

Configurez et dépannez l'enregistrement d'appels de base

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Introduction

Ce document décrit les fondements de l'enregistrement d'appels dans Cisco Unified Communications Manager (CUCM), les medias prévus circulent, les écoulements prévus d'appel pour des périphériques de Protocole SIP (Session Initiation Protocol) et de Skinny Client Control Protocol (SCCP), et un exemple d'un type commun de panne d'installation d'enregistrement d'appels.

Conditions préalables

Conditions requises

Cisco recommande que vous ayez la connaissance de CUCM intégré avec un tiers serveur d'enregistrement.

[Composants utilisés](#)

Les informations contenues dans ce document sont basées sur les versions de matériel et de logiciel suivantes :

- CUCM
- Procotole IP (Internet Protocol) de Cisco
- Serveur d'enregistrement d'appel téléphonique

Les informations contenues dans ce document ont été créées à partir des périphériques d'un environnement de laboratoire spécifique. Tous les périphériques utilisés dans ce document ont démarré avec une configuration effacée (par défaut). Si votre réseau est vivant, assurez-vous que vous comprenez l'impact potentiel de n'importe quelle commande.

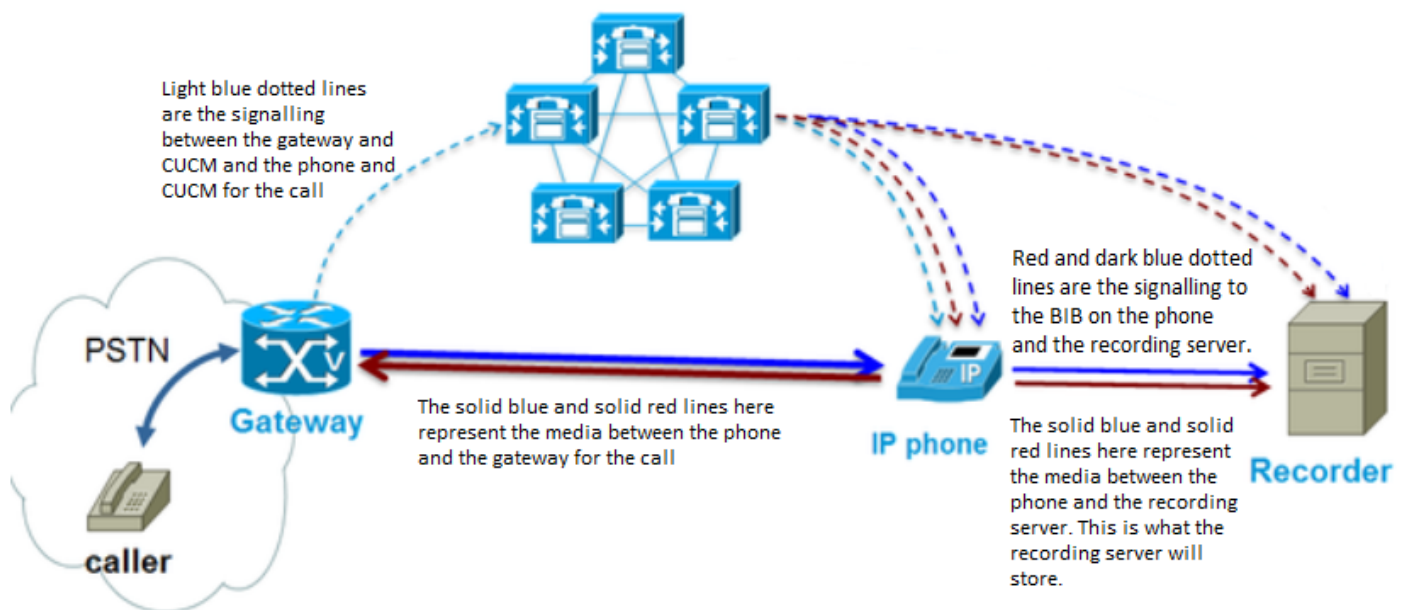
Types d'enregistrement d'appels

Automatique

Les éléments principaux de l'enregistrement d'appels automatique sont comme suit :

- Construire-Dans-passerelle d'utilisations (BAVOIR) de téléphone IP afin de bifurquer audio à la destination d'enregistrement
- Initié chaque fois que le téléphone IP place un appel ou reçoit un appel
- Exige seulement un joncteur réseau de SIP entre CUCM et destination d'enregistrement. Quelques constructeurs d'enregistrement ont besoin du couplage de la téléphonie et de l'informatique (CTI)
- Ne permet pas l'enregistrement des téléphones qui se trouvent en dehors de du réseau administré (doit avoir accès pour envoyer le RTP directement au serveur l'enregistrement et à être un téléphone IP de Cisco capable d'allouer un BAVOIR)

Dans ce diagramme, les lignes continues représentent les medias prévus circulent et les lignes tirées représentent l'écoulement de signalisation prévu :

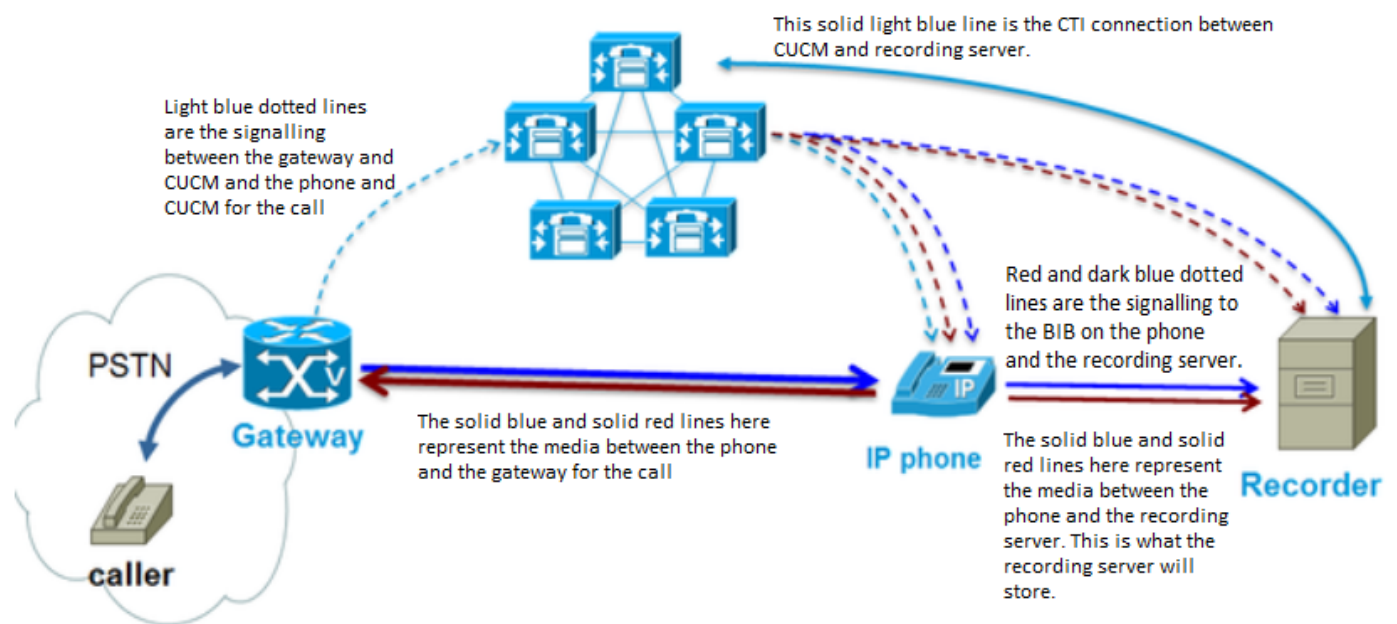


Application appelée

Les éléments principaux de l'enregistrement d'appels appelé par application sont comme suit :

- Utilise le BAVOIR du téléphone IP afin de bifurquer audio à la destination d'enregistrement
- Initié quand l'application (enregistreur) dicte qu'elle doit être initiée
- Exige le joncteur réseau de SIP et le CTI avec l'application d'enregistrement
- L'utilisateur d'application CTI doit avoir accès aux points finaux qui doivent être enregistrés
- Ne permet pas l'enregistrement des téléphones qui se trouvent en dehors de du réseau administré (doit avoir accès pour envoyer le RTP directement au serveur l'enregistrement)

Dans le diagramme ici, les lignes continues représentent les medias prévus circulent et les lignes tirées représentent l'écoulement de signalisation prévu. La ligne continue entre CUCM et le serveur d'enregistrement dénote une connexion CTI entre CUCM et l'application.

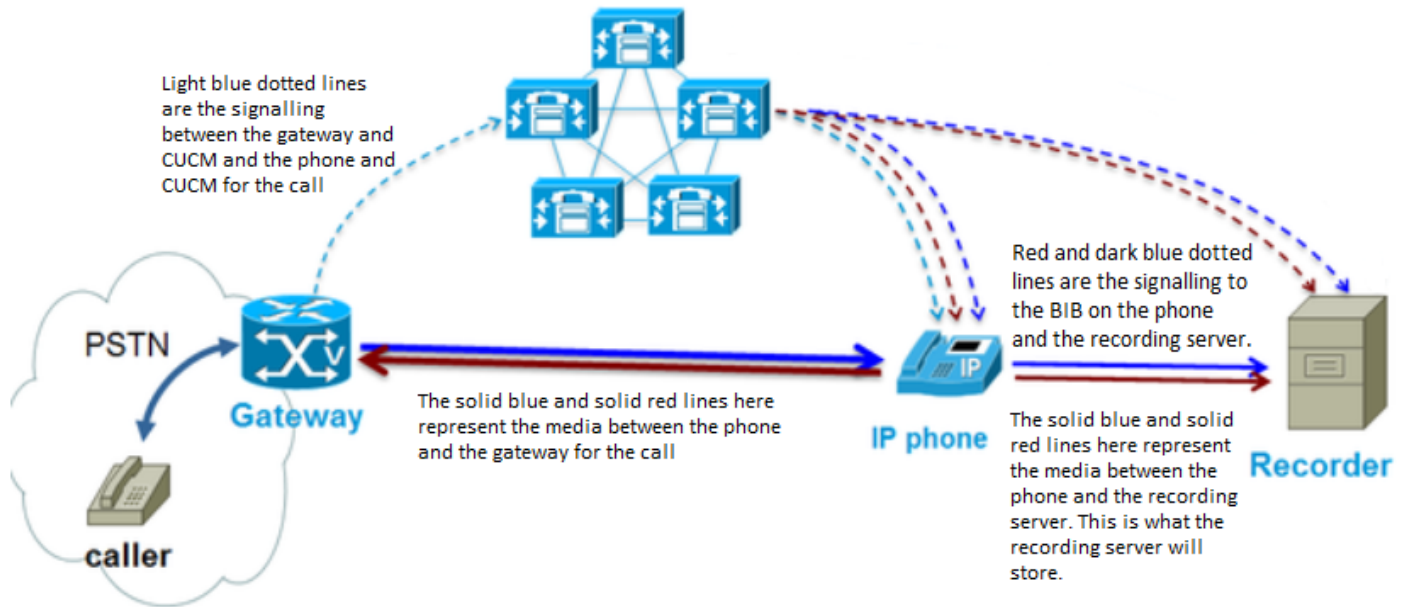


Sélectif

Les éléments principaux de l'enregistrement d'appels sélectif sont comme suit :

