在IOS-XE語音網關上配置Land Mobile Radio(LMR)/Hoot和Holler Over IP

目錄

簡介

本檔案介紹陸地行動無線電(LMR)或Hoot and Holler(Hootie)功能，此功能允許類比裝置透過啟用多點傳送的LAN與其他（類比和IP）終端通訊。

語音網關充當IP網路和模擬端點之間的分界點，促進模擬音訊和組播即時傳輸協定(RTP)之間的交流。

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必要條件

需求

思科建議您瞭解以下主題：

- 數位訊號處理器(DSP)
- [類比卡](#)
- 適用於該功能的許可證

```
!
license boot level appxk9
license boot level uck9
! or
license boot suite FoundationSuiteK9
license boot suite AdvUCSuiteK9
```

- 支援組播的LAN或WAN

    附註：本檔案未涉及LAN或WAN上多點傳送組態的許多方面。請參閱適用的文檔，以在網路路徑中啟用LAN或WAN裝置上的組播。

採用元件

- 4451-X
- NIM-4E/M
- IOS-XE 16.3或更高版本。([發行說明](#))[建議:IOS-XE 16.7或更高版本]

```
ISR4451# show inventory
NAME: "Chassis", DESCR: "Cisco ISR4451 Chassis"
PID: ISR4451-X/K9      , VID: V03  , SN: XXXXXXXXX

NAME: "NIM subslot 0/3", DESCR: "NIM-4E/M Voice Analog Module"
PID: NIM-4E/M          , VID: V01  , SN: XXXXXXXXX
```

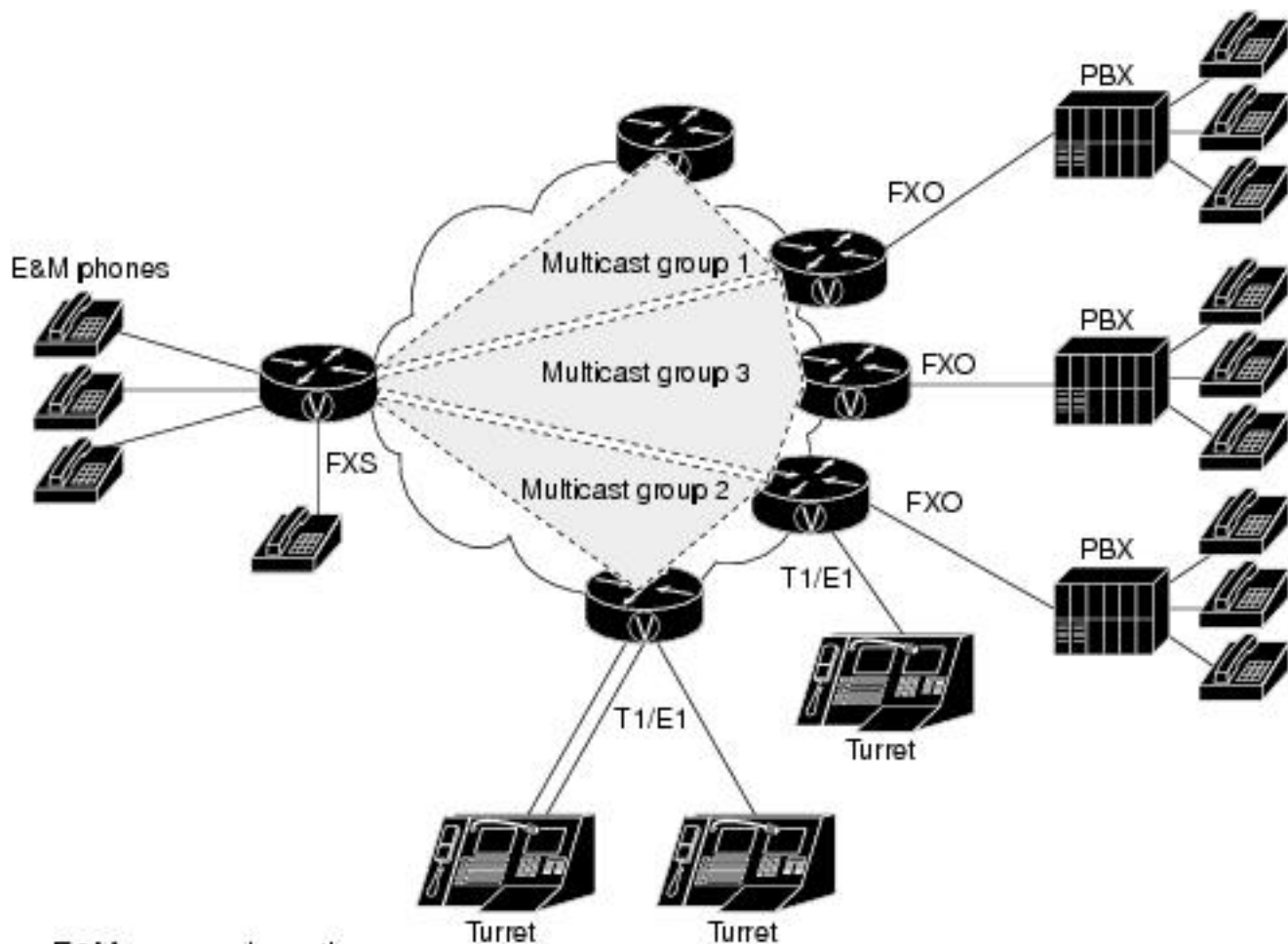   **附註**：與ISR 4000語音網關配合使用的模擬NIM卡使用本地NIM DSP。 因此不需要主機板DSP。

