

# Configure and Troubleshoot Dual Screen Feature

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## Introduction

This document describes how to setup Dual Screen feature with Cisco Meeting Server (CMS) and Cisco Telepresence Endpoints.

## Prerequisites

### Requirements

Cisco recommends that you have a knowledge of these topics:

- Callbridge component must be configured on CMS
- CMS must be running version 2.2.3 or later
- CE endpoint must be running CE9.1.3 or later
- Cisco Unified Communications Manager (CUCM) must be running 11.5.1 or later
- Calls routed via Expressway must have Expressway running 8.9 or later
- Calls must be working with CMS

### Components Used

This document is not restricted to specific software and hardware versions:

- CMS API (Application program interface)
- Postman (or any other API client)
- CUCM
- CMS
- Cisco Telepresence Endpoints (SX, MX)
- PuTTY Secure Shell (SSH) terminal emulation software for Mainboard Management Processor (MMP)
- A web browser such as Firefox, Chrome

The information in the document 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, ensure that you understand the potential impact of any command.

## Configure

Step 1. Set up an administrator user account with **API** permission or can use the administrator account for Cisco Unified Communications Manager. Please find how to create a user with API access.

You can create additional user accounts for the MMP that have admin level rights using the MMP

Add user command **user add <account name> <role>**.

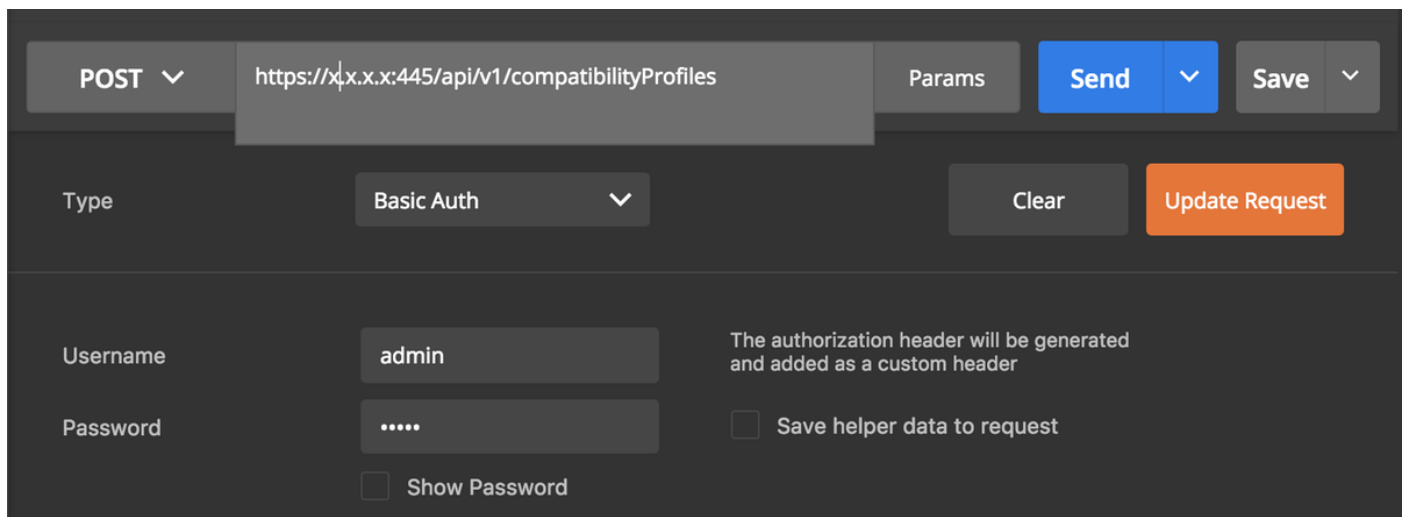
1. SSH into the MMP.
2. Add an admin level user account, for example

```
cb1> user add apiadmin admin
Please enter new password:
Please enter new password again:
Success
```

