

# 通过Collaboration Solutions Analyzer排除CUBE故障

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## 简介

本文档介绍用于使用协作解决方案分析器门户对CUBE进行故障排除的日志分析器和SIP配置文件测试器工具。

## 要求

Cisco 建议您了解以下主题：

- 思科统一边界元素(CUBE)企业版。
- 会话初始协议(SIP)。
- CUBE日志收集 ( 调试 ) 。

## 快速入门

Collaboration Solutions Analyzer (CSA)是一套工具，用于在协作解决方案的整个生命周期内为其提供支持。它有助于确定问题并在需要时提供纠正性行动计划，在协作解决方案的每个阶段提供帮助。

导航至协作解决方案分析器(<https://cway.cisco.com/csa-new/#/home>)

 注意：使用Chrome浏览器可确保工具以最佳状态运行。

## 考虑事项

这些工具专为处理SIP到SIP呼叫的CUBE设备设计。这些工具不支持任何其他语音协议。

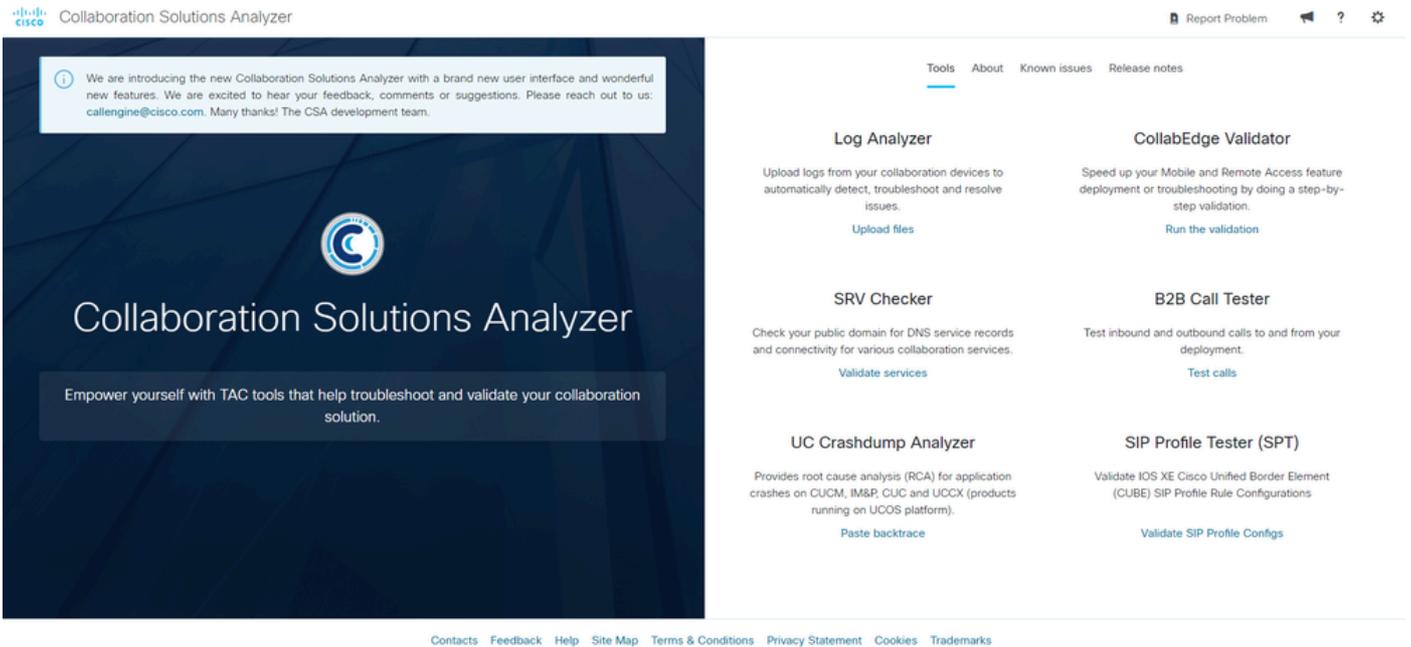
日志分析器使用CUBE日志（基于SIP消息调试）进行分析。

如需其他语音协议方面的帮助，请使用<https://supportassistant.cisco.com>上的思科支持助理进行TAC活动

## 平台说明

CSA平台提供以下CUBE工具：

- 日志分析器 -从CUBE和其他协作设备上传日志，以自动检测、故障排除和解决问题。
- SIP配置文件测试器 -验证SIP配置文件配置。



Collaboration Solutions Analyzer

Report Problem

Tools About Known issues Release notes

**Log Analyzer**  
Upload logs from your collaboration devices to automatically detect, troubleshoot and resolve issues.  
Upload files

**CollabEdge Validator**  
Speed up your Mobile and Remote Access feature deployment or troubleshooting by doing a step-by-step validation.  
Run the validation

**SRV Checker**  
Check your public domain for DNS service records and connectivity for various collaboration services.  
Validate services

**B2B Call Tester**  
Test inbound and outbound calls to and from your deployment.  
Test calls

**UC Crashdump Analyzer**  
Provides root cause analysis (RCA) for application crashes on CUCM, IM&P, CUC and UCCX (products running on UCOS platform).  
Paste backtrace

**SIP Profile Tester (SPT)**  
Validate IOS XE Cisco Unified Border Element (CUBE) SIP Profile Rule Configurations  
Validate SIP Profile Configs

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CSA主页

## 日志分析器

管理员可以通过日志分析器工具检查CUBE设备处理的呼叫信令。它提供日志文件的全面分析，包括：

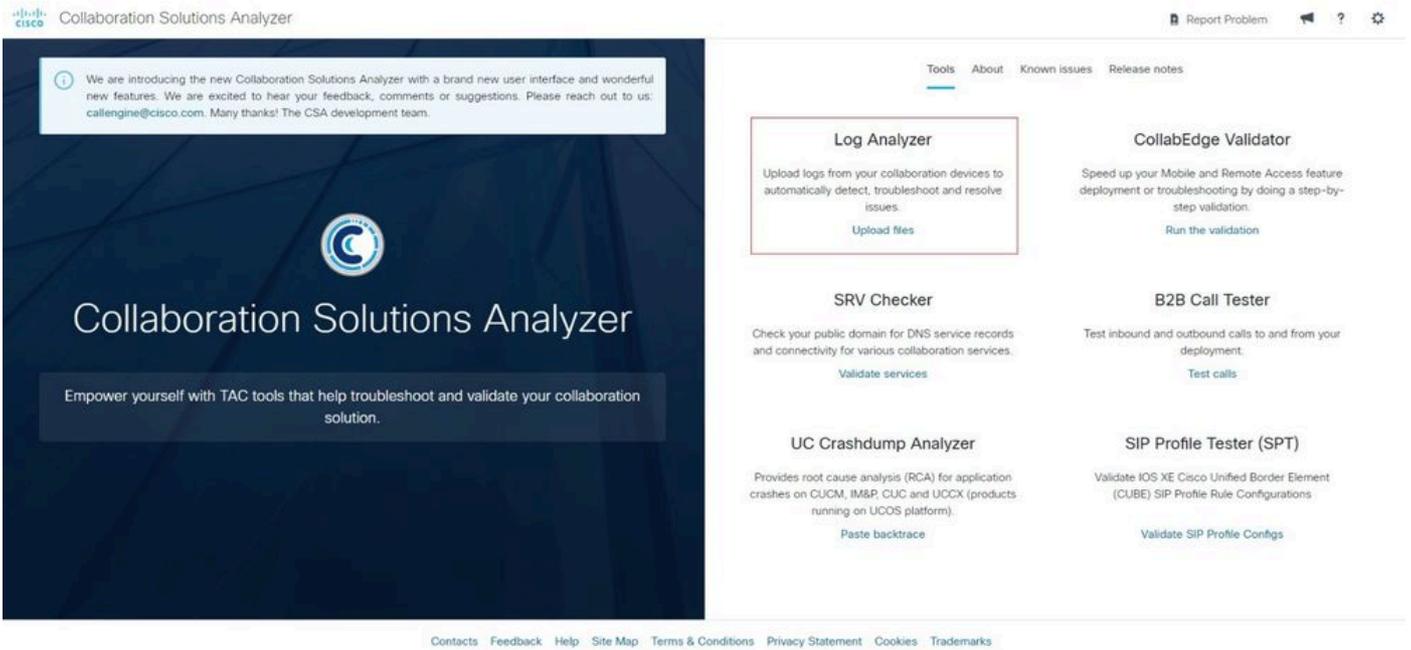
- 呼叫段信息
- 梯形图
- 信令

 注意：必须先收集由CUBE处理的呼叫中的CUBE调试(debug ccsip messages)，并将其存储在文本文件中。此文本文件中必须包括SIP调试以及任何其他输出（例如show命令）。

上传CUBE日志文件

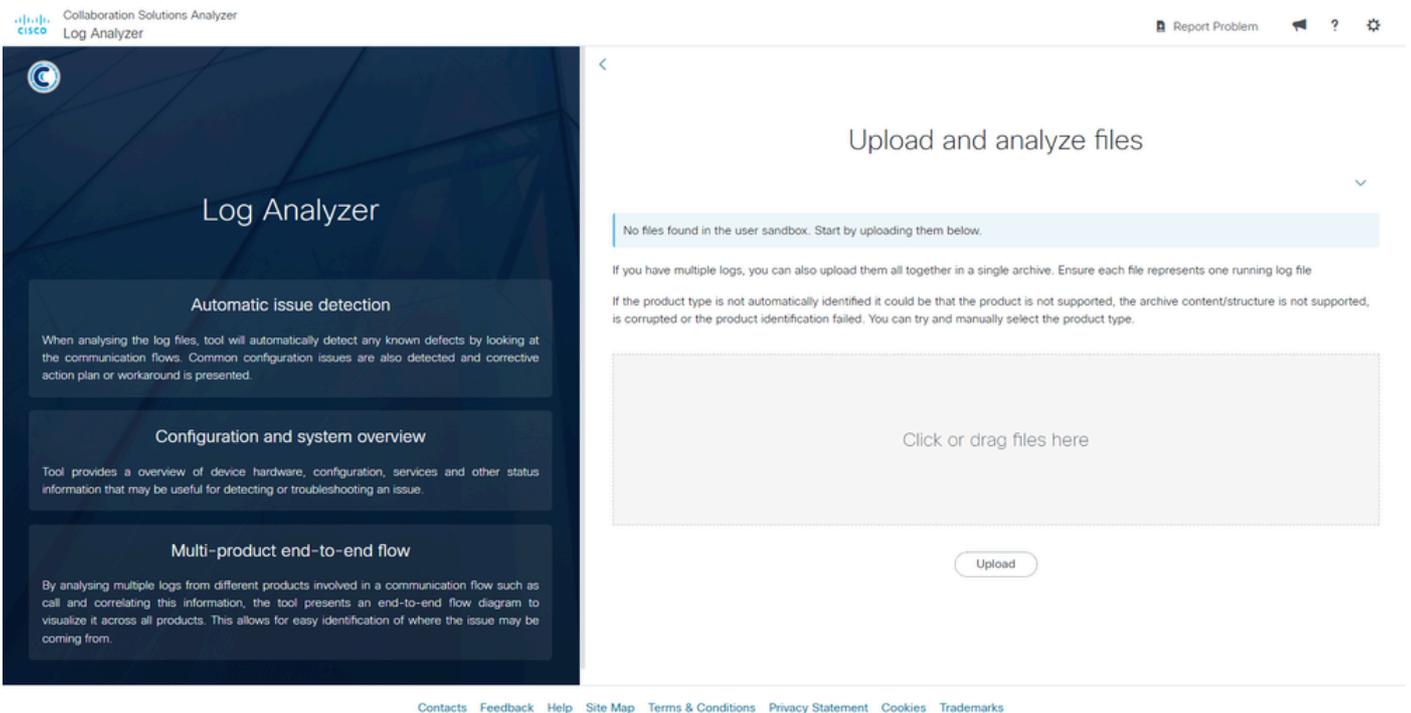
导航至协作解决方案分析器(<https://cway.cisco.com/csa-new/#/home>)

