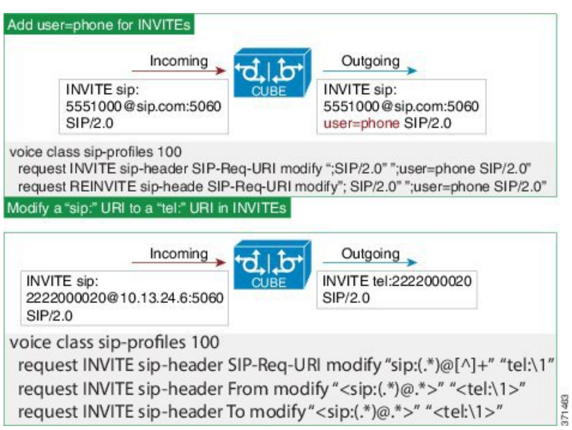
SIP Profiles

Overview

Protocol translation and repair are a key Cisco Unified Border Element (CUBE) function. CUBE can be deployed between two incompatible SIP devices to normalize messaging, ensuring end-to-end compatibility.

Service providers may have policies for which SIP messaging fields should be present (and the values they contain) before a SIP call enters their network. Similarly, enterprises and small businesses may have policies for the information that can enter or exit their networks for policy or security reasons from a service provider SIP trunk.

Figure 1: SIP Profile



CUBE Session Initiation Protocol (SIP) profiles change SIP incoming or outgoing messages so that interoperability between incompatible devices can be ensured.

You can configure SIP profiles with rules to add, remove, copy, or modify the SIP, Session Description Protocol (SDP), and peer headers that enter or leave CUBE. The rules in a SIP profile configuration can also be tagged with a unique number. Tagging the rules allows you to insert or delete rules at any position of the existing SIP profile configuration without deleting and reconfiguring the entire voice-class sip profile.

Figure 2: Incoming and Outgoing Messages Where SIP Profiles Can Be Applied

In addition to network policy compliance, CUBE SIP profiles can be used to resolve incompatibilities between SIP devices inside the enterprise network. These are some of the situations in which incompatibilities can arise:

- A device rejects an unknown header (value or parameter) instead of ignoring it.
- A device sends incorrect data in a SIP message.
- A device does not implement (or implements incorrectly) protocol procedures.
- A device expects an optional header value or parameter, or an optional protocol procedure that can be implemented in multiple ways.
- A device sends a value or parameter that must be changed or suppressed before it leaves or enters the network.
- Variations in the SIP standards on how to achieve certain functions.

The SIP profiles feature on CUBE provides a solution to these incompatibilities and customization issues.

SIP profiles can also be used to change a header name from the long form to the compact form. For example, From to f. This can be used as a way to reduce the length of a SIP message. By default, the device never sends the compact form of the SIP messages although it receives either the long or the short form.

Feature Information

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.

Table 1: Feature Information for SIP Profiles

Feature Name	Releases	Feature Information
SIP Profiles (for inbound messages)	Baseline Functionality	This feature modifies the following commands: The **inbound** keyword was added to the **sip-profiles** and **voice-class sip profiles** commands.

Important Characteristics of SIP Profiles

Given below are a few important notes for SIP Profiles:

- Session Initiation Protocol (SIP) and Session Description Protocol (SDP) headers are supported. SDP can be either a standalone body or part of a Multipurpose Internet Mail Extensions (MIME) message.
- The rules that are configured for an INVITE message are applied only to the first INVITE of a call. A special REINVITE keyword is used to manipulate subsequent INVITEs of a call.
- Manipulation of SIP headers by outbound SIP profiles occurs as the last step before the message leaves the CUBE device; that is, after destination dial-peer matching has taken place. Changes to the SIP messages are not remembered or acted on by the CUBE application. The Content-length field is recalculated after the SIP Profiles rules are applied to the outgoing message.
- If the **ANY** keyword is used in place of a header, it indicates that a rule must be applied to any message within the specified category.
- SIP header modification can be cryptic. It can sometimes be easier to remove a header and add it back (with the new value), rather than modifying it.
- To include '?' (question-mark) character as part of match-pattern or replace-pattern, you must press "Ctrl+v" keys and then type '?'. This is needed to treat ‘?’ as an input character itself instead of usual device help prompt.

Note Regex features like look-ahead, look-behind, operator, and non capturing group are not supported (for example, **?!**, **?:**, **?<=**,, and so| on).

