IOS和IOS-XE语音路由器中的RTP源验证

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简介

本文档介绍不同呼叫流和版本的Cisco IOS和IOS-XE语音路由器中RTP源验证功能的行为。

先决条件

要求

Cisco 建议您了解以下主题：

- IOS和IOS-XE软件
- H.323
- 会话初始协议 (SIP)
- Media Gateway Control Protocol (MGCP)
- 瘦呼叫控制协议(SCCP)
- 实时传输协议 (RTP)

使用的组件

本文档中的信息基于以下软件和硬件版本：

- ISRG2路由器(ISR2900、ISR3900)
- ISRG3路由器（ISR4400和ISR4300）
- ASR路由器（ASR1001-X、ASR1002-X、ASR1004、ASR1006和ASR1006-X，带RP2和ESP40）

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。如果您的网络处于活动状态，请确保您了解所有命令的潜在影响。

背景信息

了解VoIP网络和VoIP信令协议的基础知识非常重要，以便能够充分利用本文档。

RTP源验证定义和使用

RTP源验证是思科语音路由器中集成的一项功能，允许它们丢弃不受信任的入站RTP流量。

此功能的主要目标是提高设备的安全级别，并避免VoIP网络上的CrossTalk问题。

IOS语音路由器和IOS-XE语音路由器中有不同的功能和单一选项。

在IOS和IOS-XE中，此功能使语音路由器丢弃来自未知IP地址或端口的入站RTP流量，换句话说，语音路由器丢弃从未通过信令协商的IP地址或端口接收的数据包。

此功能在IOS和IOS-XE中的工作方式略有不同，因为路由器的架构以及它们被引入代码时；接下来的部分介绍这些场景。

IOS语音路由器中的RTP源验证

IOS具有两种不同的功能。

- **12.**4(6)T中引入的源滤波器
- **15.5(3)**M9、15.6(3)M6及后一版本引入的语音RTP源过滤器

  注意：请注意，下一部分介绍的场景是Cisco Unified Communications Manager(CUCM)暂候音乐(MoH)，但是，在满足要求时，同一行为会触发功能丢弃RTP的其他情况下。

源过滤器

此功能仅对SIP呼叫流可用。

配置后，如果呼叫流中使用的信令未协商RTP的IP地址和端口，则语音路由器会丢弃这些数据包。

源验证先检查**源IP地址，**然后检**查源端口**。

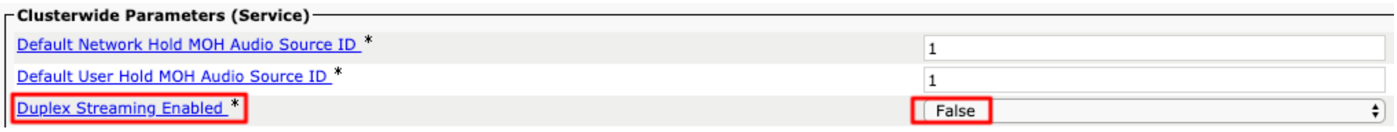
配置

```
voice service voip
 sip
  source filter
```

行为和检测

例如，CUCM将呼叫置于保持状态，并且默认情况下CUCM通过信令通告端口**4000**，但实际上从临

时端口(32768-61000)流传输RTP，因为默认情况下，**Clusterwide Parameters**下的Service Parameter **Duplex Streaming Enabled已禁用。**



Debug CCSIP Messages在语音路由器上显示**SIP ACK**消息，该消息通过会话描述协议(SDP)接收，告知路由器RTP来自**CUCM-IP-Address**和端口**4000。**

```
//-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
ACK sip:6002@Router-IP-Address:5060 SIP/2.0
Via: SIP/2.0/UDP CUCM-IP-Address:5060;branch=z9hG4bK4a424fed85
From: <sip:65002@CUCM-IP-Address>;tag=4091~842780d9-7186-4740-ada2-23e5d1b91316-46404063
To: <sip:6002@Router-IP-Address>;tag=2FF652-51D
Date: Thu, 18 Apr 2019 19:59:50 GMT
Call-ID: 3EDDD9E4-614B11E9-800D9C4B-C5465DB2@Router-IP-Address
User-Agent: Cisco-CUCM12.0
Max-Forwards: 70
CSeq: 102 ACK
Allow-Events: presence
Session-ID: 4978aa3900105000a000006cbcbcfda2;remote=836b14b48c77bfe681c0780c54ab4091
Content-Type: application/sdp
Content-Length: 191

v=0
o=CiscoSystemsCCM-SIP 4091 3 IN IP4 CUCM-IP-Address
s=SIP Call
c=IN IP4 CUCM-IP-Address (MoH Server)
t=0 0

m=audio 4000 RTP/AVP 0
a=X-cisco-media:umoh
a=ptime:20
a=rtpmap:0 PCMU/8000
a=sendonly
```

