

# Contents

[Introduction](#)

[Prerequisites](#)

[Requirements](#)

[Components Used](#)

[Background Information](#)

[Scenario](#)

[Troubleshoot](#)

[Step 1. Check the configuration on MediaSense and CUCM.](#)

[Step 2. Check if the phone is streaming media to MediaSense server.](#)

[Step 3. Verify call signaling on CUCM and MediaSense.](#)

[CUCM Log Analysis](#)

[MediaSense Log Analysis](#)

[Log collection from MediaSense](#)

[Step 1. Enable call control service trace level to debug in MediaSense serviceability.](#)

[Step 2. Enable packet capture on MediaSense.](#)

[Step 3. Collect logs using Real Time Monitoring Tool \(RTMT\)](#)

## Introduction

This document describes troubleshoot steps for media forking from a Cisco IP phone to record calls on a MediaSense server.

## Prerequisites

### Requirements

Cisco recommends that you have knowledge of these topics:

- Cisco Unified Communications Manager (CUCM)
- Cisco MediaSense

### Components Used

The information in this document is based on these software and hardware versions:

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

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, make sure that you understand the potential impact of any command.

## Background Information

Cisco MediaSense is a network-based platform that provides voice and video media recording capabilities for devices in network using Session Initiation Protocol (SIP). Fully integrated into Cisco's Unified Communications architecture, MediaSense automatically captures and stores every Voice over IP (VoIP) conversation on devices which are appropriately configured CUCM.

1. MediaSense accepts audio codec in the below formats:

- g.711  $\mu$ Law and aLaw
- g.722
- g.729, g.729a, g.729b
- Advance Audio Coding - Low Delay (AAC-LD) also known as MPEG Audio Layer 4 - Low-overhead MPEG-4 Audio Transport Multiplex (MP4A/LATM)

2. MediaSense Video in H.264 encoding

## Scenario

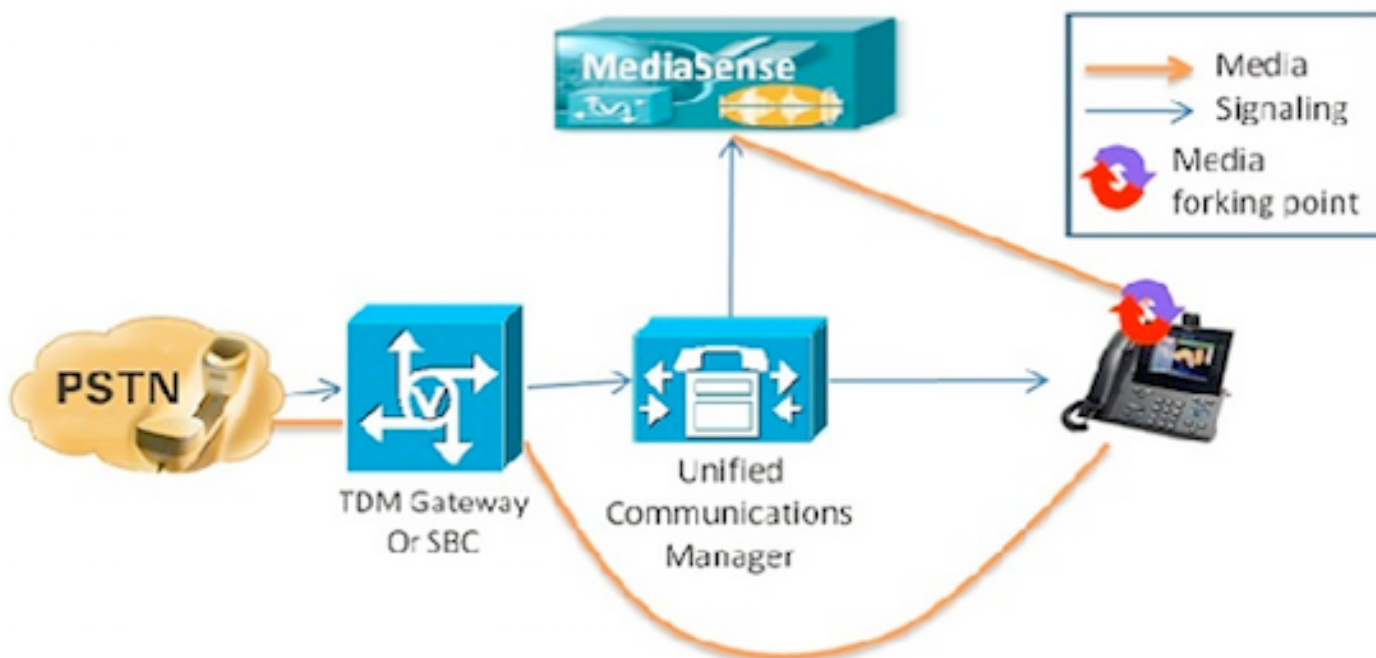
1. Basic Unified Communications Manager deployment - Internal-to-External