- For header values used to add, modify or copy a header:
  - If a whitespace occurs, the entire value must be included between double quotes. For example, “User-Agent: CISCO CUBE”
  - If double quotes occur, a back slash must prefix the double quotes. For example, “User-Agent: \”CISCO\” CUBE”
  - Basic regular expressions are supported.
- If an incoming SIP message contains certain proprietary attributes, CUBE can copy these unsupported SDP attributes or lines from incoming leg to outgoing leg using a SIP profile rule.
- The copy variable can be used in an outbound profile to add or modify the outgoing message.
- Copy Variables u01 to u99 are shared by inbound and outbound SIP Profiles.

Inbound SIP Profile:

- If the incoming message contains multiple instances of the same header, the header values are stored as a comma separated list, and this needs to be considered while modifying it.
- Modification by an inbound SIP profile takes place before regular SIP call processing happens so that behavior of CUBE would be as if it received the message directly without modification.

If inbound dial peer matching fails as required information could not be extracted from headers (like Request-URI, Via, From or To) due to issues in them, global level sip-profile config is applied. An example is a request with invalid SIP-Req-URI.

- After modification by inbound SIP Profiles, the parameters in SIP message might change, which might change the inbound dial-peer matched when actual dial-peer lookup is done.
- In the register pass-through feature, there is only one dial-peer for register and response. So both register from phone and response from registrar would go through the same inbound sip profile under the dial-peer if any.

Restrictions

- Removal or addition of mandatory headers is not supported. You can only modify mandatory headers Mandatory SIP headers include To, From, Via, CSeq, Call-Id, and Max-Forwards. Mandatory SDP headers include v, o, s, t ,c, and m.
- Addition or removal of entire Multipurpose Internet Mail Extensions (MIME) or (Session Description Protocol) SDP bodies from SIP messages is not supported.
- Syntax checking is not performed on SIP messages after SIP profile rules have been applied. Changes that are specified in the SIP profile should result in valid SIP protocol exchanges.
- The header length (including header name) after modification should not exceed 300 characters. Max header length for add value is approximately 220 characters. Max SDP length is 2048 characters. If any header length exceeds this maximum value after applying SIP profiles, then the profile is not applied.
- If a header-name is changed to its compact form, further SIP profile rules cannot be applied on that header. Thus a SIP profile rule modifying a header name to its compact form must be the last rule on that header.
- The "image" m-line attributes (m=image 16850 udptl t38) cannot be modified using SIP profiles. SIP profiles can be applied only on audio and video m-lines in SDP.
- In a high-availability (HA) scenario, SIP profiles copy variable data is not check-pointed to standby.
- Limitations and restrictions of outbound SIP profiles apply to inbound SIP profiles as well.
- You cannot configure more than 99 variables for the SIP profiles copy option.
- Once a SIP profile is configured using rule tag, you cannot add rules without tags in the same profile and conversely.
- If a SIP profile is applied to modify the SDP content of a SIP message, CUBE does not increment the "o=" line version, which may cause ITSPs to disconnect the call. CUBE does not store the modified SDP after the application of the SIP profile.

  Note that manipulation of SIP messages by outbound SIP profiles occurs as the final step before the outgoing message leaves the CUBE device, and occurs after destination dial-peer matching has taken place. Changes to SIP messages are not remembered or acted on by CUBE. The Content-length field is recalculated after SIP Profile rules are applied to outgoing messages.

How to Configure SIP Profiles

To use SIP Profiles, you must first configure the profile, then apply it either at the global (all dial-peers), tenant, or dial-peer levels. After a SIP profile is configured, it can be applied as an inbound or outbound profile.

Configure SIP Profile Rules Using Rule Tags

Configuring SIP profile rules using rule tags, allows you to perform the following tasks:

- Add a new rule at a chosen position without having to replace the whole profile.
- Modify a specific rule by specifying its rule tag.
- Remove a rule by specifying only its rule tag.

Below are the rule tag behaviors that must be considered while using rule tags in a SIP profile configuration:

- If a rule is added with the tag of an existing rule, then the existing rule is overwritten with the new rule.
- For inserting a rule at the desired position, the SIP profile configuration should be in rule format. In case the SIP profile is in nonrule format, upgrade the SIP profiles to rule format before inserting a rule.
- If a new rule is inserted, the new rule takes the position that is specified in **before** *tag*. The subsequent rules are incremented sequentially.

