多维数据集第三方互操作性传真指南

Contents

Introduction

本文描述FAX over IP (FoIP)如何在与IP服务提供商的Cisco Unified Border Element (多维数据集)呼叫流运行。

Prerequisites

Requirements

Cisco 建议您了解以下主题：

- 多维数据集企业
- **[Media Gateway Control Protocol (MGCP)](#)**

- 会话初始化协议(SIP)
- H323协议组
- T30信令

Components Used

本文档中的信息基于以下软件和硬件版本： Cisco IOS版本12.4T，15.0M，15.0T，15.1M，15.1T，15.2M，15.2T，在Cisco集成服务路由器(ISR)系列2800的15.3T，3800，2900，3900，3900e或者Cisco AS5400XM通用网关

Note:此配置示例对列出的软件版本和硬件平台没有被限制这里。

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, make sure that you understand the potential impact of any command.

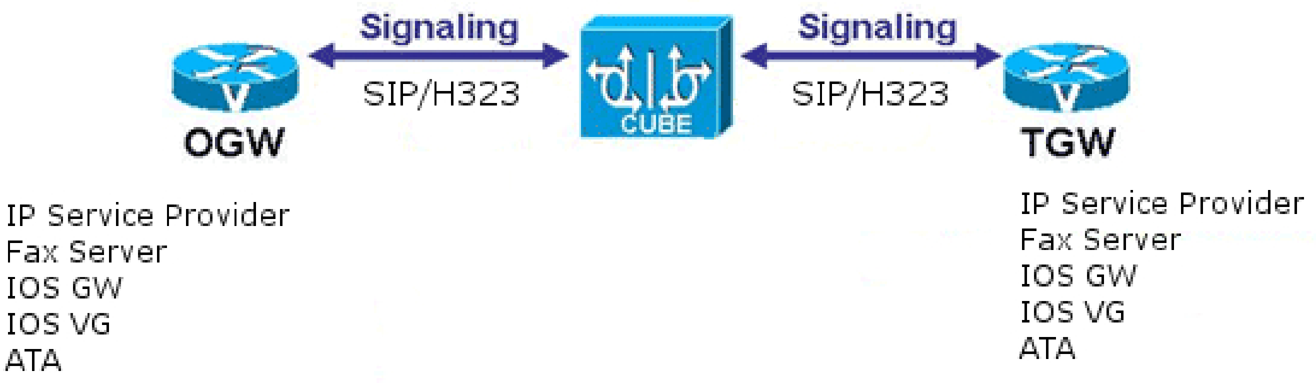
背景信息

与多维数据集的FoIP在一许多运行环境并且是被实施的为了利用可靠传真服务的当前VoIP网络。有多维数据集与一许多切换机制一起支持的多个传真协议。然而，在IP服务提供商中，您必须遵守电传由供应商支持在Cisco外面的协议和切换方法。

在FoIP呼叫流，多维数据集在终端网关(TGW)和始发网关(OGW)之间。从信令方面，多维数据集配置允许或者拒绝，从语音呼叫的切换到传真呼叫。 由于这样的事实FoIP协议是协商的端到端在VoIP环境里，重要的是从OGW到TGW配置一切为了使用同一个FoIP协议。

了解是重要的支持什么FoIP流，并且什么配置是必要的在多维数据集，以及TGWs和OGWs，为了保证可靠的传真通信。

多维数据集传真呼叫流



由于这样的事实IP服务提供商典型地有Cisco和非Cisco设备的混杂环境，重要的是您使用一个工业标准的方法为了从语音呼叫转换到传真呼叫。这意味着不可能使用命名的信令事件(NSE)，因为NSEs Cisco专利。有例外对此规则，但是他们是非常不常见的。

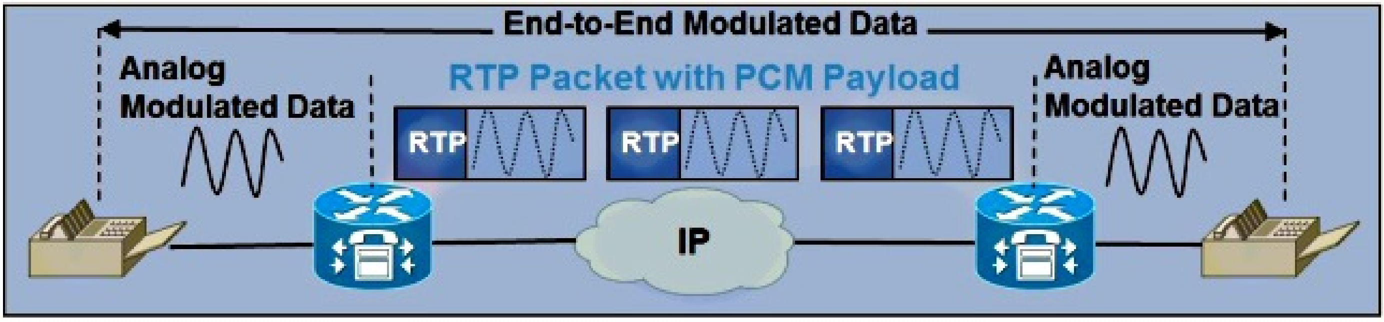
Note:使用一个基于协议的切换的无法意味着内部呼叫控制协议(SCCP)只用于对IP服务提供商的传真呼叫流与G711ulaw并且是“尽力”。

