

Configuração básica e Troubleshooting da gravação do atendimento

Índice

[Introdução](#)

[Pré-requisitos](#)

[Requisitos](#)

[Componentes Utilizados](#)

[Tipos de gravação do atendimento](#)

[Automático](#)

[Aplicativo invocado](#)

[Seletivo](#)

[Gateway-baseado](#)

[Configuração da gravação do Automatic Call para a integração do SORVO somente](#)

[Crie o tronco do SORVO ao destino da gravação](#)

[Crie o perfil da gravação](#)

[Crie a rota padrão para distribuir os atendimentos da gravação](#)

[Atribua o perfil da gravação à linha telefônica](#)

[Ajuste o BABADOR a sobre e a privacidade a fora na página da configuração telefônica](#)

[Verificar](#)

[SCCP](#)

[SORVO](#)

[Troubleshooting](#)

[Negociação de codec](#)

[Misconfiguration que inclui edições CSS e PT](#)

Introdução

Este documento descreve os princípios da gravação do atendimento dentro do gerente das comunicações unificadas de Cisco (CUCM), os media previstos flui, os fluxos da chamada esperada para dispositivos do Session Initiation Protocol (SIP) e do protocolo skinny client control (SCCP), e um exemplo de um tipo comum de falha da instalação da gravação do atendimento.

Pré-requisitos

Requisitos

CUCM integrado com um server da terceira da gravação.

[Componentes Utilizados](#)

CUCM, Cisco IP Phone (o IP é protocolo de internet), e um server da gravação do atendimento.

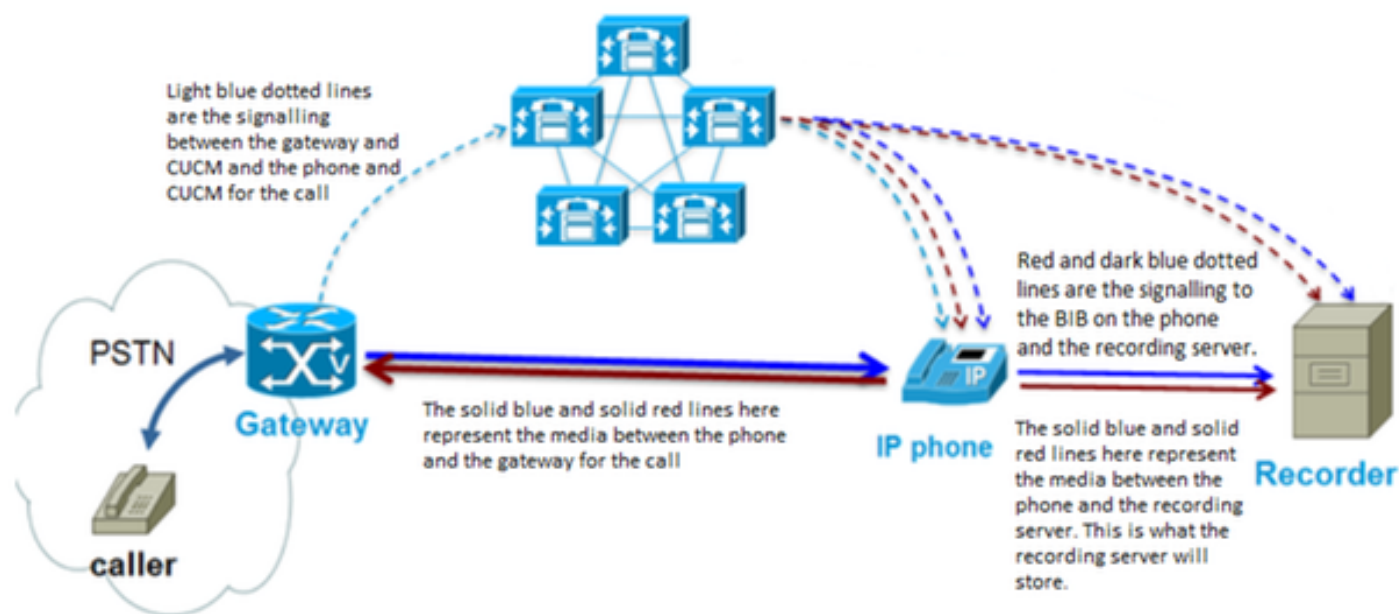
Tipos de gravação do atendimento

Automático

Os elementos chaves de gravação do Automatic Call estão abaixo:

- Usa a Construir-em-ponte do telefone IP “para bifurcar-se” áudio ao destino da gravação
- Iniciado cada vez que o telefone IP coloca um atendimento ou recebe um atendimento
- Exige somente um tronco do SORVO entre CUCM e destino da gravação. Alguns vendedores da gravação exigem a integração CTI (integração de telefonia e computador)
- Não permite a gravação dos telefones que são ficados situados fora da rede gerenciada (deve ter o acesso para enviar o RTP diretamente ao server gravar e a ser um Cisco IP Phone capaz de atribuir uma Construir-em-ponte)

No diagrama abaixo as linhas contínuas representam os media previstos fluem e as linhas tracejadas representam o fluxo previsto da sinalização:

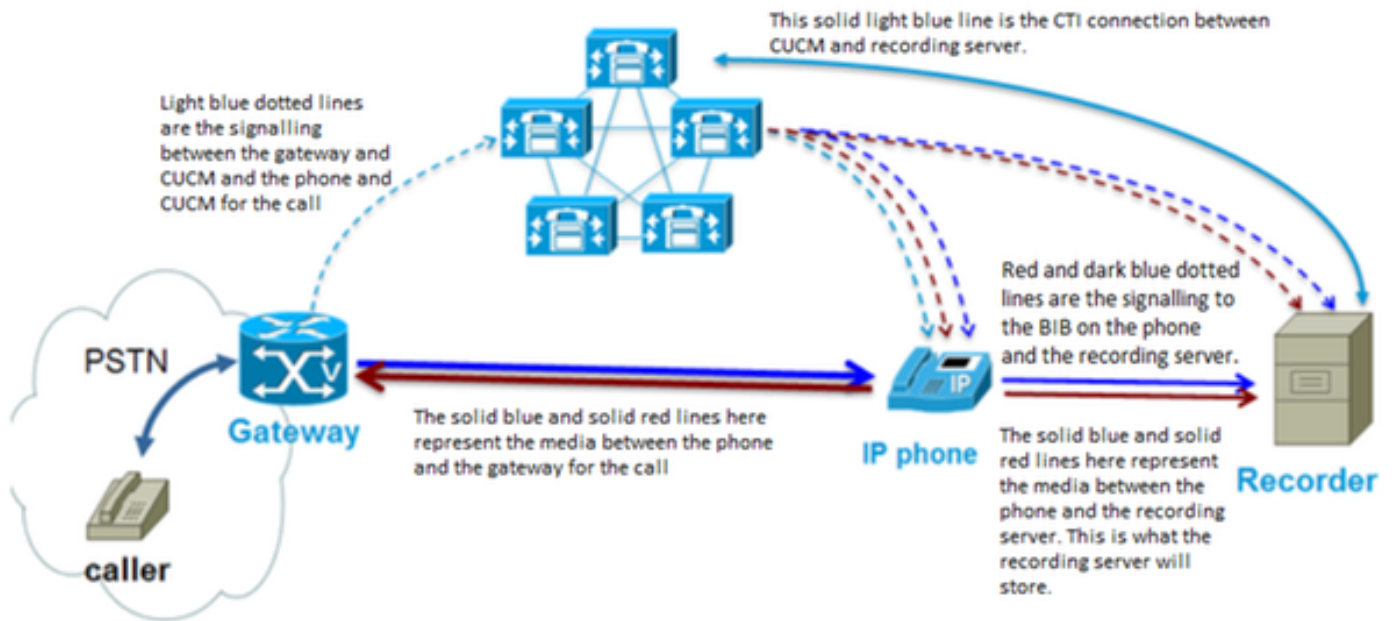


Aplicativo invocado

Os elementos chaves de gravação invocada aplicativo do atendimento estão abaixo:

- Usa a Construir-em-ponte do telefone IP “para bifurcar-se” áudio ao destino da gravação
- Iniciado quando o aplicativo (registrator) ditar que deve ser iniciado
- Exige o tronco e a Integração de CTI do SORVO com aplicativo da gravação
- O usuário do aplicativo CTI deve ter o acesso aos valores-limite que precisam de ser gravados
- Não permite a gravação dos telefones que são ficados situados fora da rede gerenciada (deve ter o acesso para enviar o RTP diretamente ao server gravar)

No diagrama abaixo as linhas contínuas representam os media previstos fluem e as linhas tracejadas representam o flo de sinalização previsto. A linha contínua entre CUCM e o server da gravação denota uma conexão CTI entre CUCM e o aplicativo.

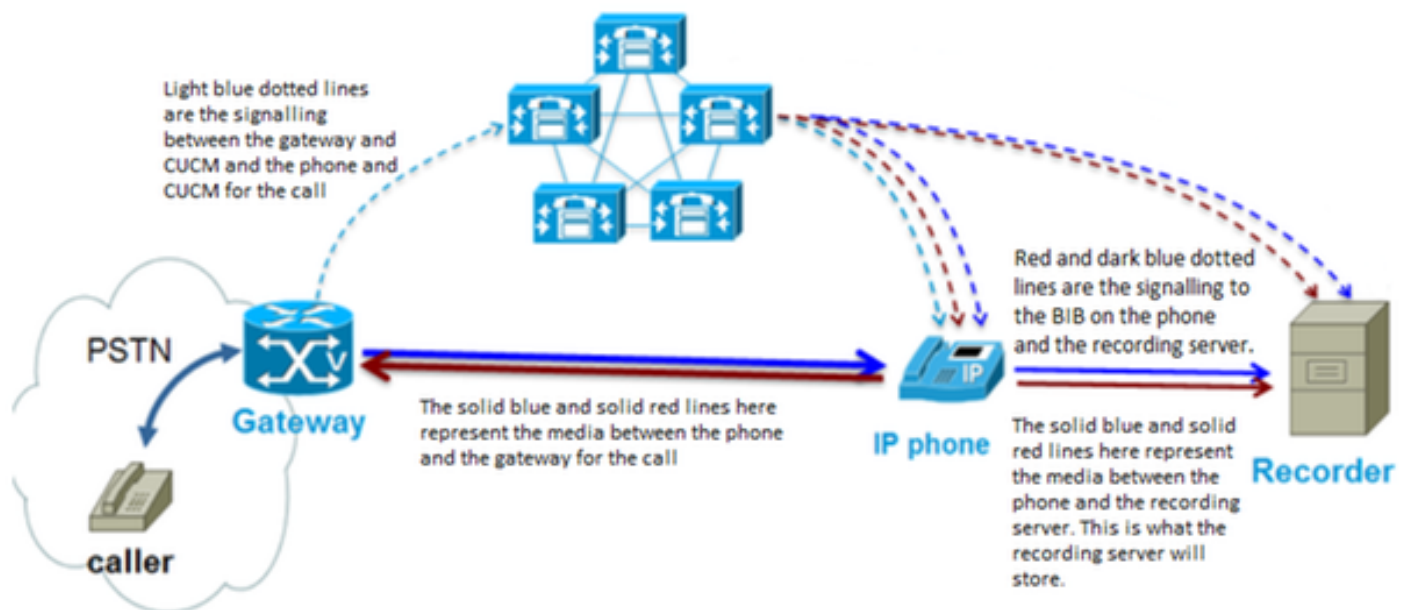


Seletivo

Os elementos chaves de gravação seletiva do atendimento estão abaixo:

- Usa a Construir-em-ponte do telefone IP “para bifurcar-se” áudio ao destino da gravação
- Iniciado cada vez que o usuário de telefone IP seleciona a opção da gravação em seu telefone IP (CUCM 9.x+) ou em um aplicativo como [nesta imagem](#)
- Exige tipicamente somente um tronco do SORVO entre CUCM e destino da gravação (segundo o vendedor do aplicativo da gravação)
- Não permite a gravação dos telefones essa mentira fora da rede gerenciada (deve ter o acesso para enviar o RTP diretamente ao server gravar)