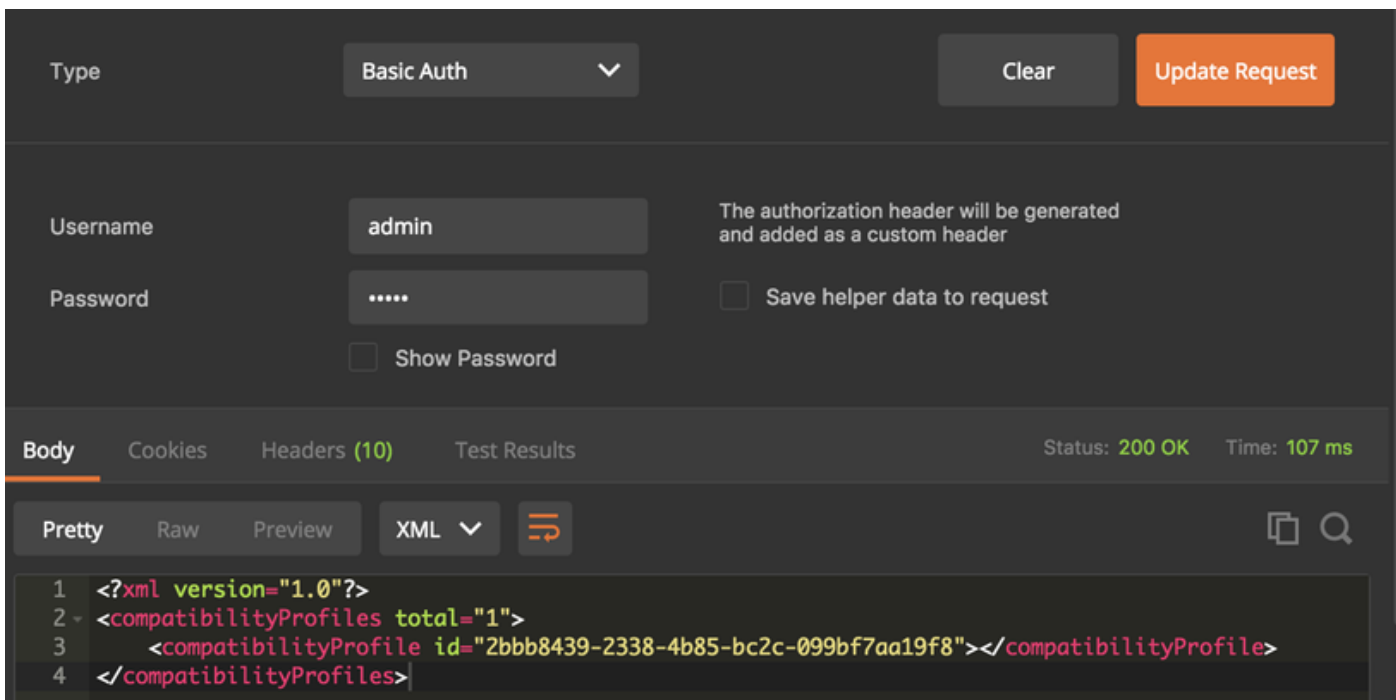
Step 2. Configure CMS to support **dual screen** feature using API.

POST, parameter **compatibilityProfiles**.

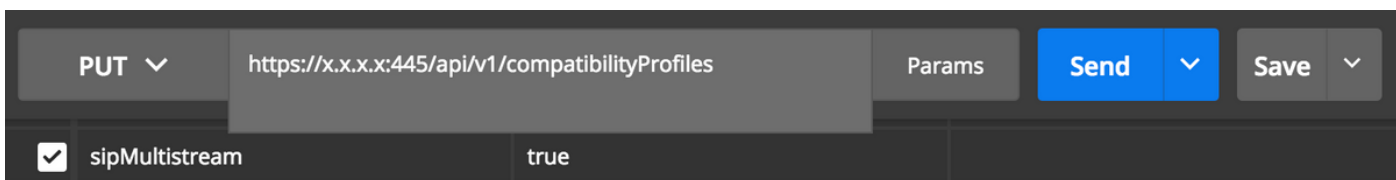
POSTMAN API is used but any API tool can be used for configuration.