FoIP传输方法

本文讨论两FoIP传输方法、传真转接和T.38传真中继。

传真转接

传真转接是T30信号和页数据通过IP网络被传输作为脉冲代码调制的传真传输方法(PCM) -编码的数据，包裹在实时传输协议(RTP)帧。

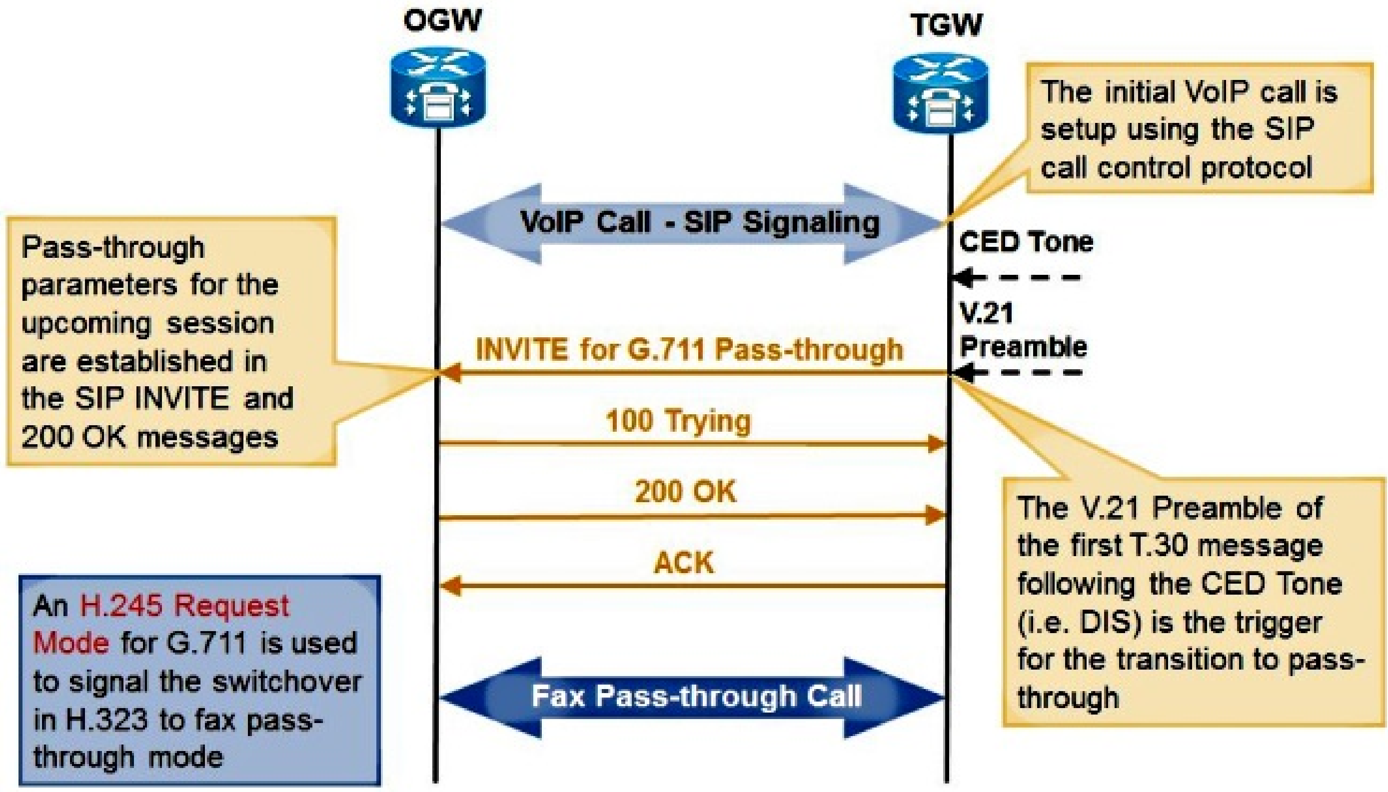


传真转接切换由V.21前导的检测TGW的触发。结果(为SIP)邀请或请求模式(H323)通过多维数据集和呼叫信令路径的其余被发送到OGW。

传真转接切换从所有语音编码转换到编码被定义在传真转接配置下(此进程在本文以后略述)。

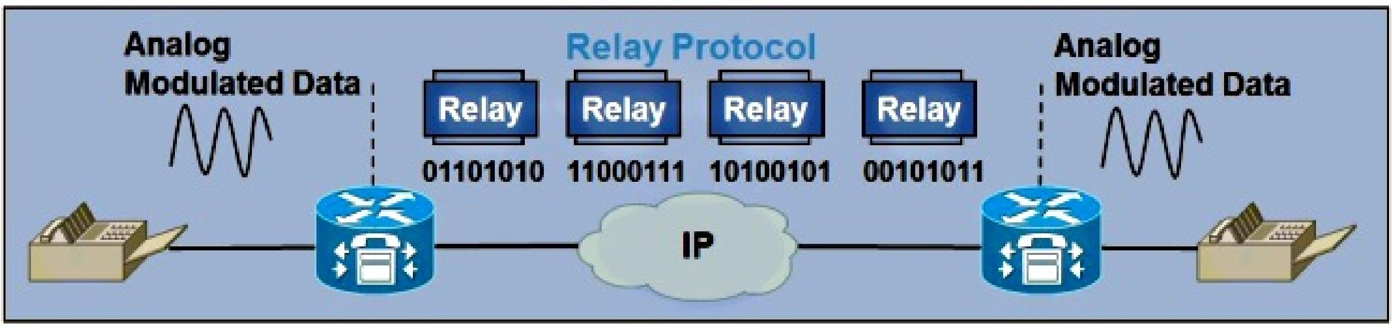
> **Note**:不可能配置MGCP网关为了起动upspeed到传真转接的G.711。所以，使用在多维数据集的转接终止到MGCP网关的所有传真必须路由与G.711编码。

> **Note**:如果最初的编码是G.711，不应该配置有传真转接H.323。当G.711已经协商时，造成一个H.245请求模式发送这换成G.711。CUCM回应H.245请求模式拒绝。



