在SDP的重复的c=线路导致与多种ITSP的断断续续单向音频

目录

简介

本文为在会话初始化协议(SIP) /SIP Cisco Unified Border Element (多维数据集)的断断续续单向音频呼出提供一解决方案给多种互联网电话服务提供商(ITSPs)。

先决条件

要求

思科建议您有SIP知识。

使用的组件

本文档中的信息基于以下软件和硬件版本：

- Cisco Unified Communications Manager (CUCM)
- 多维数据集

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

在呼出的断断续续单向音频在对多种ITSP的SIP/SIP多维数据集

呼叫流/拓扑：

创建人> CUCM (MGCP/SIP) >多维数据集(SIP/SIP) > ITSP (Megafon) >终结器。

原因/问题说明

有不支持在会话描述协议(SDP)的邮件转发代理的ITSP供应商(MTA) (好REINVITE/200的重复的c=线路)导致段的断断续续单向音频从ITSP(Tx)公共交换电话网(PSTN) phone(Rx)。

供应商：Megafon (Megacable)

情况和环境

没有SIP配置文件：

```
################################################################################Sent:INVITE sip:3114560380@200.52.198.253:5151;transport=udp SIP/2.0Via: SIP/2.0/UDP 200.52.198.15:5060;branch=z9hG4bK1BFE52263From: <sip:3396900084@200.52.198.15:5060>;tag=3DF1D23A-15D3To: sip:3114560380@200.52.198.253:5151;tag=227d2bafDate: Wed, 27 Feb 2013 19:44:31 GMTCall-ID: 000001969300063537324394105167222283261 60@10.1.56.8Supported: timer,resource-priority,replaces,sdp-anatMin-SE:  360Cisco-Guid: 3949497188-2152468962-2983459299-4054721625User-Agent: Cisco-SIPGateway/IOS-12.xAllow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,   INFO, REGISTERCSeq: 101 INVITEMax-Forwards: 70Timestamp: 1361994271Contact: <sip:3396900084@200.52.198.15:5060>Expires: 180Allow-Events: telephone-eventContent-Type: application/sdpContent-Length: 274v=0o=CiscoSystemsSIP-GW-UserAgent 8535 9331 IN IP4 200.52.198.15s=SIP Callc=IN IP4 200.52.198.15t=0 0m=audio 18504 RTP/AVP 0 101 19c=IN IP4 200.52.198.15a=rtpmap:0 PCMU/8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-16a=rtpmap:19 CN/8000a=ptime:20
```

使用已应用SIP配置文件：

注意： 连接INFO删除一审c=线路，但是不是第二。

```
################################################################################PSTN#show run | sec voice class sip-profilevoice class sip-profiles 1000 request REINVITE sdp-header Connection-Info remove response 200 sdp-header Connection-Info removeSent:INVITE sip:3310862061@200.52.198.253:5151;transport=udp SIP/2.0Via: SIP/2.0/UDP 200.52.198.15:5060;branch=z9hG4bK1BFB91A7EFrom: <sip:3396900084@200.52.198.15:5060>;tag=3DC26466-1A5FTo: MEGAFON <sip:3310862061@200.52.198.253:5151>;tag=3e3a03d7Date: Wed, 27 Feb 2013 18:52:42 GMTCall-ID: 00000195730006353421530314263322228326160@10.1.56.8Supported: timer,resource-priority,replaces,sdp-anatMin-SE: 360Cisco-Guid: 2932370470-2152010210-2968844771-4054721625User-Agent: Cisco-SIPGateway/IOS-12.xAllow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTERCSeq: 102 INVITEMax-Forwards: 70Timestamp: 1361991162Contact: <sip:3396900084@200.52.198.15:5060>Expires: 180Allow-Events: telephone-eventContent-Type: application/sdpContent-Length: 250v=0o=CiscoSystemsSIP-GW-UserAgent 1274 9443 IN IP4 200.52.198.15s=SIP Callt=0 0m=audio 21846 RTP/AVP 0 101 19c=IN IP4 200.52.198.15a=rtpmap:0 PCMU/8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-15a=rtpmap:19 CN/8000a=ptime:20
```

使用已应用SIP配置文件：

注意： 连接INFO删除第二条实例c=线路，但是不是第一。

```
###########################################################################PSTN#show run | sec voice class sip-profilevoice 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/2.0Via: SIP/2.0/UDP 200.52.198.15:5060;branch=z9hG4bK1BFB91A7EFrom: <sip:3396900084@200.52.198.15:5060>;tag=3DC26466-1A5FTo: MEGAFON <sip:3310862061@200.52.198.253:5151>;tag=3e3a03d7Date: Wed, 27 Feb 2013 18:52:42 GMTCall-ID: 000001957300063534215303142633222283261600@10.1.56.8Supported: timer,resource-priority,replaces,sdp-anatMin-SE: 360Cisco-Guid: 2932370470-2152010210-2968844771-4054721625User-Agent: Cisco-SIPGateway/IOS-12.xAllow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTERCSeq: 102 INVITEMax-Forwards: 70Timestamp: 1361991162Contact: <sip:3396900084@200.52.198.15:5060>Expires: 180Allow-Events: telephone-eventContent-Type: application/sdpContent-Length: 250v=0o=CiscoSystemsSIP-GW-UserAgent 1274 9443 IN IP4 200.52.198.15s=SIP Callc=IN IP4 200.52.198.15t=0 0m=audio 21846 RTP/AVP 0 101 19a=rtpmap:0 PCMU/8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-15a=rtpmap:19 CN/8000a=ptime:20
```

***Caveat**

SDP (RFC 2327)支持允许多条c线路，显示该多维数据集适当地实现功能。此解决方案示例起一个可能的解决方案作用对于不适当地支持RFC 2327的ITSP供应商。

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-profilevoice class sip-profiles 1000 request REINVITE sdp-header Audio-Connection-Info remove response 200 sdp-header Audio-Connection-Info remove
```

设置配置文件全局(语音服务VoIP)。

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

设置在一特定dial-peer的配置文件。在dial-peer应该设置这到/从PSTN。

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

参考本文， Cisco Unified Border Element (多维数据集)与SIP配置文件配置示例的会话初始化协议(SIP)标准化欲知更多信息。

SDP报头

这些是支持的SDP报头：

```
rtr(config-class)#response 200 sdp-header ? Attribute a= Audio-Attribute a= Audio-Bandwidth-Info b= Audio-Connection-Info c= Audio-Encryption-Key k= Audio-Media m=audio Audio-Session-Info I= Bandwidth-Key b= Connection-Info c= Email-Address e= Encrypt-Key k= Phone-Number p= Repeat-Times r= Session-Info I= Session-Name s= Session-Owner o= Time-Adjust-Key z= Time-Header t= Url-Descriptor u= Version v= Video-Attribute a= Video-Bandwidth-Info b= Video-Connection-Info c= Video-Encryption-Key k= Video-Media m=video Video-Session-Info I=
```

相关信息

- Cisco Unified Border Element (多维数据集)与SIP配置文件配置示例的会话初始化协议(SIP)标准化
- 技术支持和文档 - Cisco Systems