背景資訊

潛在使用案例：

- 包括即按即說裝置的無線電裝置和告警
- 組播資訊通告（無線廣播）
- 類比轉塔系統

    附註：以下是一些示例使用情形。應用程式不限於這些功能。

LMR的[原始設計手冊](#)未涵蓋最新一代思科語音網關所需的專案。因此，本文檔旨在介紹ISR 4300和4400系列語音網關等IOS-XE裝置的LMR/Hootie功能。

以下是拓撲示例；

E&M = ear and mouth
FXO = Foreign Exchange Office
FXS = Foreign Exchange Station

第7層信令和媒體

```
Analog Endpoint <> Ear and Mouth (E&M) Port <> Cisco Voice Gateway (4451-X) <> Multicast Enabled
LAN <> IP Endpoint.
```

提示： 請記住，因為IP骨幹網使用多點傳送，語音閘道只需要能夠成功加入所需的多點傳送群組。語音網關並不瞭解其他終端，也不直接與其通訊，因此本文檔詳細說明了一個 LMR/Hootie語音網關中的示例配置、調試、顯示命令和故障排除。

組態

步驟1。您必須首先配置操作語音和多播功能所需的IOS-XE許可證。

```
config t
!
license boot level appxk9
license boot level uck9
! or
license boot suite FoundationSuiteK9
license boot suite AdvUCSuiteK9 ! exit ! wr ! reload !
```
裝置重新通電後，驗證許可證狀態是否與以下show命令輸出匹配：

```
ISR4451# show license feature
Feature name            Enforcement  Evaluation  Subscription  Enabled  RightToUse
appxk9                  yes          yes         no            yes      yes
uck9                    yes          yes         no            yes      yes
```

步驟2.接下來，定義包含所需組播IP和埠的組播IP語音撥號對等體；

```
!
dial-peer voice 33333 voip
destination-pattern 33333
session protocol multicast
session target ipv4:239.X.X.X:21000
codec g711ulaw
vad aggressive !
```

Dial-peer命令語法：

CLI指令	說明
destination-pattern <number>	撥號對等體的Match語句。撥號對等體必須可用。
會話協定組播	指示裝置此撥號對等體用於IP組播功能。
會話目標 ipv4:<a.b.c.d.>:xxxxx	這是語音網關加入以傳送/接收組播RTP的組播組的IP和埠。
codec <codec>	定義用於多播RTP資料包的編解碼器。支援的編解碼器是G711ulaw、G711alaw、G72G726。
[no] vad [aggressive]	當您使用no vad命令禁用VAD時，請為此RTP流禁用語音活動檢測。 使用命令vad aggressive時，VAD雜訊閾值將從–78降低到–62 dBm。低於–62 dBm閾值訊被視為靜音，不會通過網路傳送。此外，未知資料包被視為靜默資料包並被丟棄。來 附註：使用vad aggressive時，由於不需要從LMR路由器傳送資料包，因此在show ip mroute中可能看不到VIF。

步驟3。為了協助多點傳送群組與此語音閘道之間的模擬連線埠永久性（一律為up）連線，您必須
定義voice-class permanent，然後將其套用到語音連線埠。

```
!
voice class permanent 1
 signal timing oos timeout disabled
 signal keepalive disabled
!
voice-port 0/1/0
voice-class permanent 1
!
```

voice class permanent command syntax

CLI指令	說明	來源
signal timing oos timeout { disabled	<seconds> }	禁用信令丟失檢測。或者，可以配置秒數。	命令語
signal keepalive {	指定keepalive信令資料包間隔（以秒為單位）。停用不會傳送keepalive。	命令語

disabled | <seconds> }

然後為E&M埠配置所需的連線型別，然後配置命令。(*E&M或本文檔未涵蓋的其他模擬特定配置有關詳細資訊，請參閱《E&M配置指南》。*)