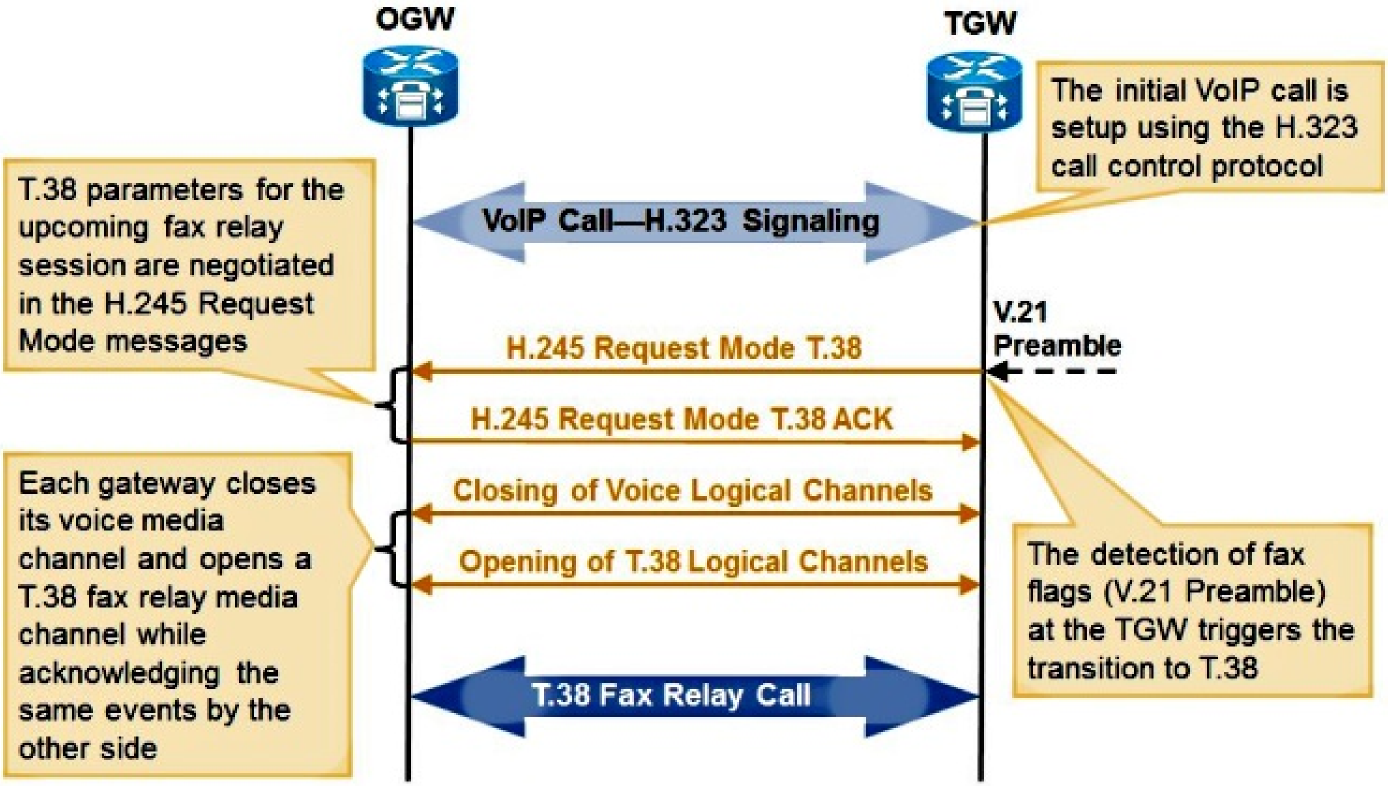
T.38传真中继

传真中继是TGWs和OGWs发现T30信号和页数据的传真传输方法。网关采取那些信号并且转换他们成中转消息，是模拟信号的数字表示法。 那些中转消息通过IP网络然后被发。



T.38传真中继切换由V.21前导的检测TGW的也触发。

- 当TGW运行与SIP时， V.21前导的检测触发T.38 ReINVITE (类似于什么以前被描述了)。
- 当TGW运行与H323时， V.21前导的检测触发T.38请求模式。
- 当TGW运行与MGCP时， V.21前导的检测触发通知(NTFY)，被发送到呼叫代理程序。 呼叫代理程序然后回应200 OK，并且发送请求模式或ReINVITE求立方，取决于使用的VoIP协议。



调试示例在本文的Troubleshoot部分。

多维数据集配置

多维数据集可以为FoIP被配置在两个地方：全局在**语音服务voip下**以及在拨号点下。在为一次特定呼叫匹配的拨号点的配置总是优先于全局配置。可以同时配置T.38的配置和传真转接，如果在不同的拨号点下，因此同时支持两个协议。

多维数据集转接配置

为了配置传真转接在**语音服务voip下**，请使用此命令(在粗体)：

```
voice service voip
no ip address trusted authenticate
allow-connections h323 to h323
allow-connections h323 to sip
allow-connections sip to h323
allow-connections sip to sip
 fax protocol pass-through g711ulaw
```

为了配置传真转接在拨号点，请使用此命令(在粗体)：

```
dial-peer voice 1 voip
description T38 Test
destination-pattern ^1000$
session protocol sipv2
session target ipv4:192.168.0.1
dtmf-relay rtp-nte
 fax protocol pass-through g711ulaw
no vad
```

Note:传真转接不是相同的象传真转接。 电传转接杠杆作用Cisco网络服务引擎(NSEs)为了从语音呼叫转换到传真呼叫。

求T.38配置的立方

Note:(超级G3传真加速) Cisco IOS版本15.1(1)T支持T.38版本3和以后。

为了配置T.38版本0 (G3传真速度)在**语音服务voip下**，请使用此命令(在粗体)：

```
voice service voip