- Utilise le BAVOIR du téléphone IP afin de bifurquer audio à la destination d'enregistrement
- Initié chaque fois que l'utilisateur de téléphone IP sélectionne l'option d'enregistrement sur leur téléphone IP (CUCM 9.x+) ou sur une application comme dans [cette image](#)
- Exige typiquement seulement un joncteur réseau de SIP entre CUCM et destination d'enregistrement (qui dépend du fournisseur d'applications d'enregistrement)
- Ne permet pas l'enregistrement des téléphones qui se trouvent en dehors de du réseau administré (doit avoir accès pour envoyer le RTP directement au serveur l'enregistrement)

Comme vous pouvez voir dans ce diagramme ici, le support et le circuit est très semblable à l'enregistrement d'appels automatique :

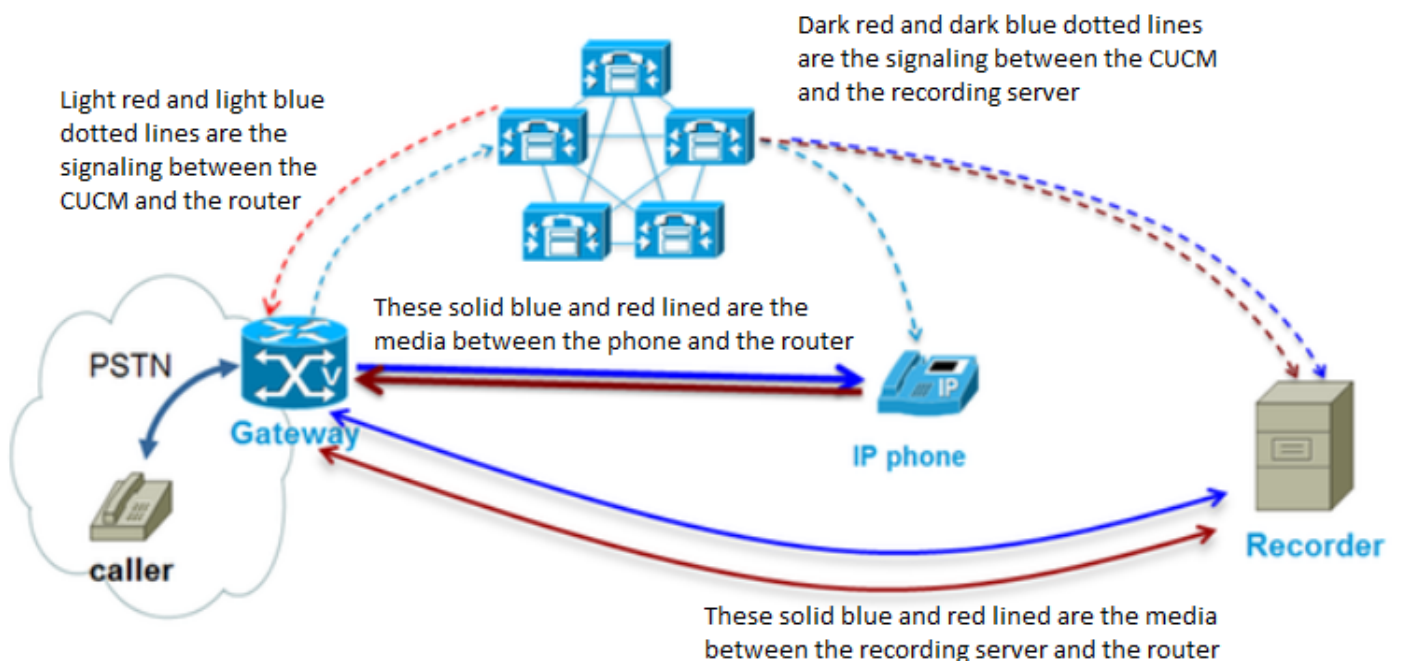


Basé sur passerelle

Les éléments principaux de l'enregistrement d'appels basé sur passerelle sont comme suit :

- Exprimez la passerelle bifurque les medias vers la destination d'enregistrement
- Inscriptions CUCM à la passerelle comme application
- CUCM emploie le HTTP afin de demander à la passerelle (gw) pour couler des medias à la destination de enregistrement
- CUCM intègre avec la destination d'enregistrement par l'intermédiaire du joncteur réseau de SIP
- Permet l'enregistrement des appels qui traversent simplement le réseau administré (par exemple, aux utilisateurs nomades) ou pour les téléphones qui ne prennent en charge pas le BAVOIR

Comme vous pouvez voir du diagramme ici, les medias circulent sont très différents des autres types d'enregistrement d'appels :

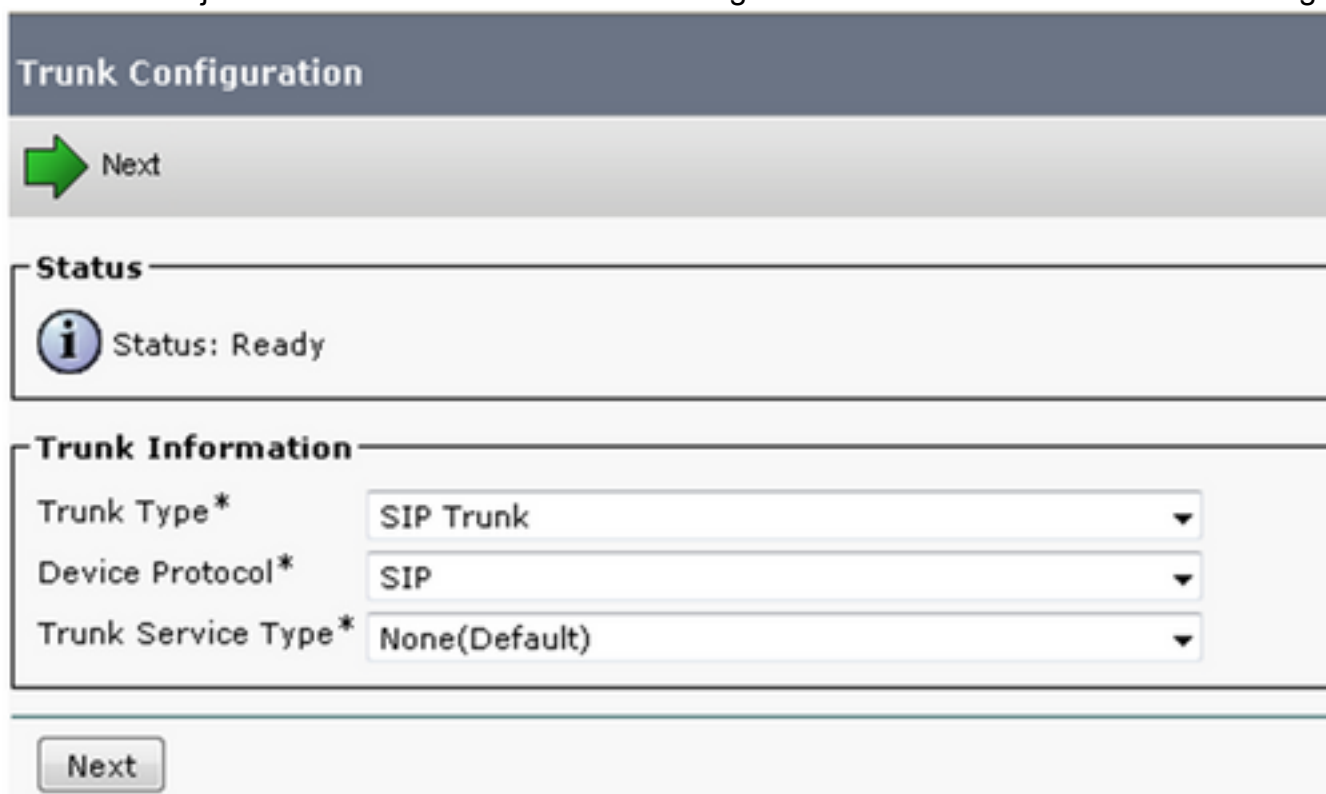


Configuration automatique d'enregistrement d'appels pour l'intégration de SIP seulement

Cette section décrit comment installer l'intégration de SIP d'un serveur d'enregistrement.

Créez le joncteur réseau de SIP à la destination de enregistrement

- Naviguez vers le **périphérique > le joncteur réseau**, choisis **ajoutent nouveau**.
- Créez un joncteur réseau de SIP avec les configurations suivant les indications de l'image.



The screenshot shows the 'Trunk Configuration' page. At the top, there is a green arrow labeled 'Next'. Below this, the 'Status' section shows an information icon and the text 'Status: Ready'. The 'Trunk Information' section contains three dropdown menus: 'Trunk Type*' is set to 'SIP Trunk', 'Device Protocol*' is set to 'SIP', and 'Trunk Service Type*' is set to 'None(Default)'. At the bottom of the form, there is a 'Next' button.

- Entrez le nom de périphérique approprié, le Pool d'appareils, le profil de Sécurité de joncteur réseau MRGL, de SIP, et le profil de SIP
- L'adresse de destination configurée est l'adresse du serveur d'applications d'enregistrement. Dans l'exemple ici, le serveur d'enregistrement est **14.48.32.170** suivant les indications de l'image.







The screenshot shows the 'SIP Information' page. Under the 'Destination' section, there is a checkbox labeled 'Destination Address is an SRV' which is unchecked. Below this, there are three input fields: 'Destination Address' with the value '14.48.32.170', 'Destination Address IPv6' which is empty, and 'Destination Port' with the value '5060'.


Créez le profil d'enregistrement

- Naviguez vers le **périphérique > les paramètres de périphérique > le profil d'enregistrement**
- L'adresse de destination de enregistrement est où les appels d'enregistrement sont envoyés suivant les indications de l'image.

Recording Profile Configuration

 Save
  Delete
  Copy
  Add New

Status

 Status: Ready

Recording Profile Information

Name*

Recording Calling Search Space

Recording Destination Address *

Créez le modèle d'artère pour conduire des appels d'enregistrement

- Créez un modèle d'artère qui apparie l'adresse de destination d'enregistrement configurée dans l'étape précédente
- Vous pouvez indiquer une liste de routage au lieu de directement au joncteur réseau de SIP, si vous souhaitez configurer les joncteurs réseau redondants de SIP

Note: La partition assignée à ce modèle d'artère doit être associée avec l'enregistrement appelle l'espace de recherche et suivant les indications de l'image.

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

Assignez le profil d'enregistrement à la ligne téléphonique

- À un téléphone déjà créé avec une extension existante, assignez le profil d'enregistrement créé
- Assignez le type d'enregistrement d'appels dans cet emplacement aussi bien
- L'exemple affiche l'enregistrement automatique, suivant les indications de l'image.

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

Placez le BAVOIR à en fonction et l'intimité à hors fonction à la page de configuration de téléphone

Tandis que sur la page de configuration de périphérique, naviguez vers la section intitulée **l'information sur le périphérique**. Placez construit dans la passerelle à **en fonction** et l'intimité à **hors fonction** suivant les indications de l'image.

