

Konfiguration und Fehlerbehebung bei grundlegenden Anrufaufzeichnungen

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Einleitung

In diesem Dokument werden die Grundlagen der Anrufaufzeichnung in Cisco Unified Communications Manager (CUCM) beschrieben.

Voraussetzungen

Anforderungen

Cisco empfiehlt, dass Sie über Kenntnisse des in einen Aufzeichnungsserver eines Drittanbieters integrierten CUCM verfügen.

Verwendete Komponenten

Die Informationen in diesem Dokument basierend auf folgenden Software- und Hardware-Versionen:

- CUCM
- Cisco Internet Protocol (IP)
- Telefon-Anrufaufzeichnungsserver

Die Informationen in diesem Dokument beziehen sich auf Geräte in einer speziell eingerichteten Testumgebung. Alle Geräte, die in diesem Dokument benutzt wurden, begannen mit einer gelöschten (Nichterfüllungs) Konfiguration. Wenn Ihr Netzwerk in Betrieb ist, stellen Sie sicher, dass Sie die möglichen Auswirkungen aller Befehle verstehen.

Hintergrundinformationen

In diesem Dokument werden auch der erwartete Medienfluss, die erwarteten Anrufflüsse für SIP- (Session Initiation Protocol) und SCCP-Geräte (Skinny Client Control Protocol) sowie ein Beispiel für einen häufigen Fehler bei der Einrichtung der Anrufaufzeichnung beschrieben.

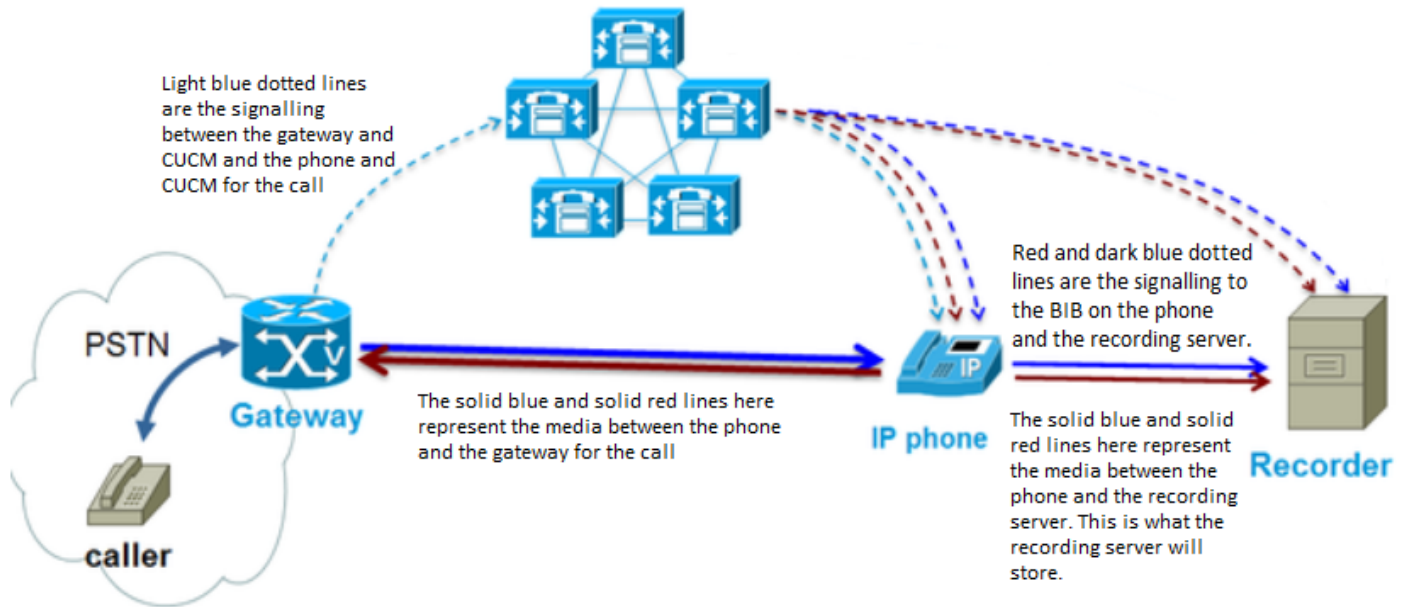
Arten der Anrufaufzeichnung

Automatisch

Die wichtigsten Elemente der automatischen Anrufaufzeichnung sind:

- Verwendet die integrierte Bridge (BIB) des IP-Telefons, um Audio an das Aufzeichnungsziel weiterzuleiten
- Wird jedes Mal initiiert, wenn das IP-Telefon einen Anruf tätigt oder empfängt
- Erfordert nur einen SIP-Trunk zwischen CUCM und dem Aufzeichnungsziel. Einige Aufzeichnungsanbieter benötigen Computer Telephony Integration (CTI).
- Keine Aufzeichnung von Telefonen außerhalb des verwalteten Netzwerks möglich (muss RTP direkt an den Aufzeichnungsserver senden können und ein Cisco IP-Telefon sein, das eine BIB zuweisen kann)

In diesem Diagramm stellen die durchgezogenen Linien den erwarteten Medienfluss und die gestrichelten Linien den erwarteten Signalisierungsfluss dar:

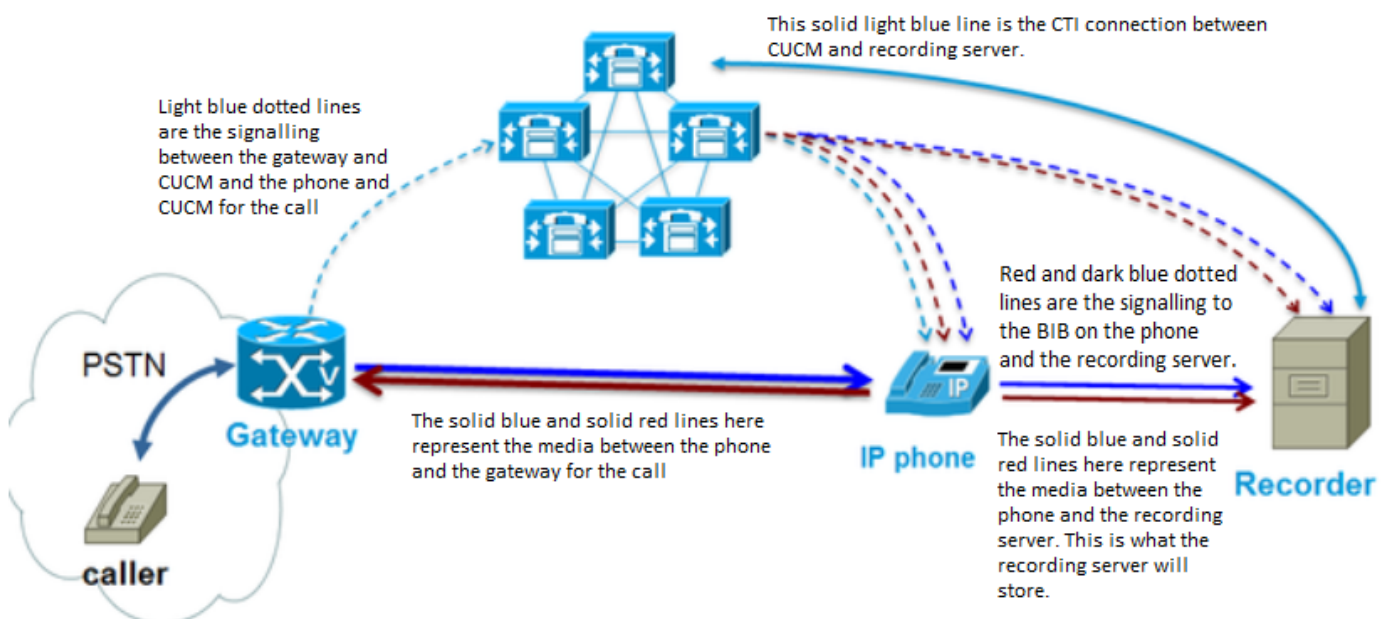


Anwendung aufgerufen

Die wichtigsten Elemente der von der Anwendung aufgerufenen Anrufaufzeichnung sind:

- Verwendet BIB des IP-Telefons, um Audio an das Aufzeichnungsziel weiterzuleiten
- Initiiert, wenn die Anwendung (Rekorder) vorschreibt, dass sie initiiert werden muss
- SIP-Trunk und CTI mit Aufzeichnungsanwendung erforderlich
- Der Benutzer der CTI-Anwendung muss Zugriff auf Endgeräte haben, die aufgezeichnet werden müssen.
- Keine Aufzeichnung von Telefonen außerhalb des verwalteten Netzwerks zulassen (muss Zugriff haben, um RTP direkt an den Aufzeichnungsserver zu senden)

Im Diagramm stellen die durchgezogenen Linien den erwarteten Medienfluss und die gestrichelten Linien den erwarteten Signalisierungsfluss dar. Die durchgezogene Linie zwischen CUCM und dem Aufzeichnungsserver kennzeichnet eine CTI-Verbindung zwischen CUCM und der Anwendung.

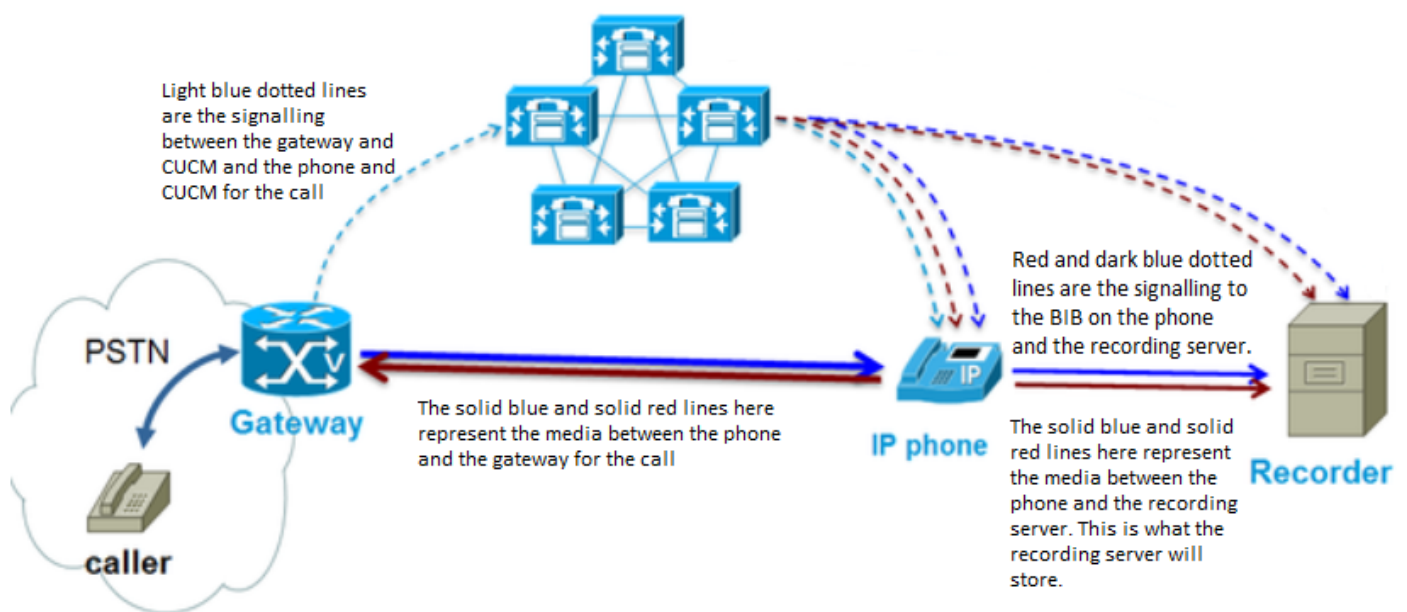


Selektiv

Die wichtigsten Elemente der selektiven Anrufaufzeichnung sind:

- Verwendet BIB des IP-Telefons, um Audio an das Aufzeichnungsziel weiterzuleiten
- Wird jedes Mal gestartet, wenn der IP-Telefonbenutzer die Aufzeichnungsoption auf seinem IP-Telefon (CUCM 9.x+) oder in einer Anwendung wie in [diesem Bild](#) auswählt.
- In der Regel ist nur ein SIP-Trunk zwischen CUCM und dem Aufzeichnungsziel erforderlich (abhängig vom Anbieter der Aufzeichnungsanwendung).
- Keine Aufzeichnung von Telefonen außerhalb des verwalteten Netzwerks zulassen (muss Zugriff haben, um RTP direkt an den Aufzeichnungsserver zu senden)