no ip address trusted authenticate
allow-connections h323 to h323
allow-connections h323 to sip
allow-connections sip to h323
allow-connections sip to sip
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
```

为了配置T.38在拨号点，请使用此命令(在粗体)：

```
dial-peer voice 1 voip
description T38 Test
destination-pattern ^1000$
session protocol sipv2
session target ipv4:192.168.0.1
dtmf-relay rtp-nte
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
```

为了配置T.38版本3，在语音服务VoIP下或在拨号点，使用此命令：

```
fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none
```

如果使用媒体传输协议(MTP)，当相互作用通过多维数据集时，必须支持编码转接。Cisco Unified通信管理器(CUCM) MTP支持版本8.6.1和以上的编码转接。Cisco IOS MTP必须有在数字式信号处理器(DSP)农厂配置的编码转接：

```
dspfarm profile 2 mtp
 codec pass-through
 codec g729r8
 maximum sessions software 50
 associate application SCCP
```

Time Division Multiplexing (TDM)相互作用的网关配置与多维数据集

对于SCCP被控制的TDM网关，此配置使用传真转接。

```
voice service voip
no modem passthrough
 fax protocol none
no fax-relay sg3-to-g3
```

Note:编码在这个区域设置为此相互作用的必须是G.711。当相互作用如注释以前，不可能设

置SCCP网关以多维数据集时，使用T.38。

为了用多维数据集配置相互作用SIP和H.323 TDM的网关的传真转接，请进入：

```
voice service voip
 no modem passthrough
 no fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
```

