Use SIP Profiles on CUBE Enterprise Common Use Cases

Contents

Introduction

This document describes how to use the Session Initiation Protocol (SIP) Profile Test Tool that is available for use on Cisco.com.

Prerequisites

Requirements

The information in this document is based on ISR platforms running Cisco IOS® and Cisco IOS® XE software.

Components Used

Cisco recommends that you have knowledge of these topics:

- Navigation through Cisco IOS®
- SIP message format and transactions

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

SIP Profiles are used in order to manipulate header information in the SIP messages. They can also be used to make changes in the Session Description Protocol (SDP), which is used to negotiate media.

Common SIP Message Normalization Scenarios

This section provides several SIP message normalization scenarios that have been seen frequently. Each scenario includes the configuration required on Cisco IOS for you reference and a screenshot from the SIP Profile Test Tool that is mentioned in the Introduction.

These scenarios can be used as references for other manipulation required on the SIP messages.

Copy Value from Diversion Header to the From Header

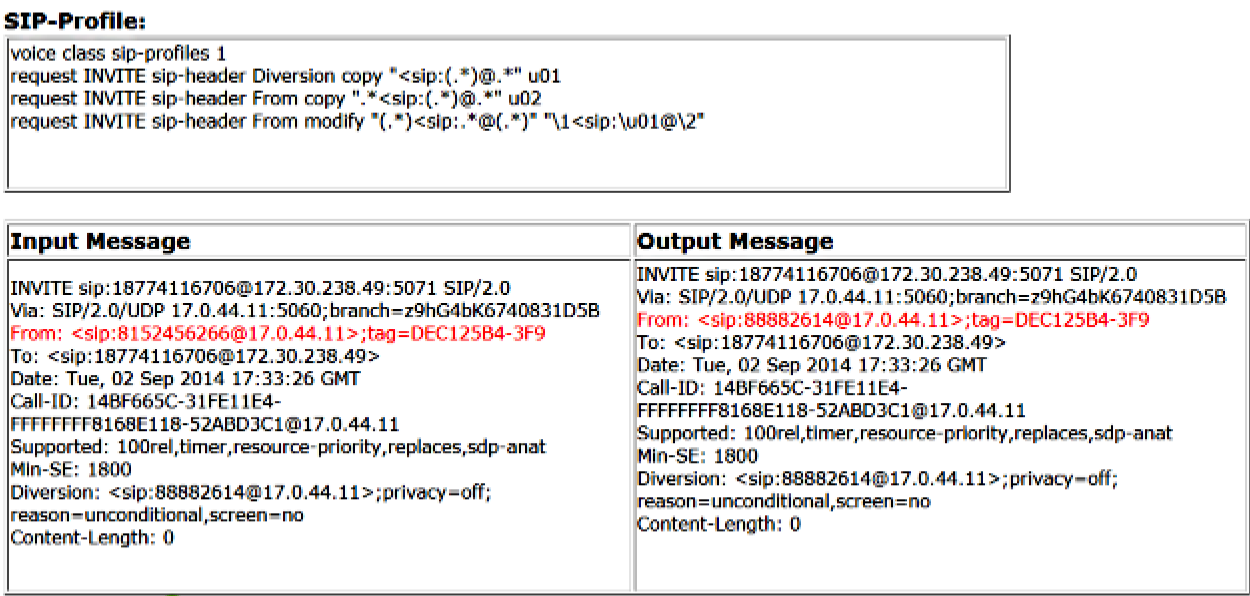
```
voice class sip-profiles 1

request INVITE sip-header Diversion copy "<sip:(.*)@.*" u01

request INVITE sip-header From copy ".*<sip:(.*)@.*" u02

request INVITE sip-header From modify "(.*)<sip:.*@(.*)" "\1<sip:\u01@\2"

request INVITE sip-header From modify "<sip:@" "<sip:\u02@"
```



Copy Number from To Header in an Incoming Invite to the REQ-URI Parameter (Prior to Cisco IOS Version 15.4)

Copy the number in the To header in an inbound Invite message and modify the outgoing INVITE:

```
voice class sip-copylist 1
```

```
sip-header TO

voice class sip-profiles 2
request INVITE peer-header sip TO copy "sip:(.*)@" u01
request INVITE sip-header SIP-Req-URI modify ".*@(.*)" "INVITE sip:\u01@\1"
```

SIP-Profile:

```
voice class sip-copylist 1
sip-header TO

voice class sip-profiles 2
request INVITE peer-header sip TO copy "sip:(.*)@" u01
request INVITE sip-header SIP-Req-URI modify ".*@(.*)" "INVITE sip:\u01@\1"
```

