SDP中的重复c=线路导致使用各种ITSP的间歇性单向音频

目录

简介

本文档提供了一种解决方案，用于通过会话初始协议(SIP)/SIP思科统一边界元素(CUBE)向各种互联网电话服务提供商(ITSP)进行间歇性单向音频出站呼叫。

先决条件

要求

思科建议您了解SIP。

使用的组件

本文档中的信息基于以下软件和硬件版本：

- 思科统一通信管理器 (CUCM)
- CUBE

本文档中的信息都是基于特定实验室环境中的设备编写的。本文档中使用的所有设备最初均采用原始（默认）配置。如果您使用的是真实网络，请确保您已经了解所有命令的潜在影响。

规则

有关文档规则的详细信息，请参阅 Cisco 技术提示规则。

问题

症状

通过SIP/SIP CUBE到各种ITSP的出站呼叫的间歇性单向音频

呼叫流/拓扑:

Originator > CUCM(MGCP/SIP)> CUBE(SIP/SIP)> ITSP(Megafon)> Terminator。

原因/问题说明

如果ITSP提供商的邮件传输代理(MTA)在会话描述协议(SDP)(REINVITE/200 OK)中不支持重复的c=线路，则会导致从ITSP(Tx)到公共交换电话网(PSTN)电话(Rx)的支路出现间歇性单向音频。

提供商：Megafon(Megacable)

条件和环境

没有SIP配置文件：

```
################################################################################
Sent:
INVITE sip:3114560380@200.52.198.253:5151;transport=udp SIP/2.0
Via: SIP/2.0/UDP 200.52.198.15:5060;branch=z9hG4bK1BFE52263
From: <sip:3396900084@200.52.198.15:5060>;tag=3DF1D23A-15D3
To: sip:3114560380@200.52.198.253:5151;tag=227d2baf
Date: Wed, 27 Feb 2013 19:44:31 GMT
Call-ID: 00000196930006353732439410516722228326160@10.1.56.8
Supported: timer,resource-priority,replaces,sdp-anat
Min-SE:  360
Cisco-Guid: 3949497188-2152468962-2983459299-4054721625
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: 1361994271
Contact: <sip:3396900084@200.52.198.15:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 274

v=0
o=CiscoSystemsSIP-GW-UserAgent 8535 9331 IN IP4 200.52.198.15
s=SIP Call
c=IN IP4 200.52.198.15
t=0 0
m=audio 18504 RTP/AVP 0 101 19
c=IN IP4 200.52.198.15
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=rtpmap:19 CN/8000
a=ptime:20
```

使用已应用的SIP配置文件：

注意： Connection-Info删除第一个实例c=行，但不删除第二个实例。

```
################################################################################
PSTN#show run | sec voice class sip-profile
voice class sip-profiles 1000
 request REINVITE sdp-header Connection-Info remove
 response 200 sdp-header Connection-Info remove

Sent:
INVITE sip:3310862061@200.52.198.253:5151;transport=udp SIP/2.0
Via: SIP/2.0/UDP 200.52.198.15:5060;branch=z9hG4bK1BFB91A7E
From: <sip:3396900084@200.52.198.15:5060>;tag=3DC26466-1A5F
To: MEGAFON <sip:3310862061@200.52.198.253:5151>;tag=3e3a03d7
Date: Wed, 27 Feb 2013 18:52:42 GMT
Call-ID: 0000195730006353421530314263322228326160@10.1.56.8
Supported: timer,resource-priority,replaces,sdp-anat
Min-SE:  360
Cisco-Guid: 2932370470-2152010210-2968844771-4054721625
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,
   INFO, REGISTER
CSeq: 102 INVITE
Max-Forwards: 70
Timestamp: 1361991162
Contact: <sip:3396900084@200.52.198.15:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 250

v=0
o=CiscoSystemsSIP-GW-UserAgent 1274 9443 IN IP4 200.52.198.15
s=SIP Call
t=0 0
m=audio 21846 RTP/AVP 0 101 19
c=IN IP4 200.52.198.15
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:19 CN/8000
a=ptime:20
```