为了用多维数据集配置相互作用SIP和H.323 TDM的网关的T.38，请进入：

```
voice service voip
no modem passthrough
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
```

Note:T.38版本3，如果在多维数据集被配置和SIP服务提供商，支持可以使用。

为了用多维数据集配置inteworking传真的转接的一个MGCP TDM网关，请进入：

```
no mgcp fax-relay sg3-to-g3
no mgcp package fxr-package
mgcp fax t38 inhibit
no mgcp modem passthrough voip mode nse
```

Note:因为MGCP网关不支持传真转接的加速，在CUCM的地区在MGCP网关和多维数据集之间必须有G.711编码。

Verify

当前没有可用于此配置的验证过程。

Troubleshoot

为了排除在多维数据集的此问题故障，这些调试一定是启用的。

SIP

Enable (event) SIP的这些调试：

```
no mgcp fax-relay sg3-to-g3
no mgcp package fxr-package
mgcp fax t38 inhibit
no mgcp modem passthrough voip mode nse
```

在语音呼叫设置后， TGW发送SIP ReINVITE到OGW通过多维数据集。如果切换是成功的，OGW回应与正确的会话描述协议(SDP)参数的SIP 200 OK。

T.38切换

```
INVITE sip:2101@10.0.0.1:5060 SIP/2.0
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK171D71
Remote-Party-ID: <sip:1101@10.0.0.2>;party=calling;screen=no;privacy=off
From: <sip:8141101@10.0.0.2>;tag=8D815D8-646
To: <sip:2101@10.0.0.1>;tag=DD4D344-21B2
Date: Fri, 25 Feb 2011 19:25:15 GMT
Call-ID: 32395B08-403E11E0-818C9D5B-499FBE40@10.0.0.1
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 786980147-1077809632-2173148507-1235205696
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: 1298661915
Contact: <sip:8141101@10.0.0.2:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 384

v=0
o=CiscoSystemsSIP-GW-UserAgent 3745 9509 IN IP4 10.0.0.2
s=SIP Call
c=IN IP4 10.0.0.2
t=0 0
m=image 17682 udptl t38
c=IN IP4 10.0.0.2
a=T38FaxVersion:0
a=T38MaxBitRate:7200
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:200
a=T38FaxMaxDatagram:180
a=T38FaxUdpEC:t38UDPRedundancy
```

!!NOTE!! Not all of the above bolded fields are required.
The above is an example of how Cisco implements T38.

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK171D71
From: <sip:8141101@10.0.0.2>;tag=8D815D8-646
To: <sip:2101@10.0.0.1>;tag=DD4D344-21B2
Date: Fri, 25 Feb 2011 17:48:05 GMT
Call-ID: 32395B08-403E11E0-818C9D5B-499FBE40@10.0.0.1
CSeq: 101 INVITE
Allow-Events: telephone-event
Server: Cisco-SIPGateway/IOS-12.x
Content-Length: 0

176443: Feb 25 17:48:05.360:
//134/2EE85D338187/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK171D71
From: <sip:8141101@10.0.0.2>;tag=8D815D8-646
To: <sip:2101@10.0.0.1>;tag=DD4D344-21B2
Date: Fri, 25 Feb 2011 17:48:05 GMT
Call-ID: 32395B08-403E11E0-818C9D5B-499FBE40@10.0.0.1
```

```
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE,
NOTIFY, INFO, REGISTER
Allow-Events: telephone-event
Remote-Party-ID: <sip:2101@10.0.0.1>
;party=called;screen=no;privacy=off
Contact: <sip:2101@10.0.0.1:5060>
Supported: replaces
Supported: sdp-anat
Server: Cisco-SIPGateway/IOS-12.x
Supported: timer
Content-Type: application/sdp
Content-Length: 384

v=0
o=CiscoSystemsSIP-GW-UserAgent 5552 9399 IN IP4 10.0.0.1
s=SIP Call
c=IN IP4 10.0.0.1
t=0 0
m=image 16710 udptl t38
c=IN IP4 10.0.0.1
a=T38FaxVersion:0
a=T38MaxBitRate:7200
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:200
a=T38FaxMaxDatagram:320
a=T38FaxUdpEC:t38UDPRedundancy

ACK sip:2101@10.0.0.1:5060 SIP/2.0
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK181B79
From: <sip:8141101@10.0.0.2>;tag=8D815D8-646
To: <sip:2101@10.0.0.1>;tag=DD4D344-21B2
Date: Fri, 25 Feb 2011 19:25:15 GMT
Call-ID: 32395B08-403E11E0-818C9D5B-499FBE40@10.0.0.1
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: telephone-event
Content-Length: 0
```

传真转接切换

