

Contents

[Introduction](#)

[Prerequisites](#)

[Requirements](#)

[Components Used](#)

[Types of Call Recording](#)

[Automatic](#)

[Application Invoked](#)

[Selective](#)

[Gateway-based](#)

[Automatic Call Recording Configuration for SIP only Integration](#)

[Create the SIP Trunk to the Recording Destination](#)

[Create the Recording Profile](#)

[Create the Route Pattern to Route the Recording Calls](#)

[Assign Recording Profile to Phone line](#)

[Set the BIB to on and Privacy to off on the Phone Configuration page](#)

[Verify](#)

[SCCP](#)

[SIP](#)

[Troubleshoot](#)

[Codec Negotiation](#)

[Misconfiguration Including CSS and PT issues](#)

Introduction

This document describes the basics of call recording within Cisco Unified Communications Manager (CUCM), the expected media flow, the expected call flows for Session Initiation Protocol (SIP) and Skinny Client Control Protocol (SCCP) devices, and an example of a common type of call recording setup failure.

Prerequisites

Requirements

CUCM integrated with a third-party recording server.

Components Used

CUCM, Cisco IP phone (IP is Internet Protocol), and a call recording server.

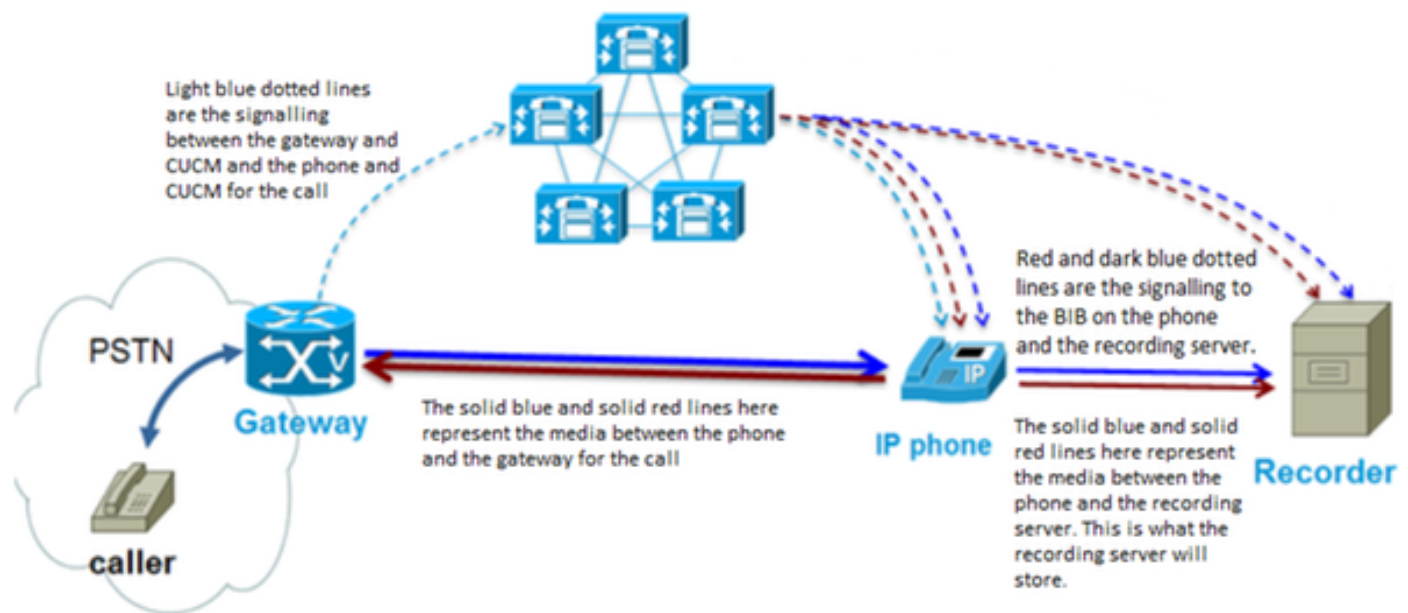
Types of Call Recording

Automatic

The key elements of automatic call recording are below:

- Uses Built-in-Bridge of IP phone to “fork” audio to the recording destination
- Initiated every time the IP phone places a call or receives a call
- Requires only a SIP trunk between CUCM and recording destination. Some recording vendors require CTI (Computer telephony integration) integration
- Does not allow recording of phones that are located outside of the managed network (must have access to send RTP directly to recording server and be a Cisco IP phone capable of allocating a Built-in-Bridge)

In the diagram below the solid lines represent the expected media flow and the dashed lines represent the expected signaling flow:

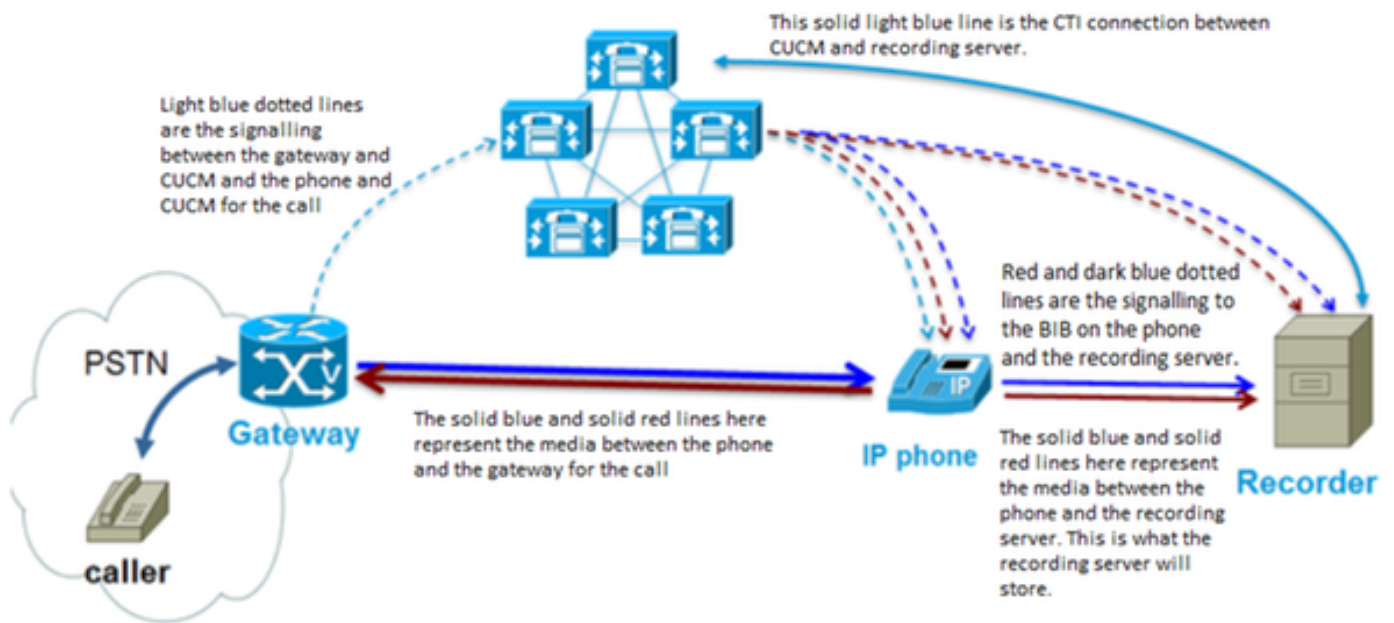


Application Invoked

The key elements of application invoked call recording are below:

- Uses Built-in-Bridge of IP phone to “fork” audio to the recording destination
- Initiated when the application (recorder) dictates that it should be initiated
- Requires SIP trunk and CTI integration with recording application
- CTI application user must have access to endpoints that need to be recorded
- Does not allow recording of phones that are located outside of the managed network (must have access to send RTP directly to recording server)

In the diagram below the solid lines represent the expected media flow and the dashed lines represent the expected signaling flow. The solid line between CUCM and the recording server denotes a CTI connection between CUCM and the application.