2. Basic Unified Communications Manager deployment - Internal-to-Internal

From the perspective of MediaSense, there is actually no difference between two scenarios.

In both cases, media forked by a phone is sent to the recording device where the forked streams are captured. They are distinguished here because there is a significant difference in their behavior at the solution level.

As shown in this image, Unified Communications Manager Deployment - Internal-to-External.

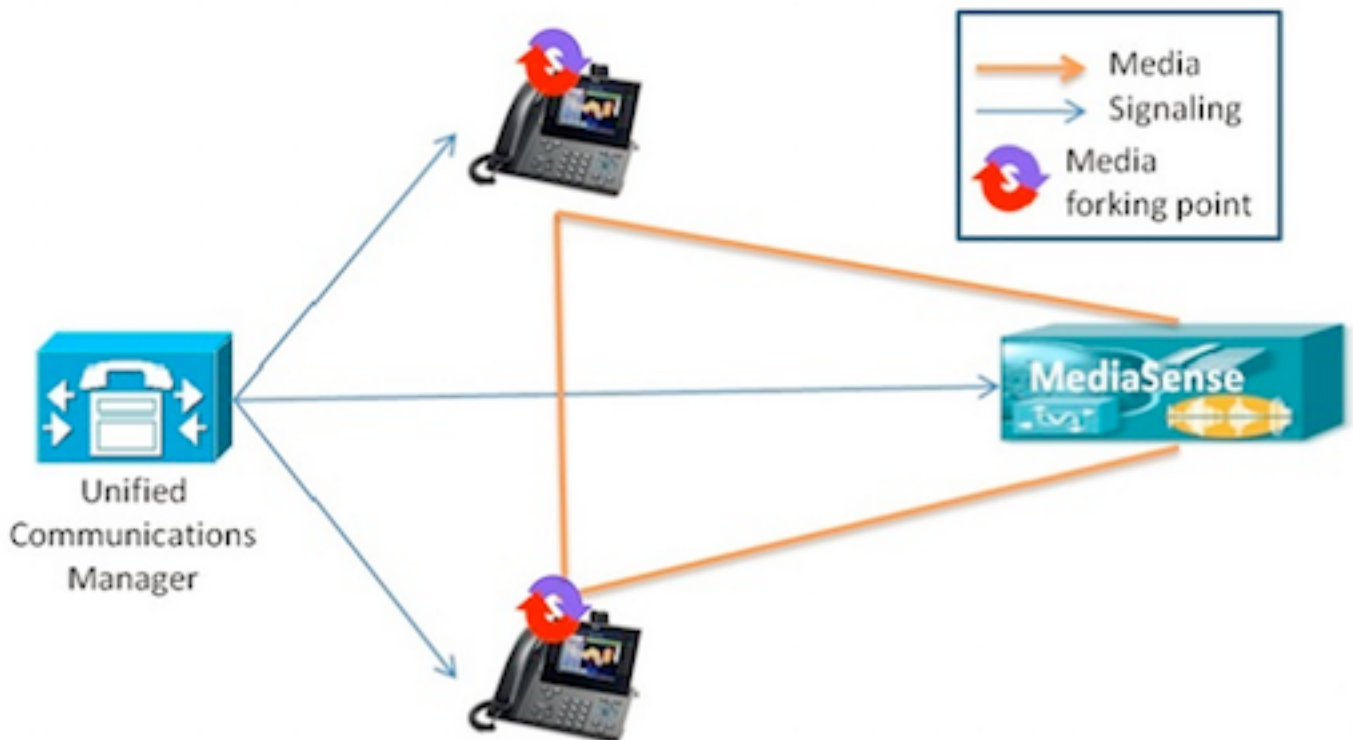


This shows a basic Unified Communications Manager deployment where Cisco IP phone call with an external caller is recorded. This applies to both inbound and outbound calls, as long as the inside phone is configured with an appropriate recording profile.

