

Configurar e solucionar problemas de gravação básica de chamadas

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Introduction

Este documento descreve os conceitos básicos de gravação de chamadas no Cisco Unified Communications Manager (CUCM).

Prerequisites

Requirements

A Cisco recomenda que você tenha conhecimento do CUCM integrado com um servidor de gravação de terceiros.

Componentes Utilizados

As informações neste documento são baseadas nestas versões de software e hardware:

- CUCM
- Protocolo de Internet da Cisco (IP)
- Servidor de Gravação de Chamada Telefônica

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. Se a rede estiver ativa, certifique-se de que você entenda o impacto potencial de qualquer comando.

Informações de Apoio

Este documento também discute o fluxo de mídia esperado, os fluxos de chamada esperados para os dispositivos Session Initiation Protocol (SIP) e Skinny Client Control Protocol (SCCP), e um exemplo de um tipo comum de falha na configuração da gravação de chamada.

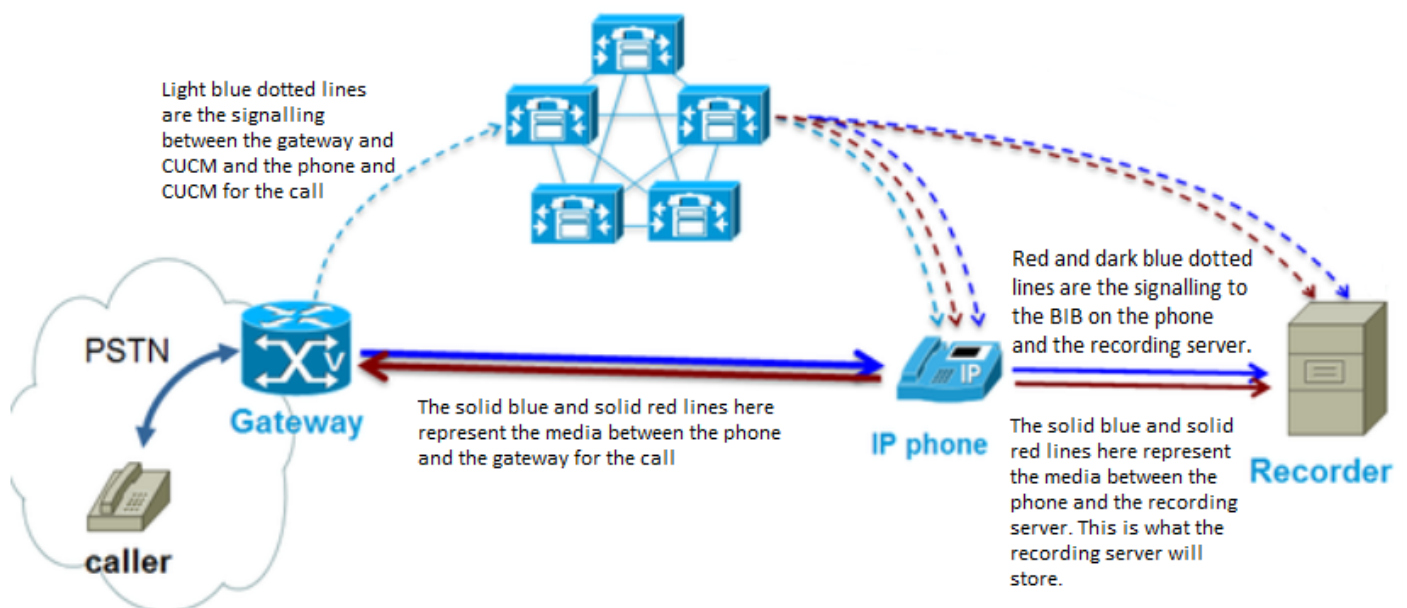
Tipos de gravação de chamada

Automático

Os principais elementos da gravação automática de chamadas são os seguintes:

- Usa o Built-In-Bridge (BIB) do telefone IP para bifurcar o áudio para o destino da gravação
- Iniciado toda vez que o telefone IP faz ou recebe uma chamada
- Requer apenas um tronco SIP entre o CUCM e o destino de gravação. Alguns fornecedores de gravação exigem a integração entre telefonia e computador (CTI)
- Não permite a gravação de telefones localizados fora da rede gerenciada (deve ter acesso para enviar RTP diretamente ao servidor de gravação e ser um telefone IP da Cisco capaz de alocar um BIB)

Neste diagrama, as linhas sólidas representam o fluxo de mídia esperado e as linhas pontilhadas representam o fluxo de sinalização esperado:

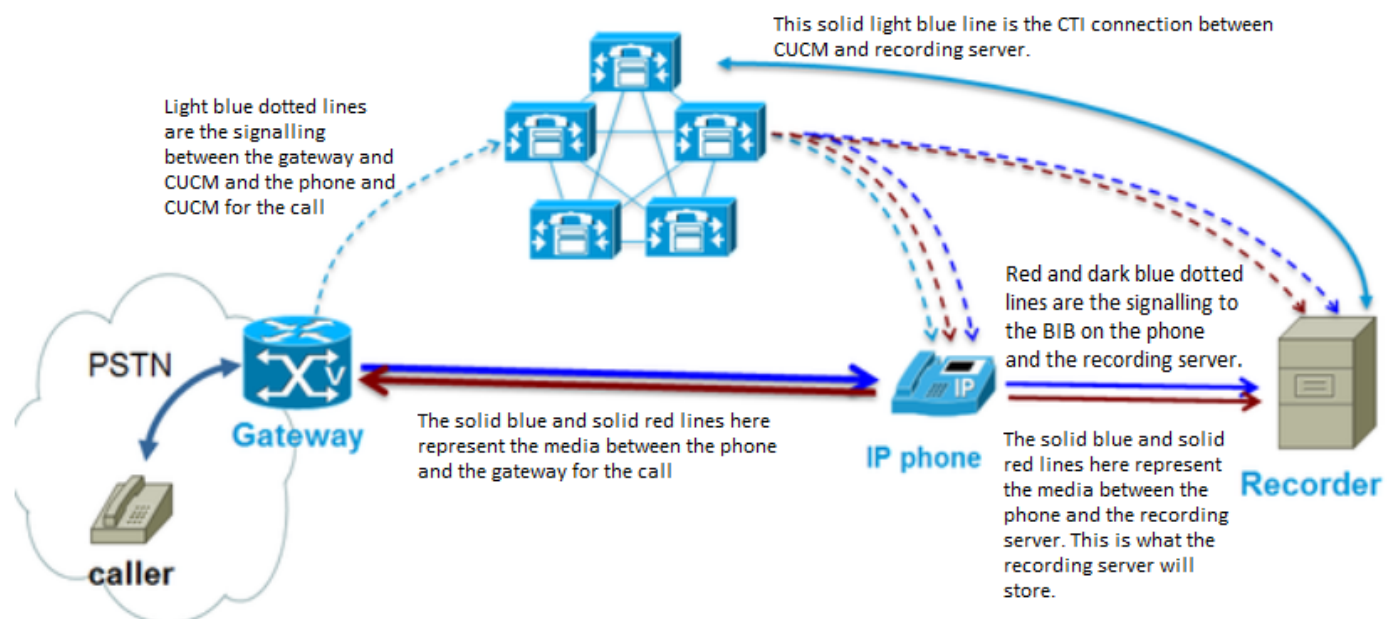


Aplicativo Chamado

Os principais elementos do aplicativo chamado gravação de chamada são os seguintes:

- Usa BIB de telefone IP para bifurcar o áudio para o destino de gravação
- Iniciado quando o aplicativo (gravador) determina que ele deve ser iniciado
- Requer tronco SIP e CTI com aplicativo de gravação
- O usuário do aplicativo CTI deve ter acesso a endpoints que precisam ser gravados
- Não permite a gravação de telefones localizados fora da rede gerenciada (deve ter acesso para enviar RTP diretamente ao servidor de gravação)

No diagrama aqui, as linhas sólidas representam o fluxo de mídia esperado e as linhas pontilhadas representam o fluxo de sinalização esperado. A linha sólida entre o CUCM e o servidor de gravação denota uma conexão CTI entre o CUCM e o aplicativo.