```
INVITE sip:2101@10.0.0.1:5060 SIP/2.0
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK154F2
Remote-Party-ID: <sip:1101@10.0.0.2>;party=calling;screen=no;privacy=off
From: <sip:8131101@10.0.0.2>;tag=8D66B94-7BF
To: <sip:2101@10.0.0.1>;tag=DD32900-5D4
Date: Fri, 25 Feb 2011 19:23:25 GMT
Call-ID: F12F0BBB-403D11E0-81869D5B-499FBE40@10.0.0.1
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 3990792353-1077744096-2172755291-1235205696
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: 1298661805
Contact: <sip:8131101@10.0.0.2:5060>
Expires: 180
Allow-Events: telephone-event
```

```
Content-Type: application/sdp
Content-Length: 174

v=0
o=CiscoSystemsSIP-GW-UserAgent 107 1892 IN IP4 10.0.0.2
s=SIP Call
c=IN IP4 10.0.0.2
t=0 0
m=audio 16464 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=silenceSupp:off - - - -

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK154F2
From: <sip:8131101@10.0.0.2>;tag=8D66B94-7BF
To: <sip:2101@10.0.0.1>;tag=DD32900-5D4
Date: Fri, 25 Feb 2011 17:46:16 GMT
Call-ID: F12F0BBB-403D11E0-81869D5B-499FBE40@10.0.0.1
CSeq: 101 INVITE
Allow-Events: telephone-event
Server: Cisco-SIPGateway/IOS-12.x
Content-Length: 0

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK154F2
From: <sip:8131101@10.0.0.2>;tag=8D66B94-7BF
To: <sip:2101@10.0.0.1>;tag=DD32900-5D4
Date: Fri, 25 Feb 2011 17:46:16 GMT
Call-ID: F12F0BBB-403D11E0-81869D5B-499FBE40@10.0.0.1
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE,
NOTIFY, INFO, REGISTER
Allow-Events: telephone-event
Remote-Party-ID: <sip:2101@10.0.0.1>;party=called;screen=no;privacy=off
Contact: <sip:2101@10.0.0.1:5060>
Supported: replaces
Supported: sdp-anat
Server: Cisco-SIPGateway/IOS-12.x
Supported: timer
Content-Type: application/sdp
Content-Length: 194

v=0
o=CiscoSystemsSIP-GW-UserAgent 4896 2709 IN IP4 10.0.0.1
s=SIP Call
c=IN IP4 10.0.0.1
t=0 0
m=audio 19054 RTP/AVP 0
c=IN IP4 10.0.0.1
a=rtpmap:0 PCMU/8000
a=silenceSupp:off - - - -

ACK sip:2101@10.0.0.1:5060 SIP/2.0
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK16A56
From: <sip:8131101@10.0.0.2>;tag=8D66B94-7BF
To: <sip:2101@10.0.0.1>;tag=DD32900-5D4
Date: Fri, 25 Feb 2011 19:23:25 GMT
Call-ID: F12F0BBB-403D11E0-81869D5B-499FBE40@10.0.0.1
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: telephone-event
Content-Length: 0
```

H323

Enable (event) H323的这些调试：

```
INVITE sip:2101@10.0.0.1:5060 SIP/2.0
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK154F2
Remote-Party-ID: <sip:1101@10.0.0.2>;party=calling;screen=no;privacy=off
From: <sip:8131101@10.0.0.2>;tag=8D66B94-7BF
To: <sip:2101@10.0.0.1>;tag=DD32900-5D4
Date: Fri, 25 Feb 2011 19:23:25 GMT
Call-ID: F12F0BBB-403D11E0-81869D5B-499FBE40@10.0.0.1
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 3990792353-1077744096-2172755291-1235205696
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: 1298661805
Contact: <sip:8131101@10.0.0.2:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 174

v=0
o=CiscoSystemsSIP-GW-UserAgent 107 1892 IN IP4 10.0.0.2
s=SIP Call
c=IN IP4 10.0.0.2
t=0 0
m=audio 16464 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=silenceSupp:off - - - -

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK154F2
From: <sip:8131101@10.0.0.2>;tag=8D66B94-7BF
To: <sip:2101@10.0.0.1>;tag=DD32900-5D4
Date: Fri, 25 Feb 2011 17:46:16 GMT
Call-ID: F12F0BBB-403D11E0-81869D5B-499FBE40@10.0.0.1
CSeq: 101 INVITE
Allow-Events: telephone-event
Server: Cisco-SIPGateway/IOS-12.x
Content-Length: 0

