

排除从思科IP电话到媒体感知的媒体分流故障

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简介

本文档介绍从Cisco IP电话分流媒体以在MediaSense服务器上记录呼叫的故障排除步骤。

先决条件

要求

Cisco 建议您了解以下主题：

- 思科统一通信管理器 (CUCM)
- 思科 MediaSense

使用的组件

本文档中的信息基于以下软件和硬件版本：

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

本文档中的信息都是基于特定实验室环境中的设备编写的。本文档中使用的所有设备最初均采用原始（默认）配置。如果您使用的是真实网络，请确保您已经了解所有命令的潜在影响。

背景信息

Cisco MediaSense是一个基于网络的平台，使用会话初始协议(SIP)为网络中的设备提供语音和视频媒体录制功能。MediaSense完全集成到思科的统一通信架构中，可自动捕获并存储每个IP语音(VoIP)会话，并将其存储在正确配置的CUCM设备上。

1. MediaSense接受以下格式的音频编解码器：

- g.711 μ Law和aLaw
- g.722
- g.729、g.729a、g.729b
- 高级音频编码 — 低延迟(AAC-LD)，也称为 MPEG音频第4层 — 低开销MPEG-4音频传输复用(MP4A/LATM)

2. H.264编码中的MediaSense视频

场景

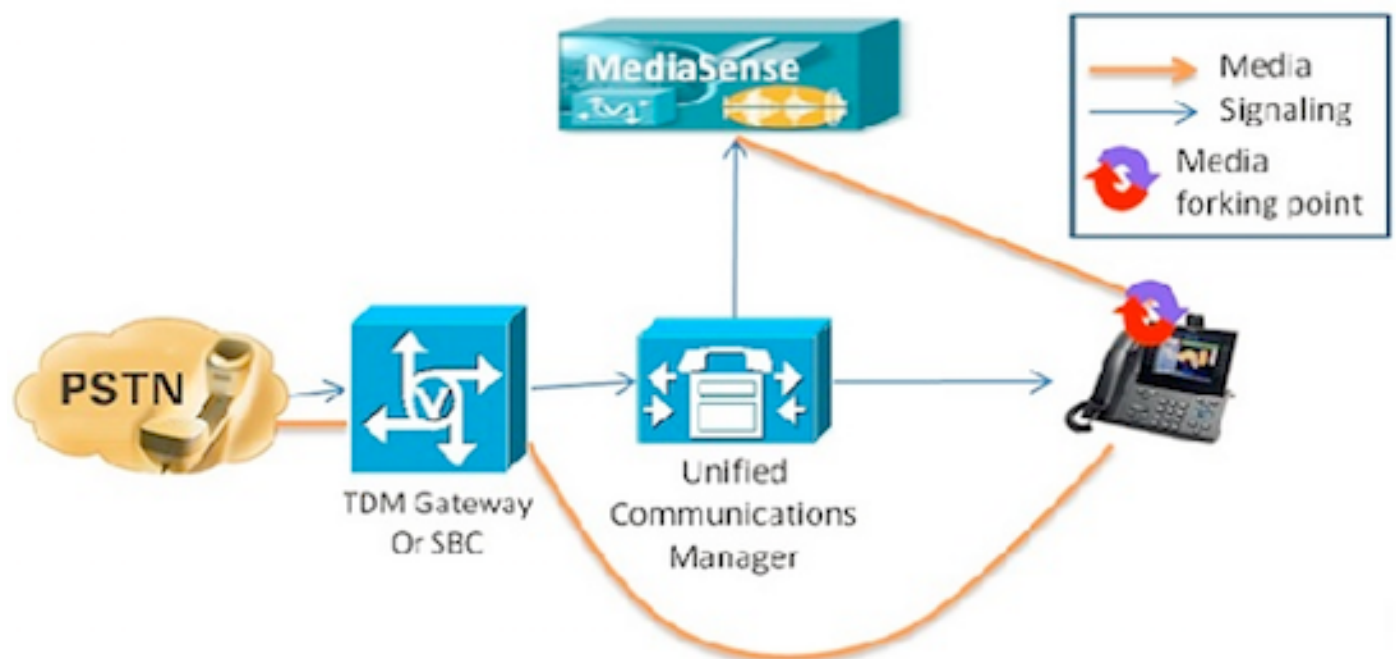
1. 基本统一通信管理器部署 — 从内部到外部

2. 基本统一通信管理器部署 — 内部到内部

从MediaSense的角度来看，两种场景实际上没有区别。

在这两种情况下，由电话分组的媒体都被发送到记录设备，在该设备中分组的流被捕获。他们之所以能在这里脱颖而出，是因为他们在解决方案级别的行为存在显著差异。

如本图所示，Unified Communications Manager部署 — 从内部到外部。

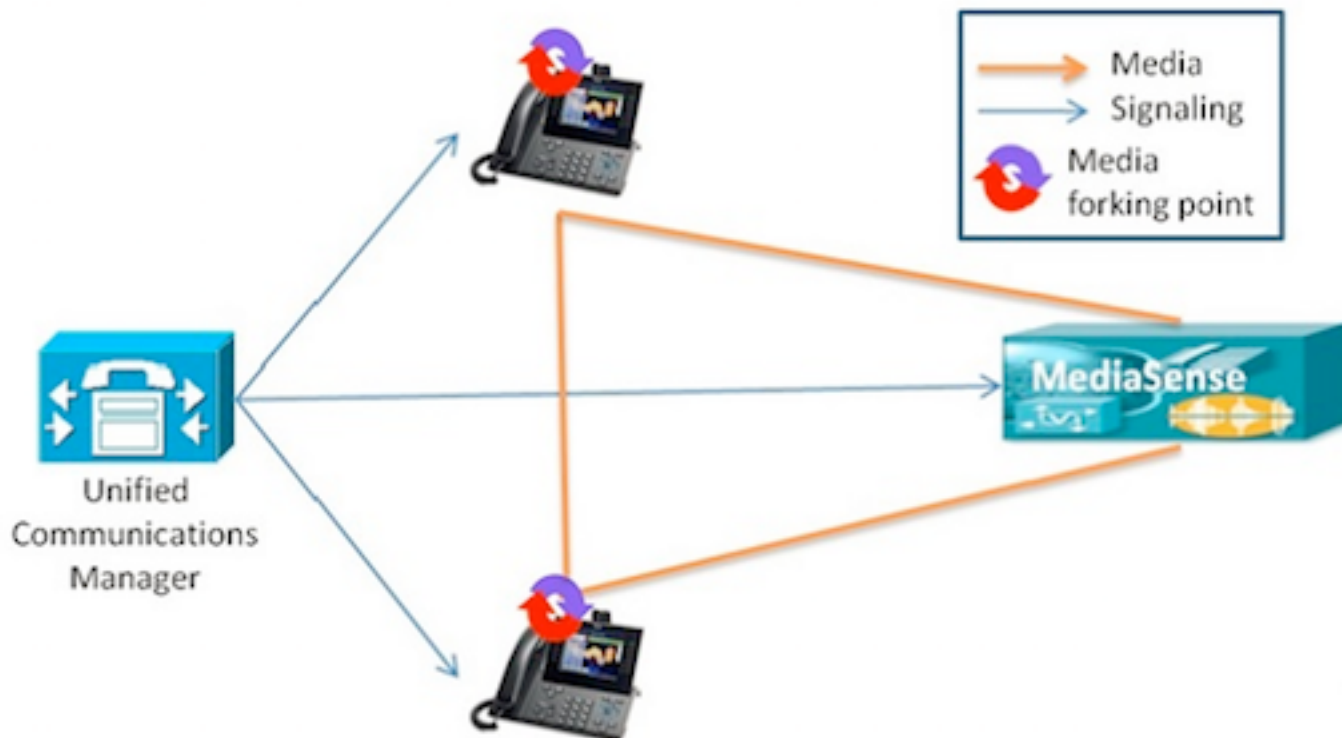


这显示了基本的Unified Communications Manager部署，其中记录了与外部呼叫方的思科IP电话呼叫。这适用于入站和出站呼叫，只要内部电话配置了适当的录制配置文件。

从信令角度建立连接后，媒体会直接从分机电话流到录制服务器。

如果呼叫从此电话转移，则录制会话将结束。仅当接听呼叫的电话配置为录制时，才会捕获呼叫的下一段。

如本图所示，Unified Communications Manager部署 — 内部到内部。



这显示了基本的Unified Communications Manager部署，其中呼叫在企业内的内部用户之间。必须将其中一部电话配置为录制。如果两部电话都配置为录制，则将捕获两个单独的录制会话。

