排除從Cisco IP電話到Media Sense的介質分叉故障

目錄

簡介

本文檔介紹在MediaSense伺服器上記錄呼叫時從Cisco IP電話進行媒體分流的故障排除步驟。

必要條件

需求

思科建議您瞭解以下主題：

- 思科整合通訊管理員(CUCM)
- Cisco MediaSense

採用元件

本文中的資訊係根據以下軟體和硬體版本：

- CUCM版本10.5.2.10000-5
- Cisco MediaSense 10.0.1.10000-95

本文中的資訊是根據特定實驗室環境內的裝置所建立。文中使用到的所有裝置皆從已清除（預設）的組態來啟動。如果您的網路正在作用，請確保您已瞭解任何指令可能造成的影響。

背景資訊

Cisco MediaSense是一個基於網路的平台，使用會話發起協定(SIP)為網路中的裝置提供語音和影片媒體錄製功能。MediaSense完全整合到思科的統一通訊架構中，可自動捕獲並儲存經過適當配置的CUCM裝置上的每個IP語音(VoIP)會話。

1. MediaSense接受以下格式的音訊編解碼器：
   - g.711 µLaw和aLaw
   - g.722
   - g.729、g.729a、g.729b
   - 高級音訊編碼 — 低延遲(AAC-LD)，也稱為 MPEG音訊第4層 — 低開銷MPEG-4音訊傳輸多路複用(MP4A/LATM)
2. 採用H.264編碼的MediaSense影片

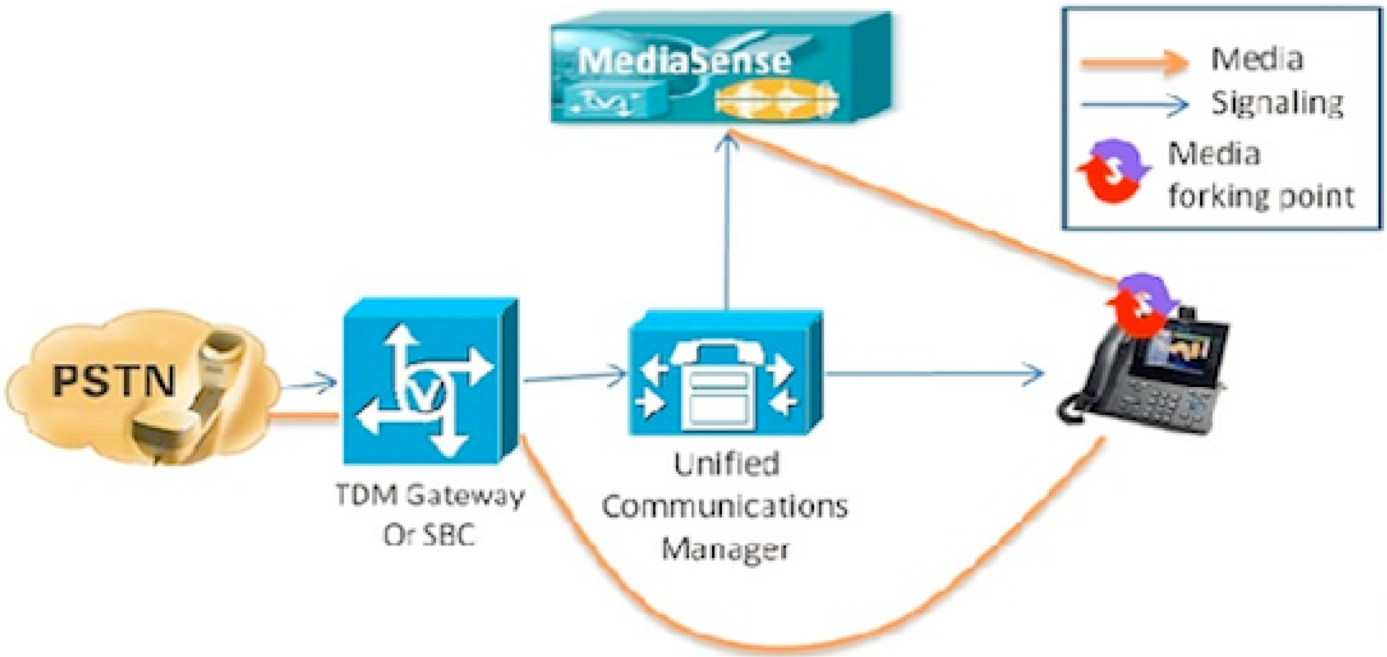
案例

1. Unified Communications Manager基本部署 — 內部到外部
2. Unified Communications Manager基本部署 — 內部到內部

從MediaSense的角度來看，兩種場景實際上沒有區別。

在這兩種情況下，由電話分叉的媒體被傳送到記錄裝置，在此捕獲分叉的流。它們之所以在此區分開來，是因為它們在解決方案級別的行為存在顯著差異。

如下圖所示，Unified Communications Manager Deployment - Internal-to-External。



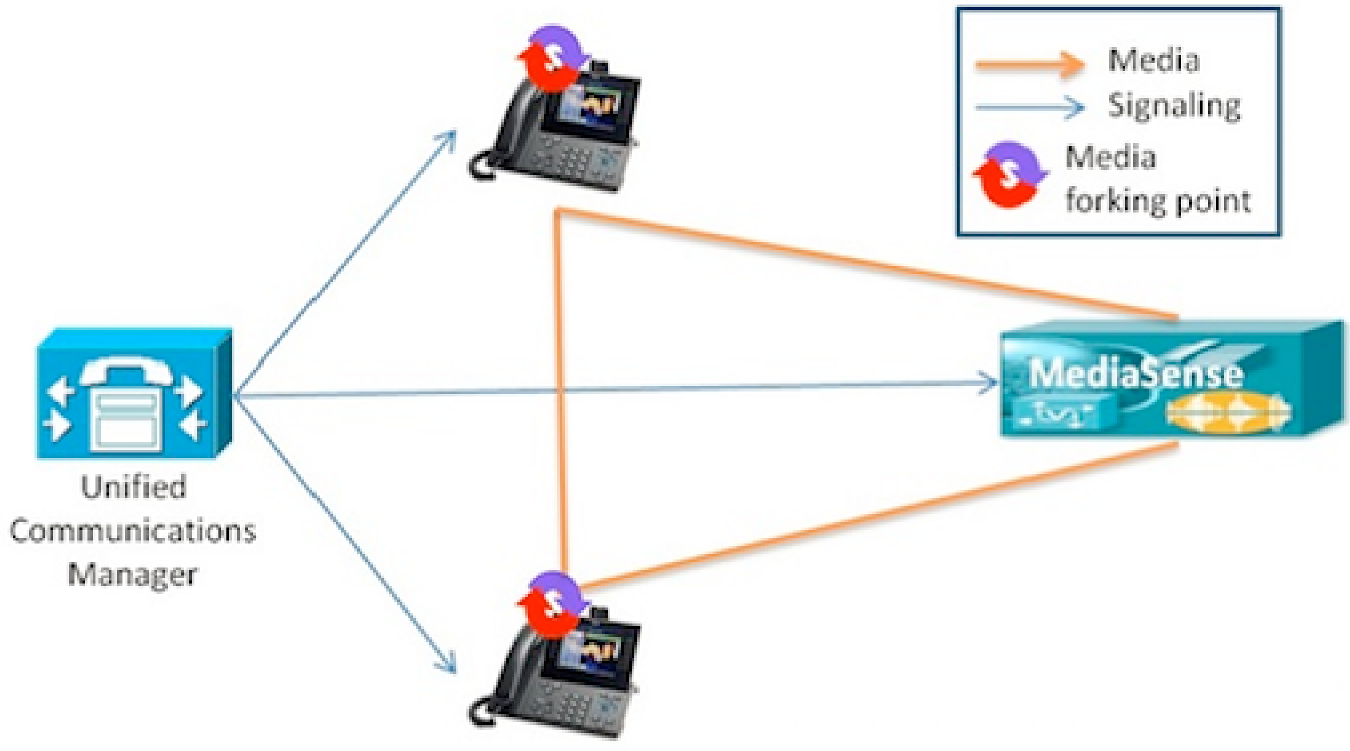
這顯示了一個基本的Unified Communications Manager部署，其中記錄了與外部呼叫者的Cisco IP電話呼叫。只要內部電話配置了相應的錄音配置檔案，此配置適用於入站和出站呼叫。