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK154F2
From: <sip:8131101@10.0.0.2>;tag=8D66B94-7BF
To: <sip:2101@10.0.0.1>;tag=DD32900-5D4
Date: Fri, 25 Feb 2011 17:46:16 GMT
Call-ID: F12F0BBB-403D11E0-81869D5B-499FBE40@10.0.0.1
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE,
NOTIFY, INFO, REGISTER
Allow-Events: telephone-event
Remote-Party-ID: <sip:2101@10.0.0.1>;party=called;screen=no;privacy=off
Contact: <sip:2101@10.0.0.1:5060>
Supported: replaces
Supported: sdp-anat
Server: Cisco-SIPGateway/IOS-12.x
Supported: timer
Content-Type: application/sdp
Content-Length: 194
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 4896 2709 IN IP4 10.0.0.1
s=SIP Call
c=IN IP4 10.0.0.1
t=0 0
m=audio 19054 RTP/AVP 0
c=IN IP4 10.0.0.1
a=rtpmap:0 PCMU/8000
a=silenceSupp:off - - - -

ACK sip:2101@10.0.0.1:5060 SIP/2.0
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK16A56
From: <sip:8131101@10.0.0.2>;tag=8D66B94-7BF
To: <sip:2101@10.0.0.1>;tag=DD32900-5D4
Date: Fri, 25 Feb 2011 19:23:25 GMT
Call-ID: F12F0BBB-403D11E0-81869D5B-499FBE40@10.0.0.1
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: telephone-event
Content-Length: 0
```

在语音呼叫设置后，TGW发送H245 RequestMode到OGW通过多维数据集。如果切换是成功的，OGW回应RequestModeAck。

T.38切换

```
INVITE sip:2101@10.0.0.1:5060 SIP/2.0
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK154F2
Remote-Party-ID: <sip:1101@10.0.0.2>;party=calling;screen=no;privacy=off
From: <sip:8131101@10.0.0.2>;tag=8D66B94-7BF
To: <sip:2101@10.0.0.1>;tag=DD32900-5D4
Date: Fri, 25 Feb 2011 19:23:25 GMT
Call-ID: F12F0BBB-403D11E0-81869D5B-499FBE40@10.0.0.1
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 3990792353-1077744096-2172755291-1235205696
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: 1298661805
Contact: <sip:8131101@10.0.0.2:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 174

v=0
o=CiscoSystemsSIP-GW-UserAgent 107 1892 IN IP4 10.0.0.2
s=SIP Call
c=IN IP4 10.0.0.2
t=0 0
m=audio 16464 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=silenceSupp:off - - - -

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK154F2
From: <sip:8131101@10.0.0.2>;tag=8D66B94-7BF
To: <sip:2101@10.0.0.1>;tag=DD32900-5D4
```

```
Date: Fri, 25 Feb 2011 17:46:16 GMT
Call-ID: F12F0BBB-403D11E0-81869D5B-499FBE40@10.0.0.1
CSeq: 101 INVITE
Allow-Events: telephone-event
Server: Cisco-SIPGateway/IOS-12.x
Content-Length: 0

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK154F2
From: <sip:8131101@10.0.0.2>;tag=8D66B94-7BF
To: <sip:2101@10.0.0.1>;tag=DD32900-5D4
Date: Fri, 25 Feb 2011 17:46:16 GMT
Call-ID: F12F0BBB-403D11E0-81869D5B-499FBE40@10.0.0.1
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE,
NOTIFY, INFO, REGISTER
Allow-Events: telephone-event
Remote-Party-ID: <sip:2101@10.0.0.1>;party=called;screen=no;privacy=off
Contact: <sip:2101@10.0.0.1:5060>
Supported: replaces
Supported: sdp-anat
Server: Cisco-SIPGateway/IOS-12.x
Supported: timer
Content-Type: application/sdp
Content-Length: 194

v=0
o=CiscoSystemsSIP-GW-UserAgent 4896 2709 IN IP4 10.0.0.1
s=SIP Call
c=IN IP4 10.0.0.1
t=0 0
m=audio 19054 RTP/AVP 0
c=IN IP4 10.0.0.1
a=rtpmap:0 PCMU/8000
a=silenceSupp:off - - - -

ACK sip:2101@10.0.0.1:5060 SIP/2.0
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK16A56
From: <sip:8131101@10.0.0.2>;tag=8D66B94-7BF
To: <sip:2101@10.0.0.1>;tag=DD32900-5D4
Date: Fri, 25 Feb 2011 19:23:25 GMT
Call-ID: F12F0BBB-403D11E0-81869D5B-499FBE40@10.0.0.1
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: telephone-event
Content-Length: 0
```

传真转接切换