Show Call Active Voice Brief不显示**RX**增量，其中RTP应来自**CUCM-IP-Address和端口4000。RTP从不同端口接收并被语音路由器丢弃。**

```
11EC : 3 3143250ms.1 (14:59:02.516 CDT Thu Apr 18 2019) +1960 pid:0 Answer 6002 active
 dur 00:47:29 tx:2330/391440 rx:64875/10380000 dscp:0 media:0 audio tos:0x0 video tos:0x0
 Tele 0/0/0:23 (3) [0/0/0.23] tx:2803960/1263780/0ms g711ulaw noise:-65 acom:3  i/0:-60/-64 dBm

11EC : 4 3143250ms.2 (14:59:02.516 CDT Thu Apr 18 2019) +1950 pid:1 Originate 65002 connected
 dur 00:47:29 tx:1686/269760 rx:2330/372800 dscp:0 media:0 audio tos:0xB8 video tos:0x0
 IP CUCM-IP-Address:4000 SRTP: off rtt:1ms pl:46150/0ms lost:0/0/0 delay:55/55/65ms g711ulaw
TextRelay: off Transcoded: No ICE: Off
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a
 long duration call detected:n long duration call duration:n/a timestamp:n/a
 LostPacketRate:0.00 OutOfOrderRate:0.00
```

Show VoIP RTP Connections将**RmtRTP**显示为**4000**，**将RemoteIP** 显示为**CUCM-IP-Address**。

路由器期望RTP来自同一源。

```
show voip rtp connections
```

```
VoIP RTP Port Usage Information:
Max Ports Available: 8091, Ports Reserved: 101, Ports in Use: 1
                                        Min   Max   Ports     Ports     Ports
Media-Address Range                     Port  Port  Available Reserved  In-use
------------------------------------------------------------------------------
Global Media Pool                       16384 32766 8091      101       1
------------------------------------------------------------------------------
VoIP RTP active connections :
No. CallId     dstCallId  LocalRTP RmtRTP   LocalIP                                  RemoteIP
MPSS
1     4           3          16386    4000   Router-IP-Address                        CUCM-IP-
Address                                 NO
Found 1 active RTP connections
```

使用嗅探器捕获，可以验证RTP实际来自何处，在本例中，它来自端口**24588**而不是**4000**，因此源验证失败，语音路由器丢弃数据包。

Source Address	Source Port	Destination Address	Destination Port	SSRC	Payload	Packets	Lost	Max Delta (ms)	Max Jitter	Mean Jitter
Remote IP Address	24588	Router IP Address	16386	0x66c	g711U	514	0 (0.0%)	29.003	1.174	0.187

语音RTP源过滤器

此功能在15.5(3)M9、15.6(3)M6 IOS版本中引入。

它的工作方式与源过**滤器相**同，它首先验证**源IP地址**，然后验**证源端**口，但有两个主要区别。

1. **语音RTP源过滤器**适用于SIP、H.323、MGCP和SCCP
2. 该功能还在调试VoIP RTP错**误中添加了错**误消息，以便轻松检测何时由于源验证失败而丢弃RTP

> **警告：**默认情况下，此功能已启用且不显示在配置中。如果有设备从不同的源发送RTP，而不是通过信令通告的源发送RTP，则升级到支持此功能的任何IOS版本可能会导致音频问题。
> 当在命令前面加上“否”**时，**该功能被禁用，然后在配置中显示。

配置

```
Configuration Terminal
 voice rtp source-filter
```

每个协议的行为和检测

对于H.323:

语音路由器上的调试H225 Asn1显示收到的**openLogicalChannelAck**，它通知路由器远程媒体地址**0.0.0.0:0**。

```
H245 MSC OUTGOING PDU ::=

value MultimediaSystemControlMessage ::= response : openLogicalChannelAck :
    {
      forwardLogicalChannelNumber 1
      forwardMultiplexAckParameters h2250LogicalChannelAckParameters :
      {
        mediaChannel unicastAddress : iPAddress :
        {
```

```
          network 'Router-IP-Address'H
          tsapIdentifier 16404 (Router's UDP Port for the RTP)
        }
        mediaControlChannel unicastAddress : iPAddress :
        {
          network 'Router-IP-Address'H
          tsapIdentifier 16405 (Router's UDP Port for the RTCP)
        }
        flowControlToZero FALSE
      }
    }
```

```
Received openLogicalChannelAck has network and tsapIdentifier for the mediaChannel in zeros
which means IP Address 0.0.0.0 and port 0.
H245 MSC INCOMING PDU ::=

value MultimediaSystemControlMessage ::= response : openLogicalChannelAck :
    {
      forwardLogicalChannelNumber 2
      forwardMultiplexAckParameters h2250LogicalChannelAckParameters :
      {
        sessionID 1
        mediaChannel unicastAddress : iPAddress :
        {
          network '00000000'H
          tsapIdentifier 0
        }
        mediaControlChannel unicastAddress : iPAddress :
        {
          network '00000000'H
          tsapIdentifier 1
        }
      }
    }
```

