppInfo |StationD: (0000114) startMediaTransmission  
conferenceID=38960750 passThruPartyID=33554452 remoteIpAddress=IpAddr.type:0  
ipAddr:0x0e3020aa000000000000000000000000(10.48.32.170) remotePortNumber=6000  
milliSecondPacketSize=20 compressType=4(Media\_Payload\_G711Ulaw64k) RFC2833PayloadType=0  
qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e302021(10.48.32.33)

### CUCM sends startMediaTransmission #2 to the called (recorded) phone telling the phone to send RTP to recording server (10.48.32.170)

03797596.001 |20:21:09.354 |AppInfo |StationD: (0000114) startMediaTransmission  
conferenceID=38960750 passThruPartyID=33554453 remoteIpAddress=IpAddr.type:0  
ipAddr:0x0e3020aa000000000000000000000000(10.48.32.170) remotePortNumber=6000  
milliSecondPacketSize=20 compressType=4(Media\_Payload\_G711Ulaw64k) RFC2833PayloadType=0  
qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e302021(10.48.32.33)

### CUCM sends outbound ACK in response to 200 OK #2

03797615.001 |20:21:09.354 |AppInfo |//SIP/SIPUdp/wait\_SdlSPISignal: Outgoing SIP UDP message to 10.48.32.170:[5060]:

[212237,NET]

ACK sip:10.48.32.170:5060;transport=UDP SIP/2.0

Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK2050183495f1

From: <sip:9110001@10.48.32.90;x-farend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-

nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=38960749;x-

farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-

farendaddr=9110006>;tag=73602~713e2333-4032-45f1-blf5-e33cf47lacec-38960757

To: <sip:8675309@10.48.32.170>;tag=2

Date: Tue, 30 Sep 2014 00:21:09 GMT

Call-ID: abbb8e00-4291f775-204d-5a20300e@10.48.32.90

User-Agent: Cisco-CUCM10.5

Max-Forwards: 70

CSeq: 101 ACK

Allow-Events: presence, kpml

Content-Type: application/sdp

Content-Length: 254

v=0

o=CiscoSystemsCCM-SIP 73602 1 IN IP4 10.48.32.90

s=SIP Call

c=IN IP4 10.48.32.33

b=TIAS:64000

b=CT:64

b=AS:64

t=0 0

m=audio 4000 RTP/AVP 0 101

a=ptime:20

a=rtpmap:0 PCMU/8000

a=sendonly

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15

### Calling phone sends CUCM the ORC ACK

03797634.001 |20:21:09.385 |AppInfo |StationInit: (0000109) OpenReceiveChannelAck Status=0,  
IpAddr=IpAddr.type:0 ipAddr:0x0e30201c000000000000000000000000(10.48.32.28), Port=17996,  
PartyID=33554450

### CUCM sends startMediaTransmission to the called (recorded) phone telling the phone to send RTP to the calling phone (10.48.32.28)

03797642.001 |20:21:09.385 |AppInfo |StationD: (0000114) startMediaTransmission  
conferenceID=38960750 passThruPartyID=33554451 remoteIpAddress=IpAddr.type:0  
ipAddr:0x0e30201c000000000000000000000000(10.48.32.28) remotePortNumber=17996  
milliSecondPacketSize=20 compressType=4(Media\_Payload\_G711Ulaw64k) RFC2833PayloadType=0  
qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e302021(10.48.32.33)

### Called (recorded) phone sends CUCM the ORC ACK

03797643.001 |20:21:09.454 |AppInfo |StationInit: (0000114) OpenReceiveChannelAck Status=0,  
IpAddr=IpAddr.type:0 ipAddr:0x0e302021000000000000000000000000(10.48.32.33), Port=32588,  
PartyID=33554451

### CUCM sends startMediaTransmission to the calling phone telling the phone to send RTP to the  
called phone (10.48.32.33)

03797655.001 |20:21:09.454 |AppInfo |StationD: (0000109) startMediaTransmission  
conferenceID=  
38960749 passThruPartyID=33554450 remoteIpAddress=IpAddr.type:0  
ipAddr:0x0e302021000000000000000000000000(10.48.32.33) remotePortNumber=32588  
milliSecondPacketSize=20 compressType=4(Media\_Payload\_G711Ulaw64k) RFC2833PayloadType=0  
qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e30201c(10.48.32.28)