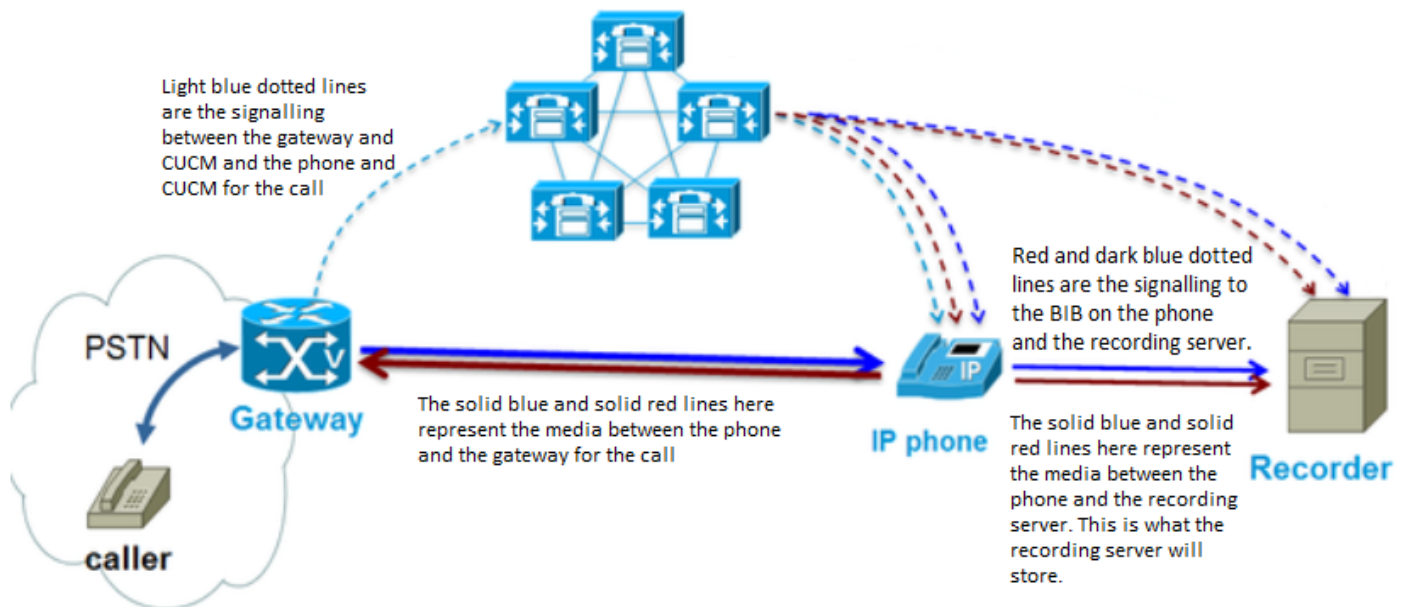


Seletivo

Os principais elementos da gravação seletiva de chamadas são os seguintes:

- Usa BIB de telefone IP para bifurcar o áudio para o destino de gravação
- Iniciado toda vez que o usuário do telefone IP seleciona a opção de gravação em seu telefone IP (CUCM 9.x+) ou em um aplicativo como [esta imagem](#)
- Geralmente requer apenas um tronco SIP entre o CUCM e o destino de gravação (que depende do fornecedor do aplicativo de gravação)
- Não permite a gravação de telefones que estão fora da rede gerenciada (deve ter acesso para enviar RTP diretamente ao servidor de gravação)

Como você pode ver neste diagrama aqui, o caminho da mídia e da sinalização é muito semelhante à gravação automática de chamadas:

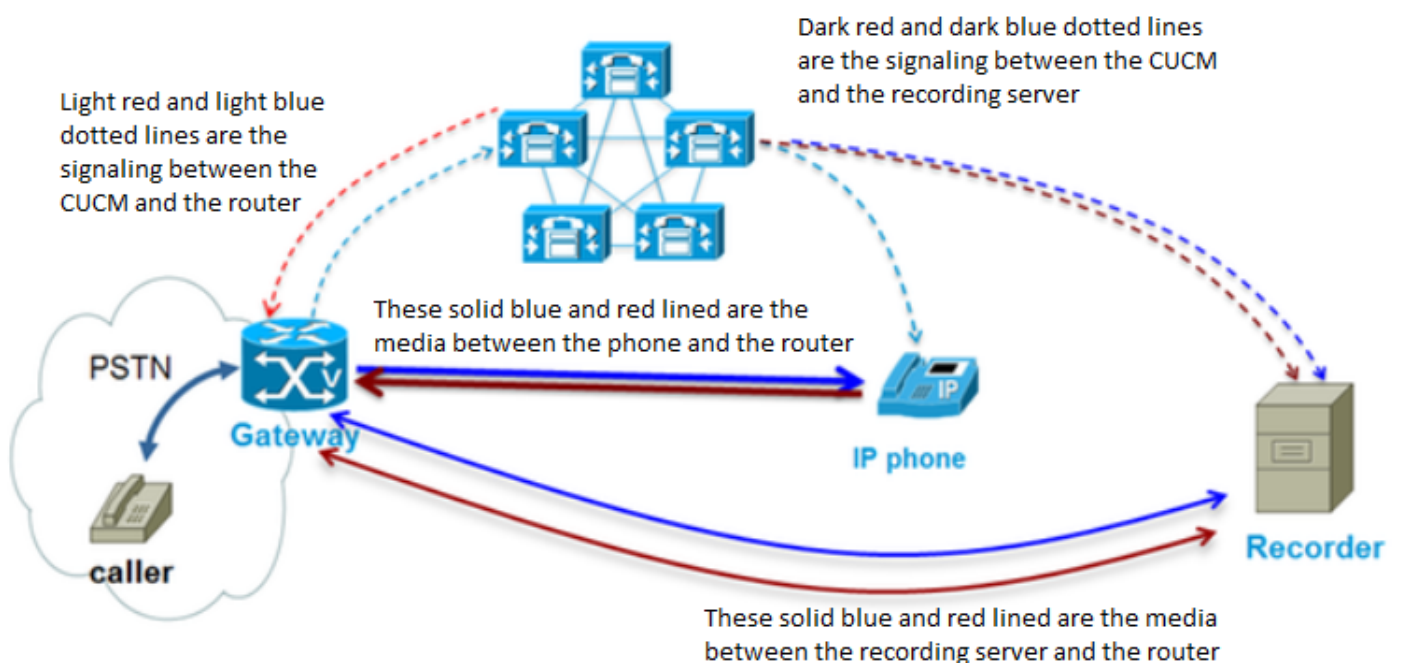


Baseado em gateway

Os principais elementos da gravação de chamada baseada em gateway são os seguintes:

- O gateway de voz bifurca a mídia em direção ao destino de gravação
- O CUCM registra o gateway como um aplicativo
- O CUCM usa o HTTP para instruir o Gateway (GW) a transmitir mídia para o destino de gravação
- O CUCM integra-se com o destino de gravação via tronco SIP
- Permite a gravação de chamadas que simplesmente passam através da rede gerenciada (por exemplo, para usuários móveis) ou para telefones que não suportam o BIB

Como você pode ver no diagrama aqui, o fluxo de mídia é bem diferente dos outros tipos de gravação de chamada:



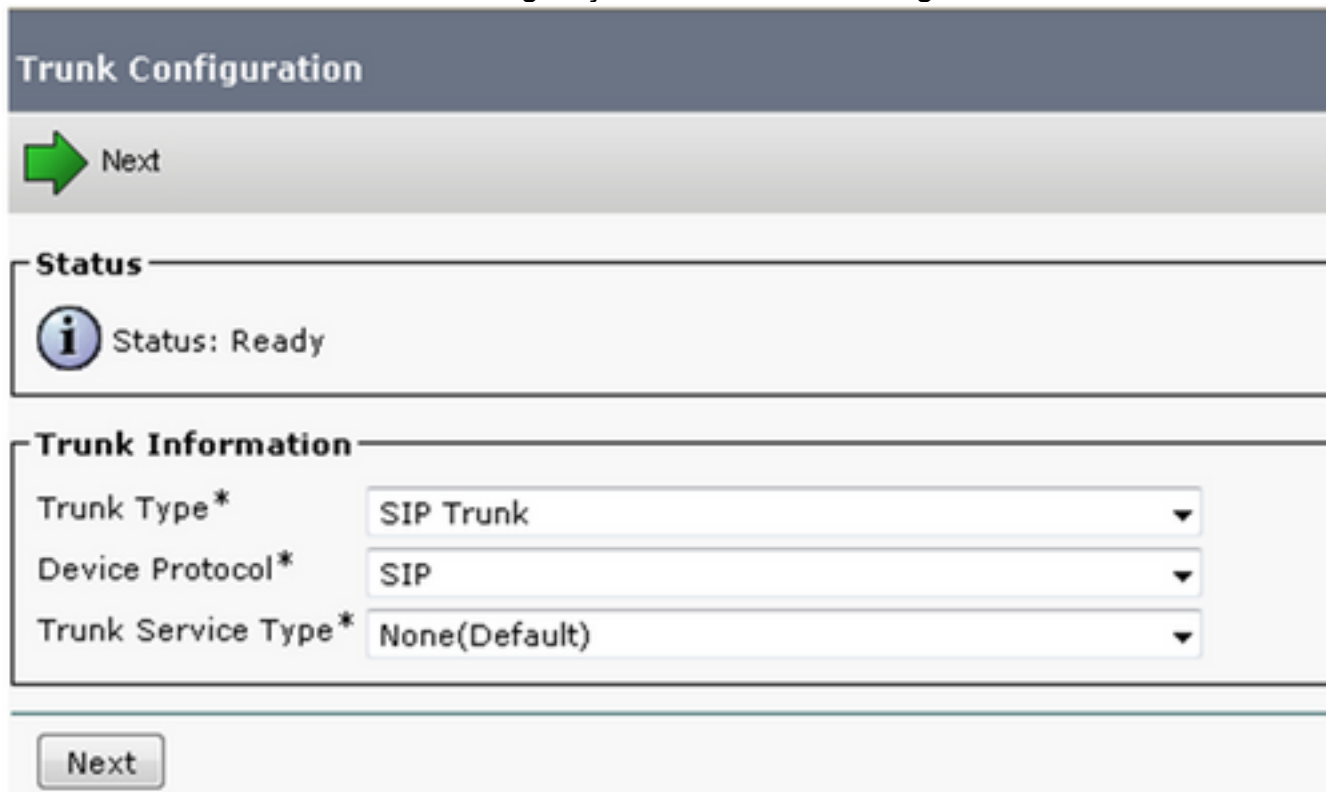
Configuração Automática de Gravação de Chamada para

Integração somente SIP

Esta seção descreve como configurar a integração SIP de um servidor de gravação.

Criar Tronco SIP para o Destino da Gravação

- Navegue até **Device > Trunk** e selecione **Add New**.
- Crie um tronco SIP com as configurações mostradas na imagem.







The screenshot shows the 'Trunk Configuration' page. At the top, there is a dark blue header with the text 'Trunk Configuration'. Below the header is a green arrow pointing right with the word 'Next' next to it. The main content area is divided into two sections: 'Status' and 'Trunk Information'. The 'Status' section shows an information icon (i) followed by the text 'Status: Ready'. The 'Trunk Information' section contains three dropdown menus: 'Trunk Type*' set to 'SIP Trunk', 'Device Protocol*' set to 'SIP', and 'Trunk Service Type*' set to 'None(Default)'. At the bottom of the form, there is a 'Next' button.

- Insira o nome do dispositivo, o pool de dispositivos, MRGL, o perfil de segurança de tronco SIP e o perfil SIP apropriados
- O endereço de destino configurado é o endereço do servidor de aplicativos de gravação.


Criar Perfil de Gravação

- Navegue até **Dispositivo > Configurações do dispositivo > Perfil de gravação**
- O endereço de destino da gravação é o local para onde as chamadas de gravação são enviadas, como mostrado na imagem.

Recording Profile Configuration

 Save
  Delete
  Copy
  Add New

Status

 Status: Ready

Recording Profile Information

Name*

Recording Calling Search Space

Recording Destination Address *

Criar padrão de rota para rotear chamadas de gravação

- Crie um padrão de rota que corresponda ao endereço de destino de gravação configurado na Etapa anterior
- Você pode apontar para uma lista de rotas em vez de apontar diretamente para o tronco SIP, se desejar configurar troncos SIP redundantes

Observação: a partição atribuída a este padrão de rota deve ser associada ao Espaço de pesquisa de chamada de gravação e como mostrado na imagem.

Pattern Definition

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

Atribuir perfil de gravação à linha telefônica

- Em um telefone já criado com um ramal existente, atribua o perfil de gravação criado
- Atribuir o tipo de gravação de chamada neste local também
- O exemplo mostra a gravação automática, como mostrado na imagem.

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

Definir BIB como Ativado e Privacidade como Desativada na Página de Configuração do Telefone

Enquanto estiver na página de configuração do dispositivo, navegue para a seção intitulada **Informações do dispositivo**. Defina Built In Bridge como **On** e Privacy como **Off** como mostrado na imagem.

Built In Bridge*	On
Privacy*	Off

Verificar

Use esta seção para confirmar se a sua configuração funciona corretamente.

Aqui estão os comportamentos esperados nos rastreamentos do Call Manager para telefones SCCP e SIP com a configuração fornecida. Esses exemplos são para um telefone que chama outro telefone no mesmo cluster enquanto um dos telefones está configurado para gravação de chamada.

Observação: os logs a serem coletados do CUCM são CTIManager, CallManager, Event Viewer App/Sys e pcaps podem ser necessários em alguns cenários.

Observação: os logs a serem coletados dos telefones são logs e pcaps do console. Você pode obter pcaps do servidor de gravação ao mesmo tempo em que obtém os pcaps do telefone.

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

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

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

CUCM extends call to phone

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

Called (recorded) phone goes off hook

03797089.001 |20:21:09.335 |AppInfo |StationD: (0000114) restart0_StationOffHook - INFO:

CI=38960750 on line=1, SPKMode=0, alwaysPrimeLine=0, alwaysUsePrimeLineForVM=0, fid=0, offHookTrigger=0.

CUCM Tells the calling phone to open the logical channel

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

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

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

CUCM Tells the calling phone to open the receive channel

03797164.002 |20:21:09.337 |AppInfo |StationD: (0000109) OpenReceiveChannel conferenceID=38960749 passThruPartyID=33554450 millisecondPacketSize=20 compressionType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0 qualifierIn=? sourceIpAddr=IpAddr.type:0 ipAddr:0x0e302021000000000000000000000000(10.48.32.33). myIP: IpAddr.type:0 ipv4Addr:0x0e30201c(10.48.32.28)

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

03797168.002 |20:21:09.337 |AppInfo |StationD: (0000114) OpenReceiveChannel conferenceID=38960750 passThruPartyID=33554451 millisecondPacketSize=20 compressionType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0 qualifierIn=? sourceIpAddr=IpAddr.type:0 ipAddr:0x0e30201c000000000000000000000000(10.48.32.28). myIP: IpAddr.type:0 ipv4Addr:0x0e302021(10.48.32.33)

CUCM allocates BIB on called (recorded) phone

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

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

03797269.001 |20:21:09.340 |AppInfo |Digit Analysis: star_DaReq: daReq.partitionSearchSpace(), filteredPartitionSearchSpaceString(), partitionSearchSpaceString()
03797269.002 |20:21:09.340 |AppInfo |Digit Analysis: star_DaReq: Matching Legacy Numeric, digits=8675309
03797269.003 |20:21:09.340 |AppInfo |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept DAMR.sstype=[0], TPcount=[0], DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]
03797269.004 |20:21:09.340 |AppInfo |Digit Analysis: getDaRes - Remote Destination [8675309] isURI[0]
03797269.005 |20:21:09.340 |AppInfo |CMUtility routeCallThroughCTIRD: no matching RemDestDynamic record exists for remdest [8675309]
03797269.006 |20:21:09.340 |AppInfo |DbMobility: getMatchedRemDest starts: cnumber = 8675309
03797269.007 |20:21:09.340 |AppInfo |DbMobility: getMatchedRemDest: full match case
03797269.008 |20:21:09.340 |AppInfo |DbMobility SelectByDestination: no matching RemDestDynamic record exists for remdest [8675309]
03797269.009 |20:21:09.340 |AppInfo |DbMobility: can't find remdest 8675309 in map
03797269.010 |20:21:09.340 |AppInfo |Digit analysis: patternUsage=5
03797269.011 |20:21:09.340 |AppInfo |Digit analysis: match(pi="1", fqcn="", cn="b00223908001", plv="5", pss="E911_PT:Phones_PT:EMERGENCY_PT:INTERNAL_PT:INFORMACAST_PT", TodFilteredPss="E911_PT:Phones_PT:EMERGENCY_PT:INTERNAL_PT:INFORMACAST_PT", dd="8675309", dac="0")
03797269.012 |20:21:09.340 |AppInfo |Digit analysis: analysis results