故障排除

本部分提供了可用于对配置进行故障排除的信息。

步骤1: MediaSenseCUCM

CUCM

- 应用用户(AXL)中的受控设备和权限信息。
- 记录配置文件和目标地址
- 指向MediaSense的SIP中继。
- 路由模式

MediaSense

在系统安装后，可以在MediaSense命令行上使用show tech call_control_service命令来验证基本配置。

此命令显示有关在系统上运行的Cisco MediaSense呼叫控制服务的信息。

Cisco MediaSense呼叫控制服务应运行以便此命令成功执行。

输出中捕获的系统信息。

```
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输出中记录会话信息。

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输出中的SIP适配器信息。

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：要设置呼叫记录，请参阅

步骤2.检查电话是否是流媒体到MediaSense服务器。


流1将是外部呼叫方的呼叫。流2将包含有关对MediaSense服务器的分组呼叫的信息。对于已分支的呼叫，接收的数据包始终保持零。

如此图所示，近端媒体流传输到MediaSense。

		<h2>Streaming Statistics</h2> <p>Cisco Unified IP Phone CP-7962G (SEP1C17D341FD21)</p>	
Device Information		Remote Address	10.106.122.178/33050
Network Configuration		Local Address	0.0.0.0/0
Network Statistics		Start Time	16:53:54
Ethernet Information		Stream Status	Not Ready
Access		Host Name	SEP1C17D341FD21
Network		Sender Packets	3888
Device Logs		Sender Octets	668736
Console Logs		Sender Codec	G.722
Core Dumps		Sender Reports Sent	14
Status Messages		Sender Report Time Sent	16:55:07
Debug Display		Rcvr Lost Packets	0
Streaming Statistics		Avg Jitter	0
Stream 1		Rcvr Codec	None
Stream 2		Rcvr Reports Sent	0
Stream 3		Rcvr Report Time Sent	00:00:00
Stream 4		Rcvr Packets	0
Stream 5		Rcvr Octets	0

远端媒体流到MediaSense

如此图所示，流1中接收的远端媒体流信息在流3中分类。

		<h2>Streaming Statistics</h2> <p>Cisco Unified IP Phone CP-7962G (SEP1C17D341FD21)</p>	
Device Information		Remote Address	10.106.122.178/57120
Network Configuration		Local Address	0.0.0.0/0
Network Statistics		Start Time	16:53:54
Ethernet Information		Stream Status	Not Ready
Access		Host Name	SEP1C17D341FD21
Network		Sender Packets	5874
Device Logs		Sender Octets	1010328
Console Logs		Sender Codec	G.722
Core Dumps		Sender Reports Sent	21
Status Messages		Sender Report Time Sent	16:55:50
Debug Display		Rcvr Lost Packets	0
Streaming Statistics		Avg Jitter	0
Stream 1		Rcvr Codec	None
Stream 2		Rcvr Reports Sent	0
Stream 3		Rcvr Report Time Sent	00:00:00
Stream 4		Rcvr Packets	0
Stream 5		Rcvr Octets	0

您可以在电话上执行数据包捕获来验证它。

如图所示，电话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=0x9471FB80,
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=0x9471FB80,
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=0x9471FB80,
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=0x9471FB80,
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=0x9471FB80,
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=0x9471FB80,
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=0x9471FB80,

提示：请参阅从[IP电话收集](#)数据包捕获

步骤3.验证CUCM和MediaSense上的呼叫信令。

此处的示例包含从分机为4011的SIP电话到分机为4009的SCCP电话的IP呼叫。录制目的号码是7878。

CUCM日志分析

从SIP电话发送的INVITE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Cisco 8945 IP电话，向CUCM发送As。

当SCCP电话应答呼叫并建立会话时，CUCM将ACK发送到SIP电话。

```
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-remotecallinfo>; 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

电话按“录音”软键，指示用户调用录制功能。

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

softKeyEvent=74 (Record)

lineInstance=1 callReference=32833194.