## SIP

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Normal CCM Traces for SIP phone to SIP phone with SIP Integrated Call Recording  
~~~~~

##### Calling phone places call

04241111.002 |11:27:41.232 |AppInfo |SIPtcp - wait\_SdlReadRsp: Incoming SIP TCP message from  
10.48.38.102 on port 50147 index 32 with 1946 bytes:  
[286938,NET]  
INVITE sip:1001@10.48.38.5;user=phone SIP/2.0  
Via: SIP/2.0/TCP 10.48.38.102:50147;branch=z9hG4bK598c2eb2  
From: "SJ User 1" <sip:1000@10.48.38.5>;tag=38ed18552a12296c00ff41e8-5fb7856e  
To: <sip:1001@10.48.38.5>  
Call-ID: 38ed1855-2a120006-78c34baf-1b81d864@10.48.38.102  
Max-Forwards: 70  
Session-ID: 1001532000105000a00038ed18552a12;remote=00000000000000000000000000000000  
Date: Tue, 27 Aug 2019 15:27:42 GMT  
CSeq: 101 INVITE  
User-Agent: Cisco-CP7861/12.1.1  
Contact: <sip:ab17ea6e-8072-927d-aad0-  
d10273906106@10.48.38.102:50147;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP38ED18552A12"  
Expires: 180  
Accept: application/sdp  
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO  
Remote-Party-ID: "SJ User 1" <sip:1000@10.48.38.5>;party=calling;id-  
type=subscriber;privacy=off;screen=yes  
Supported: replaces,join,sdp-anat,norefersub,resource-priority,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-7.0.0,X-cisco-xsi-8.5.1  
Allow-Events: kpml,dialog  
Recv-Info: conference  
Recv-Info: x-cisco-conference  
Content-Length: 687  
Content-Type: application/sdp  
Content-Disposition: session;handling=optional

v=0

o=Cisco-SIPUA 15384 0 IN IP4 10.48.38.102  
s=SIP Call  
b=AS:4064  
t=0 0  
m=audio 17904 RTP/AVP 114 9 113 115 0 8 116 18 101  
c=IN IP4 10.48.38.102  
b=TIAS:64000  
a=rtpmap:114 opus/48000/2  
a=fmtp:114 maxplaybackrate=16000;sprop-maxcapture=16000;maxaveragebitrate=64000;stereo=0;sprop-stereo=0;usedtx=0  
a=rtpmap:9 G722/8000  
a=rtpmap:113 AMR-WB/16000  
a=fmtp:113 octet-align=0;mode-change-capability=2  
a=rtpmap:115 AMR-WB/16000  
a=fmtp:115 octet-align=1;mode-change-capability=2  
a=rtpmap:0 PCMU/8000  
a=rtpmap:8 PCMA/8000  
a=rtpmap:116 iLBC/8000  
a=fmtp:116 mode=20  
a=rtpmap:18 G729/8000  
a=fmtp:18 annexb=yes  
a=rtpmap:101 telephone-event/8000  
a=fmtp:101 0-15  
a=sendrecv

##### CUCM performs digit analysis against the dialed digits (dd="1000")

04241138.007 |11:27:41.238 |AppInfo |Digit analysis: match(pi="2", fqcn="+14085251000", cn="1000", plv="5", pss="EMERGENCY\_PT:INTERNAL\_PT:SJ\_LOCAL\_PT:LD\_PT:GLOBALIZED\_PT", TodFilteredPss="EMERGENCY\_PT:INTERNAL\_PT:SJ\_LOCAL\_PT:LD\_PT:GLOBALIZED\_PT", dd="1001", dac="0")  
04241138.008 |11:27:41.238 |AppInfo |Digit analysis: analysis results  
04241138.009 |11:27:41.238 |AppInfo ||PretransformCallingPartyNumber=1000  
|CallingPartyNumber=1000  
|DialingPartition=INTERNAL\_PT  
|DialingPattern=1001  
|FullyQualifiedCalledPartyNumber=+14085251001  
|DialingPatternRegularExpression=(1001)  
|DialingWhere=  
|PatternType=Enterprise  
|PotentialMatches=NoPotentialMatchesExist  
|DialingSdlProcessId=(0,0,0)  
|PretransformDigitString=1001  
|PretransformTagsList=SUBSCRIBER  
|PretransformPositionalMatchList=1001  
|CollectedDigits=1001

##### CUCM determines call must stay on same node and go to LineControl (PID=LineControl(1,100,178,34))

04241140.001 |11:27:41.238 |AppInfo |Digit analysis: wait\_DmPidRes- Partition=[a067f454-fb26-2d1f-59da-a3f946a442c4] Pattern=[1001] Where=[],cmDeviceType=[UserDevice], OutsideDialtone =[0], DeviceOverride=[0], PID=LineControl(1,100,178,34),CI=[19301624],Sender=Cdcc(1,100,224,37)

##### CUCM sends outbound INVITE to called (recorded) phone

04241178.001 |11:27:41.242 |AppInfo |SIPTcp - wait\_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.107 on port 51902 index 52  
[286940,NET]  
INVITE sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32e829c48246  
From: "SJ User 1" <sip:1000@10.48.38.5>;tag=104952~e650e088-60ba-4195-8387-3dcc0127efdc-19301625  
To: <sip:1001@10.48.38.5>

Date: Tue, 27 Aug 2019 15:27:41 GMT  
Call-ID: 34241a00-d6514bed-327f-526300e@10.48.38.5  
Supported: timer,resource-priority,replaces  
Min-SE: 1800  
User-Agent: Cisco-CUCM11.5  
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-remotecallinfo>; security= Unknown; orientation= from; gci= 1-2029;  
isVoip; call-instance= 1  
Send-Info: conference, x-cisco-conference  
Alert-Info: <file://Bellcore-drl/>  
Session-ID: 1001532000105000a00038ed18552a12;remote=00000000000000000000000000000000  
Remote-Party-ID: "SJ User 1" <sip:1000@10.48.38.5;x-cisco-callback-  
number=1000>;party=calling;screen=yes;privacy=off  
Contact:  
<sip:1000@10.48.38.5:5060;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP38ED18552A12"  
Max-Forwards: 69  
Content-Length: 0

##### Called (recorded) phone returns 200 OK

04241233.002 |11:27:43.614 |AppInfo |SIPTcp - wait\_SdlReadRsp: Incoming SIP TCP message from  
10.48.38.107 on port 51902 index 52 with 1902 bytes:  
[286947,NET]  
SIP/2.0 200 OK  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32e829c48246  
From: "SJ User 1" <sip:1000@10.48.38.5>;tag=104952~e650e088-60ba-4195-8387-3dcc0127efdc-19301625  
To: <sip:1001@10.48.38.5>;tag=6c416a369525006f33cf6f38-43c38ad2  
Call-ID: 34241a00-d6514bed-327f-526300e@10.48.38.5  
Session-ID: 4313758700105000a0006c416a369525;remote=1001532000105000a00038ed18552a12  
Date: Tue, 27 Aug 2019 15:27:42 GMT  
CSeq: 101 INVITE  
Server: Cisco-CP7841/12.1.1  
Contact: <sip:91a43f66-ca58-9cd3-b0e5-  
588aa61a72bc@10.48.38.107:51902;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP6C416A369525"  
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO  
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=called;id-  
type=subscriber;privacy=off;screen=yes  
Supported: replaces,join,sdp-anat,norefersub,resource-priority,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-7.0.0,X-cisco-xsi-8.5.1  
Allow-Events: kpml,dialog  
Recv-Info: conference  
Recv-Info: x-cisco-conference  
Content-Length: 685  
Content-Type: application/sdp  
Content-Disposition: session;handling=optional

v=0  
o=Cisco-SIPUA 899 0 IN IP4 10.48.38.107  
s=SIP Call  
b=AS:4064  
t=0 0  
m=audio 20394 RTP/AVP 114 9 113 115 0 8 116 18 101  
c=IN IP4 10.48.38.107  
b=TIAS:64000  
a=rtpmap:114 opus/48000/2  
a=fmtp:114 maxplaybackrate=16000;sprop-  
maxcapture=16000;maxaveragebitrate=64000;stereo=0;sprop-stereo=0;usedtx=0  
a=rtpmap:9 G722/8000  
a=rtpmap:113 AMR-WB/16000



a=fmtp:113 octet-align=0;mode-change-capability=2  
a=rtpmap:115 AMR-WB/16000  
a=fmtp:115 octet-align=1;mode-change-capability=2  
a=rtpmap:0 PCMU/8000  
a=rtpmap:8 PCMA/8000  
a=rtpmap:116 iLBC/8000  
a=fmtp:116 mode=20  
a=rtpmap:18 G729/8000  
a=fmtp:18 annexb=yes  
a=rtpmap:101 telephone-event/8000  
a=fmtp:101 0-15  
a=sendrec

### CUCM sends ACK to called (recorded) phone telling the called phone to send media to the calling phone (10.48.32.28)

01314344.001 |11:18:48.652 |AppInfo |SIPTcp - wait\_SdlSPISignal: Outgoing SIP TCP message to 10.48.32.17 on port 50841 index 17  
[106320,NET]  
ACK sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@10.48.32.17:50841;transport=tcp SIP/2.0  
Via: SIP/2.0/TCP 10.48.32.90:5060;branch=z9hG4bK203c2831c118  
From: <sip:9110006@10.48.32.90>;tag=38244~713e2333-4032-45f1-b1f5-e33cf471acec-47601638  
To: <sip:9110011@10.48.32.90>;tag=b000b4d9e8cb0bba73e445ee-3cc7e650  
Date: Tue, 14 Oct 2014 15:18:44 GMT  
Call-ID: 6198e780-43d13ed4-203c-5a20300e@10.48.32.90  
User-Agent: Cisco-CUCM10.5  
Max-Forwards: 70  
CSeq: 101 ACK  
Allow-Events: presence  
Content-Type: application/sdp  
Content-Length: 243

v=0  
o=CiscoSystemsCCM-SIP 38244 1 IN IP4 10.48.32.90  
s=SIP Call  
c=IN IP4 10.48.32.28  
b=TIAS:64000  
b=CT:64  
b=AS:64  
t=0 0  
m=audio 17260 RTP/AVP 0 101  
aptime:20  
a=rtpmap:0 PCMU/8000  
a=rtpmap:101 telephone-event/8000  
a=fmtp:101 0-15### CUCM allocates BiB on called (recorded) phone

01314383.000 |11:18:48.675 |SdlSig |MrmAllocateUcbResourceReq |waiting  
|MediaResourceManager(2,100,138,1) |Cc(2,100,220,1)  
|2,100,14,20.16735^10.48.32.28^SEP0018195AA209 |[R:N-H:0,N:3,L:1,V:0,Z:0,D:0] CI=47601639  
SsType=33554461 SsKey=1 BridgeType=0 MRGLPkid= NumStream=1 Bib=c32d6714-48bd-43d7-b96f-91363aff3aa0 BibTgCi=47601638 FeatId=159 PL=5 PLDmn=0 DeviceCapability=0 NumVideoCapable=0 requestDeviceType=0 requestDeviceLocale=64 forkingDevicePosition=2 playToneDir=3

##### CUCM forwards the 200 OK to the calling phone

04241368.001 |11:27:43.624 |AppInfo |SIPTcp - wait\_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.102 on port 50147 index 32  
[286949,NET]  
SIP/2.0 200 OK  
Via: SIP/2.0/TCP 10.48.38.102:50147;branch=z9hG4bK598c2eb2  
From: "SJ User 1" <sip:1000@10.48.38.5>;tag=38ed18552a12296c00ff41e8-5fb7856e  
To: <sip:1001@10.48.38.5>;tag=104951~e650e088-60ba-4195-8387-3dcc0127efdc-19301624  
Date: Tue, 27 Aug 2019 15:27:41 GMT

Call-ID: 38ed1855-2a120006-78c34baf-1b81d864@10.48.38.102  
CSeq: 101 INVITE  
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY  
Allow-Events: presence  
Supported: replaces  
Server: Cisco-CUCM11.5  
Call-Info: <urn:x-cisco-remotecallinfo>; security= NotAuthenticated; orientation= to; gci= 1-2029; isVoip; call-instance= 1  
Send-Info: conference, x-cisco-conference  
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=called;screen=yes;privacy=off  
Session-ID: 4313758700105000a0006c416a369525;remote=1001532000105000a00038ed18552a12  
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5;user=phone>;party=x-cisco-original-called;privacy=off  
Contact:  
<sip:1001@10.48.38.5:5060;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP6C416A369525"  
Content-Type: application/sdp  
Content-Length: 223