03797269.013 |20:21:09.340 |AppInfo ||PretransformCallingPartyNumber=b00223908001
|CallingPartyNumber=b00223908001
|DialingPartition=
|DialingPattern=8675309
|FullyQualifiedCalledPartyNumber=8675309
|DialingPatternRegularExpression=(8675309)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=8675309
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=8675309
|CollectedDigits=8675309

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

03797320.001 |20:21:09.343 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message
to 10.48.32.170:[5060]:
[212231,NET]
INVITE sip:8675309@10.48.32.170:5060 SIP/2.0
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK204d520fedb3
From: <sip:9110001@10.48.32.90;x-nearend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-
nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=38960749;x-
farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-
farendaddr=9110006>;tag=73601~713e2333-4032-45f1-blf5-e33cf471acec-38960754
To: <sip:8675309@10.48.32.170>
Date: Tue, 30 Sep 2014 00:21:09 GMT
Call-ID: abbb8e00-4291f775-204c-5a20300e@10.48.32.90
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: ;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 2881195520-0000065536-0000000011-1512058894
Session-Expires: 1800
P-Asserted-Identity: <sip:9110001@10.48.32.90>
Remote-Party-ID: <sip:9110001@10.48.32.90>;party=calling;screen=yes;privacy=off
Contact: <sip:9110001@10.48.32.90:5060>;isFocus
Max-Forwards: 70
Content-Length: 0

BiB places second call to recording destination address (cn is calling party which is the
BiB cn="b00223908001" and it is dialing the recordingdestination dd="8675309")
Note that the BiB number stayed the same (b00223908001) and so did the recordingdestination
number

03797367.010 |20:21:09.344 |AppInfo |Digit analysis: patternUsage=5
03797367.011 |20:21:09.344 |AppInfo |Digit analysis: match(pi="1", fqcn="",
cn="b00223908001",plv="5", pss="E911_PT:Phones_PT:EMERGENCY_PT:INTERNAL_PT:INFORMACAST_PT",
TodFilteredPss="E911_PT:Phones_PT:EMERGENCY_PT:INTERNAL_PT:INFORMACAST_PT",
dd="8675309",dac="0")
03797367.012 |20:21:09.344 |AppInfo |Digit analysis: analysis results
03797367.013 |20:21:09.344 |AppInfo ||PretransformCallingPartyNumber=b00223908001
|CallingPartyNumber=b00223908001
|DialingPartition=
|DialingPattern=8675309
|FullyQualifiedCalledPartyNumber=8675309

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

CUCM receives 200 OK in response to INVITE #1

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

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

CUCM sends INVITE #2 to recording server (10.48.32.170)

03797445.001 |20:21:09.348 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 10.48.32.170:[5060]:
[212233,NET]
INVITE sip:8675309@10.48.32.170:5060 SIP/2.0
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK204e754eaeae
From: <sip:9110001@10.48.32.90;x-farend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=38960749;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=73602~713e2333-4032-45f1-blf5-e33cf471lacec-38960757
To: <sip:8675309@10.48.32.170>
Date: Tue, 30 Sep 2014 00:21:09 GMT
Call-ID: abbb8e00-4291f775-204d-5a20300e@10.48.32.90
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: ;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 2881195520-0000065536-0000000012-1512058894
Session-Expires: 1800
P-Asserted-Identity: <sip:9110001@10.48.32.90>

Remote-Party-ID: <sip:9110001@10.48.32.90>;party=calling;screen=yes;privacy=off
Contact: <sip:9110001@10.48.32.90:5060>;isFocus
Max-Forwards: 70
Content-Length: 0

CUCM receives 200 OK in response to INVITE #2

03797498.001 |20:21:09.350 |AppInfo |//SIP/SIPUdp/wait_SdlDataInd: Incoming SIP UDP message
size 736 from 10.48.32.170:[5060]:
[212235,NET]
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK204e754eaeae
From: <sip:9110001@10.48.32.90;x-farend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-
nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=38960749;x-
farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-
farendaddr=9110006>;tag=73602~713e2333-4032-45f1-blf5-e33cf471lacec-38960757
To: <sip:8675309@10.48.32.170>;tag=2
Call-ID: abbb8e00-4291f775-204d-5a20300e@10.48.32.90
CSeq: 101 INVITE
Contact: <sip:10.48.32.170:5060;transport=udp>
Content-Type: application/sdp
Content-Length: 135

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

CUCM sends outbound ACK in response to 200 OK #1

03797500.001 |20:21:09.351 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message
to 10.48.32.170:[5060]:
[212236,NET]
ACK sip:10.48.32.170:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK204f50bef815
From: <sip:9110001@10.48.32.90;x-nearend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-
nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=38960749;x-
farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-
farendaddr=9110006>;tag=73601~713e2333-4032-45f1-blf5-e33cf471lacec-38960754
To: <sip:8675309@10.48.32.170>;tag=1
Date: Tue, 30 Sep 2014 00:21:09 GMT
Call-ID: abbb8e00-4291f775-204c-5a20300e@10.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Type: application/sdp
Content-Length: 254

v=0
o=CiscoSystemsCCM-SIP 73601 1 IN IP4 10.48.32.90
s=SIP Call
c=IN IP4 10.48.32.33
b=TIAS:64000
b=CT:64
b=AS:64
t=0 0
m=audio 4000 RTP/AVP 0 101
a=ptime:20
a=rtpmap:0 PCMU/8000

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

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

03797479.001 |20:21:09.350 |AppInfo |StationD: (0000114) startMediaTransmission
conferenceID=38960750 passThruPartyID=33554452 remoteIpAddress=IpAddr.type:0
ipAddr:0x0e3020aa000000000000000000000000(10.48.32.170) remotePortNumber=6000
milliSecondPacketSize=20 compressType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0
qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e302021(10.48.32.33)

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

03797596.001 |20:21:09.354 |AppInfo |StationD: (0000114) startMediaTransmission
conferenceID=38960750 passThruPartyID=33554453 remoteIpAddress=IpAddr.type:0
ipAddr:0x0e3020aa000000000000000000000000(10.48.32.170) remotePortNumber=6000
milliSecondPacketSize=20 compressType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0
qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e302021(10.48.32.33)

CUCM sends outbound ACK in response to 200 OK #2

03797615.001 |20:21:09.354 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 10.48.32.170:[5060]:
[212237,NET]
ACK sip:10.48.32.170:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK2050183495f1
From: <sip:9110001@10.48.32.90;x-farend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=38960749;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=73602~713e2333-4032-45f1-b1f5-e33cf47lacec-38960757
To: <sip:8675309@10.48.32.170>;tag=2
Date: Tue, 30 Sep 2014 00:21:09 GMT
Call-ID: abbb8e00-4291f775-204d-5a20300e@10.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Type: application/sdp
Content-Length: 254

v=0
o=CiscoSystemsCCM-SIP 73602 1 IN IP4 10.48.32.90
s=SIP Call
c=IN IP4 10.48.32.33
b=TIAS:64000
b=CT:64
b=AS:64
t=0 0
m=audio 4000 RTP/AVP 0 101
a=ptime:20
a=rtpmap:0 PCMU/8000
a=sendonly
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

