IOS和IOS-XE語音路由器中的RTP源驗證

目錄

簡介

本檔案介紹Cisco IOS和IOS-XE語音路由器中適用於不同通話流程和版本的RTP來源驗證功能的行為。

必要條件

需求

思科建議您瞭解以下主題：

- IOS和IOS-XE軟體
- H.323
- 作業階段啟始通訊協定(SIP)
- 媒體閘道控制通訊協定(MGCP)
- 精簡型通話控制通訊協定(SCCP)
- 即時傳輸通訊協定(RTP)

採用元件

本文中的資訊係根據以下軟體和硬體版本：

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

ASR1006-X）

本文中的資訊是根據特定實驗室環境內的裝置所建立。文中使用到的所有裝置皆從已清除（預設）的組態來啟動。如果您的網路運作中，請確保您瞭解任何指令可能造成的影響。

背景資訊

為了能夠充分利用本文檔，瞭解VoIP網路和VoIP信令協定的基礎知識非常重要。

RTP源驗證定義和用途

RTP源驗證是Cisco語音路由器中整合的一項功能，允許它們丟棄不受信任的入站RTP流量。

此功能的主要目標是提高裝置的安全級別，同時避免VoIP網路上的CrossTalk問題。

IOS語音路由器具有不同的功能風格，而IOS-XE語音路由器只有一個選項。

在IOS和IOS-XE中，此功能使語音路由器丟棄來自未知IP地址或埠的入站RTP流量，換句話說，來自未通過信令協商的IP地址或埠的資料包將被語音路由器丟棄。

此功能在IOS和IOS-XE中的運作方式略有不同，這是因為路由器的架構以及當路由器被引入到代碼中時；下一節將介紹這些場景。

IOS語音路由器中的RTP來源驗證

IOS具有兩種不同的功能。

- 12.4(6)T中引入的源濾波器
- 15.5(3)M9、15.6(3)M6及更高版本中引入的語音RTP源過濾器

注意：請注意，以下各節中介紹的場景是Cisco Unified Communications Manager(CUCM)Music on Hold(MoH)，但在其他情況下，只要滿足要求，相同的行為就會觸發功能刪除RTP。

源篩選器

此功能僅適用於SIP呼叫流。

配置後，如果呼叫流中使用的信令未協商RTP來自的IP地址和埠，則語音路由器會丟棄這些資料包。

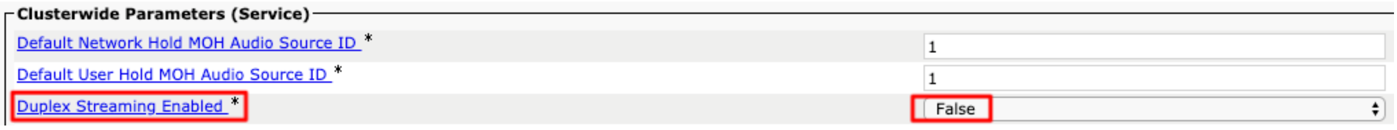
來源驗證會檢查**來源IP地址**，然後檢查**來源連線埠**。

組態

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

行為和檢測

一個很好的例子是，CUCM將呼叫置於保持狀態，並且預設情況下，CUCM通過信令通告埠**4000**，但實際上是從臨時埠(32768-61000)流傳輸RTP，因為預設情況下禁用了**Clusterwide Parameters**下的服務引數**Duplex Streaming**。



Debug CCSIP Messages在語音路由器上顯示使用會話描述協定(SDP)接收的**SIP ACK消息，告知路由器的RTP來自CUCM-IP-Address 和Port 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不會顯示RTP來自**CUCM-IP-Address 和埠4000的支路上的RX**增量。RTP從另一個埠接收並被語音路由器丟棄。

```
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將**RmtRTP**顯示為**4000 ，將RemoteIP顯示為**CUCM-IP-Address。

路由器期望RTP來自同一來源。

```
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
```

使用監聽器擷取，可以驗證RTP實際上來自何處，在本範例中，它來自連線埠**2458**，而不是**4000**，因此來源驗證失敗，且語音路由器捨棄封包。

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

語音RTP來源過濾器

此功能在15.5(3)M9、15.6(3)M6 IOS版本中引入。

其工作方式與**來源過濾器**相同，它首先驗證**來源IP位址**，然後驗證**來源連線埠**，但具有兩個主要差異。

1. **語音RTP源過濾器**適用於SIP、H.323、MGCP和SCCP
2. 此功能還在**Debug VoIP RTP Error**中新增了錯誤消息，以便輕鬆檢測由於源驗證失敗而丟棄RTP的時間

   注意：此功能預設為啟用，且不會顯示在組態中。如果裝置從不同於通過信令通告的源傳送RTP，則升級到支援此功能的任何IOS版本都可能導致音訊問題。
   當在命令前面使用**No**禁用此功能時，該功能會在配置中顯示。

組態

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

每個協定的行為和檢測

對於H.323:

在語音路由器上調試**H**225 Asn1顯示收到的**openLogicalChannelAck**，它將遠端媒體地址0.**0.0.0:0通知路由器**。