v=0  
o=CiscoSystemsCCM-SIP 104951 1 IN IP4 10.48.38.5  
s=SIP Call  
c=IN IP4 10.48.38.107  
b=AS:64  
t=0 0  
m=audio 20394 RTP/AVP 0 101  
b=TIAS:64000  
a=rtpmap:0 PCMU/8000  
a=rtpmap:101 telephone-event/8000  
a=fmtp:101 0-15

##### BiB allocation request on called (recorded) phone

04241393.000 |11:27:43.629 |SdlSig |SIPAllocateBibResourceReq |restart0  
|SIPBuiltInBridgeControl(1,100,86,15) |SIPStationCdfc(1,100,77,21)  
|1,100,14,83.39^10.48.38.107^\* |[R:N-H:0,N:1,L:0,V:0,Z:0,D:0] CI=19301626  
NumStream=1 BridgeType=0 SsType=16777246 SsKey=5 JccbId=104952 PeerAddr = 10.48.38.107:51902

##### BiB allocated on called (recorded) phone

04241400.000 |11:27:43.630 |SdlSig |MrmAllocateSharedResourceRes |wait  
|Cc(1,100,225,1) |MediaResourceManager(1,100,142,1)  
|1,100,14,83.39^10.48.38.107^\* |[R:N-H:0,N:4,L:0,V:0,Z:0,D:0] CI=19301626  
SsType=16777246 SsKey=5 DN=b0018615001 Name=1b802aa4-863d-879c-f003-9b6de9a1fae5 Pid=1,100,76,27  
BibFlag=T DeviceCapability=256 mPrimaryPartition=

##### DA for first call to activate BiB

04241418.006 |11:27:43.631 |AppInfo |Digit analysis: match(pi="1", fqcn="", cn="", plv="5",  
pss="", TodFilteredPss="", dd="b0018615001",dac="0")  
04241418.007 |11:27:43.631 |AppInfo |Digit analysis: analysis results  
04241418.008 |11:27:43.631 |AppInfo ||PretransformCallingPartyNumber=  
|CallingPartyNumber=  
|DialingPartition=  
|DialingPattern=b0018615001  
|FullyQualifiedCalledPartyNumber=b0018615001  
|DialingPatternRegularExpression=(b0018615001)  
|DialingWhere=  
|PatternType=Enterprise  
|PotentialMatches=NoPotentialMatchesExist  
|DialingSdlProcessId=(1,86,15)  
|PretransformDigitString=b0018615001

|PretransformTagsList=SUBSCRIBER  
|PretransformPositionalMatchList=b0018615001  
|CollectedDigits=b0018615001

##### CUCM sends INVITE #1 to called (recorded) phone with record-invoker=auto in Call-Info field and original Call-ID in Join field

Notice the SDP has a=inactive - even though there is no media established on the Bib yet.

04241449.001 |11:27:43.633 |AppInfo |SIPtcp - wait\_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.107 on port 51902 index 52  
[286950,NET]  
INVITE sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ea2a115cd6  
From: "Call Manager" <sip:10.48.38.5>;tag=104956~e650e088-60ba-4195-8387-3dcc0127efdc-19301628  
To: <sip:1001@10.48.38.5>  
Date: Tue, 27 Aug 2019 15:27:43 GMT  
Call-ID: 35554700-d6514bef-3280-526300e@10.48.38.5  
Supported: timer,resource-priority,replaces  
Min-SE: 1800  
User-Agent: Cisco-CUCM11.5  
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, SUBSCRIBE, NOTIFY  
CSeq: 101 INVITE  
Expires: 180  
Allow-Events: presence  
Call-Info: <urn:x-cisco-remotecallinfo>; isVoip; record-invoker=auto  
Join: 34241a00-d6514bed-327f-526300e@10.48.38.5;from-tag=6c416a369525006f33cf6f38-43c38ad2;to-tag=104952~e650e088-60ba-4195-8387-3dcc0127efdc-19301625  
Session-ID: 00000000000000000000000000000000;remote=00000000000000000000000000000000  
Remote-Party-ID: "Call Manager" <sip:10.48.38.5>;party=calling;screen=yes;privacy=off  
Contact: <sip:10.48.38.5:5060;transport=tcp>  
Max-Forwards: 70  
Content-Type: application/sdp  
Content-Length: 187

v=0  
o=CiscoSystemsCCM-SIP 104956 1 IN IP4 10.48.38.5  
s=SIP Call  
c=IN IP4 10.48.38.5  
t=0 0  
m=audio 4000 RTP/AVP 0  
a=label:X-relay-nearend  
a=rtpmap:0 PCMU/8000  
a=inactive  
a=mid:1

##### Calling phone sends CUCM an ACK in response to the 200 OK which was from when the user at the called phone answered the phone

04241455.002 |11:27:43.697 |AppInfo |SIPtcp - wait\_SdlReadRsp: Incoming SIP TCP message from 10.48.38.102 on port 50147 index 32 with 706 bytes:  
[286951,NET]  
ACK sip:1001@10.48.38.5:5060;transport=tcp SIP/2.0  
Via: SIP/2.0/TCP 10.48.38.102:50147;branch=z9hG4bK688db3c1  
From: "SJ User 1" <sip:1000@10.48.38.5>;tag=38ed18552a12296c00ff41e8-5fb7856e  
To: <sip:1001@10.48.38.5>;tag=104951~e650e088-60ba-4195-8387-3dcc0127efdc-19301624  
Call-ID: 38ed1855-2a120006-78c34baf-1b81d864@10.48.38.102  
Max-Forwards: 70  
Session-ID: 1001532000105000a00038ed18552a12;remote=4313758700105000a0006c416a369525  
Date: Tue, 27 Aug 2019 15:27:45 GMT  
CSeq: 101 ACK  
User-Agent: Cisco-CP7861/12.1.1  
Remote-Party-ID: "SJ User 1" <sip:1000@10.48.38.5>;party=calling;id-

type=subscriber;privacy=off;screen=yes  
Content-Length: 0  
Recv-Info: conference  
Recv-Info: x-cisco-conference

##### Called (recorded) phone returns 200 OK in response to the invite with "record-invoker=auto"

Notice the SDP has a=inactive - even though there is no media established on the Bib yet.

04241466.002 |11:27:43.901 |AppInfo |SIPTcp - wait\_SdlReadRsp: Incoming SIP TCP message from 10.48.38.107 on port 51902 index 52 with 1433 bytes:  
[286953,NET]  
SIP/2.0 200 OK  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ea2a115cd6  
From: "Call Manager" <sip:10.48.38.5>;tag=104956~e650e088-60ba-4195-8387-3dcc0127efdc-19301628  
To: <sip:1001@10.48.38.5>;tag=6c416a369525007019bf48f9-5901eb85  
Call-ID: 35554700-d6514bef-3280-526300e@10.48.38.5  
Session-ID: 0848153900105000a0006c416a369525;remote=00000000000000000000000000000000  
Date: Tue, 27 Aug 2019 15:27:42 GMT  
CSeq: 101 INVITE  
Server: Cisco-CP7841/12.1.1  
Contact: <sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP6C416A369525"  
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO  
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=called;id-type=subscriber;privacy=off;screen=yes  
Supported: replaces,join,sdp-anat,norefersub,resource-priority,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-7.0.0,X-cisco-xsi-8.5.1  
Allow-Events: kpml,dialog  
Recv-Info: conference  
Recv-Info: x-cisco-conference  
Content-Length: 218  
Content-Type: application/sdp  
Content-Disposition: session;handling=optional

v=0  
o=Cisco-SIPUA 2684 0 IN IP4 10.48.38.107  
s=SIP Call  
t=0 0  
m=audio 26396 RTP/AVP 0 101  
c=IN IP4 10.48.38.107  
b=TIAS:64000  
a=rtpmap:0 PCMU/8000  
a=rtpmap:101 telephone-event/8000  
a=fmtp:101 0-15  
a=inactive

##### CUCM responds to called (recorded) phone with ACK

04241469.001 |11:27:43.901 |AppInfo |SIPTcp - wait\_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.107 on port 51902 index 52 [286954,NET] ACK sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0 Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32eb34decb69 From: "Call Manager" <sip:10.48.38.5>;tag=104956~e650e088-60ba-4195-8387-3dcc0127efdc-19301628 To: <sip:1001@10.48.38.5>;tag=6c416a369525007019bf48f9-5901eb85 Date: Tue, 27 Aug 2019 15:27:43 GMT Call-ID: 35554700-d6514bef-3280-526300e@10.48.38.5 User-Agent: Cisco-CUCM11.5 Max-Forwards: 70 CSeq: 101 ACK Allow-Events: presence Content-Length: 0

##### BiB places first call to recording destination address (cn is calling party which is the BiB cn="b0018615001" and it is dialing the recordingdestination dd="7878")

04241501.011 |11:27:43.905 |AppInfo |Digit analysis: match(pi="1", fqcn="", cn="b0018615001", plv="5", pss="EMERGENCY\_PT:INTERNAL\_PT",

TodFilteredPss="EMERGENCY\_PT:INTERNAL\_PT", dd="7878",dac="0")  
04241501.012 |11:27:43.905 |AppInfo |Digit analysis: analysis results  
04241501.013 |11:27:43.905 |AppInfo ||PretransformCallingPartyNumber=b0018615001  
|CallingPartyNumber=b0018615001  
|DialingPartition=INTERNAL\_PT  
|DialingPattern=7878  
|FullyQualifiedCalledPartyNumber=7878  
|DialingPatternRegularExpression=(7878)  
|DialingWhere=  
|PatternType=Enterprise  
|PotentialMatches=NoPotentialMatchesExist  
|DialingSdlProcessId=(0,0,0)  
|PretransformDigitString=7878  
|PretransformTagsList=SUBSCRIBER  
|PretransformPositionalMatchList=7878  
|CollectedDigits=7878

##### DA for to activate BiB for the other person's side of the call

04241545.006 |11:27:43.907 |AppInfo |Digit analysis: match(pi="1", fqc=" ", cn=" ",plv="5",  
pss=" ", TodFilteredPss=" ", dd="b0018615001",dac="0")  
04241545.007 |11:27:43.907 |AppInfo |Digit analysis: analysis results  
04241545.008 |11:27:43.907 |AppInfo ||PretransformCallingPartyNumber=  
|CallingPartyNumber=  
|DialingPartition=  
|DialingPattern=b0018615001  
|FullyQualifiedCalledPartyNumber=b0018615001  
|DialingPatternRegularExpression=(b0018615001)  
|DialingWhere=  
|PatternType=Enterprise  
|PotentialMatches=NoPotentialMatchesExist  
|DialingSdlProcessId=(1,86,15)  
|PretransformDigitString=b0018615001  
|PretransformTagsList=SUBSCRIBER  
|PretransformPositionalMatchList=b0018615001  
|CollectedDigits=b0018615001

##### CUCM sends INVITE #1 to configured recording server (10.48.38.30)

04241555.001 |11:27:43.908 |AppInfo |SIPtcp - wait\_SdlSPISignal: Outgoing SIP TCP message to  
10.48.38.30 on port 5060 index 50  
[286955,NET]  
INVITE sip:7878@10.48.38.30:5060 SIP/2.0  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ecc2c802c  
From: "SJ User 2" <sip:1001@10.48.38.5;x-nearend;x-refci=19301625;x-  
nearendclusterid=StandAloneCluster;x-nearenddevice=SEP6C416A369525;x-nearendaddr=1001;x-  
farendrefci=19301624;x-farendclusterid=StandAloneCluster;x-farenddevice=SEP38ED18552A12;x-  
farendaddr=1000>;tag=104958-e650e088-60ba-4195-8387-3dcc0127efdc-19301629  
To: <sip:7878@10.48.38.30>  
Date: Tue, 27 Aug 2019 15:27:43 GMT  
Call-ID: 35554700-d6514bef-3281-526300e@10.48.38.5  
Supported: timer,resource-priority,replaces  
Min-SE: 1800  
User-Agent: Cisco-CUCM11.5  
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:10.48.38.5:5060>;method="NOTIFY;Event=telephone-event;Duration=500"  
Call-Info: <urn:x-cisco-remotec:callinfo>;x-cisco-video-traffic-class=DESKTOP  
Session-ID: 0848153900105000a0006c416a369525;remote=00000000000000000000000000000000

