排除故障分叉从Cisco IP电话的梅迪亚到梅迪亚感觉

目录

简介

本文描述排除故障分叉从Cisco IP电话的媒体的步骤记录在MediaSense服务器的呼叫。

先决条件

要求

Cisco 建议您了解以下主题：

- Cisco Unified Communications Manager (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Infromation

思科MediaSense是为在网络的设备提供记录功能的语音和视频媒介使用会话初始化协议(SIP)的一个基于网络平台。充分地集成到思科的统一通信体系结构，MediaSense自动地捕获并且存储在是适当地配置的CUCM的设备上的每次VoIP会话。

1. MediaSense接受在下面的格式的音频编码解码器：
   - g.711 μLaw和aLaw
   - g.722
   - g.729，g.729a，g.729b
   - 预先的音频编码-低迪莱(AAC-LD)亦称MPEG音频Layer4 -低开销MPEG-4音频传输多路复用(MP4A/LATM)
2. 在H.264编码的MediaSense视频

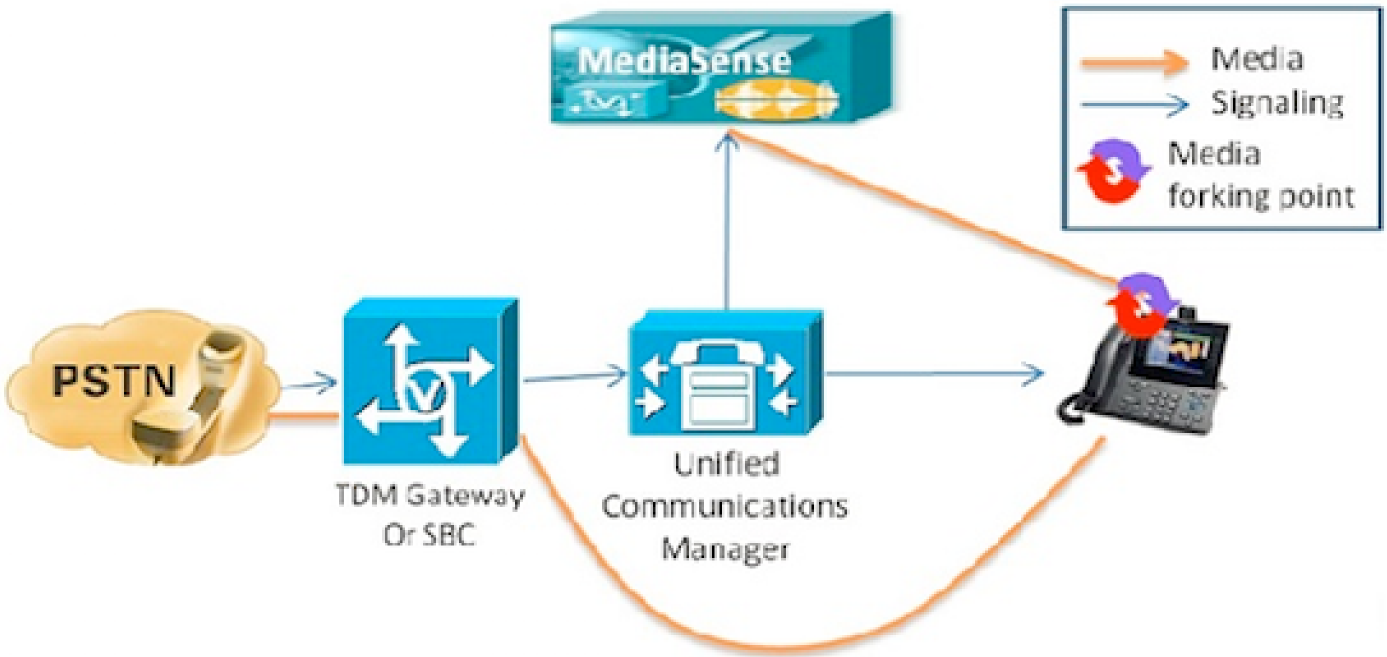
方案

1. 基本统一通信管理器部署-内部对外部
2. 基本统一通信管理器部署-内部到内部

从MediaSense的角度，实际上没有两个方案之间的差异。

在两种情况下，媒体由电话分叉了发送到分叉的数据流捕获的录音设备。因为有在他们的行为的一重大的差异在解决方案级别，他们区分得此处。