Once the connection is established from a signaling perspective, media flows directly from the forking phone to the recording server.

If the call is transferred away from this phone, the recording session ends. The next segment of the call will be captured only if the phone which takes up the call is configured for recording.

As shown in this image, Unified Communications Manager Deployment - Internal-to-Internal.



This shows a basic Unified Communications Manager deployment where the call is between internal users who are within the enterprise. It is important that one of the phones be configured for recording. In case both phones are configured for recording, then two separate recording sessions will be captured.

## Troubleshoot

This section provides information you can use in order to troubleshoot your configuration.

### Step 1. Check the configuration on MediaSense and CUCM.

CUCM

- Controlled devices and Permission information in Application user (AXL).
- Recording profile and destination address
- SIP trunk pointing to MediaSense.
- Route Pattern

### MediaSense

You can verify basic configuration using **show tech call\_control\_service** command on MediaSense command line after system installation.

This command displays information about the Cisco MediaSense Call Control Service that runs on the system.

The Cisco MediaSense Call Control Service should be running for this command to execute successfully.

## System Information captured in the output.

```
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
```

## Recording Sessions information in the **show tech call\_control\_service** output.

```
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 Adapter information in the **show tech call\_control\_service** output.

```
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
```

**Tip:** Refer to in order to setup Call Recording

## Step 2.

Stream 1 will be the call to the external caller. Stream 2 will contain the information about the forked call to the MediaSense server. Receiver packets will always remain zero for forked calls.

As shown in this image, Near end media streaming to MediaSense.

Streaming Statistics	
Cisco Unified IP Phone CP-7962G (SEP1C17D341FD21)	
Device Information	Remote Address: 10.106.122.178/33650
Network Configuration	Local Address: 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	Recv Lost Packets: 0
Streaming Statistics	Avg Jitter: 0
Stream 1	Recv Codec: None
Stream 2	Recv Reports Sent: 0
Stream 4	Recv Report Time Sent: 00:00:00
Stream 5	Recv Packets: 0
Stream 2	Recv Octets: 0

## Far end media streaming to MediaSense

As shown in this image, Streaming information for Far-end media received in Stream 1 is forked in Stream 3.

Cisco		Streaming Statistics	
		Cisco Unified IP Phone CP-7962G (SEP1C17D341FD21)	
Device Information	Remote Address	10.106.122.178/57120	
Network Configuration	Local Address	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	

You can verify it by taking Packet Capture on the phone.

As shown in this image, Phone 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	11:52:29.770061000	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,

Tip: Refer to [Collecting Packet Capture](#) from IP Phones

### Step 3.

The example taken here contains IP call from SIP phone with Extension 4011 to SCCP phone with Extension 4009. The recording destination number is 7878.

## INVITE sent from SIP phone to 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 is a Cisco 8945 IP Phone sending an As to CUCM.

## CUCM sends ACK to SIP phone when SCCP phone answers the call and the session gets established.

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-remotec: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

**Phone presses Record soft key indicating that the user invokes the recording feature.**

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

**Codec gets locked for recording.**

06053274.002 |08:39:52.681 |AppInfo | StationCdpc: star\_MediaExchangeAgenaQueryCapability -  
Device SEP1C17D341FD21, codec locked due to recording, **codecType=6**

**Built-in Bridge (BiB) resource gets allocated.**

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 dials in BiB resource.**

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

**BiB then dials to MediaSense recording number 7878.**

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

**INVITE is sent to 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 OK from MediaSense when the recording call is established.

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 to 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