步驟4. Cisco hoot和holler over IP提供永遠線上的通訊網橋。終端使用者無需撥打任何電話號碼即可與hoot組的其他成員聯絡。為了模擬此功能，Cisco IOS提供一種稱為連線中繼的功能。連線中繼提供永久語音呼叫，由於所有數字均由路由器/網關內部撥號，因此不需要終端使用者進行任何輸入。

此連線中繼將語音埠繫結到您在撥號對等體配置步驟中配置的組播地址。

```
!
voice-port 0/1/0 connection trunk 33333 !
```

Analog Port命令語法

CLI指令	說明	來
connection trunk <number>	指定模擬與PBX的永久中繼連線的連線。如果沒有任何啟用呼叫，則中繼連線將保持永久。	命

步驟5.語音配置完成後，您需要定義組播配置。

```
!
ip multicast-routing distributed
!
interface GigabitEthernet0/0/1
ip address Y.Y.Y.Y 255.255.255.0
ip pim sparse-mode
! interface Vif1
ip address 192.0.2.2 255.255.255.0
ip pim sparse-mode
!
interface Service-Engine0/1/0
ip pim sparse-mode
!
ip pim rp-address 2.x.x.x
!
```

有關組播配置的說明：

- 服務引擎介面是模擬NIM上PVDM的第3層介面。此配置需要與任何其他輸入/輸出第3層介面一樣使用Protocol Independent Multicast(PIM)命令進行配置
- 其 Service-Engine不需要IP地址
- PIM配置的型別取決於LAN上組播實施的型別
- 必須啟用組播路由，即使所有流量都在同一個VLAN中
- 對於源自路由器的組播RTP，IP必須是VIF IP減去1。因此我們的源必須為192.0.2.1，因為我們已在VIF上配置192.0.2.2 在某些情況下，這可能是VIF加1，但是對於此配置，VIF假設減1。請始終選中**show ip mroute**以檢視路由器正在使用的VIF。
- 組播PIM RP可以是同一個語音網關，但對於本實驗，組播PIM RP位於網路(2.x.x.x)中通過EIGRP獲知的其它裝置上（未顯示）

驗證

使用本節內容，確認您的組態是否正常運作。

語音驗證

當配置完成時，永久連線將停止。您可以使用此show命令輸出進行驗證；

```
ISR4451# show call active voice compact
 <callID> A/O FAX T<sec> Codec      type       Peer Address      IP R<ip>:<udp>      VRF
Total call-legs: 2
      115 ANS    T24    g711ulaw   TELE       P
      116 ORG    T0     g711ulaw   VOIP       P33333      239.X.X.X:21000


ISR4451# show voip rtp connections
VoIP RTP Port Usage Information:
Max Ports Available: 19999, Ports Reserved: 101, Ports in Use: 0
Port range not configured
                                          Min    Max   Ports     Ports     Ports
Media-Address Range                       Port   Port  Available Reserved  In-use
-------------------------------------------------------------------------------
Global Media Pool                         8000   48198 19999     101       0
-------------------------------------------------------------------------------
VoIP RTP active connections :
No. CallId     dstCallId  LocalRTP RmtRTP   LocalIP                         RemoteIP
MPSS VRF
1   116        115        15986    21000    192.0.2.1                       239.X.X.X
NO   NA
Found 1 active RTP connections

ISR4451# show voice port summary
                                          IN       OUT
PORT            CH   SIG-TYPE    ADMIN OPER STATUS   STATUS   EC
=============== == ============ ===== ==== ======== ======== ==
0/3/1           -- e&m-imd      up    up   trunked  trunked  y



ISR4451# show voice call summary
PORT            CODEC     VAD VTSP STATE            VPM STATE
=============== ========= === ==================== =====================
0/3/1           g711ulaw  y   S_CONNECT            S_TRUNKED



ISR4451# show voice call status
CallID    CID  ccVdb        Port         Slot/Bay/DSP:Ch  Called #   Codec     MLPP Dial-peers
0x73      12D0 0x7F7475CF8C08 0/3/1                0/3/1:1 33333      g711ulaw  4 777
33333777/33333
1 active call found

ISR4451# show voice trunk-conditioning supervisory
FAST SCAN
0/3/1 : state : TRUNK_SC_CONN_DEFAULT_OOS, voice : off , signal : on ,master
status: lost keepalive, trunk connected
sequence oos : idle and oos
pattern :rx_idle = 0000 rx_oos = 1111
timeout timing : idle = 0, idle_off = 0, restart = 120, standby = 0, timeout = 30
supp_all = 0, supp_voice = 0, keep_alive = 5
timer: oos_ais_timer = 46, timer = 43
```

```
ISR4451# show voice trunk-conditioning signaling

0/3/1 :
hardware-state ACTIVE signal type is NorthamericanCAS
status : lost keepalive,
forced playout pattern = 0xF
idle monitoring : disabled
tx_idle = FALSE, rx_idle = FALSE, tx_oos = FALSE, lost_keepalive = TRUE
trunk_down_timer = 0, rx_ais_duration = 0, idle_timer = 0,tx_oos_timer = 0
```