Calling phone sends CUCM the ORC ACK

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

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

03797642.001 |20:21:09.385 |AppInfo |StationD: (0000114) startMediaTransmission
conferenceID=38960750 passThruPartyID=33554451 remoteIpAddress=IpAddr.type:0
ipAddr:0x0e30201c000000000000000000000000(10.48.32.28) remotePortNumber=17996
milliSecondPacketSize=20 compressType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0
qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e302021(10.48.32.33)

Called (recorded) phone sends CUCM the ORC ACK

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

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

03797655.001 |20:21:09.454 |AppInfo |StationD: (0000109) startMediaTransmission
conferenceID=
38960749 passThruPartyID=33554450 remoteIpAddress=IpAddr.type:0
ipAddr:0x0e302021000000000000000000000000(10.48.32.33) remotePortNumber=32588
milliSecondPacketSize=20 compressType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0
qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e30201c(10.48.32.28)

SIP

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Normal CCM Traces for SIP phone to SIP phone with SIP Integrated Call Recording  
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Calling phone places call

04241111.002 |11:27:41.232 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from
10.48.38.102 on port 50147 index 32 with 1946 bytes:
[286938,NET]
INVITE sip:1001@10.48.38.5;user=phone SIP/2.0
Via: SIP/2.0/TCP 10.48.38.102:50147;branch=z9hG4bK598c2eb2
From: "SJ User 1" <sip:1000@10.48.38.5>;tag=38ed18552a12296c00ff41e8-5fb7856e
To: <sip:1001@10.48.38.5>
Call-ID: 38ed1855-2a120006-78c34baf-1b81d864@10.48.38.102
Max-Forwards: 70
Session-ID: 1001532000105000a00038ed18552a12;remote=00000000000000000000000000000000
Date: Tue, 27 Aug 2019 15:27:42 GMT
CSeq: 101 INVITE
User-Agent: Cisco-CP7861/12.1.1
Contact: <sip:abl7ea6e-8072-927d-aad0-
d10273906106@10.48.38.102:50147;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP38ED18552A12"
Expires: 180
Accept: application/sdp
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "SJ User 1" <sip:1000@10.48.38.5>;party=calling;id-
type=subscriber;privacy=off;screen=yes
Supported: replaces,join,sdp-anat,norefersub,resource-priority,extended-refer,X-cisco-
callinfo,X-cisco-serviceuri,X-cisco-escapecodes,X-cisco-service-control,X-cisco-srtp-fallback,X-
cisco-monrec,X-cisco-config,X-cisco-sis-7.0.0,X-cisco-xsi-8.5.1
Allow-Events: kpml,dialog
Recv-Info: conference
Recv-Info: x-cisco-conference
Content-Length: 687

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

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

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

04241138.007 |11:27:41.238 |AppInfo |Digit analysis: match(pi="2", fqcn="+14085251000", cn="1000", plv="5", pss="EMERGENCY_PT:INTERNAL_PT:SJ_LOCAL_PT:LD_PT:GLOBALIZED_PT", TodFilteredPss="EMERGENCY_PT:INTERNAL_PT:SJ_LOCAL_PT:LD_PT:GLOBALIZED_PT", dd="1001", dac="0")
04241138.008 |11:27:41.238 |AppInfo |Digit analysis: analysis results
04241138.009 |11:27:41.238 |AppInfo ||PretransformCallingPartyNumber=1000
|CallingPartyNumber=1000
|DialingPartition=INTERNAL_PT
|DialingPattern=1001
|FullyQualifiedCalledPartyNumber=+14085251001
|DialingPatternRegularExpression=(1001)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=1001
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=1001
|CollectedDigits=1001

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

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

CUCM sends outbound INVITE to called (recorded) phone

04241178.001 |11:27:41.242 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.107 on port 51902 index 52 [286940,NET]

INVITE sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32e829c48246
From: "SJ User 1" <sip:1000@10.48.38.5>;tag=104952~e650e088-60ba-4195-8387-3dcc0127efdc-19301625
To: <sip:1001@10.48.38.5>
Date: Tue, 27 Aug 2019 15:27:41 GMT
Call-ID: 34241a00-d6514bed-327f-526300e@10.48.38.5
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecallinfo>; security= Unknown; orientation= from; gci= 1-2029;
isVoip; call-instance= 1
Send-Info: conference, x-cisco-conference
Alert-Info: <file://Bellcore-dr1/>
Session-ID: 1001532000105000a00038ed18552a12;remote=00000000000000000000000000000000
Remote-Party-ID: "SJ User 1" <sip:1000@10.48.38.5;x-cisco-callback-
number=1000>;party=calling;screen=yes;privacy=off
Contact:
<sip:1000@10.48.38.5:5060;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP38ED18552A12"
Max-Forwards: 69
Content-Length: 0

Called (recorded) phone returns 200 OK

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

v=0
o=Cisco-SIPUA 899 0 IN IP4 10.48.38.107
s=SIP Call
b=AS:4064
t=0 0
m=audio 20394 RTP/AVP 114 9 113 115 0 8 116 18 101
c=IN IP4 10.48.38.107
b=TIAS:64000
a=rtpmap:114 opus/48000/2

a=fmtp:114 maxplaybackrate=16000;sprop-maxcapture=16000;maxaveragebitrate=64000;stereo=0;sprop-stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0;mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1;mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrec

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

01314344.001 |11:18:48.652 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.32.17 on port 50841 index 17
[106320,NET]

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

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

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

CUCM forwards the 200 OK to the calling phone

04241368.001 |11:27:43.624 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.102 on port 50147 index 32
[286949,NET]
SIP/2.0 200 OK

Via: SIP/2.0/TCP 10.48.38.102:50147;branch=z9hG4bK598c2eb2
From: "SJ User 1" <sip:1000@10.48.38.5>;tag=38ed18552a12296c00ff41e8-5fb7856e
To: <sip:1001@10.48.38.5>;tag=104951~e650e088-60ba-4195-8387-3dcc0127efdc-19301624
Date: Tue, 27 Aug 2019 15:27:41 GMT
Call-ID: 38ed1855-2a120006-78c34baf-1b81d864@10.48.38.102
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Server: Cisco-CUCM11.5
Call-Info: <urn:x-cisco-remotec:callinfo>; security= NotAuthenticated; orientation= to; gci= 1-2029; isVoip; call-instance= 1
Send-Info: conference, x-cisco-conference
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=called;screen=yes;privacy=off
Session-ID: 4313758700105000a0006c416a369525;remote=1001532000105000a00038ed18552a12
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5;user=phone>;party=x-cisco-original-called;privacy=off
Contact:
<sip:1001@10.48.38.5:5060;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP6C416A369525"
Content-Type: application/sdp
Content-Length: 223

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

BiB allocation request on called (recorded) phone

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

BiB allocated on called (recorded) phone

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

DA for first call to activate BiB

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

|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(1,86,15)
|PretransformDigitString=b0018615001
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=b0018615001
|CollectedDigits=b0018615001

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

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

04241449.001 |11:27:43.633 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.107 on port 51902 index 52
[286950,NET]

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

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

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

04241455.002 |11:27:43.697 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.48.38.102 on port 50147 index 32 with 706 bytes:
[286951,NET]

ACK sip:1001@10.48.38.5:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.48.38.102:50147;branch=z9hG4bK688db3c1
From: "SJ User 1" <sip:1000@10.48.38.5>;tag=38ed18552a12296c00ff41e8-5fb7856e
To: <sip:1001@10.48.38.5>;tag=104951~e650e088-60ba-4195-8387-3dcc0127efdc-19301624
Call-ID: 38ed1855-2a120006-78c34baf-1b81d864@10.48.38.102
Max-Forwards: 70
Session-ID: 1001532000105000a00038ed18552a12;remote=4313758700105000a0006c416a369525

Date: Tue, 27 Aug 2019 15:27:45 GMT
CSeq: 101 ACK
User-Agent: Cisco-CP7861/12.1.1
Remote-Party-ID: "SJ User 1" <sip:1000@10.48.38.5>;party=calling;id-
type=subscriber;privacy=off;screen=yes
Content-Length: 0
Recv-Info: conference
Recv-Info: x-cisco-conference

Called (recorded) phone returns 200 OK in response to the invite with "record-
invoker=auto"
Notice the SDP has a=inactive - even though there is no media established on the Bib yet.