如此镜像所显示，统一通信内部对外部管理器的部署-。



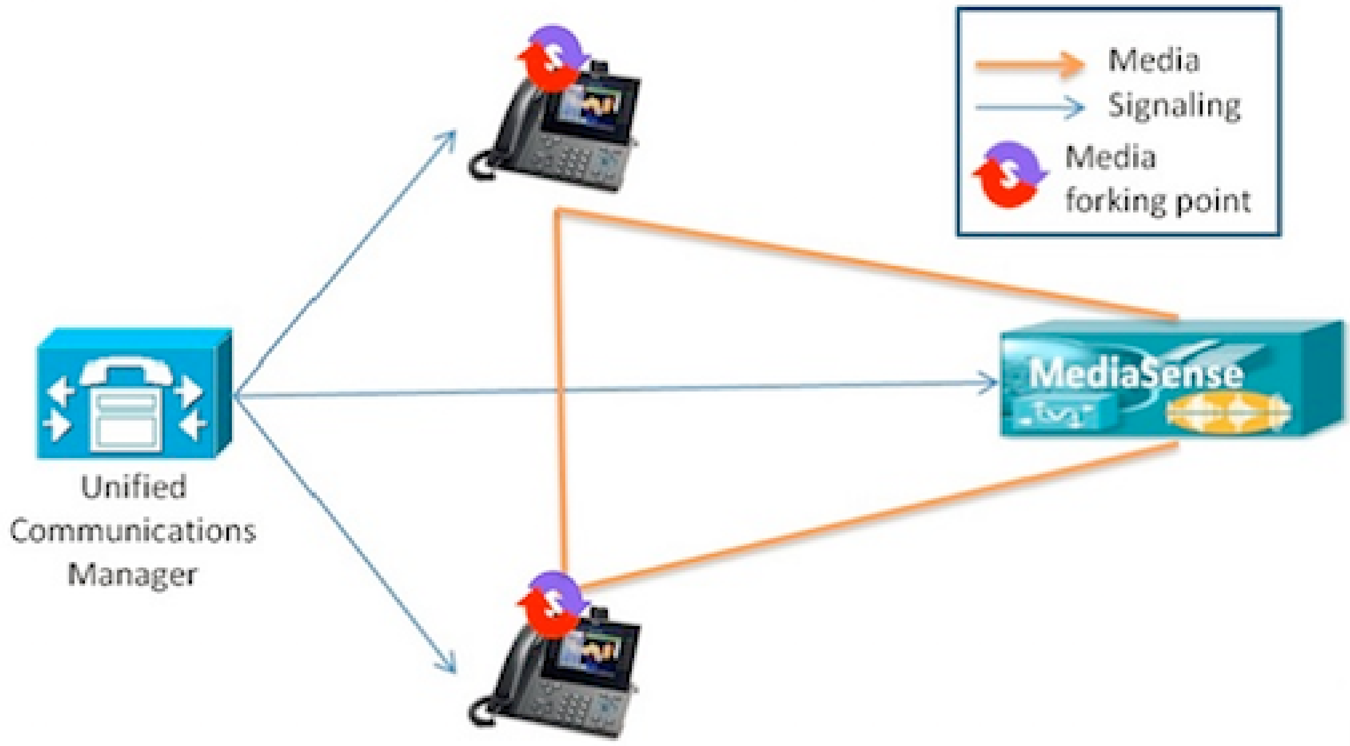
这显示一基本统一通信管理器部署与外部呼叫者的Cisco IP电话呼叫被记录的地方。只要里面电话配置与一适当的录音配置文件，这适用于两呼入和呼出电话。

一旦连接从信令方面被建立，媒体直接地从分叉的电话流到录音服务器。

如果呼叫转接远离此电话，录制会话结束。呼叫的下分段将捕获，只有当占去呼叫的电话为记录配

置。

如此镜像所显示，统一通信内部到内部管理器的部署-。



这显示一基本统一通信管理器部署呼叫在是在企业内的内部用户之间的地方。重要的是一个电话为记录配置。万一两个电话为记录配置，然后两独立的录音会话将捕获。

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

- 受控的设备和权限信息在应用程序用户(AXL)。
- 记录的配置文件和目的地址
- 指向MediaSense的SIP中继。
- 路由模式

MediaSense

您能验证基本配置使用**show tech call_control_service** on命令MediaSense line命令在系统安装以后。

此命令显示关于在系统运作的思科MediaSense呼叫控制服务的信息。

思科MediaSense呼叫控制服务应该运作为了此命令能成功执行。

在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
```

记录的会话信息在**show tech call_control_service**输出中。

