在CUBE企業常見用例上使用SIP配置檔案

目錄

簡介

本檔案介紹如何使用Cisco.com上提供的作業階段啟始通訊協定(SIP)設定檔測試工具。

必要條件

需求

本文檔中的資訊基於運行Cisco IOS®和Cisco IOS® XE軟體的ISR平台。

採用元件

思科建議您瞭解以下主題：

- 在Cisco IOS®中導航
- SIP消息格式和事務

本文中的資訊是根據特定實驗室環境內的裝置所建立。文中使用到的所有裝置皆從已清除（預設）的組態來啟動。如果您的網路運作中，請確保您瞭解任何指令可能造成的影響。

背景資訊

SIP配置檔案用於處理SIP消息中的報頭資訊。它們還可用於更改會話描述協定(SDP)，該協定用於協商介質。

常見SIP消息規範化方案

本部分提供了幾種經常出現的SIP消息規範化方案。每個場景都包括Cisco IOS上所需的配置供您參考，以及簡介中提到的SIP配置檔案測試工具的螢幕截圖。

這些場景可用作SIP消息中所需的其他操作的參考。

將值從轉移題頭複製到起始題頭

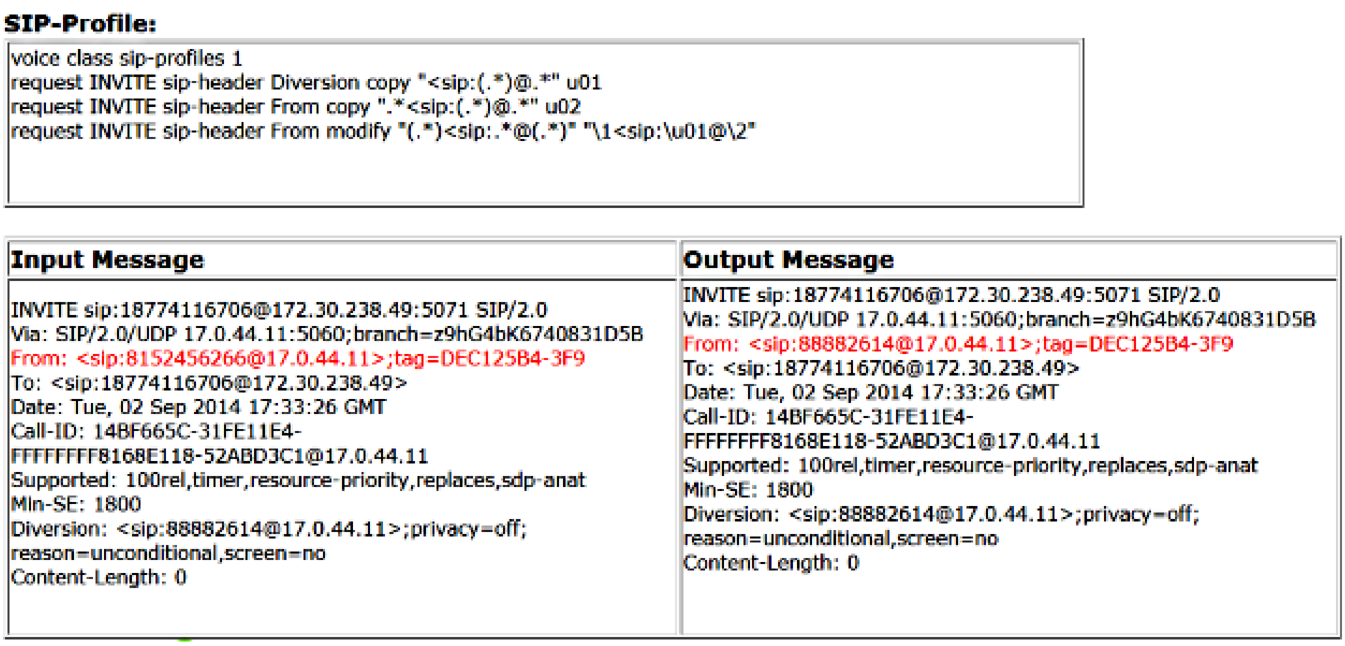
```
voice class sip-profiles 1

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@"
```



將傳入邀請中的號碼從To報頭複製到REQ-URI引數（Cisco IOS版本15.4之前）

複製入站Invite消息的To標頭中的號碼並修改傳出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

```
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
```

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
```

將傳入邀請中的號碼從到報頭複製到REQ-URI引數（帶入站SIP配置檔案）

```
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

與提供商的單向/單向音訊互操作性問題

```
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" "CUBE’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

刪除UPDATE方法支援以避免互操作性問題

```
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

```
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
```

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, REFER,
SUBSCRIBE, NOTIFY, INFO, REGISTER
Content-Length: 0
```

IP位址到網域名的轉換

```
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

```
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
```

Output Message

```
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
```

在轉接標頭中新增字首

```
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

在轉移報頭中設定DID編號

