# 適用於行動代理的CUCM網路型記錄

## 目錄

# 簡介

本檔案介紹網路型錄音(NBR)的不同案例，以及疑難排解。

# 必要條件

## 需求

思科建議您瞭解以下主題：

- 思科統一通訊管理器(CUCM)版本10.0(1)或更高版本
- 基於電話的記錄體系結構
- 基於網路的記錄體系結構

## 採用元件

本文中的資訊係根據以下軟體和硬體版本：

- 思科Call Manager版本10.5
- 客戶語音入口網站(CVP)版本10.5
- Cisco整合客服中心Express版(UCCE)10.5(2)
- 閘道3925E 15.3(3)M

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。如果您的網路正在作用，請確保您已瞭解任何指令可能造成的影響。

# 背景資訊

自CUCM 10.0(1)版起，網路記錄可用，並允許您使用網關記錄呼叫。

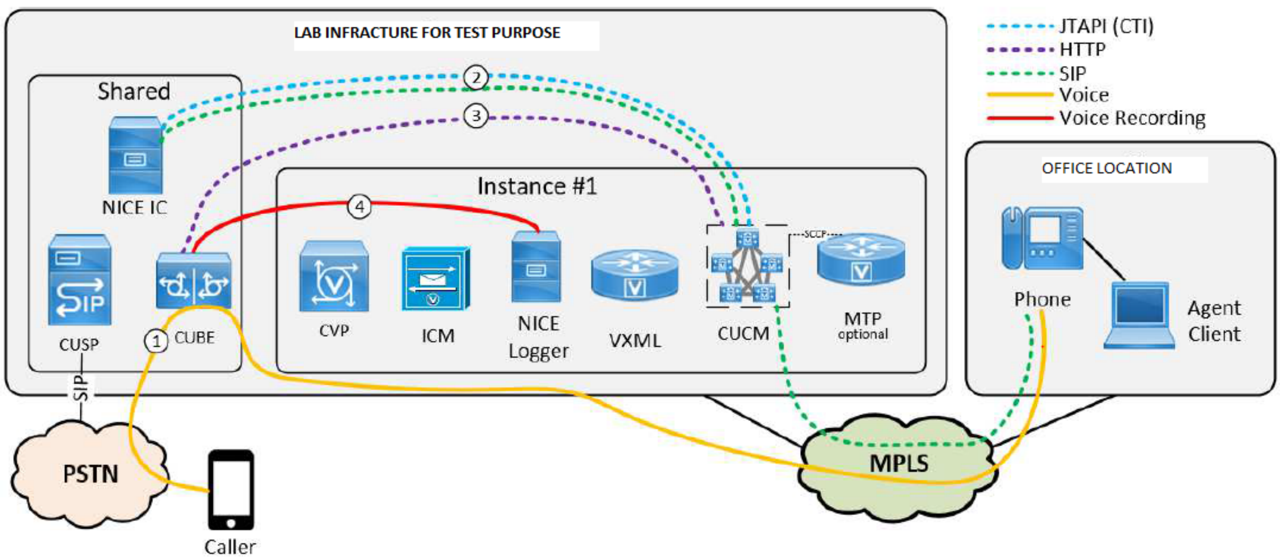
該功能允許記錄呼叫，而不考慮裝置、位置或地理位置，例如從網外擴展到移動和家庭辦公室電話的呼叫。它根據呼叫流程和呼叫參與者自動選擇正確的媒體源。

需要瞭解的是：

- SIP信令是從CUCM到CUBE以及從CUCM到錄制伺服器。
- 錄音伺服器和CUBE之間沒有直接SIP信令。
- CUBE負責將RTP流分叉到錄制伺服器。
- CUCM上記錄的終結點不需要支援內建網橋(BiB)。

CUCM使用HTTP向CUBE上的Cisco Unified Communications(UC)Services API發起呼叫記錄請求

。Cisco Unified Communications(UC)Services API為IOS網關中的不同服務提供統一Web服務介面。其中一項服務是擴展媒體分流(XMF)提供程式，它允許應用程式監控呼叫並在即時傳輸協定(RTP)和安全RTP呼叫上觸發媒體分流。




## 移動座席的工作方式

1. 通訊管理器express(CME)上的呼叫方A撥打B，該呼叫方指向網關(GW)。GW撥號對等點指向客戶語音入口網站(CVP)。
2. CVP向智慧聯絡管理器(ICM)傳送路由請求，ICM返回移動代理標籤，即本地CTI埠（LCP埠）撥號號碼(DN)。
3. CVP向CUCM傳送邀請。當LCP埠振鈴時，JTAPI網關(JGW)指示CUCM從遠端CTI埠(RCP)DN呼叫代理電話。
4. 座席應答後，座席分支將連線到通話等待音樂(MoH)。
5. JGW指示CUCM應答在LCP埠上振鈴的入站呼叫。
6. 連線LCP支路後，JGW會指示CUCM檢索代理支路。
7. JGW將即時傳輸協定(RTP)IP地址/埠詳細資訊從客戶支路傳遞到代理支路，反之亦然。
8. CUCM橋接兩個分支，並在座席和客戶之間建立RTP路徑。

## 在移動代理的情況下錄音的運作方式

- 在移動代理的情況下，可以在LCP埠或RCP埠上啟用錄製。
- 在LCP或RCP上連線呼叫並啟用錄製後，CUCM會向錄制伺服器傳送針對近端和遠端裝置的2個Invite。