```
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
```

SIP适配器信息在**show tech call_control_service**输出中。

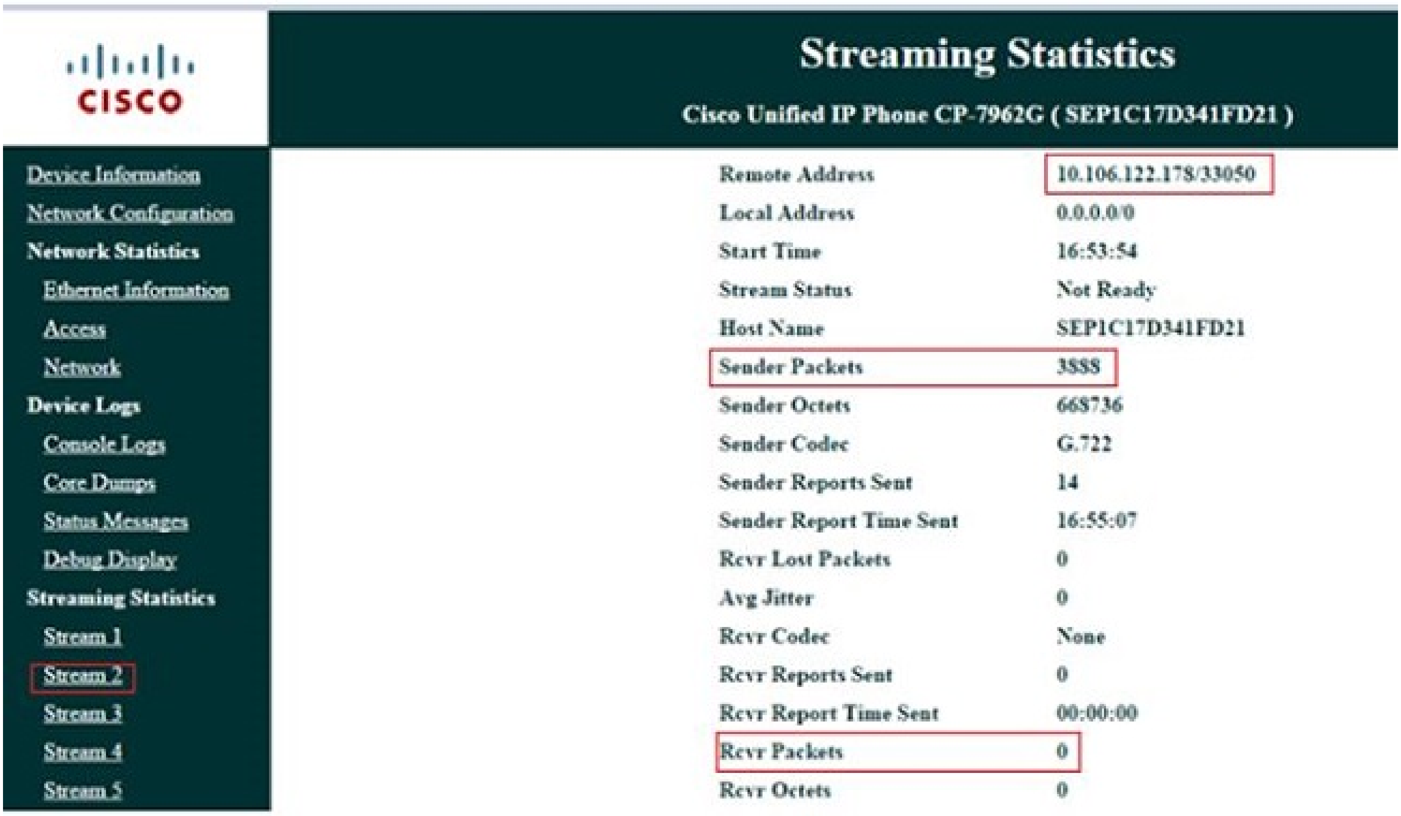
```
Sip Adapter:
LocalAddress=10.106.122.178:5060; RemoteAddresses [sip:10.106.122.174:5060
sip:10.106.122.175:5060 ], controlTransport=tcp
based on Cisco Caffeine SIP Stack, version=3.1.3.502, nonBlockingTCP=true,
closeConnectionOnTimeout=false