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

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

CUCM responds to called (recorded) phone with ACK
04241469.001 |11:27:43.901 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to
10.48.38.107 on port 51902 index 52 [286954,NET] ACK sip:91a43f66-ca58-9cd3-b0e5-
588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0 Via: SIP/2.0/TCP
10.48.38.5:5060;branch=z9hG4bK32eb34dec69 From: "Call Manager"
<sip:10.48.38.5>;tag=104956~e650e088-60ba-4195-8387-3dcc0127efdc-19301628 To:
<sip:1001@10.48.38.5>;tag=6c416a369525007019bf48f9-5901eb85 Date: Tue, 27 Aug 2019 15:27:43 GMT
Call-ID: 35554700-d6514bef-3280-526300e@10.48.38.5 User-Agent: Cisco-CUCM11.5 Max-Forwards: 70
CSeq: 101 ACK Allow-Events: presence Content-Length: 0

BiB places first call to recording destination address (cn is calling party which is the

BiB cn="b0018615001" and it is dialing the recordingdestination dd="7878")

04241501.011 |11:27:43.905 |AppInfo |Digit analysis: match(pi="1", fqcn="",
cn="b0018615001",plv="5", pss="EMERGENCY_PT:INTERNAL_PT",
TodFilteredPss="EMERGENCY_PT:INTERNAL_PT", dd="7878",dac="0")
04241501.012 |11:27:43.905 |AppInfo |Digit analysis: analysis results
04241501.013 |11:27:43.905 |AppInfo ||PretransformCallingPartyNumber=b0018615001
|CallingPartyNumber=b0018615001
|DialingPartition=INTERNAL_PT
|DialingPattern=7878
|FullyQualifiedCalledPartyNumber=7878
|DialingPatternRegularExpression=(7878)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=7878
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=7878
|CollectedDigits=7878

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

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

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

04241555.001 |11:27:43.908 |AppInfo |SIPtcp - wait_SdlSPISignal: Outgoing SIP TCP message to
10.48.38.30 on port 5060 index 50
[286955,NET]
INVITE sip:7878@10.48.38.30:5060 SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ecc2c802c
From: "SJ User 2" <sip:1001@10.48.38.5;x-nearend;x-refci=19301625;x-
nearendclusterid=StandAloneCluster;x-nearenddevice=SEP6C416A369525;x-nearendaddr=1001;x-
farendrefci=19301624;x-farendclusterid=StandAloneCluster;x-farenddevice=SEP38ED18552A12;x-
farendaddr=1000>;tag=104958-e650e088-60ba-4195-8387-3dcc0127efdc-19301629
To: <sip:7878@10.48.38.30>
Date: Tue, 27 Aug 2019 15:27:43 GMT
Call-ID: 35554700-d6514bef-3281-526300e@10.48.38.5
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback

Supported: Geolocation
Call-Info: <sip:10.48.38.5:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Call-Info: <urn:x-cisco-remotecallinfo>;x-cisco-video-traffic-class=DESKTOP
Session-ID: 0848153900105000a0006c416a369525;remote=00000000000000000000000000000000
Cisco-Guid: 0894781184-0000065536-0000000022-0086388750
Session-Expires: 1800
P-Asserted-Identity: "SJ User 2" <sip:1001@10.48.38.5>
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=calling;screen=yes;privacy=off
Contact:
<sip:1001@10.48.38.5:5060;transport=tcp>;isFocus;+u.sip!devicename.ccm.cisco.com="SEP6C416A369525"
Max-Forwards: 70
Content-Length: 0

CUCM sends INVITE #2 to called (recorded) phone with record-invoker=auto in Call-Info field and original Call-ID in Join field

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

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

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

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

04241614.002 |11:27:44.197 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.48.38.107 on port 51902 index 52 with 1434 bytes:
[286959,NET]
SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ed62f39668

From: "Call Manager" <sip:10.48.38.5>;tag=104959~e650e088-60ba-4195-8387-3dcc0127efdc-19301631
To: <sip:1001@10.48.38.5>;tag=6c416a369525007145d433c8-062b13d7
Call-ID: 35554700-d6514bef-3282-526300e@10.48.38.5
Session-ID: 56a8a95e00105000a0006c416a369525;remote=00000000000000000000000000000000
Date: Tue, 27 Aug 2019 15:27:42 GMT
CSeq: 101 INVITE
Server: Cisco-CP7841/12.1.1
Contact: <sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP6C416A369525"
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=called;id-type=subscriber;privacy=off;screen=yes
Supported: replaces,join,sdp-anat,norefersub,resource-priority,extended-refer,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-escapecodes,X-cisco-service-control,X-cisco-srtp-fallback,X-cisco-monrec,X-cisco-config,X-cisco-sis-7.0.0,X-cisco-xsi-8.5.1
Allow-Events: kpml,dialog
Recv-Info: conference
Recv-Info: x-cisco-conference
Content-Length: 219
Content-Type: application/sdp
Content-Disposition: session;handling=optional

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

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

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

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

04241651.011 |11:27:44.201 |AppInfo |Digit analysis: match(pi="1", fqcn="", cn="b0018615001",plv="5", pss="EMERGENCY_PT:INTERNAL_PT", TodFilteredPss="EMERGENCY_PT:INTERNAL_PT", dd="7878",dac="0")
04241651.012 |11:27:44.202 |AppInfo |Digit analysis: analysis results
04241651.013 |11:27:44.202 |AppInfo ||PretransformCallingPartyNumber=b0018615001
|CallingPartyNumber=b0018615001
|DialingPartition=INTERNAL_PT
|DialingPattern=7878
|FullyQualifiedCalledPartyNumber=7878

|DialingPatternRegularExpression=(7878)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=7878
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=7878
|CollectedDigits=7878

CUCM sends INVITE #2 to configured recording server

04241698.001 |11:27:44.205 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.30 on port 5060 index 50
[286961,NET]
INVITE sip:7878@10.48.38.30:5060 SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ef2867938b
From: "SJ User 2" <sip:1001@10.48.38.5;x-farend;x-refci=19301625;x-nearendclusterid=StandAloneCluster;x-nearenddevice=SEP6C416A369525;x-nearendaddr=1001;x-farendrefci=19301624;x-farendclusterid=StandAloneCluster;x-farenddevice=SEP38ED18552A12;x-farendaddr=1000>;tag=104961~e650e088-60ba-4195-8387-3dcc0127efdc-19301632
To: <sip:7878@10.48.38.30>
Date: Tue, 27 Aug 2019 15:27:44 GMT
Call-ID: 35eddd80-d6514bf0-3283-526300e@10.48.38.5
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: <sip:10.48.38.5:5060>;method="NOTIFY";Event=telephone-event;Duration=500"
Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=DESKTOP
Session-ID: 56a8a95e00105000a0006c416a369525;remote=00000000000000000000000000000000
Cisco-Guid: 0904781184-0000065536-0000000023-0086388750
Session-Expires: 1800
P-Asserted-Identity: "SJ User 2" <sip:1001@10.48.38.5>
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=calling;screen=yes;privacy=off
Contact:
<sip:1001@10.48.38.5:5060;transport=tcp>;isFocus;+u.sip!devicename.ccm.cisco.com="SEP6C416A369525"
Max-Forwards: 70
Content-Length: 0

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

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

Allow: INVITE, BYE, CANCEL, ACK, NOTIFY, INFO, UPDATE
Supported: X-cisco-srtp-fallback
Server: MediaSense/11.x

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

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

04241743.002 |11:27:44.326 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.48.38.30 on port 5060 index 50 with 1205 bytes:

[286964,NET]

SIP/2.0 200 Ok

Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ecc2c802c

To: <sip:7878@10.48.38.30>;tag=ds2c967644

From: "SJ User 2" <sip:1001@10.48.38.5;x-nearend;x-refci=19301625;x-

nearendclusterid=StandAloneCluster;x-nearenddevice=SEP6C416A369525;x-nearendaddr=1001;x-

farendrefci=19301624;x-farendclusterid=StandAloneCluster;x-farenddevice=SEP38ED18552A12;x-

farendaddr=1000>;tag=104958~e650e088-60ba-4195-8387-3dcc0127efdc-19301629

Call-ID: 35554700-d6514bef-3281-526300e@10.48.38.5

CSeq: 101 INVITE

Content-Length: 475

Contact: <sip:7878@10.48.38.30:5060;transport=TCP>

Content-Type: application/sdp

Allow: INVITE, BYE, CANCEL, ACK, NOTIFY, INFO, UPDATE

Supported: X-cisco-srtp-fallback

Server: MediaSense/11.x

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

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

Notice there is no SDP

04241825.001 |11:27:44.330 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.107 on port 51902 index 52