- 完成近端裝置的信令後，將遠端裝置SDL HTTP請求傳送到網關以指示其開始錄製。

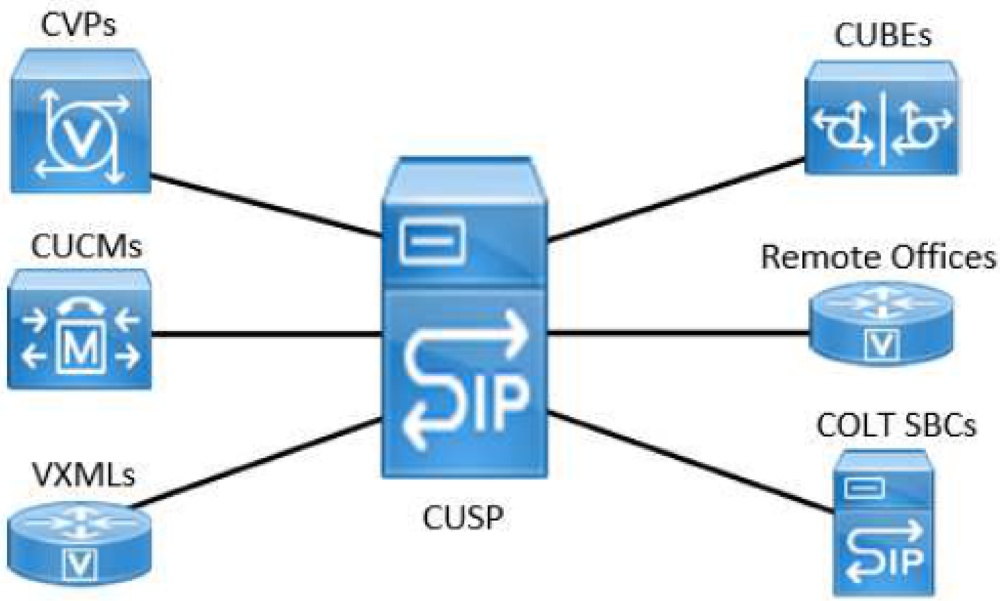
註：有些情況下，CUCM沒有直接與網關或CVP建立SIP中繼

注意：例如，CUCM可以有一個SIP中繼，其中有一個代理伺服器(CUSP)控制所有流量

注意：如果在CTI埠上啟用錄製並且呼叫在該埠上接通，錄製將正常工作。

注意：在移動代理的情況下，CTI埠確實會促進訊號傳送，然後會退出RTP流。這是RTP流經的終點。但是LCP和RCP埠永遠不會離開信令。他們的CI在通話結束前不會被銷毀。這就是即使RTP沒有流過LCP或RCP埠時，在LCP或RCP埠上記錄成功的原因

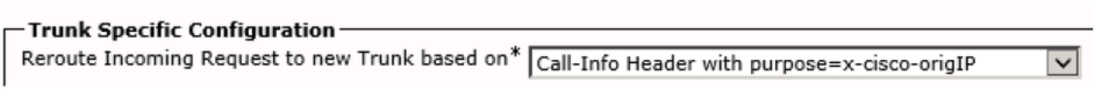
## 使用CUSP的UCCE部署（代理伺服器）




UCCE與CVP和CUSP一起部署時，使用所謂的綜合模型，CUCM和CUBE之間沒有SIP中繼。CUBE和CUCM之間的所有通訊都通過一個SIP中繼到CUSP。

CUCM需要知道呼叫來自哪個CUBE，以便知道將錄製請求傳送到何處。這是通過將請求傳送回用於呼叫的傳入SIP中繼的目標IP來實現的。但是，如果CUCM將API請求傳送回CUSP，則不會發生任何情況。要在使用CUSP的環境中解決此限制，需要實施以下CUCM配置：

- 建立到每個CUBE的虛擬SIP中繼。此中繼不會用於路由任何呼叫！
- 使用Call-Info報頭將CUSP SIP中繼上的傳入呼叫重新分類到正確的虛擬CUBE中繼。




註：此設定不影響任何呼叫處理決策 — 所有呼叫處理和呼叫類別服務決策都將完成，就像呼叫仍在CUSP SIP中繼上一樣，並且不會將任何SIP消息傳送到新匹配中繼的目的地。

註：傳入INVITE中的x-cisco-origIP值必須與目標IP地址匹配虛設中繼。

註：要為x-cisco-origIP報頭設定正確的值，必須在源多維資料集上正確設定該報頭。設定該值可以通過在CUBE上新增報頭來實現，也可以在CVP上新增報頭來實現。UCCE Direct代理指令碼已用於Call-Info報頭。因此，第二個具有所需x-cisco-origIP的Call-Info報頭將新增到直接代理指令碼的Call-Info報頭之後。測試表明，當x-cisco-origIP包含在SIP INVITE的第二個呼叫資訊報頭中時，CUCM仍會執行所需的重新分類。

# 組態

採用CUSP的UCCE部署的關鍵配置點：

## 為錄製器建立SIP中繼裝置

要將記錄器配置為SIP中繼裝置，Unified CM管理員從裝置頁面建立SIP中繼裝置，並在Destination Address欄位中輸入裝置名稱和記錄器的IP地址。

## 建立呼叫記錄配置檔案

要設定座席的線路外觀以進行呼叫記錄，應建立一個或多個呼叫記錄配置檔案。然後，為線路外觀選擇記錄配置檔案。要建立錄音配置檔案，Unified CM管理員開啟「裝置設定」頁面，然後選擇「呼叫錄音配置檔案」。在「Recording Destination Address」欄位中，管理員輸入錄製器的DN或URL。在Recording Calling Search Space欄位中，管理員輸入為錄製器配置的SIP中繼的分割槽。

## 向每個CUBE調配虛擬SIP中繼

對於需要將呼叫分叉到記錄伺服器的每個網關，必須在CUCM上配置專用虛擬中繼。請記住，此中繼不用於任何實際SIP信令，並且不影響任何呼叫決策。需要配置的重要事項包括：

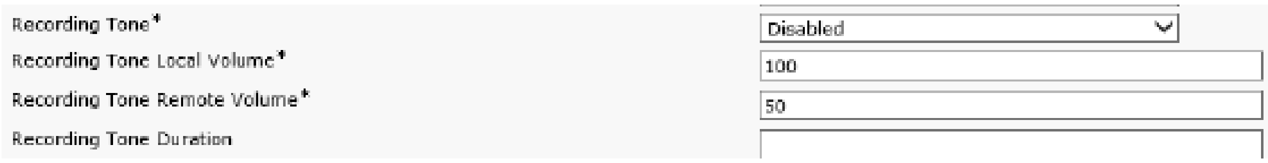
- 此中繼連線到啟用錄音的網關。
- 目標IP必須與CUBE配置為在其XMF配置中偵聽的目標IP相同