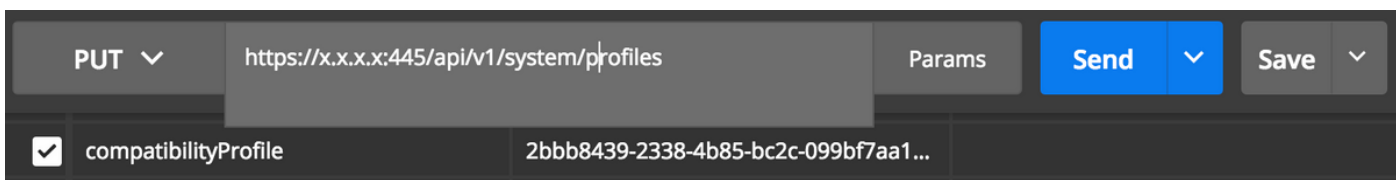
Step 3. Use **GET** operation to get the unique id for **compatibilityProfiles/<compatibilityProfiles id>**.



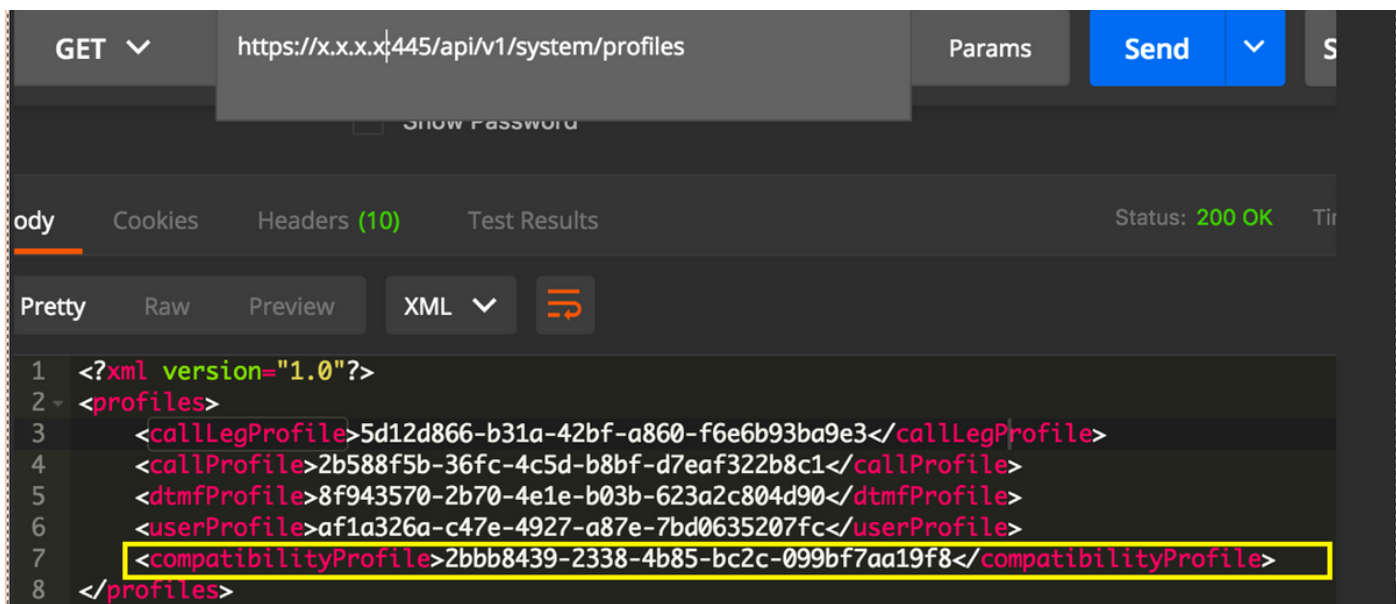
Step 4. Use **PUT** operation for **sipMultiStream=true**.



Step 5. Apply the configured **compatibilityprofile** under **system/profiles**. This is applied to the profile at top level and is used as global profile.