Cisco-Guid: 0894781184-0000065536-0000000022-0086388750  
Session-Expires: 1800  
P-Asserted-Identity: "SJ User 2" <sip:1001@10.48.38.5>  
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=calling;screen=yes;privacy=off  
Contact:  
<sip:1001@10.48.38.5:5060;transport=tcp>;isFocus;+u.sip!devicename.ccm.cisco.com="SEP6C416A369525"  
Max-Forwards: 70  
Content-Length: 0

##### CUCM sends INVITE #2 to called (recorded) phone with record-invoker=auto in Call-Info field and original Call-ID in Join field  
Notice the SDP has a=inactive - even though there is no media established on the Bib yet.

04241590.001 |11:27:43.910 |AppInfo |SIPTcp - wait\_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.107 on port 51902 index 52  
[286956,NET]  
INVITE sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ed62f39668  
From: "Call Manager" <sip:10.48.38.5>;tag=104959~e650e088-60ba-4195-8387-3dcc0127efdc-19301631  
To: <sip:1001@10.48.38.5>  
Date: Tue, 27 Aug 2019 15:27:43 GMT  
Call-ID: 35554700-d6514bef-3282-526300e@10.48.38.5  
Supported: timer,resource-priority,replaces  
Min-SE: 1800  
User-Agent: Cisco-CUCM11.5  
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, SUBSCRIBE, NOTIFY  
CSeq: 101 INVITE  
Expires: 180  
Allow-Events: presence  
Call-Info: <urn:x-cisco-remotecc:callinfo>; isVoip; record-invoker=auto  
Join: 34241a00-d6514bed-327f-526300e@10.48.38.5;from-tag=6c416a369525006f33cf6f38-43c38ad2;to-tag=104952~e650e088-60ba-4195-8387-3dcc0127efdc-19301625  
Session-ID: 00000000000000000000000000000000;remote=00000000000000000000000000000000  
Remote-Party-ID: "Call Manager" <sip:10.48.38.5>;party=calling;screen=yes;privacy=off  
Contact: <sip:10.48.38.5:5060;transport=tcp>  
Max-Forwards: 70  
Content-Type: application/sdp  
Content-Length: 186

v=0  
o=CiscoSystemsCCM-SIP 104959 1 IN IP4 10.48.38.5  
s=SIP Call  
c=IN IP4 10.48.38.5  
t=0 0  
m=audio 4000 RTP/AVP 0  
a=label:X-relay-farend  
a=rtpmap:0 PCMU/8000  
a=inactive  
a=mid:1

##### Called (recorded) phone returns 200 OK in response to INVITE #2 to invoke BiB  
Notice the SDP has a=inactive - even though there is no media established on the Bib yet.

04241614.002 |11:27:44.197 |AppInfo |SIPTcp - wait\_SdlReadRsp: Incoming SIP TCP message from 10.48.38.107 on port 51902 index 52 with 1434 bytes:  
[286959,NET]  
SIP/2.0 200 OK  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ed62f39668  
From: "Call Manager" <sip:10.48.38.5>;tag=104959~e650e088-60ba-4195-8387-3dcc0127efdc-19301631  
To: <sip:1001@10.48.38.5>;tag=6c416a369525007145d433c8-062b13d7  
Call-ID: 35554700-d6514bef-3282-526300e@10.48.38.5  
Session-ID: 56a8a95e00105000a0006c416a369525;remote=00000000000000000000000000000000

Date: Tue, 27 Aug 2019 15:27:42 GMT  
CSeq: 101 INVITE  
Server: Cisco-CP7841/12.1.1  
Contact: <sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP6C416A369525"  
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO  
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=called;id-type=subscriber;privacy=off;screen=yes  
Supported: replaces,join,sdp-anat,norefersub,resource-priority,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-7.0.0,X-cisco-xsi-8.5.1  
Allow-Events: kpml,dialog  
Recv-Info: conference  
Recv-Info: x-cisco-conference  
Content-Length: 219  
Content-Type: application/sdp  
Content-Disposition: session;handling=optional

v=0  
o=Cisco-SIPUA 13977 0 IN IP4 10.48.38.107  
s=SIP Call  
t=0 0  
m=audio 17904 RTP/AVP 0 101  
c=IN IP4 10.48.38.107  
b=TIAS:64000  
a=rtpmap:0 PCMU/8000  
a=rtpmap:101 telephone-event/8000  
a=fmtp:101 0-15  
a=inactive

##### CUCM responds with ACK for 200 OK for INVITE #2 to invoke the BiB

04241618.001 |11:27:44.199 |AppInfo |SIPTcp - wait\_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.107 on port 51902 index 52  
[286960,NET]  
ACK sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ee41b380b1  
From: "Call Manager" <sip:10.48.38.5>;tag=104959~e650e088-60ba-4195-8387-3dcc0127efdc-19301631  
To: <sip:1001@10.48.38.5>;tag=6c416a369525007145d433c8-062b13d7  
Date: Tue, 27 Aug 2019 15:27:43 GMT  
Call-ID: 35554700-d6514bef-3282-526300e@10.48.38.5  
User-Agent: Cisco-CUCM11.5  
Max-Forwards: 70  
CSeq: 101 ACK  
Allow-Events: presence  
Content-Length: 0

##### BiB places second call to recording destination address (cn is calling party which is the BiB cn="b0018615001" and it is dialing the recordingdestination dd="7878")

04241651.011 |11:27:44.201 |AppInfo |Digit analysis: match(pi="1", fqcn="", cn="b0018615001", plv="5", pss="EMERGENCY\_PT:INTERNAL\_PT", TodFilteredPss="EMERGENCY\_PT:INTERNAL\_PT", dd="7878", dac="0")  
04241651.012 |11:27:44.202 |AppInfo |Digit analysis: analysis results  
04241651.013 |11:27:44.202 |AppInfo ||PretransformCallingPartyNumber=b0018615001  
|CallingPartyNumber=b0018615001  
|DialingPartition=INTERNAL\_PT  
|DialingPattern=7878  
|FullyQualifiedCalledPartyNumber=7878  
|DialingPatternRegularExpression=(7878)  
|DialingWhere=  
|PatternType=Enterprise  
|PotentialMatches=NoPotentialMatchesExist

|DialingSdlProcessId=(0,0,0)  
|PretransformDigitString=7878  
|PretransformTagsList=SUBSCRIBER  
|PretransformPositionalMatchList=7878  
|CollectedDigits=7878

##### CUCM sends INVITE #2 to configured recording server

04241698.001 |11:27:44.205 |AppInfo |SIPTcp - wait\_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.30 on port 5060 index 50  
[286961,NET]  
INVITE sip:7878@10.48.38.30:5060 SIP/2.0  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ef2867938b  
From: "SJ User 2" <sip:1001@10.48.38.5;x-farend;x-refci=19301625;x-nearendclusterid=StandAloneCluster;x-nearenddevice=SEP6C416A369525;x-nearendaddr=1001;x-farendrefci=19301624;x-farendclusterid=StandAloneCluster;x-farenddevice=SEP38ED18552A12;x-farendaddr=1000>;tag=104961~e650e088-60ba-4195-8387-3dcc0127efdc-19301632  
To: <sip:7878@10.48.38.30>  
Date: Tue, 27 Aug 2019 15:27:44 GMT  
Call-ID: 35eddd80-d6514bf0-3283-526300e@10.48.38.5  
Supported: timer,resource-priority,replaces  
Min-SE: 1800  
User-Agent: Cisco-CUCM11.5  
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:10.48.38.5:5060>;method="NOTIFY;Event=telephone-event;Duration=500"  
Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=DESKTOP  
Session-ID: 56a8a95e00105000a0006c416a369525;remote=00000000000000000000000000000000  
Cisco-Guid: 0904781184-0000065536-0000000023-0086388750  
Session-Expires: 1800  
P-Asserted-Identity: "SJ User 2" <sip:1001@10.48.38.5>  
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=calling;screen=yes;privacy=off  
Contact:  
<sip:1001@10.48.38.5:5060;transport=tcp>;isFocus;+u.sip!devicename.ccm.cisco.com="SEP6C416A369525"  
Max-Forwards: 70  
Content-Length: 0

##### CUCM receives a 200 OK from recording server for INVITE #2

04241723.002 |11:27:44.324 |AppInfo |SIPTcp - wait\_SdlReadRsp: Incoming SIP TCP message from 10.48.38.30 on port 5060 index 50 with 1205 bytes:  
[286963,NET]  
SIP/2.0 200 Ok  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ef2867938b  
To: <sip:7878@10.48.38.30>;tag=ds1ald776c  
From: "SJ User 2" <sip:1001@10.48.38.5;x-farend;x-refci=19301625;x-nearendclusterid=StandAloneCluster;x-nearenddevice=SEP6C416A369525;x-nearendaddr=1001;x-farendrefci=19301624;x-farendclusterid=StandAloneCluster;x-farenddevice=SEP38ED18552A12;x-farendaddr=1000>;tag=104961~e650e088-60ba-4195-8387-3dcc0127efdc-19301632  
Call-ID: 35eddd80-d6514bf0-3283-526300e@10.48.38.5  
CSeq: 101 INVITE  
Content-Length: 475  
Contact: <sip:7878@10.48.38.30:5060;transport=TCP>  
Content-Type: application/sdp  
Allow: INVITE, BYE, CANCEL, ACK, NOTIFY, INFO, UPDATE  
Supported: X-cisco-srtp-fallback  
Server: MediaSense/11.x



v=0  
o=CiscoORA 707 1 IN IP4 10.48.38.30  
s=SIP Call  
c=IN IP4 10.48.38.30  
t=0 0  
m=audio 56512 RTP/SAVP 102 0 8 9 18  
a=rtpmap:102 MP4A-LATM/90000  
a=fmtp:102 profile-level-id=24;object=23;bitrate=64000  
a=rtpmap:0 PCMU/8000  
a=rtpmap:8 PCMA/8000  
a=rtpmap:9 G722/8000  
a=rtpmap:18 G729/8000  
a=recvonly  
a=crypto:XX  
a=crypto:XX

##### CUCM receives 200 OK from the recording server in response to INVITE #1

04241743.002 |11:27:44.326 |AppInfo |SIPTcp - wait\_SdlReadRsp: Incoming SIP TCP message from 10.48.38.30 on port 5060 index 50 with 1205 bytes:  
[286964,NET]  
SIP/2.0 200 Ok  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ecc2c802c  
To: <sip:7878@10.48.38.30>;tag=ds2c967644  
From: "SJ User 2" <sip:1001@10.48.38.5;x-nearend;x-refci=19301625;x-nearendclusterid=StandAloneCluster;x-nearenddevice=SEP6C416A369525;x-nearendaddr=1001;x-farendrefci=19301624;x-farendclusterid=StandAloneCluster;x-farenddevice=SEP38ED18552A12;x-farendaddr=1000>;tag=104958~e650e088-60ba-4195-8387-3dcc0127efdc-19301629  
Call-ID: 35554700-d6514bef-3281-526300e@10.48.38.5  
CSeq: 101 INVITE  
Content-Length: 475  
Contact: <sip:7878@10.48.38.30:5060;transport=TCP>  
Content-Type: application/sdp  
Allow: INVITE, BYE, CANCEL, ACK, NOTIFY, INFO, UPDATE  
Supported: X-cisco-srtp-fallback  
Server: MediaSense/11.x

v=0  
o=CiscoORA 708 1 IN IP4 10.48.38.30  
s=SIP Call  
c=IN IP4 10.48.38.30  
t=0 0  
m=audio 59058 RTP/SAVP 102 0 8 9 18  
a=rtpmap:102 MP4A-LATM/90000  
a=fmtp:102 profile-level-id=24;object=23;bitrate=64000  
a=rtpmap:0 PCMU/8000  
a=rtpmap:8 PCMA/8000  
a=rtpmap:9 G722/8000  
a=rtpmap:18 G729/8000  
a=recvonly  
a=crypto:XX  
a=crypto:XX

##### CUCM sends re-INVITE #2 to called (recorded) phone (notice there is no SDP - this is so CUCM can identify the codec the BiB is locked to)  
Notice there is no SDP

04241825.001 |11:27:44.330 |AppInfo |SIPTcp - wait\_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.107 on port 51902 index 52  
[286965,NET]  