```
INVITE sip:2101@10.0.0.1:5060 SIP/2.0
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK154F2
Remote-Party-ID: <sip:1101@10.0.0.2>;party=calling;screen=no;privacy=off
From: <sip:8131101@10.0.0.2>;tag=8D66B94-7BF
To: <sip:2101@10.0.0.1>;tag=DD32900-5D4
Date: Fri, 25 Feb 2011 19:23:25 GMT
Call-ID: F12F0BBB-403D11E0-81869D5B-499FBE40@10.0.0.1
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 3990792353-1077744096-2172755291-1235205696
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: 1298661805
Contact: <sip:8131101@10.0.0.2:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 174

v=0
o=CiscoSystemsSIP-GW-UserAgent 107 1892 IN IP4 10.0.0.2
s=SIP Call
c=IN IP4 10.0.0.2
t=0 0
m=audio 16464 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=silenceSupp:off - - - -

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK154F2
From: <sip:8131101@10.0.0.2>;tag=8D66B94-7BF
To: <sip:2101@10.0.0.1>;tag=DD32900-5D4
Date: Fri, 25 Feb 2011 17:46:16 GMT
Call-ID: F12F0BBB-403D11E0-81869D5B-499FBE40@10.0.0.1
CSeq: 101 INVITE
Allow-Events: telephone-event
Server: Cisco-SIPGateway/IOS-12.x
Content-Length: 0

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK154F2
From: <sip:8131101@10.0.0.2>;tag=8D66B94-7BF
To: <sip:2101@10.0.0.1>;tag=DD32900-5D4
Date: Fri, 25 Feb 2011 17:46:16 GMT
Call-ID: F12F0BBB-403D11E0-81869D5B-499FBE40@10.0.0.1
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE,
NOTIFY, INFO, REGISTER
Allow-Events: telephone-event
Remote-Party-ID: <sip:2101@10.0.0.1>;party=called;screen=no;privacy=off
Contact: <sip:2101@10.0.0.1:5060>
Supported: replaces
Supported: sdp-anat
Server: Cisco-SIPGateway/IOS-12.x
Supported: timer
Content-Type: application/sdp
Content-Length: 194

v=0
o=CiscoSystemsSIP-GW-UserAgent 4896 2709 IN IP4 10.0.0.1
s=SIP Call
c=IN IP4 10.0.0.1
t=0 0
m=audio 19054 RTP/AVP 0
c=IN IP4 10.0.0.1
a=rtpmap:0 PCMU/8000
a=silenceSupp:off - - - -

ACK sip:2101@10.0.0.1:5060 SIP/2.0
Via: SIP/2.0/UDP 10.0.0.2:5060;branch=z9hG4bK16A56
From: <sip:8131101@10.0.0.2>;tag=8D66B94-7BF
To: <sip:2101@10.0.0.1>;tag=DD32900-5D4
Date: Fri, 25 Feb 2011 19:23:25 GMT
Call-ID: F12F0BBB-403D11E0-81869D5B-499FBE40@10.0.0.1
Max-Forwards: 70
```

```
CSeq: 101 ACK
Allow-Events: telephone-event
Content-Length: 0
```

症状1 ：多维数据集拒绝与488的ReINVITE

如果遇到此问题，请完成这些步骤：

1. 关闭调试和为测试通话收集。
2. 验证配置得T.38或传真转接全局。
3. 如果没有配置得T.38或传真转接全局，请保证T.38或传真转接被配置在根据呼叫控制应用编程接口(CCAPI)调试的流入和流出的拨号点下。
4. 如果仍然没有解决问题， enable (event)**调试ccsip全部**(在与**logging buffered 5000000调试的**一台日志缓冲器)为了确定SIP为什么拒绝此ReINVITE。

症状2 ： 多维数据集拒绝RequestMode和RequestModeReject

如果遇到此问题，请完成这些步骤：

1. 关闭调试和为测试通话收集。
2. 验证配置得T.38或传真转接全局。
3. 如果没有配置得T.38或传真转接全局，请保证T.38或传真转接被配置在根据CCAPI调试的流入和流出的拨号点下。
4. 如果仍然没有解决问题， enable (event) **debug h225 events**、**debug h225 q931**和**debug h245 events**为了确定H323为什么拒绝此RequestMode。

卖方细节信息

韦里孙

- Cisco技术支持中心(TAC)注意，虽然韦里孙T.38的要求技术支持在SIP，他们**从未**初始化从语音呼叫的一个切换到T.38，当他们运行在TGW时。
- 这是一个已知限制在他们的环境里，并且没看来他们修正它。
- 当OGW是FoIP服务器时，您能通常设置服务器初始化切换，既使当它是OGW。
- 当Cisco GW是OGW，现在没有办法强加切换，当Cisco GW作为OGW。
- 当Cisco GW是OGW时， Cisco Bug ID CSCud72998是增强请求支持T.38切换。

Related Information

- 配置传真转接
- 配置T.38传真中继
- 了解配比在IOS平台的Inbound和Outbound Dial Peer
- Technical Support & Documentation - Cisco Systems