  For example:

  rule before 10 request INVITE sip-header From modify "(<.*:)(.*@)" "\1gateway@"

- When a rule is removed, the tags that are associated with the subsequent rules remain unchanged.
- If a rule is added to a vacant tag, the new rule gets associated with the vacant tag and the subsequent rules remain unchanged.

Upgrade or Downgrade SIP Profile Configurations

You can upgrade SIP profile rules to include rule tags or downgrade to remove them.

SUMMARY STEPS

1. **enable**
2. Enter the following to upgrade SIP profiles configurations to rule-format:
   - **voice sip sip-profiles upgrade**
3. Enter the following to downgrade SIP profiles configurations to non-rule format:
   - **voice sip sip-profiles downgrade**
4. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable**	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	Enter the following to upgrade SIP profiles configurations to rule-format: • **voice sip sip-profiles upgrade** **Example:** In EXEC(#) mode: `Device#voice sip sip-profiles upgrade`	Upgrades all SIP Profiles to rule-format configurations.
Step 3	Enter the following to downgrade SIP profiles configurations to non-rule format: • **voice sip sip-profiles downgrade** **Example:** In EXEC(#) mode: `Device#voice sip sip-profiles downgrade`	Downgrades all SIP Profiles from rule-format configurations to non-rule format configurations.
Step 4	**end**	Exits privileged EXEC mode.

What to do next

Now apply the SIP Profile as an inbound or outbound SIP profile.

Configure a SIP Profile to Manipulate SIP Request or Response Headers

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class sip-profiles** *profile-id*
4. Enter one of the following to add, remove, modify SIP headers:
   - [rule *x*] **request** *message* {**sip-header** | **sdp-header**} *header-to-add* **add** *header-value-to-add*
   - [rule *x*] **request** *message* {**sip-header** | **sdp-header**} *header-to-remove* **remove**
   - [rule *x*] **request** *message* {**sip-header** | **sdp-header**} *header-to-modify* **modify** *header-value-to-match header-value-to-replace*
5. Enter one of the following to add, remove, or modify SIP response headers:
   - [rule *x*] **response** *message* [**method** *method-type*] {**sip-header** | **sdp-header**} *header-to-add* **add** *header-value-to-add*
   - [rule *x*] **response** *message* [**method** *method-type*] {**sip-header** | **sdp-header**} *header-to-remove* **remove**

- [rule *x*] **response** *message* [**method** *method-type*] {**sip-header** | **sdp-header**} *header-to-modify* **modify** *header-value-to-match header-value-to-replace*

6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable**	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal**	Enters global configuration mode.
Step 3	**voice class sip-profiles** *profile-id* **Example:** `Device(config)# voice class sip-profiles 10`	Creates a SIP Profile and enters voice class configuration mode.

Step 4	Enter one of the following to add, remove, modify SIP headers:
	• [rule *x*] **request** *message* {**sip-header** | **sdp-header**} *header-to-add* **add** *header-value-to-add*
	• [rule *x*] **request** *message* {**sip-header** | **sdp-header**} *header-to-remove* **remove**
	• [rule *x*] **request** *message* {**sip-header** | **sdp-header**} *header-to-modify* **modify** *header-value-to-match header-value-to-replace*
		According to your choice, this step does one of the following:
		• Adds a SIP or SDP header to a SIP request.
		• Removes a SIP or SDP header from a SIP request.
		• Modifies a SIP or SDP header in a SIP request.
		• If the **ANY** is used, the rule is applied to the specified header when it appears in any message type.
		• When specifying a profile rule header value:
		  • If a value includes a space, the entire value must be included between double quotes. For example, “User-Agent: CISCO CUBE”
		  • When using double quotes in a value, delimit with a backslash.
		  • Simple regular expressions are supported.

Step 5	Enter one of the following to add, remove, or modify SIP response headers: • [rule *x*] **response** *message* [**method** *method-type*] {**sip-header** | **sdp-header**} *header-to-add* **add** *header-value-to-add* • [rule *x*] **response** *message* [**method** *method-type*] {**sip-header** | **sdp-header**} *header-to-remove* **remove** • [rule *x*] **response** *message* [**method** *method-type*] {**sip-header** | **sdp-header**} *header-to-modify* **modify** *header-value-to-match header-value-to-replace*	According to your choice, this step does one of the following: • Adds a SIP or SDP header to a SIP response. • Removes a SIP or SDP header from a SIP response. • Modifies a SIP or SDP header in a SIP response. • All notes in the previous step are applicable here.