INVITE sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f014677161  
From: "Call Manager" <sip:10.48.38.5>;tag=104959~e650e088-60ba-4195-8387-3dcc0127efdc-19301631

To: <sip:1001@10.48.38.5>;tag=6c416a369525007145d433c8-062b13d7  
Date: Tue, 27 Aug 2019 15:27:44 GMT  
Call-ID: 35554700-d6514bef-3282-526300e@10.48.38.5  
Supported: timer,resource-priority,replaces  
User-Agent: Cisco-CUCM11.5  
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, SUBSCRIBE, NOTIFY  
CSeq: 102 INVITE  
Max-Forwards: 70  
Expires: 180  
Allow-Events: presence  
Call-Info: <urn:x-cisco-remotecc:callinfo>; isVoip; record-invoker=auto  
Min-SE: 1800  
Session-ID: 00000000000000000000000000000000;remote=56a8a95e00105000a0006c416a369525  
Remote-Party-ID: "Call Manager" <sip:10.48.38.5>;party=calling;screen=yes;privacy=off  
Contact: <sip:10.48.38.5:5060;transport=tcp>  
Content-Length: 0

##### CUCM sends re-INVITE #1 to called (recorded) phone (notice there is no SDP - this is so CUCM can identify the codec the BiB is locked to)

04241866.001 |11:27:44.332 |AppInfo |SIPTcp - wait\_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.107 on port 51902 index 52  
[286966,NET]  
INVITE sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f11da4ce39  
From: "Call Manager" <sip:10.48.38.5>;tag=104956~e650e088-60ba-4195-8387-3dcc0127efdc-19301628  
To: <sip:1001@10.48.38.5>;tag=6c416a369525007019bf48f9-5901eb85  
Date: Tue, 27 Aug 2019 15:27:44 GMT  
Call-ID: 35554700-d6514bef-3280-526300e@10.48.38.5  
Supported: timer,resource-priority,replaces  
User-Agent: Cisco-CUCM11.5  
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, SUBSCRIBE, NOTIFY  
CSeq: 102 INVITE  
Max-Forwards: 70  
Expires: 180  
Allow-Events: presence  
Call-Info: <urn:x-cisco-remotecc:callinfo>; isVoip; record-invoker=auto  
Min-SE: 1800  
Session-ID: 00000000000000000000000000000000;remote=0848153900105000a0006c416a369525  
Remote-Party-ID: "Call Manager" <sip:10.48.38.5>;party=calling;screen=yes;privacy=off  
Contact: <sip:10.48.38.5:5060;transport=tcp>  
Content-Length: 0

##### Called (recorded) phone returns 200 OK for re-INVITE #2

04241872.002 |11:27:44.541 |AppInfo |SIPTcp - wait\_SdlReadRsp: Incoming SIP TCP message from 10.48.38.107 on port 51902 index 52 with 1434 bytes:  
[286969,NET]  
SIP/2.0 200 OK  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f014677161  
From: "Call Manager" <sip:10.48.38.5>;tag=104959~e650e088-60ba-4195-8387-3dcc0127efdc-19301631  
To: <sip:1001@10.48.38.5>;tag=6c416a369525007145d433c8-062b13d7  
Call-ID: 35554700-d6514bef-3282-526300e@10.48.38.5  
Session-ID: 56a8a95e00105000a0006c416a369525;remote=00000000000000000000000000000000  
Date: Tue, 27 Aug 2019 15:27:43 GMT  
CSeq: 102 INVITE  
Server: Cisco-CP7841/12.1.1  
Contact: <sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP6C416A369525"  
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO  
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=called;id-type=subscriber;privacy=off;screen=yes  
Supported: replaces,join,sdp-anat,norefersub,resource-priority,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-7.0.0,X-cisco-xsi-8.5.1  
Allow-Events: kpml,dialog  
Recv-Info: conference  
Recv-Info: x-cisco-conference  
Content-Length: 219  
Content-Type: application/sdp  
Content-Disposition: session;handling=optional

v=0  
o=Cisco-SIPUA 13977 1 IN IP4 10.48.38.107  
s=SIP Call  
t=0 0  
m=audio 17904 RTP/AVP 0 101  
c=IN IP4 10.48.38.107  
b=TIAS:64000  
a=rtpmap:0 PCMU/8000  
a=rtpmap:101 telephone-event/8000  
a=fmtp:101 0-15  
a=sendrecv

##### Called (recorded) phone returns 200 OK to re-INVITE #1

04241885.002 |11:27:44.550 |AppInfo |SIPTcp - wait\_SdlReadRsp: Incoming SIP TCP message from 10.48.38.107 on port 51902 index 52 with 1433 bytes:  
[286970,NET]  
SIP/2.0 200 OK  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f11da4ce39  
From: "Call Manager" <sip:10.48.38.5>;tag=104956~e650e088-60ba-4195-8387-3dcc0127efdc-19301628  
To: <sip:1001@10.48.38.5>;tag=6c416a369525007019bf48f9-5901eb85  
Call-ID: 35554700-d6514bef-3280-526300e@10.48.38.5  
Session-ID: 0848153900105000a0006c416a369525;remote=00000000000000000000000000000000  
Date: Tue, 27 Aug 2019 15:27:43 GMT  
CSeq: 102 INVITE  
Server: Cisco-CP7841/12.1.1  
Contact: <sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP6C416A369525"  
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO  
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=called;id-type=subscriber;privacy=off;screen=yes  
Supported: replaces,join,sdp-anat,norefersub,resource-priority,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-7.0.0,X-cisco-xsi-8.5.1  
Allow-Events: kpml,dialog  
Recv-Info: conference  
Recv-Info: x-cisco-conference  
Content-Length: 218  
Content-Type: application/sdp  
Content-Disposition: session;handling=optional

v=0  
o=Cisco-SIPUA 2684 1 IN IP4 10.48.38.107  
s=SIP Call  
t=0 0  
m=audio 26396 RTP/AVP 0 101  
c=IN IP4 10.48.38.107  
b=TIAS:64000  
a=rtpmap:0 PCMU/8000  
a=rtpmap:101 telephone-event/8000  
a=fmtp:101 0-15  
a=sendrecv

##### CUCM sends ACK to called (recorded) phone for re-INVITE #2

04241903.001 |11:27:44.552 |AppInfo |SIPTcp - wait\_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.107 on port 51902 index 52 [286971,NET]  
ACK sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f252b587f6  
From: "Call Manager" <sip:10.48.38.5>;tag=104959~e650e088-60ba-4195-8387-3dcc0127efdc-19301631  
To: <sip:1001@10.48.38.5>;tag=6c416a369525007145d433c8-062b13d7  
Date: Tue, 27 Aug 2019 15:27:44 GMT  
Call-ID: 35554700-d6514bef-3282-526300e@10.48.38.5  
User-Agent: Cisco-CUCM11.5  
Max-Forwards: 70  
CSeq: 102 ACK  
Allow-Events: presence  
Session-ID: 00000000000000000000000000000000;remote=56a8a95e00105000a0006c416a369525  
Content-Type: application/sdp  
Content-Length: 192

v=0  
o=CiscoSystemsCCM-SIP 104959 3 IN IP4 10.48.38.5  
s=SIP Call  
c=IN IP4 10.48.38.30  
b=TIAS:64000  
b=AS:64  
t=0 0  
m=audio 56512 RTP/AVP 0  
b=TIAS:64000  
a=rtpmap:0 PCMU/8000  
a=recvonly

##### CUCM sends ACK to the recording server in response to 200 OK #2

04241917.001 |11:27:44.555 |AppInfo |SIPTcp - wait\_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.30 on port 5060 index 50 [286972,NET]  
ACK sip:7878@10.48.38.30:5060;transport=TCP SIP/2.0  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f373e69393  
From: "SJ User 2" <sip:1001@10.48.38.5;x-farend;x-refci=19301625;x-nearendclusterid=StandAloneCluster;x-nearenddevice=SEP6C416A369525;x-nearendaddr=1001;x-farendrefci=19301624;x-farendclusterid=StandAloneCluster;x-farenddevice=SEP38ED18552A12;x-farendaddr=1000>;tag=104961~e650e088-60ba-4195-8387-3dcc0127efdc-19301632  
To: <sip:7878@10.48.38.30>;tag=dslald776c  
Date: Tue, 27 Aug 2019 15:27:44 GMT  
Call-ID: 35eddd80-d6514bf0-3283-526300e@10.48.38.5  
User-Agent: Cisco-CUCM11.5  
Max-Forwards: 70  
CSeq: 101 ACK  
Allow-Events: presence, kpml  
Session-ID: 56a8a95e00105000a0006c416a369525;remote=c83405810147c69016c38634ab104961  
Content-Type: application/sdp  
Content-Length: 235

v=0  
o=CiscoSystemsCCM-SIP 104961 1 IN IP4 10.48.38.5  
s=SIP Call  
c=IN IP4 10.48.38.107  
b=TIAS:64000  
b=AS:64  
t=0 0  
m=audio 17904 RTP/AVP 0 101  
a=rtpmap:0 PCMU/8000  
a=sendonly  
a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15

##### CUCM sends ACK to called (recorded) phone for re-INVITE #1

04241947.001 |11:27:44.559 |AppInfo |SIPTcp - wait\_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.107 on port 51902 index 52 [286973,NET]

ACK sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f45d25b711  
From: "Call Manager" <sip:10.48.38.5>;tag=104956~e650e088-60ba-4195-8387-3dcc0127efdc-19301628  
To: <sip:1001@10.48.38.5>;tag=6c416a369525007019bf48f9-5901eb85  
Date: Tue, 27 Aug 2019 15:27:44 GMT  
Call-ID: 35554700-d6514bef-3280-526300e@10.48.38.5  
User-Agent: Cisco-CUCM11.5  
Max-Forwards: 70  
CSeq: 102 ACK  
Allow-Events: presence  
Session-ID: 00000000000000000000000000000000;remote=0848153900105000a0006c416a369525  
Content-Type: application/sdp  
Content-Length: 192

v=0  
o=CiscoSystemsCCM-SIP 104956 3 IN IP4 10.48.38.5  
s=SIP Call  
c=IN IP4 10.48.38.30  
b=TIAS:64000  
b=AS:64  
t=0 0  
m=audio 59058 RTP/AVP 0  
b=TIAS:64000  
a=rtpmap:0 PCMU/8000  
a=recvonly

##### CUCM sends ACK to the recording server in response to 200 OK #1

04241948.001 |11:27:44.559 |AppInfo |SIPTcp - wait\_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.30 on port 5060 index 50 [286974,NET]

ACK sip:7878@10.48.38.30:5060;transport=TCP SIP/2.0  
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f573871bbb  
From: "SJ User 2" <sip:1001@10.48.38.5;x-nearend;x-refci=19301625;x-nearendclusterid=StandAloneCluster;x-nearenddevice=SEP6C416A369525;x-nearendaddr=1001;x-farendrefci=19301624;x-farendclusterid=StandAloneCluster;x-farenddevice=SEP38ED18552A12;x-farendaddr=1000>;tag=104958~e650e088-60ba-4195-8387-3dcc0127efdc-19301629  
To: <sip:7878@10.48.38.30>;tag=ds2c967644  
Date: Tue, 27 Aug 2019 15:27:43 GMT  
Call-ID: 35554700-d6514bef-3281-526300e@10.48.38.5  
User-Agent: Cisco-CUCM11.5  
Max-Forwards: 70  
CSeq: 101 ACK  
Allow-Events: presence, kpml  
Session-ID: 0848153900105000a0006c416a369525;remote=c83405810147c69016c38634ab104958  
Content-Type: application/sdp  
Content-Length: 235

v=0  
o=CiscoSystemsCCM-SIP 104958 1 IN IP4 10.48.38.5  
s=SIP Call  
c=IN IP4 10.48.38.107  
b=TIAS:64000  
b=AS:64  
t=0 0  
m=audio 26396 RTP/AVP 0 101

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

## 문제 해결

이 섹션에서는 컨피그레이션 문제를 해결하는 데 사용할 수 있는 정보를 제공합니다.

## 코덱 협상

다음은 가장 일반적인 통화 녹음 실패 유형 중 하나인 녹음된 전화기와 녹음 서버 간 코덱 불일치의 예입니다.