```
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 不顯示RX增量，且遠端IP地址和埠設定為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將RmtRTP和RemoteIP 顯示為0.0.0.0:0，因此路由器期望來自該源的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
```

透過監聽器擷取，可以驗證接收RTP的位置。在本範例中，系統從連線埠**24608** 和**CUCM-IP-Address**而不是連線埠**0**和IP位址**0.0.0.0**接收該封包。

Wireshark · RTP Streams · H

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 顯示從**CUCM-IP-Address** 而不是**0.0.0.0接收這些丟棄的資料包的原因，因此它未通過源驗證。**

```
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
```

對於SIP:

Debug CCSIP Messages在語音路由器上顯示**SIP ACK**消息，該消息通過SDP接收，指示路由器期望從**CUCM-IP-Address和Port 4000獲得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不會顯示**從CUCM-IP-Address:4000**接收RTP的支路**上的RX增量。**

由於RTP實際上來自另一個埠，因此會將其丟棄。

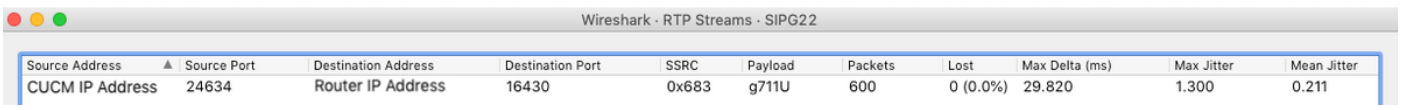
```
11F0 : 29 16672630ms.1 (18:53:43.109 CDT Fri Apr 19 2019) +1450 pid:0 Answer 6002 active
 dur 00:00:07 tx:169/28392 rx:265/42400 dscp:0 media:0 audio tos:0x0 video tos:0x0
 Tele 0/0/0:23 (29) [0/0/0.23] tx:4020/4020/0ms g711ulaw noise:-74 acom:3  i/0:-64/-64 dBm

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將RmtRTP和RemoteIP 顯示為CUCM-IP-Address:4000，路由器期望RTP來自該來源。

```
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
```

使用監聽器擷取，可以驗證RTP實際上來自何處，在本範例中，它來自連線埠**24634和CUCM-IP-Address**，而不是**CUCM-IP-Address:4000**。



Debug VoIP RTP Error 顯示從埠24634而不是埠4000接收這些丟**棄**資料包的原**因**，因此它未通過源驗證。

```
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
```

對於MGCP:

Debug MGCP Packets(調試MGCP資料包)顯示呼叫最初協商介質的時間，然後是呼叫被置於保持狀態的時間。

```
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不會顯示**RTP來自IP-Phone-IP-Address:23248的支路上的RX增量。**

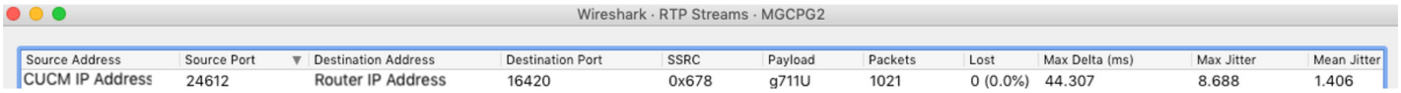
因為RTP實際上來自另一個IP地址，所以它被丟棄。

```
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將**RmtRTP和RemoteIP 顯示為IP-Phone-IP-Address:23248**，路由器希望RTP來自該來源。

```
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
```

使用監聽器擷取，可以驗證RTP實際上來自何處，在本範例中，它來自連線埠**24612和CUCM-IP-Address**，而不是**IP-Phone-IP-Address:23248**。



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 顯示從**CUCM-IP-Address(而不是IP-Phone-IP-Address)接收這些丟棄的資料包的原因，因此它無法通過源驗證。**

```
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
```

對於SCCP:

Debug SCCP Messages顯示呼叫被置於保持狀態的時間。

CUCM首先指示語音路由器使用**CloseReceiveChannel**和**StopMediaTransmission**切換到**inactive**介質。

```
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
```

然後CUCM指示語音路由器使用**OpenReceiveChannel切換**。

```
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顯示**ripaddr和**rportas **0.0.0.0:0;**路由器期望RTP來自該源**。**

```
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 顯示從**CUCM-IP-Address(而不是0.0.0.0)接收這些丟棄的資料包的原因，因此它未通過源驗證。**

```
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語音路由器上的RTP源驗證

在IOS-XE中需要強調的最重要內容是。

1. 不可配置
2. 預設情況下啟用
3. 無法禁用
4. VoIP信令中的媒體方向是允許RTP從未知源流出的唯一例外

每個協定的行為和檢測

對於H.323:

使用此協定時，MoH的RTP不起作用，因為CUCM總是傳送帶有IP地址和埠設定為零的**openLogicalChannelAck**消息，它會禁用介質。