從信令角度建立連線後，媒體直接從分叉電話流向錄制伺服器。

如果呼叫從此電話轉出，錄音會話將結束。僅當接聽呼叫的電話配置為錄製時，才會捕獲呼叫的下

一段。

如下圖所示，Unified Communications Manager Deployment - Internal-to-Internal。



這顯示了一個基本Unified Communications Manager部署，其中呼叫是在企業內部使用者之間的呼叫。配置其中一個電話進行錄製非常重要。如果兩台電話都配置為錄製，則將捕獲兩個單獨的錄製會話。

疑難排解

本節提供的資訊可用於對組態進行疑難排解。

步驟1. MediaSenseCUCM

CUCM

- 受控裝置和應用程式使用者(AXL)中的許可權資訊。
- 記錄配置檔案和目標地址
- SIP中繼指向MediaSense。
- 路由模式

MediaSense

安裝系統後，可以在MediaSense命令列上使用**show tech call_control_service**命令驗證基本配置。

此命令顯示有關系統上運行的Cisco MediaSense呼叫控制服務的資訊。

Cisco MediaSense呼叫控制服務應該正在運行，才能成功執行此命令。

輸出中捕獲的系統資訊。

```
 admin:show tech call_control_service

<html> <head> <title>mediasense</title> </head> <body> <pre>
--------------------------------



Core: ver=10.0.1


 FCS, op=SHORT
Started at Mon Jul 13 10:55:53 PDT 2015
Report at Tue Jul 21 02:05:26 PDT 2015
Running at mediasense, processors=6, pId=28270
framework: state=In Service; {AMS_ADAPTER=

IN_SERVICE

, SIP_ADAPTER=

IN_SERVICE

, RECORDING_ADAPTER=

IN_SERVICE

}
logLevel=DEBUG, traceMask=0x307, DEBUG traceMask=0x100

System Info:
Memory: used=46.509 MB(13.671 MB), alloc=790.458 MB(0.0 MB)
CPU: avrLoad=0.37, procTime=00:10:18
Threads=176, peakThreads=224
```

在show tech call_control_service輸出中記錄會話資訊。

```
SessionManagerImpl: size=0
Recording Sessions:

started=17

,

completed=17

 (100.0000%), errors=0, processing=0, maxProcessing=1, meanTime=38.310 sec, stDev=76.242 sec,
maxTime=00:05:16, lastTime=38291 mSec
Recording Setup Time:

started=17

,

completed=17

 (100.0000%), errors=0, processing=0, maxProcessing=1, meanTime=201 mSec, stDev=34 mSec,
maxTime=308 mSec, lastTime=142 mSec
```

show tech call_control_service輸出中的SIP介面卡資訊。

```
Sip Adapter:
LocalAddress=
```

10.106.122.178

```
:5060; RemoteAddresses [sip:
```

10.106.122.174

```
:
```

5060

```
 sip:
```

10.106.122.175:5060

```
 ], controlTransport=tcp
based on Cisco Caffeine SIP Stack,
```

version=3.1.3.502

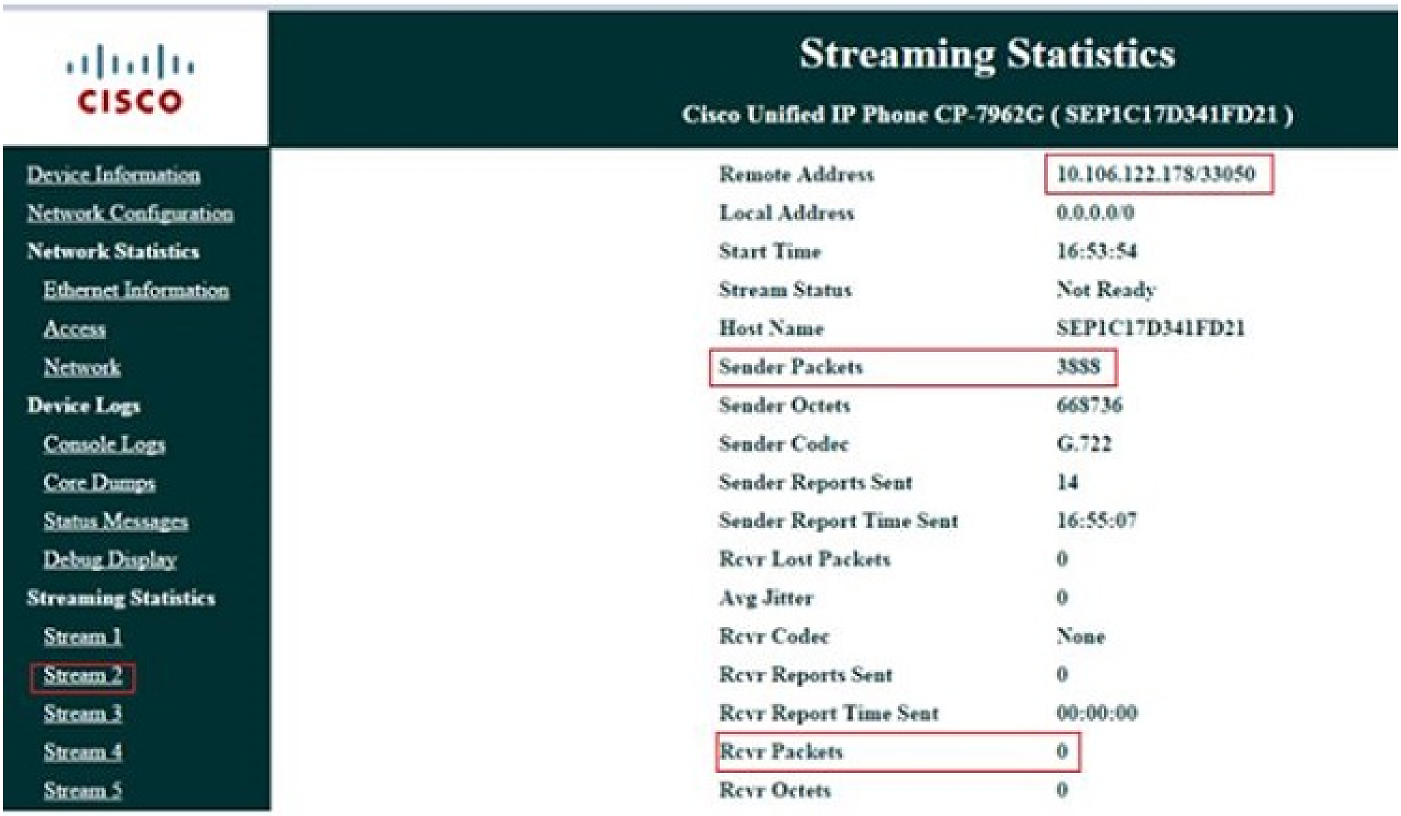
```
, nonBlockingTCP=true, closeConnectionOnTimeout=false
state=AcceptCalls, blockingMode=NONE
SdpUtil: m=audio %d RTP/AVP 102 0 8 9 18, m=video %d RTP/AVP 97
Executor: activeCount=0, poolSize=0, largestPoolSize=2, queueSize=0
```

