

Troubleshoot busyout Dial Peers on CUBE or IOS Voice Gateway

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Introduction

This document describes the issue of Cisco Unified Border Elements/voice gateway dial peer status busyout and call failures after Cisco IOS® upgrade.

Prerequisites

Requirements

There are no specific requirements for this document.

Components Used

The information in this document is based on Cisco Unified Border Elements (CUBE).

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, ensure that you understand the potential impact of any command.

Background Information

This document covers failures after Cisco IOS® upgrade to 16.12.6/17.3.5/17.6.1 or higher versions.

Problem

The calls are failing via Cisco IOS Voice Gateway or CUBE after Cisco IOS upgrade to 16.12.6/17.3.5/17.6.1/17.7.1 or higher versions.

Symptoms

When CUBE receives a SIP call ad matches an outgoing dial-peer with **session server-group** and **sip**

options-keepalive configured, the call fails at the Call Control Application Programming Interface (CCAPI) layer with "Cause Value" 188.

The CUBE does not send out outbound INVITE to the destination servers that are part of the server group.

The incoming INVITE is responded to with TRYING and 503 Service Unavailable.

The same behavior is observed even when the dial-peer shows as busyout or active **KEEPALIVE** status under **show dial-peer voice summary**.

Sample configuration/dial-peer status/debug snippet:

```
dial-peer voice 1000 voip
destination-pattern ^1000$
session protocol sipv2
session transport tcp
session server-group 1
voice-class sip options-keepalive profile 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte sip-kpm1
codec g711ulaw
ip qos dscp cs3 signaling
no vad
voice class server-group 1
ipv4 10.106.117.11
ipv4 10.106.117.6 preference 1
```

show dial-peer voice summary

TAG	TYPE	MIN	OPER	PREFIX	DEST-PATTERN	PRE	PASS	SESS-SER-GRP	OUT	STAT	PORT	KEEPALIVE	VR
3001	voip	up	up			0	syst						
1000	voip	up	up		^1000\$	0	syst	SESS-SVR-GRP: 1				busyout	NA

show dial-peer voice summary

TAG	TYPE	MIN	OPER	PREFIX	DEST-PATTERN	PRE	PASS	SESS-SER-GRP	OUT	STAT	PORT	KEEPALIVE	VR
3001	voip	up	up			0	syst						
1000	voip	up	up		^1000\$	0	syst	SESS-SVR-GRP: 1				active	NA

Debug snippet:

```
007592: Apr 7 07:28:56.046: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
```

INVITE sip:1000@10.106.117.5:5060 SIP/2.0
Via: SIP/2.0/UDP 10.106.117.2:5060;branch=z9hG4bK51889
Remote-Party-ID: <sip:3001@10.106.117.2>;party=calling;screen=no;privacy=off
From: <sip:3001@10.106.117.2>;tag=12EE76F8-154A
To: <sip:1000@10.106.117.5>
Date: Wed, 06 Apr 2022 18:28:16 GMT
Call-ID: 28E9846D-B50E11EC-8025D5B1-C2D1F237@10.106.117.2
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 0678152134-3037598188-2149635505-3268538935
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Max-Forwards: 70
Timestamp: 1649269696
Contact: <sip:3001@10.106.117.2:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 247

v=0
o=CiscoSystemsSIP-GW-UserAgent 8965 7288 IN IP4 10.106.117.2
s=SIP Call
c=IN IP4 10.106.117.2
t=0 0
m=audio 18406 RTP/AVP 0 101
c=IN IP4 10.106.117.2
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20

007649: Apr 7 07:28:56.050://-1/286BC7C68020/SIP/Info/info/2048/sipSPIGetCallConfig: Peer tag 3001 mat

007872: Apr 7 07:28:56.061: //89/286BC7C68020/CCAPI/ccCallSetupRequest:
Destination=, Calling IE Present=TRUE, Mode=0,
Outgoing Dial-peer=1000, Params=0x7FF65E441DE8, Progress Indication=NULL(0)

007935: Apr 7 07:28:56.064: //-1/xxxxxxxxxxxx/SIP/Info/critical/8192/ccsip_call_setup_request: SIP Dia

008160: Apr 7 07:28:56.073: //90/286BC7C68020/CCAPI/cc_api_call_disconnected:
Cause Value=188, Interface=0x7FF64F4542E8, Call Id=90

008199: Apr 7 07:28:56.077: //89/286BC7C68020/CCAPI/ccCallDisconnect:
Cause Value=188, Tag=0x0, Call Entry(Previous Disconnect Cause=0, Disconnect Cause=0)

008239: Apr 7 07:28:56.079: //89/286BC7C68020/SIP/Msg/ccsipDisplayMsg:

Sent:

SIP/2.0 503 Service Unavailable

Via: SIP/2.0/UDP 10.106.117.2:5060;branch=z9hG4bK51889

From: <sip:3001@10.106.117.2>;tag=12EE76F8-154A

To: <sip:1000@10.106.117.5>;tag=1C2F76-17F5

Date: Wed, 06 Apr 2022 17:28:56 GMT

Call-ID: 28E9846D-B50E11EC-8025D5B1-C2D1F237@10.106.117.2

Timestamp: 1649269696

CSeq: 101 INVITE

Allow-Events: telephone-event

Server: Cisco-SIPGateway/IOS-17.3.5

Reason: Q.850;cause=0

Session-ID: 00000000000000000000000000000000;remote=3c1f754eba075201a684fda2c51c04df

Content-Length: 0

Workaround

1. Configure the outgoing dial-peer with **session target ip4** instead of **session server-group**. If needed, create a separate dial-peer for each IP of the server group.

```
dial-peer voice 1000 voip
  session target ipv4:x.x.x.x
dial-peer voice 1001 voip
  session target ipv4:x.x.x.x
```

2. Remove the 'sip options-keepalive' on the dial-peer.

```
dial-peer voice 1000 voip
  no voice-class sip options-keepalive profile 1
```

3. Downgrade to an earlier version. This issue was introduced after the commitment of Cisco bug ID [CSCvx92872](#).

This issue is documented on Cisco bug ID [CSCvz80171](#).

Fix is available from 16.12.8/17.3.6/17.6.3/17.7.1/17.8.1