然后单击日志分析器部分中的上传文件以选择工具。

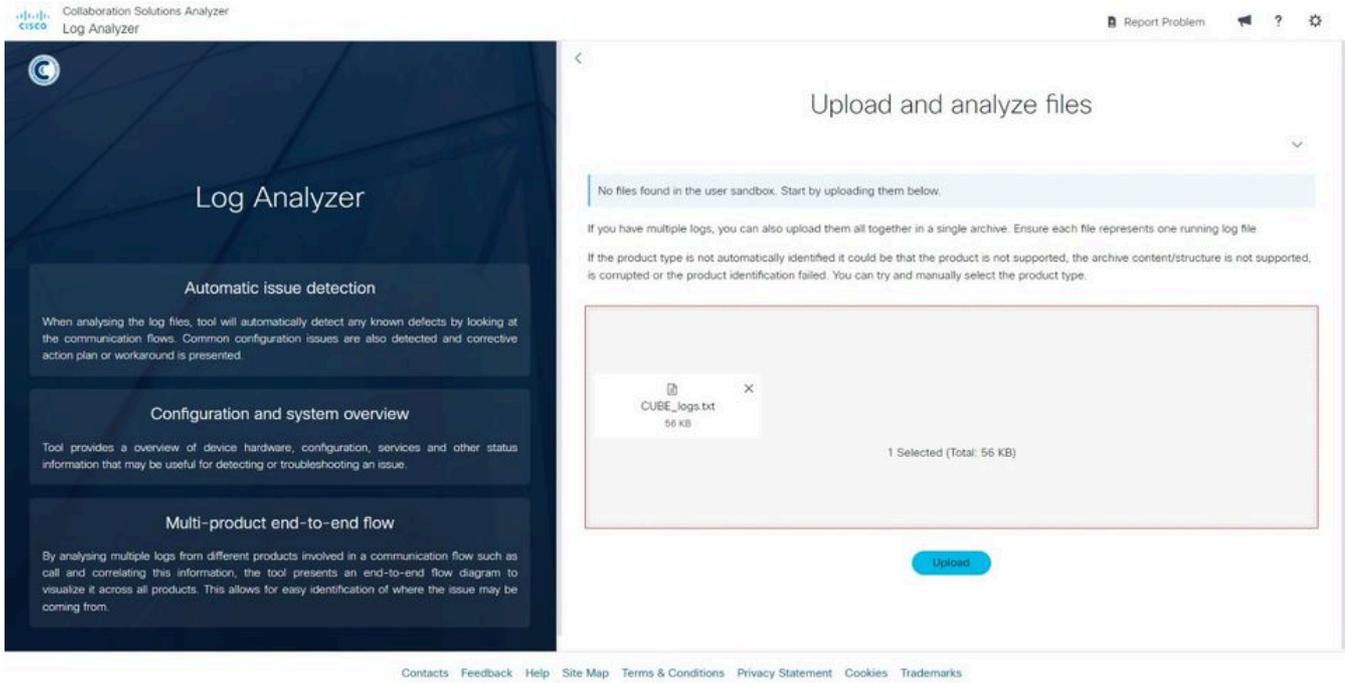


日志分析器主页

平台显示工具屏幕，可在其中选择或拖动文件。

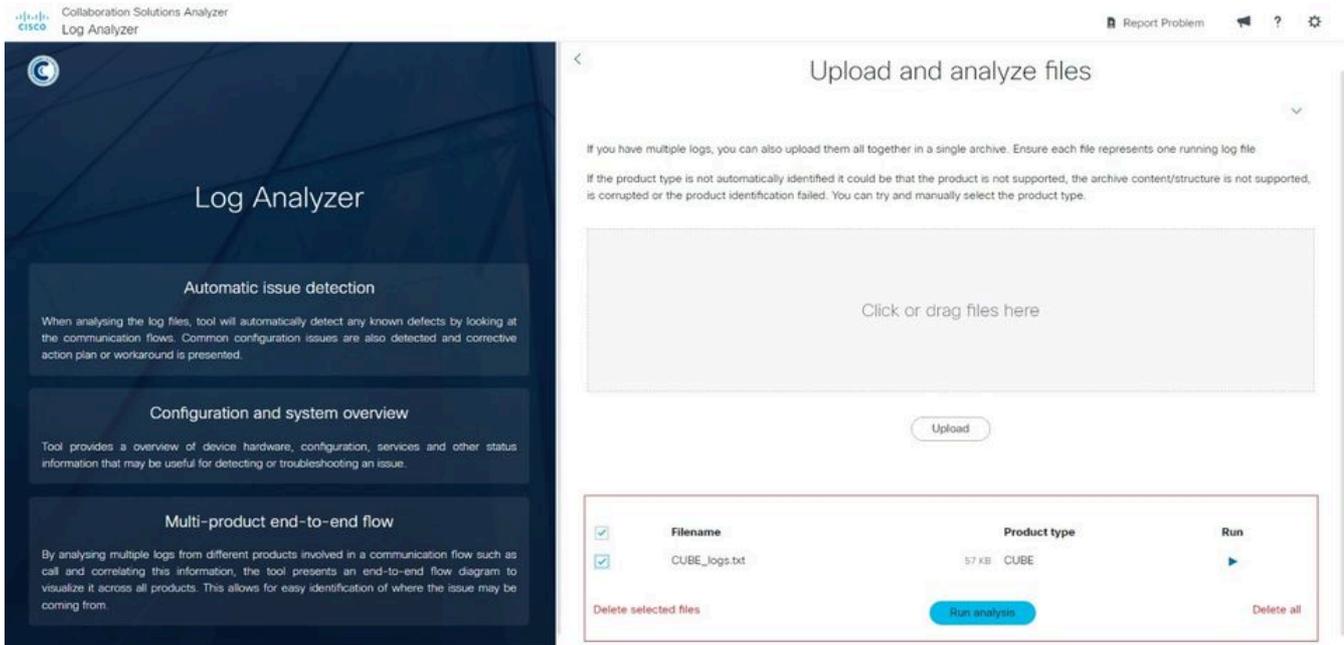


要完成上传文件以供工具分析的过程，请点击上传按钮。



将文件上传到工具后，通过选中相应的框来选择要分析的文件，然后单击Run Analysis按钮。

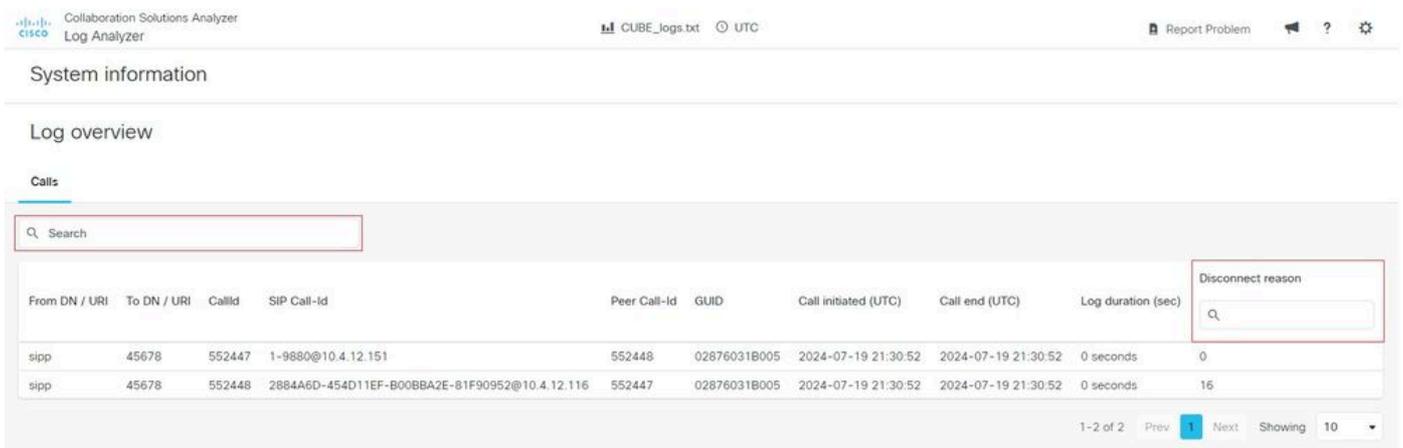
- 系统将产品类型设置为CUBE。
- 可以在同一会话中分析多个文件。



### 日志分析器产品类型

该工具分析在文本文件中捕获的所有信令呼叫，并显示所识别呼叫段的摘要。然后，可以应用两个过滤器：

- 搜索- 按特定数据（如拨叫号码）过滤呼叫会话。
- 按“断开原因”搜索- 根据呼叫断开原因过滤呼叫会话。



### 日志分析器调用过滤器

要继续详细分析，请选择您要关注的呼叫会话线路，该工具将显示展示呼叫分支信息、梯形图和信

令的完整分析。

## 呼叫段信息

第一阶段显示呼叫段信息，其中显示呼叫的概述：

- SIP呼叫段类型
- 发件人 -从INVITE消息的SIP报头获取。
- To -从INVITE消息的“TO SIP”报头获取。
- 信令源 -源设备的IP地址和端口。从INVITE消息的VIA SIP报头获取。
- 信令目标 -目标设备的IP地址和端口。从INVITE消息的URI SIP报头获取。
- 呼叫ID -从INVITE消息的SIP呼叫ID报头获取。
- 呼叫段连接- 呼叫会话时间戳。

The screenshot displays the Cisco Collaboration Solutions Analyzer Log Analyzer interface. The top navigation bar includes the Cisco logo, the text "Collaboration Solutions Analyzer Log Analyzer", and several utility icons. Below the navigation bar, there are three tabs: "Call leg info", "Ladder diagram", and "Signalling". The main content area is divided into two sections, each with a dark blue header and a table of general information.