提示：要設定呼叫錄音，請參閱

步驟2. 檢查電話是否正在將媒體流式傳輸到MediaSense伺服器。

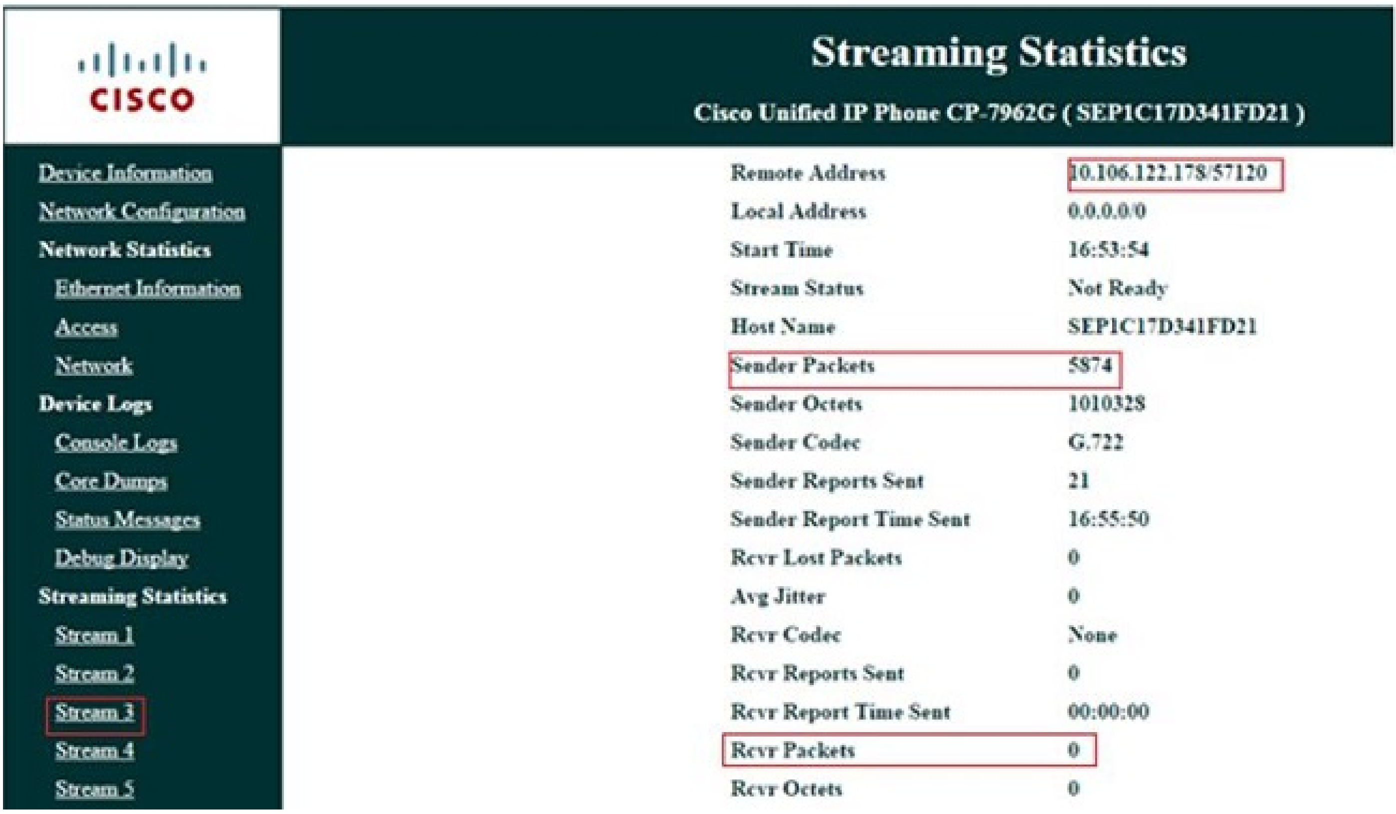
流1將成為對外部呼叫者的呼叫。流2將包含有關到MediaSense伺服器的分叉呼叫的資訊。對於分叉呼叫，接收方資料包將始終保持零。

如圖所示，近端媒體流到MediaSense。



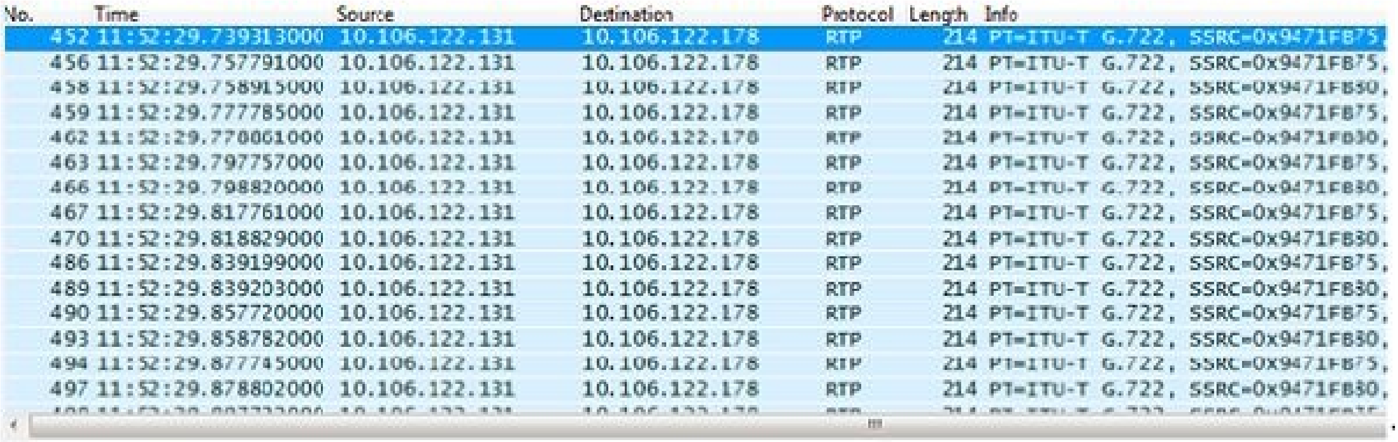
將遠端媒體流傳輸到MediaSense

如下圖所示，流1中接收的遠端媒體的流資訊在流3中被分叉。



您可以在電話上進行Packet Capture（資料包捕獲）驗證。

如下圖所示，電話PCap。



No.	Time	Source	Destination	Protocol	Length	Info
452	11:52:29.739313000	10.106.122.131	10.106.122.178	RTP	214	PT=ITU-T G.722, SSRC=0x9471FB75
456	11:52:29.757791000	10.106.122.131	10.106.122.178	RTP	214	PT=ITU-T G.722, SSRC=0x9471FB75,
458	11:52:29.758915000	10.106.122.131	10.106.122.178	RTP	214	PT=ITU-T G.722, SSRC=0x9471FB30,
459	11:52:29.777785000	10.106.122.131	10.106.122.178	RTP	214	PT=ITU-T G.722, SSRC=0x9471FB75,
462	11:52:29.778061000	10.106.122.131	10.106.122.178	RTP	214	PT=ITU-T G.722, SSRC=0x9471FB30,
463	11:52:29.797757000	10.106.122.131	10.106.122.178	RTP	214	PT=ITU-T G.722, SSRC=0x9471FB75,
466	11:52:29.798820000	10.106.122.131	10.106.122.178	RTP	214	PT=ITU-T G.722, SSRC=0x9471FB30,
467	11:52:29.817761000	10.106.122.131	10.106.122.178	RTP	214	PT=ITU-T G.722, SSRC=0x9471FB75,
470	11:52:29.818829000	10.106.122.131	10.106.122.178	RTP	214	PT=ITU-T G.722, SSRC=0x9471FB30,
486	11:52:29.839199000	10.106.122.131	10.106.122.178	RTP	214	PT=ITU-T G.722, SSRC=0x9471FB75,
489	11:52:29.839203000	10.106.122.131	10.106.122.178	RTP	214	PT=ITU-T G.722, SSRC=0x9471FB30,
490	11:52:29.857720000	10.106.122.131	10.106.122.178	RTP	214	PT=ITU-T G.722, SSRC=0x9471FB75,
493	11:52:29.858782000	10.106.122.131	10.106.122.178	RTP	214	PT=ITU-T G.722, SSRC=0x9471FB30,
494	11:52:29.877745000	10.106.122.131	10.106.122.178	RTP	214	PT=ITU-T G.722, SSRC=0x9471FB75,
497	11:52:29.878802000	10.106.122.131	10.106.122.178	RTP	214	PT=ITU-T G.722, SSRC=0x9471FB30,