Show Call Active Voice Brief不显示RX增量，并且远程IP地址和端口设置为0.0.0.0:0。

```
11F5 : 21 18903090ms.1 (16:00:48.794 CDT Fri Apr 19 2019) +1070 pid:2 Answer 6002 active
 dur 00:00:43 tx:376/63168 rx:899/137074 dscp:0 media:0 audio tos:0x0 video tos:0x0
 Tele 0/1/0:23 (21) [0/1/0.1] tx:35340/14230/0ms g711ulaw noise:-68 acom:3  i/0:-64/-63 dBm

11F5 : 22 18903090ms.2 (16:00:48.794 CDT Fri Apr 19 2019) +1070 pid:1 Originate 36004 active
 dur 00:00:43 tx:152/23047 rx:376/60160 dscp:0 media:0 audio tos:0xB8 video tos:0x0
 IP 0.0.0.0:0 SRTP: off rtt:0ms pl:0/0ms lost:0/0/0 delay:0/65/65ms g711ulaw TextRelay: off
Transcoded: No ICE: Off
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a
 long duration call detected:n long duration call duration:n/a timestamp:n/a
 LostPacketRate:0.00 OutOfOrderRate:0.00
 LocalUUID:
 RemoteUUID:
 VRF:
```

Show VoIP RTP Connections将RmtRTP和RemoteIP显示为0.0.0.0:0，因此路由器希望从该源获得RTP。

```
VoIP RTP Port Usage Information:
Max Ports Available: 8091, Ports Reserved: 101, Ports in Use: 1
Port range not configured
                                                Min    Max    Ports     Ports     Ports
```

```
Media-Address Range                        Port  Port  Available Reserved  In-use
------------------------------------------------------------------------------
Global Media Pool                          16384 32766 8091      101       1
------------------------------------------------------------------------------
VoIP RTP active connections :
No. CallId     dstCallId  LocalRTP RmtRTP   LocalIP                                     RemoteIP
MPSS  VRF
1     22         21         16404    0      Router-IP-Address                           0.0.0.0
NO    NA
Found 1 active RTP connections
```

使用嗅探器捕获，可以验证RTP的接收位置。在本示例中，它从端口**24608**和**CUCM-IP-Address**(而不是端口0和IP地址**0.0.0.0)接收**。

Source Address	Source Port	Destination Address	Destination Port	SSRC	Payload	Packets	Lost	Max Delta (ms)	Max Jitter	Mean Jitter
CUCM IP Address	24608	Router IP Address	16404	0x676	g711U	1095	0 (0.0%)	30.214	3.567	0.759

Debug VoIP RTP Error(调试VoIP RTP错误)显示从**CUCM-IP-Address(而不是0.0.0.0)接收的丢弃数据包**的原因，因此它未通过源验证。

```
voip_rtp_recv_fs_input:ERROR IP address validation failed, dropping packet.
Expected addr: 0.0.0.0, Received addr: CUCM-IP-Address
voip_rtp_recv_fs_input:ERROR IP address validation failed, dropping packet.
Expected addr: 0.0.0.0, Received addr: CUCM-IP-Address
voip_rtp_recv_fs_input:ERROR IP address validation failed, dropping packet.
Expected addr: 0.0.0.0, Received addr: CUCM-IP-Address
voip_rtp_recv_fs_input:ERROR IP address validation failed, dropping packet.
Expected addr: 0.0.0.0, Received addr: CUCM-IP-Address
```

对于SIP:

Debug CCSIP Messages在语音路由器上显**示**SDP收到的SIP ACK消息，该消息指示路由器期望从CUCM-IP-Address和端口**4000获得**RTP。

```
//-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
ACK sip:6002@Router-IP-Address:5060 SIP/2.0
Via: SIP/2.0/UDP CUCM-IP-Address:5060;branch=z9hG4bK16712e94eda
From: <sip:65002@CUCM-IP-Address>;tag=5931~842780d9-7186-4740-ada2-23e5d1b91316-46404140
To: <sip:6002@10.201.160.54>;tag=FE677E-E12
Date: Fri, 19 Apr 2019 23:53:48 GMT
Call-ID: 32798F13-623511E9-805BC9D5-801BF5C7@Router-IP-Address
User-Agent: Cisco-CUCM12.0
Max-Forwards: 70
CSeq: 102 ACK

Allow-Events: presence
Session-ID: 5fdd1bc300105000a000006cbcbcfda2;remote=761410b40eed518a94bd5f7bbccfbe40
Content-Type: application/sdp
Content-Length: 191

v=0
o=CiscoSystemsCCM-SIP 5931 3 IN IP4 CUCM-IP-Address
s=SIP Call
c=IN IP4 CUCM-IP-Address (MoH Server)
t=0 0
m=audio 4000 RTP/AVP 0
a=X-cisco-media:umoh
a=ptime:20
a=rtpmap:0 PCMU/8000
```

```
a=sendonly
```