	Command or Action	Purpose
Step 6	**end**	Exits to privileged EXEC mode

Processing Unsupported SDP Headers

To modify SDP headers that CUBE is not natively aware of, first configure SDP pass-through and then make the necessary modifications through the outbound dial-peer.

1. Configure CUBE to pass-through custom SDP on in-leg.
2. Define rule to **Copy** relevant attributes from peer SDP on out leg.
3. Define rule to **Add** or **Modify** attributes in outbound SDP with copied data.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. To enable copying of unsupported SDP attribute from incoming leg to outbound leg, you must enable one of the following commands:
   - In Global VoIP SIP configuration mode

     pass-thru content custom-sdp
   - In dial-peer configuration mode (The configuration is applied on the incoming dial-peer)

     voice-class sip pass-thru content custom-sdp
4. **voice class sip-profiles** *profile-id*
5. Enter one of the following to copy an unsupported SDP line or attribute from peer leg's SDP and add, modify, or remove in the outgoing SDP:
   - [rule *x*] **{request/response} ANY peer-header sdp mline-index** *index* **COPY** *match-pattern copy-variable*
   - [rule *x*] **{request/response} ANY sdp-header mline-index** *indexheader-name* **ADD** *copy-variable*
   - [rule *x*] **{request/response} ANY sdp-header mline-index** *indexheader-name* **MODIFY** *copy-variable + replace-pattern*
   - [rule *x*] **{ request/response} ANY sdp-header mline-index** *indexheader-name* **REMOVE**
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:**	Enters global configuration mode.

	Command or Action	Purpose
	`Device# configure terminal`	

Step 3	To enable copying of unsupported SDP attribute from incoming leg to outbound leg, you must enable one of the following commands:
	• In Global VoIP SIP configuration mode
	pass-thru content custom-sdp
	• In dial-peer configuration mode (The configuration is applied on the incoming dial-peer)
	voice-class sip pass-thru content custom-sdp
	Example:
	In Global VoIP SIP configuration mode:
	`Device(config)# voice service voip`
	`Device(conf-voi-serv)# sip`
	`Device(conf-serv-sip)# pass-thru content custom-sdp`
	Example:
	In Dial-peer configuration mode:
	`Device(config)# dial-peer voice 2 voip`
	`Device(config-dial-peer)# voice-class sip pass-thru content custom-sdp`
		Enables copying of unsupported SDP attributes per m-line to the peer leg so that it can be used in outgoing SIP messages.
		Note Enabling this command does not enable the SDP Passthrough feature.

Step 4	**voice class sip-profiles** *profile-id* **Example:** `Device(config)# voice class sip-profiles 10`	Voice class sip-profile is configured on the outbound dial-peer or as a global configuration. Creates a SIP Profile and enters voice class configuration mode.
Step 5	Enter one of the following to copy an unsupported SDP line or attribute from peer leg's SDP and add, modify, or remove in the outgoing SDP: • [rule *x*] **{request/response} ANY peer-header sdp mline-index** *index* **COPY** *match-pattern copy-variable* • [rule *x*] **{request/response} ANY sdp-header mline-index** *indexheader-name* **ADD** *copy-variable* • [rule *x*] **{request/response} ANY sdp-header mline-index** *indexheader-name* **MODIFY** *copy-variable + replace-pattern* • [rule *x*] **{ request/response} ANY sdp-header mline-index** *indexheader-name* **REMOVE**	M-line Index values: • 0 - A value of zero represents the session level. • 1–6 - A value in the range of one to six represents the m-line number in SDP. Copy: Enables copying of SDP line or attribute from peer leg SDP. Add: Enables adding the copied SDP line or attribute in the outgoing SDP. Modify: Enables modifying SDP line or attribute in the outgoing SDP. Remove: Enables removing SDP line or attribute in the outgoing SDP.
Step 6	**end**	Exits to privileged EXEC mode.