state=AcceptCalls, blockingMode=NONE
SdpUtil: m=audio %d RTP/AVP 102 0 8 9 18, m=video %d RTP/AVP 97
Executor: activeCount=0, poolSize=0, largestPoolSize=2, queueSize=0
```

提示：参考为了设置呼叫记录

第二步： 检查电话是否是流媒体到MediaSense服务器。

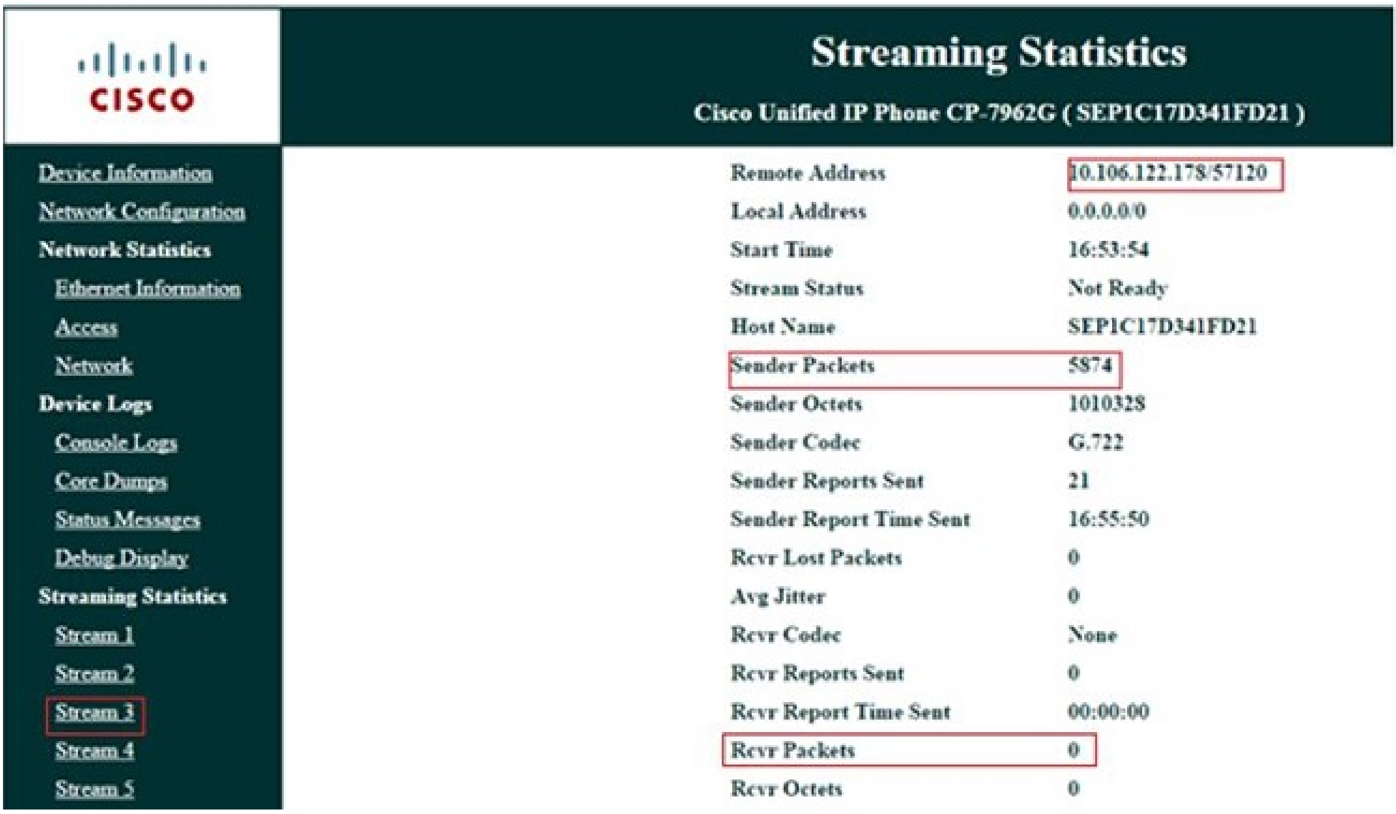
数据流1将是呼叫给外部呼叫者。数据流2将包含关于分叉的呼叫的信息到MediaSense服务器。接收方数据包永远将保持零为分叉的呼叫。

如此镜像所显示，对MediaSense的近端媒体流。



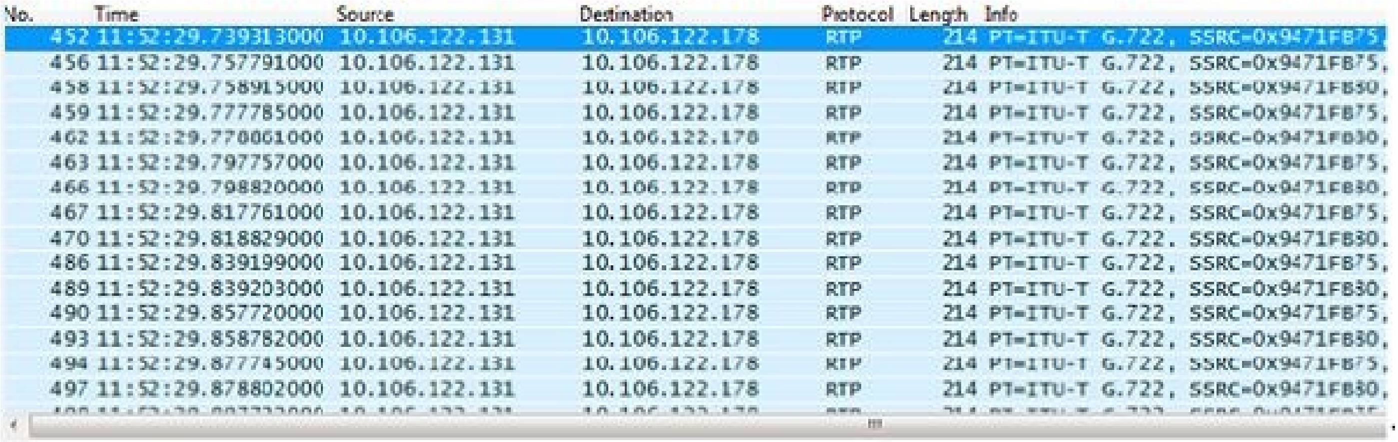
对MediaSense的远端的媒体流

如此镜像所显示，放出在数据流接收的远端的媒体的信息1在数据流3.分叉。



您能通过采取电话的数据包捕获验证它。

如此镜像所显示，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=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：参考收集数据包捕获从IP电话

步骤3.验证在CUCM和MediaSense的呼叫信令。

采取的示例此处包含从SIP电话的IP呼叫有分机的4011到有分机的4009 SCCP电话。记录的目标号码是7878。

CUCM日志分析

INVITE从SIP电话发送到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是思科8945 IP电话发送至于CUCM。

CUCM发送ACK到SIP电话，当SCCP电话应答呼叫时，并且会话被设立。

