

配置選項CUCM和CUBE之間的Ping

目錄

[簡介](#)

[必要條件](#)

[需求](#)

[採用元件](#)

[背景資訊](#)

[設定](#)

[驗證](#)

[疑難排解](#)

簡介

本檔案介紹如何在思科整合通訊管理員(CUCM)和思科整合邊界元件(CUBE)之間啟用功能選項 Ping。

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必要條件

思科建議您瞭解以下主題：

- Cisco Call Manager管理
- 思科整合邊界元件或閘道管理
- 作業階段啟始通訊協定(SIP)

採用元件

- 思科整合式服務路由器(ISR4351/K9)
- 思科整合通訊管理員12.0
- Cisco整合IP電話

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

請務必瞭解CUCM如何從SIP中繼擴展呼叫，如下所示：



CUCM - 192. .26



ISR 4351 - 192. .57

對於CUCM從SIP中繼擴展呼叫，它會繼續使用Trunk Configuration頁中指定的IP地址建立傳輸控制協定(TCP)三次握手，如下圖所示：

SIP Information

Destination

Destination Address is an SRV

Destination Address

1* 192. .57

Wireshark中的TCP三次握手如下圖所示：

Source	Destination	Protocol	Length	Info
192. .26	192. .57	TCP	74	38672 → 5060 [SYN] Seq=0 Win=14600 Len=0 MSS=1460 SACK_PERM=1
192. .57	192. .26	TCP	60	5060 → 38672 [SYN, ACK] Seq=0 Ack=1 Win=4128 Len=0 MSS=1460
192. .26	192. .57	TCP	54	38672 → 5060 [ACK] Seq=1 Ack=1 Win=14600 Len=0
192. .26	192. .57	SIP	1271	Request: INVITE sip:5123@192. .57:5060