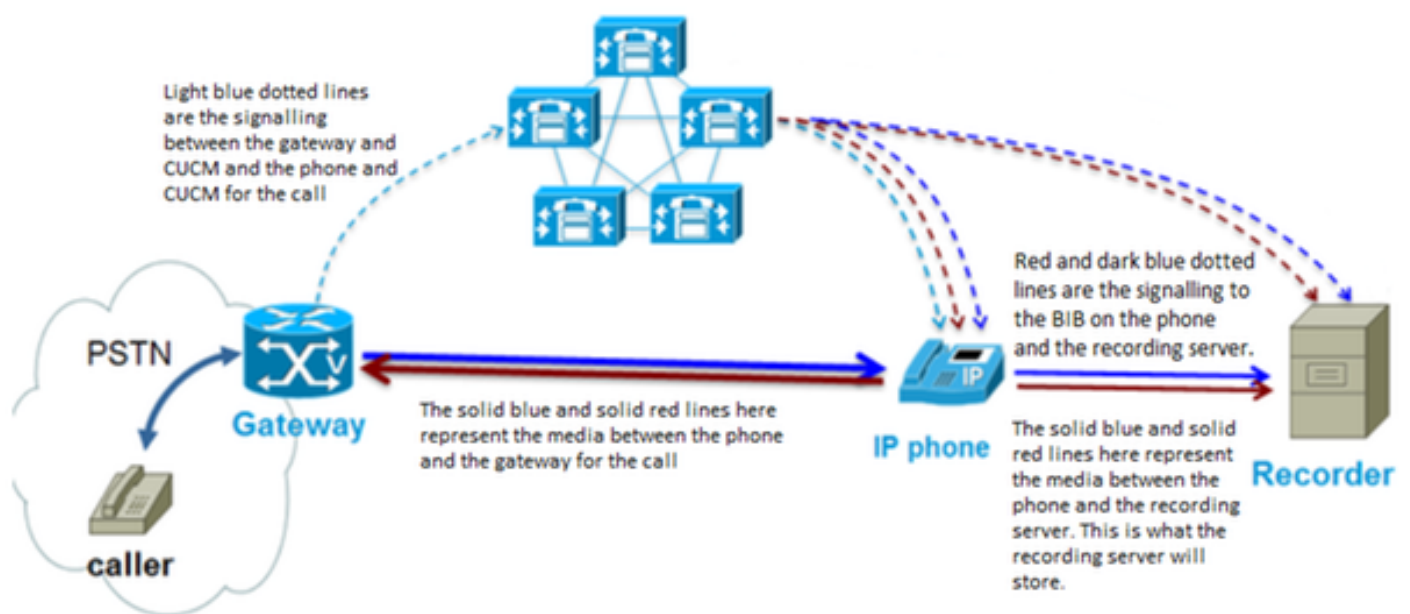


Selective

The key elements of selective call recording are below:

- Uses Built-in-Bridge of IP phone to “fork” audio to the recording destination
- Initiated every time the IP phone user selects the recording option on their IP phone (CUCM 9.x+) or on an application like in [this image](#)
- Typically requires only a SIP trunk between CUCM and recording destination (depending on recording application vendor)
- Does not allow recording of phones that lie outside of the managed network (must have access to send RTP directly to recording server)

As you can see in the diagram below, the media and signaling path is very similar to automatic call recording:

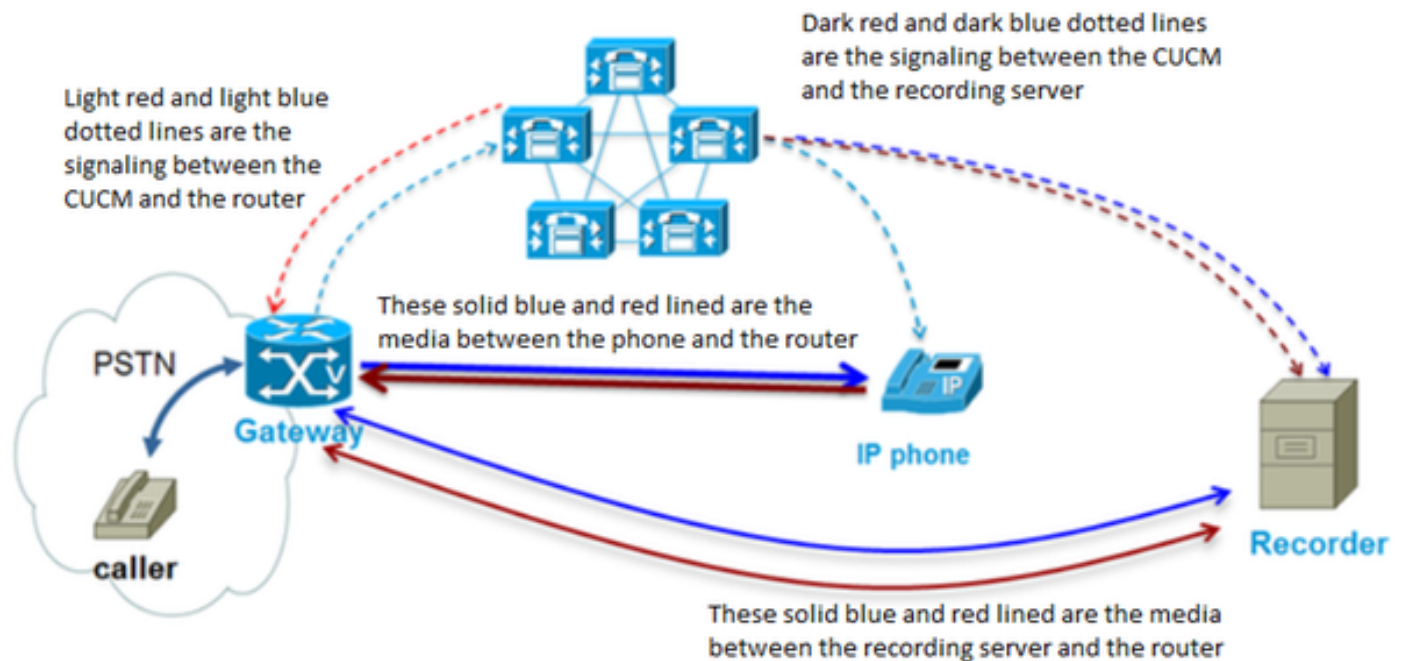


Gateway-based

The key elements of gateway-based call recording are below:

- Voice gateway forks the media towards the recording destination
- CUCM registers with gateway as an application
- CUCM uses HTTP to instruct GW to stream media to recording destination
- CUCM integrates with recording destination via SIP trunk
- Allows recording of calls that simply pass through managed network (for instance, to mobile users) or for phones that do not support the BiB

As you can see from the diagram below, the media flow is quite different from the other types of call recording:



Automatic Call Recording Configuration for SIP only Integration

This section describes how to setup the SIP integration of a recording server.