Built In Bridge*	On
Privacy*	Off

Vérifiez

Utilisez cette section pour confirmer que votre configuration fonctionne correctement.

Voici les comportements prévus dans les suivis de gestionnaire d'appel pour le SCCP et les téléphones SIP avec la configuration donnée. Ces exemples sont pour un téléphone appelle un autre téléphone sur la même batterie tandis qu'un des téléphones est installé pour l'enregistrement d'appels.

SCCP

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

### Calling phone places call

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

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

03797017.001 |20:21:08.057 |AppInfo |Digit Analysis: star_DaReq: daReq.partitionSearchSpace(),
filteredPartitionSearchSpaceString(), partitionSearchSpaceString()
03797017.002 |20:21:08.057 |AppInfo |Digit Analysis: star_DaReq: Matching Legacy Numeric,
digits=9110001
03797017.003 |20:21:08.057 |AppInfo |Digit Analysis: getDaRes data&colon; daRes.ssType=[0]
Intercept DAMR.sstype=[0], TPcount=[0], DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]
03797017.004 |20:21:08.057 |AppInfo |Digit Analysis: getDaRes - Remote Destination [] isURI[1]
03797017.005 |20:21:08.057 |AppInfo |Digit analysis: patternUsage=2
03797017.006 |20:21:08.057 |AppInfo |Digit analysis: match(pi="2", fqcn="9110006",
cn="9110006",plv="5", pss="", TodFilteredPss="", dd="9110001",dac="0")
03797017.007 |20:21:08.057 |AppInfo |Digit analysis: analysis results
03797017.008 |20:21:08.057 |AppInfo ||PretransformCallingPartyNumber=9110006
|CallingPartyNumber=9110006

```

|DialingPartition=
|DialingPattern=9110001
|FullyQualifiedCalledPartyNumber=9110001
|DialingPatternRegularExpression=(9110001)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=9110001
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=9110001
|CollectedDigits=9110001

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

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

CUCM extends call to phone

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

Called (recorded) phone goes off hook

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

CUCM Tells the calling phone to open the logical channel

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

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

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

CUCM Tells the calling phone to open the receive channel

03797164.002 |20:21:09.337 |AppInfo |StationD: (0000109) OpenReceiveChannel
conferenceID=38960749 passThruPartyID=33554450 millisecondPacketSize=20
compressionType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0 qualifierIn=?
sourceIpAddr=IpAddr.type:0 ipAddr:0x0e302021000000000000000000000000(14.48.32.33). myIP:
IpAddr.type:0 ipv4Addr:0x0e30201c(14.48.32.28)

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

03797168.002 |20:21:09.337 |AppInfo |StationD: (0000114) OpenReceiveChannel
conferenceID=38960750 passThruPartyID=33554451 millisecondPacketSize=20
compressionType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0 qualifierIn=?
sourceIpAddr=IpAddr.type:0 ipAddr:0x0e30201c000000000000000000000000(14.48.32.28). myIP:
IpAddr.type:0 ipv4Addr:0x0e302021(14.48.32.33)

CUCM allocates BIB on called (recorded) phone

03797210.000 |20:21:09.338 |SdlSig |MrmAllocateUcbResourceReq |waiting
|MediaResourceManager(2,100,138,1) |Cc(2,100,220,1)
|2,100,14,8384.91^14.48.32.33^SEP001795BDD16B |[R:N-H:0,N:1,L:0,V:0,Z:0,D:0] CI=38960751
SsType=33554461 SsKey=9 BridgeType=0 MRGLPkid= NumStream=1 Bib=89cdb152-4ef2-4d60-9e6b-
ab8c77c22618 BibTgCi=38960750 FeatId=159 PL=5 PLDmn=0 DeviceCapability=0 NumVideoCapable=0
requestDeviceType=0 requestDeviceLocale=64 forkingDevicePosition=2 playToneDir=3

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

03797269.001 |20:21:09.340 |AppInfo |Digit Analysis: star_DaReq: daReq.partitionSearchSpace(),
filteredPartitionSearchSpaceString(), partitionSearchSpaceString()
03797269.002 |20:21:09.340 |AppInfo |Digit Analysis: star_DaReq: Matching Legacy Numeric,
digits=8675309
03797269.003 |20:21:09.340 |AppInfo |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept
DAMR.sstype=[0], TPcount=[0], DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]
03797269.004 |20:21:09.340 |AppInfo |Digit Analysis: getDaRes - Remote Destination [8675309]
isURI[0]
03797269.005 |20:21:09.340 |AppInfo |CMUtility routeCallThroughCTIRD: no matching
RemDestDynamic record exists for remdest [8675309]
03797269.006 |20:21:09.340 |AppInfo |DbMobility: getMatchedRemDest starts: cnumber = 8675309
03797269.007 |20:21:09.340 |AppInfo |DbMobility: getMatchedRemDest: full match case
03797269.008 |20:21:09.340 |AppInfo |DbMobility SelectByDestination: no matching RemDestDynamic
record exists for remdest [8675309]
03797269.009 |20:21:09.340 |AppInfo |DbMobility: can't find remdest 8675309 in map
03797269.010 |20:21:09.340 |AppInfo |Digit analysis: patternUsage=5
03797269.011 |20:21:09.340 |AppInfo |Digit analysis: match(pi="1", fqcn="",
cn="b00223908001", plv="5", pss="E911_PT:Phones_PT:EMERGENCY_PT:INTERNAL_PT:INFORMACAST_PT",
TodFilteredPss="E911_PT:Phones_PT:EMERGENCY_PT:INTERNAL_PT:INFORMACAST_PT",
dd="8675309", dac="0")
03797269.012 |20:21:09.340 |AppInfo |Digit analysis: analysis results
03797269.013 |20:21:09.340 |AppInfo ||PretransformCallingPartyNumber=b00223908001
|CallingPartyNumber=b00223908001
|DialingPartition=
|DialingPattern=8675309
|FullyQualifiedCalledPartyNumber=8675309
|DialingPatternRegularExpression=(8675309)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=8675309

|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=8675309
|CollectedDigits=8675309

CUCM sends INVITE #1 to configured recording server (14.48.32.170)

03797320.001 |20:21:09.343 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 14.48.32.170:[5060]:
[212231,NET]
INVITE sip:8675309@14.48.32.170:5060 SIP/2.0
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK204d520fedb3
From: <sip:9110001@14.48.32.90;x-nearend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=38960749;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=73601~713e2333-4032-45f1-b1f5-e33cf471acec-38960754
To: <sip:8675309@14.48.32.170>
Date: Tue, 30 Sep 2014 00:21:09 GMT
Call-ID: abbb8e00-4291f775-204c-5a20300e@14.48.32.90
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: ;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 2881195520-0000065536-0000000011-1512058894
Session-Expires: 1800
P-Asserted-Identity: <sip:9110001@14.48.32.90>
Remote-Party-ID: <sip:9110001@14.48.32.90>;party=calling;screen=yes;privacy=off
Contact: <sip:9110001@14.48.32.90:5060>;isFocus
Max-Forwards: 70
Content-Length: 0

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

03797367.010 |20:21:09.344 |AppInfo |Digit analysis: patternUsage=5
03797367.011 |20:21:09.344 |AppInfo |Digit analysis: match(pi="1", fqcn="",
cn="b00223908001",plv="5", pss="E911_PT:Phones_PT:EMERGENCY_PT:INTERNAL_PT:INFORMACAST_PT",
TodFilteredPss="E911_PT:Phones_PT:EMERGENCY_PT:INTERNAL_PT:INFORMACAST_PT",
dd="8675309",dac="0")
03797367.012 |20:21:09.344 |AppInfo |Digit analysis: analysis results
03797367.013 |20:21:09.344 |AppInfo ||PretransformCallingPartyNumber=b00223908001
|CallingPartyNumber=b00223908001
|DialingPartition=
|DialingPattern=8675309
|FullyQualifiedCalledPartyNumber=8675309
|DialingPatternRegularExpression=(8675309)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=8675309
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=8675309
|CollectedDigits=8675309

CUCM receives 200 OK in response to INVITE #1

03797390.001 |20:21:09.345 |AppInfo |//SIP/SIPUdp/wait_SdlDataInd: Incoming SIP UDP message
size 737 from 14.48.32.170:[5060]:
[212232,NET]
SIP/2.0 200 OK
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK204d520fedb3
From: <sip:9110001@14.48.32.90;x-nearend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-
nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=38960749;x-
farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-
farendaddr=9110006>;tag=73601~713e2333-4032-45f1-b1f5-e33cf471acec-38960754
To: <sip:8675309@14.48.32.170>;tag=1
Call-ID: abbb8e00-4291f775-204c-5a20300e@14.48.32.90
CSeq: 101 INVITE
Contact: <sip:14.48.32.170:5060;transport=udp>
Content-Type: application/sdp
Content-Length: 135

v=0
o=user1 53655765 2353687637 IN IP4 14.48.32.170
s=-
c=IN IP4 14.48.32.170
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000

CUCM sends INVITE #2 to recording server (14.48.32.170)

03797445.001 |20:21:09.348 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message
to 14.48.32.170:[5060]:
[212233,NET]
INVITE sip:8675309@14.48.32.170:5060 SIP/2.0
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK204e754eaeae
From: <sip:9110001@14.48.32.90;x-farend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-
nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=38960749;x-
farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-
farendaddr=9110006>;tag=73602~713e2333-4032-45f1-b1f5-e33cf471acec-38960757
To: <sip:8675309@14.48.32.170>
Date: Tue, 30 Sep 2014 00:21:09 GMT
Call-ID: abbb8e00-4291f775-204d-5a20300e@14.48.32.90
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: ;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 2881195520-0000065536-0000000012-1512058894
Session-Expires: 1800
P-Asserted-Identity: <sip:9110001@14.48.32.90>
Remote-Party-ID: <sip:9110001@14.48.32.90>;party=calling;screen=yes;privacy=off
Contact: <sip:9110001@14.48.32.90:5060>;isFocus
Max-Forwards: 70
Content-Length: 0

CUCM receives 200 OK in response to INVITE #2

03797498.001 |20:21:09.350 |AppInfo |//SIP/SIPUdp/wait_SdlDataInd: Incoming SIP UDP message
size 736 from 14.48.32.170:[5060]:
[212235,NET]
SIP/2.0 200 OK

Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK204e754eaeae
From: <sip:9110001@14.48.32.90;x-farend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=38960749;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=73602~713e2333-4032-45f1-b1f5-e33cf471acec-38960757
To: <sip:8675309@14.48.32.170>;tag=2
Call-ID: abbb8e00-4291f775-204d-5a20300e@14.48.32.90
CSeq: 101 INVITE
Contact: <sip:14.48.32.170:5060;transport=udp>
Content-Type: application/sdp
Content-Length: 135

v=0
o=user1 53655765 2353687637 IN IP4 14.48.32.170
s=-
c=IN IP4 14.48.32.170
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000

CUCM sends outbound ACK in response to 200 OK #1

03797500.001 |20:21:09.351 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 14.48.32.170:[5060]:

[212236,NET]

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

Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK204f50bef815
From: <sip:9110001@14.48.32.90;x-nearend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=38960749;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=73601~713e2333-4032-45f1-b1f5-e33cf471acec-38960754
To: <sip:8675309@14.48.32.170>;tag=1
Date: Tue, 30 Sep 2014 00:21:09 GMT
Call-ID: abbb8e00-4291f775-204c-5a20300e@14.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Type: application/sdp
Content-Length: 254

v=0
o=CiscoSystemsCCM-SIP 73601 1 IN IP4 14.48.32.90
s=SIP Call
c=IN IP4 14.48.32.33
b=TIAS:64000
b=CT:64
b=AS:64
t=0 0
m=audio 4000 RTP/AVP 0 101
a=ptime:20
a=rtpmap:0 PCMU/8000
a=sendonly
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

CUCM sends startMediaTransmission to the called (recorded) phone telling the phone to send RTP to recording server (14.48.32.170)

03797479.001 |20:21:09.350 |AppInfo |StationD: (0000114) startMediaTransmission
conferenceID=38960750 passThruPartyID=33554452 remoteIpAddress=IpAddr.type:0