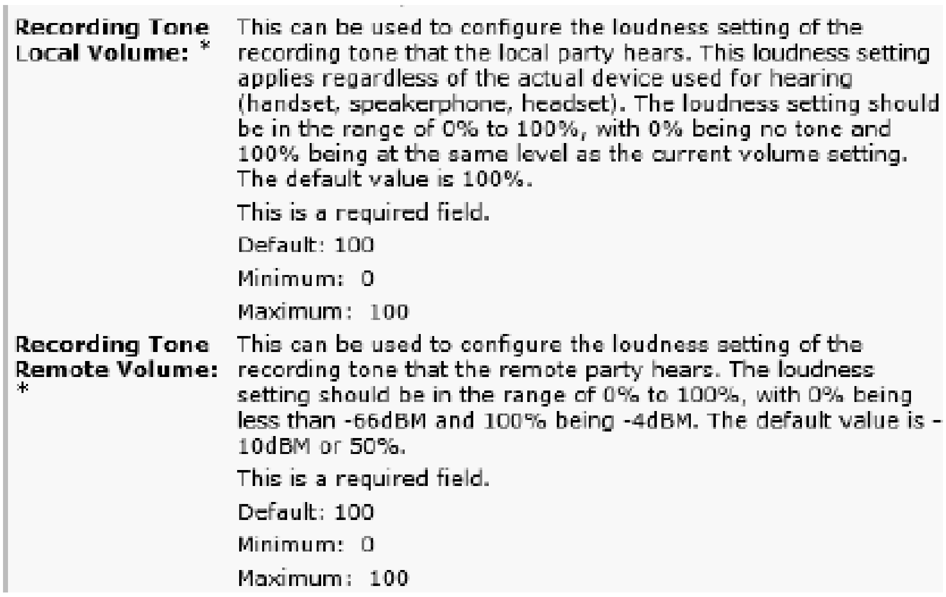
## 為記錄器設定路由模式

要設定記錄器的路由模式，管理員開啟路由模式配置頁，然後根據記錄器DN輸入路由模式。管理員為記錄器選擇SIP中繼裝置，然後儲存路由模式。如果記錄器地址指定為SIP URL，且URL的RHS不屬於Unified CM集群，則應配置SIP路由模式。模式欄位應為記錄器的域或ip地址（記錄器URL的RHS部分），SIP Trunk欄位應為記錄器的SIP中繼。

## 設定錄音呼叫通知音選項

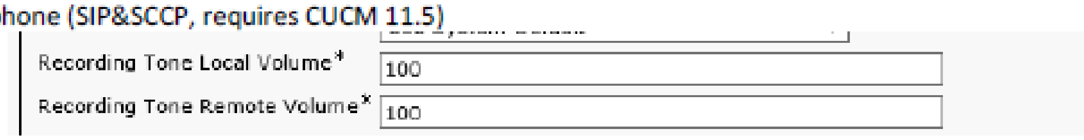
要設定錄製通知音的集群範圍服務引數，管理員開啟Unified CM管理的「服務引數」頁，並找到Play Recording Notification Tone到Observed Target的條目。管理員輸入Yes或No。然後，管理員會查詢已觀察到的已連線目標的播放錄音通知音的條目。管理員輸入Yes或No。



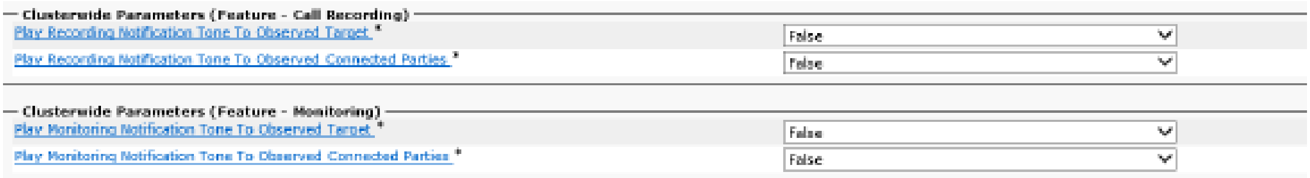




US: Tone=Enabled; Local Volume = 0 ; Remote Volume= 1

Softphone (SIP&SCCP, requires CUCM 11.5)




Service Parameter



**Clusterwide Parameters (Feature - Call Recording)**

| | |
|---|---|
| **Play Recording Notification Tone To Observed Target:** * | This parameter specifies whether to enable the Recording Tone will be played to the Observed Target. Valid values specify False (no tones) or True (tone is played). The system uses this parameter during the initiation of Recording Feature to determine whether the tone will be played. Changes in this parameter will not affect currently registered devices. To get changes of this parameter to currently registered devices, the devices have to be restarted.<br>This is a required field.<br>Default: False |
| **Play Recording Notification Tone To Observed Connected Parties:** * | This parameter specifies whether to enable the Recording Tone will be played to the Observed Connected Parties. Valid values specify False (no tones) or True (tone is played). The system uses this parameter during the initiation of Recording Feature to determine whether the tone will be played. Changes in this parameter will not affect currently registered devices. To get changes of this parameter to currently registered devices, the devices have to be restarted.<br>This is a required field.<br>Default: False |

## 設定CUBE XMF提供程式

這些配置啟用HTTP通訊和XMF提供程式配置：

CUBE001 :

ip http server
no ip http secure-server
ip http max-connections 1000
ip http timeout-policy idle 600 life 86400 requests 86400
ip http client source-interface Port-channel20.307
uc wsapi
message-exchange max-failures 2
source-address 10.106.230.20
探測間隔keepalive 5
探測max-failures 5
!
提供商xmf
remote-url 1 http://10.106.97.140:8090/ucm_xmf
remote-url 2 http://10.106.97.141:8090/ucm_xmf
remote-url 3 http://10.106.97.143:8090/ucm_xmf
remote-url 4 http://10.106.97.144:8090/ucm_xmf