提示：請參閱從IP電話收集資料包捕獲

步驟3. 驗證CUCM和MediaSense上的呼叫信令。

此處的示例包含從分機4011的SIP電話到分機4009的SCCP電話的IP呼叫。錄音目的號碼是7878。

CUCM日誌分析

INVITE從SIP電話傳送到CUCM。

```
 06053008.002 |08:39:47.013 |AppInfo  |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from
10.106.122.153 on port 53979 index 44 with 2126 bytes:
[50171,NET]
INVITE sip:4009@10.106.122.174;user=phone SIP/2.0
Via: SIP/2.0/TCP 10.106.122.153:53979;branch=z9hG4bK22e1618f
From: "4011" <sip:4011@10.106.122.174>;tag=203a0782d99f04115d77007a-7abfc08c
To: <sip:4009@10.106.122.174>
Call-ID: 203a0782-d99f000c-57711fea-6ba95503@10.106.122.153
Max-Forwards: 70
Date: Thu, 16 Jul 2015 15:39:46 GMT
CSeq: 101 INVITE


User-Agent: Cisco-CP8945/9.4.2


Contact: <sip:48a499a0-f78e-4baa-a287-5c6eeb0f2fe7@10.106.122.153:53979;transport=tcp>;video
Expires: 180
Accept: application/sdp
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "4011" <sip:4011@10.106.122.174>;party=calling;id-
type=subscriber;privacy=off;screen=yes
Supported: replaces,join,sdp-anat,norefersub,resource-priority,extended-refer,X-cisco-
callinfo,X-cisco-serviceuri,X-cisco-escapecodes,X-cisco-service-control,X-cisco-srtp-fallback,X-
cisco-monrec,X-cisco-config,X-cisco-sis-7.0.0,X-cisco-xsi-8.5.1
Allow-Events: kpml,dialog
Recv-Info: conference
Recv-Info: x-cisco-conference
Content-Length: 986
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
```

```
o=Cisco-SIPUA 15743 0 IN IP4 10.106.122.153
s=SIP Call
b=AS:2000
t=0 0
m=audio
```

16420

```
 RTP/AVP 102 9 0 8 116 18 101
c=IN IP4
```

10.106.122.153

```
a=trafficclass:conversational.audio.avconf.aq:admitted
a=rtpmap:102 L16/16000
a=rtpmap:9 G722/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

UserAgent是向CUCM傳送As的Cisco 8945 IP電話。

當SCCP電話應答呼叫並且會話建立時，CUCM會向SIP電話傳送ACK。

```
06053236.001 |08:39:49.777 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to
10.106.122.153 on port 53979 index 44
[50174,NET]
SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.106.122.153:53979;branch=z9hG4bK22e1618f
From: "4011" <sip:4011@10.106.122.174>;tag=203a0782d99f04115d77007a-7abfc08c
To: <sip:4009@10.106.122.174>;tag=16789~78868996-a8aa-4784-b765-86098b176d95-32833193
Date: Thu, 16 Jul 2015 15:39:47 GMT
Call-ID: 203a0782-d99f000c-57711fea-6ba95503@10.106.122.153
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Server: Cisco-CUCM10.5
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; gci= 1-
7171; isVoip; call-instance= 1
Send-Info: conference, x-cisco-conference
Remote-Party-ID: <sip:4009@10.106.122.174>;party=called;screen=yes;privacy=off
Remote-Party-ID: <sip:4009@10.106.122.174;user=phone>;party=x-cisco-original-called;privacy=off
Contact: <sip:4009@10.106.122.174:5060;transport=tcp>
Content-Type: application/sdp
Content-Length: 435

v=0
o=CiscoSystemsCCM-SIP 16789 1 IN IP4 10.106.122.174
s=SIP Call
c=IN IP4
```

10.106.122.131

```
b=AS:64
t=0 0
m=audio
```

18840

```
 RTP/AVP 9 101
a=ptime:20
a=rtpmap:9 G722/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=trafficclass:conversational.audio.aq:admitted
```

電話按下「錄音」軟鍵，指示使用者呼叫錄音功能。

```
06053271.001 |08:39:52.681 |AppInfo  |StationInit: (0000045) SoftKeyEvent
```

softKeyEvent=74(Record)

```
 lineInstance=1 callReference=32833194.
```

為錄製鎖定了編解碼器。

```
06053274.002 |08:39:52.681 |AppInfo  | StationCdpc: star_MediaExchangeAgenaQueryCapability -
Device SEP1C17D341FD21, codec locked due to recording,
```

codecType=6

分配內建網橋(BiB)資源。

```
06053309.000 |08:39:52.682 |SdlSig   |AllocateBibResourceRes
|resource_rsvp                  |MediaResourceCdpc(1,100,139,52)
|BuiltInBridgeControl(1,100,239,6) |1,100,14,269032.3452^10.106.122.131^SEP1C17D341FD21 |[R:N-
H:0,N:0,L:0,V:0,Z:0,D:0]  CI=32833195 BridgeDn=
```

b00123906001

```
 Pid=100,1,63,45 SsType=16777245 SsKey=43 deviceCap=0
```

CUCM在BiB資源中撥號。

```
06053318.008 |08:39:52.683 |AppInfo  ||PretransformCallingPartyNumber=
|CallingPartyNumber=
|DialingPartition=
|DialingPattern=
```

b00123906001

```
|FullyQualifiedCalledPartyNumber=
```

b00123906001

然後BiB撥打到MediaSense錄製號碼7878。

```
06053358.013 |08:39:52.686 |AppInfo  ||PretransformCallingPartyNumber=b00123906001
|CallingPartyNumber=
```

b00123906001

```
|DialingPartition=
|DialingPattern=
```

7878

```
|FullyQualifiedCalledPartyNumber=
```

7878

INVITE被傳送到MediaSense。

```
06053416.001 |08:39:52.690 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to
10.106.122.178 on port 5060 index 71
[50176,NET]
INVITE sip:7878@10.106.122.178:5060 SIP/2.0
Via: SIP/2.0/TCP 10.106.122.174:5060;branch=z9hG4bK14432e0a687
From: <sip:
```

4009

```
@10.106.122.174;x-nearend;x-refci=32833194;x-nearendclusterid=StandAloneCluster;x-
nearenddevice=SEP1C17D341FD21;x-nearendaddr=4009;x-farendrefci=32833193;x-
farendclusterid=StandAloneCluster;x-farenddevice=SEP203A0782D99F;x-
farendaddr=4011>;tag=16791~78868996-a8aa-4784-b765-86098b176d95-32833198
To: <sip:7878@10.106.122.178>
Date: Thu, 16 Jul 2015 15:39:52 GMT
Call-ID: e4fb9980-5a71d048-b0-ae7a6a0a@10.106.122.174
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: <sip:10.106.122.174:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 3841694080-0000065536-0000000071-2927258122
Session-Expires: 1800
P-Asserted-Identity: <sip:4009@10.106.122.174>
Remote-Party-ID: <sip:4009@10.106.122.174>;party=calling;screen=yes;privacy=off
Contact: <sip:4009@10.106.122.174:5060;transport=tcp>;isFocus
Max-Forwards: 70
Content-Length: 0
```

建立錄製呼叫時，從MediaSense獲得200 OK。