ipAddr:0x0e3020aa000000000000000000000000(14.48.32.170) remotePortNumber=6000
milliSecondPacketSize=20 compressType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0

qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e302021(14.48.32.33)

CUCM sends startMediaTransmission #2 to the called (recorded) phone telling the phone to send RTP to recording server (14.48.32.170)

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

CUCM sends outbound ACK in response to 200 OK #2

03797615.001 |20:21:09.354 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message
to 14.48.32.170:[5060]:
[212237,NET]
ACK sip:14.48.32.170:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK2050183495f1
From: <sip:9110001@14.48.32.90;x-farend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-
nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=38960749;x-
farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-
farendaddr=9110006>;tag=73602~713e2333-4032-45f1-b1f5-e33cf471acec-38960757
To: <sip:8675309@14.48.32.170>;tag=2
Date: Tue, 30 Sep 2014 00:21:09 GMT
Call-ID: abbb8e00-4291f775-204d-5a20300e@14.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Type: application/sdp
Content-Length: 254

v=0
o=CiscoSystemsCCM-SIP 73602 1 IN IP4 14.48.32.90
s=SIP Call
c=IN IP4 14.48.32.33
b=TIAS:64000
b=CT:64
b=AS:64
t=0 0
m=audio 4000 RTP/AVP 0 101
a=ptime:20
a=rtpmap:0 PCMU/8000
a=sendonly
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

Calling phone sends CUCM the ORC ACK

03797634.001 |20:21:09.385 |AppInfo |StationInit: (0000109) OpenReceiveChannelAck Status=0,
IpAddr=IpAddr.type:0 ipAddr:0x0e30201c000000000000000000000000(14.48.32.28), Port=17996,
PartyID=33554450

CUCM sends startMediaTransmission to the called (recorded) phone telling the phone to send RTP to the calling phone (14.48.32.28)

03797642.001 |20:21:09.385 |AppInfo |StationD: (0000114) startMediaTransmission
conferenceID=38960750 passThruPartyID=33554451 remoteIpAddress=IpAddr.type:0
ipAddr:0x0e30201c000000000000000000000000(14.48.32.28) remotePortNumber=17996
milliSecondPacketSize=20 compressType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0
qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e302021(14.48.32.33)

Called (recorded) phone sends CUCM the ORC ACK

03797643.001 |20:21:09.454 |AppInfo |StationInit: (0000114) OpenReceiveChannelAck Status=0,
IpAddr=IpAddr.type:0 ipAddr:0x0e302021000000000000000000000000(14.48.32.33), Port=32588,
PartyID=33554451

CUCM sends startMediaTransmission to the calling phone telling the phone to send RTP to the
called phone (14.48.32.33)

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

SIP

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

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Calling phone places call

01314118.001 |11:18:44.472 |AppInfo |StationInit: (0000004) EnblocCall calledParty=9110011.

CUCM performs digit analysis against the dialed digits (dd="9110011")

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

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

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

CUCM sends outbound INVITE to called (recorded) phone

01314173.001 |11:18:44.754 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 14.48.32.17 on port 50841 index 17
[106316,NET]
INVITE sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK203b13880683
From: <sip:9110006@14.48.32.90>;tag=38244~713e2333-4032-45f1-b1f5-e33cf471acec-47601638
To: <sip:9110011@14.48.32.90>
Date: Tue, 14 Oct 2014 15:18:44 GMT
Call-ID: 6198e780-43d13ed4-203c-5a20300e@14.48.32.90
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Call-Info: ; security= Unknown; orientation= from; gci= 2-6001; isVoip; call-instance= 1
Send-Info: conference, x-cisco-conference
Alert-Info:
Remote-Party-ID: <sip:9110006@14.48.32.90;x-cisco-callback-number=9110006>;party=calling;screen=yes;privacy=off
Contact: <sip:9110006@14.48.32.90:5060;transport=tcp>
Max-Forwards: 70
Content-Length: 0

Called (recorded) phone returns 100 Trying

01314174.002 |11:18:44.758 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 14.48.32.17 on port 50841 index 17 with 802 bytes:
[106317,NET]
SIP/2.0 100 Trying
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK203b13880683
From: <sip:9110006@14.48.32.90>;tag=38244~713e2333-4032-45f1-b1f5-e33cf471acec-47601638
To: <sip:9110011@14.48.32.90>
Call-ID: 6198e780-43d13ed4-203c-5a20300e@14.48.32.90
Date: Tue, 14 Oct 2014 15:18:51 GMT
CSeq: 101 INVITE
Server: Cisco-CP8841/10.2.1
Contact: <sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp>
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE, SUBSCRIBE, INFO
Supported: replaces, join, sdp-anat, norefersub, resource-priority, extended-refer, X-cisco-callinfo, X-cisco-serviceuri, X-cisco-escapecodes, X-cisco-service-control, X-cisco-srtp-fallback, X-cisco-monrec, X-cisco-config, X-cisco-sis-7.0.0, X-cisco-xsi-8.5.1
Allow-Events: kpml, dialog
Content-Length: 0

Called (recorded) phone returns 180 Ringing

01314178.002 |11:18:45.357 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 14.48.32.17 on port 50841 index 17 with 950 bytes:
[106318,NET]
SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK203b13880683
From: <sip:9110006@14.48.32.90>;tag=38244~713e2333-4032-45f1-b1f5-e33cf471acec-47601638
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bba73e445ee-3cc7e650
Call-ID: 6198e780-43d13ed4-203c-5a20300e@14.48.32.90
Date: Tue, 14 Oct 2014 15:18:51 GMT
CSeq: 101 INVITE
Server: Cisco-CP8841/10.2.1
Contact: <sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp>
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE, SUBSCRIBE, INFO

Remote-Party-ID: "9110011" <sip:9110011@14.48.32.90>;party=called;id-type=subscriber;privacy=off;screen=yes
Supported: replaces,join,sdp-anat,norefersub,resource-priority,extended-refer,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-escapecodes,X-cisco-service-control,X-cisco-srtp-fallback,X-cisco-monrec,X-cisco-config,X-cisco-sis-7.0.0,X-cisco-xsi-8.5.1
Allow-Events: kpml,dialog
Content-Length: 0

Called (recorded) phone returns 200 OK

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

[106319,NET]

SIP/2.0 200 OK

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

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

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

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

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

CSeq: 101 INVITE

Server: Cisco-CP8841/10.2.1

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

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

Remote-Party-ID: "9110011" <sip:9110011@14.48.32.90>;party=called;id-

type=subscriber;privacy=off;screen=yes

Supported: replaces,join,sdp-anat,norefersub,resource-priority,extended-refer,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-escapecodes,X-cisco-service-control,X-cisco-srtp-fallback,X-cisco-monrec,X-cisco-config,X-cisco-sis-7.0.0,X-cisco-xsi-8.5.1

Allow-Events: kpml,dialog

Content-Length: 404

Content-Type: application/sdp

Content-Disposition: session;handling=optional

v=0

o=Cisco-SIPUA 15076 0 IN IP4 14.48.32.17

s=SIP Call

t=0 0

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

c=IN IP4 14.48.32.17

a=rtpmap:0 PCMU/8000

a=rtpmap:8 PCMA/8000

a=rtpmap:18 G729/8000

a=fmtp:18 annexb=yes

a=rtpmap:102 L16/16000

a=rtpmap:9 G722/8000

a=rtpmap:116 iLBC/8000

a=fmtp:116 mode=20

a=rtpmap:124 ISAC/16000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15

a=sendrecv

CUCM Tells the calling phone to open the logical channel

01314284.001 |11:18:48.599 |AppInfo |StationD: (0000004) SEP0018195AA209 , star_MediaExchangeAgenaOpenLogicalChannel packetSize=20, codec=4, ci=47601637

CUCM Tells the calling phone to open the receive channel

01314294.002 |11:18:48.599 |AppInfo |StationD: (0000004) OpenReceiveChannel conferenceID=47601637 passThruPartyID=33554433 millisecondPacketSize=20 compressionType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=101 qualifierIn=? sourceIpAddr=IpAddr.type:0 ipAddr:0x0e302011000000000000000000000000(14.48.32.17). myIP:

IpAddr.type:0 ipv4Addr:0x0e30201c(14.48.32.28)

CUCM sends startMediaTransmission to the calling phone telling the phone to send RTP to the called (recorded) phone (14.48.32.17)

01314295.001 |11:18:48.599 |AppInfo |StationD: (0000004) startMediaTransmission
conferenceID=47601637 passThruPartyID=33554433 remoteIpAddress=IpAddr.type:0
ipAddr:0x0e302011000000000000000000000000(14.48.32.17) remotePortNumber=28354
milliSecondPacketSize=20 compressType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=101
qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e30201c(14.48.32.28)

CUCM sends ACK to called (recorded) phone telling the called phone to send media to the calling phone (14.48.32.28)

01314344.001 |11:18:48.652 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 14.48.32.17 on port 50841 index 17

[106320,NET]

ACK sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK203c2831c118
From: <sip:9110006@14.48.32.90>;tag=38244~713e2333-4032-45f1-b1f5-e33cf471acec-47601638
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bba73e445ee-3cc7e650
Date: Tue, 14 Oct 2014 15:18:44 GMT
Call-ID: 6198e780-43d13ed4-203c-5a20300e@14.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 243

v=0

o=CiscoSystemsCCM-SIP 38244 1 IN IP4 14.48.32.90
s=SIP Call
c=IN IP4 14.48.32.28
b=TIAS:64000
b=CT:64
b=AS:64
t=0 0
m=audio 17260 RTP/AVP 0 101
a=ptime:20
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

CUCM allocates BiB on called (recorded) phone

01314383.000 |11:18:48.675 |SdlSig |MrmAllocateUcbResourceReq |waiting
|MediaResourceManager(2,100,138,1) |Cc(2,100,220,1)
|2,100,14,20.16735^14.48.32.28^SEP0018195AA209 |[R:N-H:0,N:3,L:1,V:0,Z:0,D:0] CI=47601639
SsType=33554461 SsKey=1 BridgeType=0 MRGLPkid= NumStream=1 Bib=c32d6714-48bd-43d7-b96f-
91363aff3aa0 BibTgCi=47601638 FeatId=159 PL=5 PLDmn=0 DeviceCapability=0 NumVideoCapable=0
requestDeviceType=0 requestDeviceLocale=64 forkingDevicePosition=2 playToneDir=3

CUCM sends INVITE #1 to called (recorded) phone with record-invoker=auto in Call-Info field and original Call-ID in Join field
Notice the SDP has a=inactive to tear down the media

01314446.001 |11:18:48.682 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 14.48.32.17 on port 50841 index 17

[106321,NET]

INVITE sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK203d55363a7c
From: "Call Manager" ;tag=38246~713e2333-4032-45f1-b1f5-e33cf471acec-47601641

To: <sip:9110011@14.48.32.90>
Date: Tue, 14 Oct 2014 15:18:48 GMT
Call-ID: 63fb4180-43d13ed8-203d-5a20300e@14.48.32.90
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Call-Info: ; isVoip; record-invoker=auto
Join: 6198e780-43d13ed4-203c-5a20300e@14.48.32.90;from-tag=b000b4d9e8cb0bba73e445ee-3cc7e650;to-tag=38244~713e2333-4032-45f1-b1f5-e33cf471acec-47601638
Contact: <sip:14.48.32.90:5060;transport=tcp>
Remote-Party-ID: "Call Manager" ;party=calling;screen=yes;privacy=off
Max-Forwards: 70
Content-Type: application/sdp
Content-Length: 188

v=0
o=CiscoSystemsCCM-SIP 38246 1 IN IP4 14.48.32.90
s=SIP Call
c=IN IP4 14.48.32.90
t=0 0
m=audio 4000 RTP/AVP 0
a=label:X-relay-nearend
a=rtpmap:0 PCMU/8000
a=inactive
a=mid:1

Called (recorded) phone returns 200 OK
Notice the SDP has a=inactive to tear down the media

01314449.002 |11:18:48.702 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 14.48.32.17 on port 50841 index 17 with 1235 bytes:

[106323,NET]

SIP/2.0 200 OK

Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hg4bK203d55363a7c

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