**SIP - outgoing** (Ladder tags, Use for signaling and ladder)

General information	
SIP call leg type	Call
From	sipp@10.4.12.116
To	45678@10.4.12.151
Signaling source	10.4.12.116 : 5060
Signaling destination	10.4.12.151 : 5060
Call ID	2884A6D-454D11EF-B00BBA2E-81F90952@10.4.12.116
Call leg connects	✓ 2024-07-19 21:30:52 UTC

**SIP - incoming** (Ladder tags, Use for signaling and ladder)

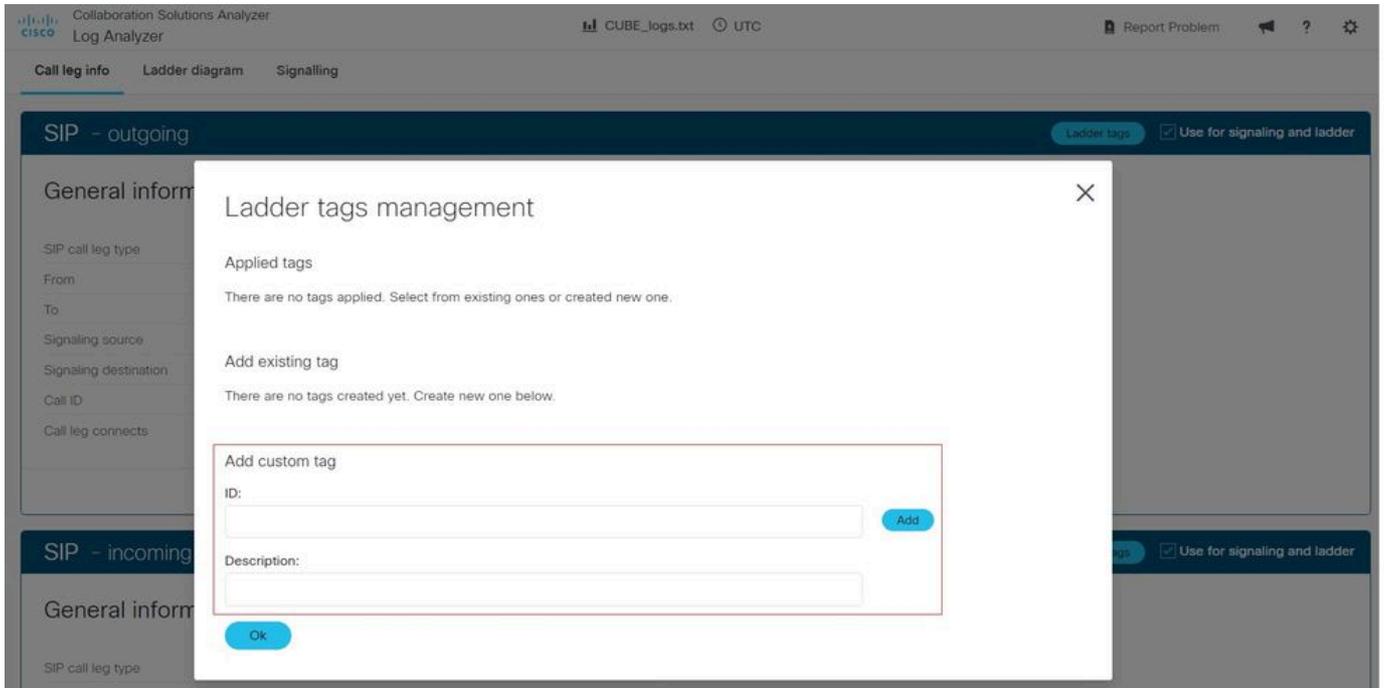
General information	
SIP call leg type	Call
From	sipp@10.4.12.151:5061
To	45678@10.4.12.116:5060
Signaling source	10.4.12.151 : 5061
Signaling destination	10.4.12.116 : 5060
Call ID	1-9880@10.4.12.151

日志分析器呼叫段信息

在此部分，可以启用梯形标记以突出显示梯形图中的消息。该应用程序有2个字段：

- ID -输入要突出显示的特定参数。
- Description -添加参数的说明。

单击Add按钮完成此过程。



日志分析器梯形标签

## 梯形图

在第二阶段，显示梯形图，直观地描述呼叫过程中交换的SIP消息。这些消息采用颜色编码，以方便识别：

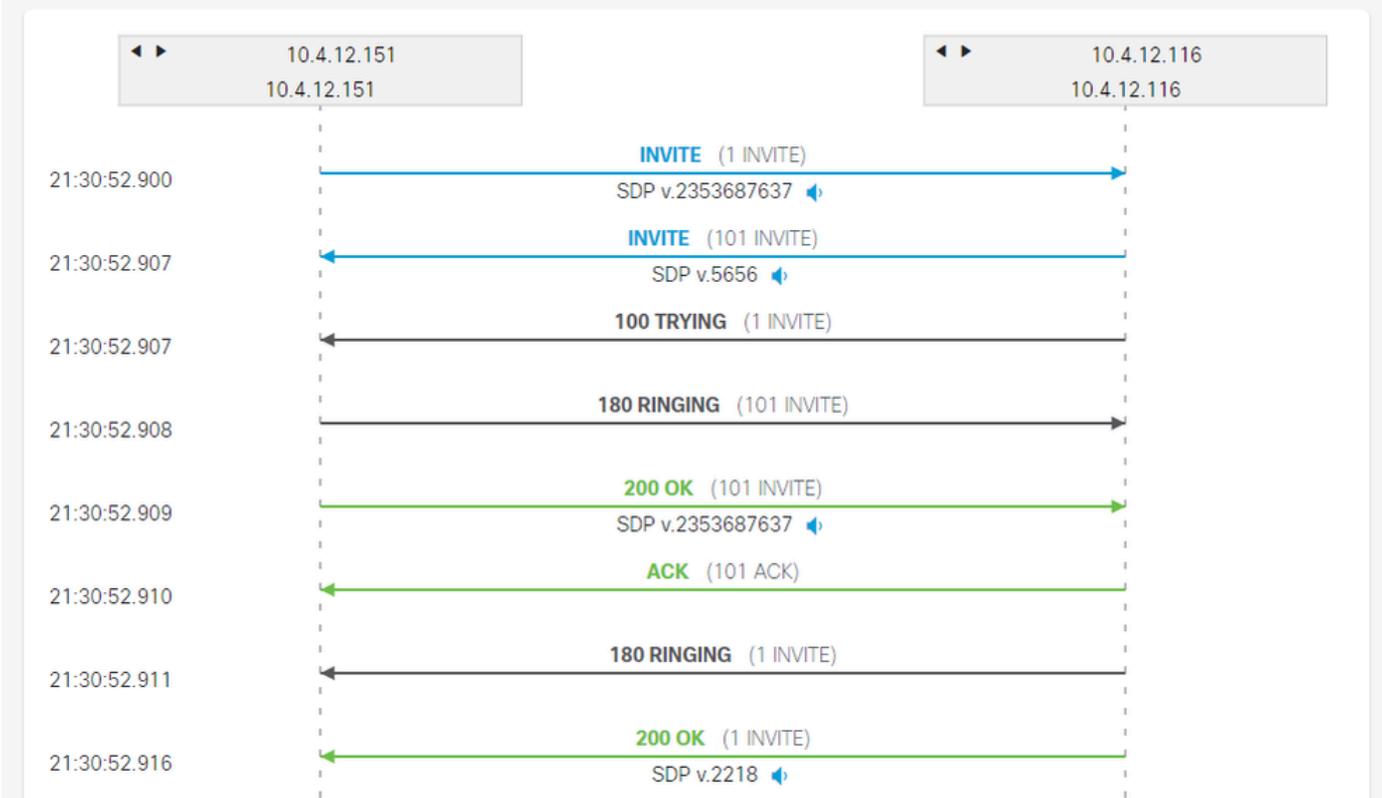
- 蓝色- SIP INVITE消息。
- 绿色 - SIP 200 OK和ACK消息。
- 红色- SIP BYE消息。

要下载图的副本，请单击Download Ladder按钮。系统随即会下载该图并将其保存为PNG图像文件。请注意，此选项仅在使用Google Chrome浏览器时可用。

# Call

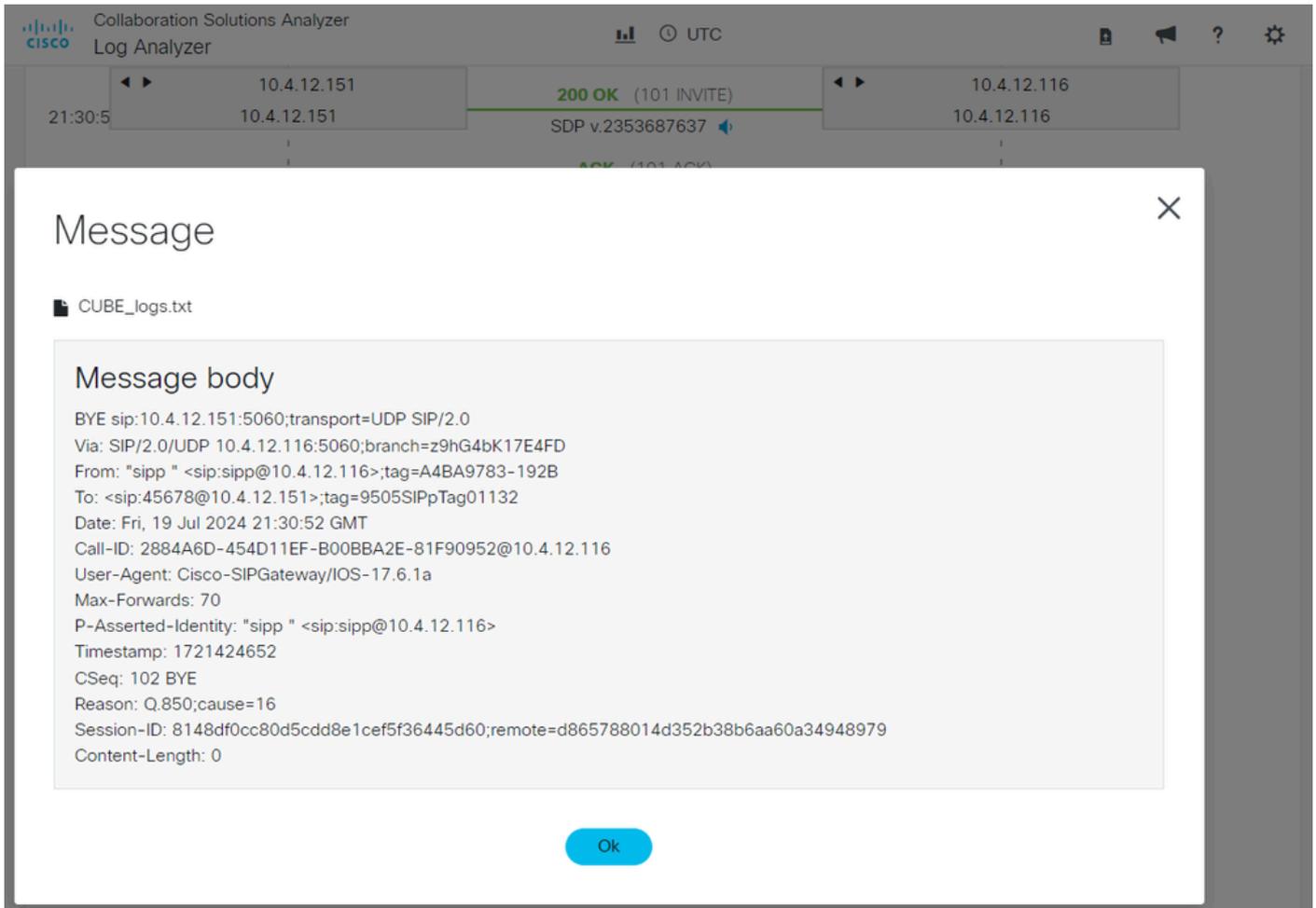
Call leg info   **Ladder diagram**   Signalling