Show Call Active Voice Brief不显示**RX增量**，该RTP预期从CUCM-IP-Address:**4000收到**。

由于RTP实际上来自另一个端口，因此它被丢弃。

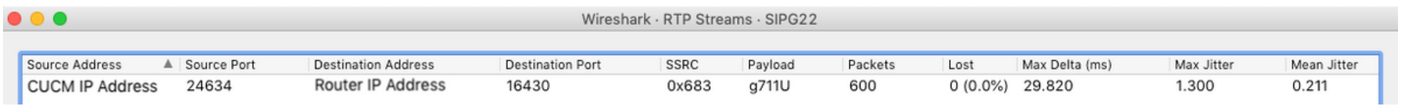
```
11F0 : 29 16672630ms.1 (18:53:43.109 CDT Fri Apr 19 2019) +1450 pid:0 Answer 6002 active
 dur 00:00:07 tx:169/28392 rx:265/42400 dscp:0 media:0 audio tos:0x0 video tos:0x0
 Tele 0/0/0:23 (29) [0/0/0.23] tx:4020/4020/0ms g711ulaw noise:-74 acom:3  i/0:-64/-64 dBm

11F0 : 30 16672630ms.2 (18:53:43.109 CDT Fri Apr 19 2019) +1450 pid:1 Originate 65002 connected
 dur 00:00:07 tx:64/10240 rx:169/27040 dscp:0 media:0 audio tos:0xB8 video tos:0x0
 IP CUCM-IP-Address:4000 SRTP: off rtt:0ms pl:3200/0ms lost:0/0/0 delay:0/55/65ms g711ulaw
TextRelay: off Transcoded: No ICE: Off
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a
 long duration call detected:n long duration call duration:n/a timestamp:n/a
 LostPacketRate:0.00 OutOfOrderRate:0.00
 LocalUUID:5fdd1bc300105000a000006cbcbcfda2
 RemoteUUID:761410b40eed518a94bd5f7bbccfbe40
 VRF: NA
```

Show VoIP RTP Connections将RmtRTP和 RemoteIP **显示为CUCM-IP-Address:4000**，路由器期望RTP来自该源。

```
show voip rtp connections
VoIP RTP Port Usage Information:
Max Ports Available: 8091, Ports Reserved: 101, Ports in Use: 1
Port range not configured
                                        Min   Max   Ports     Ports     Ports
Media-Address Range                     Port  Port  Available Reserved  In-use
------------------------------------------------------------------------------
Global Media Pool                       16384 32766 8091      101       1
------------------------------------------------------------------------------
VoIP RTP active connections :
No. CallId     dstCallId   LocalRTP RmtRTP   LocalIP                                        RemoteIP
MPSS   VRF
1      30          29          16430     4000   Router-IP-Address                              CUCM-IP-
Address                                 NO     NA
Found 1 active RTP connections
```

通过嗅探器捕获，可以验证RTP实际来自何处，在本例中，它来自端口**24634 和CUCM-IP-Address**，而不是**CUCM-IP-Address:4000**。


Wireshark · RTP Streams · SIPG22

Source Address	Source Port	Destination Address	Destination Port	SSRC	Payload	Packets	Lost	Max Delta (ms)	Max Jitter	Mean Jitter
CUCM IP Address	24634	Router IP Address	16430	0x683	g711U	600	0 (0.0%)	29.820	1.300	0.211

Debug VoIP RTP Error(调试VoIP RTP错误)显示从端口24634（而不是端口4000）接收的丢弃数据包的原因，**因此**它未通过源验证。

```
voip_rtp_recv_fs_input:ERROR Port validation failed, dropping RTP packet.
Expected port: 4000, Received port: 24634
voip_rtp_recv_fs_input:ERROR Port validation failed, dropping RTP packet.
Expected port: 4000, Received port: 24634
voip_rtp_recv_fs_input:ERROR Port validation failed, dropping RTP packet.
Expected port: 4000, Received port: 24634
voip_rtp_recv_fs_input:ERROR Port validation failed, dropping RTP packet.
Expected port: 4000, Received port: 24634
```

对于MGCP:

Debug MGCP Packets(调试MGCP数据包)显示呼叫最初协商的介质的时间，以及呼叫被置于保持状态的时间。

```
When the call initially connects, it negotiates the media capabilities through SDP.

MGCP Packet received from CUCM-IP-Address:2427---> MDCX 1324 S0/SU1/DS1-1/23@3945-A.luirami2.lab
MGCP 0.1 C: D000000002c4139b000000F500000008 I: 10 X: 17 L: p:20, a:PCMU, s:off, t:b8 M:
sendrecv
R: D/[0-9ABCD*#]
S:
Q: process,loop

v=0
o=- 16 0 IN EPN S0/SU1/DS1-1/23@3945-A.luirami2.lab
s=Cisco SDP 0
t=0 0
m=audio 23248 RTP/AVP 0
c=IN IP4 IP-Phone-IP-Address
<---

MGCP Packet sent to CUCM-IP-Address:2427--->
200 1324 OK
<---


Then when it is placed on hold, CUCM only changes the direction of the media.
MGCP Packet received from CUCM-IP-Address:2427---> MDCX 1325 S0/SU1/DS1-1/23@3945-A.luirami2.lab
MGCP 0.1 C: D000000002c4139b000000F500000008 I: 10 X: 17 M: recvonly
R: D/[0-9ABCD*#]
Q: process,loop
<---

MGCP Packet sent to CUCM-IP-Address:2427--->
200 1325 OK
<---
```