Create the SIP Trunk to the Recording Destination

- Under Device > Trunk, select add new
- Create a SIP trunk with the following settings:

Trunk Configuration



Status

 Status: Ready

Trunk Information

Trunk Type*

Device Protocol*

Trunk Service Type*

Next

- Input the appropriate Device Name, Device Pool, MRGL, SIP trunk security profile, and SIP profile
- The destination address configured will be the address of the recording application server. In the example below the recording server is 14.48.32.170

-SIP Information

Destination





Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	<input type="text" value="14.48.32.170"/>	<input type="text"/>	<input type="text" value="5060"/>


Create the Recording Profile

- Under Device > Device Settings > Recording Profile
- Recording destination address is where the recording calls will be sent

Recording Profile Configuration

 Save
  Delete
  Copy
  Add New

Status

 Status: Ready

Recording Profile Information

Name*

Recording Calling Search Space

Recording Destination Address *

Create the Route Pattern to Route the Recording Calls

- Create a route pattern that matches the recording destination address configured in the previous step
- You can point to a route list instead of directly at the SIP trunk, if you wish to configure redundant SIP trunks

Please note that the the partition assigned to this Route Pattern must be associated with the **Recording Calling Search Space**.

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

Route Class*

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

Route Option Route this pattern

Assign Recording Profile to Phone line

- On an already created phone with an existing extension, assign the recording profile created
- Assign the type of call recording in this location as well
- This example shows automatic recording

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

Set the BIB to on and Privacy to off on the Phone Configuration page

While on the device configuration page navigate to the section titled **Device Information**. Set Built In Bridge to on and Privacy to off.

Built In Bridge*	On
Privacy*	Off

Verify

The below are the expected behaviors in the CallManager traces for SCCP and SIP phones given the above configuration. These examples are for a phone calling another phone on the same cluster while one of the phones is set up for call recording.

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(14.48.32.33). myIP: IpAddr.type:0 ipv4Addr:0x0e30201c(14.48.32.28)### 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(14.48.32.28). myIP: IpAddr.type:0 ipv4Addr:0x0e302021(14.48.32.33)### 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)
|2,100,14,8384.91^14.48.32.33^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 (14.48.32.170)

03797320.001 |20:21:09.343 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message
to 14.48.32.170:[5060]:
[212231,NET]
INVITE sip:8675309@14.48.32.170:5060 SIP/2.0
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hg4bk204d520fedb3
From: <sip:9110001@14.48.32.90;x-nearend;x-refci=38960750;x-nearendclusterid=glencucm10-5;x-
nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=38960749;x-
farendclusterid=glencucm10-5;x-farenddevice=sep0018195aa209;x-
farendaddr=9110006>;tag=73601~713e2333-4032-45f1-b1f5-e33cf471acec-38960754
To: <sip:8675309@14.48.32.170>
Date: Tue, 30 Sep 2014 00:21:09 GMT
Call-ID: abbb8e00-4291f775-204c-5a20300e@14.48.32.90
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@14.48.32.90>
Remote-Party-ID: <sip:9110001@14.48.32.90>;party=calling;screen=yes;privacy=off
Contact: <sip:9110001@14.48.32.90:5060>;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 14.48.32.170:[5060]:
[212232,NET]
SIP/2.0 200 OK
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK204d520fedb3
From: <sip:9110001@14.48.32.90;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-e33cf471lacec-38960754
To: <sip:8675309@14.48.32.170>;tag=1
Call-ID: abbb8e00-4291f775-204c-5a20300e@14.48.32.90
CSeq: 101 INVITE
Contact: <sip:14.48.32.170:5060;transport=udp>
Content-Type: application/sdp
Content-Length: 135

v=0
o=user1 53655765 2353687637 IN IP4 14.48.32.170
s=-
c=IN IP4 14.48.32.170
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000 ### CUCM sends INVITE #2 to recording server (14.48.32.170)

03797445.001 |20:21:09.348 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 14.48.32.170:[5060]:

[212233,NET]

INVITE sip:8675309@14.48.32.170:5060 SIP/2.0

Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK204e754eaeae

From: <sip:9110001@14.48.32.90;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-e33cf471acec-38960757

To: <sip:8675309@14.48.32.170>

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

Call-ID: abbb8e00-4291f775-204d-5a20300e@14.48.32.90

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@14.48.32.90>

Remote-Party-ID: <sip:9110001@14.48.32.90>;party=calling;screen=yes;privacy=off

Contact: <sip:9110001@14.48.32.90:5060>;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 14.48.32.170:[5060]:

[212235,NET]

SIP/2.0 200 OK

Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK204e754eaeae

From: <sip:9110001@14.48.32.90;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-e33cf471acec-38960757

To: <sip:8675309@14.48.32.170>;tag=2

Call-ID: abbb8e00-4291f775-204d-5a20300e@14.48.32.90

CSeq: 101 INVITE

Contact: <sip:14.48.32.170:5060;transport=udp>

Content-Type: application/sdp

Content-Length: 135

v=0

o=user1 53655765 2353687637 IN IP4 14.48.32.170

s=-

c=IN IP4 14.48.32.170

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 14.48.32.170:[5060]:

[212236,NET]

ACK sip:14.48.32.170:5060;transport=UDP SIP/2.0

Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK204f50bef815

From: <sip:9110001@14.48.32.90;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@14.48.32.170>;tag=1

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

Call-ID: abbb8e00-4291f775-204c-5a20300e@14.48.32.90
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 14.48.32.90
s=SIP Call
c=IN IP4 14.48.32.33
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 (14.48.32.170)

03797479.001 |20:21:09.350 |AppInfo |StationD: (0000114) startMediaTransmission
conferenceID=38960750 passThruPartyID=33554452 remoteIpAddress=IpAddr.type:0
ipAddr:0x0e3020aa000000000000000000000000(14.48.32.170) remotePortNumber=6000
milliSecondPacketSize=20 compressType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0
qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e302021(14.48.32.33)### CUCM sends
startMediaTransmission #2 to the called (recorded) phone telling the phone to send RTP to
recording server (14.48.32.170)

03797596.001 |20:21:09.354 |AppInfo |StationD: (0000114) startMediaTransmission
conferenceID=38960750 passThruPartyID=33554453 remoteIpAddress=IpAddr.type:0
ipAddr:0x0e3020aa000000000000000000000000(14.48.32.170) remotePortNumber=6000
milliSecondPacketSize=20 compressType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0
qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e302021(14.48.32.33)### 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 14.48.32.170:[5060]:

[212237,NET]

ACK sip:14.48.32.170:5060;transport=UDP SIP/2.0

Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK2050183495f1

From: <sip:9110001@14.48.32.90;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-e33cf471acec-38960757

To: <sip:8675309@14.48.32.170>;tag=2

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

Call-ID: abbb8e00-4291f775-204d-5a20300e@14.48.32.90

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 14.48.32.90
s=SIP Call
c=IN IP4 14.48.32.33
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(14.48.32.28), Port=17996, PartyID=33554450### CUCM sends startMediaTransmission to the called (recorded) phone telling the phone to send RTP to the calling phone (14.48.32.28)

03797642.001 |20:21:09.385 |AppInfo |StationD: (0000114) startMediaTransmission conferenceID=38960750 passThruPartyID=33554451 remoteIpAddress=IpAddr.type:0 ipAddr:0x0e30201c000000000000000000000000(14.48.32.28) remotePortNumber=17996 milliSecondPacketSize=20 compressType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0 qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e302021(14.48.32.33)### 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(14.48.32.33), Port=32588, PartyID=33554451### CUCM sends startMediaTransmission to the calling phone telling the phone to send RTP to the called phone (14.48.32.33)