[286965,NET]

INVITE sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f014677161
From: "Call Manager" <sip:10.48.38.5>;tag=104959~e650e088-60ba-4195-8387-3dcc0127efdc-19301631
To: <sip:1001@10.48.38.5>;tag=6c416a369525007145d433c8-062b13d7
Date: Tue, 27 Aug 2019 15:27:44 GMT
Call-ID: 35554700-d6514bef-3282-526300e@10.48.38.5
Supported: timer,resource-priority,replaces
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, SUBSCRIBE, NOTIFY
CSeq: 102 INVITE
Max-Forwards: 70
Expires: 180
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecallinfo>; isVoip; record-invoker=auto
Min-SE: 1800
Session-ID: 00000000000000000000000000000000;remote=56a8a95e00105000a0006c416a369525
Remote-Party-ID: "Call Manager" <sip:10.48.38.5>;party=calling;screen=yes;privacy=off
Contact: <sip:10.48.38.5:5060;transport=tcp>
Content-Length: 0

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

04241866.001 |11:27:44.332 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.107 on port 51902 index 52

[286966,NET]

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

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

04241872.002 |11:27:44.541 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.48.38.107 on port 51902 index 52 with 1434 bytes:

[286969,NET]

SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f014677161
From: "Call Manager" <sip:10.48.38.5>;tag=104959~e650e088-60ba-4195-8387-3dcc0127efdc-19301631
To: <sip:1001@10.48.38.5>;tag=6c416a369525007145d433c8-062b13d7
Call-ID: 35554700-d6514bef-3282-526300e@10.48.38.5
Session-ID: 56a8a95e00105000a0006c416a369525;remote=00000000000000000000000000000000
Date: Tue, 27 Aug 2019 15:27:43 GMT
CSeq: 102 INVITE
Server: Cisco-CP7841/12.1.1
Contact: <sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP6C416A369525"

Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=called;id-
type=subscriber;privacy=off;screen=yes
Supported: replaces,join,sdp-anat,norefersub,resource-priority,extended-refer,X-cisco-
callinfo,X-cisco-serviceuri,X-cisco-escapecodes,X-cisco-service-control,X-cisco-srtp-fallback,X-
cisco-monrec,X-cisco-config,X-cisco-sis-7.0.0,X-cisco-xsi-8.5.1
Allow-Events: kpml,dialog
Recv-Info: conference
Recv-Info: x-cisco-conference
Content-Length: 219
Content-Type: application/sdp
Content-Disposition: session;handling=optional

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

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

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

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

a=fmtp:101 0-15
a=sendrecv

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

04241903.001 |11:27:44.552 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.107 on port 51902 index 52
[286971,NET]

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

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

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

04241917.001 |11:27:44.555 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.30 on port 5060 index 50
[286972,NET]

ACK sip:7878@10.48.38.30:5060;transport=TCP SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f373e69393
From: "SJ User 2" <sip:1001@10.48.38.5;x-farend;x-refci=19301625;x-nearendclusterid=StandAloneCluster;x-nearenddevice=SEP6C416A369525;x-nearendaddr=1001;x-farendrefci=19301624;x-farendclusterid=StandAloneCluster;x-farenddevice=SEP38ED18552A12;x-farendaddr=1000>;tag=104961~e650e088-60ba-4195-8387-3dcc0127efdc-19301632
To: <sip:7878@10.48.38.30>;tag=ds1ald776c
Date: Tue, 27 Aug 2019 15:27:44 GMT
Call-ID: 35eddd80-d6514bf0-3283-526300e@10.48.38.5
User-Agent: Cisco-CUCM11.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Session-ID: 56a8a95e00105000a0006c416a369525;remote=c83405810147c69016c38634ab104961
Content-Type: application/sdp
Content-Length: 235

v=0
o=CiscoSystemsCCM-SIP 104961 1 IN IP4 10.48.38.5
s=SIP Call
c=IN IP4 10.48.38.107
b=TIAS:64000
b=AS:64
t=0 0

m=audio 17904 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=sendonly
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

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

04241947.001 |11:27:44.559 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.107 on port 51902 index 52
[286973,NET]
ACK sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f45d25b711
From: "Call Manager" <sip:10.48.38.5>;tag=104956~e650e088-60ba-4195-8387-3dcc0127efdc-19301628
To: <sip:1001@10.48.38.5>;tag=6c416a369525007019bf48f9-5901eb85
Date: Tue, 27 Aug 2019 15:27:44 GMT
Call-ID: 35554700-d6514bef-3280-526300e@10.48.38.5
User-Agent: Cisco-CUCM11.5
Max-Forwards: 70
CSeq: 102 ACK
Allow-Events: presence
Session-ID: 00000000000000000000000000000000;remote=0848153900105000a0006c416a369525
Content-Type: application/sdp
Content-Length: 192

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

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

04241948.001 |11:27:44.559 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.30 on port 5060 index 50
[286974,NET]
ACK sip:7878@10.48.38.30:5060;transport=TCP SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f573871bbb
From: "SJ User 2" <sip:1001@10.48.38.5;x-nearend;x-refci=19301625;x-nearendclusterid=StandAloneCluster;x-nearenddevice=SEP6C416A369525;x-nearendaddr=1001;x-farendrefci=19301624;x-farendclusterid=StandAloneCluster;x-farenddevice=SEP38ED18552A12;x-farendaddr=1000>;tag=104958~e650e088-60ba-4195-8387-3dcc0127efdc-19301629
To: <sip:7878@10.48.38.30>;tag=ds2c967644
Date: Tue, 27 Aug 2019 15:27:43 GMT
Call-ID: 35554700-d6514bef-3281-526300e@10.48.38.5
User-Agent: Cisco-CUCM11.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Session-ID: 0848153900105000a0006c416a369525;remote=c83405810147c69016c38634ab104958
Content-Type: application/sdp
Content-Length: 235