```
06053554.002 |08:39:52.831 |AppInfo  |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from
10.106.122.178 on port 5060 index 71 with 1013 bytes:
[50181,NET]
SIP/2.0 200 Ok
Via: SIP/2.0/TCP 10.106.122.174:5060;branch=z9hG4bK14432e0a687
To: <sip:7878@10.106.122.178>;tag=ds606d34cb
From: <sip:4009@10.106.122.174;x-nearend;x-refci=32833194;x-
nearendclusterid=StandAloneCluster;x-nearenddevice=SEP1C17D341FD21;x-nearendaddr=4009;x-
farendrefci=32833193;x-farendclusterid=StandAloneCluster;x-farenddevice=SEP203A0782D99F;x-
farendaddr=4011>;tag=16791~78868996-a8aa-4784-b765-86098b176d95-32833198
Call-ID: e4fb9980-5a71d048-b0-ae7a6a0a@10.106.122.174
CSeq: 101 INVITE
Content-Length: 313
Contact: <sip:7878@10.106.122.178:5060;transport=tcp>
Content-Type: application/sdp
Allow: INVITE, BYE, CANCEL, ACK, NOTIFY, INFO, UPDATE
Server: MediaSense/10.x

v=0
o=CiscoORA 3197 1 IN IP4 10.106.122.178
s=SIP Call
c=IN IP4
```

10.106.122.178

```
t=0 0
m=audio
```

42120

```
 RTP/AVP 102 0 8 9 18
a=rtpmap:102 MP4A-LATM/90000
a=fmtp:102 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=
```

recvonly

返回MediaSense。

```
 06053719.001 |08:39:52.842 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to
10.106.122.178 on port 5060 index 71
[50183,NET]
ACK sip:7878@10.106.122.178:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.106.122.174:5060;branch=z9hG4bK147605d100d
From: <sip:4009@10.106.122.174;x-nearend;x-refci=32833194;x-
nearendclusterid=StandAloneCluster;x-nearenddevice=SEP1C17D341FD21;x-nearendaddr=4009;x-
farendrefci=32833193;x-farendclusterid=StandAloneCluster;x-farenddevice=SEP203A0782D99F;x-
farendaddr=4011>;tag=16791~78868996-a8aa-4784-b765-86098b176d95-32833198
To: <sip:7878@10.106.122.178>;tag=ds606d34cb
Date: Thu, 16 Jul 2015 15:39:52 GMT
Call-ID: e4fb9980-5a71d048-b0-ae7a6a0a@10.106.122.174
User-Agent: Cisco-CUCM10.5
```

```
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Type: application/sdp
Content-Length: 260

v=0
o=CiscoSystemsCCM-SIP 16791 1 IN IP4 10.106.122.174
s=SIP Call
c=IN IP4
```

10.106.122.131

```
b=TIAS:64000
b=CT:64
b=AS:64
t=0 0
m=audio
```

4000

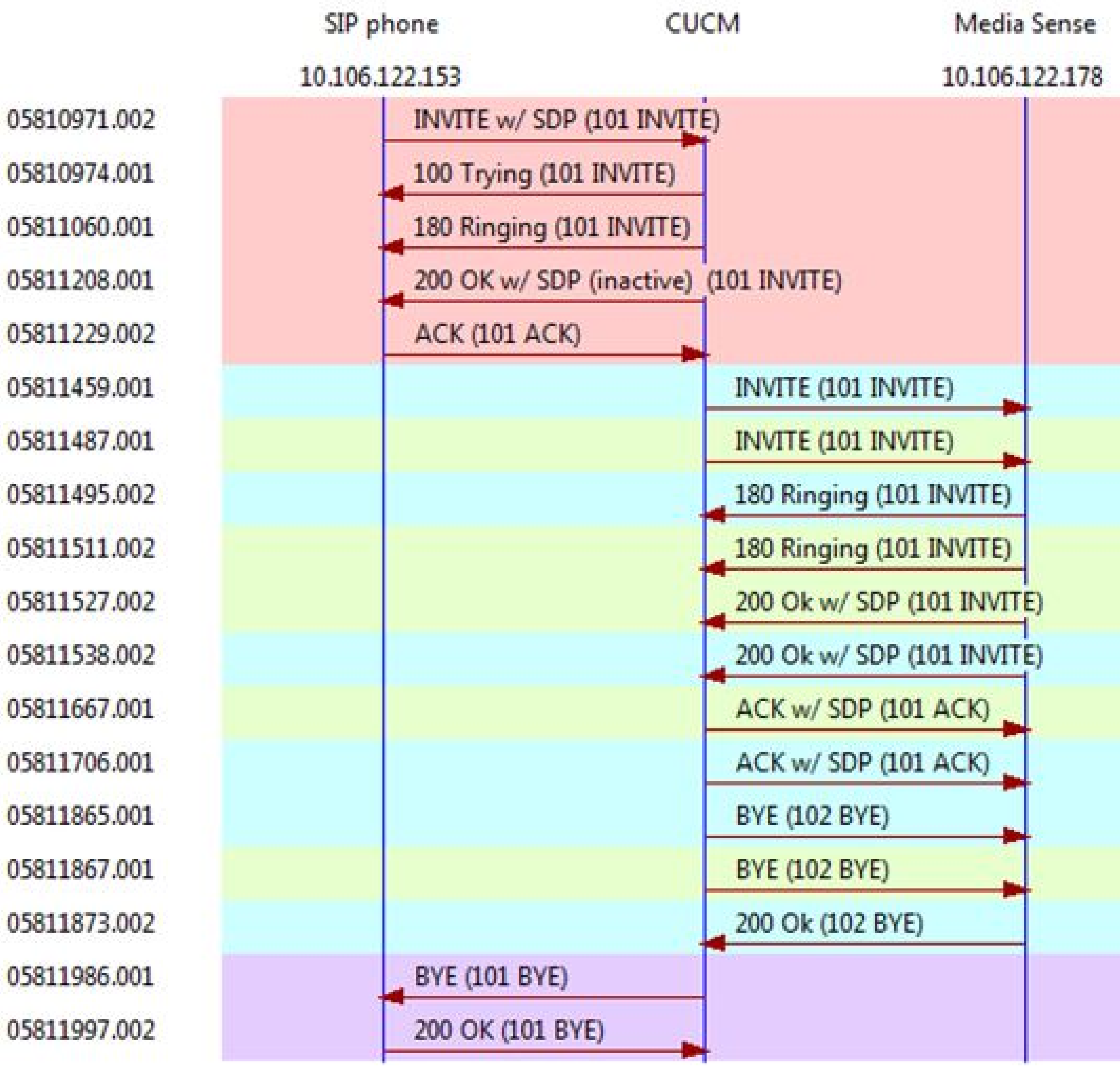
```
 RTP/AVP 9 101
a=ptime:20
a=rtpmap:9 G722/8000
a=
```

sendonly

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

對遠端流重複相同的過程。在BiB中，CUCM撥號，BiB將撥打錄製號碼，並在CUCM和MediaSense之間建立SIP會話。

如本圖所示，訊號圖表。



MediaSense日誌分析

從CUCM邀請，為近端建立呼叫記錄（來自SIP IP電話的音訊）