要驗證IP到模擬複製，首先檢查新的IOS-XE命令：

```
ISR4451# show platform hardware qfp active feature sbc hootie group
SBC Hootie structure :
-------------------------------------

  VRF                               = 0
  IP                                = 239.X.X.X
  Port                              = 21000
  Protocol                          = 1
  Calls in group                    = 1

SBC Hootie group Statistics
-------------------------------------

  Total RTP packets received              = 2873
  Total RTP octects received              = 573520
  Total RTP packets replicated            = 2873
  Total RTP octects replicated            = 573520
  Total RTP packets dropped         = 0
  Total RTP octects dropped         = 0


ISR4451# show platform hardware qfp active feature sbc hootie group
SBC Hootie structure :
-------------------------------------

  VRF                               = 0
  IP                                = 239.X.X.X
  Port                              = 21000
  Protocol                          = 1
  Calls in group                    = 1

SBC Hootie group Statistics
-------------------------------------

  Total RTP packets received              = 3111
  Total RTP octects received              = 621032
  Total RTP packets replicated            = 3111
  Total RTP octects replicated            = 621032
  Total RTP packets dropped         = 0
  Total RTP octects dropped         = 0
```

多點傳送驗證

驗證PIM鄰居：

```
ISR4451# show ip pim neighbor
PIM Neighbor Table
Mode: B - Bidir Capable, DR - Designated Router, N - Default DR Priority,
      P - Proxy Capable, S - State Refresh Capable, G - GenID Capable,
      L - DR Load-balancing Capable
Neighbor          Interface              Uptime/Expires    Ver   DR
Address                                                          Prio/Mode
Y.Y.Y.Y       GigabitEthernet0/0/1       00:20:13/00:01:41 v2    1 / DR S P G
```

驗證mroute輸出是否正確：

```
ISR4451# show ip mroute
[snip]

(192.0.2.1, 239.X.X.X), 00:01:08/00:02:20, flags: FT
  Incoming interface: Vif1, RPF nbr 0.0.0.0
  Outgoing interface list:
    GigabitEthernet0/0/1, Forward/Sparse, 00:01:08/00:03:19
```

驗證清單中是否有多點傳送RP:

```
ISR4451# show ip igmp member
Flags: A  - aggregate, T - tracked
       L  - Local, S - static, V - virtual, R - Reported through v3
       I - v3lite, U - Urd, M - SSM (S,G) channel
       1,2,3 - The version of IGMP, the group is in
Channel/Group-Flags:
       / - Filtering entry (Exclude mode (S,G), Include mode (G))
Reporter:
       <mac-or-ip-address> - last reporter if group is not explicitly tracked
       <n>/<m>      - <n> reporter in include mode, <m> reporter in exclude

 Channel/Group              Reporter       Uptime   Exp.  Flags  Interface
 *,239.X.X.X                192.0.2.2    00:01:16 01:43 2VA    Vi1
```

驗證組播資料包複製：

```
RP# show ip mroute count
[snip]
Group: 239.X.X.X, Source count: 1, Packets forwarded: 2107, Packets received: 2108
  RP-tree: Forwarding: 2/0/56/0, Other: 2/0/0
  Source: 192.168.19.1/32, Forwarding: 2105/50/158/80, Other: 2106/0/1

RP# show ip mroute count
[snip]
Group: 239.X.X.X, Source count: 1, Packets forwarded: 2190, Packets received: 2191
  RP-tree: Forwarding: 2/0/56/0, Other: 2/0/0
  Source: 192.168.19.1/32, Forwarding: 2188/50/159/80, Other: 2189/0/1
```

[Cisco CLI Analyzer（僅供已註冊客戶使用）支援某些](#) show 指令。使用 Cisco CLI Analyzer 檢視 show 指令輸出的分析。