Allow horizontal scroll   [Download ladder](#)



日志分析器梯形图

此工具允许管理员打开SIP消息并查看其内容。单击一条消息将其打开。



日志分析器梯形图消息

管理员可以在呼叫分支信息部分添加阶梯标记以使SIP消息具有明显的点标记。SIP消息中包含的任何参数都可用于标记。

在本例中，ID参数使用IP地址，并且添加了说明。包含IP地址的SIP消息以圆点标记突出显示，以便与其他消息区分。

## Ladder tags management

Applied tags

ID	Description	Visual	Action
10.4.12.151	Service Provider	●	🗑️

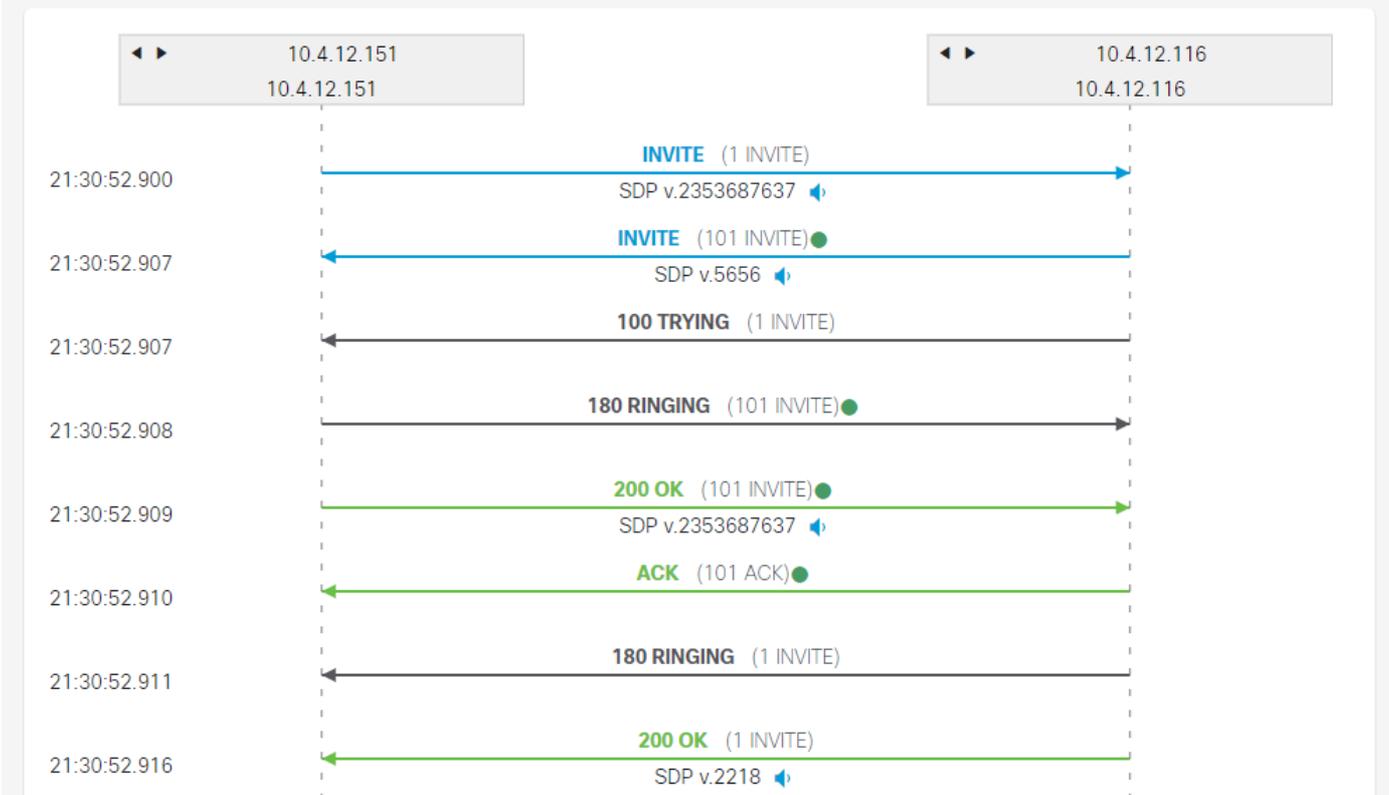
日志分析器梯形标签1

# Call

Call leg info **Ladder diagram** Signalling

Allow horizontal scroll

Legend: ■ Service Provider



日志分析器梯形标签2

可用于区分SIP消息与其他消息的另一个过滤器是语音编解码器。

## Ladder tags management



Applied tags

ID	Description	Visual	Action
PCMU	Voice Codec G711ulaw	●	🗑️

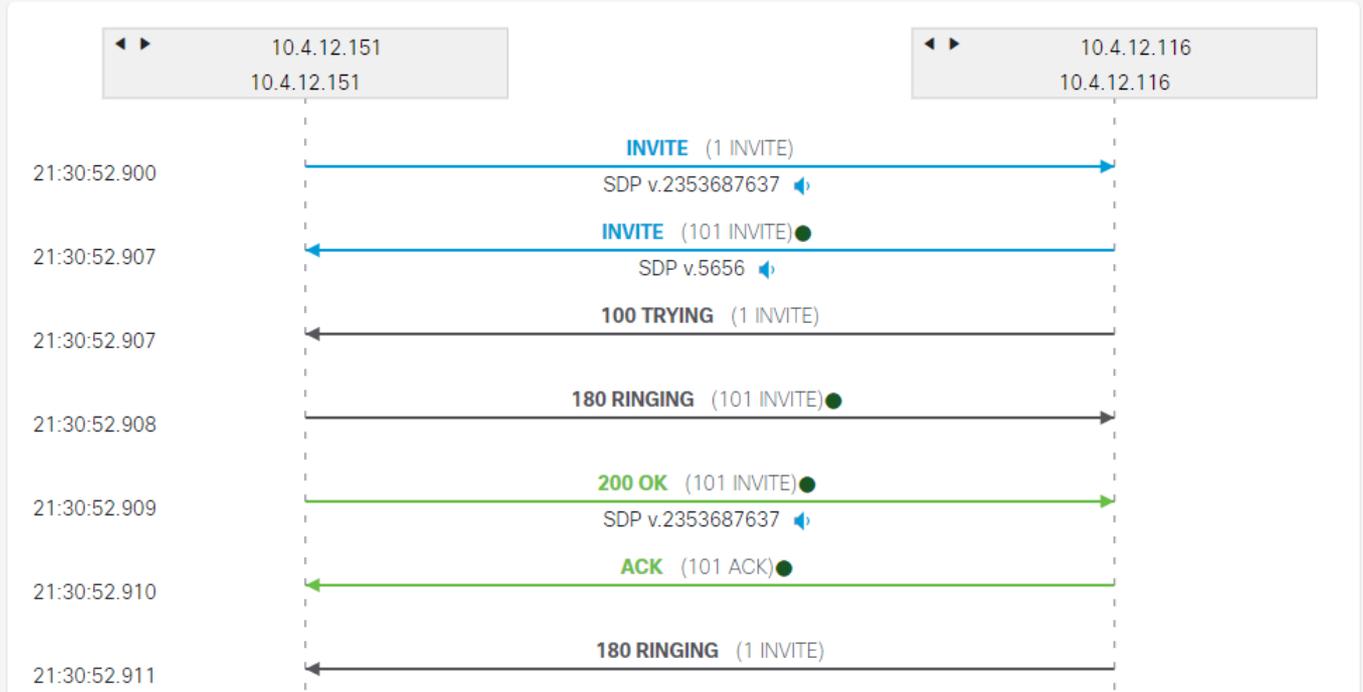
日志分析器梯形标签3

# Call

Call leg info **Ladder diagram** Signalling

Allow horizontal scroll

Legend: ■ Voice Codec G711ulaw



日志分析器梯形标签4

## 信令

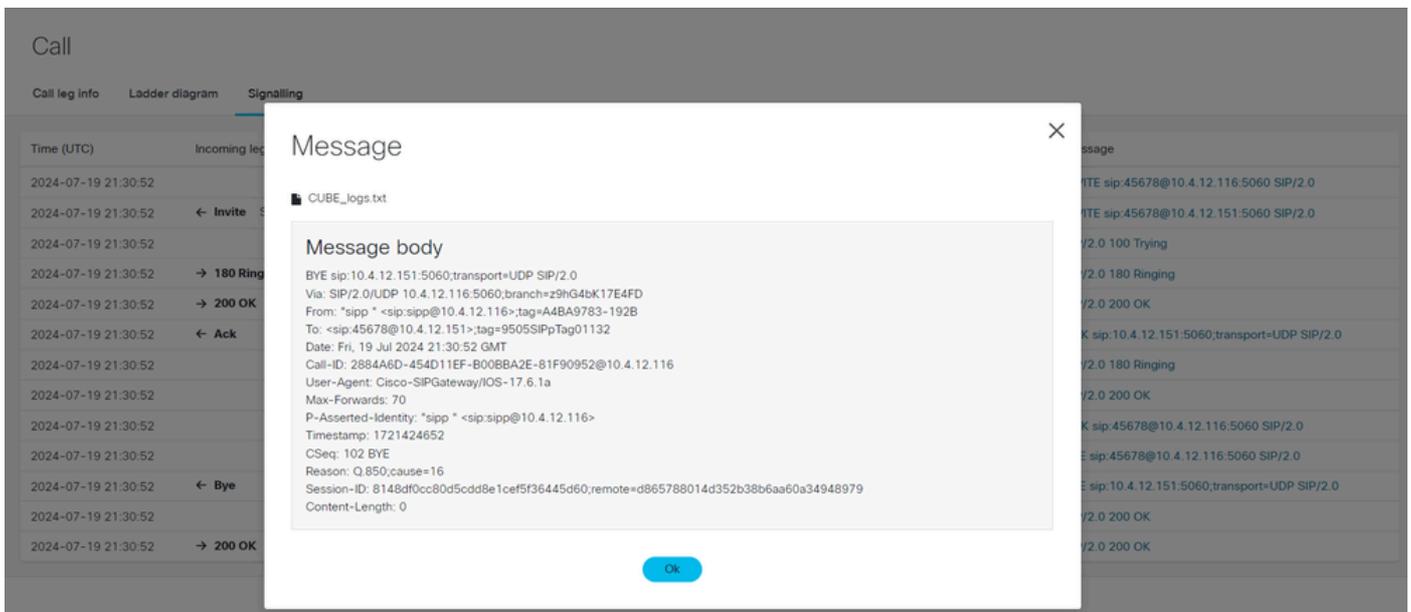
最后阶段是信令，它显示CUBE段（传入和传出）的SIP消息。它包含源IP地址和目的IP地址。单击以查看消息。