Example: Configuring SIP Profile Rules (Attribute Passing)

```
rule 10 response ANY peer-header sdp mline-index 4 copy “(a=ixmap:0.*)” u01
rule 20 response ANY sdp-header mline-index 4 a=ixmap add “\u01”
```

Example: Configuring SIP Profile Rules (Parameter Passing)

```
rule 30 response ANY peer-header sdp mline-index 2 copy "a=fmtp:126 .*(max-fps=....)" u04
rule 40 response ANY sdp-header mline-index 2 a=fmtp:126 modify ";" ";\u04;"
```

Example: Configuration to Remove an Attribute

```
rule 50 response ANY sdp-header mline-index 4 a=test REMOVE
```

Use Non-standard SIP Headers in SIP Profiles

In addition to the standard set of headers available when creating SIP profile rules, it is possible to carry out a similar set of functions for any other non-standard header.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class sip-profiles** *profile-id*
4. Enter one of the following to add, copy, remove, or modify non-standard SIP request headers:
   - [rule *x*] **request** *message* **sip-header** *non-standard-header-to-add* **add** *non-standard-header-value-to-add*
   - [rule *x*] **request** *message* **sip-header** *non-standard-header-to-copy* **copy** *non-standard-header-value-to-match copy-variable*
   - [rule *x*] **request** *message* **sip-header** *non-standard-header-to-remove* **remove**]
   - [rule *x*] **request** *message* {**sip-header** } *non-standard-header-to-modify* **modify** *non-standard-header-value-to-match non-standard-header-value-to-replace*
5. Enter one of the following to add, copy, remove, or modify non-standard SIP response headers:
   - [rule *x*] **response** *message* [**method** *method-type*] **sip-header** *non-standard-header-to-add* **add** *non-standard-header-value-to-add*
   - [rule *x*] **response** *message* [**method** *method-type*] **sip-header** *non-standard-header-to-copy* **copy** *non-standard-header-value-to-match copy-variable*
   - [rule *x*] **response** *message* [**method** *method-type*] **sip-header** *non-standard-header-to-remove* **remove**
   - [rule *x*] **response** *message* [**method** *method-type*] **sip-header** *non-standard-header-to-modify* **modify** *non-standard-header-value-to-match non-standard-header-value-to-replace*
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable**	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal**	Enters global configuration mode.
Step 3	**voice class sip-profiles** *profile-id* **Example:** `Device(config)# voice class sip-profiles 10`	Creates a SIP Profiles and enters voice class configuration mode.

Step 4	Enter one of the following to add, copy, remove, or modify non-standard SIP request headers:
	• [rule *x*] **request** *message* **sip-header** *non-standard-header-to-add* **add** *non-standard-header-value-to-add*
	• [rule *x*] **request** *message* **sip-header** *non-standard-header-to-copy* **copy** *non-standard-header-value-to-match copy-variable*
	• [rule *x*] **request** *message* **sip-header** *non-standard-header-to-remove* **remove**]
	• [rule *x*] **request** *message* {**sip-header** } *non-standard-header-to-modify* **modify** *non-standard-header-value-to-match non-standard-header-value-to-replace*
		According to your choice, this step does one of the following:
		• Adds a non-standard SIP header to a SIP request.
		• Copies a non-standard SIP header value to a copy variable.
		• Removes a non-standard SIP header from a SIP request.
		• Modifies a non-standard SIP header in a SIP request.
		• If the **ANY** message keyword is used, the rule is applied to the specified header when it appears in any message type.
		• For *non-standard-header-value-to-add* used to add a non-standard header, *non-standard-header-value-to-match* or *non-standard-header-value-to-replace* used to modify a non-standard header when specifying a profile rule header value:
		  • If a value includes a space , the entire value must be included between double quotes. For example, "User-Agent: CISCO CUBE"
		  • When using double quotes in a value, delimit with a backslash.
		  • Simple regular expressions are supported.

Step 5	Enter one of the following to add, copy, remove, or modify non-standard SIP response headers: • [rule *x*] **response** *message* [**method** *method-type*] **sip-header** *non-standard-header-to-add* **add** *non-standard-header-value-to-add*	According to your choice, this step does one of the following: • Adds a non-standard header to a SIP response message. • Copies contents from a non-standard SIP header to a SIP response.

	Command or Action	Purpose
	• [rule *x*] **response** *message* [**method** *method-type*] **sip-header** *non-standard-header-to-copy* **copy** *non-standard-header-value-to-match copy-variable* • [rule *x*] **response** *message* [**method** *method-type*] **sip-header** *non-standard-header-to-remove* **remove** • [rule *x*] **response** *message* [**method** *method-type*] **sip-header** *non-standard-header-to-modify* **modify** *non-standard-header-value-to-match non-standard-header-value-to-replace*	• Removes a non-standard header to a SIP response. • Modifies a non-standard SIP header in a SIP response. • All notes from the previous step are applicable here.
Step 6	**end**	Exits to privileged EXEC mode

Configure a SIP Profile as an Outbound Profile

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. Apply the SIP profile to a dial peer:
    - **voice-class sip profiles** *profile-id* in the dial-peer configuration mode.
    - **sip-profiles** *profile-id* in the global VoIP configuration mode
4. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable**	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal**	Enters global configuration mode.