Como você pode ver no diagrama abaixo, o trajeto dos media e do sinal é muito similar à gravação do Automatic Call:

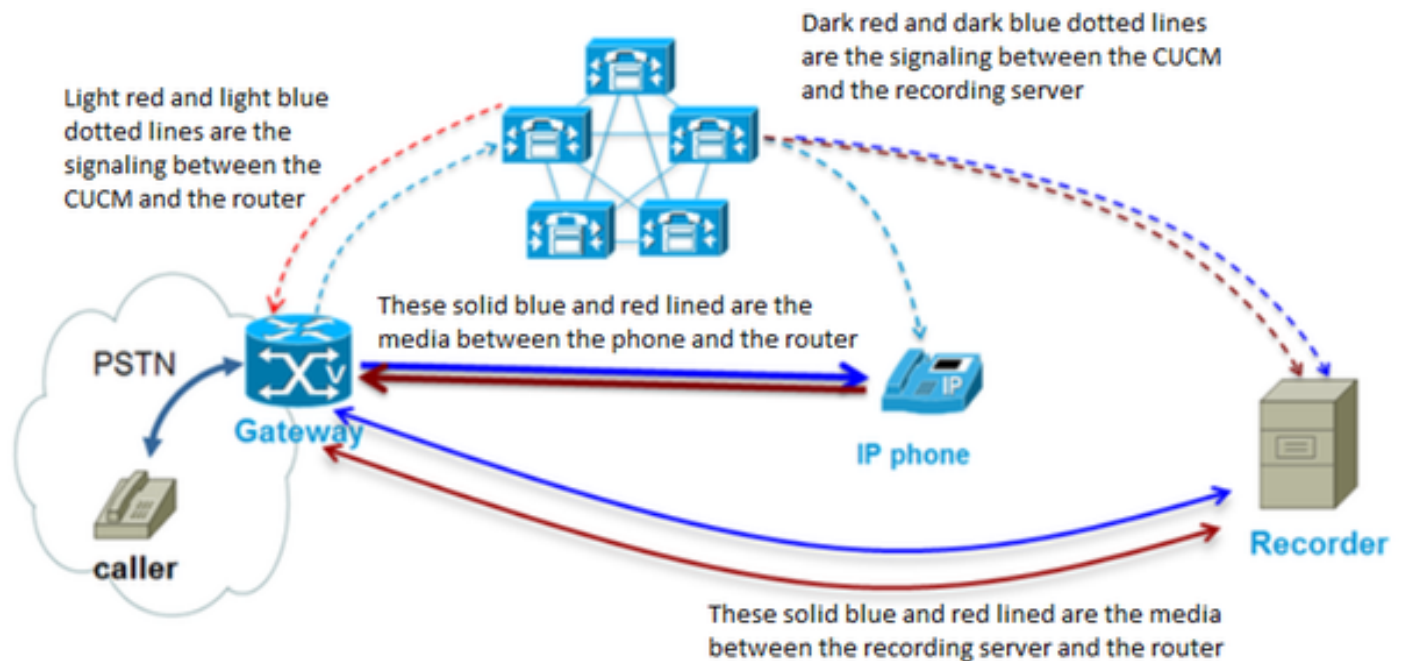


Gateway-baseado

Os elementos chaves de gravação do gateway-based call estão abaixo:

- O gateway de voz bifurca-se os media para o destino da gravação
- Registros CUCM com gateway como um aplicativo
- CUCM usa o HTTP para instruir o GW para fluir media ao destino de gravação
- CUCM integra com o destino da gravação através do tronco do SORVO
- Permite a gravação dos atendimentos isso simplesmente passagem através da rede gerenciada (por exemplo, aos usuários móveis) ou os telefones que não apoiam o babador

Como você pode ver do diagrama abaixo, os media fluem são bastante diferentes dos outros tipos de gravação do atendimento:



Configuração da gravação do Automatic Call para a integração do SORVO somente

Esta seção descreve como setup a integração do SORVO de um server da gravação.