Input Message	**Output Message**
INVITE sip:+18774116700@172.30.238.49:5071 SIP/2.0 Via: SIP/2.0/UDP 17.0.44.11:5060;branch=z9hG4bK6740831D5B From: <sip:8152456266@17.0.44.11>;tag=DEC125B4-3F9 To: <sip:18774116706@172.30.238.49> Date: Tue, 02 Sep 2014 17:33:26 GMT Call-ID: 14BF665C-31FE11E4- FFFFFFFF8168E118-52ABD3C1@17.0.44.11 Supported: 100rel,timer,resource-priority,replaces,sdp-anat Min-SE: 1800 Diversion: <sip:88882614@17.0.44.11>;privacy=off; reason=unconditional,screen=no Content-Length: 0	INVITE sip:18774116706@172.30.238.49:5071 SIP/2.0 Via: SIP/2.0/UDP 17.0.44.11:5060;branch=z9hG4bK6740831D5B From: <sip:8152456266@17.0.44.11>;tag=DEC125B4-3F9 To: <sip:18774116706@172.30.238.49> Date: Tue, 02 Sep 2014 17:33:26 GMT Call-ID: 14BF665C-31FE11E4- FFFFFFFF8168E118-52ABD3C1@17.0.44.11 Supported: 100rel,timer,resource-priority,replaces,sdp-anat Min-SE: 1800 Diversion: <sip:88882614@17.0.44.11>;privacy=off; reason=unconditional,screen=no Content-Length: 0

Copy Number from To Header in an Incoming Invite to the REQ-URI Parameter (with Inbound SIP Profiles)

```
voice class sip-profiles 1
 request INVITE sip-header TO copy "sip:(.*)@" u01
 request INVITE sip-header SIP-Req-URI modify ".*@(.*)" "INVITE sip:\u01@\1"

voice service voip
sip
sip-profiles inbound
sip-profiles 1 inbound
```

SIP-Profile:

```
voice class sip-profiles 1
request INVITE sip-header TO copy "sip:(.*)@" u01
request INVITE sip-header SIP-Req-URI modify ".*@(.*)" "INVITE sip:\u01@\1"

voice service voip
sip
sip-profiles inbound
sip-profiles 1 inbound
```

Input Message	Output Message
INVITE sip:+18774116700@172.30.238.49:5071 SIP/2.0 Via: SIP/2.0/UDP 17.0.44.11:5060;branch=z9hG4bK6740831D5B From: <sip:8152456266@17.0.44.11>;tag=DEC125B4-3F9 To: <sip:18774116706@172.30.238.49> Date: Tue, 02 Sep 2014 17:33:26 GMT Call-ID: 14BF665C-31FE11E4-FFFFFFFF8168E118-52ABD3C1@17.0.44.11 Supported: 100rel,timer,resource-priority,replaces,sdp-anat Min-SE: 1800 Diversion: <sip:88882614@17.0.44.11>;privacy=off;reason=unconditional,screen=no Content-Length: 0	INVITE sip:18774116706@172.30.238.49:5071 SIP/2.0 Via: SIP/2.0/UDP 17.0.44.11:5060;branch=z9hG4bK6740831D5B From: <sip:8152456266@17.0.44.11>;tag=DEC125B4-3F9 To: <sip:18774116706@172.30.238.49> Date: Tue, 02 Sep 2014 17:33:26 GMT Call-ID: 14BF665C-31FE11E4-FFFFFFFF8168E118-52ABD3C1@17.0.44.11 Supported: 100rel,timer,resource-priority,replaces,sdp-anat Min-SE: 1800 Diversion: <sip:88882614@17.0.44.11>;privacy=off;reason=unconditional,screen=no Content-Length: 0

One-way / No-way Audio Interoperability Issues with Provider

```
voice class sip-profiles 200
request ANY sdp-header Audio-Attribute modify "a=inactive" "a=sendrecv"
request ANY sdp-header Audio-Connection-Info modify "0.0.0.0" "CUBEâ€™s IP"
```

SIP-Profile:

```
voice class sip-profiles 200
request ANY sdp-header Audio-Attribute modify "a=inactive" "a=sendrecv"
request ANY sdp-header Audio-Connection-Info modify "0.0.0.0" "10.10.10.1"
```