编解码器被锁定进行录制。

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

codecType=6

已分配内置网桥(BiB)资源。

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资源中拨号。

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

b00123906001

|FullyQualifiedCalledPartyNumber=

b00123906001

然后BiB拨打至MediaSense录制号码7878。

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

b00123906001

|DialingPartition=
|DialingPattern=

7878

|FullyQualifiedCalledPartyNumber=

7878

INVITE将发送到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

建立录制呼叫时，从MediaSense返回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

ACK到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

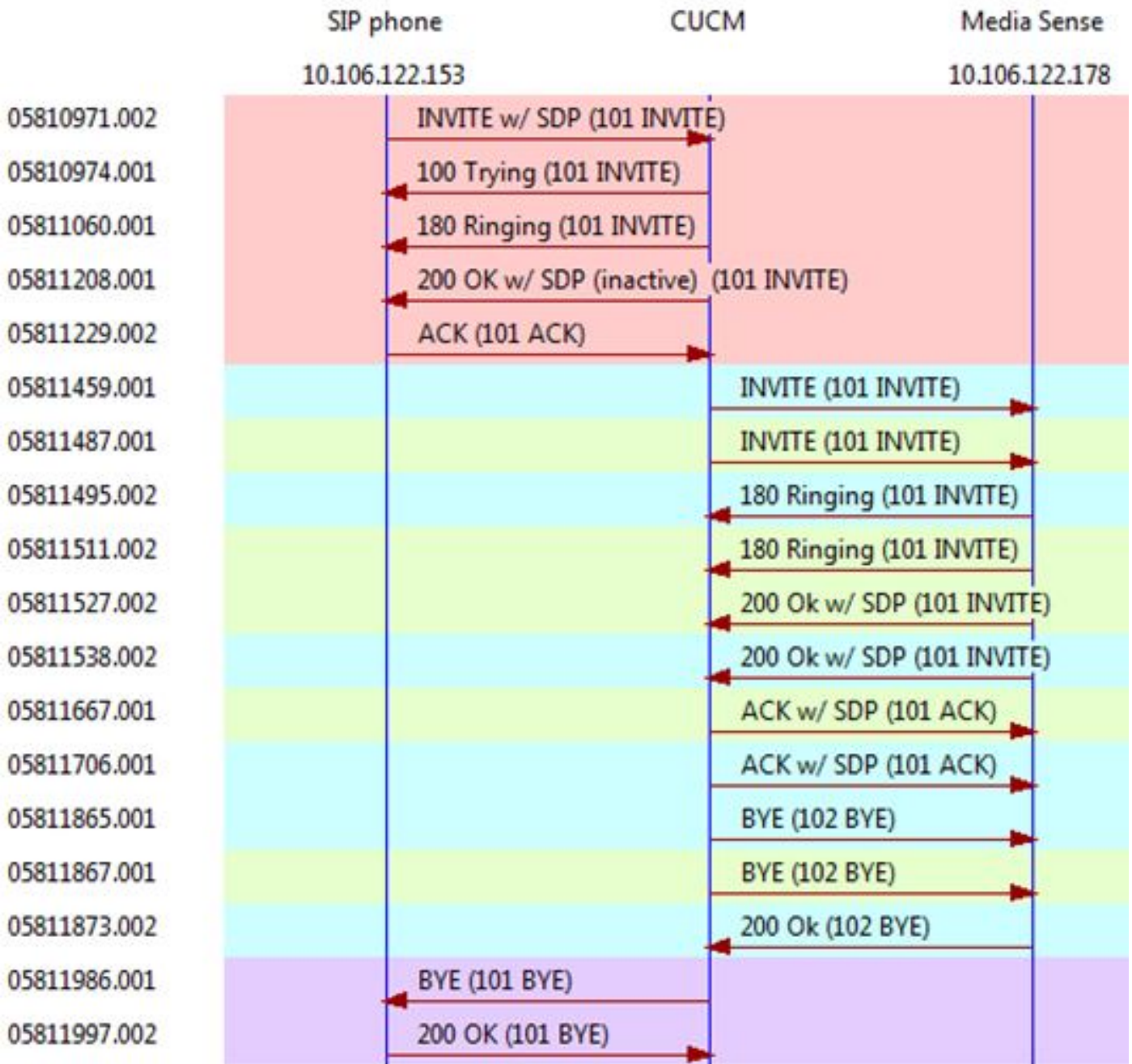
RTP/AVP 9 101
a=ptime:20
a=rtpmap:9 G722/8000
a=

sendonly

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

对远端流重复相同的过程。在BiB中，CUCM拨号，BiB将拨打录制号码，并在CUCM和MediaSense之间建立SIP会话。

如图所示，信令图。



MediaSense日志分析

从CUCM邀请建立近端呼叫记录 (来自SIP IP电话的音频)

```
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

]

从CUCM邀请建立远端呼叫记录 (来自SCCP IP电话的音频)。

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, ucCiscoGuid=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上捕获近端和远端录制信息的SIP支路后，为呼叫创建的会话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，用于近端呼叫。

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
```