疑難排解

本節提供的資訊可用於對組態進行疑難排解。

呼叫設定問題

 如果連線未建立，請首先透過以下偵錯驗證訊號：


```
debug vpm signal
debug voip vtsp session
debug voip ccapi inout
```
調試示例：


```
123165: Oct  XX 13:21:55.563: htsp_process_event: [0/3/1, S_DOWN, E_HTSP_IF_INSERVICE]
123166: Oct  XX 13:21:55.564: %LINK-3-UPDOWN: Interface recEive and transMit 0/3/1, changed
state to up
123167: Oct  XX 13:21:55.564: recEive and transMit 0/3/1 rx_signal_map:
 0 0 0 0
 0 0 0 0
 8 8 8 8
 8 8 8 8
123168: Oct  XX 13:21:55.564: recEive and transMit 0/3/1 tx_signal_map:
 0 0 0 0
 0 0 0 0
 C C C C
 C C C C
123169: Oct  XX 13:21:55.564: htsp_process_event: [0/3/1, S_OPEN_PEND,
E_HTSP_GO_TRUNK]em_trunk_null_init
123170: Oct  XX 13:21:55.564: flex_set_Legerity_impedance: [0/3/1] impedance = 0
123171: Oct  XX 13:21:55.704: htsp_process_event: [0/3/1, S_TRUNK_NULL,
E_HTSP_INSERVE]default_trunk_down
123172: Oct  XX 13:21:55.704: htsp_timer - 6204 msec
123173: Oct  XX 13:21:55.919: %SYS-5-CONFIG_I: Configured from console by vty3 (192.168.19.2)
123174: Oct  XX 13:22:01.908: htsp_process_event: [0/3/1, S_TRUNK_PEND, E_HTSP_EVENT_TIMER]
123175: Oct  XX 13:22:01.908: htsp_timer_stop htsp_setup_ind
123176: Oct  XX 13:22:01.908: [0/3/1] get_local_station_id calling num= calling name= calling
time=10/08 13:22  orig called=
123177: Oct  XX 13:22:01.908: htsp_timer - 2000 msec

123181: Oct  XX 13:22:01.909: //-1/80F08D0180E8/CCAPI/cc_api_call_setup_ind_common:
   Interface=0x7F7475CF8C08, Call Info(
   Calling Number=,(Calling Name=)(TON=Unknown, NPI=Unknown, Screening=Not Screened,
Presentation=Allowed),
   Called Number=33333(TON=Unknown, NPI=Unknown),
   Calling Translated=FALSE, Subscriber Type Str=RegularLine, FinalDestinationFlag=TRUE,
   Incoming Dial-peer=777, Progress Indication=ORIGINATING SIDE IS NON ISDN(3), Calling IE
Present=FALSE,
   Source Trkgrp Route Label=, Target Trkgrp Route Label=, CLID Transparent=FALSE), Call Id=-1

123203: Oct  XX 13:22:01.911: //115/80F08D0180E8/CCAPI/ccCallSetupRequest:
   Calling Number=(TON=Unknown, NPI=Unknown, Screening=Not Screened, Presentation=Allowed),
   Called Number=33333(TON=Unknown, NPI=Unknown),
   Redirect Number=, Display Info=
   Account Number=, Final Destination Flag=TRUE,
   Guid=80F08D01-CA55-11E8-80E8-8E0AC3C8E4C4, Outgoing Dial-peer=33333

123252: Oct  XX 13:22:01.914: //116/80F08D0180E8/CCAPI/cc_api_caps_ack:
   Destination Interface=0x7F7475CF8C08, Destination Call Id=115, Source Call Id=116,
   Caps(Codec=g711ulaw(0x1), Fax Rate=FAX_RATE_VOICE(0x2), Fax Version:=0, Vad=AGGRESSIVE(0x4),
   Modem=OFF(0x0), Codec Bytes=160, Signal Type=2, Seq Num Start=2165)
123253: Oct  XX 13:22:01.914: //115/80F08D0180E8/CCAPI/cc_api_caps_ack:
   Destination Interface=0x7F7471175B68, Destination Call Id=116, Source Call Id=115,
   Caps(Codec=g711ulaw(0x1), Fax Rate=FAX_RATE_VOICE(0x2), Fax Version:=0, Vad=AGGRESSIVE(0x4),
   Modem=OFF(0x0), Codec Bytes=160, Signal Type=2, Seq Num Start=2165)
```

```
123255: Oct XX 13:22:01.914: //115/80F08D0180E8/VTSP:(0/3/1):-1:1:1/vtsp_call_connect: Connected
Name
123256: Oct  XX 13:22:01.914: //115/80F08D0180E8/VTSP:(0/3/1):-1:1:1/vtsp_call_connect:
Connected Number 33333
123257: Oct  XX 13:22:01.914: //115/80F08D0180E8/VTSP:(0/3/1):-1:1:1/vtsp_call_connect:
Connected oct3a  0
123258: Oct  XX 13:22:01.914: //115/80F08D0180E8/CCAPI/ccCallConnect:
   Call Entry(Connected=TRUE, Responsed=TRUE)


123265: Oct  XX 13:22:01.916: htsp_process_event: [0/3/1, S_TRUNK_W_CUTTHRU,
E_HTSP_VOICE_CUT_THROUGH]
123266: Oct  XX 13:22:01.916: send_trunk_dsp_voice_chnl_mapping:[0/3/1], 1/0/0
123267: Oct  XX 13:22:01.916: send_trunk_dsp_sig_chnl_mapping:[0/3/1], 129/0/0
123268: Oct  XX 13:22:01.916: recEive and transMit 0/3/1 rx_signal_map:
 0 0 0 0
 0 0 0 0
 0 0 0 0
 0 0 0 8 default_trunk_up
123269: Oct  XX 13:22:01.916: recEive and transMit 0/3/1 tx_signal_map:
 0 0 0 0
 0 0 0 0
 F F F F
 F F F F default_trunk_updefault_trunk_up
123270: Oct  XX 13:22:01.916: recEive and transMit 0/3/1 rx_signal_map:
 0 0 0 0
 0 0 0 0
 0 0 0 0
 0 0 0 8 default_trunk_up
123271: Oct  XX 13:22:01.916: recEive and transMit 0/3/1 tx_signal_map:
 0 0 0 0
 0 0 0 0
 F F F F
 F F F F default_trunk_up
123272: Oct XX 13:22:01.916: %HTSP-5-UPDOWN: Trunk port(channel) [0/3/1] is up
```