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

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

Date: Tue, 14 Oct 2014 15:18:55 GMT

CSeq: 101 INVITE

Server: Cisco-CP8841/10.2.1

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

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

Remote-Party-ID: "9110011" <sip:9110011@14.48.32.90>;party=called;id-type=subscriber;privacy=off;screen=yes

Supported: replaces,join,sdp-anat,norefersub,resource-priority,extended-refer,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-escapecodes,X-cisco-service-control,X-cisco-srtp-fallback,X-cisco-monrec,X-cisco-config,X-cisco-sis-7.0.0,X-cisco-xsi-8.5.1

Allow-Events: kpml,dialog

Content-Length: 202

Content-Type: application/sdp

Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 4077 0 IN IP4 14.48.32.17
s=SIP Call
t=0 0
m=audio 28512 RTP/AVP 0 101
c=IN IP4 14.48.32.17
a=rtpmap:0 PCMU/8000

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

CUCM responds to called (recorded) phone with ACK

01314452.001 |11:18:48.702 |AppInfo |SIPtcp - wait_SdlsPISignal: Outgoing SIP TCP message to 14.48.32.17 on port 50841 index 17
[106324,NET]
ACK sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK203e9999fc7
From: "Call Manager" ;tag=38246~713e2333-4032-45f1-b1f5-e33cf471acec-47601641
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bbb4457e725-6869188a
Date: Tue, 14 Oct 2014 15:18:48 GMT
Call-ID: 63fb4180-43d13ed8-203d-5a20300e@14.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Length: 0

BiB places first call to recording destination address (cn is calling party which is the BiB cn="b0028310001" and it is dialing the recordingdestination dd="8675309")

01314484.003 |11:18:48.753 |AppInfo |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept DAMR.sstype=[0], TPcount=[0], DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]
01314484.004 |11:18:48.753 |AppInfo |Digit Analysis: getDaRes - Remote Destination [8675309] isURI[0]
01314484.005 |11:18:48.765 |AppInfo |CMUtility routeCallThroughCTIRD: no matching RemDestDynamic record exists for remdest [8675309]
01314484.006 |11:18:48.765 |AppInfo |DbMobility: getMatchedRemDest starts: cnumber = 8675309
01314484.007 |11:18:48.765 |AppInfo |DbMobility: getMatchedRemDest: full match case
01314484.008 |11:18:48.765 |AppInfo |DbMobility SelectByDestination: no matching RemDestDynamic record exists for remdest [8675309]
01314484.009 |11:18:48.765 |AppInfo |DbMobility: can't find remdest 8675309 in map
01314484.010 |11:18:48.765 |AppInfo |Digit analysis: patternUsage=5
01314484.011 |11:18:48.765 |AppInfo |Digit analysis: match(pi="1", fqcn="", cn="b0028310001", plv="5", pss="E911_PT:Phones_PT:EMERGENCY_PT:INTERNAL_PT:INFORMACAST_PT", TodFilteredPss="E911_PT:Phones_PT:EMERGENCY_PT:INTERNAL_PT:INFORMACAST_PT", dd="8675309", dac="1")
01314484.012 |11:18:48.765 |AppInfo |Digit analysis: analysis results
01314484.013 |11:18:48.765 |AppInfo ||PretransformCallingPartyNumber=b0028310001
|CallingPartyNumber=b0028310001
|DialingPartition=
|DialingPattern=8675309
|FullyQualifiedCalledPartyNumber=8675309
|DialingPatternRegularExpression=(8675309)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=8675309
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=8675309
|CollectedDigits=8675309

CUCM sends INVITE #1 to configured recording server (14.48.32.170)

01314552.001 |11:18:48.795 |AppInfo |//SIP/SIPUdp/wait_SdlsPISignal: Outgoing SIP UDP message to 14.48.32.170:[5060]:
[106325,NET]
INVITE sip:8675309@14.48.32.170:5060 SIP/2.0

Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK203f3135e715
From: <sip:9110011@14.48.32.90;x-nearend;x-refci=47601638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sepb000b4d9e8cb;x-nearendaddr=9110011;x-farendrefci=47601637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=38248~713e2333-4032-45f1-b1f5-e33cf471acec-47601642
To: <sip:8675309@14.48.32.170>
Date: Tue, 14 Oct 2014 15:18:48 GMT
Call-ID: 63fb4180-43d13ed8-203e-5a20300e@14.48.32.90
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: ;method="NOTIFY;Event=telephone-event;Duration=500"
Call-Info: ;x-cisco-video-traffic-class=DESKTOP
Cisco-Guide: 1677410688-0000065536-0000000001-1512058894
Session-Expires: 1800
P-Asserted-Identity: <sip:9110011@14.48.32.90>
Remote-Party-ID: <sip:9110011@14.48.32.90>;party=calling;screen=yes;privacy=off
Contact: <sip:9110011@14.48.32.90:5060>;isFocus
Max-Forwards: 70
Content-Length: 0

CUCM sends INVITE #2 to called (recorded) phone with record-invoker=auto in Call-Info field and original Call-ID in Join field
Notice the SDP has a=inactive to tear down the media

01314575.001 |11:18:48.796 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 14.48.32.17 on port 50841 index 17
[106326,NET]
INVITE sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK20401b237b36
From: "Call Manager" ;tag=38249~713e2333-4032-45f1-b1f5-e33cf471acec-47601644
To: <sip:9110011@14.48.32.90>
Date: Tue, 14 Oct 2014 15:18:48 GMT
Call-ID: 63fb4180-43d13ed8-203f-5a20300e@14.48.32.90
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Call-Info: ; isVoip; record-invoker=auto
Join: 6198e780-43d13ed4-203c-5a20300e@14.48.32.90;from-tag=b000b4d9e8cb0bba73e445ee-3cc7e650;to-tag=38244~713e2333-4032-45f1-b1f5-e33cf471acec-47601638
Contact: <sip:14.48.32.90:5060;transport=tcp>
Remote-Party-ID: "Call Manager" ;party=calling;screen=yes;privacy=off
Max-Forwards: 70
Content-Type: application/sdp
Content-Length: 187

v=0
o=CiscoSystemsCCM-SIP 38249 1 IN IP4 14.48.32.90
s=SIP Call
c=IN IP4 14.48.32.90
t=0 0
m=audio 4000 RTP/AVP 0
a=label:X-relay-farend

a=rtpmap:0 PCMU/8000
a=inactive
a=mid:1

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

01314583.001 |11:18:48.862 |AppInfo |//SIP/SIPUdp/wait_SdlDataInd: Incoming SIP UDP message size 737 from 14.48.32.170:[5060]:

[106328,NET]

SIP/2.0 200 OK

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

From: <sip:9110011@14.48.32.90;x-nearend;x-refci=47601638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sepb000b4d9e8cb;x-nearendaddr=9110011;x-farendrefci=47601637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=38248~713e2333-4032-45f1-b1f5-e33cf471acec-47601642

To: <sip:8675309@14.48.32.170>;tag=1

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

CSeq: 101 INVITE

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

Content-Type: application/sdp

Content-Length: 135

v=0

o=user1 53655765 2353687637 IN IP4 14.48.32.170

s=-

c=IN IP4 14.48.32.170

t=0 0

m=audio 6000 RTP/AVP 0

a=rtpmap:0 PCMU/8000

CUCM sends re-INVITE to called (recorded) phone for call #1 to invoke the BiB (notice there is no SDP)

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

[106329,NET]

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

Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK204176d717cd

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

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

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

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

Supported: timer,resource-priority,replaces

User-Agent: Cisco-CUCM10.5

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

CSeq: 102 INVITE

Max-Forwards: 70

Expires: 180

Allow-Events: presence

Call-Info: ; isVoip; record-invoker=auto

Min-SE: 1800

Remote-Party-ID: "Call Manager" ;party=calling;screen=yes;privacy=off

Contact: <sip:14.48.32.90:5060;transport=tcp>

Content-Length: 0

Called (recorded) phone returns 200 OK in response to INVITE #2 to invoke BiB
Notice the SDP has a=inactive to tear down the media

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

[106330,NET]

SIP/2.0 200 OK

Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK20401b237b36

From: "Call Manager" ;tag=38249~713e2333-4032-45f1-b1f5-e33cf471acec-47601644
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bbc4d5b7fc6-3ab2172f
Call-ID: 63fb4180-43d13ed8-203f-5a20300e@14.48.32.90
Date: Tue, 14 Oct 2014 15:18:55 GMT
CSeq: 101 INVITE
Server: Cisco-CP8841/10.2.1
Contact: <sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "9110011" <sip:9110011@14.48.32.90>;party=called;id-
type=subscriber;privacy=off;screen=yes
Supported: replaces,join,sdp-anat,norefersub,resource-priority,extended-refer,X-cisco-
callinfo,X-cisco-serviceuri,X-cisco-escapecodes,X-cisco-service-control,X-cisco-srtp-fallback,X-
cisco-monrec,X-cisco-config,X-cisco-sis-7.0.0,X-cisco-xsi-8.5.1
Allow-Events: kpml,dialog
Content-Length: 203
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 11326 0 IN IP4 14.48.32.17
s=SIP Call
t=0 0
m=audio 19696 RTP/AVP 0 101
c=IN IP4 14.48.32.17
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=inactive

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

01314648.001 |11:18:48.866 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to
14.48.32.17 on port 50841 index 17
[106331,NET]
ACK sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK20424175effe
From: "Call Manager" ;tag=38249~713e2333-4032-45f1-b1f5-e33cf471acec-47601644
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bbc4d5b7fc6-3ab2172f
Date: Tue, 14 Oct 2014 15:18:48 GMT
Call-ID: 63fb4180-43d13ed8-203f-5a20300e@14.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Length: 0

BiB places second call to recording destination address (cn is calling party which is the
BiB cn="b0028310001" and it is dialing the recordingdestination dd="8675309")
Note that the BiB number stayed the same (b0028310001) and so did the recordingdestination
number

01314680.003 |11:18:48.867 |AppInfo |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept
DAMR.sstype=[0], TPcount=[0], DAMR.NotifyCount=[0], DaRes.NotifyCount=[0]
01314680.004 |11:18:48.867 |AppInfo |Digit Analysis: getDaRes - Remote Destination [8675309]
isURI[0]
01314680.005 |11:18:48.867 |AppInfo |CMUtility routeCallThroughCTIRD: no matching
RemDestDynamic record exists for remdest [8675309]
01314680.006 |11:18:48.867 |AppInfo |DbMobility: getMatchedRemDest starts: cnumber = 8675309
01314680.007 |11:18:48.867 |AppInfo |DbMobility: getMatchedRemDest: full match case
01314680.008 |11:18:48.867 |AppInfo |DbMobility SelectByDestination: no matching RemDestDynamic
record exists for remdest [8675309]
01314680.009 |11:18:48.867 |AppInfo |DbMobility: can't find remdest 8675309 in map
01314680.010 |11:18:48.867 |AppInfo |Digit analysis: patternUsage=5

01314680.011 |11:18:48.867 |AppInfo |Digit analysis: match(pi="1", fqcn="",
cn="b0028310001", plv="5", pss="E911_PT:Phones_PT:EMERGENCY_PT:INTERNAL_PT:INFORMACAST_PT",
TodFilteredPss="E911_PT:Phones_PT:EMERGENCY_PT:INTERNAL_PT:INFORMACAST_PT",
dd="8675309", dac="1")

01314680.012 |11:18:48.867 |AppInfo |Digit analysis: analysis results
01314680.013 |11:18:48.867 |AppInfo ||PretransformCallingPartyNumber=b0028310001
|CallingPartyNumber=b0028310001
|DialingPartition=
|DialingPattern=8675309
|FullyQualifiedCalledPartyNumber=8675309
|DialingPatternRegularExpression=(8675309)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=8675309
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=8675309
|CollectedDigits=8675309

CUCM sends INVITE #2 to configured recording server

01314731.001 |11:18:48.870 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message
to 14.48.32.170:[5060]:
[106333,NET]
INVITE sip:8675309@14.48.32.170:5060 SIP/2.0
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK20432a53d34c
From: <sip:9110011@14.48.32.90;x-farend;x-refci=47601638;x-nearendclusterid=glenscucm10-5;x-
nearenddevice=sepb000b4d9e8cb;x-nearendaddr=9110011;x-farendrefci=47601637;x-
farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-