CUBE002:

ip http server

no ip http secure-server
ip http max-connections 1000
ip http timeout-policy idle 600 life 86400 requests 86400
ip http client source-interface Port-channel20.307
uc wsapi
message-exchange max-failures 2
source-address 10.106.230.20
探測間隔keepalive 5
探測max-failures 5
!
提供商xmf
remote-url 1 http://10.106.97.140:8090/ucm_xmf
remote-url 2 http://10.106.97.141:8090/ucm_xmf
remote-url 3 http://10.106.97.143:8090/ucm_xmf
remote-url 4 http://10.106.97.144:8090/ucm_xmf

| Parameter | Description |
| --- | --- |
| ip http client source-interface | set to match the uc wsapi source address |
| ip http max-connections 1000 | please set accordingly with the expected calls |
| source-address x.x.x.x | This is the IP Address to which the CUCM sends the http XMF messages. This IP Address must match the destination IP in the CUCM SIP Trunk configuration for the "dummy" CUBE. |
| probing interval keepalive 5 | note that any other message sent by the gateway will be treated as a keepalive |
| probing interval negative 5 | default value, shown for completeness |
| Remote-url | call processing servers, max 32 entries |

## 為呼叫資訊報頭調配CUBE SIP配置檔案

要為x-cisco-origIP報頭設定正確的值，必須注意在源CUBE上正確設定它。可以通過多種方式設定該值，而且無需在CUBE上設定，例如，也可以在CVP上設定。 這是一個示例SIP配置檔案，它將傳出INVITE from CUBE中的x-cisco-origIP值靜態設定為CUSP。

—
語音類sip配置檔案666
請求INVITE sip-header Call-Info add "Call-Info: <sip:10.106.242.27>;PURPOSE=x-cisco-origIP"
—

如果UCCE系統已依賴呼叫資訊報頭，則使用具有所需xcisco-origIP的第二個呼叫資訊報頭。測試表明，當x-cisco-origIP包含在SIP INVITE的第二個呼叫資訊報頭中時，CUCM仍會執行所需的重新分類。同樣的測試也顯示，如果把新的呼叫資訊報頭放在首位，其他系統將會停止工作。該配置檔案需要應用於指向CUSP的出站撥號對等體。

如需詳細組態，請參閱以下連結：

# 疑難排解

## 日誌分析

### 來自客戶語音門戶(CVP)的傳入邀請

```
01382866.006 |12:52:49.858 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.106.97.
[105066,NET]
INVITE sip:9876@eu91.voip.test SIP/2.0
Via: SIP/2.0/TCP 10.106.97.135:5060;branch=z9hG4bKc7z5eWQrKkRtP5FKnbAb6w~~780271
Via: SIP/2.0/TCP 10.106.97.136:5062;branch=z9hG4bKhYyfmvtY8.fM7CSyQd9K4Q~~48611
Max-Forwards: 63
Record-Route: <sip:rr$n=cvp@10.106.97.135:5060;transport=tcp;lr>
To: <sip:9876@CVP001.eu91.lab.test;transport=tcp>
From: +1234567890 <sip:+1234567890@10.106.97.136:5062>;tag=dsf816dd0c
Contact: <sip:+1234567890@10.106.97.136:5062;transport=tcp>
Expires: 60
Diversion: <sip:+123459876@10.106.97.137>;reason=unconditional;screen=yes;privacy=off
Call-ID: 694646BC1D2311E7A8D2826ACB31D85A-149182876973312598@10.106.97.136
CSeq: 1 INVITE
Content-Length: 250
User-Agent: CVP 10.5 (1) ES-18 Build-36
Date: Mon, 10 Apr 2017 12:52:38 GMT
Min-SE: 1800
Cisco-Guid: 1766213308-0488837607-2832368234-3409041498
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
Allow-Events: telephone-event
P-Asserted-Identity: <sip:+1234567890@10.106.97.138>
Session-Expires: 1800
Content-Disposition: session;handling=required
History-Info: <sip:\u95>
History-Info: <sip:\u95>
Call-Info: <sip:10.106.97.138>;purpose=x-cisco-origIP
Cisco-Gucid: 694646BC1D2311E7A8D2826ACB31D85A
Supported: timer
Supported: resource-priority
Supported: replaces
Supported: sdp-anat
Content-Type: application/sdp
App-Info: <10.106.97.136:8000:8443>

v=0
o=CiscoSystemsSIP-GW-UserAgent 2790 2026 IN IP4 10.106.97.138
s=SIP Call
c=IN IP4 10.106.242.1
t=0 0
m=audio 16552 RTP/AVP 8 101
c=IN IP4 10.106.242.1
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
```

### 來電的數字分析

```
01382890.009 |12:52:49.861 |AppInfo ||PretransformCallingPartyNumber=+1234567890
|CallingPartyNumber=+1234567890
|DialingPartition=SYS-DN-PlainE164-PT
|DialingPattern=9876
|FullyQualifiedCalledPartyNumber=9876
|DialingPatternRegularExpression=(9876)
|DialingWhere=
```

用於呼叫號和本地CTI埠(LCP)的呼叫識別符號(CI)關聯

```
01382897.001 |12:52:49.862 |AppInfo  |LBMIF: CI: 43358624 ASSOC    43358625
01382897.002 |12:52:49.862 |AppInfo  |LBMIF: CI: 43358625 ASSOC'   43358624
```

已選擇LCP

```
01382902.001 |12:52:49.862 |AppInfo  |LineCdpc(135): -dispatchToAllDevices-, sigName=CcSetupReq, device
01382905.002 |12:52:49.862 |AppInfo  |StationCdpc(59): StationCtiCdpc-CtiEnableReq CH=0|0 DevName=LCP_4
```