```

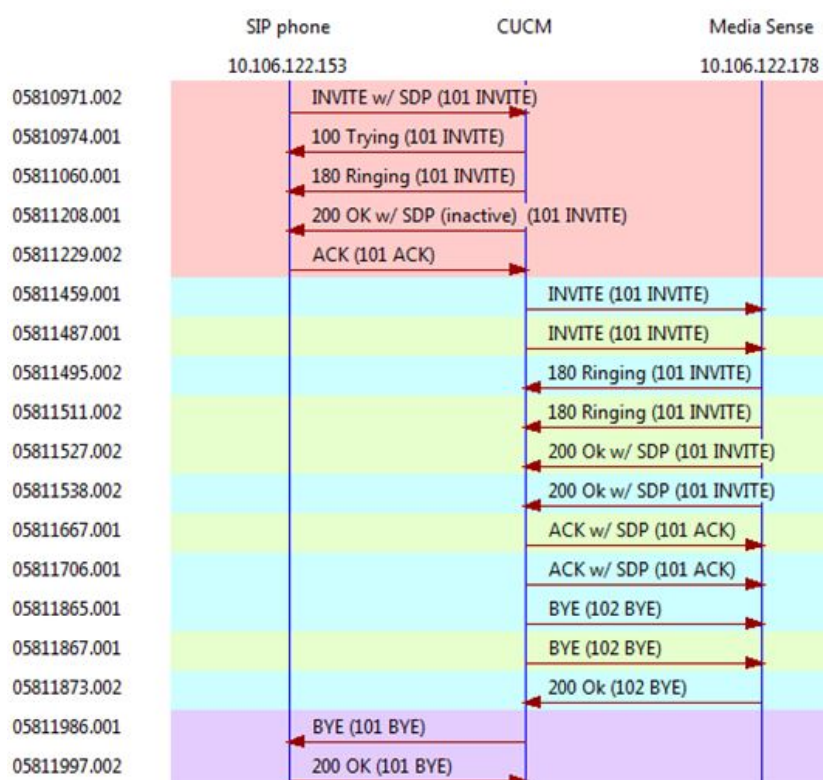
```

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

```

Same process is repeated for far end stream. CUCM dials in BiB, BiB will dial the recording number and an SIP session will be established between CUCM and MediaSense.

As shown in this image, the Signaling diagram.



# MediaSense Log Analysis

## INVITE from CUCM to establish call-recording for near end (Audio from SIP IP Phone)

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**]

## INVITE from CUCM to establish call-recording for far end (Audio from SCCP IP Phone).

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**

## **Session ID created for the call once both SIP leg for Near End and Far End recording information is captured on MediaSense.**

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: **d14e97859bfff1**, 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 and ACK for near end call.**

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
```

Similar event will be captured once the Media Sense answers the call. Note that ACK sent contains port 4000 and indicates **sendonly**.

### Session information after both SIP dialog established.

```
{"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

```

## When the call is disconnected phone stops recording.

```

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>

```

**Note:** In this area, you notice that there is a size in the recording attributes. This example shows that **size="1"**, which means MediaSense did receive the audio from CUCM. If you notice **size="0"**, it means MediaSense did not receive the audio from CUCM.

## Finally the session closes.