```
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
```

电话按表明的记录软键用户调用录音功能。

```
06053271.001 |08:39:52.681 |AppInfo  |StationInit: (0000045) SoftKeyEvent
softKeyEvent=74(Record) lineInstance=1 callReference=32833194.
```

为记录锁定的编码获得。

```
06053274.002 |08:39:52.681 |AppInfo  | StationCdpc: star_MediaExchangeAgenaQueryCapability -
Device SEP1C17D341FD21, codec locked due to recording,  codecType=6
```

内置的网桥(围嘴)资源得到分配。

```
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在围嘴资源拨号。

```
06053318.008 |08:39:52.683 |AppInfo  ||PretransformCallingPartyNumber=
|CallingPartyNumber=
|DialingPartition=
|DialingPattern= b00123906001
|FullyQualifiedCalledPartyNumber= b00123906001
```

围嘴然后拨号对MediaSense记录编号7878。

```
06053358.013 |08:39:52.686 |AppInfo  ||PretransformCallingPartyNumber=b00123906001
|CallingPartyNumber= b00123906001
|DialingPartition=
|DialingPattern= 7878
|FullyQualifiedCalledPartyNumber= 7878
```

INVITE发送对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
```

200从MediaSense的OK，当录音呼叫建立。

```
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-
```

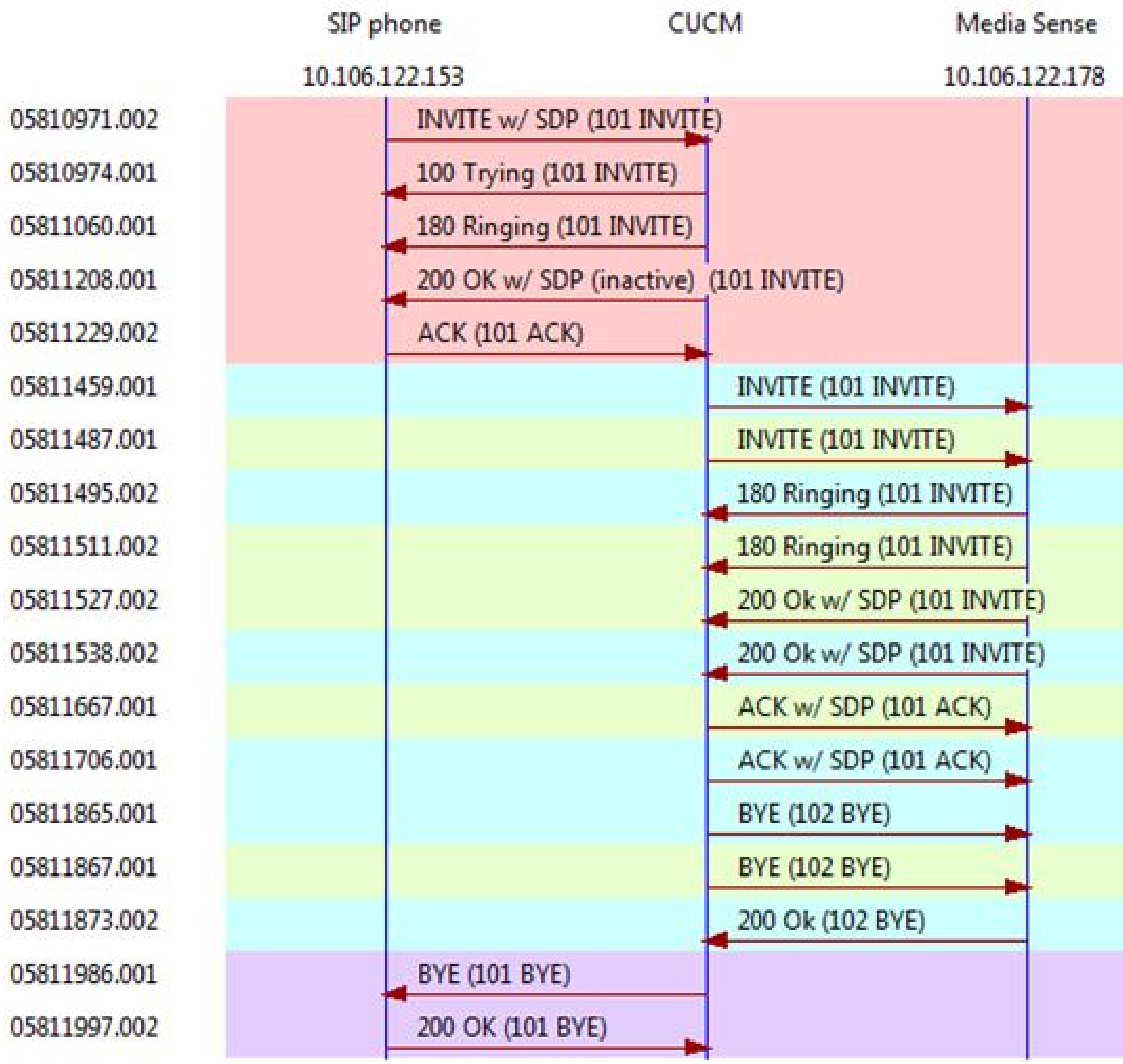
```
farendrefci=32833193;x-farendclusterid=StandAloneCluster;x-farenddevice=SEP203A0782D99F;x-
farendaddr=4011>;tag=16791~78868996-a8aa-4784-b765-86098b176d95-32833198
Call-ID: e4fb9980-5a71d048-b0-ae7a6a0a@10.106.122.174
CSeq: 101 INVITE
Content-Length: 313
Contact: <sip:7878@10.106.122.178:5060;transport=tcp>
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
```

对MediaSense的ACK。

```
 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
```

同样进程为远端的数据流被重复。CUCM在围嘴拨号，围嘴将拨号记录编号，并且SIP会话将建立在CUCM和MediaSense之间。

如此镜像所显示，信令图表。



MediaSense日志分析

从CUCM邀请设立近端的(从SIP IP电话的音频呼叫录音)

```
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邀请设立远端的(从SCCP IP电话的音频呼叫录音)。

```
 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
```

一旦近端和远端的SIP段记录信息在MediaSense，捕获会话ID为呼叫创建。

```
 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近端呼叫的好和ACK。

```
 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
```

一旦梅迪亚感觉应答呼叫，相似的事件将捕获。注意发送的ACK包含端口4000并且指示**sendonly**。

在设立的两个SIP对话以后的会话信息。