180振鈴已傳送到CVP

```
01382949.001 |12:52:49.865 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.106.97.
[105068,NET]
SIP/2.0 180 Ringing
Via: SIP/2.0/TCP 10.106.97.135:5060;branch=z9hG4bKc7z5eWQrKkRtP5FKnbAb6w~~780271,SIP/2.0/TCP 10.106.97.
From: +1234567890 <sip:+1234567890@10.106.97.136:5062>;tag=dsf816dd0c
To: <sip:9876@CVP001.eu91.lab.test;transport=tcp>;tag=46359~8c66ebf6-153f-456b-a6e8-0bf5f687ce1f-433586
Date: Mon, 10 Apr 2017 12:52:49 GMT
Call-ID: 694646BC1D2311E7A8D2826ACB31D85A-149182876973312598@10.106.97.136
CSeq: 1 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Record-Route: <sip:rr$n=cvp@10.106.97.135:5060;transport=tcp;lr>
Server: Cisco-CUCM10.5
Supported: X-cisco-srtp-fallback
Supported: Geolocation
P-Asserted-Identity: <sip:9876@10.107.28.14>
Remote-Party-ID: <sip:9876@10.107.28.14>;party=called;screen=yes;privacy=off
Contact: <sip:9876@10.107.28.14:5060;transport=tcp>
Content-Length: 0
```

RCP將呼叫擴展到被叫號碼

LCP和主叫號碼鈴聲和遠端CTI埠(RCP)將呼叫擴展到被叫號碼（即座席）。

```
01382957.000 |12:52:49.882 |SdlSig   |CtiEnableReq                           |null0
01382957.001 |12:52:49.882 |AppInfo  |StationCdpc(2,100,64,60): StationCtiCdpc::StationCtiCdpc
01382957.002 |12:52:49.882 |AppInfo  |StationCdpc(60): StationCtiCdpc-CtiEnableReq CH=0|0 DevName=RCP_4
01382958.000 |12:52:49.882 |SdlSig   |StationOutputSetRinger                 |restart0
01382958.001 |12:52:49.882 |AppInfo  |StationD:    (0000245) SetRinger ringMode=1(RingOff).
```

RCP呼叫代理的DIgit分析

```
01383005.013 |12:52:49.885 |AppInfo ||PretransformCallingPartyNumber=9876
|CallingPartyNumber=9876
|DialingPartition=TE-PSTNInternational-PT
|DialingPattern=+.[1-9]!
|FullyQualifiedCalledPartyNumber=+1122334455
|DialingPatternRegularExpression=(+)([1-9][0-9]+)
```

RCP和代理的呼叫識別符號(CI)關聯

```
01383012.001 |12:52:49.885 |AppInfo  |LBMIF: CI: 43358626 ASSOC    43358627
01383012.002 |12:52:49.885 |AppInfo  |LBMIF: CI: 43358627 ASSOC'   43358626
```

已傳送座席的邀請：

```
01383048.001 |12:52:49.888 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.241.242
[105069,NET]
INVITE sip:1122334455@10.106.22.199:5060 SIP/2.0
Via: SIP/2.0/TCP 10.107.28.14:5060;branch=z9hG4bK6b0870d07a53
From: <sip:9876@10.107.28.14>;tag=46360~8c66ebf6-153f-456b-a6e8-0bf5f687ce1f-43358627
To: <sip:1122334455@10.106.22.199>
Date: Mon, 10 Apr 2017 12:52:49 GMT
Call-ID: 98b4ac00-8eb18021-67f3-c2e4110a@10.107.28.14
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback,X-cisco-original-called
Call-Info: <sip:10.107.28.14:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=VIDEO_UNSPECIFIED
```

```
Cisco-Guid: 2561977344-0000065536-0000000138-3269726474
Session-Expires: 1800
P-Asserted-Identity: <sip:9876@10.107.28.14>
Remote-Party-ID: <sip:9876@10.107.28.14>;party=calling;screen=yes;privacy=off
Contact: <sip:9876@10.107.28.14:5060;transport=tcp>;DeviceName="RCP_47483708"
Max-Forwards: 70
Content-Length: 0
```

```
01383182.002 |12:53:00.624 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.106.22.
[105079,NET]
SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.107.28.14:5060;branch=z9hG4bK6b0870d07a53
Record-Route: <sip:rr$n=cube-pool-int@10.106.22.199:5060;transport=tcp;lr>
To: <sip:1122334455@10.106.22.199>;tag=AD1038-15B8
From: <sip:9876@10.107.28.14>;tag=46360~8c66ebf6-153f-456b-a6e8-0bf5f687ce1f-43358627
Contact: <sip:1122334455@10.106.97.138:5060;transport=tcp>
Require: timer
Remote-Party-ID: <sip:+1122334455@10.106.97.138>;party=called;screen=no;privacy=off
Call-ID: 98b4ac00-8eb18021-67f3-c2e4110a@10.107.28.14
CSeq: 101 INVITE
Content-Length: 250
Date: Mon, 10 Apr 2017 12:52:49 GMT
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
Allow-Events: telephone-event
Supported: replaces
Supported: sdp-anat
Supported: timer
Server: Cisco-SIPGateway/IOS-15.4.3.M5
Session-Expires: 1800;refresher=uac
Content-Type: application/sdp
Content-Disposition: session;handling=required

v=0
o=CiscoSystemsSIP-GW-UserAgent 6311 9012 IN IP4 10.106.97.138
s=SIP Call
c=IN IP4 10.106.242.1
t=0 0
m=audio 16554 RTP/AVP 8 101
c=IN IP4 10.106.242.1
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
```