### Call

Call leg info **Ladder diagram** **Signalling**

Time (UTC)	Incoming legs	Outgoing legs	Sequence	Source	Destination	Message
2024-07-19 21:30:52		← Invite SDP v.2353687637	1 INVITE	10.4.12.151:5061	10.4.12.116:5060	INVITE sip:45678@10.4.12.116:5060 SIP/2.0
2024-07-19 21:30:52	← Invite SDP v.5656		101 INVITE	10.4.12.116:5060	10.4.12.151:5060	INVITE sip:45678@10.4.12.151:5060 SIP/2.0
2024-07-19 21:30:52		→ 100 Trying	1 INVITE	10.4.12.116:5060	10.4.12.151:5061	SIP/2.0 100 Trying
2024-07-19 21:30:52		→ 180 Ringing	101 INVITE	10.4.12.151:5060	10.4.12.116:5060	SIP/2.0 180 Ringing
2024-07-19 21:30:52		→ 200 OK SDP v.2353687637	101 INVITE	10.4.12.151:5060	10.4.12.116:5060	SIP/2.0 200 OK
2024-07-19 21:30:52	← Ack		101 ACK	10.4.12.116:5060	10.4.12.151:5060	ACK sip:10.4.12.151:5060;transport=UDP SIP/2.0
2024-07-19 21:30:52		→ 180 Ringing	1 INVITE	10.4.12.116:5060	10.4.12.151:5061	SIP/2.0 180 Ringing
2024-07-19 21:30:52		→ 200 OK SDP v.2218	1 INVITE	10.4.12.116:5060	10.4.12.151:5061	SIP/2.0 200 OK
2024-07-19 21:30:52		← Ack	1 ACK	10.4.12.151:5061	10.4.12.116:5060	ACK sip:45678@10.4.12.116:5060 SIP/2.0
2024-07-19 21:30:52		← Bye	2 BYE	10.4.12.151:5061	10.4.12.116:5060	BYE sip:45678@10.4.12.116:5060 SIP/2.0
2024-07-19 21:30:52	← Bye		102 BYE	10.4.12.116:5060	10.4.12.151:5060	BYE sip:10.4.12.151:5060;transport=UDP SIP/2.0
2024-07-19 21:30:52		→ 200 OK	2 BYE	10.4.12.116:5060	10.4.12.151:5061	SIP/2.0 200 OK
2024-07-19 21:30:52		→ 200 OK	102 BYE	10.4.12.151:5060	10.4.12.116:5060	SIP/2.0 200 OK

日志分析器信令



日志分析器信令消息

## 诊断

从日志分析的所有数据都将根据诊断签名运行，这些签名可识别已知缺陷、常见问题或错误配置，并提供纠正措施计划。

选择在日志中捕获的呼叫显示呼叫摘要分析后，CSA平台应显示诊断部分，其中包含以下信息：

- 发现的问题
- 缺少信息
- 潜在问题

可激活切换按钮以仅过滤和显示缺陷。

Collaboration Solutions Analyzer  
Log Analyzer

CUBE\_logs.txt UTC

Report Problem ?

## Log overview

Calls

Search

From DN / URI	To DN / URI	CallId	SIP Call-Id	Peer Call-Id	GUID	Call initiated (UTC)	Call end (UTC)	Log duration (sec)	Disconnect reason
sipp	45678	5524 47	1-9880@10.4.12.1 51	552448	02876 031B0 05	2024-07-19 21:3 0:52	2024-07-19 2 1:30:52	0 seconds	0
sipp	45678	5524 48	2884A6D-454D11E F-B00BBA2E-81F9 0952@10.4.12.116	552447	02876 031B0 05	2024-07-19 21:3 0:52	2024-07-19 2 1:30:52	0 seconds	16

1-2 of 2 Prev 1 Next Showing 10

日志分析器诊断程序主页

Collaboration Solutions Analyzer  
Log Analyzer

UTC

Report Problem ?

## Diagnostic overview

Issues found No issue Not applicable Missing information Potential problem

Search

**Result Category** ^

- Call (8)
- MRA (0)
- Configuration (0)

Defects only

✓ No issues were found.

You can still view the diagnostic signatures that were run but did not find any issue by selecting different result type tabs above.

Click on any of the below to see details or [continue to analysis](#).

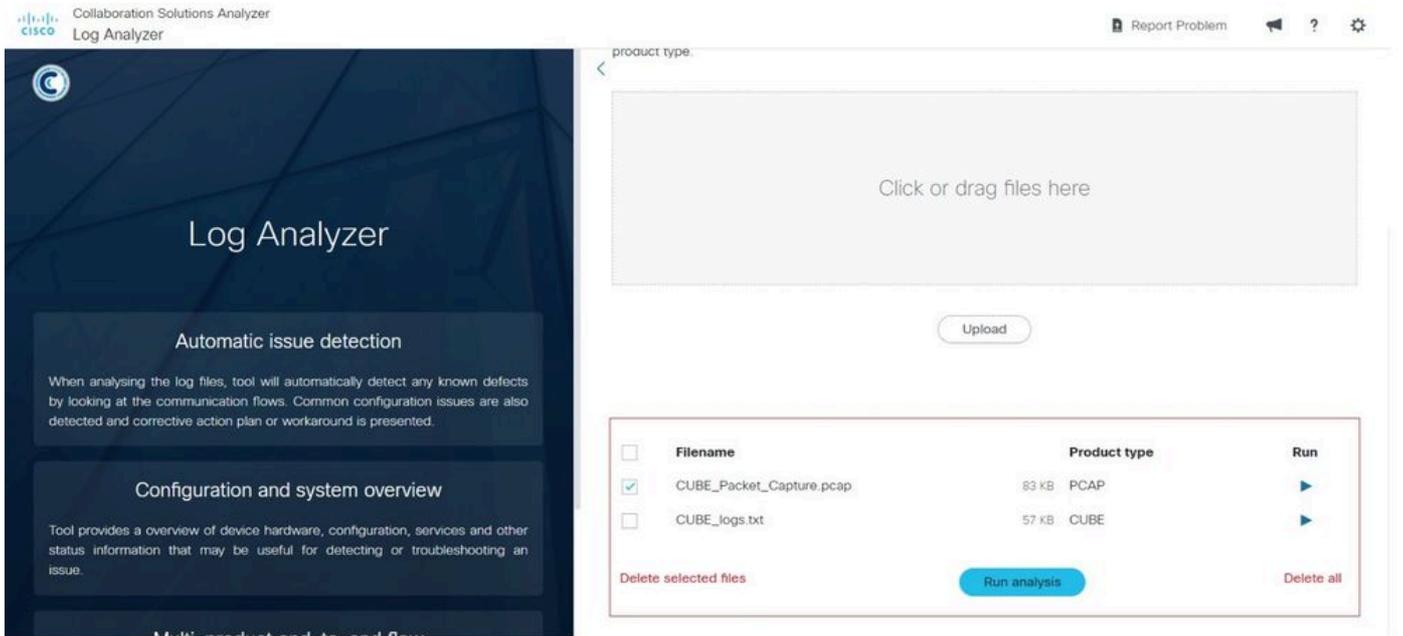
日志分析器诊断概述

## CUBE数据包捕获

数据包捕获是一种文件缓冲区，创建它的目的是在CUBE网络接口或任何语音网络设备上收集实际数据包的副本。此文件可以通过网络分析器软件(如Wireshark)打开和分析。

日志分析工具已通过数据包捕获分析工具进行增强，该工具可以处理pcap或pcpng文件格式扩展，提供从呼叫中收集的会话和网络统计信息的摘要。

数据包捕获文件必须以CUBE日志文件相同的方式上传到日志分析器工具。系统将产品类型确定为PCAP。



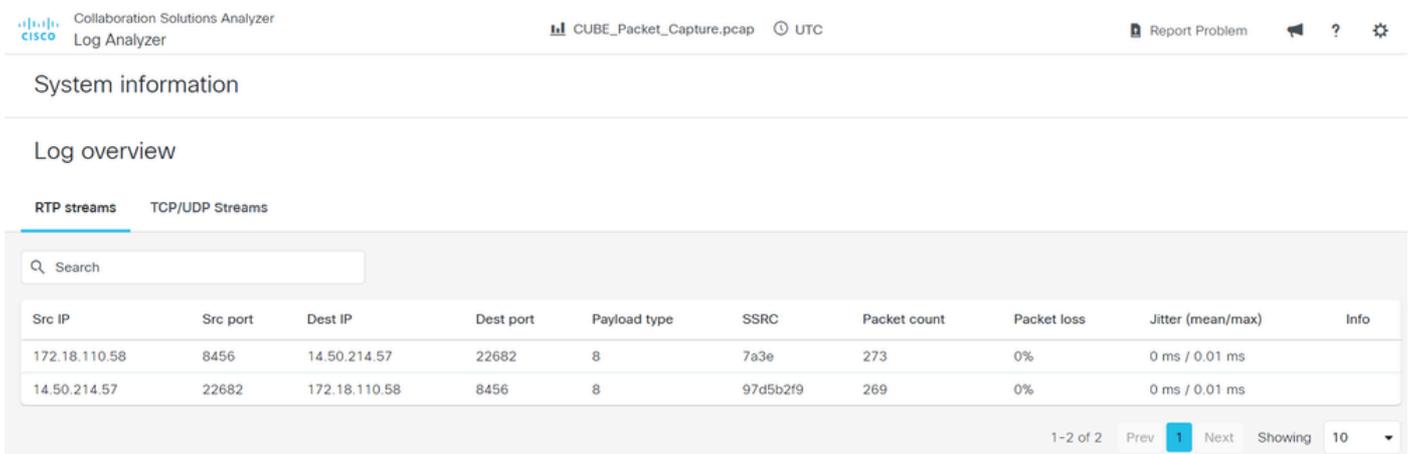
日志分析器数据包捕获文件

激活运行分析按钮后，日志分析器工具将分析信息并在两列中提供所捕获会话的摘要：

- RTP流
- TCP/UDP数据流

 注意：如果数据包捕获包含SRTP流，则它将显示在“RTP流”列中，并执行网络分析。SRTP流的音频部分未解码。

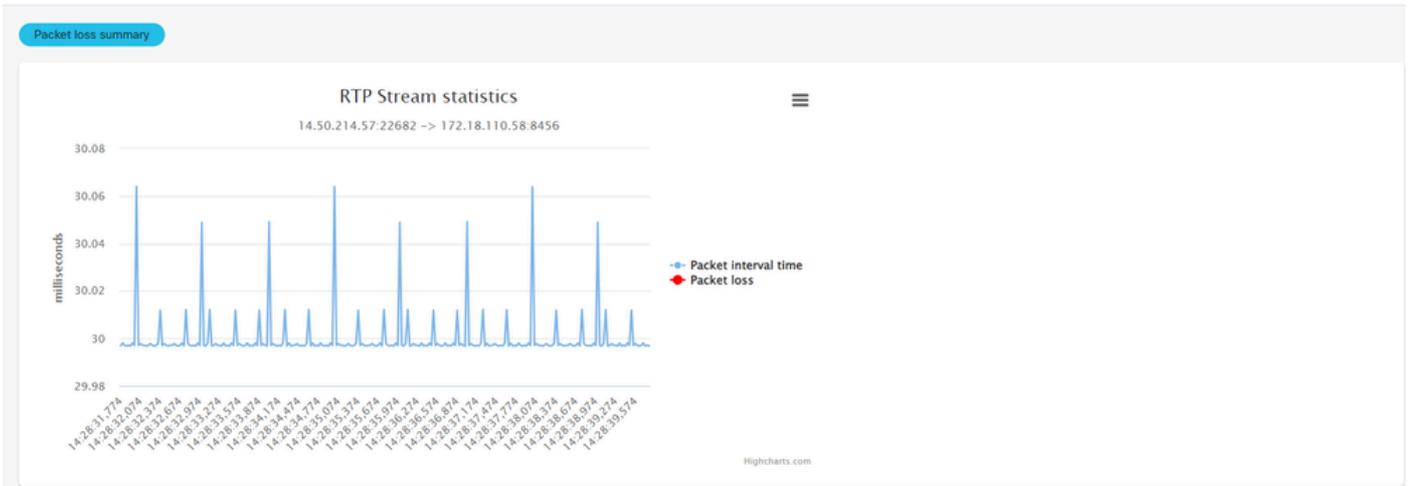
从RTP流列中选择一个会话，工具将显示该连接的RTP流统计信息。如果数据流受到网络条件的影响，则丢包参数应标有红点。