```
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

移除轉接標頭

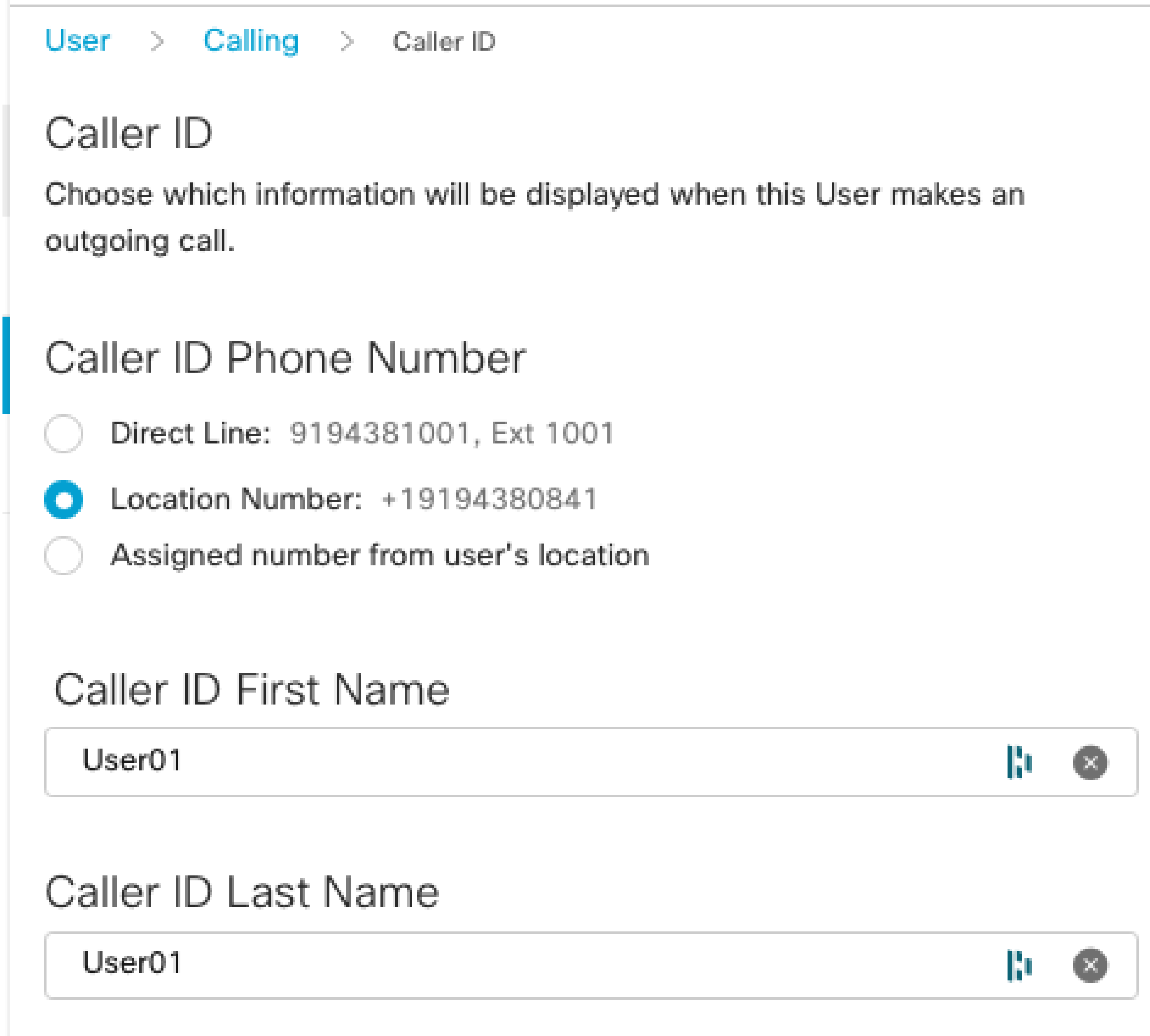
```
voice class sip-profiles 1
 request INVITE sip-header Diversion remove
```

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

本地網關中呼叫者ID的複製位置號碼（美國、加拿大和波多黎各的Webex呼叫部署）




```
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:SSE1717393632007211706552365@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:5080 SIP/2.0 Via: SIP/2.0/UDP 1.1.1.1:5060;branch=z9hG4bK13CA141F20 From: "User01 User01" <sip:+19194380841@pstn.com>;tag=CB0B7295-DB7 To: <sip:+19199614190@pstn.com> Date: Tue, 20 Jul 2021 17:59:26 GMT Call-ID: E50FFB7-E8BB11EB-B57BD6D5-6AE138B@1.1.1.1 Contact: <sip:+19194380841@1.1.1.1:5060> Allow-Events: telephone-event Max-Forwards: 68 **P-Asserted-Identity: "User01 User01" <sip:+19194380841@1.1.1.1>**

可能的問題

以下是您可以遇到的一些可能問題。

- 在Cisco IOS版本15.4之後，還引入SIP配置檔案功能來修改入站SIP消息。
- Cisco IOS版本15.3及更低版本僅支援出站方向的SIP配置檔案。

相關資訊

[Cisco IOS和IOS-XE呼叫路由的深入說明]

[瞭解IOS平台上的入站和出站撥號對等體匹配]

關於此翻譯

思科已使用電腦和人工技術翻譯本文件，讓全世界的使用者能夠以自己的語言理解支援內容。請注意，即使是最佳機器翻譯，也不如專業譯者翻譯的內容準確。Cisco Systems, Inc. 對這些翻譯的準確度概不負責，並建議一律查看原始英文文件（提供連結）。