### RCP處於保持狀態，LCP和主叫方已連線

```
01383470.004 |12:53:00.650 |AppInfo |StationD: (0000388) INFO- sendSignalNow, sigName=StationOffHook, c
01383471.000 |12:53:00.651 |SdlSig-O |CtiLineCallAnswerRes |NA RemoteSignal |UnknownProcessName(2,200,2
01383472.000 |12:53:00.651 |SdlSig |StationOutputSetRinger |restart0 |StationD(2,100,63,388) |StationD(
01383472.001 |12:53:00.651 |AppInfo |StationD: (0000388) SetRinger ringMode=1(RingOff).
```

### 呼叫方和LCP的媒體連線請求

```
01383497.001 |12:53:00.651 |AppInfo  |ARBTRY-ConnectionManager-wait_MediaConnectRequest(43358624,433586
01383497.002 |12:53:00.651 |AppInfo  |ARBTRY-ConnectionManager- storeMediaInfo(CI=43358624): ADD NEW EN
01383497.003 |12:53:00.651 |AppInfo  |ARBTRY-ConnectionManager- storeMediaInfo(CI=43358625): ADD NEW EN
```

媒體終端點(MTP)分配給LCP和呼叫方

```
01383508.002 |12:53:00.652 |AppInfo  |MediaResourceCdpc(185)::waiting_MrmAllocateMtpResourceReq - CI=43
```

在LCP埠上啟用錄製

```
01383607.002 |12:53:00.655 |AppInfo | StationCdpc: startRecordingIfNeeded - Device LCP_47483708, starte

01383614.016 |12:53:00.655 |AppInfo | StationCdpc: startRecordingIfNeeded - Device LCP_47483708, lockin
01383614.017 |12:53:00.655 |AppInfo | StationCdpc: star_MediaExchangeAgenaQueryCapability - Device LCP_
01383614.018 |12:53:00.655 |AppInfo | StationCdpc: startRecordingIfNeeded - Device LCP_47483708, starte
01383614.019 |12:53:00.655 |AppInfo |StatiopnCdpc::StartRecordingIfNeeded DeviceName =LCP_47483708 Reco
01383614.020 |12:53:00.655 |AppInfo | StationCdpc: startRecordingIfNeeded - Device LCP_47483708. FinalT
```

記錄啟動的簽名

```
01383640.003 |12:53:00.657 |AppInfo |RecordManager::- await_SsDataInd lParties=(43358624,43358625)
01383641.000 |12:53:00.657 |SdlSig |SsDataInd |await_recordingFeatureData |Recording(2,100,100,77) |Rec
01383641.001 |12:53:00.657 |AppInfo |Recording::- (0000077) -await_recordingFeatureData_SsDataInd: mRec
01383641.002 |12:53:00.657 |AppInfo |Recording::- (0000077) -await_recordingFeatureData_SsDataInd: Trig

01383645.001 |12:53:00.657 |AppInfo |Recording::- (0000077) -processGWPreferred ....
01383645.002 |12:53:00.657 |AppInfo |Recording::- (0000077) -getRecordingAnchorMode: PeerBib=[1];peerCM
01383645.003 |12:53:00.657 |AppInfo |Recording::- (0000077) -processGWPreferred: GW Recording - sideABi
```

內建橋接器的數字分析(Bib)

```
1383671.008 |12:53:00.658 |AppInfo ||PretransformCallingPartyNumber=
|CallingPartyNumber=
|DialingPartition=
|DialingPattern=b0026901001
|FullyQualifiedCalledPartyNumber=b0026901001
|DialingPatternRegularExpression=(b0026901001)
```

這裡SIPBIB建立SIPBIBCDPC錄製流程

```
01383681.000 |12:53:00.658 |SdlSig |CcSetupReq |restart0 |SIPvBIB(2,100,69,1) |Cdcc(2,100,219,295)
01383681.001 |12:53:00.658 |AppInfo |SIPvBIB::restart0_CcSetupReq: primCallCi=43358624 primCallBranch=0
01383682.000 |12:53:00.658 |SdlSig |CcSetupReq |restart0 |SIPvBIBCdpc(2,100,68,55) |SIPvBIB(2,100,69,1)
```

適用於LCP和呼叫方的200 OK

```
01383761.001 |12:53:00.668 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.106.97.
[105082,NET]
SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.106.97.135:5060;branch=z9hG4bKc7z5eWQrKkRtP5FKnbAb6w~~780271,SIP/2.0/TCP 10.106.97.
From: +1234567890 <sip:+1234567890@10.106.97.136:5062>;tag=dsf816dd0c
To: <sip:9876@CVP001.eu91.lab.test;transport=tcp>;tag=46359~8c66ebf6-153f-456b-a6e8-0bf5f687ce1f-433586
Date: Mon, 10 Apr 2017 12:52:49 GMT
Call-ID: 694646BC1D2311E7A8D2826ACB31D85A-149182876973312598@10.106.97.136
CSeq: 1 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence, kpml
Record-Route: <sip:rr$n=cvp@10.106.97.135:5060;transport=tcp;lr>
Supported: replaces
Server: Cisco-CUCM10.5
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Session-Expires: 1800;refresher=uas
Require: timer
P-Asserted-Identity: <sip:9876@10.107.28.14>
Remote-Party-ID: <sip:9876@10.107.28.14>;party=called;screen=yes;privacy=off
Contact: <sip:9876@10.107.28.14:5060;transport=tcp>;DeviceName="LCP_47483708"
Content-Type: application/sdp
Content-Length: 246

v=0
o=CiscoSystemsCCM-SIP 46359 1 IN IP4 10.107.28.14
s=SIP Call
c=IN IP4 10.17.229.27
b=TIAS:64000
b=CT:64
b=AS:64
t=0 0
m=audio 23304 RTP/AVP 8 101
a=ptime:20
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

錄製詳細資訊

此處錄製是首選網關：

```
01383780.001 |12:53:00.669 |AppInfo |Recording::- (0000077) -setMetaDataWithLocalPhoneOrGWForking: fork
01383780.002 |12:53:00.669 |AppInfo |Recording::- (0000077) -buildOtherParm: OtherParm=[x-nearend;x-ref
```

記錄編號的數字分析