```
~~~~~
Codec Negotiation Failure
~~~~~

### Calling phone places call

00019629.001 |12:48:34.510 |AppInfo |StationInit: (0000005) EnblocCall calledParty=9110001.

### CUCM performs digit analysis against the dialed digits (dd="9110001")

00019638.001 |12:48:34.511 |AppInfo |Digit Analysis: star_DaReq: daReq.partitionSearchSpace(),
filteredPartitionSearchSpaceString(), partitionSearchSpaceString()
00019638.002 |12:48:34.511 |AppInfo |Digit Analysis: star_DaReq: Matching Legacy Numeric,
digits=9110001
00019638.003 |12:48:34.522 |AppInfo |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept
DAMR.sstype=[0], TPcount=[0], DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]
00019638.004 |12:48:34.522 |AppInfo |Digit Analysis: getDaRes - Remote Destination [] isURI[1]
00019638.005 |12:48:34.522 |AppInfo |Digit analysis: patternUsage=2
00019638.006 |12:48:34.522 |AppInfo |Digit analysis: match(pi="2", fqcn="9110006",
cn="9110006",plv="5", pss="", TodFilteredPss="", dd="9110001",dac="1")
00019638.007 |12:48:34.522 |AppInfo |Digit analysis: analysis results
00019638.008 |12:48:34.522 |AppInfo ||PretransformCallingPartyNumber=9110006
|CallingPartyNumber=9110006
|DialingPartition=
|DialingPattern=9110001
|FullyQualifiedCalledPartyNumber=9110001
|DialingPatternRegularExpression=(9110001)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=9110001
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=9110001
|CollectedDigits=9110001

### CUCM determines call must stay on same node and go to LineControl
(PID=LineControl(2,100,174,19))

00019640.001 |12:48:34.522 |AppInfo |Digit analysis: wait_DmPidRes- Partition=[]
```

Pattern=[9110001] Where=[],cmDeviceType=[UserDevice], OutsideDialtone =[0], DeviceOverride=[0], PID=LineControl(2,100,174,7),CI=[49613637],Sender=Cdcc(2,100,219,1)

### CUCM extends the call to the called phone

00019657.003 |12:48:34.560 |AppInfo |StationD: (0000007) DEBUG whatToDo: line=1 calls=0 limit=4, busy=2. GCI=(2, 7001), cm\_PL=(5, 0).  
00019657.004 |12:48:34.560 |AppInfo |StationD: (0000007) DEBUG whatToDo: busy trigger not hit... send to open appearance  
00019657.005 |12:48:34.560 |AppInfo |preFilterCapCount =[11], preFilterCaps :: (Cap)= (25) (6) (4) (2) (7) (8) (15) (16) (11) (12) (257) Filtering Caps due to Service Parameter Configuration postFilterCapCount =[8], postFilterCaps :: (Cap)= (25) (4) (2) (15) (16) (11) (12) (257)  
00019657.006 |12:48:34.560 |AppInfo |preFilterCapCount =[0], preFilterCaps :: (Cap)= Filtering Caps due to Service Parameter Configuration postFilterCapCount =[0], postFilterCaps :: (Cap)=  
00019657.007 |12:48:34.560 |Created | | |  
|StationCdpc(2,100,64,2) |StationD(2,100,63,7) | | |  
|NumOfCurrentInstances: 2  
00019657.008 |12:48:34.560 |AppInfo |StationD: (0000007) DEBUG- getLineRingSetting: retVal=4.  
00019657.009 |12:48:34.560 |AppInfo |StationD: (0000007) DEBUG- saveRinger for: ci=49613638, line=1, mode=2, cm\_precedence=5, callPhase=5.  
00019657.010 |12:48:34.560 |AppInfo |StationD: (0000007) DEBUG- saveRinger: ci=49613638, line=1, mode=2, cm\_precedence=5, callPhase=5, modifier=0  
00019657.011 |12:48:34.560 |AppInfo |StationD: (0000007) INFO sendCallAcceptReq: Try to send StationLineCallAccept to cdpc=2 .  
00019657.012 |12:48:34.560 |AppInfo |StationD: (0000007) playRinger for: ci=49613638.  
00019657.013 |12:48:34.560 |AppInfo |StationD: (0000007) DEBUG- getLineRingSetting: retVal=4.  
00019657.014 |12:48:34.560 |AppInfo |StationD: (0000007) DEBUG- getLineRingSetting: retVal=4.  
00019657.015 |12:48:34.560 |AppInfo |StationD: (0000007) DEBUG- getLineRingSetting: retVal=4.

### The Called (recorded) phone goes off hook

00019709.001 |12:48:36.042 |AppInfo |StationD: (0000007) restart0\_StationOffHook - INFO: CI=49613638 on line=1, SPKMode=0, alwaysPrimeLine=0, alwaysUsePrimeLineForVM=0, fid=9999, offHookTrigger=1.

### CUCM Tells the calling phone to open the logical channel

00019773.001 |12:48:36.061 |AppInfo |StationD: (0000005) SEP0018195AA209 , star\_MediaExchangeAgenaOpenLogicalChannel packetSize=20, codec=4, ci=49613637

### CUCM Tells the called (recorded) to open the logical channel

00019776.001 |12:48:36.061 |AppInfo |StationD: (0000007) SEP001795BDD16B , star\_MediaExchangeAgenaOpenLogicalChannel packetSize=20, codec=4, ci=49613638

### CUCM Tells the calling phone to open the receive channel

00019784.002 |12:48:36.062 |AppInfo |StationD: (0000005) OpenReceiveChannel conferenceID=49613637 passThruPartyID=33554433 millisecondPacketSize=20 compressionType=4(Media\_Payload\_G711Ulaw64k) RFC2833PayloadType=0 qualifierIn=? sourceIpAddr=IpAddr.type:0 ipAddr:0x0e302021000000000000000000000000(10.48.32.33). myIP: IpAddr.type:0 ipv4Addr:0x0e30201c(10.48.32.28)

### Codec locked due to recording on called (recorded) phone

00019785.003 |12:48:36.062 |AppInfo | StationCdpc: star\_MediaExchangeAgenaQueryCapability - Device SEP001795BDD16B, codec locked due to recording, codecType=4

### CUCM Tells the called (recorded) phone to open the receive channel

00019788.002 |12:48:36.062 |AppInfo |StationD: (0000007) OpenReceiveChannel  
conferenceID=49613638 passThruPartyID=33554434 millisecondPacketSize=20  
compressionType=4(Media\_Payload\_G711Ulaw64k) RFC2833PayloadType=0 qualifierIn=?  
sourceIpAddr=IpAddr.type:0 ipAddr:0x0e30201c000000000000000000000000(10.48.32.28). myIP:  
IpAddr.type:0 ipv4Addr:0x0e302021(10.48.32.33)

### CUCM allocates the BiB on the called (recorded) phone

00019830.000 |12:48:36.074 |SdlSig |MrmAllocateUcbResourceReq |waiting  
|MediaResourceManager(2,100,138,1) |Cc(2,100,220,1)  
|2,100,14,19.206^10.48.32.33^SEP001795BDD16B |[R:N-H:0,N:1,L:0,V:0,Z:0,D:0] CI=49613639  
SsType=33554461 SsKey=1 BridgeType=0 MRGLPkid= NumStream=1 Bib=89cdb152-4ef2-4d60-9e6b-  
ab8c77c22618 BibTgCi=49613638 FeatId=159 PL=5 PLDmn=0 DeviceCapability=0 NumVideoCapable=0  
requestDeviceType=0 requestDeviceLocale=64 forkingDevicePosition=2 playToneDir=3

### BiB places it's first call to recording destination address (cn is calling number which is  
the BiB cn="b00223906001" and it is dialing the recordingdestination dd="8675309")

00019889.001 |12:48:36.100 |AppInfo |Digit Analysis: star\_DaReq: daReq.partitionSearchSpace(),  
filteredPartitionSearchSpaceString(), partitionSearchSpaceString()  
00019889.002 |12:48:36.100 |AppInfo |Digit Analysis: star\_DaReq: Matching Legacy Numeric,  
digits=8675309  
00019889.003 |12:48:36.100 |AppInfo |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept  
DAMR.sstype=[0], TPcount=[0], DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]  
00019889.004 |12:48:36.100 |AppInfo |Digit Analysis: getDaRes - Remote Destination [8675309]  
isURI[0]  
00019889.005 |12:48:36.100 |AppInfo |CMUtility routeCallThroughCTIRD: no matching  
RemDestDynamic record exists for remdest [8675309]  
00019889.006 |12:48:36.100 |AppInfo |DbMobility: getMatchedRemDest starts: cnumber = 8675309  
00019889.007 |12:48:36.100 |AppInfo |DbMobility: getMatchedRemDest: full match case  
00019889.008 |12:48:36.100 |AppInfo |DbMobility SelectByDestination: no matching RemDestDynamic  
record exists for remdest [8675309]  
00019889.009 |12:48:36.100 |AppInfo |DbMobility: can't find remdest 8675309 in map  
00019889.010 |12:48:36.100 |AppInfo |Digit analysis: patternUsage=5  
00019889.011 |12:48:36.100 |AppInfo |Digit analysis: match(pi="1", fqcn="",  
cn="b00223906001",plv="5", pss="E911\_PT:Phones\_PT:EMERGENCY\_PT:INTERNAL\_PT:INFORMACAST\_PT",  
TodFilteredPss="E911\_PT:Phones\_PT:EMERGENCY\_PT:INTERNAL\_PT:INFORMACAST\_PT",  
dd="8675309",dac="1")  
00019889.012 |12:48:36.100 |AppInfo |Digit analysis: analysis results  
00019889.013 |12:48:36.100 |AppInfo ||PretransformCallingPartyNumber=b00223906001  
|CallingPartyNumber=b00223906001  
|DialingPartition=  
|DialingPattern=8675309  
|FullyQualifiedCalledPartyNumber=8675309  
|DialingPatternRegularExpression=(8675309)  
|DialingWhere=  
|PatternType=Enterprise  
|PotentialMatches=NoPotentialMatchesExist  
|DialingSdlProcessId=(0,0,0)  
|PretransformDigitString=8675309  
|PretransformTagsList=SUBSCRIBER  
|PretransformPositionalMatchList=8675309  
|CollectedDigits=8675309

### Calling phone sends CUCM the ORC ACK

00019912.001 |12:48:36.139 |AppInfo |StationInit: (0000005) OpenReceiveChannelAck Status=0,  
IpAddr=IpAddr.type:0 ipAddr:0x0e30201c000000000000000000000000(10.48.32.28), Port=31678,  
PartyID=33554433

### CUCM sends startMediaTransmission to the called (recorded) phone telling the phone to send



RTP to the calling phone (10.48.32.28)

```
00019920.001 |12:48:36.139 |AppInfo |StationD: (0000007) startMediaTransmission
conferenceID=49613638 passThruPartyID=33554434 remoteIpAddress=IpAddr.type:0
ipAddr:0x0e30201c000000000000000000000000(10.48.32.28) remotePortNumber=31678
milliSecondPacketSize=20 compressType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0
qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e302021(10.48.32.33)
```

### Called (recorded) phone sends CUCM the ORC ACK

```
00019959.001 |12:48:36.145 |AppInfo |StationInit: (0000007) OpenReceiveChannelAck Status=0,
IpAddr=IpAddr.type:0 ipAddr:0x0e302021000000000000000000000000(10.48.32.33), Port=28360,
PartyID=33554434
```

### CUCM sends startMediaTransmission to the calling phone telling the phone to send RTP to the called phone (10.48.32.33)