```

{"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

```

## Log collection from MediaSense

### Step 1.

As shown in this image, MediaSense Serviceability.



## Step 2.

Please run **utils network capture eth0 file packets count 100000 size all** in order to enable packet capture on MediaSense.

As shown in this image, Packet capture on MediaSense.

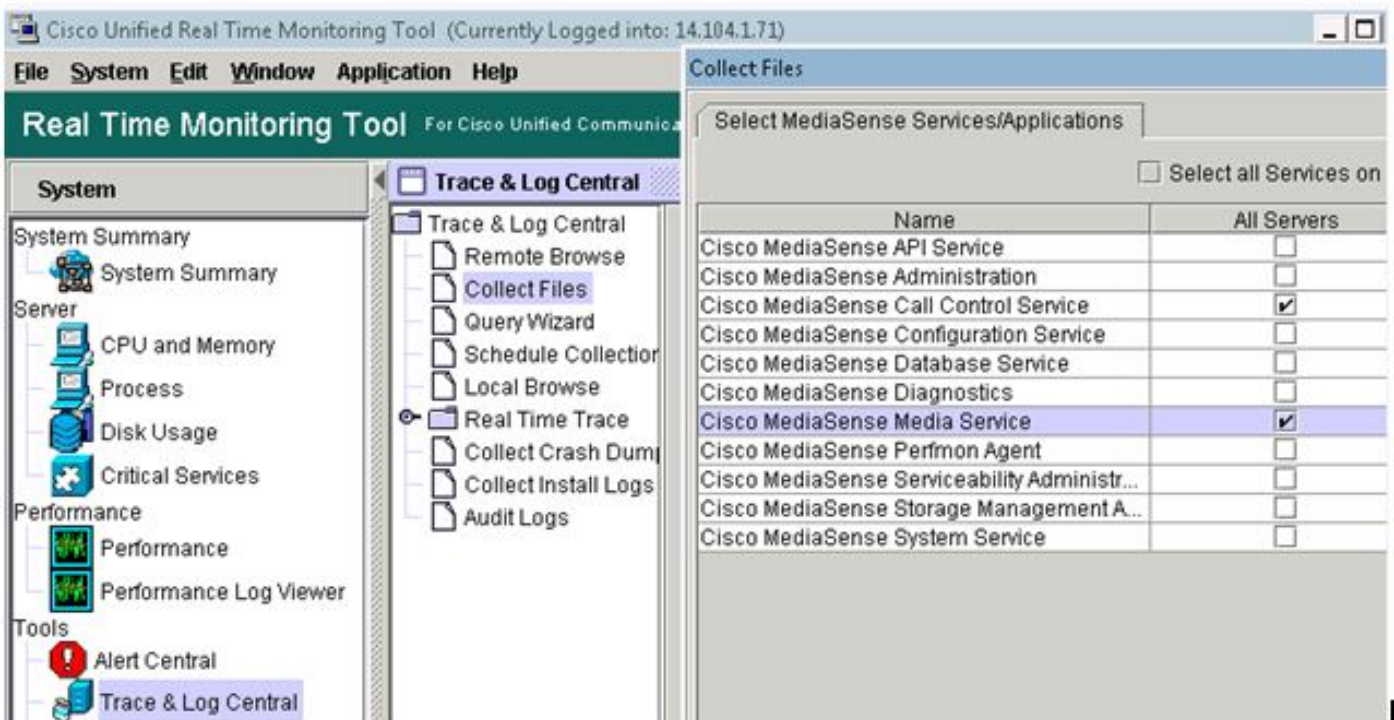
```
admin:utils network capture eth0 file packets count 100000 size all
Executing command with options:
  size=ALL          count=100000          interface=eth0
  src=              dest=              port=
  ip=
Control-C pressed
admin: █
```

## Step 3.

Connect to MediaSense server using RTMT.

Navigate to **Trace & Log Central > Collect Files**

As shown in this image, Real Time Monitoring Tool.



Click **Next** and select **packet capture**

As shown in this image, Real Time Monitoring Tool.

NTP 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"/>
SEI Unix logs	<input type="checkbox"/>	<input type="checkbox"/>

Select time accordingly.

## Some useful commands:

### 1. **utils media recording\_sessions**

The **utils media recording\_sessions** file fileName command generates an html file with a detailed list of the last 100 recording sessions processed by this Cisco MediaSense server. Confirm that the Cisco MediaSense Call Control Service is running before you execute this command. The file is saved to the platform/cli/ folder and can be downloaded using the file get activelog platform/cli/fileName command.

Command: **utils media recording\_sessions** file fileName

Details:

- **file** is a mandatory parameter that outputs the information to a file.
- **fileName** is a mandatory parameter that defines the name of the .html file.
- When you issue this command, you get the following response: Cisco MediaSense Call Control Service Recording sessions saved to platform/cli/<filename>.html. You can now download it using: file get activelog platform/cli/<filename>.html You can then retrieve the file from that directory and save it to a location of your choice.

Example:

- **utils media recording\_sessions** file sessions.html Cisco MediaSense. Call Control Service Recording sessions saved to platform/cli/sessions.html. You can now download it using: file get activelog platform/cli/sessions.html

### 2. **utils system maintenance**

The command **utils system maintenance** operation enables or disables maintenance mode on Cisco MediaSense , or displays the Cisco MediaSense maintenance mode status. While it is in maintenance mode, Cisco MediaSense cannot process any recording requests or API requests.

Cisco MediaSense reboots when it enters maintenance mode. Any streaming activities end abruptly. Any active recordings end in a CLOSED\_ERROR state. Cisco MediaSense reboots again when maintenance mode is disabled and it re-enters normal mode.

Command: **utils system maintenance** operation

Details: operation specifies what the command does.

Valid operations include:

- enable
- disable
- status

Examples:

- **utils system maintenance enable**
- **utils system maintenance disable**
- **utils system maintenance status**



## **Some basic issues**

[MediaSense Doc Wiki](#)

### **Known Defects**

[CSCup24364](#) : C all recording not working for calls with no caller id get error message.

[CSCui13760](#) : MediaSense does not support removal of node from cluster.

[CSCtn45420](#) : MediaSense call recording fails with Camelot SIP endpoint.

[CSCut09446](#) : MediaSense UI does not populate CUCM configuration & API user config.

[CSCuo95309](#) : MediaSense Search and Play Recordings not populated from other node.

[CSCuq20108](#) : From header to getting truncated when using escaped characters.