疑難解答CUBE SP拒絕轉發到PSTN號碼的內部呼叫

目錄

簡介

本文檔介紹如何在思科統一邊界要素（SP版）(CUBE SP)拒絕內部呼叫（配置轉發為PSTN號碼）時進行故障排除。

呼叫流： 內部IP電話4002呼叫內部IP電話4001,ip電話4001上的所有呼叫都轉移到已配置的PSTN號碼。

問題：從IP電話4002到4001的呼叫時主叫方聽到快速忙音

呼叫方使用IP電話1呼叫另一個IP電話2,IP電話2配置為將所有呼叫轉接到外部PSTN號碼。呼叫無法連線PSTN電話，PSTN電話未振鈴且呼叫方聽到快速忙音。

解決方案

以下是進行問題疑難排解的步驟。

步驟1.思科統一通訊管理器(CUCM)日誌分析。

在CUCM日誌中，可以看到來自CUBE SP的錯誤消息。

```
SIP/2.0 604 Does Not Exist Anywhere
```

詳細消息：

```
SIP/2.0 604 Does Not Exist Anywhere from cube SP
 82645958.001 |13:08:46.297 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to
10.4.15.253 on port 5060 index 18491 [19580587,NET] INVITE sip:+612xxxxxxxx@10.x.x.x:5060
SIP/2.0 Via: SIP/2.0/TCP 10.4.15.5:5060;branch=z9hG4bK3cc7264a831cc4 From:
<sip:+612xxxxxxx@10.x.x.x>;tag=8162255~9cbf8c07-9c9b-758f-e658-bebd74e53d96-40280558 To:
<sip:+614xxxxxxx@10.4.15.253> Date: Fri, 17 Nov 2017 02:08:46 GMT Call-ID: 3cf99080-a0e144ae-
3692cb-50f040a@10.x.x.x Supported: 100rel,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 Supported: Geolocation Call-Info:
<sip:10.x.x.x:5060>;method="NOTIFY;Event=telephone-event;Duration=500" Call-Info: <urn:x-cisco-
remotecc:callinfo>;x-cisco-video-traffic-class=VIDEO_UNSPECIFIED Cisco-Guid: 1022988416-
0000065536-0000118822-0084870154 Session-Expires: 1800 Diversion:
```

```
P-Asserted-Identity: x <sip:+612xxxxxxxx@10.x.x.x>
Remote-Party-ID: x <sip:+612xxxxxxxxx@10.x.x.x>;party=calling;screen=yes;privacy=off
```

步驟2. CUBE SP日誌分析。

在CUBE SP日誌中，您可以看到呼叫未通過源號碼分析，因為它與任何條目都不匹配。

inside na-src-prefix-table

轉移：**<sip:9180@10.x.x.x>;reason=unknown;privacy=off;screen=yes**

```
Routing fails.
SBC Index = 0X00000001
Config set Index = 0X0000270F
Source Account = CUCM-TL1
Source Adjacency = CUCM-cust01-1
Calling Address Type = 0X00030000
Called Address Type = 0X00030000
Calling Address = 9180
Called Address = +614xxxxxxxx
```

步驟3.根據步驟1和2的疑難排解確認進入Bug。

這會影響已知的[錯誤CSCup67940](https://bst.cloudapps.cisco.com/bugsearch/bug/CSCup67940/?referring_site=bugquickviewredir)

CUCM需要傳送**extend&connect**的轉移報頭中的E.164號碼。

https://bst.cloudapps.cisco.com/bugsearch/bug/CSCup67940/?referring_site=bugquickviewredir

因應措施：

除非我們在CUBE中進行修改以接受來自轉移報頭的邀請，否則包含電話DN，如**26708 <sip:26708@58.162.59.181>;reason=unknown;privacy=off;screen=yes**

因應措施

根據解決方法，允許轉接標頭中的編號。

執行此操作可以在此**na-src-prefix-table**中新增新條目。

```
na-src-prefix-table   xxxxx

     entry  10
     action accept
     match-prefix  9
```

應用替代方法後的新問題

應用此工作區後，呼叫已成功連線，但會向服務提供商傳送一個五位分機號。

使用SIP報頭編輯器解決此問題

在實驗室中經過測試，當您使用SIP報頭編輯器修改CUBE SP中的Distribution報頭時，它會成功連線呼叫並向服務提供商傳送e164號碼。

程式

在實驗室測試中，IP電話4002呼叫4001，在IP電話4001上呼叫全部到60006009(PSTN)號碼。

```
sip header-editor donnietest
    store-rule entry 1
     condition header-name Diversion header-value regex-match "sip:4\(...\)" store-as
diversionuri
    header diversion entry 1
     action replace-value value
"<sip:+888888884${diversionuri}@10.66.75.51>;reason=unconditional;

privacy=off;screen=yes"
      condition header-name Diversion header-value regex-match "sip:4\(...\)@"



 adjacency sip donniecucm
    editor-type editor
    header-editor inbound donnietest
```