```
00019977.001 |12:48:36.146 |AppInfo |StationD: (0000005) startMediaTransmission
conferenceID=49613637 passThruPartyID=33554433 remoteIpAddress=IpAddr.type:0
ipAddr:0x0e302021000000000000000000000000(10.48.32.33) remotePortNumber=28360
milliSecondPacketSize=20 compressType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0
qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e30201c(10.48.32.28)
```

```
### BiB places second call to recording destination address (cn is calling number which is the
BiB cn="b00223906001" and it is dialing the recordingdestination dd="8675309")
Note that the BiB number stayed the same (b00223906001) and so did the recordingdestination
number 00020002.001 |12:48:36.147 |AppInfo |Digit Analysis: star_DaReq:
daReq.partitionSearchSpace(), filteredPartitionSearchSpaceString(), partitionSearchSpaceString()
00020002.002 |12:48:36.147 |AppInfo |Digit Analysis: star_DaReq: Matching Legacy Numeric,
digits=8675309 00020002.003 |12:48:36.147 |AppInfo |Digit Analysis: getDaRes data:
daRes.ssType=[0] Intercept DAMR.sstype=[0], TPcount=[0], DAMR.NotifyCount=[0],
DaRes.NotifyCount=[0] 00020002.004 |12:48:36.147 |AppInfo |Digit Analysis: getDaRes - Remote
Destination [8675309] isURI[0] 00020002.005 |12:48:36.147 |AppInfo |CMUtility
routeCallThroughCTIRD: no matching RemDestDynamic record exists for remdest [8675309]
00020002.006 |12:48:36.147 |AppInfo |DbMobility: getMatchedRemDest starts: cnumber = 8675309
00020002.007 |12:48:36.147 |AppInfo |DbMobility: getMatchedRemDest: full match case 00020002.008
|12:48:36.147 |AppInfo |DbMobility SelectByDestination: no matching RemDestDynamic record exists
for remdest [8675309] 00020002.009 |12:48:36.147 |AppInfo |DbMobility: can't find remdest
8675309 in map 00020002.010 |12:48:36.147 |AppInfo |Digit analysis: patternUsage=5 00020002.011
|12:48:36.147 |AppInfo |Digit analysis: match(pi="1", fqc="", cn="b00223906001",plv="5",
pss="E911_PT:Phones_PT:EMERGENCY_PT:INTERNAL_PT:INFORMACAST_PT",
TodFilteredPss="E911_PT:Phones_PT:EMERGENCY_PT:INTERNAL_PT:INFORMACAST_PT",
dd="8675309",dac="1") 00020002.012 |12:48:36.147 |AppInfo |Digit analysis: analysis results
00020002.013 |12:48:36.147 |AppInfo ||PretransformCallingPartyNumber=b00223906001
|CallingPartyNumber=b00223906001 |DialingPartition= |DialingPattern=8675309
|FullyQualifiedCalledPartyNumber=8675309 |DialingPatternRegularExpression=(8675309)
|DialingWhere= |PatternType=Enterprise |PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0) |PretransformDigitString=8675309 |PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=8675309 |CollectedDigits=8675309 |UnconsumedDigits=
|TagsList=SUBSCRIBER |PositionalMatchList=8675309
```

### CUCM sends INVITE #1 to configured recording server (10.48.32.170)

```
00020086.001 |12:48:36.156 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message
to 10.48.32.170:[5060]:
[901,NET]
INVITE sip:8675309@10.48.32.170:5060 SIP/2.0
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hg4bk4f2a857d3d
From: <sip:9110001@10.48.32.90;x-nearend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-
nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-
farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-
farendaddr=9110006>;tag=351~713e2333-4032-45f1-b1f5-e33cf47lacec-49613642
To: <sip:8675309@10.48.32.170>
```

Date: Tue, 14 Oct 2014 16:48:36 GMT  
Call-ID: ef7acf80-43d153e4-50-5a20300e@10.48.32.90  
Supported: timer,resource-priority,replaces  
Min-SE: 1800  
User-Agent: Cisco-CUCM10.5  
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: ;method="NOTIFY;Event=telephone-event;Duration=500"  
Cisco-Guid: 4017803136-0000065536-0000000001-1512058894  
Session-Expires: 1800  
P-Asserted-Identity: <sip:9110001@10.48.32.90>  
Remote-Party-ID: <sip:9110001@10.48.32.90>;party=calling;screen=yes;privacy=off  
Contact: <sip:9110001@10.48.32.90:5060>;isFocus  
Max-Forwards: 70  
Content-Length: 0

### CUCM sends INVITE #2 to configured recording server (10.48.32.170)

00020088.001 |12:48:36.157 |AppInfo |//SIP/SIPUdp/wait\_SdlSPISignal: Outgoing SIP UDP message to 10.48.32.170:[5060]:

[902,NET]

INVITE sip:8675309@10.48.32.170:5060 SIP/2.0

Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK5014378d0b

From: <sip:9110001@10.48.32.90;x-farend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=352~713e2333-4032-45f1-b1f5-e33cf47lacec-49613645

To: <sip:8675309@10.48.32.170>

Date: Tue, 14 Oct 2014 16:48:36 GMT

Call-ID: ef7acf80-43d153e4-51-5a20300e@10.48.32.90

Supported: timer,resource-priority,replaces

Min-SE: 1800

User-Agent: Cisco-CUCM10.5

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: ;method="NOTIFY;Event=telephone-event;Duration=500"

Cisco-Guid: 4017803136-0000065536-0000000002-1512058894

Session-Expires: 1800

P-Asserted-Identity: <sip:9110001@10.48.32.90>

Remote-Party-ID: <sip:9110001@10.48.32.90>;party=calling;screen=yes;privacy=off

Contact: <sip:9110001@10.48.32.90:5060>;isFocus

Max-Forwards: 70

Content-Length: 0

### CUCM receives a 200 OK from recording server for INVITE #1

00020089.001 |12:48:36.161 |AppInfo |//SIP/SIPUdp/wait\_SdlDataInd: Incoming SIP UDP message size 731 from 10.48.32.170:[5060]:

[903,NET]

SIP/2.0 200 OK

Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK4f2a857d3d

From: <sip:9110001@10.48.32.90;x-nearend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=351~713e2333-4032-45f1-b1f5-e33cf47lacec-49613642

To: <sip:8675309@10.48.32.170>;tag=1  
Call-ID: ef7acf80-43d153e4-50-5a20300e@10.48.32.90  
CSeq: 101 INVITE  
Contact: <sip:10.48.32.170:5060;transport=udp>  
Content-Type: application/sdp  
Content-Length: 135

v=0  
o=user1 53655765 2353687637 IN IP4 10.48.32.170  
s=-  
c=IN IP4 10.48.32.170  
t=0 0  
m=audio 6000 RTP/AVP 0  
a=rtpmap:0 PCMU/8000

### CUCM receives a 200 OK from recording server for INVITE #2

00020092.001 |12:48:36.161 |AppInfo |//SIP/SIPUdp/wait\_SdlDataInd: Incoming SIP UDP message size 730 from 10.48.32.170:[5060]:  
[905,NET]

SIP/2.0 200 OK

Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK5014378d0b

From: <sip:9110001@10.48.32.90;x-farend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=352~713e2333-4032-45f1-b1f5-e33cf47lacec-49613645

To: <sip:8675309@10.48.32.170>;tag=2

Call-ID: ef7acf80-43d153e4-51-5a20300e@10.48.32.90

CSeq: 101 INVITE

Contact: <sip:10.48.32.170:5060;transport=udp>

Content-Type: application/sdp

Content-Length: 135

v=0  
o=user1 53655765 2353687637 IN IP4 10.48.32.170  
s=-  
c=IN IP4 10.48.32.170  
t=0 0  
m=audio 6000 RTP/AVP 0  
a=rtpmap:0 PCMU/8000

### Region information for connecting audio for recording call, both appear to support G.711. Note that the bandwidth capabilities printed is kbps=8 meaning the region relationship between the two regions is limited to codecs using 8kbps or less. 00020160.005 |12:48:36.190 |AppInfo |DET-RegionsServer::matchCapabilities-- savedOption=3, PREF\_NONE, regionA=(null) regionB=(null) latentCaps(A=0, B=0) kbps=8, capACount=1, capBCount=1 00020160.006 |12:48:36.190 |AppInfo |DET-MediaManager-(2)::checkAudioPassThru, param(bPostMTPAllocation=0,chkTrp=1), capCount(1,1), mtpPT=1, aPT=2 00020160.007 |12:48:36.190 |AppInfo |DET-MediaManager-(2)::preCheckCapabilities, **region1=Default, region2=RecordingTrunk, Pty1** capCount=1 (Cap,ptime)= **(4,20)**, **Pty2** capCount=1 (Cap,ptime)= **(4,20)** 00020160.008 |12:48:36.190 |AppInfo |DET-RegionsServer::matchCapabilities-- savedOption=0, PREF\_NONE, regionA=(null) regionB=(null) latentCaps(A=0, B=0) **kbps=8**, capACount=1, capBCount=1

### CUCM determines 2 transcoders are required and attempts to allocate

00020160.011 |12:48:36.190 |AppInfo |DET-MediaManager-(2)::preCheckCapabilities, **caps mismatch! Xcoder Req'd. kbps(8)**, filtered A[capCount=0 (Cap,ptime)=], B[capCount=0 (Cap,ptime)=] allowMTP=0 **numXcoderRequired=2** xcodingSide=0

### No transcoder is configured which can cause this call to fail

00020162.003 |12:48:36.190 |AppInfo |MediaResourceManager::sendAllocationResourceErr - ERROR - no transcoder device configured

### CUCM sendt the ACK and BYE to the recording server in response to INVITE #1  
Note the Q.850 cause code

00020210.001 |12:48:36.216 |AppInfo |//SIP/SIPUdp/wait\_SdlSPISignal: Outgoing SIP UDP message to 10.48.32.170:[5060]:

[906,NET]

ACK sip:10.48.32.170:5060;transport=UDP SIP/2.0

Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK51257b2b47

From: <sip:9110001@10.48.32.90;x-nearend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=351~713e2333-4032-45f1-b1f5-e33cf47lacec-49613642

To: <sip:8675309@10.48.32.170>;tag=1

Date: Tue, 14 Oct 2014 16:48:36 GMT

Call-ID: ef7acf80-43d153e4-50-5a20300e@10.48.32.90

User-Agent: Cisco-CUCM10.5

Max-Forwards: 70

CSeq: 101 ACK

Allow-Events: presence, kpml

Content-Length: 0

00020211.001 |12:48:36.216 |AppInfo |//SIP/SIPUdp/wait\_SdlSPISignal: Outgoing SIP UDP message to 10.48.32.170:[5060]:

[907,NET]

BYE sip:10.48.32.170:5060;transport=UDP SIP/2.0

Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK526f3d2afa

From: <sip:9110001@10.48.32.90;x-nearend;x-refci=49613638;x-nearendclusterid=GlensCUCM10-5;x-nearenddevice=SEP001795BDD16B;x-nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=GlensCUCM10-5;x-farenddevice=SEP0018195AA209;x-farendaddr=9110006>;tag=351~713e2333-4032-45f1-b1f5-e33cf47lacec-49613642

To: <sip:8675309@10.48.32.170>;tag=1

Date: Tue, 14 Oct 2014 16:48:36 GMT

Call-ID: ef7acf80-43d153e4-50-5a20300e@10.48.32.90

User-Agent: Cisco-CUCM10.5

Max-Forwards: 70

P-Asserted-Identity: <sip:9110001@10.48.32.90>

CSeq: 102 BYE

**Reason: Q.850;cause=47**

Content-Length: 0

### CUCM sendt the ACK and BYE to the recording server in response to INVITE #2  
Note the Q.850 cuase code in the BYE

00020248.001 |12:48:36.218 |AppInfo |//SIP/SIPUdp/wait\_SdlSPISignal: Outgoing SIP UDP message to 10.48.32.170:[5060]:

[908,NET]

ACK sip:10.48.32.170:5060;transport=UDP SIP/2.0

Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK531df920a6

From: <sip:9110001@10.48.32.90;x-farend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=352~713e2333-4032-45f1-b1f5-e33cf47lacec-49613645

To: <sip:8675309@10.48.32.170>;tag=2

Date: Tue, 14 Oct 2014 16:48:36 GMT

Call-ID: ef7acf80-43d153e4-51-5a20300e@10.48.32.90

User-Agent: Cisco-CUCM10.5

Max-Forwards: 70

CSeq: 101 ACK

Allow-Events: presence, kpml

Content-Length: 0