Step 3	Apply the SIP profile to a dial peer:
	• **voice-class sip profiles** *profile-id* in the dial-peer configuration mode.
	• **sip-profiles** *profile-id* in the global VoIP configuration mode
	Example:
	In dial-peer configuration mode
	`!Applying SIP profiles to one dial peer only`
	`Device (config)# dial-peer voice 10 voip`
	`Device (config-dial-peer)# voice-class sip profiles 30`
	`Device (config-dial-peer)# end`	

	Command or Action	Purpose

	Example:
	In global VoIP SIP mode
	`! Applying SIP profiles globally`
	`Device(config)# voice service voip`
	`Device (config-voi-serv)# sip`
	`Device (config-voi-sip)# sip-profiles 20`
	`Device (config-voi-sip)# end`	

Step 4	**end**	Exits to privileged EXEC mode .

Configure a SIP Profile as an Inbound Profile

You can configure a SIP profile as an inbound profile applied globally or to multiple inbound dial peers. Inbound SIP profiles feature must be enabled before applying it.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **sip-profiles inbound**
6. Apply the SIP profile to a dial peer:
   - **voice-class sip profiles** *profile-id* **inbound** in the dial-peer configuration mode.
   - **sip-profiles** *profile-id* **inbound** in the global VoIP configuration mode
7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable**	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal**	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Device(config)# voice service voip`	Enters global VoIP configuration mode.
Step 4	**sip** **Example:** `Device(config-voi-serv)# sip`	Enters global VoIP SIP configuration mode.

	Command or Action	Purpose
Step 5	**sip-profiles inbound** **Example:** `Device(config-voi-sip)# sip-profiles inbound`	Enables inbound SIP profiles feature.

Step 6	Apply the SIP profile to a dial peer:
	• **voice-class sip profiles** *profile-id* **inbound** in the dial-peer configuration mode.
	• **sip-profiles** *profile-id* **inbound** in the global VoIP configuration mode
	Example:
	In dial-peer configuration mode
	`!Applying SIP profiles to one dial peer only`
	`Device (config)# dial-peer voice 10 voip`
	`Device (config-dial-peer)# voice-class sip profiles 30 inbound`
	`Device (config-dial-peer)# end`
	Example:
	In global VoIP SIP mode
	`! Applying SIP profiles globally`
	`Device(config)# voice service voip`
	`Device (config-voi-serv)# sip`
	`Device (config-voi-sip)# sip-profiles 20 inbound`
	`Device (config-voi-sip)# end`	

Step 7	**end**	Exits to privileged EXEC mode

Supported SIP Messages

This section provides the CLI options of the SIP messages that you can process with the CUBE SIP profiles feature.

SIP Requests

Supported SIP requests are the following:

```
ACK        sip ack
ANY        any sip request
BYE        sip bye
CANCEL     sip cancel
COMET      sip comet
INFO       sip info
INVITE     sip invite
NOTIFY     sip notify
OPTIONS    sip options
PRACK      sip prack
PUBLISH    sip publish
```

```
REFER      sip refer
REGISTER   sip register
REINVITE   sip reinvite
SUBSCRIBE  sip subscribe
UPDATE     sip info
```

SIP Responses

Supported SIP responses are the following:

```
100  Response code 100
180  Response code 180
181  Response code 181
182  Response code 182
183  Response code 183
200  Response code 200
202  Response code 202
300  Response code 300
301  Response code 301
302  Response code 302
305  Response code 305
380  Response code 380
400  Response code 400
401  Response code 401
402  Response code 402
403  Response code 403
404  Response code 404
405  Response code 405
406  Response code 406
407  Response code 407
408  Response code 408
409  Response code 409
410  Response code 410
412  Response code 412
413  Response code 413
414  Response code 414
415  Response code 415
416  Response code 416
417  Response code 417
420  Response code 420
421  Response code 421
422  Response code 422
423  Response code 423
480  Response code 480
481  Response code 481
482  Response code 482
483  Response code 483
484  Response code 484
485  Response code 485
486  Response code 486
487  Response code 487
488  Response code 488
489  Response code 489
491  Response code 491
493  Response code 493
500  Response code 500
501  Response code 501
502  Response code 502
503  Response code 503
504  Response code 504
505  Response code 505
513  Response code 513
580  Response code 580
600  Response code 600
```

```
603  Response code 603
604  Response code 604
606  Response code 606
ANY  Any Response
```