当Media Sense应答呼叫后，将捕获类似事件。请注意，发送的ACK包含端口4000并指示**sendonly**。

两个SIP对话建立后的会话信息。

```
{ "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
```

当呼叫断开时，电话停止录音。

```
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>
```

注意：在此区域中，您注意到录制属性中有一个大小。此示例显示**size="1"**，这意味着MediaSense确实从CUCM接收音频。如果您注意到**size="0"**，则表示MediaSense未从CUCM接收音频。

最后，会话关闭。

```
{"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
```

从MediaSense收集日志

步骤1.启用呼叫控制服务跟踪级别以在MediaSense适用性中调试。

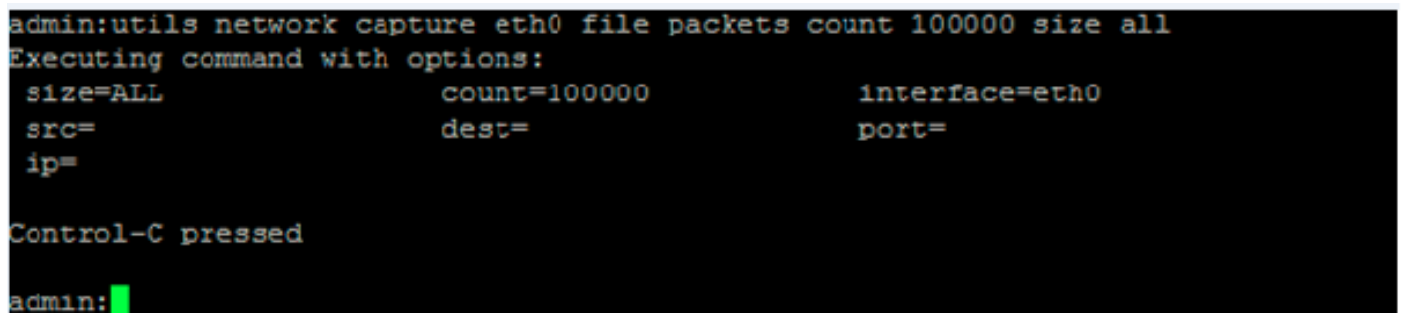
如本图所示，MediaSense服务。



步骤2.在MediaSense上启用数据包捕获。

请运行utils network capture eth0 file packets count 100000 size all以在MediaSense上启用数据包捕获。