Crie o tronco do SORVO ao destino da gravação

- Sob o dispositivo > o tronco, seletos adicionar novo
- Crie um tronco do SORVO com os seguintes ajustes:

Trunk Configuration



Status

 Status: Ready

Trunk Information

Trunk Type*

Device Protocol*

Trunk Service Type*

Next

- Entre o nome de dispositivo, o pool de dispositivos, o perfil de segurança do tronco MRGL, de SORVO, e o perfil apropriados do SORVO
- O endereço de destino configurado será o endereço do server de aplicativo da gravação. No exemplo abaixo do server da gravação são 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"/>


Crie o perfil da gravação

- Sob o dispositivo > os ajustes do dispositivo > o perfil da gravação
- O endereço de destino de gravação é o lugar aonde os atendimentos da gravação serão enviados

Recording Profile Configuration

 Save
  Delete
  Copy
  Add New

Status

 Status: Ready

Recording Profile Information

Name*

Recording Calling Search Space

Recording Destination Address *

Crie a rota padrão para distribuir os atendimentos da gravação

- Crie uma rota padrão que combine o endereço de destino da gravação configurado na etapa precedente
- Você pode apontar a uma lista da rota em vez de diretamente no tronco do SORVO, se você deseja configurar troncos redundantes do SORVO

Note por favor que o a separação atribuída a esta rota padrão deve ser associada com o **Calling Search Space** da gravação.

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

Atribua o perfil da gravação à linha telefônica

- Em um telefone já criado com uma extensão existente, atribua o perfil da gravação criado
- Atribua o tipo de gravação do atendimento neste lugar também
- Este exemplo mostra a gravação automática

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

Ajuste o BABADOR a sobre e a privacidade a fora na página da configuração telefônica

Quando na página da configuração de dispositivo navegue ao seicoint intitulado **informação do dispositivo**. Ajuste construído na ponte a sobre e na privacidade a fora.

Built In Bridge*	On
Privacy*	Off

Verificar

O abaixo são os comportamentos esperados nos rastreamentos do CallManager para o SCCP e SORVEM os telefones dados a configuração acima. Estes exemplos são para uma chamada de telefone um outro telefone no mesmo conjunto quando um dos telefones se estabelecer para a gravação do atendimento.

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-blf5-e33cf47lacec-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-blf5-e33cf47lacec-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-e33cf47lacec-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-e33cf47lacec-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-blf5-e33cf471lacec-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-blf5-e33cf471lacec-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)

SORVO

~~~~~  
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 |SdlSig |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-e33cf471lacec-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-e33cf471lacec-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-

faarendclusterid=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-blf5-e33cf47lacec-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-blf5-e33cf47lacec-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-blf5-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-blf5-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=sep000b4d9e8cb;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
```

Troubleshooting

Negociação de codec

O abaixo é um exemplo de um da maioria de tipo comum de falhas da gravação do atendimento - incompatibilidade de codec entre o telefone gravado e o server da gravação:

```
~~~~~
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:0x0e30201c00000000000000000000(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:0x0e30201c00000000000000000000(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:0x0e30202100000000000000000000(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:0x0e30202100000000000000000000(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-e33cf47lacec-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-e33cf47lacec-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-e33cf47lacec-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=0
numXcoderRequired=2 xcodingSide=0### 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### 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-e33cf47lacec-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-e33cf47lacec-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 que inclui edições CSS e PT

Os comandos abaixo permitem que a maioria das configurações da gravação seja revista rapidamente com somente conhecer o MAC address de um telefone que não esteja gravando atendimentos. Substitua simplesmente parte do comando **"MAC_of_Phone"** com o MAC address real do telefone como nos exemplos abaixo.