v=0
o=CiscoSystemsCCM-SIP 104958 1 IN IP4 10.48.38.5
s=SIP Call
c=IN IP4 10.48.38.107

```
b=TIAS:64000
b=AS:64
t=0 0
m=audio 26396 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=sendonly
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

Troubleshoot

Esta seção disponibiliza informações para a solução de problemas de configuração.

Negociação de codec

Este é um exemplo de um dos tipos mais comuns de falhas de gravação de chamada - incompatibilidade de codec entre o telefone gravado e o servidor de 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_G711Ululaw64k) RFC2833PayloadType=0 qualifierIn=?
sourceIpAddr=IpAddr.type:0 ipAddr:0x0e302021000000000000000000000000(10.48.32.33). myIP:
IpAddr.type:0 ipv4Addr:0x0e30201c(10.48.32.28)

Codec locked due to recording on called (recorded) phone

00019785.003 |12:48:36.062 |AppInfo | StationCdp: star_MediaExchangeAgenaQueryCapability - Device SEP001795BDD16B, codec locked due to recording, codecType=4

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

00019788.002 |12:48:36.062 |AppInfo |StationD: (0000007) OpenReceiveChannel
conferenceID=49613638 passThruPartyID=33554434 millisecondPacketSize=20
compressionType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0 qualifierIn=?
sourceIpAddr=IpAddr.type:0 ipAddr:0x0e30201c000000000000000000000000(10.48.32.28). myIP:
IpAddr.type:0 ipv4Addr:0x0e302021(10.48.32.33)

CUCM allocates the BiB on the called (recorded) phone

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

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

00019889.001 |12:48:36.100 |AppInfo |Digit Analysis: star_DaReq: daReq.partitionSearchSpace(),
filteredPartitionSearchSpaceString(), partitionSearchSpaceString()
00019889.002 |12:48:36.100 |AppInfo |Digit Analysis: star_DaReq: Matching Legacy Numeric,
digits=8675309
00019889.003 |12:48:36.100 |AppInfo |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept
DAMR.sstype=[0], TPcount=[0], DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]
00019889.004 |12:48:36.100 |AppInfo |Digit Analysis: getDaRes - Remote Destination [8675309]
isURI[0]
00019889.005 |12:48:36.100 |AppInfo |CMUtility routeCallThroughCTIRD: no matching
RemDestDynamic record exists for remdest [8675309]
00019889.006 |12:48:36.100 |AppInfo |DbMobility: getMatchedRemDest starts: cnumber = 8675309
00019889.007 |12:48:36.100 |AppInfo |DbMobility: getMatchedRemDest: full match case
00019889.008 |12:48:36.100 |AppInfo |DbMobility SelectByDestination: no matching RemDestDynamic
record exists for remdest [8675309]
00019889.009 |12:48:36.100 |AppInfo |DbMobility: can't find remdest 8675309 in map
00019889.010 |12:48:36.100 |AppInfo |Digit analysis: patternUsage=5
00019889.011 |12:48:36.100 |AppInfo |Digit analysis: match(pi="1", fqcn="",
cn="b00223906001",plv="5", pss="E911_PT:Phones_PT:EMERGENCY_PT:INTERNAL_PT:INFORMACAST_PT",
TodFilteredPss="E911_PT:Phones_PT:EMERGENCY_PT:INTERNAL_PT:INFORMACAST_PT",
dd="8675309",dac="1")
00019889.012 |12:48:36.100 |AppInfo |Digit analysis: analysis results
00019889.013 |12:48:36.100 |AppInfo ||PretransformCallingPartyNumber=b00223906001
|CallingPartyNumber=b00223906001
|DialingPartition=
|DialingPattern=8675309
|FullyQualifiedCalledPartyNumber=8675309
|DialingPatternRegularExpression=(8675309)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=8675309
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=8675309
|CollectedDigits=8675309

Calling phone sends CUCM the ORC ACK

00019912.001 |12:48:36.139 |AppInfo |StationInit: (0000005) OpenReceiveChannelAck Status=0,

IpAddr=IpAddr.type:0 ipAddr:0x0e30201c000000000000000000000000(10.48.32.28), Port=31678,
PartyID=33554433

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

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

Called (recorded) phone sends CUCM the ORC ACK

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

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

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

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

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

00020086.001 |12:48:36.156 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message
to 10.48.32.170:[5060]:
[901,NET]
INVITE sip:8675309@10.48.32.170:5060 SIP/2.0
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK4f2a857d3d
From: <sip:9110001@10.48.32.90;x-nearend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-

nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=351~713e2333-4032-45f1-b1f5-e33cf47lacec-49613642
To: <sip:8675309@10.48.32.170>
Date: Tue, 14 Oct 2014 16:48:36 GMT
Call-ID: ef7acf80-43d153e4-50-5a20300e@10.48.32.90
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: ;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 4017803136-0000065536-0000000001-1512058894
Session-Expires: 1800
P-Asserted-Identity: <sip:9110001@10.48.32.90>
Remote-Party-ID: <sip:9110001@10.48.32.90>;party=calling;screen=yes;privacy=off
Contact: <sip:9110001@10.48.32.90:5060>;isFocus
Max-Forwards: 70
Content-Length: 0

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

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

[902,NET]

INVITE sip:8675309@10.48.32.170:5060 SIP/2.0

Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hg4bk5014378d0b

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

To: <sip:8675309@10.48.32.170>

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

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

Supported: timer,resource-priority,replaces

Min-SE: 1800

User-Agent: Cisco-CUCM10.5

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

CSeq: 101 INVITE

Expires: 180

Allow-Events: presence, kpml

Supported: X-cisco-srtp-fallback

Supported: Geolocation

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

Cisco-Guid: 4017803136-0000065536-0000000002-1512058894

Session-Expires: 1800

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

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

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

Max-Forwards: 70

Content-Length: 0

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

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

[903,NET]

SIP/2.0 200 OK

Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hg4bk4f2a857d3d

From: <sip:9110001@10.48.32.90;x-nearend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=351~713e2333-4032-45f1-b1f5-e33cf47lacec-49613642
To: <sip:8675309@10.48.32.170>;tag=1
Call-ID: ef7acf80-43d153e4-50-5a20300e@10.48.32.90
CSeq: 101 INVITE
Contact: <sip:10.48.32.170:5060;transport=udp>
Content-Type: application/sdp
Content-Length: 135

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

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

00020092.001 |12:48:36.161 |AppInfo |//SIP/SIPUdp/wait_SdlDataInd: Incoming SIP UDP message size 730 from 10.48.32.170:[5060]:

[905,NET]

SIP/2.0 200 OK

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

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

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

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

CSeq: 101 INVITE

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

Content-Type: application/sdp

Content-Length: 135

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

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

CUCM determines 2 transcoders are required and attempts to allocate

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

No transcoder is configured which can cause this call to fail

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

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

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

[906,NET]

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

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

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

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

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

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

User-Agent: Cisco-CUCM10.5

Max-Forwards: 70

CSeq: 101 ACK

Allow-Events: presence, kpml

Content-Length: 0

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

[907,NET]

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

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

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

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

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

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

User-Agent: Cisco-CUCM10.5

Max-Forwards: 70

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

CSeq: 102 BYE

Reason: Q.850;cause=47

Content-Length: 0

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

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

[908,NET]

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

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

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

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

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

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

User-Agent: Cisco-CUCM10.5

Max-Forwards: 70

CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Length: 0

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

Configuração incorreta que inclui problemas de CSS e PT

Os comandos aqui permitem que a maioria das configurações de gravação sejam revisadas rapidamente apenas com o endereço MAC conhecido de um telefone que não esteja gravando chamadas. Basta substituir a parte do comando **MAC_of_Phone** pelo endereço MAC real do telefone, como nos exemplos vistos aqui.

Isso fornece o DN (todos eles, se houver mais de um) para o MAC que você procura, o MAC do telefone apenas para confirmação, a configuração BIB, a configuração de privacidade, o tipo de gravação (faça referência aos valores listados nos exemplos do laboratório), o perfil de gravação em uso pelo telefone, o nome dos Espaços de Pesquisa de Chamadas (CSS) de gravação, o destino de gravação para esse perfil de gravação e a partição à qual o destino de gravação está associado com base no MAC que você procura:

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

Isso fornece a lista de partições associadas ao CSS de gravação no perfil de gravação associado ao MAC do telefone que você procura.

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

Aqui estão exemplos da saída do laboratório para um telefone com endereço MAC

SEPC80084AA8743:

Nesse comando, você pode ver que o telefone tem apenas um DN nele, que é 2003, também vemos que o BIB está ativado, a privacidade está desativada, o tipo de gravação é automático, a origem preferencial é o telefone, o perfil de gravação é **Test Recording Profile**, o espaço de pesquisa de chamada de gravação é **INTERNAL_CSS**, o padrão de rota para chamadas gravadas é 8675309 e esse padrão está associado à partição **INTERNAL_PT**.

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

Com a saída desse comando, você pode verificar todas as partições do CSS de gravação e do perfil de gravação associado ao telefone de interesse. Você pode ver aqui que a partição **INTERNAL_PT** é uma das partições associadas ao espaço de pesquisa de chamada **INTERNAL_CSS**. Isso significa que não deve haver problemas com o BIB do telefone que é capaz de chamar o padrão de rota de 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 callingsearchspace_member as csm on
csm.fkcallingsearchspace = css.pkid inner join routepartition as rp on csm.fkroutepartition =
rp.pkid inner join recordingprofile as rcrdpro on rcrdpro.fkcallingsearchspace_callrecording =
css.pkid inner join devicenumplanmap as devnumplanmap on rcrdpro.pkid =
devnumplanmap.fkreordingprofile inner join device as dev on devnumplanmap.fkdevice = dev.pkid
where css.pkid = rcrdpro.fkcallingsearchspace_callrecording and dev.name='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
```

Informações Relacionadas

- [Designs de rede de referência da solução Cisco Collaboration System 11.x \(SRND\)](#)

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