SIP Headers

Supported SIP headers are the following:

Note Non-standard SIP headers are also supported.

```
Accept-Contact            SIP header Accept-Contact
Accept-Encoding           SIP header Accept-Encoding
Accept-Header             SIP header Accept
Accept-Language           SIP header Accept-Language
Accept-Resource-Priority  SIP header Accept-Resource-Priority
Alert-Info                SIP header Alert-Info
Allow-Events              SIP header Allow-Events
Allow-Header              SIP header Allow
Also                      SIP header Also
Authorization             SIP header Authorization
CC-Diversion              SIP header CC-Diversion
CC-Redirect               SIP header CC-Redirect
CSeq                      SIP header CSeq
Call-ID                   SIP header Call-ID
Call-Info                 SIP header Call-Info
Cisco-Gcid                SIP header Cisco-Gcid
Cisco-Guid                SIP header Cisco-Guid
Contact                   SIP header contact
Content-Disposition       SIP header Content-Disposition
Content-Encoding          SIP header Content-Encoding
Content-Id                SIP header Content-Id
Content-Length            SIP header Content-Length
Content-Type              SIP header Content-Type
Date                      SIP header Date
Diversion                 SIP header Diversion
Event                     SIP header Event
Expires                   SIP header Expires
From                      SIP header FROM
History-Info              SIP header History-Info
Location                  SIP header Location
MIME-Version              SIP header MIME-Version
Max-Forwards              SIP header Max-Forwards
Min-Expires               SIP header Min-Expires
Min-SE                    SIP header Min-SE
Orig-dial-plan            SIP header Orig-dial-plan
P-Asserted-Identity       SIP header P-Asserted-Identity
P-Preferred-Identity      SIP header P-Preferred-Identity
P-RTP-Stat                SIP Header P-RTP-Stat
Privacy                   SIP header Privacy
Proxy-Authenticate        SIP header Proxy-Authenticate
Proxy-Authorization       SIP header Proxy-Authorization
Proxy-Require             SIP header Proxy-Require
Rack                      SIP header Rack
Reason                    SIP header Reason
Record-Route              SIP header Record-Route
Refer-To                  SIP header Refer-To
Referred-By               SIP header Referred-By
Reject-Contact            SIP header Reject-Contact
```

```
Remote-Party-ID           SIP header Remote-Party-ID
Replaces                  SIP header Replaces
Request-Disposition       SIP header Request-Disposition
Requested-By              SIP header Requested-By
Require                   SIP header Require
Resource-Priority         SIP header Resource-Priority
Retry-After               SIP header Retry-After
Route                     SIP header Route
Rseq                      SIP header Rseq
SIP-ETag                  SIP header SIP-ETag
SIP-If-Match              SIP header SIP-If-Match
SIP-Req-URI               SIP Request URI
SIP-StatusLine            SIP Status-Line
Server                    SIP header Server
Session-Expires           SIP header Session-Expires
Session-Header            SIP header Session
Session-ID                SIP header Session ID
Subscription-State        SIP header Subscription-State
Supported                 SIP header Supported
Term-dial-plan            SIP header Term-dial-plan
Timestamp                 SIP header Timestamp
To                        SIP header TO
Unsupported               SIP header Unsupported
User-Agent                SIP header User-Agent
Via                       SIP header Via
WORD                      Any other SIP header name
WWW-Authenticate          SIP header WWW-Authenticate
Warning                   SIP header Warning
```