Show Call Active Voice Brief不显示RX增量，该RTP预期来自IP-Phone-IP-Address:23248。

由于RTP实际上来自另一个IP地址，因此它被丢弃。

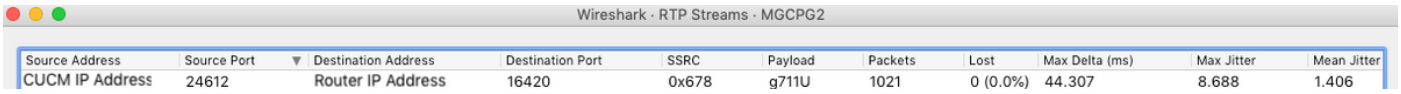
```
11FD : 38 31140580ms.1 (19:24:46.254 CDT Fri Apr 19 2019) +0 pid:0 Originate  connecting
 dur 00:00:36 tx:289/46240 rx:272/43520 dscp:0 media:0 audio tos:0xB8 video tos:0x0
 IP IP-Phone-IP-Address:23248 SRTP: off rtt:1ms pl:5440/70ms lost:0/0/0 delay:0/55/65ms g711ulaw
TextRelay: off Transcoded: No ICE: Off
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a
 long duration call detected:n long duration call duration:n/a timestamp:n/a
 LostPacketRate:0.00 OutOfOrderRate:0.00
 LocalUUID:
 RemoteUUID:
 VRF:
11FD : 37 31140580ms.2 (19:24:46.252 CDT Fri Apr 19 2019) +0 pid:0 Originate  active
 dur 00:00:36 tx:272/45696 rx:1832/293120 dscp:0 media:0 audio tos:0x0 video tos:0x0
 Tele 0/1/1:23 (37) [0/1/1.23] tx:36630/36630/0ms g711ulaw noise:-68 acom:6  i/0:-65/-60 dBm
```

Show VoIP RTP Connections将RmtRTP和RemoteIP显示为IP-Phone-IP-Address:23248，路由器预期RTP来自该源。

```
show voip rtp connections
VoIP RTP Port Usage Information:
Max Ports Available: 8091, Ports Reserved: 101, Ports in Use: 1
```

```
Port range not configured
                                        Min   Max   Ports     Ports    Ports
Media-Address Range                     Port  Port  Available Reserved In-use
------------------------------------------------------------------------------
Global Media Pool                       16384 32766 8091      101      1
------------------------------------------------------------------------------
VoIP RTP active connections :
No. CallId     dstCallId  LocalRTP RmtRTP   LocalIP                                                RemoteIP
MPSS   VRF
1      38         37        16420   23248    Router-IP-Address                                      IP-
Phone-IP-Address                                NO     NA
Found 1 active RTP connections
```

通过嗅探器捕获，可以验证RTP实际来自何处，在本例中，它来自端口**24612和CUCM-IP-Address**，而不是**IP-Phone-IP-Address:23248**。



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Source Address	Source Port	Destination Address	Destination Port	SSRC	Payload	Packets	Lost	Max Delta (ms)	Max Jitter	Mean Jitter
CUCM IP Address	24612	Router IP Address	16420	0x678	g711U	1021	0 (0.0%)	44.307	8.688	1.406

Debug VoIP RTP Error(调试VoIP RTP错误)显示从CUCM-IP-Address(而非**IP-Phone-IP-Address)接收的丢弃数据包**的原因，因此它未通过源验证。

```
voip_rtp_recv_fs_input:ERROR IP address validation failed, dropping packet.
Expected addr: IP-Phone-IP-Address, Received addr: CUCM-IP-Address
voip_rtp_recv_fs_input:ERROR IP address validation failed, dropping packet.
Expected addr: IP-Phone-IP-Address, Received addr: CUCM-IP-Address
voip_rtp_recv_fs_input:ERROR IP address validation failed, dropping packet.
Expected addr: IP-Phone-IP-Address, Received addr: CUCM-IP-Address
voip_rtp_recv_fs_input:ERROR IP address validation failed, dropping packet.
Expected addr: IP-Phone-IP-Address, Received addr: CUCM-IP-Address
```

对于SCCP:

调试SCCP消息显示呼叫处于保持状态的时间。

CUCM首先指示语音路由器使用**CloseReceiveChannel和StopMediaTransmission**切换到**非活**动的**媒体**。

```
SCCP:rcvd CloseReceiveChannel
CloseReceiveChannelMsg Info:
conference_id = 33554439, pass_through_party_id = 33554541, call_ref = 46404215, port_handling =
0

SCCP:rcvd StopMediaTransmission
StopMediaTransmissionMsg Info:
conference_id = 33554439, pass_through_party_id = 33554541, call_ref = 46404215, port_handling =
0
```

然后，CUCM指示语音路由器使用OpenReceiveChannel**切换**到只**接收**。

```
SCCP:rcvd OpenReceiveChannel
OpenReceiveChannelMsg Info:
conference_id = 33554439, pass_through_party_id = 33554542
msec_pkt_size = 20, compression_type = 4
qualifier_in.ecvalue = 0, g723_bitrate = 0, call_ref = 46404215
stream_pass_through_id = 16777216, rfc2833_payload_type = 0
codec_dynamic_payload = 0, codec_mode = 0
Encryption Info :: algorithm_id 0, key_len 0, salt_len 0
requestedAddrType = 0, source_ip_addr.ipAddrType = 0, source_ip_addr = CUCM-IP-Address,
```

```
source_port_number = 4000,
audio_level_adjustment = 0

SCCP:send OpenReceiveChannelAck
OpenReceiveChannelAck Info:
pass_through_party_id=33554542, status=0(ok), host_ip_addr= Router-IP-Address, port=16390
```