03797655.001 |20:21:09.454 |AppInfo |StationD: (0000109) startMediaTransmission conferenceID=38960749 passThruPartyID=33554450 remoteIpAddress=IpAddr.type:0 ipAddr:0x0e302021000000000000000000000000(14.48.32.33) remotePortNumber=32588 milliSecondPacketSize=20 compressType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0 qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e30201c(14.48.32.28)

SIP

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

Calling phone places call

01314118.001 |11:18:44.472 |AppInfo |StationInit: (0000004) EnblocCall calledParty=9110011.
CUCM performs digit analysis against the dialed digits (dd="9110011")

01314127.001 |11:18:44.473 |AppInfo |Digit Analysis: star_DaReq: daReq.partitionSearchSpace(), filteredPartitionSearchSpaceString(), partitionSearchSpaceString()
01314127.002 |11:18:44.473 |AppInfo |Digit Analysis: star_DaReq: Matching Legacy Numeric, digits=9110011
01314127.003 |11:18:44.499 |AppInfo |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept DAMR.sstype=[0], TPcount=[0], DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]
01314127.004 |11:18:44.499 |AppInfo |Digit Analysis: getDaRes - Remote Destination [] isURI[1]
01314127.005 |11:18:44.506 |AppInfo |Digit analysis: patternUsage=2
01314127.006 |11:18:44.506 |AppInfo |Digit analysis: match(pi="2", fqcn="9110006", cn="9110006", plv="5", pss="", TodFilteredPss="", dd="9110011", dac="1")
01314127.007 |11:18:44.506 |AppInfo |Digit analysis: analysis results
01314127.008 |11:18:44.506 |AppInfo ||PretransformCallingPartyNumber=9110006
|CallingPartyNumber=9110006
|DialingPartition=
|DialingPattern=9110011
|FullyQualifiedCalledPartyNumber=9110011
|DialingPatternRegularExpression=(9110011)
|DialingWhere=

|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=9110011
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=9110011
|CollectedDigits=9110011 ### CUCM determines call must stay on same node and go to LineControl
(PID=LineControl(2,100,174,19))

01314129.001 |11:18:44.506 |AppInfo |Digit analysis: wait_DmPidRes- Partition=[]
Pattern=[9110011] Where=[],cmDeviceType=[UserDevice], OutsideDialtone =[0], DeviceOverride=[0],
PID=LineControl(2,100,174,19),CI=[47601637],Sender=Cdcc(2,100,219,1)### CUCM sends outbound
INVITE to called (recorded) phone

01314173.001 |11:18:44.754 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to
14.48.32.17 on port 50841 index 17
[106316,NET]
INVITE sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK203b13880683
From: <sip:9110006@14.48.32.90>;tag=38244~713e2333-4032-45f1-b1f5-e33cf471acec-47601638
To: <sip:9110011@14.48.32.90>
Date: Tue, 14 Oct 2014 15:18:44 GMT
Call-ID: 6198e780-43d13ed4-203c-5a20300e@14.48.32.90
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
Call-Info: ; security= Unknown; orientation= from; gci= 2-6001; isVoip; call-instance= 1
Send-Info: conference, x-cisco-conference
Alert-Info:
Remote-Party-ID: <sip:9110006@14.48.32.90;x-cisco-callback-
number=9110006>;party=calling;screen=yes;privacy=off
Contact: <sip:9110006@14.48.32.90:5060;transport=tcp>
Max-Forwards: 70
Content-Length: 0 ### Called (recorded) phone returns 100 Trying

01314174.002 |11:18:44.758 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from
14.48.32.17 on port 50841 index 17 with 802 bytes:
[106317,NET]
SIP/2.0 100 Trying
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK203b13880683
From: <sip:9110006@14.48.32.90>;tag=38244~713e2333-4032-45f1-b1f5-e33cf471acec-47601638
To: <sip:9110011@14.48.32.90>
Call-ID: 6198e780-43d13ed4-203c-5a20300e@14.48.32.90
Date: Tue, 14 Oct 2014 15:18:51 GMT
CSeq: 101 INVITE
Server: Cisco-CP8841/10.2.1
Contact: <sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
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
Content-Length: 0 ### Called (recorded) phone returns 180 Ringing

01314178.002 |11:18:45.357 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from
14.48.32.17 on port 50841 index 17 with 950 bytes:
[106318,NET]
SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK203b13880683

From: <sip:9110006@14.48.32.90>;tag=38244~713e2333-4032-45f1-b1f5-e33cf471acec-47601638
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bba73e445ee-3cc7e650
Call-ID: 6198e780-43d13ed4-203c-5a20300e@14.48.32.90
Date: Tue, 14 Oct 2014 15:18:51 GMT
CSeq: 101 INVITE
Server: Cisco-CP8841/10.2.1
Contact: <sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "9110011" <sip:9110011@14.48.32.90>;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
Content-Length: 0 ### Called (recorded) phone returns 200 OK

01314217.002 |11:18:48.466 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from
14.48.32.17 on port 50841 index 17 with 1430 bytes:

[106319,NET]

SIP/2.0 200 OK

Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK203b13880683

From: <sip:9110006@14.48.32.90>;tag=38244~713e2333-4032-45f1-b1f5-e33cf471acec-47601638

To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bba73e445ee-3cc7e650

Call-ID: 6198e780-43d13ed4-203c-5a20300e@14.48.32.90

Date: Tue, 14 Oct 2014 15:18:54 GMT

CSeq: 101 INVITE

Server: Cisco-CP8841/10.2.1

Contact: <sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp>

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

Remote-Party-ID: "9110011" <sip:9110011@14.48.32.90>;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

Content-Length: 404

Content-Type: application/sdp

Content-Disposition: session;handling=optional

v=0

o=Cisco-SIPUA 15076 0 IN IP4 14.48.32.17

s=SIP Call

t=0 0

m=audio 28354 RTP/AVP 0 8 18 102 9 116 124 101

c=IN IP4 14.48.32.17

a=rtpmap:0 PCMU/8000

a=rtpmap:8 PCMA/8000

a=rtpmap:18 G729/8000

a=fmtp:18 annexb=yes

a=rtpmap:102 L16/16000

a=rtpmap:9 G722/8000

a=rtpmap:116 iLBC/8000

a=fmtp:116 mode=20

a=rtpmap:124 ISAC/16000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15

a=sendrecv ### CUCM Tells the calling phone to open the logical channel

01314284.001 |11:18:48.599 |AppInfo |StationD: (0000004) SEP0018195AA209 ,
star_MediaExchangeAgenaOpenLogicalChannel packetSize=20, codec=4, ci=47601637### CUCM Tells the
calling phone to open the receive channel

01314294.002 |11:18:48.599 |AppInfo |StationD: (0000004) OpenReceiveChannel

conferenceID=47601637 passThruPartyID=33554433 millisecondPacketSize=20
compressionType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=101 qualifierIn=?
sourceIpAddr=IpAddr.type:0 ipAddr:0x0e302011000000000000000000000000(14.48.32.17). myIP:
IpAddr.type:0 ipv4Addr:0x0e30201c(14.48.32.28)### CUCM sends startMediaTransmission to the
calling phone telling the phone to send RTP to the called (recorded) phone (14.48.32.17)