Input Message	Output Message
INVITE sip:18774116706@172.30.238.49:5071 SIP/2.0 Content-Disposition: session;handling=required Content-Length: 261 v=0 o=CiscoSystemsSIP-GW-UserAgent 1796 4793 IN IP4 17.0.44.11 s=SIP Call c=IN IP4 17.0.44.11 t=0 0 m=audio 0 RTP/AVP 0 101 19 c=IN IP4 0.0.0.0 a=rtpmap:0 PCMU/8000 a=inactive a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-16 a=rtpmap:19 CN/8000 a=ptime:20	INVITE sip:18774116706@172.30.238.49:5071 SIP/2.0 Content-Disposition: session;handling=required Content-Length: 273 v=0 o=CiscoSystemsSIP-GW-UserAgent 1796 4793 IN IP4 17.0.44.11 s=SIP Call c=IN IP4 17.0.44.11 t=0 0 m=audio 0 RTP/AVP 0 101 19 c=IN IP4 10.10.10.1 a=rtpmap:0 PCMU/8000 a=sendrecv a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-16 a=rtpmap:19 CN/8000 a=ptime:20

Remove the UPDATE Method Support to Avoid Interoperability Issues

```
voice class sip-profiles 200
 request ANY sip-header Allow-Header modify ", UPDATE" ""
```

SIP-Profile:

```
voice class sip-profiles 200
request ANY sip-header Allow-Header modify ", UPDATE" ""
```

Input Message	Output Message
INVITE sip:18774116706@172.30.238.49:5071 SIP/2.0 Via: SIP/2.0/UDP 17.0.44.11:5060;branch=z9hG4bK6740831D5B From: <sip:8152456266@17.0.44.11>;tag=DEC125B4-3F9 To: <sip:18774116706@172.30.238.49> Date: Tue, 02 Sep 2014 17:33:26 GMT Call-ID: 14BF665C-31FE11E4-FFFFFFFF8168E118-52ABD3C1@17.0.44.11 Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER Content-Length: 0	INVITE sip:18774116706@172.30.238.49:5071 SIP/2.0 Via: SIP/2.0/UDP 17.0.44.11:5060;branch=z9hG4bK6740831D5B From: <sip:8152456266@17.0.44.11>;tag=DEC125B4-3F9 To: <sip:18774116706@172.30.238.49> Date: Tue, 02 Sep 2014 17:33:26 GMT Call-ID: 14BF665C-31FE11E4-FFFFFFFF8168E118-52ABD3C1@17.0.44.11 Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER Content-Length: 0

IP Address to Domain Name Conversion

```
voice class sip-profiles 1
 request ANY sip-header SIP-Req-URI modify "10.67.138.241:5060" "sipp.cisco.com"
```

SIP-Profile:

```
voice class sip-profiles 1
request ANY sip-header SIP-Req-URI modify "10.67.138.241:5060" "sipp.cisco.com"
```

Input Message	Output Message
INVITE sip:9819940331@10.67.138.241:5060 SIP/2.0 Via: SIP/2.0/UDP 17.0.44.11:5060;branch=z9hG4bK6740831D5B From: <sip:8152456266@17.0.44.11>;tag=DEC125B4-3F9 To: <sip:18774116706@172.30.238.49> Date: Tue, 02 Sep 2014 17:33:26 GMT Call-ID: 14BF665C-31FE11E4-FFFFFFFF8168E118-52ABD3C1@17.0.44.11 Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER Content-Length: 0	INVITE sip:9819940331@sipp.cisco.com SIP/2.0 Via: SIP/2.0/UDP 17.0.44.11:5060;branch=z9hG4bK6740831D5B From: <sip:8152456266@17.0.44.11>;tag=DEC125B4-3F9 To: <sip:18774116706@172.30.238.49> Date: Tue, 02 Sep 2014 17:33:26 GMT Call-ID: 14BF665C-31FE11E4-FFFFFFFF8168E118-52ABD3C1@17.0.44.11 Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER Content-Length: 0

Add a Prefix in the Diversion Header

```
voice class sip-profiles 1
 request ANY sip-header Diversion modify "sip:(.*)@" "sip:704264\1@"
```

SIP-Profile:

```
voice class sip-profiles 1
request ANY sip-header Diversion modify "sip:(.*)@" "sip:704264\1@"
```