Show SCCP Connections显示**ripaddr**和rportas **0.0.0.0:0**；**路由**器预期RTP来自该源。

```
show sccp connections
sess_id     conn_id      stype mode     codec   sport rport ripaddr conn_id_tx

33554439    33554542     mtp   recvonly g711u   16390 0     0.0.0.0
33554439    33554540     mtp   sendrecv g711u   16386 16384 10.201.160.54

Total number of active session(s) 1, and connection(s) 2
```

Debug VoIP RTP Error(调试VoIP RTP错误)显示从**CUCM-IP-Address(**而不是**0.0.0.0)接收的丢弃数据包**的原因，因此它未通过源验证。

```
000147: Apr 24 11:49:22.499: voip_rtp_recv_fs_input:ERROR IP address validation failed, dropping
packet.
Expected addr: 0.0.0.0, Received addr: CUCM-IP-Address
000148: Apr 24 11:49:22.519: voip_rtp_recv_fs_input:ERROR IP address validation failed, dropping
packet.
Expected addr: 0.0.0.0, Received addr: CUCM-IP-Address
000149: Apr 24 11:49:22.539: voip_rtp_recv_fs_input:ERROR IP address validation failed, dropping
packet.
Expected addr: 0.0.0.0, Received addr: CUCM-IP-Address
000150: Apr 24 11:49:22.559: voip_rtp_recv_fs_input:ERROR IP address validation failed, dropping
packet.
Expected addr: 0.0.0.0, Received addr: CUCM-IP-Address
```

IOS-XE语音路由器上的RTP源验证

在IOS-XE中，最需要重点介绍的是。

1. 不可配置
2. 默认情况下已启用
3. 无法禁用
4. VoIP信令中的媒体方向是唯一允许RTP从未知源流动的例外

每个协议的行为和检测

对于H.323:

使用此协议时，来自MoH的RTP不起作用，因为CUCM始终将IP地址和端口设置为零的**openLogicalChannelAck**消息发送，从而禁用媒体。

```
H245 MSC INCOMING PDU ::=

value MultimediaSystemControlMessage ::= response : openLogicalChannelAck :
    {
      forwardLogicalChannelNumber 6
      forwardMultiplexAckParameters h2250LogicalChannelAckParameters :
      {
```

```
sessionID 1
mediaChannel unicastAddress : iPAddress :
{
  network '00000000'H
  tsapIdentifier 0
}
mediaControlChannel unicastAddress : iPAddress :
{
  network '00000000'H
  tsapIdentifier 1
```

使用**Show Call Active Voice Brief** 可以验证同一情况，以检查**RX**增量值如何停止以及远程媒体地址**是IP 0.0.0.0:0**。

```
11F3 : 17 8703830ms.1 (13:00:22.060 CDT Tue Apr 23 2019) +2150 pid:2 Answer 6002 active
 dur 00:15:22 tx:19014/9213600 rx:1/3836010 dscp:0 media:0 audio tos:0x0 video tos:0x0
 Tele 0/1/1:23 (17) [0/1/1.23] tx:158740/106870/0ms g711ulaw noise:-68 acom:22  i/0:-57/-61 dBm

11F3 : 18 8703830ms.2 (13:00:22.060 CDT Tue Apr 23 2019) +2150 pid:1 Originate 55002 active
 dur 00:15:22 tx:19709/3836010 rx:46068/9213600 dscp:0 media:0 audio tos:0xB8 video tos:0x0
 IP 0.0.0.0:0 SRTP: off rtt:0ms pl:0/0ms lost:0/0/0 delay:0/0/0ms g711ulaw TextRelay: off
Transcoded: No ICE: Off
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a
 long duration call detected:n long duration call duration:n/a timestamp:n/a
 LostPacketRate:0.00 OutOfOrderRate:0.00
```

> **警告：RX**和**TX**在IOS-XE平台中不会增加，除非在语音服务VoIP下配置**Media Bulk-Stats** 命令，但请注意，此命令可能影响路由器的性能，因此建议仅在进行故障排除时启用它，然后禁用它。

Debug Voip FPI Inout不显示此处启**用的网络地址转换(NAT)标**志，因为**openLogicalChannelAck禁用了媒体**，可以使用消息端**:SIDE_A、rtp_type:0：检查媒体禁用**。

```
//18/7F507F32800A/VOIPFPI:():voip_fpi_get_snd_rcv_enable_flag: side:SIDE_A, rtp_type:0:  send:0
recv:0
//18/7F507F32800A/VOIPFPI:():voip_fpi_get_snd_rcv_enable_flag: destAddr == 0, rcv and send both
set to FALSE
```

show platform hardware qfp active feature sbc global | s丢弃的数据包总数|丢弃的数据包数：显示一个表，其中呼叫处于暂候**时入口流接收**被禁用的所有丢弃数据包数增加。

```
show platform hardware qfp active feature sbc global | s Total packets dropped|Dropped packets:
  Total packets dropped                      = 138512
Dropped packets:
  No associated flow                         = 0
  Wrong source for flow                      = 0
  Ingress flow receive disabled              = 138512
  Egress flow send disabled                  = 0
  Not conforming to flowspec                 = 0
```