```
 {"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

当呼叫被断开时电话停止记录。

```
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，您注意有在录音属性的一个大小。此示例显示该**size="1"**，含义MediaSense接收从CUCM的音频。如果注意**size="0"**，含义MediaSense没有接收从CUCM的音频。

最终会话关闭。

{"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的日志集

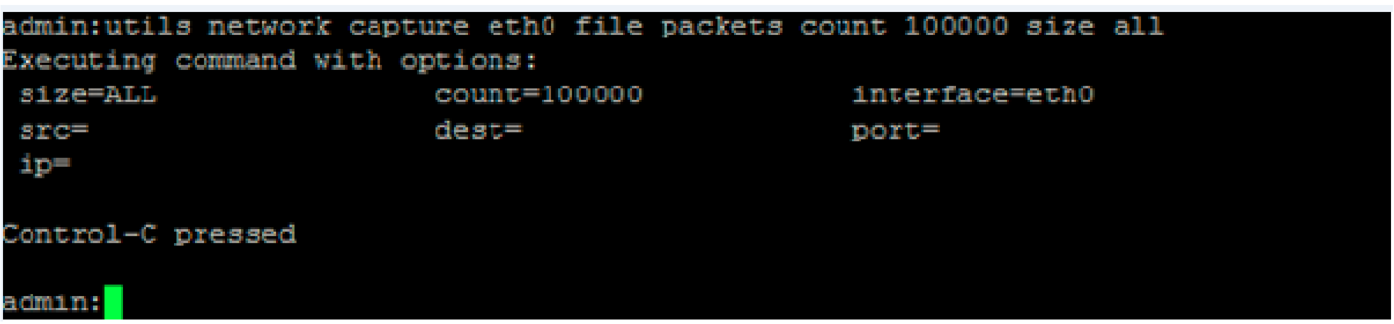
步骤1. Enable (event)呼叫控制调试的服务跟踪级别在MediaSense维护性。

如此镜像所显示， MediaSense Serviceabilty。

步骤2. MediaSense的Enable (event)数据包捕获。

请运行**使用情况网络捕捉eth0文件信息包计数100000大小全部**为了启用MediaSense的数据包捕获。

如此镜像所显示， MediaSense的数据包捕获。



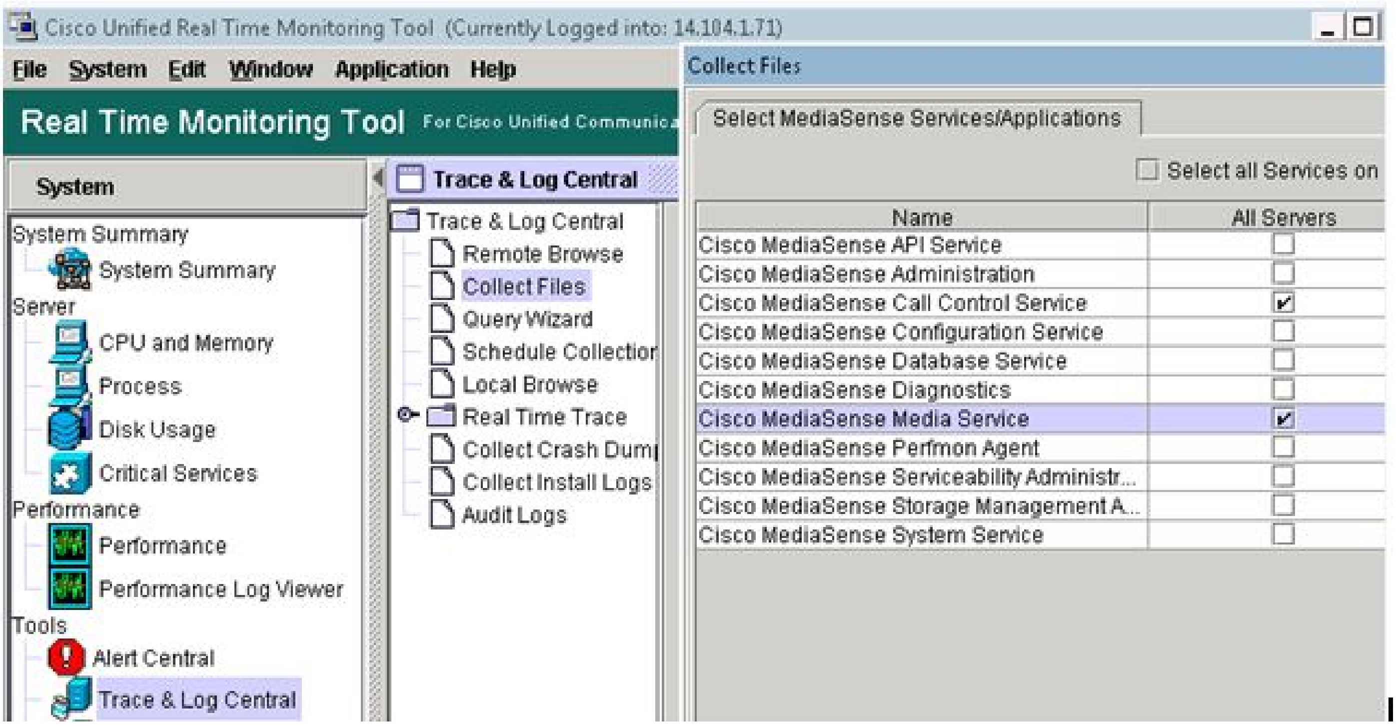
```
admin:utils network capture eth0 file packets count 100000 size all
Executing command with options:
 size=ALL                    count=100000               interface=eth0
 src=                        dest=                      port=
 ip=