Input Message	Output Message
INVITE sip:9819940331@10.67.138.241:5060 SIP/2.0 Via: SIP/2.0/UDP 17.0.44.11:5060;branch=z9hG4bK6740831D5B From: <sip:8152456266@17.0.44.11>;tag=DEC125B4-3F9 To: <sip:18774116706@172.30.238.49> Date: Tue, 02 Sep 2014 17:33:26 GMT Call-ID: 14BF665C-31FE11E4-FFFFFFFF8168E118-52ABD3C1@17.0.44.11 Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER Diversion: <sip:2614@17.0.44.11>;privacy=off;reason=unconditional,screen=no Content-Length: 0	INVITE sip:9819940331@10.67.138.241:5060 SIP/2.0 Via: SIP/2.0/UDP 17.0.44.11:5060;branch=z9hG4bK6740831D5B From: <sip:8152456266@17.0.44.11>;tag=DEC125B4-3F9 To: <sip:18774116706@172.30.238.49> Date: Tue, 02 Sep 2014 17:33:26 GMT Call-ID: 14BF665C-31FE11E4-FFFFFFFF8168E118-52ABD3C1@17.0.44.11 Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER Diversion: <sip:7042642614@17.0.44.11>;privacy=off;reason=unconditional,screen=no Content-Length: 0

Set DID Number in Diversion Header

```
voice class sip-profiles 1
 request INVITE sip-header Diversion modify "sip:(.*)@" "sip:7042642614@"
```

SIP-Profile:

```
voice class sip-profiles 1
 request INVITE sip-header Diversion modify "sip:(.*)@" "sip:7042642614@"
```

Input Message	Output Message
INVITE sip:18774116706@172.30.238.49:5060 SIP/2.0 Via: SIP/2.0/UDP 17.0.44.11:5060;branch=z9hG4bKD23DB From: <sip:8152456266@17.0.44.11>;tag=28B470-1CC0 To: <sip:18774116706@172.30.238.49> Date: Thu, 10 Sep 2020 06:02:45 GMT Call-ID: 1462FCC6-F26211EA-813AE871-299EC8ED@17.0.44.11 Supported: timer,resource-priority,replaces,sdp-anat Min-SE: 1800 **Diversion: <sip:88882614@17.0.44.11>;privacy=off;reason-unconditional,screen=no** Content-Length: 0	INVITE sip:18774116706@172.30.238.49:5060 SIP/2.0 Via: SIP/2.0/UDP 17.0.44.11:5060;branch=z9hG4bKD23DB From: <sip:8152456266@17.0.44.11>;tag=28B470-1CC0 To: <sip:18774116706@172.30.238.49> Date: Thu, 10 Sep 2020 06:02:45 GMT Call-ID: 1462FCC6-F26211EA-813AE871-299EC8ED@17.0.44.11 Supported: timer,resource-priority,replaces,sdp-anat Min-SE: 1800 **Diversion: <sip:7042642614@17.0.44.11>;privacy=off;reason-unconditional,screen=no** Content-Length: 0

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Remove Diversion Header

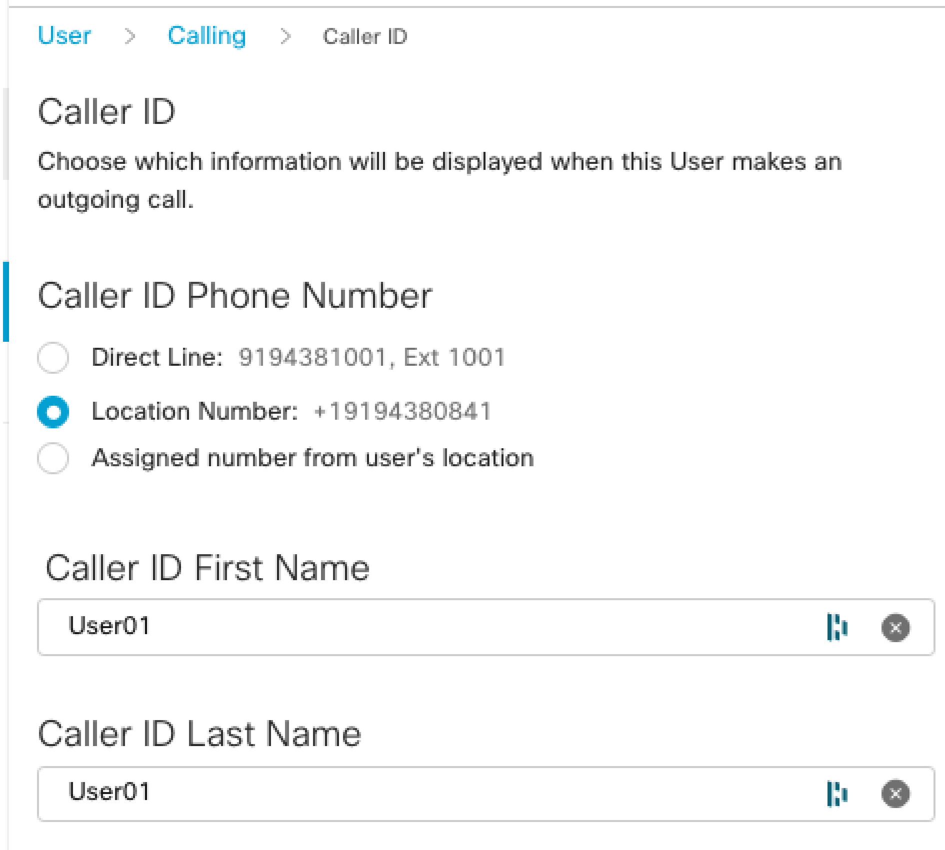
```
voice class sip-profiles 1
 request INVITE sip-header Diversion remove
```