```
0000010803: 10.106.122.178: Jul 16 2015 08:39:52.694 -0700: %CCBU_CALL_CONTROL-6-BORDER_MESSAGE:
{Thrd=Pool-sip-thread-25} %[message_string=process new Invitation: SipCall-25,
INBOUND_RECORDING, null, State=ALERTED: , processing=1
INVITE sip:7878@10.106.122.178:5060 SIP/2.0
Via: SIP/2.0/TCP 10.106.122.174:5060;branch=z9hG4bK14432e0a687
Max-Forwards: 69
To: <sip:7878@10.106.122.178>
From: <sip:4009@10.106.122.174;x-nearend;x-refci=32833194;x-
nearendclusterid=StandAloneCluster;x-nearenddevice=SEP1C17D341FD21;x-nearendaddr=4009;x-
farendrefci=32833193;x-farendclusterid=StandAloneCluster;x-farenddevice=SEP203A0782D99F;x-
farendaddr=4011>;tag=16791~78868996-a8aa-4784-b765-86098b176d95-32833198
Call-ID: e4fb9980-5a71d048-b0-ae7a6a0a@10.106.122.174
CSeq: 101 INVITE
Content-Length: 0
```

```
Date: Thu, 16 Jul 2015 15:39:52 GMT
Supported: timer,resource-priority,replaces
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Expires: 180
Allow-Events: presence, kpml
Call-Info: <sip:10.106.122.174:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 3841694080-0000065536-0000000071-2927258122
Session-Expires: 1800
P-Asserted-Identity: <sip:4009@10.106.122.174>
Remote-Party-ID: <sip:4009@10.106.122.174>;party=calling;screen=yes;privacy=off
Contact: <sip:4009@10.106.122.174:5060;transport=tcp>;isfocus


 ]: Border Message
0000010804: 10.106.122.178: Jul 16 2015 08:39:52.694 -0700: %CCBU_CALL_CONTROL-7-TRACE:
{Thrd=Pool-sip-thread-25} -preProcessInvitation SipCall-25, INBOUND_RECORDING, null,
State=ALERTED: ciscoGuidHeader=Cisco-Guid: 3841694080-0000065536-0000000071-2927258122

0000010808: 10.106.122.178: Jul 16 2015 08:39:52.695 -0700: %CCBU_CALL_CONTROL-7-TRACE:
{Thrd=Pool-sip-thread-25} -postProcessInvitation SipCall-25, INBOUND_RECORDING, NEAR_END,
State=ALERTED: from=4009, displayName=null, xRefci=32833194,
```

endPointType=NEAR_END

```
, xNearDevice=SEP1C17D341FD21, ucmCiscoGuid=null, nearEndClusterId=StandAloneCluster, and
farEndClusterId=StandAloneCluster


0000010809: 10.106.122.178: Jul 16 2015 08:39:52.695 -0700: %CCBU_CALL_CONTROL-7-TRACE:
{Thrd=Pool-sip-thread-25} -postProcessInvitation SipCall-25, INBOUND_RECORDING, NEAR_END,
State=ALERTED: created MediaResources: [AUDIO-MediaResource-25: SipCall-25, INBOUND_RECORDING,
NEAR_END, State=ALERTED, weight=1, ip=
```

10.106.122.174

```
]
```

從CUCM發出邀請，為遠端建立呼叫記錄（來自SCCP IP電話的音訊）。

```
 0000010818: 10.106.122.178: Jul 16 2015 08:39:52.700 -0700: %CCBU_CALL_CONTROL-6-
BORDER_MESSAGE: {Thrd=Pool-sip-thread-26} %[message_string=process new Invitation: SipCall-26,
INBOUND_RECORDING, null, State=ALERTED: , processing=2
INVITE sip:7878@10.106.122.178:5060 SIP/2.0
Via: SIP/2.0/TCP 10.106.122.174:5060;branch=z9hG4bK14578497f79
Max-Forwards: 69
To: <sip:7878@10.106.122.178>
From: <sip:4009@10.106.122.174;x-farend;x-refci=32833194;x-nearendclusterid=StandAloneCluster;x-
nearenddevice=SEP1C17D341FD21;x-nearendaddr=4009;x-farendrefci=32833193;x-
farendclusterid=StandAloneCluster;x-farenddevice=SEP203A0782D99F;x-
farendaddr=4011>;tag=16792~78868996-a8aa-4784-b765-86098b176d95-32833201
Call-ID: e4fb9980-5a71d048-b1-ae7a6a0a@10.106.122.174
CSeq: 101 INVITE
Content-Length: 0
Date: Thu, 16 Jul 2015 15:39:52 GMT
Supported: timer,resource-priority,replaces
Supported: X-cisco-srtp-fallback
```

```
Supported: Geolocation
Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Expires: 180
Allow-Events: presence, kpml
Call-Info: <sip:10.106.122.174:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 3841694080-0000065536-0000000072-2927258122
Session-Expires: 1800
P-Asserted-Identity: <sip:4009@10.106.122.174>
Remote-Party-ID: <sip:4009@10.106.122.174>;party=calling;screen=yes;privacy=off
Contact: <sip:4009@10.106.122.174:5060;transport=tcp>;isfocus
```

```
 ]: Border Message
0000010819: 10.106.122.178: Jul 16 2015 08:39:52.700 -0700: %CCBU_CALL_CONTROL-7-TRACE:
{Thrd=Pool-sip-thread-26} -preProcessInvitation SipCall-26, INBOUND_RECORDING, null,
State=ALERTED: ciscoGuidHeader=Cisco-Guid: 3841694080-0000065536-0000000072-2927258122

0000010823: 10.106.122.178: Jul 16 2015 08:39:52.701 -0700: %CCBU_CALL_CONTROL-7-TRACE:
{Thrd=Pool-sip-thread-26} -postProcessInvitation SipCall-26, INBOUND_RECORDING, NEAR_END,
State=ALERTED: from=4009, displayName=null, xRefci=32833194,
```

endPointType=FAR_END

```
, xNearDevice=null, ucmCiscoGuid=null, nearEndClusterId=StandAloneCluster, and
farEndClusterId=StandAloneCluster

0000010824: 10.106.122.178: Jul 16 2015 08:39:52.701 -0700: %CCBU_CALL_CONTROL-7-TRACE:
{Thrd=Pool-sip-thread-26} -postProcessInvitation SipCall-26, INBOUND_RECORDING, NEAR_END,
State=ALERTED: created MediaResources: [AUDIO-MediaResource-26: SipCall-26, INBOUND_RECORDING,
FAR_END, State=ALERTED, weight=1, ip=
```

10.106.122.174

在MediaSense上捕獲近端和遠端記錄資訊的SIP支路後，為呼叫建立的會話ID。

```
 0000010830: 10.106.122.178: Jul 16 2015 08:39:52.703 -0700: %CCBU_CALL_CONTROL-7-TRACE:
{Thrd=Pool-sip-thread-26} -Core: dispatch StartRecordingRequestEvent: SipRequestContextImpl-76,
type=Sip, Session:
```

d14e97859bff1

```
, INITIALIZING, call=SipCall-26, INBOUND_RECORDING, FAR_END, State=ALERTED, firstCall=SipCall-
25, INBOUND_RECORDING, NEAR_END, State=ALERTED, requestedAudioPorts=2, requestedVideoPorts=0,
append=false, audioSdp=null to Recording Adapter
```

200 OK和ACK用於近端呼叫。

```
 0000010846: 10.106.122.178: Jul 16 2015 08:39:52.829 -0700: %CCBU_CALL_CONTROL-6-
BORDER_MESSAGE: {Thrd=Pool-capture-thread-38} %[message_string=SipCall-25, INBOUND_RECORDING,
NEAR_END, State=ALERTED send 200 Ok:
SIP/2.0 200 Ok
Via: SIP/2.0/TCP 10.106.122.174:5060;branch=z9hG4bK14432e0a687
To: <sip:7878@10.106.122.178>;tag=ds606d34cb
From: <sip:4009@10.106.122.174;x-nearend;x-refci=32833194;x-
nearendclusterid=StandAloneCluster;x-nearenddevice=SEP1C17D341FD21;x-nearendaddr=4009;x-
```