使用已应用的SIP配置文件：

注意： Connection-Info删除第二个实例c=行，但不删除第一个实例。

```
################################################################################
PSTN#show run | sec voice class sip-profile
voice class sip-profiles 1000
 request REINVITE sdp-header Audio-Connection-Info remove
 response 200 sdp-header Audio-Connection-Info remove

Sent:
INVITE sip:3310862061@200.52.198.253:5151;transport=udp SIP/2.0
Via: SIP/2.0/UDP 200.52.198.15:5060;branch=z9hG4bK1BFB91A7E
From: <sip:3396900084@200.52.198.15:5060>;tag=3DC26466-1A5F
To: MEGAFON <sip:3310862061@200.52.198.253:5151>;tag=3e3a03d7
Date: Wed, 27 Feb 2013 18:52:42 GMT
Call-ID: 0000195730006353421530314263322228326160@10.1.56.8
```

```
Supported: timer,resource-priority,replaces,sdp-anat
Min-SE:  360
Cisco-Guid: 2932370470-2152010210-2968844771-4054721625
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,
   INFO, REGISTER
CSeq: 102 INVITE
Max-Forwards: 70
Timestamp: 1361991162
Contact: <sip:3396900084@200.52.198.15:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 250

v=0
o=CiscoSystemsSIP-GW-UserAgent 1274 9443 IN IP4 200.52.198.15
s=SIP Call
c=IN IP4 200.52.198.15
t=0 0
m=audio 21846 RTP/AVP 0 101 19
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:19 CN/8000
a=ptime:20
```

***警告**

SDP(RFC 2327)支持允许多条c线路，这表明CUBE已正确实施了该功能。此解决方案示例是不正确支持RFC 2327的ITSP提供商的可能解决方案。

从RFC:

```
Session description
        v=  (protocol version)
        o=  (owner/creator and session identifier).
        s=  (session name)
        i=* (session information)
        u=* (URI of description)
        e=* (email address)
        p=* (phone number)
        c=* (connection information - not required if included in all media)
        b=* (bandwidth information)
        One or more time descriptions (see below)
        z=* (time zone adjustments)
        k=* (encryption key)
        a=* (zero or more session attribute lines)
        Zero or more media descriptions (see below)

Time description
        t=  (time the session is active)
        r=* (zero or more repeat times)

Media description
        m=  (media name and transport address)
        i=* (media title)
        c=* (connection information - optional if included at session-level)
        b=* (bandwidth information)
        k=* (encryption key)
        a=* (zero or more media attribute lines)
```

解决方案

使用此解决方法解决问题。

```
PSTN#show run | sec voice class sip-profile
voice class sip-profiles 1000
 request REINVITE sdp-header Audio-Connection-Info remove
 response 200 sdp-header Audio-Connection-Info remove
```

全局设置配置文件（语音服务VoIP）。

```
##################################
PSTN#show run | sec voice service voip
voice service voip
 sip
  sip-profiles 1000
```

在特定拨号对等体上设置配置文件。应在拨号对等体与PSTN之间进行设置。

```
###################################################################
PSTN#show run | sec dial-peer voice 5566
dial-peer voice 5566 voip
 destination-pattern 6666
 session target ipv4:1.1.1.1
 voice-class sip profiles 1000
```

有关详细信息，请参阅文档“思科统一边界元素(CUBE)会话初始协议(SIP)SIP配置文件规范化配置示例”。

SDP报头

以下是支持的SDP报头：

```
rtr(config-class)#response 200 sdp-header ?
  Attributea=
  Audio-Attributea=
  Audio-Bandwidth-Infob=
  Audio-Connection-Infoc=
  Audio-Encryption-Keyk=
  Audio-Mediam=audio
  Audio-Session-InfoI=
  Bandwidth-Keyb=
  Connection-Infoc=
  Email-Addresse=
  Encrypt-Keyk=
  Phone-Numberp=
  Repeat-Timesr=
  Session-InfoI=
  Session-Names=
  Session-Ownero=
  Time-Adjust-Keyz=
  Time-Headert=
  Url-Descriptoru=
  Versionv=
  Video-Attributea=
  Video-Bandwidth-Infob=
  Video-Connection-Infoc=
```

```
Video-Encryption-Keyk=
Video-Mediam=video
Video-Session-InfoI=
```

相关信息

- 思科统一边界元素(CUBE)会话初始协议(SIP)规范化与SIP配置文件配置示例
- 技术支持和文档 - Cisco Systems