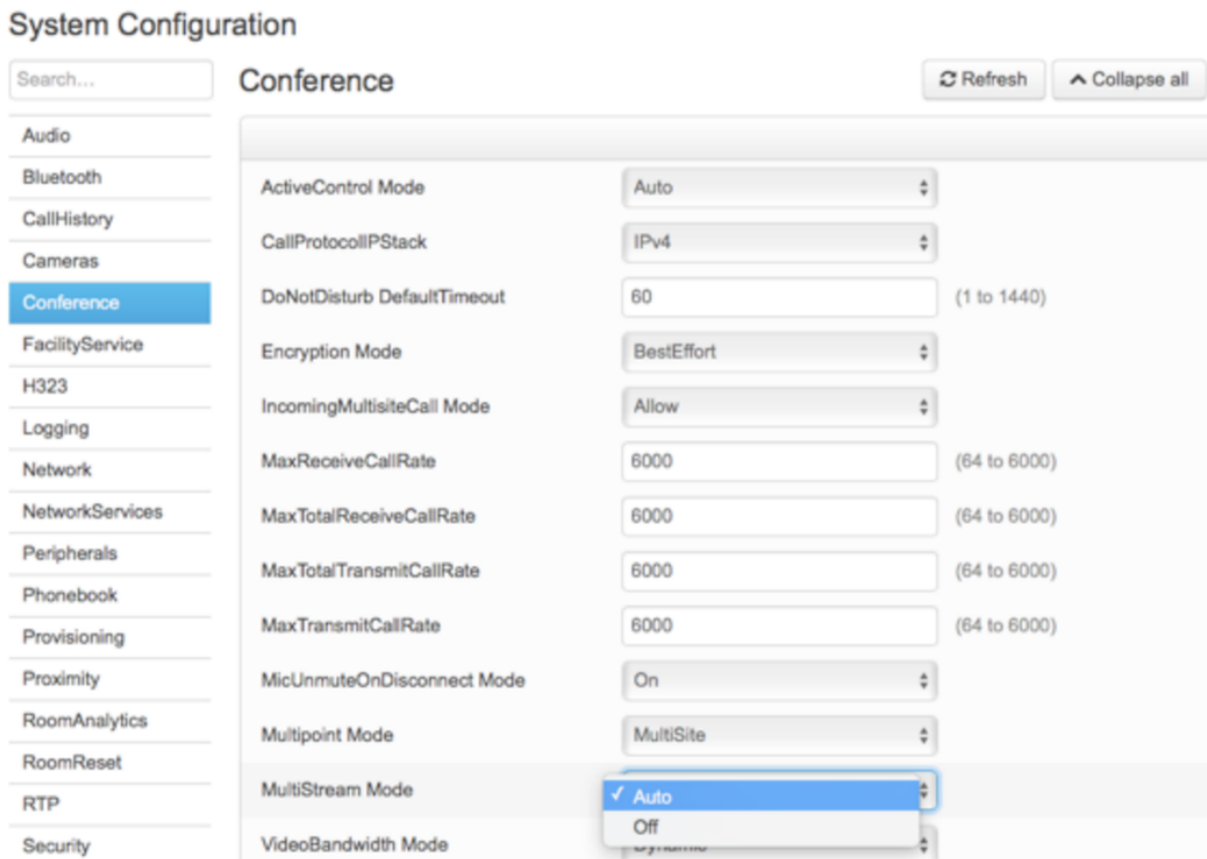


This image shows the **compatibilityprofile** being applied successfully.