如果您看到此錯誤，是因為撥號對等體上的session protocol multicast命令不可用。


```
%VOICE_IEC-3-GW: H323: Internal Error (H225 chn, sock fail in RAS): IEC=1.1.186.5.81.0
```

音訊問題

如果問題出在無音訊上，請驗證語音網關是否正確加入組播組。有關工作裝置的基線輸出，請參閱本文檔驗證部分中的命令輸出。特定組播組的show ip mroute命令的傳出介面決不能為Null。如果您看到Null傳出介面，請檢視組播LAN的適用網路配置，因為這表明語音網關無法正確加入組播組。

Null傳出介面示例：


```
Router# show ip mroute 239.X.X.X
(*, 239.X.X.X), 00:22:02/stopped, RP 10.188.0.1, flags: SJCF
  Incoming interface: GigabitEthernet0/0/1, RPF nbr X.X.X.X
  Outgoing interface list:
    Vif1, Forward/Sparse-Dense, 00:18:27/00:02:32

(A.B.C.D, 239.X.X.X), 00:20:34/00:01:23, flags: PFT
  Incoming interface: Vif1, RPF nbr 0.0.0.0
  Outgoing interface list: Null
```
如果裝置在組播組中正確但音訊問題依然存在，請使用命令show platform hardware qfp active

feature sbc hootie group數次驗證裝置是否能夠接收和複製資料包。每次運行該命令時，計數器必須增加。或者，也可以運行show platform hardware qfp active statistics drop命令以檢視語音網關是否丟棄流量。若要清除這些計數器，請運行show platform hardware qfp active statistics drop clear命令。

如果沒有設定**IP multicast-routing**，則Ipv4mcNoRoute增量的捨棄原因如下：

```
4451# show platform hardware qfp active statistics drop
-----------------------------------------------------------------------
Global Drop Stats                        Packets              Octets
-----------------------------------------------------------------------
Ipv4mcNoRoute                               728                145272
```

其他音訊問題也可能發生，例如網關無法將模擬端接收的組播RTP資料包複製到IP端，因為組播配置有問題。當觀察到這些丟棄時，這些問題可能表現為丟棄原因FIAError。觀察到這些配置時，請檢查適用的組播配置，確保網關可以正確加入組播組，並且**show ip mroute** 命令具有有效的輸出介面。請參閱本文檔的多播部分，瞭解基線命令輸出。

```
4451# show platform hardware qfp active statistics drop
-----------------------------------------------------------------------
Global Drop Stats                        Packets              Octets
-----------------------------------------------------------------------
FIAError                                    724                144800
```
如果未啟用組播路由，則show ip mroute states的輸出也如此。

```
ISR4451# sh ip mroute
IP Multicast Forwarding is not enabled.
[snip]
```

PCM捕獲

為了驗證是否在語音埠上傳送或接收模擬音訊，您可以捕獲PCM。[完整PCM文檔](#)

```
conf t
voice pcm capture buffer 200000
voice pcm capture destination bootflash:
exit
!
test voice port 0/1/0 pcm-dump caplog ffffff duration 255
! send audio test voice port 0/1/0 pcm-dump disable ! copy flash:/<filename>.dat
[ftp://user:pass@ip.address/filename.pcap | tftp://a.b.c.d/filename] ! TAC is required to decode
the binary .dat file into SIN/SOUT/RIN audio streams
```

封包擷取(PCAP)