01314295.001 |11:18:48.599 |AppInfo |StationD: (0000004) startMediaTransmission
conferenceID=47601637 passThruPartyID=33554433 remoteIpAddress=IpAddr.type:0
ipAddr:0x0e302011000000000000000000000000(14.48.32.17) remotePortNumber=28354
milliSecondPacketSize=20 compressType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=101
qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e30201c(14.48.32.28)### CUCM sends ACK to called
(recorded) phone telling the called phone to send media to the calling phone (14.48.32.28)

01314344.001 |11:18:48.652 |AppInfo |SIPTcp - wait_SdlsPISignal: Outgoing SIP TCP message to
14.48.32.17 on port 50841 index 17
[106320,NET]

ACK sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK203c2831c118
From: <sip:9110006@14.48.32.90>;tag=38244~713e2333-4032-45f1-b1f5-e33cf471acec-47601638
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bba73e445ee-3cc7e650
Date: Tue, 14 Oct 2014 15:18:44 GMT
Call-ID: 6198e780-43d13ed4-203c-5a20300e@14.48.32.90
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 14.48.32.90
s=SIP Call
c=IN IP4 14.48.32.28
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 |SdlsSig |MrmAllocateUcbResourceReq |waiting
|MediaResourceManager(2,100,138,1) |Cc(2,100,220,1)
|2,100,14,20.16735^14.48.32.28^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 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 to tear down the media

01314446.001 |11:18:48.682 |AppInfo |SIPTcp - wait_SdlsPISignal: Outgoing SIP TCP message to
14.48.32.17 on port 50841 index 17
[106321,NET]

INVITE sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK203d55363a7c
From: "Call Manager" ;tag=38246~713e2333-4032-45f1-b1f5-e33cf471acec-47601641
To: <sip:9110011@14.48.32.90>
Date: Tue, 14 Oct 2014 15:18:48 GMT
Call-ID: 63fb4180-43d13ed8-203d-5a20300e@14.48.32.90
Supported: timer,resource-priority,replaces

Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Call-Info: ; isVoip; record-invoker=auto
Join: 6198e780-43d13ed4-203c-5a20300e@14.48.32.90;from-tag=b000b4d9e8cb0bba73e445ee-3cc7e650;to-tag=38244~713e2333-4032-45f1-b1f5-e33cf471acec-47601638
Contact: <sip:14.48.32.90:5060;transport=tcp>
Remote-Party-ID: "Call Manager" ;party=calling;screen=yes;privacy=off
Max-Forwards: 70
Content-Type: application/sdp
Content-Length: 188

v=0
o=CiscoSystemsCCM-SIP 38246 1 IN IP4 14.48.32.90
s=SIP Call
c=IN IP4 14.48.32.90
t=0 0
m=audio 4000 RTP/AVP 0
a=label:X-relay-nearend
a=rtpmap:0 PCMU/8000
a=inactive
a=mid:1 ### Called (recorded) phone returns 200 OK
Notice the SDP has a=inactive to tear down the media

01314449.002 |11:18:48.702 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 14.48.32.17 on port 50841 index 17 with 1235 bytes:
[106323,NET]
SIP/2.0 200 OK
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK203d55363a7c
From: "Call Manager" ;tag=38246~713e2333-4032-45f1-b1f5-e33cf471acec-47601641
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bbb4457e725-6869188a
Call-ID: 63fb4180-43d13ed8-203d-5a20300e@14.48.32.90
Date: Tue, 14 Oct 2014 15:18:55 GMT
CSeq: 101 INVITE
Server: Cisco-CP8841/10.2.1
Contact: <sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp>
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE, SUBSCRIBE, INFO
Remote-Party-ID: "9110011" <sip:9110011@14.48.32.90>;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
Content-Length: 202
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 4077 0 IN IP4 14.48.32.17
s=SIP Call
t=0 0
m=audio 28512 RTP/AVP 0 101
c=IN IP4 14.48.32.17
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

01314452.001 |11:18:48.702 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 14.48.32.17 on port 50841 index 17

[106324,NET]

ACK sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp SIP/2.0

Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK203e9999fc7

From: "Call Manager" ;tag=38246~713e2333-4032-45f1-b1f5-e33cf471acec-47601641

To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bbb4457e725-6869188a

Date: Tue, 14 Oct 2014 15:18:48 GMT

Call-ID: 63fb4180-43d13ed8-203d-5a20300e@14.48.32.90

User-Agent: Cisco-CUCM10.5

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="b0028310001" and it is dialing the recordingdestination dd="8675309")

01314484.003 |11:18:48.753 |AppInfo |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept

DAMR.sstype=[0], TPcount=[0], DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]

01314484.004 |11:18:48.753 |AppInfo |Digit Analysis: getDaRes - Remote Destination [8675309]

isURI[0]

01314484.005 |11:18:48.765 |AppInfo |CMUtility routeCallThroughCTIRD: no matching

RemDestDynamic record exists for remdest [8675309]

01314484.006 |11:18:48.765 |AppInfo |DbMobility: getMatchedRemDest starts: cnumber = 8675309

01314484.007 |11:18:48.765 |AppInfo |DbMobility: getMatchedRemDest: full match case

01314484.008 |11:18:48.765 |AppInfo |DbMobility SelectByDestination: no matching RemDestDynamic

record exists for remdest [8675309]

01314484.009 |11:18:48.765 |AppInfo |DbMobility: can't find remdest 8675309 in map

01314484.010 |11:18:48.765 |AppInfo |Digit analysis: patternUsage=5

01314484.011 |11:18:48.765 |AppInfo |Digit analysis: match(pi="1", fqcn="",

cn="b0028310001", 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")

01314484.012 |11:18:48.765 |AppInfo |Digit analysis: analysis results

01314484.013 |11:18:48.765 |AppInfo ||PretransformCallingPartyNumber=b0028310001

|CallingPartyNumber=b0028310001

|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 (14.48.32.170)

01314552.001 |11:18:48.795 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message

to 14.48.32.170:[5060]:

[106325,NET]

INVITE sip:8675309@14.48.32.170:5060 SIP/2.0

Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK203f3135e715

From: <sip:9110011@14.48.32.90;x-nearend;x-refci=47601638;x-nearendclusterid=glenscucm10-5;x-

nearenddevice=sepb000b4d9e8cb;x-nearendaddr=9110011;x-farendrefci=47601637;x-

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

farendaddr=9110006>;tag=38248~713e2333-4032-45f1-b1f5-e33cf471acec-47601642

To: <sip:8675309@14.48.32.170>

Date: Tue, 14 Oct 2014 15:18:48 GMT

Call-ID: 63fb4180-43d13ed8-203e-5a20300e@14.48.32.90

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"
Call-Info: ;x-cisco-video-traffic-class=DESKTOP
Cisco-Guid: 1677410688-0000065536-0000000001-1512058894
Session-Expires: 1800
P-Asserted-Identity: <sip:9110011@14.48.32.90>
Remote-Party-ID: <sip:9110011@14.48.32.90>;party=calling;screen=yes;privacy=off
Contact: <sip:9110011@14.48.32.90:5060>;isFocus
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 to tear down the media

01314575.001 |11:18:48.796 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to
14.48.32.17 on port 50841 index 17
[106326,NET]

INVITE sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK20401b237b36
From: "Call Manager" ;tag=38249~713e2333-4032-45f1-b1f5-e33cf471acec-47601644
To: <sip:9110011@14.48.32.90>
Date: Tue, 14 Oct 2014 15:18:48 GMT
Call-ID: 63fb4180-43d13ed8-203f-5a20300e@14.48.32.90
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180