```
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** 檢查如何停止RX增量值，以及遠端介質地址為**IP 0.0.0.0:0**可以檢驗相同的情況。

```
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：IOS-XE平台中的RX和TX**不會遞增，除非在**語音服務VoIP**下配置了**Media Bulk-Stats命令，但請注意，此命令可能會影響路由器的效能，因此建議僅在進行故障排除時啟用此命令，並在之後將其禁用。**

Debug Voip FPI Inout不顯示Network Address Translation(NAT)Flag在此處啟用，因為使用**openLogicalChannelAck**禁用了媒體，可以使用消息**side:SIDE_A， rtp_type:0:**檢查禁用了媒體。

```
//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 Total packets dropped|Dropped packets：顯示一個表，其中包含所有丟棄的**包，其中入口流接收被禁**用的資料包在呼叫保持期間遞增。

```
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
```

對於SIP

使用SIP時，CUCM會在SDP中將**CUCM-IP-Address**、Port **4000**和方向媒體屬性傳送為**a=sendonly**，這指示路由器只接收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將媒體方向設定為語音路由器視角的**recvonly**，這將觸發**NAT標誌**功能，該功能仍然允許RTP通過，即使它來自其他源。

這可透過**Debug VoIP FPI Inout**檢查。

```
//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
```

如果發生這種情況，向語音路由器傳送了不同的媒體方向屬性，將不會啟用**NAT標誌**功能，並且資料包將被丟棄，因為它們來自不同的源。

Debug CCSIP Messages顯示在此示例**a=sendrecv中**。

```
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
```

Debug VoIP FPI Inout顯示媒體方向已設**置為rtp_type:3:SENDRECV**，且無**NAT標誌**功能。

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

由於沒有NAT**標誌**,show platform hardware qfp active feature sbc global **| s Total packets dropped|Dropped packets：顯示Wrong source for flow**部分中**的增量數**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
```

對於MGCP:

使用MGCP時，CUCM會傳送一個MDCX，以便更改呼叫最初連線時已經協商的媒體方向，因此IP地址或信令沒有更改，但在MDCX之後，RTP現在會從另一個源進行流式傳輸。

自**M:recvonly**被傳送到語音路由器，**NAT標誌**功能被啟用。

```
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
<---
```

Debug VoIP FPI Inout顯示媒體方向設定為**rtp_type:2:RECVONLY和NAT標誌**功能，允許RTP通過。

```
//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
```

如果發生這種情況，向語音路由器傳送了不同的媒體方向屬性，將不會啟用**NAT標誌**功能，並且資料包將被丟棄，因為它們來自不同的源。

Debug MGCP Packets顯示在此範例**M中：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
<---
```

Debug VoIP FPI Inout顯示媒體方向已設**置為rtp_type:3:SENDRECV**，且無**NAT標誌**功能。

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

由於沒有NAT**標誌,**show platform hardware qfp active feature sbc global **| s Total packets dropped|Dropped packets：顯示Wrong source for flow**部分中**的增量數**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
```

對於SCCP:

Debug SCCP Messages顯示呼叫被置於保持狀態的時間。

CUCM首先指示語音路由器使用**CloseReceiveChannel**和**StopMediaTransmission**切換到非活動媒體。

```
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
```

然後CUCM指示語音路由器使用**OpenReceiveChannel**進行恢復。

```
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顯示**ripaddr和**rportas **0.0.0.0:0;**路由器期望RTP來自該源**。**

```
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
```

Debug VoIP FPI Inout顯示媒體方向設定為**rtp_type:2:RECVONLY和NAT標誌**功能，允許RTP通過。

```
//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消息用於指示語音路由器接收RTP，而語音路由器通過OpenReceiveChannelAck通知CUCM要接收該媒體的位置。**
> **StartMediaTransmission**消息用於指示語音路由器將RTP傳送到指定的目標。
>
> 換句話說，如果只交換**OpenReceiveChannel**，是一種告知媒體資源它只接收RTP(**recvonly**)的方式，如果只交換**StartMediaTransmission**，是一種告知媒體資源它只傳送RTP(sendonly)的方式，但如果兩者都交換，則等於sendrecv。

如果媒體方向設定為**sendonly**或**sendrecv**，並且RTP來自其他源，則不啟用**NAT**標誌，並且show platform hardware qfp active feature sbc global **| s Total packets dropped|Dropped packets：顯示** packets dropped。

> **提示：**如果需要允許來自不同地址的RTP，而不是通過信令協商的地址，且無法使用**recvonly**，則可以使用**voice Service Voip**下的**nat force-on**、**Sip**來新增手動期望。以前不能正常工作，但已修復缺陷 [CSCvo15141](https://bst.cloudapps.cisco.com/bugsearch/bug/CSCvo15141) .請記住，這僅適用於SIP。

> 警告：如果在**語音服務voip下配置sip，則傳遞內容sdp，這將不允許在收到**recvonly**時FPI層啟用NAT標誌**功能。

> **提示：**某些情況下，如果呼叫和音訊的**NAT標誌**處於活動狀態，則在**show platform hardware qfp active feature sbc global**下丟棄資料包值 **|s丟棄的資料包總數|丟棄的資料包：**仍然能夠以低得多的速率增加，這是因為在某些情況和呼叫流程中，即時控制協定(RTCP)仍可傳送到語音路由器並從其他源傳送，這將導致此行為。