日志分析器PCAP分析

RTP流统计信息可以以包含数据包丢失摘要的文本文件格式下载。单击Packet Loss Summary按钮以下载文件。

## RTP Stream



日志分析器PCAP RTP流

对于TCP/UDP流，系统显示捕获的会话的摘要。

## System information

### Log overview

RTP streams TCP/UDP Streams

Search

Protocol	Src IP	Src port	Dest IP	Dest port	Packet count	2-way communication	OCSIP
UDP	172.18.110.58	49782	172.18.110.48	5060	4	✖	
UDP	172.18.110.48	5060	172.18.110.58	5060	4	✖	
UDP	172.18.110.59	32771	172.18.110.1	5060	2	✖	

1-3 of 3 Prev 1 Next Showing 10

日志分析器PCAP TCP UDP数据流

## SIP配置文件测试器(SPT)

会话发起协议(SIP)配置文件用于修改传入或传出SIP消息，以确保不同设备之间的兼容性。“SIP配置文件测试器”工具允许您在实时环境中部署配置之前验证配置。

SIP配置文件工具包含3个部分：

- SIP配置文件规则 -用于插入要测试的SIP配置文件规则的窗口。
- SIP Message to Apply Rules -用于将SIP消息粘贴到要应用规则的位置的窗口。
- SIP message to copy from - ( 可选 ) 用于粘贴SIP消息以备测试复制列表配置时使用。复制列表配置将设备接收的入站报头内容复制到出站报头。

该工具包含2个用于管理测试的按钮：

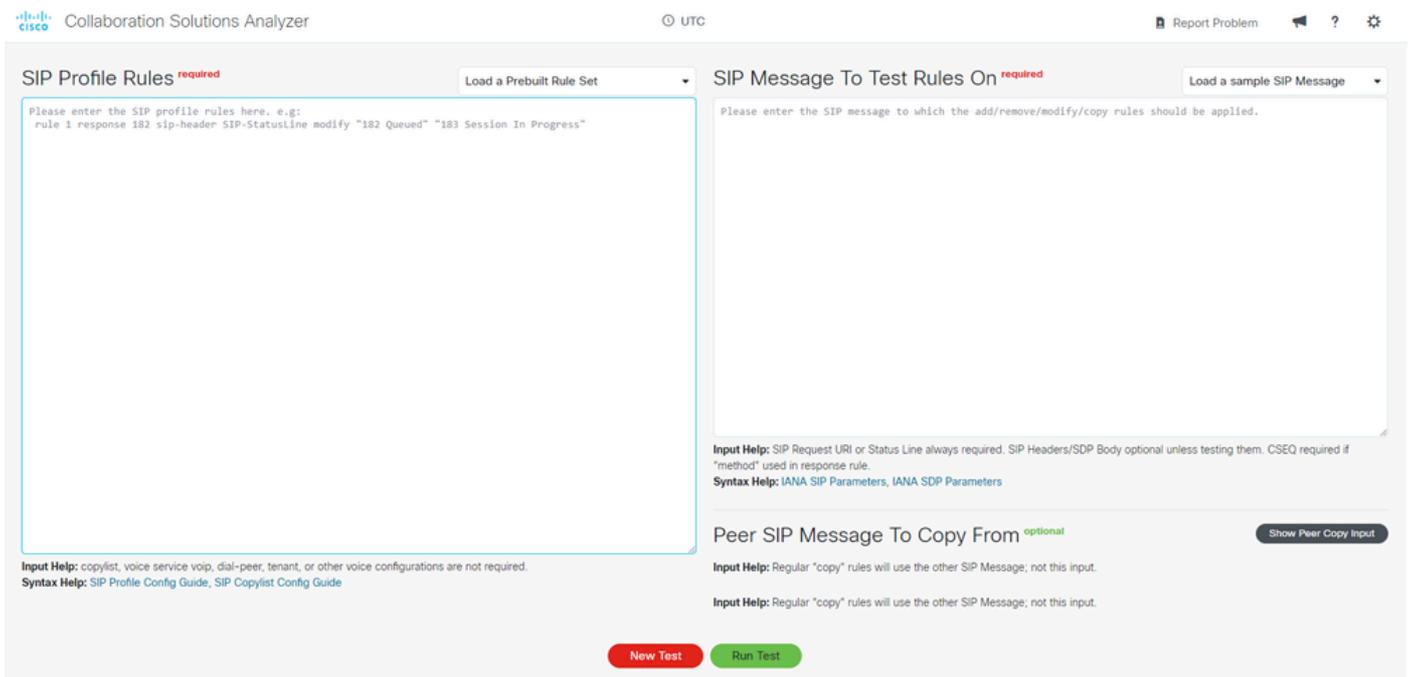
- 绿色按钮- 运行测试。
- 红色按钮- 用于重置和清除设置。

选择绿色按钮运行测试后，该工具将显示以下选项：

- 红色按钮- 新测试
- 蓝色按钮- 显示输入

突出显示原始/修改的SIP消息结果：

- 蓝色- 修改的SIP报头或SDP正文在两个消息区域中均以蓝色突出显示。
- 绿色- 添加的SIP报头或SDP正文仅在修改的SIP消息结果中突出显示为绿色。
- 红色- 删除的SIP报头或SDP正文仅在原始SIP消息结果中突出显示为红色。



SIP配置文件主页

## 预构建SIP配置文件示例

该工具提供预建示例来简化测试。在每个窗口的顶部，都有一个用于选择这些示例的应用程序框。

以下是如何使用预定义的配置：

1. 点击加载预构建规则集并选择添加：SIP报头。
2. 点击加载示例SIP消息并选择邀请（无SDP）。
3. 选择绿色的Run Test 按钮以执行测试。

### SIP Profile Rules required

rule 100 request ANY sip-header Diversion Add "Diversion: < sip:8675309@cisco.com>"

**Input Help:** copylist, voice service voip, dial-peer, tenant, or other voice configurations are not required.  
**Syntax Help:** SIP Profile Config Guide, SIP Copylist Config Guide

### SIP Message To Test Rules On required

```
INVITE sip:8675309@192.168.11.10:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.10.10:5060;branch=z9hG4kK16242110
Via: SIP/2.0/UDP 192.168.10.9:5060;branch=z9hG4kK00002579
From: "CallerID_Name" < sip:123456789@192.168.10.10>;tag=4EDF0008-CA0
To: < sip:8675309@192.168.11.10>
Call-ID: 07E43511-335111EF-85780A40-687E8AC0@192.168.10.10
Session-ID: 2d390a8000105000a000247e1266c26d;remote=3b954a1e00105000a0006c416a369498
Cisco-Guid: 3622027175-0860951023-223888512-1803467483
Cseq: 101 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
Allow-Events: telephone-event, kpml, dialog
Supported: 100rel, timer, resource-priority, replaces, sdp-anat
Require: timer
Subject: SIP Profile Test
Session: Media
User-Agent: Cisco-SIPGateway/IOS-17.14.1a
Date: Thu, 27 Jun 2024 00:20:07 GMT
Timestamp: 1719447607
Expires: 180
Min-SE: 1800
Session-Expires: 1800;refresher=uac
Max-Forwards: 69
Contact: < sip:111111111@192.168.10.10:5060;transport=tcp>
Diversion: < sip:222222222@192.168.10.10>;privacy=off;reason=unconditional;counter=1;screen=no
Remote-Party-ID: "CallerID_Name" < sip:333333333@192.168.10.10>;party=calling;screen=no;privacy=off
P-Asserted-Identity: "CallerID_Name" < sip:444444444@192.168.10.10>
P-Preferred-Identity: "CallerID_Name" < sip:555555555@192.168.10.10>
CustomHeader: "CallerID_Name" < sip:777777777@192.168.10.10>
Accept: application/sdp
Content-Disposition: session;handling=required
Content-Length: 0
```

**Input Help:** Regular "copy" rules will use the other SIP Message; not this input.

**Peer SIP Message To Copy From optional**

**Input Help:** Regular "copy" rules will use the other SIP Message; not this input.

Show Peer Copy Input

New Test
Run Test

预置SIP配置文件

该工具将显示一个新屏幕，显示测试结果：

修改的SIP消息

ADDED (GREEN) - Diversion: < sip:8675309@cisco.com

Cisco Collaboration Solutions Analyzer

Report Problem ? ? ?

New Test
Show Inputs

#### Original SIP Message:

```
1 INVITE sip:8675309@192.168.11.10:5060 SIP/2.0
2 Via: SIP/2.0/UDP 192.168.10.10:5060;branch=z9hG4kK16242110,SIP/2.0/UDP 192.168.10.9:5060;branch=z9hG4kK00002579
3 From: "CallerID_Name" < sip:123456789@192.168.10.10>;tag=4EDF0008-CA0
4 To: < sip:8675309@192.168.11.10>
5 Call-ID: 07E43511-335111EF-85780A40-687E8AC0@192.168.10.10
6 Session-ID: 2d390a8000105000a000247e1266c26d;remote=3b954a1e00105000a0006c416a369498
7 Cisco-Guid: 3622027175-0860951023-223888512-1803467483
8 Cseq: 101 INVITE
9 Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
10 Allow-Events: telephone-event, kpml, dialog
11 Supported: 100rel, timer, resource-priority, replaces, sdp-anat
12 Require: timer
13 Subject: SIP Profile Test
14 Session: Media
15 User-Agent: Cisco-SIPGateway/IOS-17.14.1a
16 Date: Thu, 27 Jun 2024 00:20:07 GMT
17 Timestamp: 1719447607
18 Expires: 180
19 Min-SE: 1800
20 Session-Expires: 1800;refresher=uac
21 Max-Forwards: 69
22 Contact: < sip:111111111@192.168.10.10:5060;transport=tcp>
23 Diversion: < sip:222222222@192.168.10.10>;privacy=off;reason=unconditional;counter=1;screen=no
24 Remote-Party-ID: "CallerID_Name" < sip:333333333@192.168.10.10>;party=calling;screen=no;privacy=off
25 P-Asserted-Identity: "CallerID_Name" < sip:444444444@192.168.10.10>
26 P-Preferred-Identity: "CallerID_Name" < sip:555555555@192.168.10.10>
27 CustomHeader: "CallerID_Name" < sip:777777777@192.168.10.10>
28 Accept: application/sdp
29 Content-Disposition: session;handling=required
30 Content-Length: 0
```