這是基於每個呼叫、每個節點完成的；因此，CUCM在嘗試使用備用中繼或GW（網關）之前，必須等待同步(SYN)消息超時或SIP服務出錯。

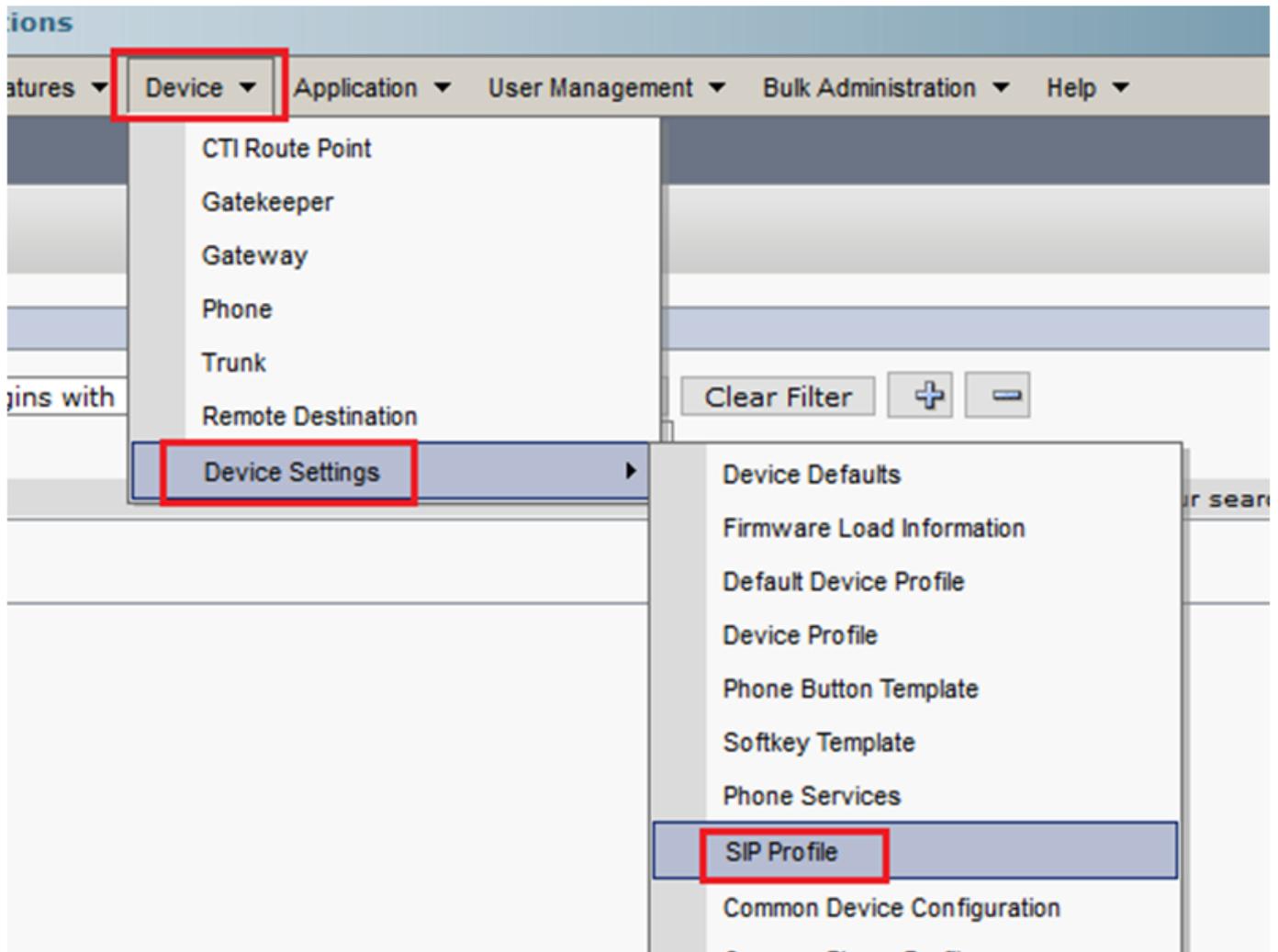
為了解決此問題，您可以啟用選項Ping並主動檢查SIP中繼的狀態。

當您在SIP中繼上啟用Options Ping時，您還會新增SIP中繼狀態和運行時間統計資訊，以便監控每個SIP中繼的狀態，並對中繼關閉的時刻進行故障排除。這些統計資訊顯示在SIP Trunk Configuration頁面上。

設定

步驟1.在SIP配置中啟用SIP選項Ping:

- 導覽至Cisco Unified CM Administration >> Device >> Device Settings >> SIP Profile，如下圖所示：



- 按一下「查詢」，然後決定是否要建立新的SIP配置檔案、編輯已存在的SIP配置檔案還是製作SIP配置檔案的副本。在本示例中，請建立標準SIP配置檔案的副本，如下圖所示：

A screenshot of the 'SIP Profile Configuration' page in a network management interface. The page has a dark header with the title 'SIP Profile Configuration'. Below the header is a toolbar with several icons and labels: 'Copy' (highlighted with a red box), 'Reset', 'Apply Config', and 'Add New'. Below the toolbar is a 'Status' section with two information icons and text: 'Status: Ready' and 'All SIP devices using this profile must be restarted before any changes will take affect.' Below the status section is the 'SIP Profile Information' section, which contains three input fields: 'Name*' with the value 'Standard SIP Profile', 'Description' with the value 'Default SIP Profile', and 'Default MTP Telephony Event Payload Type*' with the value '101'.

- 重新命名新的SIP配置檔案並啟用選項Ping，如下圖所示：

SIP Profile Configuration

 Save

Status

 Status: Ready

 All SIP devices using this profile must be restarted before any changes will take affect.