Isto dá-nos o DN (todo se há mais de um) para o MAC que nós estamos procurando sobre, o MAC do telefone apenas para a confirmação, o ajuste do babador, o ajuste da privacidade, o tipo de gravação (proveja os valores alistados nos exemplos de meu laboratório) o perfil da gravação no uso pelo telefone, o nome do CSS de gravação, o destino da gravação para esse perfil da gravação, e a separação que gravando o destino está associado com baseado no MAC nos está procurando sobre:

execute sql n1.dnorpattern seletto como phone_dn, dev.name como o phone_mac, o CASO dev.tkstatus_builtonbridge QUANDO '1' ENTÃO o "babador está em" QUANDO '0' ENTÃO o "babador está fora" da EXTREMIDADE OUTRA "NA" como o is_bib_on, o CASO dev.resettoggle QUANDO "t" ENTÃO "privacidade está em" QUANDO "f" ENTÃO "privacidade está fora" da EXTREMIDADE OUTRA "NA" como o is_privacy_on, o CASO recordynam.tkrecordingflag QUANDO EXTREMIDADE OUTRA "seletiva" "NA" do '2' do '1' do '0' ENTÃO a "gravação desabilitou" QUANDO ENTÃO "automático" QUANDO ENTÃO como o recording_type, o CASO

devnumplanmap.tkpreferredmediasource QUANDO '1' ENTÃO o "gateway preferiu" QUANDO o '2' ENTÃO "telefonam" à EXTREMIDADE OUTRA preferida "NA" como Recording_Media_Source, rcrdpro.name como o recording_profile_name, o css.name como css_used_by_recording_profile, o rcrdpro.recorderdestination como recording_route_pattern, o rp.name tão required_partition_for_css_used_by_recording_profile de recordingprofile como o callingsearchspace da junta interna do rcrdpro como o css em junta interna rcrdpro.fkcallingsearchspace_callrecording = css.pkid numplan como n em routepartition da junta interna n.dnorpattern = rcrdpro.recorderdestination como o rp em devicenumplanmap da junta interna rp.pkid = n.fkroutepartition como o devnumplanmap em junta interna rcrdpro.pkid = devnumplanmap.fkrecordingprofile recordingdynamic como o recordynam em dispositivo da junta interna devnumplanmap.pkid = recordynam.fkdevicenumplanmap como o colaborador em junta interna devnumplanmap.fkdevice = dev.pkid numplan como n1 em devnumplanmap.fknumplan = n1.pkid onde css.pkid = rcrdpro.fkcallingsearchspace_callrecording e dev.name= MAC_of_Phone

Isto dá-nos a lista de separações que são associadas com o CSS de gravação no perfil da gravação que é associado com o MAC do telefone que nós estamos procurando contra.

execute sql css.name seletor como name_of_the_recording_css, rp.name como partitions_in_recording_css, csm.sortorder do callingsearchspace como o callingsearchspace_member da junta interna css como o csm em routepartition da junta interna csm.fkcallingsearchspace = css.pkid como o rp em junta interna csm.fkroutepartition = rp.pkid recordingprofile como o rcrdpro em devicenumplanmap da junta interna rcrdpro.fkcallingsearchspace_callrecording = css.pkid como o devnumplanmap em dispositivo da junta interna rcrdpro.pkid = devnumplanmap.fkrecordingprofile como o colaborador em devnumplanmap.fkdevice = dev.pkid onde css.pkid = rcrdpro.fkcallingsearchspace_callrecording e dev.name= MAC_of_Phone

Estão aqui os exemplos da saída de meu laboratório para um telefone com MAC address **SEPC80084AA8743**:

Neste comando nós podemos ver o telefone tem somente um DN nele qual é **2003**, nós igualmente vê que o babador está ligada, a privacidade está, o tipo de gravação é automático, o origem de energia é telefone, o perfil da gravação é **perfil da gravação do teste**, o Calling Search Space da gravação é **INTERNAL_CSS**, a rota padrão para atendimentos gravados é **8675309** e esse teste padrão é associado com a separação **INTERNAL_PT**.

```
### 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-blf5-e33cf471lacec-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

Com a saída deste comando nós estamos verificando todas as separações do CSS de gravação do perfil da gravação associado com o telefone do interesse. Nós podemos ver que aqui a separação **INTERNAL_PT** é uma das separações associadas com o Calling Search Space **INTERNAL_CSS**. Isto significa que não deve haver nenhuma edição com o babador do telefone que pode chamar a rota padrão da gravação.

```
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
```