对于SIP

当使用SIP时，CUCM在SDP中发送**CUCM-IP-Address、Port 4000**和方向的媒体属性**a=sendonly**，指示路由器仅接收RTP。

```
v=0
o=CiscoSystemsCCM-SIP 72019 3 IN IP4 CUCM-IP-Address
```

```
s=SIP Call
c=IN IP4 CUCM-IP-Address (MoH Server)
t=0 0
m=audio 4000 RTP/AVP 0
a=X-cisco-media:umoh
a=ptime:20
a=rtpmap:0 PCMU/8000
a=sendonly
```

a=sendonly将语音路由器的媒体方向设置为**recvonly**，这会触发**NAT标志**功能，即使RTP来自其他源，该功能仍允许RTP通过。

这可以通过调试VoIP FPI**输出来检查**。

```
//25/3EAF69800000/VOIPFPI:():voip_fpi_get_snd_rcv_enable_flag: side:SIDE_A, rtp_type:2:RECVONLY
send:0 recv:2
//25/3EAF69800000/VOIPFPI:():voip_fpi_get_snd_rcv_enable_flag: recvonly mode - setting NAT flag
```

如果发生此情况时将不同的Attribute for Media Direction发送到语音路由器，则不会激活**NAT标志**功能，并且数据包将被丢弃，因为它们来自不同的源。

调试CCSIP消息在本示例中**显示a=sendrecv。**

```
v=0
o=CiscoSystemsCCM-SIP 72019 3 IN IP4 CUCM-IP-Address
s=SIP Call
c=IN IP4 CUCM-IP-Address (MoH Server)
t=0 0
m=audio 4000 RTP/AVP 0
a=X-cisco-media:umoh
a=ptime:20
a=rtpmap:0 PCMU/8000
a=sendrecv
```

调试VoIP FPI输出显示媒体方向设置为**rtp_type:3:SENDRECV，**无NAT**标志**功能。

```
//27/F56119000000/VOIPFPI:():voip_fpi_get_snd_rcv_enable_flag: side:SIDE_A, rtp_type:3:SENDRECV
send:1 recv:2
```

由于没有NAT标**志，因**此**show platform hardware qfp active feature sbc global | s已丢弃的数据包总数|已丢弃的数据包数：**在"流的源错**误"部分显示增**量。

```
4351-A#show platform hardware qfp active feature sbc global | s Total packets dropped|Dropped
packets:
  Total packets dropped                        = 33496
Dropped packets:
  No associated flow                           = 0
  Wrong source for flow                        = 33196
  Ingress flow receive disabled                = 0
  Egress flow send disabled                    = 0
  Not conforming to flowspec                   = 0
```

对于MGCP:

当使用MGCP时，CUCM发送MDCX以更改最初连接呼叫时已协商的媒体方向，因此IP地址或信令没有更改，但在MDCX之后，RTP现在从另一个源流传输。

自**M:recvonly**被发送到语音路由器，**NAT标**志功能启用。

```
MGCP Packet received from CUCM-IP-Address:2427--->
MDCX 1529 S0/SU1/DS1-1/23@4351-A.luirami2.lab MGCP 0.1
C: D000000002c4151d000000F50000000a
I: B
X: 17
M: recvonly
R: D/[0-9ABCD*#]
Q: process,loop
<---
```

调试VoIP FPI输出显示媒体方向设置为**rtp_type:2:RECVONLY和**NAT标志功**能，**允许RTP通过。

```
//30/xxxxxxxxxxxx/VOIPFPI:():voip_fpi_get_snd_rcv_enable_flag: side:SIDE_A, rtp_type:2:RECVONLY
send:0 recv:2
//30/xxxxxxxxxxxx/VOIPFPI:():voip_fpi_get_snd_rcv_enable_flag: recvonly mode - setting NAT flag
```

如果发生此情况时将不同的Attribute for Media Direction发送到语音路由器，则不会激活**NAT标志**功能，并且数据包将被丢弃，因为它们来自不同的源。

调试MGCP数据包显示在本示**例M:sendrecv**。

```
MGCP Packet received from CUCM-IP-Address:2427--->
MDCX 1530 S0/SU1/DS1-1/23@4351-A.luirami2.lab MGCP 0.1
C: D000000002c4151d000000F50000000a
I: B
X: 17
M: sendrecv
R: D/[0-9ABCD*#]
Q: process,loop
<---
```

调试VoIP FPI输出显示媒体方向设置为**rtp_type:3:SENDRECV，**无NAT**标志**功能。

```
//29/F56119000000/VOIPFPI:():voip_fpi_get_snd_rcv_enable_flag: side:SIDE_A, rtp_type:3:SENDRECV
send:1 recv:2
```

