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Introduction

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

Prerequisites

Requirements

The information in this document is based on Cisco Integrated Service Router (ISR) Generation 1 (2800/3800) or Generation 2 (2900/3900) series.

Components Used

Cisco recommends that you have knowledge of these topics:

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

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

SIP-Profile:

```
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"
```

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 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:88882614@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 (Prior to Cisco IOS Vesion 15.4)

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

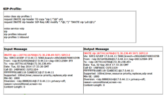
SIP-Profile:

```
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)



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

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

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

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

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

Possible Issues

Here are some possible issues you might 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.