為了驗證是否傳送或接收了組播RTP，您可以在物理介面上執行資料包捕獲(PCAP)。完整[的EPC文檔。](#)

```
! NOT IN CONFIGURATION TERMINAL monitor capture TAC int gig0/0/1 both monitor capture TAC match
any ! monitor capture TAC start ! send audio monitor capture TAC stop ! monitor capture TAC
export [flash:/filename.pcap | ftp://user:pass@ip.address/filename.pcap |
tftp://a.b.c.d/filename] ! monitor capture TAC clear
```

DSP測試音

如果需要，語音網關上的DSP/PVDM可以在所需方向（網路 — IP-LAN端或本地模擬埠端）生成測試音。

此音調可以定向到DSP以指向IP LAN組播地址。這些命令可用於啟用/禁用。連線必須處於活動狀態，並且必須指定用於測試的模擬埠。

```
test voice port 0/1/0 inject-tone network 1000
! A 1000hz tone is now being generated from the analog port to the IP LAN Multicast Address test
voice port 0/1/0 inject-tone network disable
```

為了從DSP輸出模擬埠生成音調，可以使用這些命令來啟用/禁用。連線必須處於活動狀態，並且必須為測試指定模擬埠。

```
test voice port 0/1/0 inject-tone local 1000
! A 1000hz tone is now being generated out of the analog port. test voice port 0/1/0 inject-tone
local disable
```
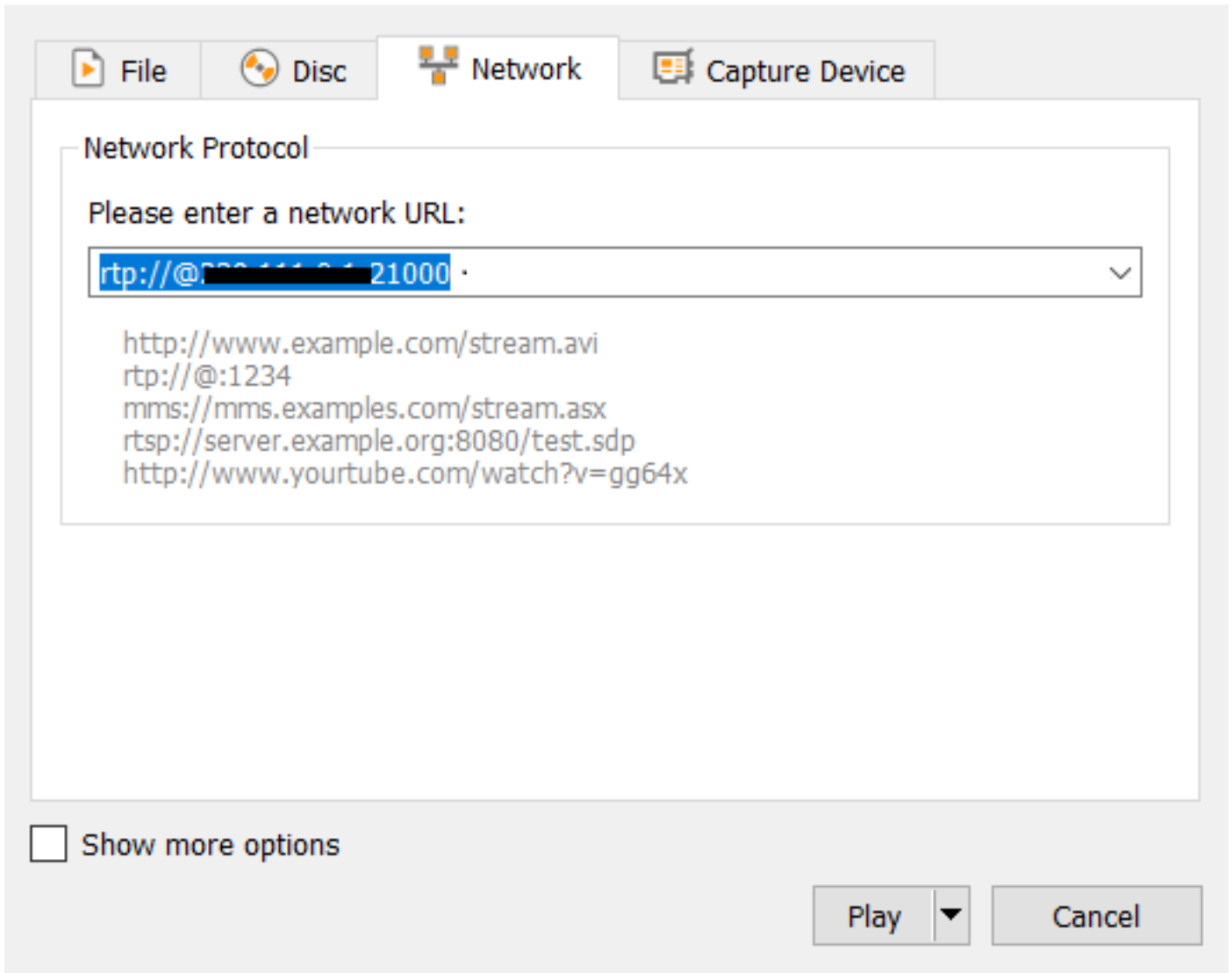
使用VLC媒體播放器測試組播接收

