

Troubleshoot CUBE SP Rejects Internal Call which is Forwarded to a PSTN Number

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Introduction

This document describes how to troubleshoot Cisco Unified Border Element (SP Edition) (CUBE SP) when it rejects the internal call, which is config forwarded to PSTN number.

Call Flow: Internal IP phone 4002 calls internal IP phone 4001, all calls on ip phone 4001 are forward to a configured PSTN number.

Problem: Caller Hears Fast Busy Tone when Calls from IP Phone 4002 to 4001

Caller uses IP Phone 1 to call another IP Phone 2, the IP Phone 2 is configured to forward all calls to an external PSTN number. The call failed to connect the PSTN Phone, PSTN phone does not ring and the caller hears fast busy tone.

Solution

These are the steps to troubleshoot the issue.

Step 1. Cisco Unified Communication Manager (CUCM) Log Analysis.

From CUCM logs, can see error message coming from CUBE SP.

Detail Message:

```
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:+612xxxxxxxx@10.x.x.x>;tag=8162255-9cbf8c07-9c9b-758f-e658-bebd74e53d96-40280558 To:
<sip:+614xxxxxxxx@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:
<sip:9180@10.x.x.x>;reason=unconditional;privacy=off;screen=yes
P-Asserted-Identity: x <sip:+612xxxxxxxx@10.x.x.x>
Remote-Party-ID: x <sip:+612xxxxxxxx@10.x.x.x>;party=calling;screen=yes;privacy=off
```

Step 2. CUBE SP Log Analysis.

From CUBE SP logs, you can see that the call did not pass source number analysis, as it does not match any entry.

inside na-src-prefix-table

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

```
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:+612xxxxxxxx@10.x.x.x>;tag=8162255-9cbf8c07-9c9b-758f-e658-bebd74e53d96-40280558 To:
<sip:+614xxxxxxxx@10.4.15.253> Date: Fri, 17 Nov 2017 02:08:46 GMT Call-ID: 3cf99080-a0e144ae-
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Remote-Party-ID: x <sip:+612xxxxxxxx@10.x.x.x>;party=calling;screen=yes;privacy=off
```

Step 3. Base on Troubleshoot Step 1 and 2 Confirm it Hit Bug.

This hits the konwn bug [CSCup67940](#)

CUCM needs to send E.164 number in diversion header for **extend&connect**.

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

Workaround:

Unless we make modification in the CUBE to accept the invite from diversion header contains phone DN such as **26708**

<sip:26708@58.162.59.181>;reason=unknown;privacy=off;screen=yes

Workaround

According to the workaround, allow the number in Diversion header.

This can be done to add a new entry in this **na-src-prefix-table**.

```
SIP/2.0 604 Does Not Exist Anywhere from cube SP
82645958.001 |13:08:46.297 |AppInfo |SIPTcp - wait_Sd1SPISignal: 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:+612xxxxxxxx@10.x.x.x>;tag=8162255~9cbf8c07~9c9b-758f-e658-bebd74e53d96-40280558 To:
<sip:+614xxxxxxxx@10.4.15.253> Date: Fri, 17 Nov 2017 02:08:46 GMT Call-ID: 3cf99080-a0e144ae-
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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:
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<sip:9180@10.x.x.x>;reason=unconditional;privacy=off;screen=yes
P-Asserted-Identity: x <sip:+612xxxxxxxx@10.x.x.x>
Remote-Party-ID: x <sip:+612xxxxxxxx@10.x.x.x>;party=calling;screen=yes;privacy=off
```

New Issue after you Apply the Workaround

After you apply this workaroud, the call is successfully connected but a five digit extension number is sent to the Service Provider.

Use SIP Header-Editor to Fix this Issue

Tested in the lab, as you use SIP header-editor to modify Diversion header in CUBE SP, it connects the call successfully and sends e164 number to service provider.

Procedure

In the lab testing, IP Phone 4002 calls 4001, on IP phone 4001 call foward all to 60006009 (PSTN) number.

```
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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:+612xxxxxxxx@10.x.x.x>;tag=8162255~9cbf8c07~9c9b-758f-e658-bebd74e53d96-40280558 To:
<sip:+614xxxxxxxx@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:** <sip:9180@10.x.x.x>;reason=unconditional;privacy=off;screen=yes P-Asserted-Identity: x <sip:+612xxxxxxxx@10.x.x.x> Remote-Party-ID: x <sip:+612xxxxxxxx@10.x.x.x>;party=calling;screen=yes;privacy=off

Verify

No Diversion Header Modification

Without any Diversion Header modification, you can see the invite from CUCM the Diversion Header is below

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:+612xxxxxxxx@10.x.x.x>;tag=8162255-9cbf8c07-9c9b-758f-e658-bebd74e53d96-40280558 To: <sip:+614xxxxxxxx@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:** <sip:9180@10.x.x.x>;reason=unconditional;privacy=off;screen=yes P-Asserted-Identity: x <sip:+612xxxxxxxx@10.x.x.x> Remote-Party-ID: x <sip:+612xxxxxxxx@10.x.x.x>;party=calling;screen=yes;privacy=off

SIP Header-Editor Match Diversion Header

SIP header-editor match the Diversion Header Start with **sip:4xxx@** , then make it +E164 format

It can be seen after sip header-editor. In Diversion Header, 4001 has been modified to +888888884001

Diversion: <sip:[+888888884001@10.66.75.51](#)>;reason=unconditional;privacy=off;screen=yes

MSG-6401-0027-69FECA-0747 at 01:48:38, 20 November 2017 (491542613 ms):
0X01000E2059EBD60A

A module has returned a message after editing.

Editor name = donnietest

Editor config set = 0X00000000

This is the message after you edit.

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SIP/2.0 Via: SIP/2.0/TCP 10.4.15.5:5060;branch=z9hG4bK3cc7264a831cc4 From:
<sip:+612xxxxxxxx@10.x.x.x>;tag=8162255-9cbf8c07-9c9b-758f-e658-bebd74e53d96-40280558 To:
<sip:+614xxxxxxxx@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:
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<sip:9180@10.x.x.x>;reason=unconditional;privacy=off;screen=yes
P-Asserted-Identity: x <sip:+612xxxxxxxx@10.x.x.x>
Remote-Party-ID: x <sip:+612xxxxxxxx@10.x.x.x>;party=calling;screen=yes;privacy=off
MSG-6401-0028-69FECA-0885 at 01:48:38, 20 November 2017 (491542613 ms):
0X01000E2059EBD60A

The edits are made on the message.

This is the message after you edit

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:+612xxxxxxxx@10.x.x.x>;tag=8162255-9cbf8c07-9c9b-758f-e658-bebd74e53d96-40280558 To:
<sip:+614xxxxxxxx@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:**
<sip:9180@10.x.x.x>;reason=unconditional;privacy=off;screen=yes
P-Asserted-Identity: x <sip:+612xxxxxxxx@10.x.x.x>
Remote-Party-ID: x <sip:+612xxxxxxxx@10.x.x.x>;party=calling;screen=yes;privacy=off