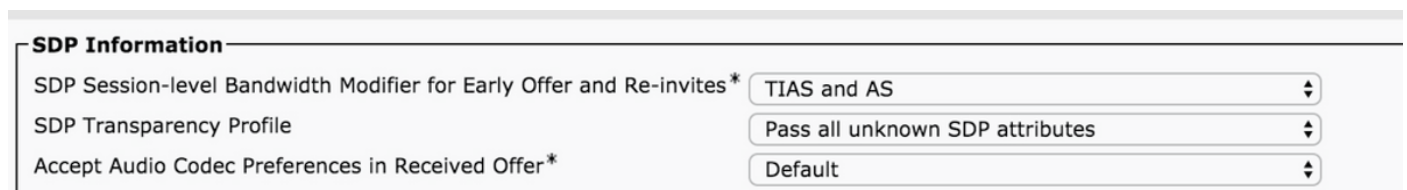
The above is the required configuration to setup **DualScreen** feature on CMS. Now, you need to also configure an endpoint with required configuration. The endpoint must run CE9.1.3 or later version of software code.

Step 6. MultiStream Mode on an endpoint must be set as **AUTO** as shown in this image.



Step 7. On Call Manager when configuring Sip trunk please keep in mind the sip profile used must have following parameters.

SDP Transparency Profile **Pass all unknown SDP attributes.**



IX, must be enabled on the **Sip Profile** for trunk.



On Call Manager for endpoints, the Sip Profile used should be either [Standard SIP Profile For](#)

[TelePresence Endpoint](#) or if you customize the sip profile to use for endpoints ensure that you have these parameters checked.

#### SDP Information

- Send send-receive SDP in mid-call INVITE
- Allow Presentation Sharing using BFCP
- Allow iX Application Media
- Allow multiple codecs in answer SDP

**Note:** When the system is in a three-screen setup, for example the Cisco Telepresence SX80, MX700 or MX800, the third screen is reserved for content while in a dual screen call.

## Verify

Use this section in order to confirm that your configuration works properly

Consider, SX, MX700/800 to work as a dual screen endpoint. The layout of participants would appear as shown in the image and if you have an additional third monitor connected, the presentation should appear on the third monitor.



Without multistream



With multistream

## Troubleshoot

Ensure to verify the software version for an endpoint, CUCM & CMS. Once confirmed the versions are supported then further troubleshooting is required

<https://www.cisco.com/c/dam/en/us/td/docs/telepresence/endpoint/software/ce9/release-notes/ce-software-release-notes-ce9.pdf>

[https://www.cisco.com/c/dam/en/us/td/docs/conferencing/ciscoMeetingServer/Release\\_Notes/Version-2-2/Cisco-Meeting-Server-Release-Notes-2-2-5.pdf](https://www.cisco.com/c/dam/en/us/td/docs/conferencing/ciscoMeetingServer/Release_Notes/Version-2-2/Cisco-Meeting-Server-Release-Notes-2-2-5.pdf)

Scenario 1. Dual screen feature does not work.

Collect Sip detailed traces on CMS

RTMT from Call Manager

Log bundle from the endpoint

Check logs for a scenario where dual screen feature was not working for the customer. Analyzing, to check the cause of the issue

Invite message sent by an endpoint

```
2017-08-24T11:25:31.709+08:00 SX80 appl[1660]: 3939.20 SipPacket I: SIP Msg: Outgoing => INVITE,
CSeq: 100 INVITE, Remote: 172.16.19.110:5060, CallId: 280004cfb801730726ecla9e9941d0d8
2017-08-24T11:25:31.709+08:00 SX80 appl[1660]: 3939.20 SipPacket INVITE
sip:8001@172.16.19.110 SIP/2.0
2017-08-24T11:25:31.709+08:00 SX80 appl[1660]: 3939.20 SipPacket Via: SIP/2.0/TCP
172.16.19.116:5060;branch=z9hG4bKe04a77c1ce5008a9c69d4c621c705bb6;rport
2017-08-24T11:25:31.709+08:00 SX80 appl[1660]: 3939.20 SipPacket Call-ID:
280004cfb801730726ecla9e9941d0d8
2017-08-24T11:25:31.710+08:00 SX80 appl[1660]: 3939.21 SipPacket CSeq: 100 INVITE
2017-08-24T11:25:31.710+08:00 SX80 appl[1660]: 3939.21 SipPacket Contact:
<sip:1000@172.16.19.116:55245;transport=tcp>;sip.cisco.multistream;x-cisco-multiple-screen=2

2017-08-24T11:25:31.726+08:00 SX80 appl[1660]: 3939.22 SipPacket a=rtcp-fb:* ccm cisco-scr
2017-08-24T11:25:31.726+08:00 SX80 appl[1660]: 3939.22 SipPacket a=sendrecv
2017-08-24T11:25:31.727+08:00 SX80 appl[1660]: 3939.22 SipPacket a=sprop-simul:1 1 *
2017-08-24T11:25:31.727+08:00 SX80 appl[1660]: 3939.22 SipPacket a=sprop-source:1
csi=3364746240
2017-08-24T11:25:31.727+08:00 SX80 appl[1660]: 3939.22 SipPacket m=video 2390 RTP/AVP 99 97
126 96 34 31 123
```