```
farendrefci=32833193;x-farendclusterid=StandAloneCluster;x-farenddevice=SEP203A0782D99F;x-
farendaddr=4011>;tag=16791~78868996-a8aa-4784-b765-86098b176d95-32833198
Call-ID: e4fb9980-5a71d048-b0-ae7a6a0a@10.106.122.174
CSeq: 101 INVITE
Content-Length: 313
Contact: <sip:7878@10.106.122.178:5060;transport=tcp>
Content-Type: application/sdp
Allow: INVITE, BYE, CANCEL, ACK, NOTIFY, INFO, UPDATE
Server: MediaSense/10.x

v=0
o=CiscoORA 3197 1 IN IP4 10.106.122.178
s=SIP Call
c=IN IP4
```

10.106.122.178

```
t=0 0
m=audio
```

42120

```
 RTP/AVP 102 0 8 9 18
a=rtpmap:102 MP4A-LATM/90000
a=fmtp:102 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=
```

recvonly

```
 ACK sip:7878@10.106.122.178:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.106.122.174:5060;branch=z9hG4bK147605d100d
Max-Forwards: 69
To: <sip:7878@10.106.122.178>;tag=ds606d34cb
From: <sip:4009@10.106.122.174;x-nearend;x-refci=32833194;x-
nearendclusterid=StandAloneCluster;x-nearenddevice=SEP1C17D341FD21;x-nearendaddr=4009;x-
farendrefci=32833193;x-farendclusterid=StandAloneCluster;x-farenddevice=SEP203A0782D99F;x-
farendaddr=4011>;tag=16791~78868996-a8aa-4784-b765-86098b176d95-32833198
Call-ID: e4fb9980-5a71d048-b0-ae7a6a0a@10.106.122.174
CSeq: 101 ACK
Content-Length: 260
Date: Thu, 16 Jul 2015 15:39:52 GMT
User-Agent: Cisco-CUCM10.5
Allow-Events: presence, kpml
Content-Type: application/sdp

v=0
o=CiscoSystemsCCM-SIP 16791 1 IN IP4 10.106.122.174
s=SIP Call
c=IN IP4
```

10.106.122.131

```
b=TIAS:64000
```

```
b=CT:64
b=AS:64
t=0 0
m=audio
```

4000

```
 RTP/AVP 9 101
a=ptime:20
a=rtpmap:9 G722/8000
a=
```

sendonly

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

Media Sense應答呼叫後，將捕獲類似事件。請注意，傳送的ACK包含埠4000並表示**sendonly。**

建立兩個SIP對話方塊後的會話資訊。


```
 {"sessionData": {
"callControllerIP": "10.106.122.174",
"callControllerType": "Cisco-CUCM",
"endPoints": [
{
"clusterid": "StandAloneCluster",
"conference": false,
"device": "
```

SEP1C17D341FD21

```
",
"dn": "
```

4009

```
",
"startDate": 1437061192882,
"tracks": [{
"codec": "
```

G722

```
",
"location": "/common",
"mediaState": "
```

ACTIVE

```
",
"startDate": 1437061192882,
"track": 0,
"type": "AUDIO"
}],
"type": "
```

NEAR_END

",
"xRefci": "32833194"
},
{
"clusterid": "StandAloneCluster",
"conference": false,
"device": "

SEP203A0782D99F

",
"dn": "

4011

",
"startDate": 1437061192882,
"tracks": [{
"codec": "G722",
"location": "/common",
"mediaState": "ACTIVE",
"startDate": 1437061192882,
"track": 1,
"type": "AUDIO"
}],
"type": "

FAR_END

",
"xRefci": "32833193"
}
],
"operationType": "

ADD

",
"recordingServer": "10.106.122.178",
"rtspUrl": "rtsp://10.106.122.178/d14e97859bff1",
"sessionName": "

d14e97859bff1

",
"sipServer": "10.106.122.178",
"startDate": 1437061192882,
"state": "

ACTIVE

",
"version": 7

當呼叫斷開時，電話將停止錄製。

```
 0000010897: 10.106.122.178: Jul 16 2015 08:40:01.525 -0700: %CCBU_CALL_CONTROL-7-TRACE:
{Thrd=DIALOG_CALLBACK.7} -Core: dispatch

StopRecordingRequestEvent

: SipRequestContextImpl-78, type=Sip, Session:

d14e97859bff1

, ACTIVE, call=SipCall-26, INBOUND_RECORDING, FAR_END, State=DISCONNECTED, firstCall=null to
Recording Adapter
0000009368: 10.106.122.178: Jul 16 2015 08:40:01.762 -0700: %CCBU_COMMON-6-VSMS HTTP Info:
{Thrd=Pool-capture-thread-39} %[HTTP Response Body=<Session>
<diskusage>
<recording name="

d14e97859bff1

-TRACK0"

size="1"

 repository="/common" />
<recording name="

d14e97859bff1

-TRACK1"

size="1"

 repository="/common" />
</diskusage>
<rtsplink>/archive/

d14e97859bff1

</rtsplink>
```

> **註**：在此區域中，您注意到記錄屬性中有一個大小。此範例顯示**size="1"**，這表示MediaSense確實從CUCM接收了音訊。如果您注意到**size="0"**，則表示MediaSense未從CUCM接收音訊。

會話最終會關閉。


```
 {"sessionData": {
"callControllerIP": "10.106.122.174",
"callControllerType": "Cisco-CUCM",
"endDate": 1437061201522,
"endPoints": [
{
"clusterid": "StandAloneCluster",
"conference": false,
"device": "

SEP1C17D341FD21
```

",
"dn": "

4009

",
"startDate": 1437061192882,
"tracks": [{
"codec": "G722",
"location": "/common",
"mediaState": "ACTIVE",
"size": 1,
"startDate": 1437061192882,
"track": 0,
"type": "AUDIO"
}],
"type": "

NEAR_END

",
"xRefci": "32833194"
},
{
"clusterid": "StandAloneCluster",
"conference": false,
"device": "

SEP203A0782D99F

",
"dn": "

4011

",
"startDate": 1437061192882,
"tracks": [{
"codec": "G722",
"location": "/common",
"mediaState": "ACTIVE",
"size": 1,
"startDate": 1437061192882,
"track": 1,
"type": "AUDIO"
}],
"type": "

FAR_END

",
"xRefci": "32833193"
}
],
"operationType": "EXISTING",
"recordingServer": "10.106.122.178",
"rtspUrl": "rtsp://10.106.122.178/archive/d14e97859bff1",
"sessionName": "

d14e97859bff1

```
",
"sipServer": "10.106.122.178",
"startDate": 1437061192882,
"state": "
```

CLOSED