由于没有**NAT**标志,因**此show platform hardware qfp active feature sbc global | s已丢弃的数据包总数|已丢弃的数据包数：**在"流的源错**误"部分显示增**量。

```
show platform hardware qfp active feature sbc global | s Total packets dropped|Dropped packets:
  Total packets dropped                              = 33596
Dropped packets:
  No associated flow                                 = 0
  Wrong source for flow                              = 33296
  Ingress flow receive disabled                      = 0
  Egress flow send disabled                          = 0
  Not conforming to flowspec                         = 0
```

对于SCCP:

调试SCCP消息显示呼叫处于保持状态的时间。

CUCM首先指示语音路由器使用CloseReceiveChannel和StopMediaTransmission切换**到不**活动**的媒体。**

```
SCCP:rcvd CloseReceiveChannel
CloseReceiveChannelMsg Info:
```

```
conference_id = 33554436, pass_through_party_id = 33554500, call_ref = 46405010, port_handling =
0

SCCP:rcvd StopMediaTransmission
StopMediaTransmissionMsg Info:
conference_id = 33554436, pass_through_party_id = 33554500, call_ref = 46405010, port_handling =
0
```

然后，CUCM指示语音路由器切换为仅与OpenReceiveChannel**进行恢复。**

```
SCCP:rcvd OpenReceiveChannel
OpenReceiveChannelMsg Info:
conference_id = 33554436, pass_through_party_id = 33554501
msec_pkt_size = 20, compression_type = 4
qualifier_in.ecvalue = 0, g723_bitrate = 0, call_ref = 46405010
stream_pass_through_id = 16777216, rfc2833_payload_type = 0
codec_dynamic_payload = 0, codec_mode = 0
Encryption Info :: algorithm_id 0, key_len 0, salt_len 0
requestedAddrType = 0, source_ip_addr.ipAddrType = 0, source_ip_addr = CUCM-IP-Address,
source_port_number = 4000,
audio_level_adjustment = 0

SCCP:send OpenReceiveChannelAck
OpenReceiveChannelAck Info:
pass_through_party_id=33554501, status=0(ok), host_ip_addr= Router-IP-Address, port=8028
```

Show SCCP Connections显示**ripaddr**和rportas **0.0.0.0:0；路由**器预期RTP来自该源。

```
show sccp connections
sess_id    conn_id      stype mode     codec   sport rport ripaddr conn_id_tx

33554436   33554501     mtp   recvonly g711u   8028  0     0.0.0.0
33554436   33554499     mtp   sendrecv g711u   8022  8024  Router-IP-Address

Total number of active session(s) 1, and connection(s) 2
```

调试VoIP FPI输出显示媒体方向设置为**rtp_type:2:RECVONLY和**NAT标志功**能，**允许RTP通过。

```
//18/xxxxxxxxxxxx/VOIPFPI:():voip_fpi_get_snd_rcv_enable_flag: side:SIDE_A, rtp_type:1:SENDONLY
send:1 recv:0
//15/xxxxxxxxxxxx/VOIPFPI:():voip_fpi_get_snd_rcv_enable_flag: side:SIDE_B, rtp_type:3:SENDRECV
send:1 recv:2
//19/xxxxxxxxxxxx/VOIPFPI:():voip_fpi_get_snd_rcv_enable_flag: side:SIDE_A, rtp_type:2:RECVONLY
send:0 recv:2
//19/xxxxxxxxxxxx/VOIPFPI:():voip_fpi_get_snd_rcv_enable_flag: recvonly mode - setting NAT flag
//15/xxxxxxxxxxxx/VOIPFPI:():voip_fpi_get_snd_rcv_enable_flag: side:SIDE_B, rtp_type:3:SENDRECV
send:1 recv:2
```

> **提示**: **OpenReceiveChannel** 消息用于指示语音路由器接收RTP，而语音路由器通过**OpenReceiveChannelAck**告知CUCM要在何处接收该媒体。**StartMediaTransmission消**息用于指示语音路由器将RTP发送到指定的目标。
>
> 换句话说，如果只交换**OpenReceiveChannel**是通知媒体资源它只接收RTP(**recvonly**)的一种方式，并且如果只交换**StartMediaTransmission**，则是告知媒体资源它只发送RTP（仅发送）的一种方式，但如果交换了这两种方式，则它等于发送recv

如果媒体方向设置为**sendonly**或**sendrecv**，并且RTP来自不同的源，则不会激活**NAT** 标志，并且show platform hardware qfp active feature sbc global **| s丢弃的数据包总数|丢弃的数据包：**显示丢弃的数据包。

提示：如果需要允许来自不同于通过信令协商的地址的RTP，并且**recvonly** 不能使用，则**语音服务Voip** 下的**nat force-on** ,**Sip**可用于添加手动扩展。此问题以前无法正常工作，但已修复缺陷 CSCvo15141 。请记住，这仅适用于SIP。

警告：如果在语音服务voip下配置了sip ，则这不允许FPI层在收到recvonly时激活NAT标志功能。

提示：在NAT标志对呼**叫处于活动状态**且音频工作正常的某些情况下，在show platform hardware qfp active feature sbc global下丢弃**的数据包值 |s丢弃的数据包总数|丢弃的数据包数：**但速率仍然会提高，这是因为在某些情况下和呼叫流中，实时控制协议(RTCP)仍然可以发送到语音路由器，并且可以从其他源发送，这会导致此行为。