SIP Profile Information

Name*	<input type="text" value="Options Ping SIP Profile"/>
Description	<input type="text" value="Default SIP Profile"/>
Default MTP Telephony Event Payload Type*	<input type="text" value="101"/>
Early Offer for G.Clear Calls*	<input type="text" value="Disabled"/>
User-Agent and Server header information*	<input type="text" value="Send Unified CM Version Information as User-Agent"/>
Version in User Agent and Server Header*	<input type="text" value="Major And Minor"/>
Dial String Interpretation*	<input type="text" value="Phone number consists of characters 0-9, *, #, and"/>
Confidential Access Level Headers*	<input type="text" value="Disabled"/>

SIP OPTIONS Ping

<input checked="" type="checkbox"/> Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"	
Ping Interval for In-service and Partially In-service Trunks (seconds)*	<input type="text" value="60"/>
Ping Interval for Out-of-service Trunks (seconds)*	<input type="text" value="120"/>
Ping Retry Timer (milliseconds)*	<input type="text" value="500"/>
Ping Retry Count*	<input type="text" value="6"/>

步驟2.將SIP配置檔案新增到有問題的SIP中繼並單擊Save:

附註：請記住，此中繼必須先前已配置。如果您需要有關如何配置SIP中繼的指導，請訪問以下連結：[系統配置指南](#)

- 導覽至Device >> Trunk，然後選擇要編輯的中繼，如下圖所示：

Cisco Unified CM Administration

For Cisco Unified Communications Solutions

Call Routing ▾ Media Resources ▾ Advanced Features ▾ **Device ▾** Application ▾ User Management ▾

Device Configuration

 Delete  Copy  Reset  Apply Config

CTI Route Point

Gatekeeper

Gateway

Phone

Trunk

Remote Destination

Device Settings ▶

successful

IP devices using this profile must be restarted before any

File Information

Options Ping SIP Profile

Default SIP Profile

TP Telephony Event Payload Type* 101

er for G.Clear Calls* Disabled ▾

nt and Server header information* Send Unified CM Version Information as User-Agen' ▾

1 User Agent and Server Header* Major And Minor ▾

g Interpretation* Phone number consists of characters 0-9, *, #, and ▾

Find and List Trunks

 Add New  Select All  Clear All  Delete Selected  Reset Selected

Status

 1 records found

Trunks (1 - 1 of 1)

Find Trunks where Device Name ▾ begins with ▾ TAC Find
Select item or enter search text ▾

<input type="checkbox"/>	Name ▲	Description	Calling Search Space
<input type="checkbox"/>	 TAC-SIP-Trunk	TAC SIP Trunk	