Wie Sie in diesem Diagramm sehen können, ähnelt der Medien- und Signalisierungspfad der automatischen Anrufaufzeichnung:

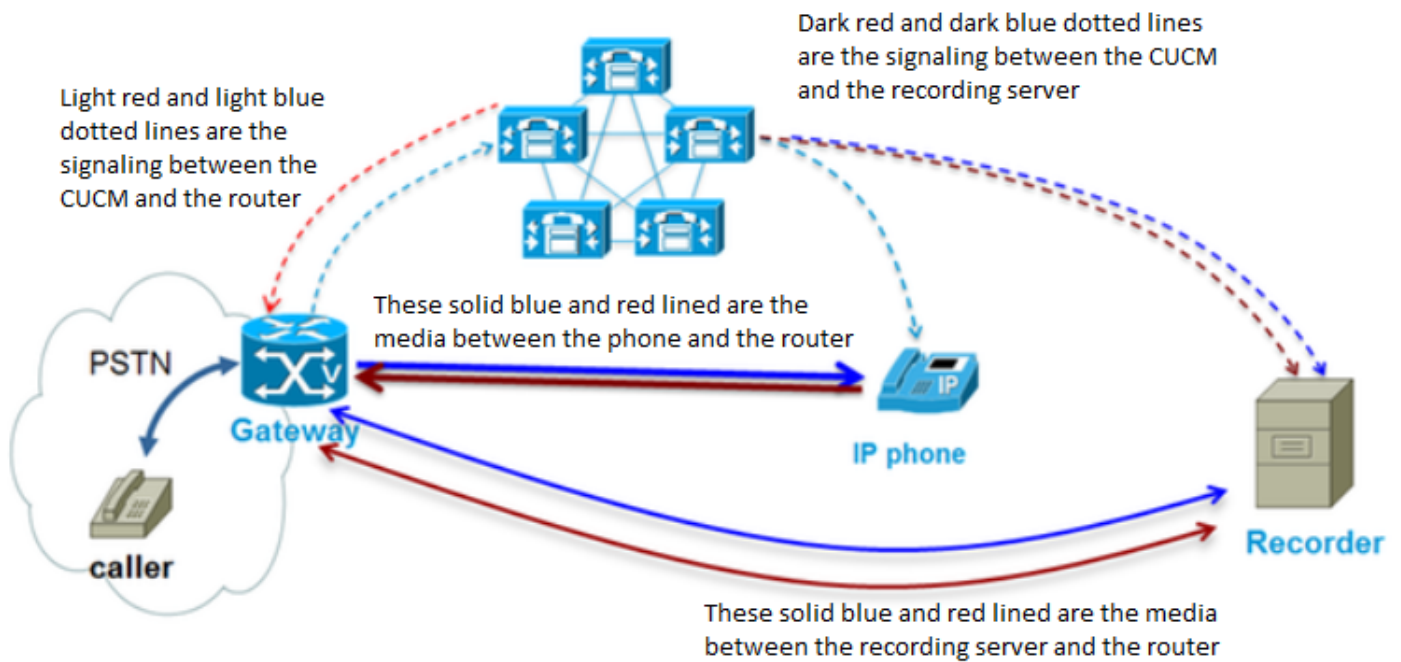


Gateway-basiert

Die wichtigsten Elemente der Gateway-basierten Anrufaufzeichnung sind:

- Voice Gateway leitet Medien zum Aufzeichnungsziel weiter
- CUCM registriert sich beim Gateway als Anwendung
- CUCM weist Gateway (GW) über HTTP an, Medien zum Aufzeichnungsziel zu streamen.
- CUCM wird über SIP-Trunk in das Aufzeichnungsziel integriert
- Ermöglicht die Aufzeichnung von Anrufen, die einfach über ein verwaltetes Netzwerk (z. B. an mobile Benutzer) weitergeleitet werden, oder für Telefone, die das BIB nicht unterstützen

Wie Sie dem Diagramm hier entnehmen können, unterscheidet sich der Medienfluss erheblich von den anderen Arten der Anrufaufzeichnung:



Automatische Anrufaufzeichnungskonfiguration für SIP Only-Integration

In diesem Abschnitt wird beschrieben, wie Sie die SIP-Integration eines Aufzeichnungsservers einrichten.