The invite looks good which suggested the required software and configuration is correct on and endpoint.

It was a delayed offer from CUCM to CMS. CMS sent 200 OK

```
Aug 24 11:25:29 user.info< http://user.info>; acano host:server: INFO : SIP trace: connection
23: outgoing SIP TCP data to 172.16.19.110:52560 from 172.16.19.123:5060, size 3830:
Aug 24 11:25:29 user.info< http://user.info>; acano host:server: INFO : SIP trace: SIP/2.0 200
OK
Aug 24 11:25:29 user.info< http://user.info>; acano host:server: INFO : SIP trace: Via:
SIP/2.0/TCP 172.16
Aug 24 11:25:29 user.info< http://user.info>; acano host:server: INFO : SIP trace: Max-
Forwards: 70
```

In SDP

```
Aug 24 11:25:29 user.info< http://user.info>; acano host:server: INFO : SIP trace: a=sendrecv
Aug 24 11:25:29 user.info< http://user.info>; acano host:server: INFO : SIP trace: a=sprop-
source:1 count=2;policies=cs:1
Aug 24 11:25:29 user.info< http://user.info>; acano host:server: INFO : SIP trace: a=sprop-
simul:1 1 *
Aug 24 11:25:29 user.info< http://user.info>; acano host:server: INFO : SIP trace: a=rtcp-fb:*
nack pli
```

```

Aug 24 11:25:29 user.info< http://user.info>; acano host:server: INFO : SIP trace: a=rtcp-fb:*
ccm fir
Aug 24 11:25:29 user.info< http://user.info>; acano host:server: INFO : SIP trace: a=rtcp-fb:*
ccm cisco-scr
Aug 24 11:25:29 user.info< http://user.info>; acano host:server: INFO : SIP trace: a=extmap:1
http://protocols.cisco.com/virtualid
Aug 24 11:25:29 user.info< http://user.info>; acano host:server: INFO : SIP trace: a=extmap:2
http://protocols.cisco.com/frameamarking
Aug 24 11:25:29 user.info< http://user.info>; acano host:server: INFO : SIP trace: a=rtpmap:97
H264/90000
Aug 24 11:25:29 user.info< http://user.info>; acano host:server: INFO : SIP trace: a=fmtp:97

```

The 200 OK from CMS has required attributes listed. The following attributes must be received by an endpoint to make the "dualscreen" feature to work effectively.

On endpoint when we checked the 200 Ok. We found the attributes were missing

```

2017-08-24T11:25:31.823+08:00 SX80 appl[1660]: 3939.32 SipPacket m=video 34794 RTP/AVP 97 116
96 34 31
2017-08-24T11:25:31.823+08:00 SX80 appl[1660]: 3939.32 SipPacket b=TIAS:1889000
2017-08-24T11:25:31.823+08:00 SX80 appl[1660]: 3939.32 SipPacket a=label:11
2017-08-24T11:25:31.823+08:00 SX80 appl[1660]: 3939.32 SipPacket a=rtpmap:97 H264/90000
2017-08-24T11:25:31.824+08:00 SX80 appl[1660]: 3939.32 SipPacket a=fmtp:97 profile-level-
id=428014;max-mbps=489600;max-fs=8160;max-dpb=4752;max-fps=6000
2017-08-24T11:25:31.824+08:00 SX80 appl[1660]: 3939.32 SipPacket a=rtpmap:116 H264/90000
2017-08-24T11:25:31.824+08:00 SX80 appl[1660]: 3939.32 SipPacket a=fmtp:116 profile-level-
id=428014;packetization-mode=1;max-mbps=489600;max-fs=8160;max-dpb=4752;max-fps=6000