farendaddr=9110006>;tag=38251~713e2333-4032-45f1-b1f5-e33cf471acec-47601645
To: <sip:8675309@14.48.32.170>
Date: Tue, 14 Oct 2014 15:18:48 GMT
Call-ID: 63fb4180-43d13ed8-2040-5a20300e@14.48.32.90
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: ;method="NOTIFY;Event=telephone-event;Duration=500"
Call-Info: ;x-cisco-video-traffic-class=DESKTOP
Cisco-Guid: 1677410688-0000065536-0000000002-1512058894
Session-Expires: 1800
P-Asserted-Identity: <sip:9110011@14.48.32.90>
Remote-Party-ID: <sip:9110011@14.48.32.90>;party=calling;screen=yes;privacy=off
Contact: <sip:9110011@14.48.32.90:5060>;isFocus
Max-Forwards: 70
Content-Length: 0

CUCM receives 200 OK in response to INVITE #2 from configured recording server

01314751.001 |11:18:48.871 |AppInfo |//SIP/SIPUdp/wait_SdlDataInd: Incoming SIP UDP message
size 736 from 14.48.32.170:[5060]:
[106335,NET]
SIP/2.0 200 OK
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK20432a53d34c
From: <sip:9110011@14.48.32.90;x-farend;x-refci=47601638;x-nearendclusterid=glenscucm10-5;x-
nearenddevice=sepb000b4d9e8cb;x-nearendaddr=9110011;x-farendrefci=47601637;x-
farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-

faresaddr=9110006>;tag=38251~713e2333-4032-45f1-b1f5-e33cf471acec-47601645
To: <sip:8675309@14.48.32.170>;tag=2
Call-ID: 63fb4180-43d13ed8-2040-5a20300e@14.48.32.90
CSeq: 101 INVITE
Contact: <sip:14.48.32.170:5060;transport=udp>
Content-Type: application/sdp
Content-Length: 135

v=0
o=user1 53655765 2353687637 IN IP4 14.48.32.170
s=-
c=IN IP4 14.48.32.170
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000

CUCM sends re-INVITE #2 to called (recorded) phone for second BiB invocation call
Notice there is no SDP

01314828.001 |11:18:48.875 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to
14.48.32.17 on port 50841 index 17
[106336,NET]

INVITE sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK20443475e621
From: "Call Manager" ;tag=38249~713e2333-4032-45f1-b1f5-e33cf471acec-47601644
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bbc4d5b7fc6-3ab2172f
Date: Tue, 14 Oct 2014 15:18:48 GMT
Call-ID: 63fb4180-43d13ed8-203f-5a20300e@14.48.32.90
Supported: timer,resource-priority,replaces
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, SUBSCRIBE, NOTIFY
CSeq: 102 INVITE
Max-Forwards: 70
Expires: 180
Allow-Events: presence
Call-Info: ; isVoip; record-invoker=auto
Min-SE: 1800
Remote-Party-ID: "Call Manager" ;party=calling;screen=yes;privacy=off
Contact: <sip:14.48.32.90:5060;transport=tcp>
Content-Length: 0

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

01314829.002 |11:18:48.876 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from
14.48.32.17 on port 50841 index 17 with 1235 bytes:

[106337,NET]
SIP/2.0 200 OK
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK204176d717cd
From: "Call Manager" ;tag=38246~713e2333-4032-45f1-b1f5-e33cf471acec-47601641
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bbb4457e725-6869188a
Call-ID: 63fb4180-43d13ed8-203d-5a20300e@14.48.32.90
Date: Tue, 14 Oct 2014 15:18:55 GMT
CSeq: 102 INVITE
Server: Cisco-CP8841/10.2.1
Contact: <sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "9110011" <sip:9110011@14.48.32.90>;party=called;id-
type=subscriber;privacy=off;screen=yes
Supported: replaces,join,sdp-anat,norefersub,resource-priority,extended-refer,X-cisco-
callinfo,X-cisco-serviceuri,X-cisco-escapecodes,X-cisco-service-control,X-cisco-srtp-fallback,X-
cisco-monrec,X-cisco-config,X-cisco-sis-7.0.0,X-cisco-xsi-8.5.1
Allow-Events: kpml,dialog
Content-Length: 202

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

v=0
o=Cisco-SIPUA 4077 1 IN IP4 14.48.32.17
s=SIP Call
t=0 0
m=audio 28512 RTP/AVP 0 101
c=IN IP4 14.48.32.17
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv

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

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

v=0
o=CiscoSystemsCCM-SIP 38246 3 IN IP4 14.48.32.90
s=SIP Call
c=IN IP4 14.48.32.170
b=TIAS:64000
b=AS:64
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=recvonly

CUCM sends ACK to configured recording server for INVITE #1

01314875.001 |11:18:48.880 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 14.48.32.170:[5060]:
[106339,NET]
ACK sip:14.48.32.170:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK20467ee6be7
From: <sip:9110011@14.48.32.90;x-nearend;x-refci=47601638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sepb000b4d9e8cb;x-nearendaddr=9110011;x-farendrefci=47601637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=38248~713e2333-4032-45f1-b1f5-e33cf471acec-47601642
To: <sip:8675309@14.48.32.170>;tag=1
Date: Tue, 14 Oct 2014 15:18:48 GMT
Call-ID: 63fb4180-43d13ed8-203e-5a20300e@14.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Type: application/sdp
Content-Length: 234

v=0
o=CiscoSystemsCCM-SIP 38248 1 IN IP4 14.48.32.90
s=SIP Call
c=IN IP4 14.48.32.17
b=TIAS:64000
b=AS:64
t=0 0
m=audio 28512 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=sendonly
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

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

01314878.005 |11:18:48.881 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 14.48.32.17 on port 50841 index 17 with 1236 bytes:
[106341,NET]
SIP/2.0 200 OK
Via: SIP/2.0/TCP 14.48.32.90:5060;branch=z9hG4bK20443475e621
From: "Call Manager" ;tag=38249~713e2333-4032-45f1-b1f5-e33cf471acec-47601644
To: <sip:9110011@14.48.32.90>;tag=b000b4d9e8cb0bbc4d5b7fc6-3ab2172f
Call-ID: 63fb4180-43d13ed8-203f-5a20300e@14.48.32.90
Date: Tue, 14 Oct 2014 15:18:55 GMT
CSeq: 102 INVITE
Server: Cisco-CP8841/10.2.1
Contact: <sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@14.48.32.17:50841;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "9110011" <sip:9110011@14.48.32.90>;party=called;id-type=subscriber;privacy=off;screen=yes
Supported: replaces,join,sdp-anat,norefersub,resource-priority,extended-refer,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-escapecodes,X-cisco-service-control,X-cisco-srtp-fallback,X-cisco-monrec,X-cisco-config,X-cisco-sis-7.0.0,X-cisco-xsi-8.5.1
Allow-Events: kpml,dialog
Content-Length: 203
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 11326 1 IN IP4 14.48.32.17
s=SIP Call
t=0 0
m=audio 19696 RTP/AVP 0 101
c=IN IP4 14.48.32.17
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv

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

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

CSeq: 102 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 178

v=0
o=CiscoSystemsCCM-SIP 38249 3 IN IP4 14.48.32.90
s=SIP Call
c=IN IP4 14.48.32.170
b=TIAS:64000
b=AS:64
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=recvonly

CUCM sends ACK to configured recording server for INVITE #2

01314909.001 |11:18:48.883 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 14.48.32.170:[5060]:

[106343,NET]

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

Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK204854e1b53f

From: <sip:9110011@14.48.32.90;x-farend;x-refci=47601638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sepb000b4d9e8cb;x-nearendaddr=9110011;x-farendrefci=47601637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=38251~713e2333-4032-45f1-b1f5-e33cf471acec-47601645

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

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

Call-ID: 63fb4180-43d13ed8-2040-5a20300e@14.48.32.90

User-Agent: Cisco-CUCM10.5

Max-Forwards: 70

CSeq: 101 ACK

Allow-Events: presence, kpml

Content-Type: application/sdp

Content-Length: 234

v=0
o=CiscoSystemsCCM-SIP 38251 1 IN IP4 14.48.32.90
s=SIP Call
c=IN IP4 14.48.32.17
b=TIAS:64000
b=AS:64
t=0 0
m=audio 19696 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=sendonly
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

Dépannez

Cette section fournit des informations que vous pouvez utiliser pour dépanner votre configuration.

[Négociation de Codec](#)

C'est un exemple d'un des types les plus communs de pannes d'enregistrement d'appels - non-concordance de codecs entre le téléphone enregistré et le serveur d'enregistrement :

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

Calling phone places call

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

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

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

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

00019640.001 |12:48:34.522 |AppInfo |Digit analysis: wait_DmPidRes- Partition=[]
Pattern=[9110001] Where=[],cmDeviceType=[UserDevice], OutsideDialtone =[0], DeviceOverride=[0],
PID=LineControl(2,100,174,7),CI=[49613637],Sender=Cdcc(2,100,219,1)

CUCM extends the call to the called phone

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

00019657.010 |12:48:34.560 |AppInfo |StationD: (0000007) DEBUG- saveRinger: ci=49613638, line=1, mode=2, cm_precedence=5, callPhase=5, modifier=0
00019657.011 |12:48:34.560 |AppInfo |StationD: (0000007) INFO sendCallAcceptReq: Try to send StationLineCallAccept to cdpc=2 .
00019657.012 |12:48:34.560 |AppInfo |StationD: (0000007) playRinger for: ci=49613638.
00019657.013 |12:48:34.560 |AppInfo |StationD: (0000007) DEBUG- getLineRingSetting: retVal=4.
00019657.014 |12:48:34.560 |AppInfo |StationD: (0000007) DEBUG- getLineRingSetting: retVal=4.
00019657.015 |12:48:34.560 |AppInfo |StationD: (0000007) DEBUG- getLineRingSetting: retVal=4.

The Called (recorded) phone goes off hook

00019709.001 |12:48:36.042 |AppInfo |StationD: (0000007) restart0_StationOffHook - INFO: CI=49613638 on line=1, SPKMode=0, alwaysPrimeLine=0, alwaysUsePrimeLineForVM=0, fid=9999, offHookTrigger=1.

CUCM Tells the calling phone to open the logical channel

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

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

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

CUCM Tells the calling phone to open the receive channel

00019784.002 |12:48:36.062 |AppInfo |StationD: (0000005) OpenReceiveChannel conferenceID=49613637 passThruPartyID=33554433 millisecondPacketSize=20 compressionType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0 qualifierIn=? sourceIpAddr=IpAddr.type:0 ipAddr:0x0e302021000000000000000000000000(14.48.32.33). myIP: IpAddr.type:0 ipv4Addr:0x0e30201c(14.48.32.28)

Codec locked due to recording on called (recorded) phone

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

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

00019788.002 |12:48:36.062 |AppInfo |StationD: (0000007) OpenReceiveChannel conferenceID=49613638 passThruPartyID=33554434 millisecondPacketSize=20 compressionType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0 qualifierIn=? sourceIpAddr=IpAddr.type:0 ipAddr:0x0e30201c000000000000000000000000(14.48.32.28). myIP: IpAddr.type:0 ipv4Addr:0x0e302021(14.48.32.33)

CUCM allocates the BiB on the called (recorded) phone

00019830.000 |12:48:36.074 |SdlSig |MrmAllocateUcbResourceReq |waiting
|MediaResourceManager(2,100,138,1) |Cc(2,100,220,1)
|2,100,14,19.206^14.48.32.33^SEP001795BDD16B |[R:N-H:0,N:1,L:0,V:0,Z:0,D:0] CI=49613639
SsType=33554461 SsKey=1 BridgeType=0 MRGLPkid= NumStream=1 Bib=89cdb152-4ef2-4d60-9e6b-ab8c77c22618 BibTgCi=49613638 FeatId=159 PL=5 PLDmn=0 DeviceCapability=0 NumVideoCapable=0 requestDeviceType=0 requestDeviceLocale=64 forkingDevicePosition=2 playToneDir=3

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

00019889.001 |12:48:36.100 |AppInfo |Digit Analysis: star_DaReq: daReq.partitionSearchSpace(),

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

Calling phone sends CUCM the ORC ACK