SIP-Profile:

```
voice class sip-profiles 1
 request INVITE sip-header Diversion remove
```

Input Message	Output Message
INVITE sip:18774116706@172.30.238.49:5060 SIP/2.0 Via: SIP/2.0/UDP 17.0.44.11:5060;branch=z9hG4bKD23DB From: <sip:8152456266@17.0.44.11>;tag=28B470-1CC0 To: <sip:18774116706@172.30.238.49> Date: Thu, 10 Sep 2020 06:02:45 GMT Call-ID: 1462FCC6-F26211EA-813AE871-299EC8ED@17.0.44.11 Supported: timer,resource-priority,replaces,sdp-anat Min-SE: 1800 **Diversion: <sip:88882614@17.0.44.11>;privacy=off; reason-unconditional,screen=no** Content-Length: 0	INVITE sip:18774116706@172.30.238.49:5060 SIP/2.0 Via: SIP/2.0/UDP 17.0.44.11:5060;branch=z9hG4bKD23DB From: <sip:8152456266@17.0.44.11>;tag=28B470-1CC0 To: <sip:18774116706@172.30.238.49> Date: Thu, 10 Sep 2020 06:02:45 GMT Call-ID: 1462FCC6-F26211EA-813AE871-299EC8ED@17.0.44.11 Supported: timer,resource-priority,replaces,sdp-anat Min-SE: 1800 Content-Length: 0

Copy Location Number for Caller ID in Local Gateway (Webex Calling Deployments in United States, Canada, and Puerto Rico)




```
voice service voip
 sip
  sip-profile inbound
```

```
voice class sip-profiles 201
 rule 1 request INVITE sip-header From copy "<sip:(.*)@" u01
 rule 2 request INVITE sip-header P-Asserted-Identity modify "<sip:.*@(.*)>" "<sip:\u01@\1>"

voice class tenant 200
 sip-profiles 201 inbound
```

SIP-Profile:

```
voice class sip-profiles 201
 rule 1 request INVITE sip-header From copy "<sip:(.*)@" u01
 rule 2 request INVITE sip-header P-Asserted-Identity modify "<sip:.*@(.*)>" "<sip:\u01@\1>"
```

Input Message	Output Message
INVITE sip:+19199614190@1.1.1.1:5061;transport=tls;dtg=rtplgw9687_lgu SIP/2.0 Via:SIP/2.0/TLS 139.177.65.12:8934;branch=z9hG4bKBroadworksSSE.-1.1.1.1V57722-0-100-973405068-1626801459363- From:"User01 User01"<sip:+19194380841@139.177.65.12;user=phone>;tag=973405068-1626801459363- To:<sip:+19199614190@90444895.cisco-bcld.com;user=phone> Call-ID:SSE171739363200721170655236 5@139.177.65.12 CSeq:100 INVITE Contact:<sip:139.177.65.12:8934;transport=tls> P-Asserted-Identity:"User01 User01"<sip:+19194381001@10.21.0.214;user=phone>	INVITE sip:+19199614190@pstn.com:5 Via: SIP/2.0/UDP 1.1.1.1:5060;branch=z From: "User01 User01" <sip:+1919438084 To: <sip:+19199614190@pstn.com> Date: Tue, 20 Jul 2021 17:59:26 GMT Call-ID: E50FFB7-E8BB11EB-B57BD6 Contact: <sip:+19194380841@1.1.1.1:50 Allow-Events: telephone-event Max-Forwards: 68 P-Asserted-Identity: "User01 User01"

Possible Issues

Here are some possible issues you can encounter.

- After Cisco IOS Version 15.4, the SIP profile feature is introduced to modify inbound SIP messages as well.
- Cisco IOS Versions 15.3 and earlier only support SIP profiles in the outbound direction.

Related Information

[In Depth Explanation of Cisco IOS and IOS-XE Call Routing]

[Understanding Inbound and Outbound Dial Peers Matching on IOS Platforms]