#### Modified SIP Message:

```
1 INVITE sip:8675309@192.168.11.10:5060 SIP/2.0
2 Via: SIP/2.0/UDP 192.168.10.10:5060;branch=z9hG4kK16242110,SIP/2.0/UDP 192.168.10.9:5060;branch=z9hG4kK00002579
3 From: "CallerID_Name" < sip:123456789@192.168.10.10>;tag=4EDF0008-CA0
4 To: < sip:8675309@192.168.11.10>
5 Call-ID: 07E43511-335111EF-85780A40-687E8AC0@192.168.10.10
6 Session-ID: 2d390a8000105000a000247e1266c26d;remote=3b954a1e00105000a0006c416a369498
7 Cisco-Guid: 3622027175-0860951023-223888512-1803467483
8 Cseq: 101 INVITE
9 Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
10 Allow-Events: telephone-event, kpml, dialog
11 Supported: 100rel, timer, resource-priority, replaces, sdp-anat
12 Require: timer
13 Subject: SIP Profile Test
14 Session: Media
15 User-Agent: Cisco-SIPGateway/IOS-17.14.1a
16 Date: Thu, 27 Jun 2024 00:20:07 GMT
17 Timestamp: 1719447607
18 Expires: 180
19 Min-SE: 1800
20 Session-Expires: 1800;refresher=uac
21 Max-Forwards: 69
22 Contact: < sip:111111111@192.168.10.10:5060;transport=tcp>
23 Diversion: < sip:222222222@192.168.10.10>;privacy=off;reason=unconditional;counter=1;screen=no
24 Remote-Party-ID: "CallerID_Name" < sip:333333333@192.168.10.10>;party=calling;screen=no;privacy=off
25 P-Asserted-Identity: "CallerID_Name" < sip:444444444@192.168.10.10>
26 P-Preferred-Identity: "CallerID_Name" < sip:555555555@192.168.10.10>
27 CustomHeader: "CallerID_Name" < sip:777777777@192.168.10.10>
28 Accept: application/sdp
29 Content-Disposition: session;handling=required
30 [Diversion: < sip:8675309@cisco.com>]
31 Content-Length: 0
```

#### Logs:

Action	Before	After	Rule
ADD		Diversion: < sip:8675309@cisco.com>	rule 100 request ANY sip-header Diversion Add "Diversion: < sip:8675309@cisco.com>"

SIP配置文件预构建添加示例

下面是修改/添加/删除突出显示的示例：

## SIP配置文件规则

```
rule 100 request ANY sip-header Diversion Add "Diversion: <sip:8675309@cisco.com>"
rule 200 request ANY sip-header P-Asserted-Identity modify "sip:4444444444@" "sip:5555555555@"
rule 300 request ANY sip-header P-Preferred-Identity remove
```

## 测试规则的Sip消息

```
INVITE sip:8675309@192.168.11.10:5060 SIP/2.0
Via: SIP/2.0/TCP 192.168.10.10:5060;branch=z9hG4bK16242110
Via: SIP/2.0/UDP 192.168.10.9:5060;branch=z9hG4bK00002579
From: "CallerID_Name" <sip:123456789@192.168.10.10>;tag=4EDF0DD8-CA0
To: <sip:8675309@192.168.11.10>
Call-ID: D7E43511-335111EF-8578BA40-6B7EBADB@192.168.10.10
Session-ID: 2d390a8000105000a000247e1266c26d;remote=3b954a1e00105000a0006c416a369498
Cisco-Guid: 3622027175-0860951023-2238888512-1803467483
Cseq: 101 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
Allow-Events: telephone-event,kpml,dialog
Supported: 100rel,timer,resource-priority,replaces
Supported: sdp-anat
Require: timer
Subject: SIP Profile Test
Session: Media
User-Agent: Cisco-SIPGateway/IOS-17.14.1a
Date: Thu, 27 Jun 2024 00:20:07 GMT
Timestamp: 1719447607
Expires: 180
Min-SE: 1800
Session-Expires: 1800;refresher=uac
Max-Forwards: 69
Contact: <sip:1111111111@192.168.10.10:5060;transport=tcp>
Diversion: <sip:2222222222@192.168.10.10>;privacy=off;reason=unconditional;counter=1;screen=no
Remote-Party-ID: "CallerID_Name" <sip:3333333333@192.168.10.10>;party=calling;screen=no;privacy=off
P-Asserted-Identity: "CallerID_Name" <sip:4444444444@192.168.10.10>
P-Preferred-Identity: "CallerID_Name" <sip:5555555555@192.168.10.10>
CustomHeader: "CallerID_Name" <sip:7777777777@192.168.10.10>
Accept: application/sdp
Content-Disposition: session;handling=required
Content-Length: 0
```

### SIP Profile Rules required

Load a Prebuilt Rule Set

```
rule 100 request ANY sip-header Diversion Add "Diversion: <sip:8675309@cisco.com>"
rule 200 request ANY sip-header P-Asserted-Identity modify "sip:444444444@ " "sip:555555555@"
rule 300 request ANY sip-header P-Preferred-Identity remove
```

**Input Help:** copylist, voice service voip, dial-peer, tenant, or other voice configurations are not required.  
**Syntax Help:** SIP Profile Config Guide, SIP Copylist Config Guide

### SIP Message To Test Rules On required

Load a sample SIP Message

```
INVITE sip:8675309@192.168.11.10:5060 SIP/2.0
Via: SIP/2.0/TCP 192.168.10.10:5060;branch=z9hG4bK16242110
Via: SIP/2.0/UDP 192.168.10.9:5060;branch=z9hG4bK00002579
From: "CallerID_Name" <sip:123456789@192.168.10.10>;tag=4EDF80D8-CA0
To: <sip:8675309@192.168.11.10>
Call-ID: 07E43511-335111EF-85788A40-687E8AD0@192.168.10.10
Session-ID: 2d390a800018500a000247e126c26d;remote=3b954a1e00105000a000c416a369498
Cisco-UUID: 3622027175-0860951023-2238888512-1803467483
Cseq: 101 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
Allow-Events: telephone-event, kmel, dialtone
Supported: 100rel,timer,resource-priority,replaces
Supported: sdp-anat
Require: timer
Subject: SIP Profile Test
Session: Media
User-Agent: Cisco-SIPGateway/IOS-17.14.1a
Date: Thu, 27 Jun 2024 00:20:07 GMT
Timestamp: 1719447607
Expires: 180
Min-SE: 1800
Session-Expires: 1800;refresher=uac
Max-Forwards: 69
Contact: <sip:111111111@192.168.10.10:5060;transport=tcp>
Diversion: <sip:222222222@192.168.10.10>;privacy=off;reason-unconditional;counter=1;screen=no
Input Help: SIP Request URI or Status Line always required. SIP Headers/SDP Body optional unless testing them. CSEQ required if "method" used in response rule.
Syntax Help: IANA SIP Parameters, IANA SDP Parameters
```

**Peer SIP Message To Copy From optional** Show Peer Copy Input

**Input Help:** Regular "copy" rules will use the other SIP Message; not this input.

New Test
Run Test

SIP配置文件修改添加删除示例

要查看结果，请单击Run Test。

原始SIP消息

MODIFIED (BLUE) - P-Asserted-Identity: "CallerID\_Name"

4444444444@192.168.10.10>

REMOVED (RED) - P-Preferred-Identity: "CallerID\_Name" <sip:555555555@192.168.10.10>

修改的SIP消息

MODIFIED (BLUE) - P-Asserted-Identity: "CallerID\_Name" <sip:555555555@192.168.10.10>  
 ADDED (GREEN) - Diversion: <sip:8675309@cisco.com>

### Original SIP Message:

```

1 INVITE sip:8675309@192.168.11.10:5060 SIP/2.0
2 Via: SIP/2.0/TCP 192.168.10.10:5060;branch=z9hG4k16242110,SIP/2.0/UDP 192.168.10.9:5060;branch=z9hG4k00002579
3 From: "CallerID_Name" <sip:123456789@192.168.10.10>;tag=4E0F0C08-CA0
4 To: <sip:8675309@192.168.11.10>
5 Call-ID: 07643511-335111F1-85788440-687E8AD0@192.168.10.10
6 Session-ID: 1d790a000105000a000147e126c16d;remote=30954e1e00105000a000c416a369498
7 Cisco-Guid: 3622027175-0860951023-223888512-1803467483
8 Cseq: 101 INVITE
9 Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
10 Allow-Events: telephone-event,kgml,dialog
11 Supported: 100rel,timer,resource-priority,replaces,sdp-angat
12 Require: timer
13 Subject: SIP Profile Test
14 Session: Media
15 User-Agent: Cisco-SIPGateway/IOS-17.14.1a
16 Date: Thu, 27 Jun 2024 00:20:07 GMT
17 Timestamp: 1719447607
18 Expires: 180
19 Min-SE: 1800
20 Session-Expires: 1800;refresher=uac
21 Max-Forwards: 69
22 Contact: <sip:111111111@192.168.10.10:5060;transport=tcp>
23 Diversion: <sip:222222222@192.168.10.10>;privacy=off;reason=unconditional;counter=1;screen=no
24 Remote-Party-ID: "CallerID_Name" <sip:333333333@192.168.10.10>;party=calling;screen=no;privacy=off
25 P-Asserted-Identity: "CallerID_Name" <sip:444444444@192.168.10.10>
26 P-Preferred-Identity: "CallerID_Name" <sip:555555555@192.168.10.10>
27 CustomHeader: "CallerID_Name" <sip:777777777@192.168.10.10>
28 Accept: application/sdp
29 Content-Disposition: session;handling=required
30 Content-Length: 0
                
```

### Modified SIP Message:

```

1 INVITE sip:8675309@192.168.11.10:5060 SIP/2.0
2 Via: SIP/2.0/TCP 192.168.10.10:5060;branch=z9hG4k16242110,SIP/2.0/UDP 192.168.10.9:5060;branch=z9hG4k00002579
3 From: "CallerID_Name" <sip:123456789@192.168.10.10>;tag=4E0F0C08-CA0
4 To: <sip:8675309@192.168.11.10>
5 Call-ID: 07643511-335111F1-85788440-687E8AD0@192.168.10.10
6 Session-ID: 1d790a000105000a000147e126c16d;remote=30954e1e00105000a000c416a369498
7 Cisco-Guid: 3622027175-0860951023-223888512-1803467483
8 Cseq: 101 INVITE
9 Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
10 Allow-Events: telephone-event,kgml,dialog
11 Supported: 100rel,timer,resource-priority,replaces,sdp-angat
12 Require: timer
13 Subject: SIP Profile Test
14 Session: Media
15 User-Agent: Cisco-SIPGateway/IOS-17.14.1a
16 Date: Thu, 27 Jun 2024 00:20:07 GMT
17 Timestamp: 1719447607
18 Expires: 180
19 Min-SE: 1800
20 Session-Expires: 1800;refresher=uac
21 Max-Forwards: 69
22 Contact: <sip:111111111@192.168.10.10:5060;transport=tcp>
23 Diversion: <sip:222222222@192.168.10.10>;privacy=off;reason=unconditional;counter=1;screen=no
24 Remote-Party-ID: "CallerID_Name" <sip:333333333@192.168.10.10>;party=calling;screen=no;privacy=off
25 P-Asserted-Identity: "CallerID_Name" <sip:555555555@192.168.10.10>
26 CustomHeader: "CallerID_Name" <sip:777777777@192.168.10.10>
27 Accept: application/sdp
28 Content-Disposition: session;handling=required
29 Diversion: <sip:8675309@cisco.com>
30 Content-Length: 0
                
```

### Logs:

Action	Before	After	Rule
ADD		Diversion: <sip:8675309@cisco.com>	rule 100 request ANY sip-header Diversion Add "Diversion: <sip:8675309@cisco.com>"
MODIFY	P-Asserted-Identity: "CallerID_Name" <sip:444444444@192.168.10.10>	P-Asserted-Identity: "CallerID_Name" <sip:555555555@192.168.10.10>	rule 200 request ANY sip-header P-Asserted-Identity modify "sip:444444444@" "sip:555555555@"
REMOVE	P-Preferred-Identity: "CallerID_Name" <sip:555555555@192.168.10.10>		rule 300 request ANY sip-header P-Preferred-Identity remove

## SIP配置文件修改添加删除示例2

## Copylist SIP配置文件

要将设备收到的传入报头中的内容复制到传出报头（SIP复制列表），可以使用以下工具输入：

- 流程图：传入SIP消息 —> CUBE —> 修改的SIP消息
- Peer SIP Message To Copy From -要从中复制的SIP消息。
- Sip Message To Test Rules On -用于应用规则的SIP消息。

要启用要从中复制的对等SIP消息部分，必须启用Show Peer Copy Input选项。您可以单击Hide Peer Copy Input隐藏此部分。

Cisco Collaboration Solutions Analyzer

UTC

Report Problem ?