驗證

沒有修改轉接報頭

如果不修改任何轉移報頭，您可以從CUCM看到以下轉移報頭邀請

```
Diversion: <sip:4001@10.66.75.51>;reason=unconditional;privacy=off;screen=yes
INVITE sip:60006099@10.66.75.33:5068 SIP/2.0
Via: SIP/2.0/TCP 10.66.75.51:5060;branch=z9hG4bK1ef607cac8bd6
```

```
From: "agent2-4002" <sip:4002@10.66.75.51>;tag=194346~4c742393-721f-476b-82c3-bc13f8a9c6cd-
22765770
To: <sip:60006099@10.66.75.33>
Date: Sun, 19 Nov 2017 23:39:16 GMT
Call-ID: d9ad6f80-a1211624-1eee8-334b420a@10.66.75.51
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM11.5
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
Supported: Geolocation
Call-Info: <sip:10.66.75.51:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=DESKTOP
Session-ID: 223cb8ec818c0c0dd669d19baa194344;remote=00000000000000000000000000000000
Cisco-Guid: 3652022144-0000065536-0000000027-0860570122
Session-Expires:  1800
Diversion: <sip:4001@10.66.75.51>;reason=unconditional;privacy=off;screen=yes
P-Asserted-Identity: "agent2-4002" <sip:4002@10.66.75.51>
Remote-Party-ID: "agent2-4002" <sip:4002@10.66.75.51>;party=calling;screen=yes;privacy=off
Contact: <sip:4002@10.66.75.51:5060;transport=tcp>
Max-Forwards: 69
Content-Length: 0
```

SIP後台編輯器匹配轉接標頭

SIP報頭編輯器將轉移報頭以**sip:4xxx@**匹配，然後將其變成+E164格式

可在sip標頭編輯器後看到。在轉移標頭中，4001已修改為+888888884001

轉移：<sip:+888888884001@10.66.75.51>;reason=unconditional;privacy=off;screen=yes

MSG-6401-0027-69FECA-0747,2017年11月20日01:48:38(491542613毫秒
): 0X01000E2059EBD60A

模組在編輯後返回了消息。
編輯器名稱= donnietest
編輯器配置集= 0X00000000
這是您編輯後的消息。

```
INVITE sip:60006099@10.66.75.33:5068 SIP/2.0
Supported: X-cisco-srtp-fallback
Via: SIP/2.0/TCP 10.66.75.51:5060;branch=z9hG4bK1f11c18671c97
From: "agent2-4002" <sip:4002@10.66.75.51>;tag=194931~4c742393-721f-476b-82c3-bc13f8a9c6cd-
22765859
To: <sip:60006099@10.66.75.33>
Date: Mon, 20 Nov 2017 02:13:12 GMT
Call-ID: 5ac33180-a1213a38-1f045-334b420a@10.66.75.51
Min-SE: 1800
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
```

```
Allow-Events: presence, kpml
Call-Info: <sip:10.66.75.51:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=DESKTOP
Session-ID: 223cb8ec818c0c0dd669d19baa194929;remote=00000000000000000000000000000000
Cisco-Guid: 1522741632-0000065536-0000000050-0860570122
Session-Expires: 1800
Diversion: <sip:+888888884001@10.66.75.51>;reason=unconditional;privacy=off;screen=yes
P-Asserted-Identity: "agent2-4002" <sip:4002@10.66.75.51>

Remote-Party-ID: "agent2-4002" <sip:4002@10.66.75.51>;party=calling;screen=yes;privacy=off
Contact: <sip:4002@10.66.75.51:5060;transport=tcp>
Max-Forwards: 69
Content-Length: 0
```

MSG-6401-0028-69FECA-0885,2017年11月20日01:48:38(491542613毫秒): 0X01000E2059EBD60A

對消息進行編輯。

這是您編輯後的消息

```
INVITE sip:60006099@10.66.75.33:5068 SIP/2.0
Supported: X-cisco-srtp-fallback
Via: SIP/2.0/TCP 10.66.75.51:5060;branch=z9hG4bK1f11c18671c97
From: "agent2-4002" <sip:4002@10.66.75.51>;tag=194931~4c742393-721f-476b-82c3-bc13f8a9c6cd-
22765859
To: <sip:60006099@10.66.75.33>
Date: Mon, 20 Nov 2017 02:13:12 GMT
Call-ID: 5ac33180-a1213a38-1f045-334b420a@10.66.75.51
Min-SE: 1800
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Call-Info: <sip:10.66.75.51:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=DESKTOP
Session-ID: 223cb8ec818c0c0dd669d19baa194929;remote=00000000000000000000000000000000
Cisco-Guid: 1522741632-0000065536-0000000050-0860570122
Session-Expires: 1800
Diversion: <sip:+888888884001@10.66.75.51>;reason=unconditional;privacy=off;screen=yes
P-Asserted-Identity: "agent2-4002" <sip:4002@10.66.75.51>
Remote-Party-ID: "agent2-4002" <sip:4002@10.66.75.51>;party=calling;screen=yes;privacy=off
Contact: <sip:4002@10.66.75.51:5060;transport=tcp>
Max-Forwards: 69
Content-Length: 0
```