```
",
"version": 11
```


從MediaSense收集日誌

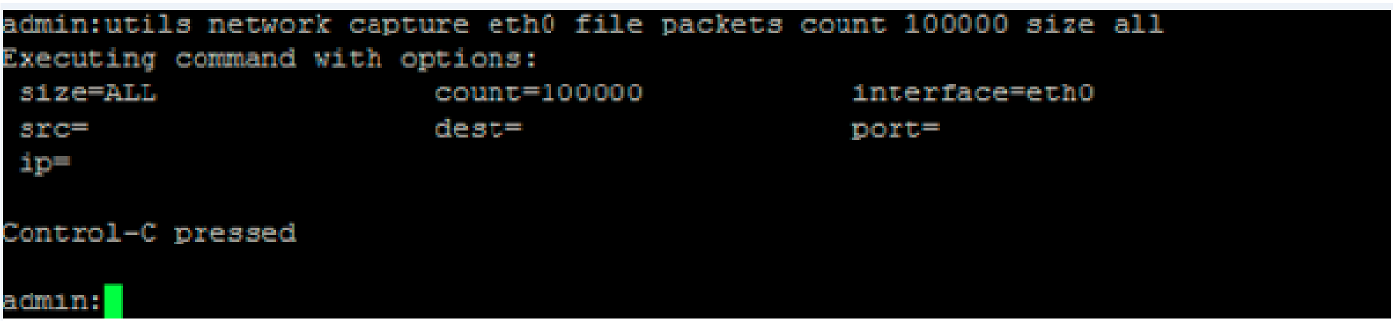
步驟1. 在MediaSense可維護性中啟用呼叫控制服務跟蹤級別進行調試。

如本圖所示，MediaSense可維護性。

步驟2. 在MediaSense上啟用資料包捕獲。

請運行**utils network capture eth0 file packets count 100000 size all**，以便在MediaSense上啟用資料包捕獲。

如圖所示，MediaSense上的資料包捕獲。

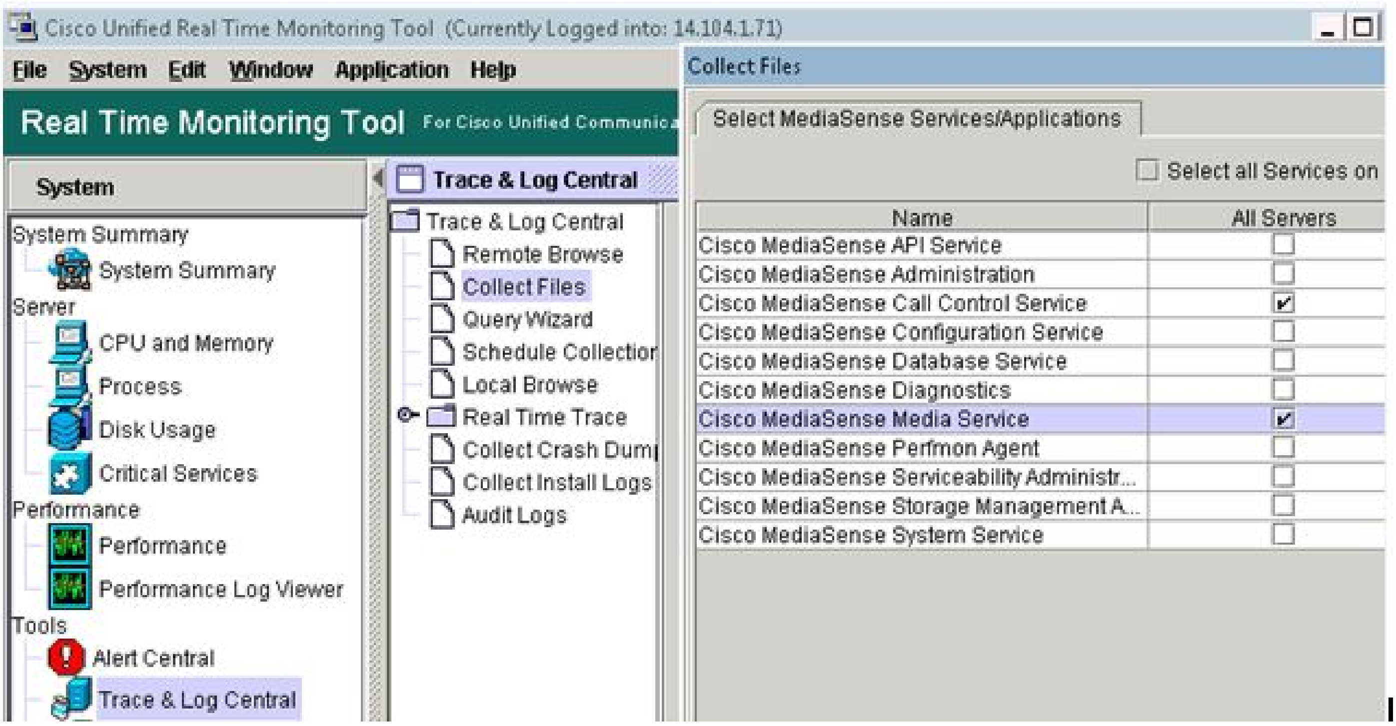


步驟3. 使用即時監控工具(RTMT)收集日誌

使用RTMT連線到MediaSense伺服器。

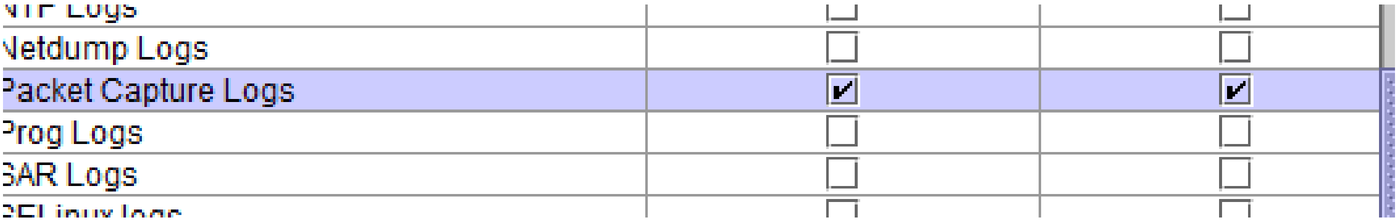
導覽至**追蹤與記錄中心 > 收集檔案**

如本圖所示，即時監視工具。



按一下**Next**並選擇**packet capture**

如本圖所示，即時監視工具。



相應地選擇時間。

一些有用的命令：

1.利**用媒體recording_sessions**

utils media recording_sessions file fileName命令生成一個html檔案，其中包含此Cisco MediaSense伺服器處理的最後100個記錄會話的詳細清單。執行此命令之前，請確認Cisco MediaSense呼叫控制服務正在運行。檔案將儲存到platform/cli/資料夾中，並可使用file get activelog platform/cli/fileName命令下載。

命令：**utils media recording_sessions** fileName

詳細資訊：

- **file**是將資訊輸出到檔案的必需引數。
- **fileName**是定義.html檔名稱的必需引數。
- 當您發出此命令時，會收到以下響應：儲存到platform/cli/<filename>.html的Cisco MediaSense呼叫控制服務記錄會話。現在可以使用：file get activelog platform/cli/<filename>.html下載檔案，然後從該目錄檢索檔案並將其儲存到您選擇的位置。

範例：

- **utils media recording_sessions** file sessions.html Cisco MediaSense。呼叫控制服務錄製會話儲存到platform/cli/sessions.html。現在可以使用file get activelog platform/cli/sessions.html下載它

2. utils**系統維護**

命令**utils system maintenance**操作啟用或禁用Cisco MediaSense上的維護模式，或顯示Cisco MediaSense維護模式狀態。當處於維護模式時，Cisco MediaSense無法處理任何錄製請求或API請求。

Cisco MediaSense在進入維護模式時重新啟動。任何流媒體活動都會突然結束。任何活動的錄音以CLOSED_ERROR狀態結束。Cisco MediaSense在禁用維護模式並重新進入正常模式後再次重新啟動。

命令：使**用系統維護操**作

詳細資訊：操作指定命令執行的操作。

有效的操作包括：

- 啟用
- 禁用
- 狀態

示例：

- utils system maintenance enable
- utils系統維護禁用
- utils系統維護狀態

一些基本問題

MediaSense文檔維基

已知瑕疵

CSCup24364:C all recording not working for calls with no caller id get錯誤消息。

CSCui13760: MediaSense不支援從群集中刪除節點。

CSCtn45420:MediaSense呼叫記錄失敗，使用camelot SIP終端。

CSCut09446: MediaSense UI未填充CUCM配置和API使用者配置。

CSCuo95309:未從其他節點填充MediaSense搜尋和播放錄製。

CSCuq20108:使用跳脫字元時，從標頭到被截斷。

關於此翻譯