```

To check further, check the Call Manager traces. Analysed the following attributes are unrecognized.

```

00267759.030 |13:55:03.641 |AppInfo |DET-SDPMsg- TCL_UNSPECIFIED (0)
00267759.031 |13:55:03.641 |AppInfo |DET-SDPMsg- Unrecognized attributes list: a=extmap:1
http://protocols.cisco.com/virtualid a=extmap:14 http://protocols.cisco.com/timestamp#100us
a=rtcp-fb:* ccm cisco-scr a=sprop-simul:1 1 * a=sprop-source:1 csi=51132416
00267759.032 |13:55:03.641 |AppInfo |DET-SDPMsg- mAudiomLines(i).bandwidth.enabledMask=TIAS,
TIAS=128000, AS=0, CT=0, RS=0, RR=0
00267759.033 |13:55:03.641 |AppInfo |DET-SDPMsg- nVideo=2
00267759.034 |13:55:03.641 |AppInfo |DET-SDPMsg- remoteIpAddress=172.16.19.116
remoteRtpPortNumber=2370 stackIdx=2 telephonyEvent=0 silenceSuppressionFlag=0 mSDPMode=0
idleFlag=0 vcId=1 mid=-1

```

Checked the Sip Profile to make sure the following parameters as mentioned above in the document are checked.

SDP Transparency Profile **Pass all unknown SDP attributes** . This parameter is setup on the configured sip profile. However, **Allow iX Application Media** is not checked.

Check the **Allow iX Application Media** and this fixed the issue.

Scenario 2. Dual screen feature not working.

In second scenario the problem is the same. However, the cause is different.

Endpoint sends the INVITE with required header and attributes in SDP. However; CUCM is unable to recognized the attributes in the SDP.

INVITE sip:95101@192.168.11.2<mailto:sip%3A95101@192.168.11.2> SIP/2.0  
Via: SIP/2.0/TCP 192.168.11.9:58911;branch=z9hg4bK64fdaf0987c59765f74b7f8f2673adfe;rport  
Call-ID: ca81ed904b80cf18528e5b0a4e4a4c01  
CSeq: 100 INVITE  
Contact: <sip:7436254f-c370-ccad-745d-110f8f59bee2@192.168.11.9<mailto:110f8f59bee2@192.168.11.9>:58911;transport=tcp>;**sip.cisco.multi-stream;x-cisco-multiple-screen=2**  
From: "Sala 5 Cota"  
<sip:571317@192.168.11.2<mailto:sip%3A571317@192.168.11.2>>;tag=0edc947e1b7a916a  
To: <sip:95101@192.168.11.2<mailto:sip%3A95101@192.168.11.2>>  
Max-Forwards: 70  
Route: <sip:192.168.11.2;lr>  
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,INFO,OPTIONS,REFER,NOTIFY  
User-Agent: TANDBERG/529 (ce9.1.4.3ae3106) Cisco-MX700ST  
Supported: replaces,100rel,timer,gruu,path,outbound,X-cisco-serviceuri,X-cisco-callinfo,X-cisco-service-control,X-cisco-sis-7.1.1,norefersub,extended-refer,sdp-anat  
Recv-Info: x-cisco-conference  
Session-Expires: 1800  
Allow-Events: dialog  
Remote-Party-ID: "Sala 5 Cota"  
<sip:571317@192.168.11.2<mailto:sip%3A571317@192.168.11.2>>;privacy=off;id-type=subscriber;screen=yes;party=calling  
Content-Type: application/sdp  
Content-Length: 4166