### SIP Profile Rules required

Load a Prebuilt Rule Set

Please enter the SIP profile rules here. e.g:  
rule 1 response 182 sip-header SIP-Statusline modify "182 Queued" "183 Session In Progress"

**Input Help:** copylist, voice service voip, dial-peer, tenant, or other voice configurations are not required.  
**Syntax Help:** SIP Profile Config Guide, SIP Copylist Config Guide

### SIP Message To Test Rules On required

Load a sample SIP Message

Please enter the SIP message to which the add/remove/modify/copy rules should be applied.

**Input Help:** SIP Request URI or Status Line always required. SIP Headers/SDP Body optional unless testing them. CSEQ required if "method" used in response rule.  
**Syntax Help:** IANA SIP Parameters, IANA SDP Parameters

### Peer SIP Message To Copy From optional

Hide Peer Copy Input

Please enter the peer SIP message here to copy values from when using "peer-header" type rules.



To: <sip:5678@10.15.0.2>  
 Call-ID: a0f63500-1f013804-1344e15-16000e0a@10.14.0.1  
 Supported: 100rel,timer,resource-priority,replaces  
 Min-SE: 1800  
 User-Agent: Cisco-CUCM12.5  
 Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY  
 CSeq: 101 INVITE  
 Expires: 180  
 Allow-Events: presence, kpm1  
 Supported: X-cisco-srtp-fallback,X-cisco-original-called  
 Call-Info: <sip:10.14.0.1:5060>;method="NOTIFY;Event=telephone-event;Duration=500"  
 Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=DESKTOP  
 Session-ID: 1629a67700105000885a92d9a7fe;remote=00000000000000000000000000000000  
 Cisco-Guid: 2700489984-0000065536-0000126777-1234102346  
 Session-Expires: 1800  
 P-Asserted-Identity: "Cisco" <sip:1234@10.14.0.1>  
 Remote-Party-ID: "Cisco" <sip:1234@10.14.0.1>;party=calling;screen=yes;privacy=off  
 Contact: <sip:1234@10.14.0.1:5060>;+u.sip!devicename.ccm.cisco.com="SEP885A92D9A7FE"  
 Max-Forwards: 69  
 Content-Length: 0

The screenshot shows the Cisco Collaboration Solutions Analyzer interface. It is divided into two main sections: 'SIP Profile Rules' and 'SIP Message To Test On'.  
 - The 'SIP Profile Rules' section on the left contains two rules: 'request INVITE peer-header sip to copy \*sip:(.\*)@\* u01' and 'request INVITE sip-header SIP-Req-URI modify \*sip:(.\*)@\* sip:\u01@'. Below the rules are 'Input Help' and 'Syntax Help' links.  
 - The 'SIP Message To Test On' section on the right displays a detailed SIP INVITE message header and body. The header includes fields like 'Via: SIP/2.0/UDP 192.0.2.0:5060', 'From: "Cisco" <sip:1234@10.14.0.1>', 'To: <sip:5678@10.15.0.2>', 'Call-ID: 7835570f-193811ef-acc18962-0503ec18@192.0.2.0', 'Supported: 100rel,timer,resource-priority,replaces,sdp-anat', 'Min-SE: 1800', 'Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER', 'CSeq: 101 INVITE', 'Timestamp: 1716577979', 'Contact: <sip:1234@192.0.2.0:5060>', 'Expires: 180', 'Allow-Events: telephone-event', 'Max-Forwards: 69', 'P-Asserted-Identity: "Cisco" <sip:9876@192.0.2.0>', 'Session-ID: 1629a67700105000a00009a7fe;remote=00000000000000000000000000000000', 'Session-Expires: 1800', 'Content-Type: application/sdp', 'Content-Disposition: session;handling=required', and 'Content-Length: 243'. The body contains 'v=0', 'o=CiscoSystemsSIP-Gatekeeper 3681 9882 IN IP4 192.0.2.0', 's=SIP Call', and 'c=urn:ietf:params:ipp:session-description'. Below the message is 'Input Help' and 'Syntax Help' text.  
 - At the bottom of the interface, there are two buttons: 'New Test' (red) and 'Run Test' (green).

SIP配置文件复制列表示例

单击Run Test 按钮继续以启动工具。

复制寄存器

Register: u01  
 Value: 5678

原始SIP消息

MODIFIED (BLUE) - INVITE sip:235678@10.16.0.5:5060 SIP/2.0

### 修改的SIP消息

MODIFIED (BLUE) - INVITE sip:5678@10.16.0.5:5060 SIP/2.0

Collaboration Solutions Analyzer

Report Problem
?
⚙

**Original SIP Message:**

```

1 INVITE sip:235678@10.16.0.5:5060 SIP/2.0
2 Via: SIP/2.0/UDP 192.0.2.0:5060;branch=19NG4KA7155C
3 From: "Cisco" <sip:1234@10.16.0.3>;tag=8125CE72-1184
4 To: <sip:5678@10.16.0.5>
5 Call-ID: 7835570F-193811EF-AAC18962-0503EC18@192.0.2.0
6 Supported: 100rel,timer,resource-priority,replaces,sdp-annat
7 Min-SE: 1800
8 Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
9 CSeq: 101 INVITE
10 Timestamp: 1716577979
11 Contact: <sip:1234@192.0.2.0:5060>
12 Expires: 180
13 Allow-Events: telephone-event
14 Max-Forwards: 68
15 P-Asserted-Identity: "Cisco" <sip:9876@192.0.2.0>
16 Session-ID: 1629e67700105000a000a7fe;remote=00000000000000000000000000000000
17 Session-Expires: 1800
18 Content-Type: application/sdp
19 Content-Disposition: session;handling=required
20 Content-Length: 243
21
22 v=0
23 o=CiscoSystemsSIP-GU-UserAgent 3601 9082 IN IP4 192.0.2.0
24 s=SIP Call
25 c=IN IP4 192.0.2.0
26 t=0
27 m=audio 8482 RTP/AVP 0 101
28 c=IN IP4 192.0.2.0
29 a=rtpmap:0 PCMU/8000
30 a=rtpmap:101 telephone-event/8000
31 a=fmtp:101 0-16

```

**Modified SIP Message:**

```

1 INVITE sip:5678@10.16.0.5:5060 SIP/2.0
2 Via: SIP/2.0/UDP 192.0.2.0:5060;branch=29HG4KA7155C
3 From: "Cisco" <sip:1234@10.16.0.3>;tag=8125CE72-1184
4 To: <sip:5678@10.16.0.5>
5 Call-ID: 7835570F-193811EF-AAC18962-0503EC18@192.0.2.0
6 Supported: 100rel,timer,resource-priority,replaces,sdp-annat
7 Min-SE: 1800
8 Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
9 CSeq: 101 INVITE
10 Timestamp: 1716577979
11 Contact: <sip:1234@192.0.2.0:5060>
12 Expires: 180
13 Allow-Events: telephone-event
14 Max-Forwards: 68
15 P-Asserted-Identity: "Cisco" <sip:9876@192.0.2.0>
16 Session-ID: 1629e67700105000a000a7fe;remote=00000000000000000000000000000000
17 Session-Expires: 1800
18 Content-Type: application/sdp
19 Content-Disposition: session;handling=required
20 Content-Length: 225
21
22 v=0
23 o=CiscoSystemsSIP-GU-UserAgent 3601 9082 IN IP4 192.0.2.0
24 s=SIP Call
25 c=IN IP4 192.0.2.0
26 t=0
27 m=audio 8482 RTP/AVP 0 101
28 c=IN IP4 192.0.2.0
29 a=rtpmap:0 PCMU/8000
30 a=rtpmap:101 telephone-event/8000
31 a=fmtp:101 0-16

```

**Copy Registers:**

Register	Value
u01	5678

**Logs:**

Action	Before	After	Rule
COPY	To: <sip:5678@10.16.0.5>	5678	request INVITE peer-header sip To copy "sip:(*)@" u01
MODIFY	INVITE sip:235678@10.16.0.5:5060 SIP/2.0	INVITE sip:5678@10.16.0.5:5060 SIP/2.0	request INVITE sip-header SIP-Req-URI modify "sip:(*)@" sip:u01@

SIP配置文件复制列表示例2

### 报告问题

在CSA平台顶部，Report A Problem部分允许您共享工具中检测到的任何问题。

此外，管理员还可以通过向CSA开发团队处理信息的位置发送邮件来提供反馈、意见或建议。

 We are introducing the new Collaboration Solutions Analyzer with a brand new user interface and wonderful new features. We are excited to hear your feedback, comments or suggestions. Please reach out to us: [callengine@cisco.com](mailto:callengine@cisco.com). Many thanks! The CSA development team.



# Collaboration Solutions Analyzer

Empower yourself with TAC tools that help troubleshoot and validate your collaboration solution.

**Tools** About Known issues Release notes

### Log Analyzer

Upload logs from your collaboration devices to automatically detect, troubleshoot and resolve issues.

[Upload files](#)

### CollabEdge Validator

Speed up your Mobile and Remote Access feature deployment or troubleshooting by doing a step-by-step validation.

[Run the validation](#)

### SRV Checker

Check your public domain for DNS service records and connectivity for various collaboration services.

[Validate services](#)

### B2B Call Tester

Test inbound and outbound calls to and from your deployment.

[Test calls](#)

报告问题主页

## Report an issue ✕

Product

Issue

Details about an issue

[Cancel](#) [Send](#)

报告问题

已启用三个图标，以允许用户提供反馈（麦克风图标）、查看用户文档（问号图标）和打开用户设置（齿轮图标）。



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图标

## 支持相关信息

[配置CUBE和TDM网关的调试集合](#)

[贯穿思科IOS XE 17.5的思科统一边界元素配置指南](#)

[章节：SIP配置文件](#)

[在CUBE企业常见使用案例上使用SIP配置文件](#)

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请注意：即使是最好的机器翻译，其准确度也不及专业翻译人员的水平。

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