下載VLC Media Player並導航到**Media > Open Network Stream**
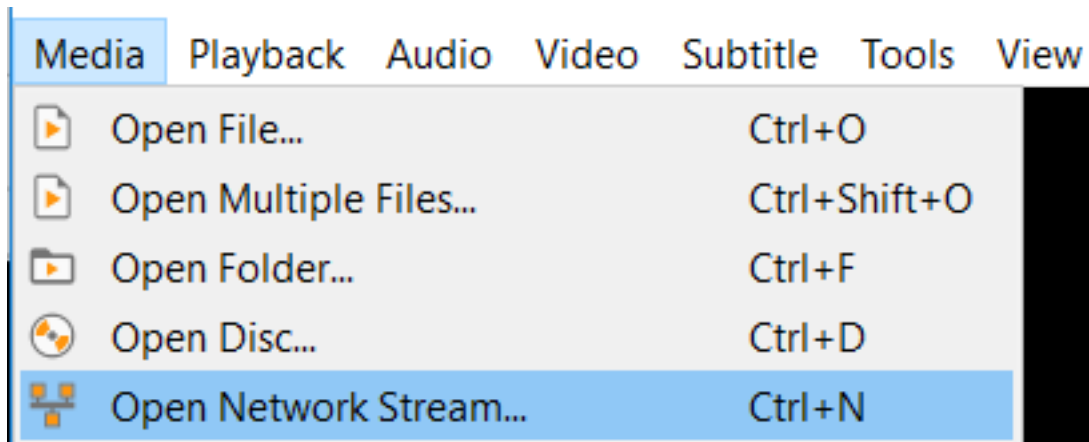
按此格式輸入組播RTP IP地址並按播放

```
rtp://@239.X.X.X:21000
```

然後下載並開啟Wireshark。然後選擇資料包捕獲所需的特定介面。

使用rtp過濾器開始捕獲。

如果一切順利，您必須加入組播RP。（可以從RP運行相同的組播命令以驗證加入組播組的PC）。

通過tone命令生成音訊或讓模擬終端發言。

現在，您必須在wireshark中看到資料包。請記住，來源IP必須是VIF IP減去1，因此我們的測試必須是192.0.2.2 -1 = 192.0.2.1。

No.	Time	Source	Destination	Destination Port	Protocol	Length	Info
33	14:08:31.960373	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3718, Time=669534125, Mark
34	14:08:31.980461	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3719, Time=669534285
35	14:08:32.000448	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3720, Time=669534445
36	14:08:32.020594	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3721, Time=669534605
37	14:08:32.040123	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3722, Time=669534765
38	14:08:32.060368	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3723, Time=669534925
39	14:08:32.080459	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3724, Time=669535085
40	14:08:32.100577	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3725, Time=669535245
42	14:08:32.120098	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3726, Time=669535405
43	14:08:32.140343	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3727, Time=669535565
44	14:08:32.160470	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3728, Time=669535725
45	14:08:32.180532	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3729, Time=669535885
46	14:08:32.200625	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3730, Time=669536045
47	14:08:32.220073	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3731, Time=669536205
48	14:08:32.240231	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3732, Time=669536365
49	14:08:32.260346	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3733, Time=669536525
50	14:08:32.280352	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3734, Time=669536685
51	14:08:32.300434	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3735, Time=669536845
52	14:08:32.320509	192.168.19.1	230.111.8.1	21000	RTP	214	PT=ITU-T G.711 PCMU, SSRC=0x79D4, Seq=3736, Time=669537005

Cisco CLI Analyzer（僅供已註冊客戶使用）支援某些 show 指令。使用 Cisco CLI Analyzer 檢視 show 指令輸出的分析。

附註：使用 debug 指令之前，請先參閱有關 Debug 指令的重要資訊。

相關資訊

- 已知瑕疵

CSCvd18792 - ISR4K - Hoot和Holler E&M埠不能與組播集線器共置

CSCve66876 - ISR4K — 為來自DSP的資料包丟棄組播RP註冊

CSCve71893 - ISR4K - Hoot和Holler組播複製問題

- 技術支援與檔案 — Cisco System