04323021.031 |21:05:59.460 |AppInfo |//SIP/SIPHandler/ccbId=0/scbId=0/getTrunInfoByRouteHdr:  
Route header userPart is missing  
04323021.032 |21:05:59.460 |AppInfo |//SIP/SIPHandler/ccbId=0/scbId=0/getRelIxxType: No  
matching SIP trunk found in hash table, returning relIxx disabled  
04323021.033 |21:05:59.460 |AppInfo  
|//SIP/SIPHandler/ccbId=4294967295/scbId=0/sipSPIGetCallExtensionSupported:  
SIPRelIxxEnabledServiceParamSetting=0 , ccb->pId.outboundRelIxx=1  
04323021.034 |21:05:59.460 |AppInfo |//SIP/SIPHandler/ccbId=0/scbId=0/sip\_stop\_timer:  
timerContext=0xdbc4a3c type=SIP\_TIMER\_EXPIRES value=1800000 retries=0  
04323021.035 |21:05:59.461 |AppInfo |//SIP/SIPHandler/ccbId=0/scbId=0/sip\_start\_timer:  
timerContext=0xdbc4a3c type=SIP\_TIMER\_EXPIRES value=1800000 retries=0  
04323021.036 |21:05:59.461 |AppInfo |//SIP/SIPHandler/ccbId=0/scbId=0/extractAssertedInfo:  
parseResult[1]  
**04323021.037 |21:05:59.475 |AppInfo |//SIP/SDPLib/Warning/0x0/sdp\_parse\_attr\_rtcpfb: rtcp-fb  
ccm has unrecognized param token: cisco-scr**  
**04323021.038 |21:05:59.475 |AppInfo |//SIP/SDPLib/Warning/0x0/sdp\_parse\_attr\_rtcpfb: rtcp-fb  
ccm has unrecognized param token: cisco-scr**  
**04323021.039 |21:05:59.475 |AppInfo |//SIP/SDPLib/Warning/0x0/sdp\_parse\_attr\_rtcpfb: rtcp-fb  
ccm has unrecognized param token: cisco-scr**  
**04323021.040 |21:05:59.476 |AppInfo |//SIP/SDPLib/Warning/0x0/sdp\_parse\_fmtp\_line\_params:  
Warning: Invalid maxbr specified for fmtp attribute.**  
**04323021.041 |21:05:59.476 |AppInfo |//SIP/SDPLib/Warning/0x0/sdp\_parse\_fmtp\_line\_params:  
Warning: Invalid maxbr specified for fmtp attribute.**  
**04323021.042 |21:05:59.476 |AppInfo |//SIP/SDPLib/Warning/0x0/sdp\_parse\_fmtp\_line\_params:  
Warning: Invalid maxbr specified for fmtp attribute.**  
**04323021.043 |21:05:59.476 |AppInfo |//SIP/SDPLib/Warning/0x0/sdp\_parse\_fmtp\_line\_params:  
Warning: Invalid maxbr specified for fmtp attribute.**  
**04323021.044 |21:05:59.476 |AppInfo |//SIP/SDPLib/Warning/0x0/sdp\_parse\_fmtp\_line\_params:  
Warning: Invalid maxbr specified for fmtp attribute.**  
04323021.045 |21:05:59.476 |AppInfo |//SIP/SDPLib/Warning/0x0/sdp\_parse\_fmtp\_line\_params:  
Warning: Invalid maxbr specified for fmtp attribute.  
04323021.046 |21:05:59.477 |AppInfo |//SIP/SIPHandler/ccbId=0/scbId=0/getMP4ALATMParameters:  
Saved payload(107) as Media\_Payload\_MP4ALATM\_128, clock=90000, profile=25,

CUCM is unable to recognized parameter like **cisco-scr** which is required for **DualScreen** feature to work. As the following endpoint is registered to Call Manager and there is no trunk in between. Checked the "SipProfile" configured for an endpoint analysed and found the setup using **Standard Sip Profile** rather it should use "Standard SIP Profile For Telepresence Endpoint"



Change to correct Sip Profile.