如本图所示，在MediaSense上捕获数据包。

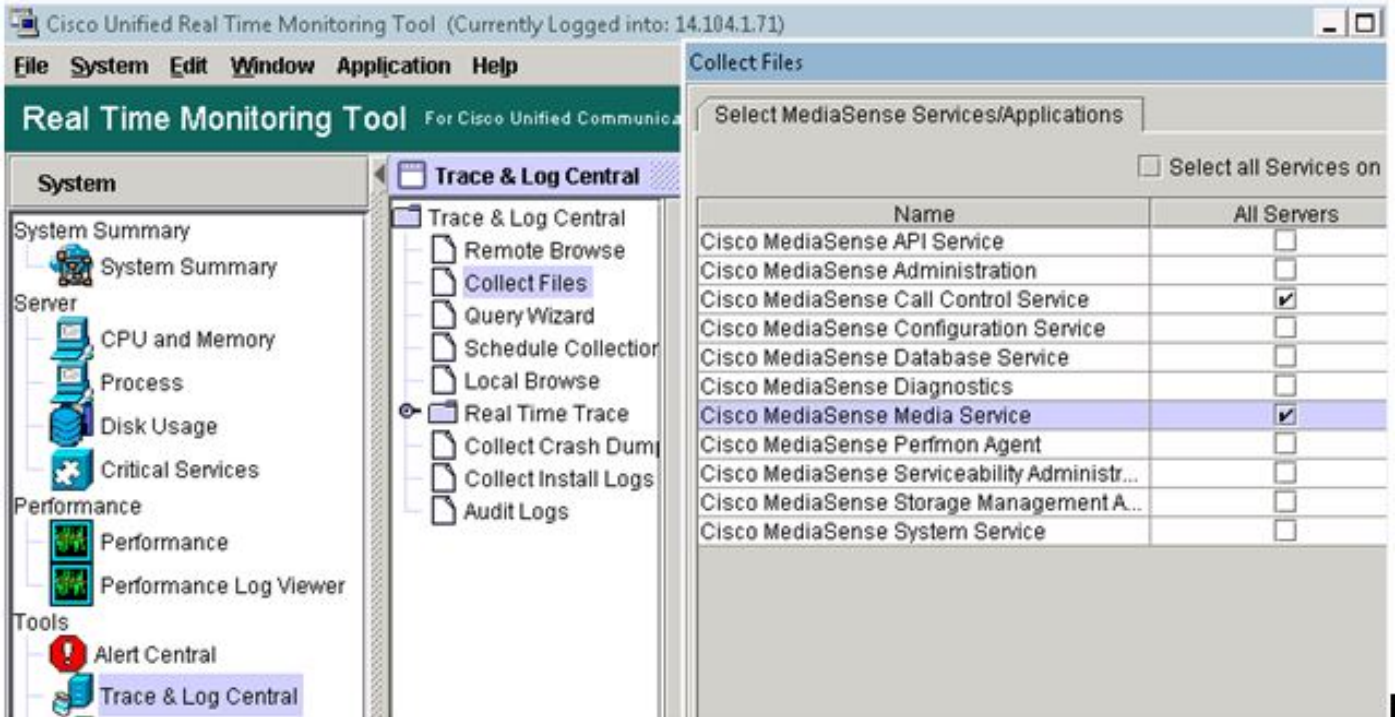


步骤3.使用实时监控工具(RTMT)收集日志

使用RTMT连接到MediaSense服务器。

导航至跟踪和日志中心>收集文件

如图所示，实时监控工具。



单击“下一步”并选择数据包捕获

如图所示，实时监控工具。

VIF Logs	<input type="checkbox"/>	<input type="checkbox"/>
Netdump Logs	<input type="checkbox"/>	<input type="checkbox"/>
Packet Capture Logs	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
Prog Logs	<input type="checkbox"/>	<input type="checkbox"/>
SAR Logs	<input type="checkbox"/>	<input type="checkbox"/>
SELinux Logs	<input type="checkbox"/>	<input type="checkbox"/>

相应地选择时间。

一些有用的命令：

1. utils media recording_sessions

utils media recording_sessions file fileName命令生成一个html文件，其中包含此Cisco MediaSense服务器处理的最后100个录制会话的详细列表。在执行此命令之前，请确认Cisco MediaSense呼叫控制服务正在运行。文件将保存到platform/cli/文件夹，并可使用file get activelog platform/cli/fileName命令下载。

命令：`utils media recording_sessions文件fileName`

详细信息：

- **file**是将信息输出到文件的必需参数。
- **fileName**是定义.html文件名称的必需参数。
- 发出此命令时，您会得到以下响应：Cisco MediaSense呼叫控制服务录制会话保存到平台/platform/cli/<filename>.html。现在，您可以使用：`file get activelog platform/cli/<filename>.html`然后，您可以从该目录检索文件并将其保存到您选择的位置。

示例：

- **utils media recording_sessions**文件sessions.html Cisco MediaSense。呼叫控制服务录制会话已保存到platform/cli/sessions.html。现在，您可以使用：文件获取活动日志
platform/cli/sessions.html

2. utis系统维护

命令**使用系统维护**操作启用或禁用Cisco MediaSense上的维护模式，或显示Cisco MediaSense维护模式状态。当处于维护模式时，Cisco MediaSense无法处理任何录制请求或API请求。

当Cisco MediaSense进入维护模式时，它将重新启动。任何流活动都会突然结束。任何活动录制都以CLOSED_ERROR状态结束。当维护模式被禁用时，Cisco MediaSense将再次重新启动，并重新进入正常模式。

命令：**utils系统维护操作**

详细信息：operation指定命令的作用。

有效操作包括：

- 启用
- 禁用
- 状态

示例：

- utis系统维护启用
- utis system maintenance disable
- utils系统维护状态

一些基本问题

[MediaSense文档维基](#)

已知缺陷

[CSCup24364](#) :C所有录音都无法处理没有呼叫方id的呼叫获取错误消息。

[CSCui13760](#): MediaSense不支持从群集中删除节点。

[CSCtn45420](#): MediaSense呼叫记录在Camelot SIP终端上失败。

[CSCut09446](#): MediaSense UI不填充CUCM配置和API用户配置。

[CSCuo95309](#): MediaSense搜索和播放录制未从其他节点填充。

[CSCuq20108](#): 使用转义字符时，从头到被截断。