```
01383793.012 |12:53:00.669 |AppInfo |Digit analysis: analysis results
01383793.013 |12:53:00.669 |AppInfo ||PretransformCallingPartyNumber=b0026901001
|CallingPartyNumber=b0026901001
|DialingPartition=SYS-NiceRecording-PT
|DialingPattern=123456789
|FullyQualifiedCalledPartyNumber=123456789
|DialingPatternRegularExpression=(123456789)
```

呼叫擴展至路由清單

```
01383807.001 |12:53:00.670 |AppInfo  |RouteListControl::idle_CcSetupReq - RouteList(NICERecording-01-RL
```

邀請已傳送到近端裝置的錄制伺服器

```
01383831.001 |12:53:00.671 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.17.230.
[105083,NET]
INVITE sip:123456789@10.17.230.4:5060 SIP/2.0
Via: SIP/2.0/TCP 10.107.28.14:5060;branch=z9hG4bK6b0d30bfa6ec
From: <sip:+1234567890@10.107.28.14;x-nearend;x-refci=43358625;x-nearendclusterid=eu91;x-nearenddevice=
To: <sip:123456789@10.17.230.4>
Date: Mon, 10 Apr 2017 12:53:00 GMT
Call-ID: 9f432380-8eb1802c-67f6-c2e4110a@10.107.28.14
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Cisco-Guid: 2671977344-0000065536-0000000139-3269726474
Session-Expires: 1800
P-Asserted-Identity: <sip:+1234567890@10.107.28.14>
Remote-Party-ID: <sip:+1234567890@10.107.28.14>;party=calling;screen=yes;privacy=off
Contact: <sip:+1234567890@10.107.28.14:5060;transport=tcp>;isFocus
Max-Forwards: 70
Content-Length: 0
```

從錄音伺服器收到200 OK

```
SIP/2.0 200 OK
From: <sip:+1234567890@10.107.28.14;x-nearend;x-refci=43358625;x-nearendclusterid=eu91;x-nearenddevice=
To: <sip:123456789@10.17.230.4>;tag=ea1fb60-0-13c4-5506-90037-9c2acf-90037
Call-ID: 9f432380-8eb1802c-67f6-c2e4110a@10.107.28.14
CSeq: 101 INVITE
Via: SIP/2.0/TCP 10.107.28.14:5060;branch=z9hG4bK6b0d30bfa6ec
Supported: timer
Contact: <sip:123456789@10.17.230.4:5060;transport=TCP>
Session-Expires: 1800;refresher=uas
Content-Type: application/sdp
Content-Length: 119

v=0
o=VRSP 0 0 IN IP4 127.0.0.1
s=NICE VRSP
c=IN IP4 127.0.0.1
t=0 0
m=audio 1000 RTP/AVP 0 4 8 9 18
a=recvonly


01383896.001 |12:53:00.673 |AppInfo |Recording::- (0000077) -setMetaDataWithLocalPhoneOrGWForking: fork
01383896.002 |12:53:00.673 |AppInfo |Recording::- (0000077) -buildOtherParm: OtherParm=[x-farend;x-refc
```

從CUCM傳送的確認(ACK)

```
01384017.001 |12:53:00.678 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.17.230.
[105086,NET]
ACK sip:123456789@10.17.230.4:5060;transport=TCP SIP/2.0
Via: SIP/2.0/TCP 10.107.28.14:5060;branch=z9hG4bK6b0e716815d6
From: <sip:+1234567890@10.107.28.14;x-nearend;x-refci=43358625;x-nearendclusterid=eu91;x-nearenddevice=
To: <sip:123456789@10.17.230.4>;tag=ea1fb60-0-13c4-5506-90037-9c2acf-90037
Date: Mon, 10 Apr 2017 12:53:00 GMT
Call-ID: 9f432380-8eb1802c-67f6-c2e4110a@10.107.28.14
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 232

v=0
o=CiscoSystemsCCM-SIP 46365 1 IN IP4 10.107.28.14
s=SIP Call
c=IN IP4 10.106.242.1
b=TIAS:0
b=AS:0
t=0 0
m=audio 7000 RTP/AVP 8 101
a=rtpmap:8 PCMA/8000
a=sendonly
a=rtpmap:101 telephone-event/8000
```

```
a=fmtp:101 0-15
```

## CUCM向錄制伺服器傳送針對遠端裝置的邀請

```
01384043.001 |12:53:00.679 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.17.230.4
[105087,NET]
INVITE sip:123456789@10.17.230.4:5060 SIP/2.0
Via: SIP/2.0/TCP 10.107.28.14:5060;branch=z9hG4bK6b0f5120dbe5
From: <sip:+1234567890@10.107.28.14;x-farend;x-refci=43358625;x-nearendclusterid=eu91;x-nearenddevice=LC
To: <sip:123456789@10.17.230.4>
Date: Mon, 10 Apr 2017 12:53:00 GMT
Call-ID: 9f432380-8eb1802c-67f7-c2e4110a@10.107.28.14
Supported: timer,resource-priority,replaces
Min-SE: 1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Cisco-Guid: 2671977344-0000065536-0000000140-3269726474
Session-Expires: 1800
P-Asserted-Identity: <sip:+1234567890@10.107.28.14>
Remote-Party-ID: <sip:+1234567890@10.107.28.14>;party=calling;screen=yes;privacy=off
Contact: <sip:+1234567890@10.107.28.14:5060;transport=tcp>;isFocus
Max-Forwards: 70
Content-Length: 0
```

## 200 OK from Recording Server