Allow-Events: presence
Call-Info: ; isVoip; record-invoker=auto
Join: 6198e780-43d13ed4-203c-5a20300e@14.48.32.90;from-tag=b000b4d9e8cb0bba73e445ee-3cc7e650;to-
tag=38244~713e2333-4032-45f1-b1f5-e33cf471acec-47601638
Contact: <sip:14.48.32.90:5060;transport=tcp>
Remote-Party-ID: "Call Manager" ;party=calling;screen=yes;privacy=off
Max-Forwards: 70
Content-Type: application/sdp
Content-Length: 187

v=0
o=CiscoSystemsCCM-SIP 38249 1 IN IP4 14.48.32.90
s=SIP Call
c=IN IP4 14.48.32.90
t=0 0
m=audio 4000 RTP/AVP 0
a=label:X-relay-farend
a=rtpmap:0 PCMU/8000
a=inactive
a=mid:1 ### CUCM receives 200 OK in response to INVITE #1 to recording server

01314583.001 |11:18:48.862 |AppInfo |//SIP/SIPUdp/wait_SdlDataInd: Incoming SIP UDP message
size 737 from 14.48.32.170:[5060]:
[106328,NET]
SIP/2.0 200 OK
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK203f3135e715
From: <sip:9110011@14.48.32.90;x-nearend;x-refci=47601638;x-nearendclusterid=glenscucm10-5;x-
nearenddevice=sepb000b4d9e8cb;x-nearendaddr=9110011;x-farendrefci=47601637;x-
farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-
farendaddr=9110006>;tag=38248~713e2333-4032-45f1-b1f5-e33cf471acec-47601642
To: <sip:8675309@14.48.32.170>;tag=1

Call-ID: 63fb4180-43d13ed8-203e-5a20300e@14.48.32.90
CSeq: 101 INVITE
Contact: <sip:14.48.32.170:5060;transport=udp>
Content-Type: application/sdp
Content-Length: 135

v=0
o=user1 53655765 2353687637 IN IP4 14.48.32.170
s=-
c=IN IP4 14.48.32.170
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000 ### CUCM sends re-INVITE to called (recorded) phone for call #1 to invoke the BiB (notice there is no SDP)

01314644.001 |11:18:48.864 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 14.48.32.17 on port 50841 index 17

[106329,NET]
INVITE sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK204176d717cd
From: "Call Manager" ;tag=38246~713e2333-4032-45f1-b1f5-e33cf471acec-47601641
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bbb4457e725-6869188a
Date: Tue, 14 Oct 2014 15:18:48 GMT
Call-ID: 63fb4180-43d13ed8-203d-5a20300e@14.48.32.90
Supported: timer,resource-priority,replaces
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, SUBSCRIBE, NOTIFY
CSeq: 102 INVITE
Max-Forwards: 70
Expires: 180
Allow-Events: presence
Call-Info: ; isVoip; record-invoker=auto
Min-SE: 1800
Remote-Party-ID: "Call Manager" ;party=calling;screen=yes;privacy=off
Contact: <sip:14.48.32.90:5060;transport=tcp>
Content-Length: 0 ### Called (recorded) phone returns 200 OK in response to INVITE #2 to invoke BiB
Notice the SDP has a=inactive to tear down the media

01314645.002 |11:18:48.865 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 14.48.32.17 on port 50841 index 17 with 1236 bytes:

[106330,NET]
SIP/2.0 200 OK
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK20401b237b36
From: "Call Manager" ;tag=38249~713e2333-4032-45f1-b1f5-e33cf471acec-47601644
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bbc4d5b7fc6-3ab2172f
Call-ID: 63fb4180-43d13ed8-203f-5a20300e@14.48.32.90
Date: Tue, 14 Oct 2014 15:18:55 GMT
CSeq: 101 INVITE
Server: Cisco-CP8841/10.2.1
Contact: <sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "9110011" <sip:9110011@14.48.32.90>;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
Content-Length: 203
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0

o=Cisco-SIPUA 11326 0 IN IP4 14.48.32.17
s=SIP Call
t=0 0
m=audio 19696 RTP/AVP 0 101
c=IN IP4 14.48.32.17
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

01314648.001 |11:18:48.866 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 14.48.32.17 on port 50841 index 17 [106331,NET]
ACK sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK20424175effe
From: "Call Manager" ;tag=38249~713e2333-4032-45f1-b1f5-e33cf471acec-47601644
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bbc4d5b7fc6-3ab2172f
Date: Tue, 14 Oct 2014 15:18:48 GMT
Call-ID: 63fb4180-43d13ed8-203f-5a20300e@14.48.32.90
User-Agent: Cisco-CUCM10.5
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="b0028310001" and it is dialing the recordingdestination dd="8675309") Note that the BiB number stayed the same (b0028310001) and so did the recordingdestination number

01314680.003 |11:18:48.867 |AppInfo |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept DAMR.sstype=[0], TPcount=[0], DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]
01314680.004 |11:18:48.867 |AppInfo |Digit Analysis: getDaRes - Remote Destination [8675309] isURI[0]
01314680.005 |11:18:48.867 |AppInfo |CMUtility routeCallThroughCTIRD: no matching RemDestDynamic record exists for remdest [8675309]
01314680.006 |11:18:48.867 |AppInfo |DbMobility: getMatchedRemDest starts: cnumber = 8675309
01314680.007 |11:18:48.867 |AppInfo |DbMobility: getMatchedRemDest: full match case
01314680.008 |11:18:48.867 |AppInfo |DbMobility SelectByDestination: no matching RemDestDynamic record exists for remdest [8675309]
01314680.009 |11:18:48.867 |AppInfo |DbMobility: can't find remdest 8675309 in map
01314680.010 |11:18:48.867 |AppInfo |Digit analysis: patternUsage=5
01314680.011 |11:18:48.867 |AppInfo |Digit analysis: match(pi="1", fqcn="", cn="b0028310001", 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")
01314680.012 |11:18:48.867 |AppInfo |Digit analysis: analysis results
01314680.013 |11:18:48.867 |AppInfo ||PretransformCallingPartyNumber=b0028310001
|CallingPartyNumber=b0028310001
|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 #2 to configured recording server

01314731.001 |11:18:48.870 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 14.48.32.170:[5060]: [106333,NET]

INVITE sip:8675309@14.48.32.170:5060 SIP/2.0
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK20432a53d34c
From: <sip:9110011@14.48.32.90;x-farend;x-refci=47601638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sepb000b4d9e8cb;x-nearendaddr=9110011;x-farendrefci=47601637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=38251~713e2333-4032-45f1-b1f5-e33cf471acec-47601645
To: <sip:8675309@14.48.32.170>
Date: Tue, 14 Oct 2014 15:18:48 GMT
Call-ID: 63fb4180-43d13ed8-2040-5a20300e@14.48.32.90
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"
Call-Info: ;x-cisco-video-traffic-class=DESKTOP
Cisco-Guid: 1677410688-0000065536-0000000002-1512058894
Session-Expires: 1800
P-Asserted-Identity: <sip:9110011@14.48.32.90>
Remote-Party-ID: <sip:9110011@14.48.32.90>;party=calling;screen=yes;privacy=off
Contact: <sip:9110011@14.48.32.90:5060>;isFocus
Max-Forwards: 70
Content-Length: 0 ### CUCM receives 200 OK in response to INVITE #2 from configured recording server