```
00019912.001 |12:48:36.139 |AppInfo |StationInit: (0000005) OpenReceiveChannelAck Status=0,
IpAddr=IpAddr.type:0 ipAddr:0x0e30201c000000000000000000000000(14.48.32.28), Port=31678,
PartyID=33554433
```

CUCM sends startMediaTransmission to the called (recorded) phone telling the phone to send RTP to the calling phone (14.48.32.28)

```
00019920.001 |12:48:36.139 |AppInfo |StationD: (0000007) startMediaTransmission
conferenceID=49613638 passThruPartyID=33554434 remoteIpAddress=IpAddr.type:0
ipAddr:0x0e30201c000000000000000000000000(14.48.32.28) remotePortNumber=31678
milliSecondPacketSize=20 compressType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0
qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e302021(14.48.32.33)
```

Called (recorded) phone sends CUCM the ORC ACK

```
00019959.001 |12:48:36.145 |AppInfo |StationInit: (0000007) OpenReceiveChannelAck Status=0,
IpAddr=IpAddr.type:0 ipAddr:0x0e302021000000000000000000000000(14.48.32.33), Port=28360,
PartyID=33554434
```

CUCM sends startMediaTransmission to the calling phone telling the phone to send RTP to the called phone (14.48.32.33)

```
00019977.001 |12:48:36.146 |AppInfo |StationD: (0000005) startMediaTransmission
conferenceID=49613637 passThruPartyID=33554433 remoteIpAddress=IpAddr.type:0
ipAddr:0x0e302021000000000000000000000000(14.48.32.33) remotePortNumber=28360
milliSecondPacketSize=20 compressType=4(Media_Payload_G711Ulaw64k) RFC2833PayloadType=0
```

qualifierOut=?. myIP: IpAddr.type:0 ipv4Addr:0x0e30201c(14.48.32.28)

```
### BiB places second call to recording destination address (cn is calling number which is the
BiB cn="b00223906001" and it is dialing the recordingdestination dd="8675309")
Note that the BiB number stayed the same (b00223906001) and so did the recordingdestination
number 00020002.001 |12:48:36.147 |AppInfo |Digit Analysis: star_DaReq:
daReq.partitionSearchSpace(), filteredPartitionSearchSpaceString(), partitionSearchSpaceString()
00020002.002 |12:48:36.147 |AppInfo |Digit Analysis: star_DaReq: Matching Legacy Numeric,
digits=8675309 00020002.003 |12:48:36.147 |AppInfo |Digit Analysis: getDaRes data:
daRes.ssType=[0] Intercept DAMR.sstype=[0], TPcount=[0], DAMR.NotifyCount=[0],
DaRes.NotifyCount=[0] 00020002.004 |12:48:36.147 |AppInfo |Digit Analysis: getDaRes - Remote
Destination [8675309] isURI[0] 00020002.005 |12:48:36.147 |AppInfo |CMUtility
routeCallThroughCTIRD: no matching RemDestDynamic record exists for remdest [8675309]
00020002.006 |12:48:36.147 |AppInfo |DbMobility: getMatchedRemDest starts: cnumber = 8675309
00020002.007 |12:48:36.147 |AppInfo |DbMobility: getMatchedRemDest: full match case 00020002.008
|12:48:36.147 |AppInfo |DbMobility SelectByDestination: no matching RemDestDynamic record exists
for remdest [8675309] 00020002.009 |12:48:36.147 |AppInfo |DbMobility: can't find remdest
8675309 in map 00020002.010 |12:48:36.147 |AppInfo |Digit analysis: patternUsage=5 00020002.011
|12:48:36.147 |AppInfo |Digit analysis: match(pi="1", 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 (14.48.32.170)

```
00020086.001 |12:48:36.156 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message
to 14.48.32.170:[5060]:
[901,NET]
INVITE sip:8675309@14.48.32.170:5060 SIP/2.0
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK4f2a857d3d
From: <sip:9110001@14.48.32.90;x-nearend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-
nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-
farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-
farendaddr=9110006>;tag=351-713e2333-4032-45f1-b1f5-e33cf471acec-49613642
To: <sip:8675309@14.48.32.170>
Date: Tue, 14 Oct 2014 16:48:36 GMT
Call-ID: ef7acf80-43d153e4-50-5a20300e@14.48.32.90
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: ;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 4017803136-0000065536-0000000001-1512058894
Session-Expires: 1800
P-Asserted-Identity: <sip:9110001@14.48.32.90>
Remote-Party-ID: <sip:9110001@14.48.32.90>;party-calling;screen=yes;privacy=off
Contact: <sip:9110001@14.48.32.90:5060>;isFocus
Max-Forwards: 70
Content-Length: 0
```

CUCM sends INVITE #2 to configured recording server (14.48.32.170)

00020088.001 |12:48:36.157 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 14.48.32.170:[5060]:
[902,NET]
INVITE sip:8675309@14.48.32.170:5060 SIP/2.0
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK5014378d0b
From: <sip:9110001@14.48.32.90;x-farend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=352-713e2333-4032-45f1-b1f5-e33cf47lacec-49613645
To: <sip:8675309@14.48.32.170>
Date: Tue, 14 Oct 2014 16:48:36 GMT
Call-ID: ef7acf80-43d153e4-51-5a20300e@14.48.32.90
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: ;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 4017803136-0000065536-0000000002-1512058894
Session-Expires: 1800
P-Asserted-Identity: <sip:9110001@14.48.32.90>
Remote-Party-ID: <sip:9110001@14.48.32.90>;party=calling;screen=yes;privacy=off
Contact: <sip:9110001@14.48.32.90:5060>;isFocus
Max-Forwards: 70
Content-Length: 0

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

00020089.001 |12:48:36.161 |AppInfo |//SIP/SIPUdp/wait_SdlDataInd: Incoming SIP UDP message size 731 from 14.48.32.170:[5060]:
[903,NET]
SIP/2.0 200 OK
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK4f2a857d3d
From: <sip:9110001@14.48.32.90;x-nearend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=351-713e2333-4032-45f1-b1f5-e33cf47lacec-49613642
To: <sip:8675309@14.48.32.170>;tag=1
Call-ID: ef7acf80-43d153e4-50-5a20300e@14.48.32.90
CSeq: 101 INVITE
Contact: <sip:14.48.32.170:5060;transport=udp>
Content-Type: application/sdp
Content-Length: 135

v=0
o=user1 53655765 2353687637 IN IP4 14.48.32.170
s=-
c=IN IP4 14.48.32.170
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000

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

00020092.001 |12:48:36.161 |AppInfo |//SIP/SIPUdp/wait_SdlDataInd: Incoming SIP UDP message size 730 from 14.48.32.170:[5060]:
[905,NET]
SIP/2.0 200 OK

Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK5014378d0b
From: <sip:9110001@14.48.32.90;x-farend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=352-713e2333-4032-45f1-b1f5-e33cf47lacec-49613645
To: <sip:8675309@14.48.32.170>;tag=2
Call-ID: ef7acf80-43d153e4-51-5a20300e@14.48.32.90
CSeq: 101 INVITE
Contact: <sip:14.48.32.170:5060;transport=udp>
Content-Type: application/sdp
Content-Length: 135

v=0
o=user1 53655765 2353687637 IN IP4 14.48.32.170
s=-
c=IN IP4 14.48.32.170
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000

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

CUCM determines 2 transcoders are required and attempts to allocate

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

CUCM determines 2 transcoders are required and attempts to allocate

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

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

00020210.001 |12:48:36.216 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 14.48.32.170:[5060]:
[906,NET]

ACK sip:14.48.32.170:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK51257b2b47
From: <sip:9110001@14.48.32.90;x-nearend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=351-713e2333-4032-45f1-b1f5-e33cf47lacec-49613642
To: <sip:8675309@14.48.32.170>;tag=1
Date: Tue, 14 Oct 2014 16:48:36 GMT
Call-ID: ef7acf80-43d153e4-50-5a20300e@14.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml

Content-Length: 0

00020211.001 |12:48:36.216 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 14.48.32.170:[5060]:

[907,NET]

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

Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK526f3d2afa

From: <sip:9110001@14.48.32.90;x-nearend;x-refci=49613638;x-nearendclusterid=GlensCUCM10-5;x-nearenddevice=SEP001795BDD16B;x-nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=GlensCUCM10-5;x-farenddevice=SEP0018195AA209;x-farendaddr=9110006>;tag=351~713e2333-4032-45f1-b1f5-e33cf471acec-49613642

To: <sip:8675309@14.48.32.170>;tag=1

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

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

User-Agent: Cisco-CUCM10.5

Max-Forwards: 70

P-Asserted-Identity: <sip:9110001@14.48.32.90>

CSeq: 102 BYE

Reason: Q.850;cause=47

Content-Length: 0

CUCM sendt the ACK and BYE to the recording server in response to INVITE #2

Note the Q.850 cuase code in the BYE

00020248.001 |12:48:36.218 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 14.48.32.170:[5060]:

[908,NET]

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

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

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

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

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

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

User-Agent: Cisco-CUCM10.5

Max-Forwards: 70

CSeq: 101 ACK

Allow-Events: presence, kpml

Content-Length: 0

00020249.001 |12:48:36.218 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 14.48.32.170:[5060]: [909,NET] BYE sip:14.48.32.170:5060;transport=UDP SIP/2.0 Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK5462aba807 From: <sip:9110001@14.48.32.90;x-farend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=352~713e2333-4032-45f1-b1f5-e33cf471acec-49613645 To: <sip:8675309@14.48.32.170>;tag=2 Date: Tue, 14 Oct 2014 16:48:36 GMT Call-ID: ef7acf80-43d153e4-51-5a20300e@14.48.32.90 User-Agent: Cisco-CUCM10.5 Max-Forwards: 70 P-Asserted-Identity: <sip:9110001@14.48.32.90> CSeq: 102 BYE **Reason: Q.850;cause=47**

Content-Length: 0

Content-Length: 0

Mauvaise configuration qui inclut des questions CSS et pinte

Les commandes ici permettent la majorité des configurations d'enregistrement à passer en revue rapidement avec connaître seulement l'adresse MAC d'un téléphone qui n'enregistre pas des appels. Remplacez simplement la partie de la commande **MAC_of_Phone** par l'adresse MAC réelle du téléphone comme dans les exemples vus ici.

Ceci te donne le DN (tous s'il y a plus d'un) pour le MAC que vous recherchez en fonction, le MAC

du téléphone juste pour la confirmation, la configuration de BAVOIR, la configuration d'intimité, le type d'enregistrement (mettez en référence les valeurs répertoriées dans les exemples du laboratoire), le profil d'enregistrement en service par le téléphone, le nom des espaces de recherche d'appel d'enregistrement (CSS), la destination d'enregistrement pour ce profil d'enregistrement, et la partition que la destination de enregistrement est associée avec basé sur le MAC vous recherchez en fonction :

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

```
00020248.001 |12:48:36.218 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message
to 14.48.32.170:[5060]:
[908,NET]
ACK sip:14.48.32.170:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK531df920a6
From: <sip:9110001@14.48.32.90;x-farend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-
nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-
farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-
farendaddr=9110006>;tag=352-713e2333-4032-45f1-b1f5-e33cf471acec-49613645
To: <sip:8675309@14.48.32.170>;tag=2
Date: Tue, 14 Oct 2014 16:48:36 GMT
Call-ID: ef7acf80-43d153e4-51-5a20300e@14.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Length: 0
```

```
00020249.001 |12:48:36.218 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to
14.48.32.170:[5060]: [909,NET] BYE sip:14.48.32.170:5060;transport=UDP SIP/2.0 Via: SIP/2.0/UDP
14.48.32.90:5060;branch=z9hG4bK5462aba807 From: <sip:9110001@14.48.32.90;x-farend;x-
refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-
nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-
farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=352-713e2333-4032-45f1-b1f5-e33cf471acec-
49613645 To: <sip:8675309@14.48.32.170>;tag=2 Date: Tue, 14 Oct 2014 16:48:36 GMT Call-ID:
ef7acf80-43d153e4-51-5a20300e@14.48.32.90 User-Agent: Cisco-CUCM10.5 Max-Forwards: 70 P-
Asserted-Identity: <sip:9110001@14.48.32.90> CSeq: 102 BYE Reason: Q.850;cause=47
Content-Length: 0
```

Ceci te donne la liste de partitions qui sont associées avec le CSS de enregistrement sur le profil d'enregistrement qui est associé avec le MAC du téléphone que vous recherchez contre.

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

```
00020248.001 |12:48:36.218 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message
to 14.48.32.170:[5060]:
[908,NET]
ACK sip:14.48.32.170:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK531df920a6
From: <sip:9110001@14.48.32.90;x-farend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-
nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-
farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-
farendaddr=9110006>;tag=352-713e2333-4032-45f1-b1f5-e33cf471acec-49613645
To: <sip:8675309@14.48.32.170>;tag=2
Date: Tue, 14 Oct 2014 16:48:36 GMT
Call-ID: ef7acf80-43d153e4-51-5a20300e@14.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
```

Allow-Events: presence, kpml
Content-Length: 0

```
00020249.001 |12:48:36.218 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to
14.48.32.170:[5060]: [909,NET] BYE sip:14.48.32.170:5060;transport=UDP SIP/2.0 Via: SIP/2.0/UDP
14.48.32.90:5060;branch=z9hG4bK5462aba807 From: <sip:9110001@14.48.32.90;x-farend;x-
refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-
nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-
farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=352~713e2333-4032-45f1-b1f5-e33cf471acec-
49613645 To: <sip:8675309@14.48.32.170>;tag=2 Date: Tue, 14 Oct 2014 16:48:36 GMT Call-ID:
ef7acf80-43d153e4-51-5a20300e@14.48.32.90 User-Agent: Cisco-CUCM10.5 Max-Forwards: 70 P-
Asserted-Identity: <sip:9110001@14.48.32.90> CSeq: 102 BYE Reason: Q.850;cause=47
Content-Length: 0
```

Voici les exemples de la sortie du laboratoire pour un téléphone avec l'adresse MAC SEPC80084AA8743 :

Dans cette commande, vous pouvez voir le téléphone a seulement un DN là-dessus ce qui est **2003**, nous voient également que le BAVOIR est allumé, l'intimité est éteinte, le type d'enregistrement est automatique, la source préférée est téléphone, le profil d'enregistrement est **profil d'enregistrement de test**, l'enregistrement appelle l'espace de recherche est **INTERNAL_CSS**, le modèle d'artère pour des appels enregistrés est **8675309** et ce modèle est associé avec la partition **INTERNAL_PT**.

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

```
00020248.001 |12:48:36.218 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message
to 14.48.32.170:[5060]:
[908,NET]
ACK sip:14.48.32.170:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 14.48.32.90:5060;branch=z9hG4bK531df920a6
From: <sip:9110001@14.48.32.90;x-farend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-
nearenddevice=sep001795bdd16b;x-nearendaddr=9110001;x-farendrefci=49613637;x-
farendclusterid=glenscucm10-5;x-farenddevice=sep0018195aa209;x-
farendaddr=9110006>;tag=352~713e2333-4032-45f1-b1f5-e33cf471acec-49613645
To: <sip:8675309@14.48.32.170>;tag=2
Date: Tue, 14 Oct 2014 16:48:36 GMT
Call-ID: ef7acf80-43d153e4-51-5a20300e@14.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Length: 0
```

```
00020249.001 |12:48:36.218 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to
14.48.32.170:[5060]: [909,NET] BYE sip:14.48.32.170:5060;transport=UDP SIP/2.0 Via: SIP/2.0/UDP
14.48.32.90:5060;branch=z9hG4bK5462aba807 From: <sip:9110001@14.48.32.90;x-farend;x-
refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevice=sep001795bdd16b;x-
nearendaddr=9110001;x-farendrefci=49613637;x-farendclusterid=glenscucm10-5;x-
farenddevice=sep0018195aa209;x-farendaddr=9110006>;tag=352~713e2333-4032-45f1-b1f5-e33cf471acec-
49613645 To: <sip:8675309@14.48.32.170>;tag=2 Date: Tue, 14 Oct 2014 16:48:36 GMT Call-ID:
ef7acf80-43d153e4-51-5a20300e@14.48.32.90 User-Agent: Cisco-CUCM10.5 Max-Forwards: 70 P-
Asserted-Identity: <sip:9110001@14.48.32.90> CSeq: 102 BYE Reason: Q.850;cause=47
Content-Length: 0
```

Avec la sortie de cette commande, vous pouvez vérifier toutes les partitions du CSS de enregistrement et du profil d'enregistrement associé avec le téléphone d'intérêt. Vous pouvez voir ici que la partition **INTERNAL_PT** est l'une des partitions associées avec l'espace de recherche appelant **INTERNAL_CSS**. Ceci signifie qu'il ne doit y avoir aucune question avec le BAVOIR du téléphone pouvant appeler le modèle d'artère d'enregistrement.

```

run sql select css.name as name_of_the_recording_css, rp.name as partitions_in_recording_css,
csm.sortorder from callingsearchspace as css inner join callingsearchspacemember as csm on
csm.fkcallingsearchspace = css.pkid inner join routepartition as rp on csm.fkroutepartition =
rp.pkid inner join recordingprofile as rcrdpro on rcrdpro.fkcallingsearchspace_callrecording =
css.pkid inner join devicenumplanmap as devnumplanmap on rcrdpro.pkid =
devnumplanmap.fkrecordingprofile inner join device as dev on devnumplanmap.fkdevice = dev.pkid
where css.pkid = rcrdpro.fkcallingsearchspace_callrecording and dev.name='SEPC80084AA8743'
name_of_the_recording_css partitions_in_recording_css sortorder

```

```

=====
INTERNAL_CSS          E911_PT              1
INTERNAL_CSS          Phones_PT            2
INTERNAL_CSS          EMERGENCY_PT        3
INTERNAL_CSS         INTERNAL_PT         4
INTERNAL_CSS          INFORMACAST_PT      5

```