- 請注意，「狀態」、「狀態原因」和「持續時間」均設定為N/A。
- 選擇正確的SIP配置檔案，然後點選儲存

SIP Information

Destination

Destination Address is an SRV

Destination Address: 192.X.X.57 Destination Address IPv6: Destination Port: 5060

Status	Status Reason	Duration
N/A	N/A	N/A

MTP Preferred Originating Codec*: 711ulaw

BLF Presence Group*: Standard Presence group

SIP Trunk Security Profile*: Non Secure SIP Trunk Profile

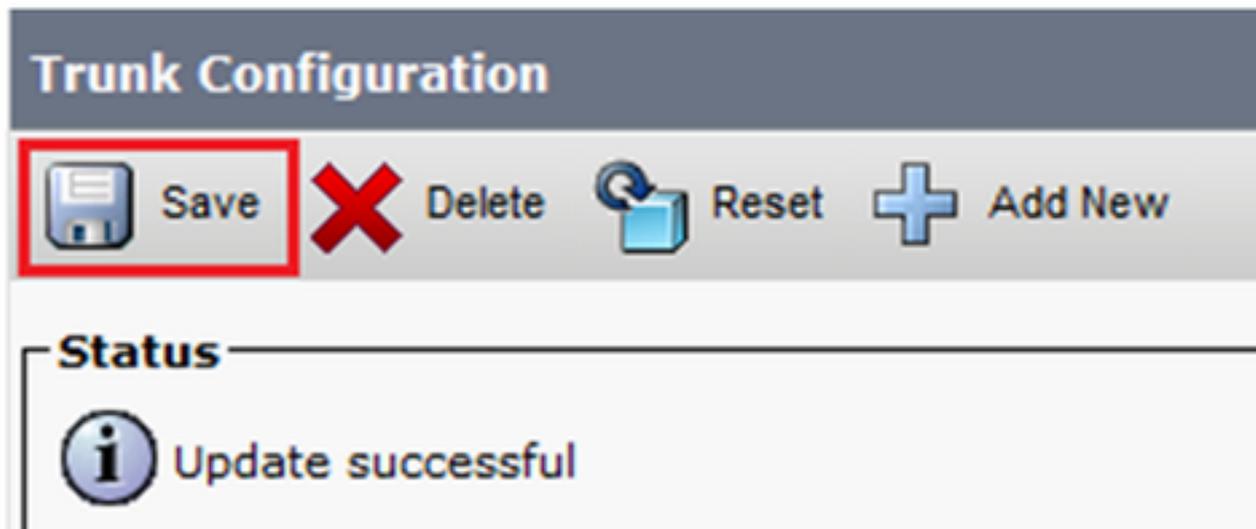
Rerouting Calling Search Space: < None >

Out-Of-Dialog Refer Calling Search Space: < None >

SUBSCRIBE Calling Search Space: < None >

SIP Profile*: Options Ping SIP Profile [View Details](#)

DTMF Signaling Method*: No Preference



- 此時，CUCM必須能夠監控SIP中繼的狀態，如下圖所示：

Trunks (1 - 1 of 1)

Find Trunks where Device Name begins with tac Find Clear Filter

Name	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status	SIP Trunk Duration
TAC-SIP-Trunk	TAC SIP Trunk		Default	XXXX				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 2 minutes

SIP Information

Destination

Destination Address is an SRV

Destination Address: 192.X.X.57 Destination Address IPv6: Destination Port: 5060

Status	Status Reason	Duration
up		Time Up: 0 day 0 hour 4 minutes

步驟3. (可選) 在SIP中繼的遠端啟用SIP選項Ping。在這種情況下：192.X.X.57(ISR 4351)

- 導航到ISR Cisco Unified Border Element或Gateway，並確認您要將Ping選項新增到哪個撥號對等體，如下圖所示：

```
LESQUIVE-4351-A(config)#do show run | sec dial-peer voice 100
dial-peer voice 100 voip
description CUCM dial-peer
session protocol sipv2
session target ipv4:192.X.X.26
dtmf-relay rtp-nte sip-kpml
codec g711ulaw
```

- 使用命令新增選項Ping:voice-class sip options-keepalive，如下圖所示：

```

LESQUIVE-4351-A(config)#do show run | sec dial-peer voice 100
dial-peer voice 100 voip
description CUCM dial-peer
session protocol sipv2
session target ipv4:192.168.1.26
dtmf-relay rtp-nte sip-kpml
codec g711ulaw
LESQUIVE-4351-A(config)#dial-peer voice 100
LESQUIVE-4351-A(config-dial-peer)#voice-class sip options-keepalive

```

驗證

使用本節內容，確認已正確交換選項消息。

附註：如果您需要瞭解如何在CUCM eth0埠上運行資料包捕獲，請按照以下連結中的說明操作：[CUCM裝置型號上的資料包捕獲](#)

- 請注意，TCP三次握手僅執行一次，當中繼重新啟動後，我們僅將OPTIONS消息從CUCM傳送到ISR，該消息應使用200 OK作為響應。預設情況下，這些消息每60秒交換一次。