01314751.001 |11:18:48.871 |AppInfo |//SIP/SIPUdp/wait_SdlDataInd: Incoming SIP UDP message size 736 from 14.48.32.170:[5060]:
[106335,NET]
SIP/2.0 200 OK
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK20432a53d34c
From: <sip:9110011@14.48.32.90;x-farend;x-refci=47601638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sepb000b4d9e8cb;x-nearendaddr=9110011;x-farendrefci=47601637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=38251~713e2333-4032-45f1-b1f5-e33cf471acec-47601645
To: <sip:8675309@14.48.32.170>;tag=2
Call-ID: 63fb4180-43d13ed8-2040-5a20300e@14.48.32.90
CSeq: 101 INVITE
Contact: <sip:14.48.32.170:5060;transport=udp>
Content-Type: application/sdp
Content-Length: 135

v=0
o=user1 53655765 2353687637 IN IP4 14.48.32.170
s=-
c=IN IP4 14.48.32.170
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000 ### CUCM sends re-INVITE #2 to called (recorded) phone for second BiB invocation call
Notice there is no SDP

01314828.001 |11:18:48.875 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 14.48.32.17 on port 50841 index 17
[106336,NET]
INVITE sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK20443475e621
From: "Call Manager" ;tag=38249~713e2333-4032-45f1-b1f5-e33cf471acec-47601644
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bbc4d5b7fc6-3ab2172f
Date: Tue, 14 Oct 2014 15:18:48 GMT

Call-ID: 63fb4180-43d13ed8-203f-5a20300e@14.48.32.90
Supported: timer,resource-priority,replaces
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, SUBSCRIBE, NOTIFY
CSeq: 102 INVITE
Max-Forwards: 70
Expires: 180
Allow-Events: presence
Call-Info: ; isVoip; record-invoker=auto
Min-SE: 1800
Remote-Party-ID: "Call Manager" ;party=calling;screen=yes;privacy=off
Contact: <sip:14.48.32.90:5060;transport=tcp>
Content-Length: 0 ### Called (recorded) phone returns 200 OK to re-INVITE #1

01314829.002 |11:18:48.876 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from
14.48.32.17 on port 50841 index 17 with 1235 bytes:
[106337,NET]
SIP/2.0 200 OK
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK204176d717cd
From: "Call Manager" ;tag=38246~713e2333-4032-45f1-b1f5-e33cf471acec-47601641
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bbb4457e725-6869188a
Call-ID: 63fb4180-43d13ed8-203d-5a20300e@14.48.32.90
Date: Tue, 14 Oct 2014 15:18:55 GMT
CSeq: 102 INVITE
Server: Cisco-CP8841/10.2.1
Contact: <sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "9110011" <sip:9110011@14.48.32.90>;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
Content-Length: 202
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 4077 1 IN IP4 14.48.32.17
s=SIP Call
t=0 0
m=audio 28512 RTP/AVP 0 101
c=IN IP4 14.48.32.17
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 #1

01314873.001 |11:18:48.880 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to
14.48.32.17 on port 50841 index 17
[106338,NET]
ACK sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK204521531f4b
From: "Call Manager" ;tag=38246~713e2333-4032-45f1-b1f5-e33cf471acec-47601641
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bbb4457e725-6869188a
Date: Tue, 14 Oct 2014 15:18:48 GMT
Call-ID: 63fb4180-43d13ed8-203d-5a20300e@14.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 102 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 178

v=0
o=CiscoSystemsCCM-SIP 38246 3 IN IP4 14.48.32.90
s=SIP Call
c=IN IP4 14.48.32.170
b=TIAS:64000
b=AS:64
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=recvonly### CUCM sends ACK to configured recording server for INVITE #1

01314875.001 |11:18:48.880 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 14.48.32.170:[5060]:

[106339,NET]
ACK sip:14.48.32.170:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK20467ee6be7
From: <sip:9110011@14.48.32.90;x-nearend;x-refci=47601638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sepb000b4d9e8cb;x-nearendaddr=9110011;x-farendrefci=47601637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=38248-713e2333-4032-45f1-b1f5-e33cf471acec-47601642
To: <sip:8675309@14.48.32.170>;tag=1
Date: Tue, 14 Oct 2014 15:18:48 GMT
Call-ID: 63fb4180-43d13ed8-203e-5a20300e@14.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Type: application/sdp
Content-Length: 234

v=0
o=CiscoSystemsCCM-SIP 38248 1 IN IP4 14.48.32.90
s=SIP Call
c=IN IP4 14.48.32.17
b=TIAS:64000
b=AS:64
t=0 0
m=audio 28512 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=sendonly
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15 ### Called (recorded) phone returns 200 OK for re-INVITE #2

01314878.005 |11:18:48.881 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 14.48.32.17 on port 50841 index 17 with 1236 bytes:

[106341,NET]
SIP/2.0 200 OK
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK20443475e621
From: "Call Manager" ;tag=38249~713e2333-4032-45f1-b1f5-e33cf471acec-47601644
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bbc4d5b7fc6-3ab2172f
Call-ID: 63fb4180-43d13ed8-203f-5a20300e@14.48.32.90
Date: Tue, 14 Oct 2014 15:18:55 GMT
CSeq: 102 INVITE
Server: Cisco-CP8841/10.2.1
Contact: <sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "9110011" <sip:9110011@14.48.32.90>;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

Content-Length: 203
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 11326 1 IN IP4 14.48.32.17
s=SIP Call
t=0 0
m=audio 19696 RTP/AVP 0 101
c=IN IP4 14.48.32.17
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

01314907.001 |11:18:48.883 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 14.48.32.17 on port 50841 index 17
[106342,NET]
ACK sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK204755ae79c7
From: "Call Manager" ;tag=38249~713e2333-4032-45f1-b1f5-e33cf471acec-47601644
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bbc4d5b7fc6-3ab2172f
Date: Tue, 14 Oct 2014 15:18:48 GMT
Call-ID: 63fb4180-43d13ed8-203f-5a20300e@14.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 102 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 178

v=0
o=CiscoSystemsCCM-SIP 38249 3 IN IP4 14.48.32.90
s=SIP Call
c=IN IP4 14.48.32.170
b=TIAS:64000
b=AS:64
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=recvonly ### CUCM sends ACK to configured recording server for INVITE #2

01314909.001 |11:18:48.883 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 14.48.32.170:[5060]:
[106343,NET]
ACK sip:14.48.32.170:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK204854e1b53f
From: <sip:9110011@14.48.32.90;x-farend;x-refci=47601638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sepb000b4d9e8cb;x-nearendaddr=9110011;x-farendrefci=47601637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=38251~713e2333-4032-45f1-b1f5-e33cf471acec-47601645
To: <sip:8675309@14.48.32.170>;tag=2
Date: Tue, 14 Oct 2014 15:18:48 GMT
Call-ID: 63fb4180-43d13ed8-2040-5a20300e@14.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Type: application/sdp
Content-Length: 234

v=0
o=CiscoSystemsCCM-SIP 38251 1 IN IP4 14.48.32.90

```
s=SIP Call
c=IN IP4 14.48.32.17
b=TIAS:64000
b=AS:64
t=0 0
m=audio 19696 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=sendonly
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

Troubleshoot

Codec Negotiation

The below is an example of one of the most common type of call recording failures - codec mismatch between the recorded phone and the recording server:

```
~~~~~
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(14.48.32.33). myIP: IpAddr.type:0 ipv4Addr:0x0e30201c(14.48.32.28)### 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(14.48.32.28). myIP: IpAddr.type:0 ipv4Addr:0x0e302021(14.48.32.33)### 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)
|2,100,14,19.206^14.48.32.33^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(14.48.32.28), Port=31678, PartyID=33554433### CUCM sends startMediaTransmission to the called (recorded) phone telling the phone to send RTP to the calling phone (14.48.32.28)

00019920.001 |12:48:36.139 |AppInfo |StationD: (0000007) startMediaTransmission conferenceID=49613638 passThruPartyID=33554434 remoteIpAddress=IpAddr.type:0 ipAddr:0x0e30201c000000000000000000000000(14.48.32.28) remotePortNumber=31678 milliSecondPacketSize=20 compressType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0 qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e302021(14.48.32.33)### 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(14.48.32.33), Port=28360, PartyID=33554434### CUCM sends startMediaTransmission to the calling phone telling the phone to send RTP to the called phone (14.48.32.33)

00019977.001 |12:48:36.146 |AppInfo |StationD: (0000005) startMediaTransmission conferenceID=49613637 passThruPartyID=33554433 remoteIpAddress=IpAddr.type:0 ipAddr:0x0e302021000000000000000000000000(14.48.32.33) remotePortNumber=28360 milliSecondPacketSize=20 compressType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0 qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e30201c(14.48.32.28)### 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 (14.48.32.170)

00020086.001 |12:48:36.156 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message
to 14.48.32.170:[5060]:

[901,NET]

INVITE sip:8675309@14.48.32.170:5060 SIP/2.0

Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK4f2a857d3d

From: <sip:9110001@14.48.32.90;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@14.48.32.170>

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

Call-ID: ef7acf80-43d153e4-50-5a20300e@14.48.32.90

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@14.48.32.90>

Remote-Party-ID: <sip:9110001@14.48.32.90>;party=calling;screen=yes;privacy=off

Contact: <sip:9110001@14.48.32.90:5060>;isFocus

Max-Forwards: 70

Content-Length: 0 ### CUCM sends INVITE #2 to configured recording server (14.48.32.170)

00020088.001 |12:48:36.157 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message
to 14.48.32.170:[5060]:

[902,NET]

INVITE sip:8675309@14.48.32.170:5060 SIP/2.0

Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK5014378d0b

From: <sip:9110001@14.48.32.90;x-farend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-
nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-
farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-

farendaddr=9110006>;tag=352-713e2333-4032-45f1-b1f5-e33cf471acec-49613645
To: <sip:8675309@14.48.32.170>
Date: Tue, 14 Oct 2014 16:48:36 GMT
Call-ID: ef7acf80-43d153e4-51-5a20300e@14.48.32.90
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@14.48.32.90>
Remote-Party-ID: <sip:9110001@14.48.32.90>;party=calling;screen=yes;privacy=off
Contact: <sip:9110001@14.48.32.90:5060>;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 14.48.32.170:[5060]:

[903,NET]

SIP/2.0 200 OK

Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK4f2a857d3d

From: <sip:9110001@14.48.32.90;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@14.48.32.170>;tag=1

Call-ID: ef7acf80-43d153e4-50-5a20300e@14.48.32.90

CSeq: 101 INVITE

Contact: <sip:14.48.32.170:5060;transport=udp>

Content-Type: application/sdp

Content-Length: 135

v=0

o=user1 53655765 2353687637 IN IP4 14.48.32.170

s=-

c=IN IP4 14.48.32.170

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 14.48.32.170:[5060]:

[905,NET]

SIP/2.0 200 OK

Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK5014378d0b

From: <sip:9110001@14.48.32.90;x-farend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=352-713e2333-4032-45f1-b1f5-e33cf471acec-49613645

To: <sip:8675309@14.48.32.170>;tag=2

Call-ID: ef7acf80-43d153e4-51-5a20300e@14.48.32.90

CSeq: 101 INVITE

Contact: <sip:14.48.32.170:5060;transport=udp>

Content-Type: application/sdp

Content-Length: 135

v=0

o=user1 53655765 2353687637 IN IP4 14.48.32.170
s=-
c=IN IP4 14.48.32.170
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=**numXcoderRequired=2** xcodingSide=0 ### No transcoder is configured which will 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 14.48.32.170:[5060]:
[906,NET]
ACK sip:14.48.32.170:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK51257b2b47
From: <sip:9110001@14.48.32.90;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@14.48.32.170>;tag=1
Date: Tue, 14 Oct 2014 16:48:36 GMT
Call-ID: ef7acf80-43d153e4-50-5a20300e@14.48.32.90
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 14.48.32.170:[5060]:
[907,NET]
BYE sip:14.48.32.170:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK526f3d2afa
From: <sip:9110001@14.48.32.90;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@14.48.32.170>;tag=1
Date: Tue, 14 Oct 2014 16:48:36 GMT
Call-ID: ef7acf80-43d153e4-50-5a20300e@14.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
P-Asserted-Identity: <sip:9110001@14.48.32.90>
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 14.48.32.170:[5060]:

[908,NET]

ACK sip:14.48.32.170:5060;transport=UDP SIP/2.0

Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK531df920a6

From: <sip:9110001@14.48.32.90;x-farend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=352-713e2333-4032-45f1-b1f5-e33cf471acec-49613645

To: <sip:8675309@14.48.32.170>;tag=2

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

Call-ID: ef7acf80-43d153e4-51-5a20300e@14.48.32.90

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 14.48.32.170:[5060]: [909,NET] BYE sip:14.48.32.170:5060;transport=UDP SIP/2.0 Via: SIP/2.0/UDP

14.48.32.90:5060;branch=z9hG4bK5462aba807 From: <sip:9110001@14.48.32.90;x-farend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=352-713e2333-4032-45f1-b1f5-e33cf471acec-49613645 To: <sip:8675309@14.48.32.170>;tag=2 Date: Tue, 14 Oct 2014 16:48:36 GMT Call-ID:

ef7acf80-43d153e4-51-5a20300e@14.48.32.90 User-Agent: Cisco-CUCM10.5 Max-Forwards: 70 P-

Asserted-Identity: <sip:9110001@14.48.32.90> CSeq: 102 BYE **Reason: Q.850;cause=47** Content-

Length: 0

Misconfiguration Including CSS and PT issues

The commands below allow the majority of the recording configurations to be reviewed quickly with only knowing the MAC address of a phone that isn't recording calls. Simply replace the part of the command '**MAC_of_Phone**' with the actual MAC address of the phone as in the examples below.

This gives us the DN (all of them if there is more than one) for the MAC we are searching on, the MAC of the phone just for confirmation, the bib setting, the privacy setting, the recording type (reference the values listed in the examples from my lab) the recording profile in use by the phone, the name of the recording CSS, the recording destination for that recording profile, and the partition that recording destination is associated with based on the MAC we are searching on:

```
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='MAC_of_Phone'
```

This gives us the list of partitions that are associated with the recording CSS on the recording profile that is associated with the MAC of the phone we are searching against.

```
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'
```

Here are examples of the output from my lab for a phone with MAC address **SEPC80084AA8743**:

In this command we can see the phone has only one DN on it which is **2003**, we also see the BiB is on, privacy is off, the recording type is automatic, the preferred source is phone, the recording profile is **Test Recording Profile**, the recording calling search space is **INTERNAL_CSS**, the route pattern for recorded calls is **8675309** and that pattern is associated with the partition **INTERNAL_PT**.

```
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
```

With the output of this command we are checking all of the partitions of the recording CSS of the recording profile associated with the phone of interest. We can see here the partition **INTERNAL_PT** is one of the partitions associated with the calling search space **INTERNAL_CSS**.

This means there should be no issues with the BiB of the phone being able to call the recording route pattern.

```
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
```