Erstellen eines SIP-Trunks zum Aufzeichnungsziel

- Navigieren Sie zu **Gerät > Trunk**, und wählen Sie **Neu hinzufügen** aus.
- Erstellen Sie einen SIP-Trunk mit den im Bild gezeigten Einstellungen.

Trunk Configuration

➔ Next

Status

i Status: Ready

Trunk Information

Trunk Type*	SIP Trunk
Device Protocol*	SIP
Trunk Service Type*	None(Default)

Next

- Geben Sie den entsprechenden Gerätenamen, Gerätepool, MRGL, SIP-Trunk-

Sicherheitsprofil und SIP-Profil ein.

- Die konfigurierte Zieladresse ist die Adresse des Aufzeichnungsanwendungsservers.

Aufzeichnungsprofil erstellen

- Navigieren Sie zu **Device > Device Settings > Recording Profile**.
- Die Zieladresse für die Aufzeichnung ist der Ort, an den die Aufzeichnungsaufrufe gesendet werden, wie im Bild gezeigt.

Recording Profile Configuration

Save Delete Copy Add New

Status

i Status: Ready

Recording Profile Information

Name* Test Recording Profile

Recording Calling Search Space INTERNAL_CSS

Recording Destination Address * 8675309

Save Delete Copy Add New

Routenmuster für Anrufe zur Routenaufzeichnung erstellen

- Erstellen eines Routenmusters, das mit der im vorherigen Schritt konfigurierten Aufzeichnungszieladresse übereinstimmt
- Sie können auf eine Routenliste anstatt direkt auf den SIP-Trunk verweisen, wenn Sie redundante SIP-Trunks konfigurieren möchten.

Hinweis: Die diesem Routenmuster zugewiesene Partition muss dem RecordingCallingSearch Space zugeordnet sein und im Bild dargestellt sein.

Pattern Definition	
Route Pattern*	8675309
Route Partition	INTERNAL_PT
Description	
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	RecordingTrunk (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern

Zuweisen des Aufzeichnungsprofils zu einer Telefonleitung

- Weisen Sie auf einem bereits erstellten Telefon mit einer vorhandenen Durchwahl das erstellte Aufzeichnungsprofil zu.
- Weisen Sie auch hier die Art der Anrufaufzeichnung zu.
- Das Beispiel zeigt die automatische Aufzeichnung, wie im Bild dargestellt.

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

Auf der Telefonkonfigurationsseite BIB auf "Ein" und Datenschutz "Aus" setzen

Navigieren Sie auf der Seite für die Gerätekonfiguration zum Abschnitt **Geräteinformationen**. Legen Sie Built In Bridge auf **On (Ein)** und Privacy auf **Off** (Wie im Bild gezeigt) fest.

Built In Bridge*	On
Privacy*	Off

Überprüfung

Verwenden Sie diesen Abschnitt, um zu überprüfen, ob Ihre Konfiguration ordnungsgemäß funktioniert.

Nachfolgend sind die erwarteten Verhaltensweisen in den Call Manager-Ablaufverfolgungen für SCCP- und SIP-Telefone mit der angegebenen Konfiguration aufgeführt. Diese Beispiele gelten für ein Telefon, das ein anderes Telefon im gleichen Cluster anruft, während eines der Telefone für die Anrufaufzeichnung eingerichtet ist.

Hinweis: Die vom CUCM zu erfassenden Protokolle sind CTIManager, CallManager, Event Viewer App/Sys, und in einigen Szenarien können pcap erforderlich sein.

Hinweis: Bei den Protokollen, die von Telefonen gesammelt werden, handelt es sich um Konsolenprotokolle und pcaps. Sie können pcaps vom Aufnahmeserver gleichzeitig mit den pcaps vom Telefon abrufen.