```
SIP/2.0 200 OK
From: <sip:+1234567890@10.107.28.14;x-farend;x-refci=43358625;x-nearendclusterid=eu91;x-nearenddevice=LC
To: <sip:123456789@10.17.230.4>;tag=ea1f830-0-13c4-5506-90037-22ea55b6-90037
Call-ID: 9f432380-8eb1802c-67f7-c2e4110a@10.107.28.14
CSeq: 101 INVITE
Via: SIP/2.0/TCP 10.107.28.14:5060;branch=z9hG4bK6b0f5120dbe5
Supported: timer
Contact: <sip:123456789@10.17.230.4:5060;transport=TCP>
Session-Expires: 1800;refresher=uas
Content-Type: application/sdp
Content-Length: 119

v=0
o=VRSP 0 0 IN IP4 10.10.1.10
s=NICE VRSP
c=IN IP4 127.0.0.1
t=0 0
m=audio 1000 RTP/AVP 0 4 8 9 18
a=recvonly
```

從CUCM傳送的ACK

```
01384207.001 |12:53:00.882 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.17.230.4
[105091,NET]
ACK sip:123456789@10.17.230.4:5060;transport=TCP SIP/2.0
Via: SIP/2.0/TCP 10.107.28.14:5060;branch=z9hG4bK6b1013a924b6
From: <sip:+1234567890@10.107.28.14;x-farend;x-refci=43358625;x-nearendclusterid=eu91;x-nearenddevice=LC
To: <sip:123456789@10.17.230.4>;tag=ea1f830-0-13c4-5506-90037-22ea55b6-90037
Date: Mon, 10 Apr 2017 12:53:00 GMT
Call-ID: 9f432380-8eb1802c-67f7-c2e4110a@10.107.28.14
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 232

v=0
o=CiscoSystemsCCM-SIP 46366 1 IN IP4 10.107.28.14
s=SIP Call
c=IN IP4 10.106.242.1
b=TIAS:0
b=AS:0
t=0 0
m=audio 7000 RTP/AVP 8 101
a=rtpmap:8 PCMA/8000
a=sendonly
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

座席最終呼叫該號碼

RCP埠偵聽MOH ，然後稍後斷開與MOH的連線，然後連線回代理，最後將代理連線到呼叫號碼。

```
01384484.001 |12:53:04.609 |AppInfo  |ARBTRY-ConnectionManager-wait_MediaConnectRequest(43358626,433586
01384484.002 |12:53:04.609 |AppInfo  |ARBTRY-ConnectionManager- storeMediaInfo(CI=43358626): EXISTING E
01384484.003 |12:53:04.609 |AppInfo  |ARBTRY-ConnectionManager- storeMediaInfo(CI=43358627): EXISTING E
```

CUCM傳送SDL HTTP請求

僅在近端和遠端裝置Invite發生200 OK後，CUCM才會傳送SDL Http請求以啟動錄製

LCP錄製的SDL HTTP請求

```
01384808.000 |12:53:04.672 |SdlSig |SdlHTTPReq |wait |SdlHTTPService(2,100,6,1) |CayugaInterface(2,100,
```

```
<soapenv:Envelope xmlns:soapenv="http://www.w3.org/2003/05/soap-envelope">
<soapenv:Body>
<RequestXmfConnectionMediaForking xmlns="http://www.cisco.com/schema/cisco_xmf/v1_0">
<msgHeader>
<transactionID>Cisco:UCM:CayugaIf:1:69</transactionID>
<registrationID>C094:XMF:Unified CM 10.5.2.12901-1:1</registrationID>
</msgHeader>
<callID>42</callID>
<connID>554</connID>
<action>
<enableMediaForking>
<nearEndAddr>
<ipv4>10.17.230.5</ipv4>
<port>42095</port>
</nearEndAddr>
<farEndAddr>
<ipv4>10.17.230.5</ipv4>
<port>42094</port>
</farEndAddr>
<preserve>true</preserve>
</enableMediaForking>
</action>
</RequestXmfConnectionMediaForking>
</soapenv:Body>
</soapenv:Envelope>
```


```
01384843.001 |12:53:04.674 |AppInfo |Recording::- (0000077) - Media Setup Complete: mRecordingCallInfo
01384843.002 |12:53:04.674 |AppInfo |RCD_RecordingCallInfo::print: resourceInfo
01384843.003 |12:53:04.674 |AppInfo |RCD_ResourceInfo::print: nodeId=2
01384843.004 |12:53:04.674 |AppInfo |RCD_ResourceInfo::print: bNum
01384843.005 |12:53:04.674 |AppInfo |RCD_Utility::printCcPtyNum: CcPtyNum contains only Directory Numbe
01384843.006 |12:53:04.674 |AppInfo |RCD_RecordingCallInfo::print: recordedPartyInfo
01384843.007 |12:53:04.674 |AppInfo |RCD_RecordedPartyInfo::print: ssAe
01384843.008 |12:53:04.674 |AppInfo |RCD_Utility::printSsAe: ss=43358625, nodeId=2
01384843.009 |12:53:04.674 |AppInfo |RCD_RecordedPartyInfo::print: partyNum
01384843.010 |12:53:04.674 |AppInfo |RCD_Utility::printCcPtyNum: CcPtyNum contains only Directory Numbe
01384843.011 |12:53:04.674 |AppInfo |RCD_RecordedPartyInfo::print: deviceName = LCP_47483708

01384843.023 |12:53:04.674 |AppInfo |RCD_Utility::printCcPtyNum: CcPtyNum contains only Directory Numbe
01384843.024 |12:53:04.674 |AppInfo |RCD_RecorderPartyInfo::print: partition = 812fe5de-3a9b-4d67-9fdd-
```

# 相關資訊

- http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cust_contact/contact_center/mediasense/10/sr srnd/CUMS_BK_MC36D963_00_mediasense-srnd_chapter_0111.html
- http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/voi-cube-uc-gateway-services.html
- http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/voi-ntwk-based.html

- 技術支援與文件 - Cisco Systems

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思科已使用電腦和人工技術翻譯本文件，讓全世界的使用者能夠以自己的語言理解支援內容。請注意，即使是最佳機器翻譯，也不如專業譯者翻譯的內容準確。Cisco Systems, Inc. 對這些翻譯的準確度概不負責，並建議一律查看原始英文文件（提供連結）。