Source	Destination	Protocol	Length	Info
192.168.1.26	192.168.1.57	TCP	74	46535 → 5060 [SYN] Seq=0 Win=14600 Len=0 MSS=1460
192.168.1.57	192.168.1.26	TCP	60	5060 → 46535 [SYN, ACK] Seq=0 Ack=1 Win=4128 Len=0
192.168.1.26	192.168.1.57	TCP	54	46535 → 5060 [ACK] Seq=1 Ack=1 Win=14600 Len=0
192.168.1.26	192.168.1.57	SIP	451	Request: OPTIONS sip:192.168.1.57:5060
192.168.1.57	192.168.1.26	TCP	60	5060 → 46535 [ACK] Seq=1 Ack=398 Win=3731 Len=0
192.168.1.57	192.168.1.26	SIP/SDP	1014	Status: 200 OK

- 請注意，選項消息僅從192.X.X.26(CUCM)傳送到192.X.X.57(ISR)，因為只有CUCM配置為監控中繼狀態：

Time	Source	Destination	Protocol	Length	Info
13:37:46.029581	192.168.1.26	192.168.1.57	SIP	451	Request: OPTIONS sip:192.168.1.57:5060
13:37:46.031672	192.168.1.57	192.168.1.26	SIP/SDP	1014	Status: 200 OK
13:38:47.552245	192.168.1.26	192.168.1.57	SIP	451	Request: OPTIONS sip:192.168.1.57:5060
13:38:47.554691	192.168.1.57	192.168.1.26	SIP/SDP	513	Status: 200 OK
13:39:48.895232	192.168.1.26	192.168.1.57	SIP	452	Request: OPTIONS sip:192.168.1.57:5060
13:39:48.897399	192.168.1.57	192.168.1.26	SIP/SDP	1014	Status: 200 OK
13:40:50.418479	192.168.1.26	192.168.1.57	SIP	451	Request: OPTIONS sip:192.168.1.57:5060
13:40:50.420957	192.168.1.57	192.168.1.26	SIP/SDP	1014	Status: 200 OK
13:41:51.014881	192.168.1.26	192.168.1.57	SIP	451	Request: OPTIONS sip:192.168.1.57:5060
13:41:51.017117	192.168.1.57	192.168.1.26	SIP/SDP	1013	Status: 200 OK
13:42:52.389610	192.168.1.26	192.168.1.57	SIP	451	Request: OPTIONS sip:192.168.1.57:5060

- 現在進行呼叫時，CUCM已經知道中繼處於運行狀態並立即傳送Invite：

192.168.1.57	192.168.1.26	SIP/SDP	1013	Status: 200 OK
192.168.1.26	192.168.1.57	SIP	451	Request: OPTIONS sip:192.168.1.57:5060
192.168.1.57	192.168.1.26	SIP/SDP	1013	Status: 200 OK
192.168.1.26	192.168.1.57	SIP	1271	Request: INVITE sip:5123@192.168.1.57:5060

- 如果您執行了步驟3（在CUBE上執行可選配置），您將看到雙向傳送的Options消息：

192.168.1.26	SIP	440	Request: OPTIONS sip:192.168.1.26:5060
192.168.1.57	SIP	449	Status: 200 OK
192.168.1.57	SIP	452	Request: OPTIONS sip:192.168.1.57:5060
192.168.1.26	SIP/SDP	1014	Status: 200 OK

疑難排解

— 為了對CUCM中的選項Ping進行故障排除，您需要：

- 最佳開始選項是從CUCM Eth0埠捕獲資料包，更多詳細資訊：[CUCM裝置型號上的資料包捕獲](#)使用第三方自由軟體Wireshark開啟捕獲，並使用SIP進行過濾
- 您還可以檢查詳細的Cisco Callmanager跟蹤，使用RTMT下載它們，在此處找到步驟：[如何收集CUCM 9.x或更高版本的跟蹤](#)
- 驗證此連結中的SIPTrunkOOS原因代碼：系[統錯誤消息](#)
 - Local=1 (請求超時)
 - local=2 (本地SIP堆疊無法與遠端對等體建立套接字連線)
 - Local=3 (DNS查詢失敗)

— 為了對ISR4351中的Ping選項進行故障排除，您需要：

- 調試ccsip消息
- Debug ccapi inout
- 從指向CUCM的介面捕獲資料包