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-b1f5-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-e33cf47lacec-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-b1f5-e33cf47lacec-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-b1f5-e33cf47lacec-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-b1f5-e33cf47lacec-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_G711Ul原因64k) 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_G711Ul原因64k) 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-b1f5-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  
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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
a=ptime: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-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: 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=z9hG4bK32eb34dec69 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", fqcn="", 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-remotecc: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-remotecallinfo>;x-cisco-video-traffic-class=DESKTOP
Session-ID: 56a8a95e00105000a0006c416a369525;remote=00000000000000000000000000000000
Cisco-Guid: 0904781184-0000065536-000000023-0086388750
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-remotecallinfo>; 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-remotecallinfo>; 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=ds1ald776c
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
```

Fehlerbehebung

Dieser Abschnitt enthält Informationen, die Sie zur Fehlerbehebung bei Ihrer Konfiguration verwenden können.

Codec-Aushandlung

Dies ist ein Beispiel für einen der häufigsten Fehler bei der Anrufaufzeichnung - Abweichung des Codecs zwischen dem aufgezeichneten Telefon und dem Aufzeichnungsserver:

```
~~~~~
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-e33cf471lacec-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-e33cf471lacec-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-e33cf471acec-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-e33cf471acec-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-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 P-
Asserted-Identity: <sip:9110001@10.48.32.90> CSeq: 102 BYE Reason: Q.850;cause=47
Content-Length: 0
```

Fehlerhafte Konfiguration mit CSS- und PT-Problemen

Mit diesen Befehlen können die meisten Aufzeichnungskonfigurationen schnell überprüft werden, wobei nur die bekannte MAC-Adresse eines Telefons verwendet wird, das keine Anrufe aufzeichnet. Ersetzen Sie einfach den Teil des Befehls **MAC_of_Phone** durch die tatsächliche MAC-Adresse des Telefons, wie in den hier gezeigten Beispielen.

Sie erhalten die DN (alle, falls mehrere vorhanden sind) für die von Ihnen gesuchte MAC, die MAC des Telefons nur für die Bestätigung, die BIB-Einstellung, die Datenschutzeinstellung, den Aufzeichnungstyp (Verweis auf die in den Beispielen der Übung aufgeführten Werte), das vom Telefon verwendete Aufzeichnungsprofil, den Namen der aufzeichnenden Call Search Spaces (CSS), das Aufzeichnungsziel für dieses Aufzeichnungsprofil und die Partition, der das Aufzeichnungsziel zugeordnet ist. basierend auf der MAC-Adresse, nach der Sie suchen:

```
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 n1 on
devnumplanmap.fknumplan = n1.pkid where css.pkid = rcrdpro.fkcallingsearchspace_callrecording
and dev.name='MAC_of_Phone'
```

Dadurch erhalten Sie eine Liste der Partitionen, die dem Aufzeichnungs-CSS im Aufzeichnungsprofil zugeordnet sind, das der MAC-Adresse des Telefons zugeordnet ist, nach dem Sie suchen.

```
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='MAC_of_Phone'
```

Hier sind Beispiele für die Ausgabe aus der Übung für ein Telefon mit der MAC-Adresse **SEPC80084AA8743**:

In diesem Befehl können Sie sehen, dass das Telefon nur eine DN hat, die 2003 ist. Wir sehen auch, dass die BIB Ein ist, Privatsphäre Aus ist, der Aufzeichnungstyp automatisch ist, die bevorzugte Quelle Telefon ist, das Aufzeichnungsprofil **Test Recording Profile** ist, der Aufzeichnungs-Calling Search Space ist **INTERNAL_CSS**, das Routenmuster für aufgezeichnete Anrufe ist **8675 309** und dieses Muster ist mit der Partition **INTERNAL_PT** verknüpft.

```
run sql select n1.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 n1 on
devnumplanmap.fknumplan = n1.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
```

Mit der Ausgabe dieses Befehls können Sie alle Partitionen des Aufzeichnungs-CSS und des Aufzeichnungsprofils überprüfen, die dem gewünschten Telefon zugeordnet sind. Sie können hier

sehen, dass die Partition **INTERNAL_PT** eine der Partitionen ist, die dem aufrufenden Suchraum **INTERNAL_CSS** zugeordnet sind. Das bedeutet, dass es keine Probleme mit der BIB des Telefons geben darf, die in der Lage ist, das Aufzeichnungs-Routenmuster anzurufen.

```
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
```

Zugehörige Informationen

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

Informationen zu dieser Übersetzung

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