```
00020249.001 |12:48:36.218 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to
10.48.32.170:[5060]: [909,NET] BYE sip:10.48.32.170:5060;transport=UDP SIP/2.0 Via: SIP/2.0/UDP
10.48.32.90:5060;branch=z9hG4bK5462aba807 From: <sip:9110001@10.48.32.90;x-farend;x-
refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-
nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-
farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=352~713e2333-4032-45f1-b1f5-e33cf471acec-
49613645 To: <sip:8675309@10.48.32.170>;tag=2 Date: Tue, 14 Oct 2014 16:48:36 GMT Call-ID:
ef7acf80-43d153e4-51-5a20300e@10.48.32.90 User-Agent: Cisco-CUCM10.5 Max-Forwards: 70 P-
Asserted-Identity: <sip:9110001@10.48.32.90> CSeq: 102 BYE Reason: Q.850;cause=47
Content-Length: 0
```

## CSS 및 PT 문제를 포함한 잘못된 컨피그레이션

이 명령을 사용하면 통화를 기록하지 않는 전화기의 알려진 MAC 주소만으로 대부분의 녹음 컨피그레이션을 빠르게 검토할 수 있습니다. 여기 예시에서처럼 명령 **MAC\_of\_Phone**의 부분을 전화기의 실제 MAC 주소로 바꾸면 됩니다.

이렇게 하면 검색하는 MAC의 DN(둘 이상의 경우 모두), 확인용 전화기의 MAC, BIB 설정, 프라이버시 설정, 녹음 유형(실습의 예에 나열된 값 참조), 전화기에서 사용 중인 녹음 프로파일, 녹음 CSS(Call Search Space) 이름, 녹음 프로파일에 대한 녹음 대상, 검색한 MAC을 기반으로 녹음 대상이 연결된 파티션을 제공합니다.

```
run sql select nl.dnorpattern as phone_dn, dev.name as phone_mac, CASE
dev.tkstatus_builtinbridge WHEN '1' THEN 'BiB is on' WHEN '0' THEN 'BiB is off' ELSE 'NA' END as
is_bib_on, CASE dev.resettoggle WHEN 't' THEN 'Privacy is on' WHEN 'f' THEN 'Privacy is off'
ELSE 'NA' END as is_privacy_on, CASE recordynam.tkrecordingflag WHEN '0' THEN 'Recording
Disabled' WHEN '1' THEN 'Automatic' WHEN '2' THEN 'Selective' ELSE 'NA' END as recording_type,
CASE devnumplanmap.tkpreferredmediasource WHEN '1' THEN 'Gateway Preferred' WHEN '2' THEN 'Phone
Preferred' ELSE 'NA' END as Recording_Media_Source, rcrdpro.name as recording_profile_name,
css.name as css_used_by_recording_profile, rcrdpro.recorderdestination as
recording_route_pattern, rp.name as required_partition_for_css_used_by_recording_profile from
recordingprofile as rcrdpro inner join callingsearchspace as css on
rcrdpro.fkcallingsearchspace_callrecording = css.pkid inner join numplan as n on n.dnorpattern =
rcrdpro.recorderdestination inner join routepartition as rp on rp.pkid = n.fkroutepartition
inner join devicenumplanmap as devnumplanmap on rcrdpro.pkid = devnumplanmap.fkreordingprofile
inner join recordingdynamic as recordynam on devnumplanmap.pkid = recordynam.fkdevicenumplanmap
inner join device as dev on devnumplanmap.fkdevice = dev.pkid inner join numplan as nl on
devnumplanmap.fknumplan = nl.pkid where css.pkid = rcrdpro.fkcallingsearchspace_callrecording
and dev.name='MAC_of_Phone'
```

그러면 검색 대상 전화기의 MAC과 연결된 녹음 프로파일의 녹음 CSS와 연결된 파티션 목록이 제공 됩니다.

```
run sql select css.name as name_of_the_recording_css, rp.name as partitions_in_recording_css,
csm.sortorder from callingsearchspace as css inner join callingsearchspace_member as csm on
csm.fkcallingsearchspace = css.pkid inner join routepartition as rp on csm.fkroutepartition =
rp.pkid inner join recordingprofile as rcrdpro on rcrdpro.fkcallingsearchspace_callrecording =
css.pkid inner join devicenumplanmap as devnumplanmap on rcrdpro.pkid =
devnumplanmap.fkreordingprofile inner join device as dev on devnumplanmap.fkdevice = dev.pkid
where css.pkid = rcrdpro.fkcallingsearchspace_callrecording and dev.name='MAC_of_Phone'
```

다음은 MAC 주소가 **SEPC80084AA8743**인 전화기에 대한 랩 출력의 예입니다.

이 명령에서 전화기에 2003이라는 하나의 DN만 있는 것을 볼 수 있습니다. 또한 BIB가 On이고, 프라이버시가 Off이고, 녹음 유형이 자동이고, 기본 소스가 전화기이며, 녹음 프로파일이 **Test Recording Profile**이고, 녹음 발신 검색 공간이 **INTERNAL\_CSS**이고, 녹음된 통화에 대한 경로 패턴이 **8675309**이고 이 패턴은 파티션 **INTERNAL\_PT**와 연결되어 있습니다.

```

run sql select nl.dnorpattern as phone_dn, dev.name as phone_mac, CASE
dev.tkstatus_builtinbridge WHEN '1' THEN 'BiB is on' WHEN '0' THEN 'BiB is off' ELSE 'NA' END as
is_bib_on, CASE dev.resettoggle WHEN 't' THEN 'Privacy is on' WHEN 'f' THEN 'Privacy is off'
ELSE 'NA' END as is_privacy_on, CASE recordynam.tkrecordingflag WHEN '0' THEN 'Recording
Disabled' WHEN '1' THEN 'Automatic' WHEN '2' THEN 'Selective' ELSE 'NA' END as recording_type,
CASE devnumplanmap.tkpreferredmediasource WHEN '1' THEN 'Gateway Preferred' WHEN '2' THEN 'Phone
Preferred' ELSE 'NA' END as Recording_Media_Source, rcrdpro.name as recording_profile_name,
css.name as css_used_by_recording_profile, rcrdpro.recorderdestination as
recording_route_pattern, rp.name as required_partition_for_css_used_by_recording_profile from
recordingprofile as rcrdpro inner join callingsearchspace as css on
rcrdpro.fkcallingsearchspace_callrecording = css.pkid inner join numplan as n on n.dnorpattern =
rcrdpro.recorderdestination inner join routepartition as rp on rp.pkid = n.fkroutepartition
inner join devicenumplanmap as devnumplanmap on rcrdpro.pkid = devnumplanmap.fkrecordingprofile
inner join recordingdynamic as recordynam on devnumplanmap.pkid = recordynam.fkdevicenumplanmap
inner join device as dev on devnumplanmap.fkdevice = dev.pkid inner join numplan as nl on
devnumplanmap.fknumplan = nl.pkid where css.pkid = rcrdpro.fkcallingsearchspace_callrecording
and dev.name='SEPC80084AA8743'
phone_dn phone_mac is_bib_on is_privacy_on recording_type recording_media_source
recording_profile_name css_used_by_recording_profile recording_route_pattern
required_partition_for_css_used_by_recording_profile
=====
=====
=====

```

```

2003 SEPC80084AA8743 BiB is on Privacy is off Automatic Phone Preferred Test Recording Profile
INTERNAL_CSS 8675309 INTERNAL_PT

```

이 명령의 출력으로 녹음 CSS 및 관심 전화기와 연결된 녹음 프로파일의 모든 파티션을 확인할 수 있습니다. 여기서 INTERNAL\_PT 파티션이 발신 검색 공간 INTERNAL\_CSS와 연결된 파티션 중 하나임을 확인할 수 있습니다. 즉, 녹음 경로 패턴을 호출할 수 있는 전화기의 BIB에 문제가 없어야 합니다.

```

run sql select css.name as name_of_the_recording_css, rp.name as partitions_in_recording_css,
csm.sortorder from callingsearchspace as css inner join callingsearchspacemember as csm on
csm.fkcallingsearchspace = css.pkid inner join routepartition as rp on csm.fkroutepartition =
rp.pkid inner join recordingprofile as rcrdpro on rcrdpro.fkcallingsearchspace_callrecording =
css.pkid inner join devicenumplanmap as devnumplanmap on rcrdpro.pkid =
devnumplanmap.fkrecordingprofile inner join device as dev on devnumplanmap.fkdevice = dev.pkid
where css.pkid = rcrdpro.fkcallingsearchspace_callrecording and dev.name='SEPC80084AA8743'
name_of_the_recording_css partitions_in_recording_css sortorder
=====
INTERNAL_CSS          E911_PT              1
INTERNAL_CSS          Phones_PT            2
INTERNAL_CSS          EMERGENCY_PT        3
INTERNAL_CSS          INTERNAL_PT          4
INTERNAL_CSS          INFORMACAST_PT      5

```

## 관련 정보

- [Cisco Collaboration System 11.x SRND\(Solution Reference Network Design\)](#)

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