Control-C pressed

admin:
```

步骤3.收集的日志使用实时监控工具(RTMT)

对MediaSense服务器的连接使用RTMT。

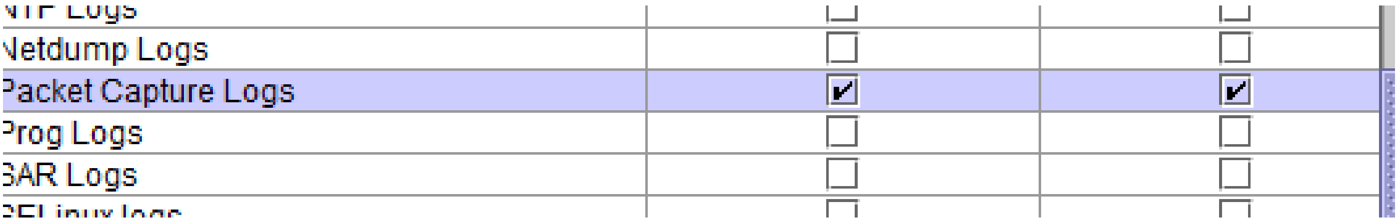
导航**跟踪&记录中央印制厂>收集的文件**

如此镜像所显示，实时监控工具。



其次单击并且选择**数据包捕获**

如此镜像所显示，实时监控工具。



相应挑选时间。

一些有用的命令：

1.**使用情况媒体recording_sessions**

file filename命令**使用情况媒体的recording_sessions**生成与此Cisco MediaSense服务器处理的最后100记录的会话的详细清单的一个HTML文件。确认思科MediaSense呼叫控制服务运作，在您执行此命令前。文件保存到平台/cli/文件夹，并且可以下载使用文件获得activelog平台/cli/filename命令。

指令：**使用情况媒体recording_sessions**文件文件名

详细信息：

- **文件**是输出信息到文件的必选参数。
- **文件名**是定义了.html文件的名称的必选参数。
- 当您发出此命令时，您得到以下答复：思科MediaSense呼叫控制服务纪录会话保存到平台/cli/<filename>.html。您能当前下载它使用：文件上activelog平台/您能从该目录然后检索文件和保存它到您的选择的位置的cli/<filename>.html。

示例：

- **使用情况媒体recording_sessions**文件sessions.html思科MediaSense。呼叫控制服务纪录会话保存对平台/cli/sessions.html。您能当前下载它使用：文件获得activelog平台/cli/sessions.html

2. **使用情况系统维护**

命令**使用情况系统维护**操作启动或禁用在思科MediaSense的维护模式或者显示思科MediaSense维护模式状态。当它在维护模式时，思科MediaSense不能处理任何记录请求或API请求。

思科MediaSense重新启动，当它输入维护模式。所有流活动突然结束。所有活动录音在CLOSED_ERROR状态结束。思科MediaSense再重新启动，当维护模式禁用时，并且重新输入正常模式。

指令：**使用情况系统维护**操作

详细信息：操作指定什么命令。

有效操作包括：

- enable (event)
- 禁用
- 状态

示例：

- 使用情况系统维护enable (event)
- 使用情况系统维护禁用
- 使用情况系统维护状态

一些基本问题

[MediaSense文档维基]

已知缺陷

CSCup24364 ：C所有记录不工作没有呼叫方ID的呼叫的收到错误消息。

CSCui13760 ： MediaSense不支持节点删除从集群。

CSCtn45420 ： MediaSense呼叫记录失效与Camelot SIP终端。

CSCut09446 ： MediaSense UI不填充CUCM配置& API用户设置。

CSCuo95309 ： MediaSense从其他节点没填充的搜索和作用录音。

CSCuq20108 ： 从报头到获得削，当曾经退出的字符时。