SDP Headers

Supported SDP headers are the following:

```
Attribute               SDP header Attribute
Audio-Attribute         SDP Audio Attribute
Audio-Bandwidth-Info    SDP Audio Bandwidth Info
Audio-Connection-Info   SDP Audio Connection Info
Audio-Encryption-Key    SDP Audio Encript key
Audio-Media             SDP Audio Media
Audio-Session-Info      SDP Audio Session Info
Bandwidth-Key           SDP header Bandwidth-Key
Connection-Info         SDP header Connection-Info
Email-Address           SDP header Email-Address
Encrypt-Key             SDP header Encrypt-Key
Phone-Number            SDP header Phone-Number
Repeat-Times            SDP header Repeat-Times
Session-Info            SDP header Session-info
Session-Name            SDP header Session-Name
Session-Owner           SDP header Session
Time-Adjust-Key         SDP header Time-Adjust-Key
Time-Header             SDP header Time
Url-Descriptor          SDP header Url-Descriptor
Version                 SDP Version
Video-Attribute         SDP Video Attribute
Video-Bandwidth-Info    SDP Video Bandwidth Info
Video-Connection-Info   SDP Video Connection Info
Video-Encryption-Key    SDP Video Encryption Key
Video-Media             SDP Video Media
Video-Session-Info      SDP Video Session Info
mline-index             M-Line index for SDP Line
```

Verify SIP Profiles

SUMMARY STEPS

1. **show dial-peer voice** *id* **| include profile**

DETAILED STEPS

show dial-peer voice *id* **| include profile**

Displays information related to SIP profiles configured on the specified dial peer.

Example:

```
Device# show dial-peer voice 10 | include sip profile
        voice class sip profiles = 11
        voice class sip profiles inbound = 10
```

Troubleshoot SIP Profiles

SUMMARY STEPS

1. **debug ccsip all**

DETAILED STEPS

The following debugs can also be used:

- **debug ccsip info**
- **debug ccsip feature sip-profiles**
- **debug ccsip error**

debug ccsip all

This command displays the applied SIP profiles.

Example:

Applied SIP profile is highlighted in the example below.

```
Device# debug ccsip all
…
Oct 12 06:51:53.619: //-1/735085DC8F3D/SIP/Info/sipSPIGetShrlPeer:
                     Try match incoming dialpeer for Calling number:
                     : sippOct 12 06:51:53.619:
                     //-1/735085DC8F3D/SIP/Info/sipSPIGetCallConfig:
                     Peer tag 2 matched for incoming call
```

```
Oct 12 06:51:53.619: //-1/xxxxxxxxxxxx/SIP/Info/sipSPIGetCallConfig:
                     voice class SIP profiles tag is set : 1
Oct 12 06:51:53.619: //-1/735085DC8F3D/SIP/Info/sipSPIGetCallConfig:
                     Not using Voice Class Codec
Oct 12 06:51:53.619: //-1/735085DC8F3D/SIP/Info/sipSPIGetCallConfig:
                     xcoder high-density disabled
Oct 12 06:51:53.619: //-1/735085DC8F3D/SIP/Info/sipSPIGetCallConfig:
                     Flow Mode set to FLOW_THROUGH
```

This command also displays the modifications that are performed by the SIP profile configuration, by preceding the modification information with the word sip_profiles, as highlighted in the following example.

Example:

```
Device# debug ccsip all
…
Oct 12 06:51:53.647: //-1/xxxxxxxxxxxx/SIP/Info/
sip_profiles_application_change_sdp_line:
New SDP header is added : b=AS: 1600
Oct 12 06:51:53.647: //-1/xxxxxxxxxxxx/SIP/Info/
sip_profiles_update_content_length:
Content length header before modification :
Content-Length: 290
Oct 12 06:51:53.647: //-1/xxxxxxxxxxxx/SIP/Info/
sip_profiles_update_content_length:
Content length header after modification :
Content-Length: 279
```

Examples: Adding, Modifying, Removing SIP Profiles

Example: Adding a SIP, SDP, or Peer Header

Example: Adding "b=AS:4000" SDP header to the video-media Header of the INVITE SDP Request Messages

```
Device(config)# voice class sip-profiles 10
Device(config-class)# request INVITE sdp-header Video-Bandwidth-Info add "b=AS:4000"
Device(config-class)# end
```

Example: Adding "b=AS:4000" SDP header to the video-media Header of the INVITE SDP Request Messages in rule format

```
Device(config)# voice class sip-profiles 10
Device(config-class)# rule 1 request INVITE sdp-header Video-Bandwidth-Info add "b=AS:4000"
Device(config-class)# end
```

Example: Adding the Retry-After Header to the SIP 480 Response Messages

```
Device(config)# voice class sip-profiles 20
```

```
Device(config-class)# response 480 sip-header Retry-After add “Retry-After: 60”
Device(config-class)# end
```

Example: Adding the Retry-After Header to the SIP 480 Response Messages in rule format

```
Device(config)# voice class sip-profiles 20
Device(config-class)# rule 1 response 480 sip-header Retry-After add “Retry-After: 60”
Device(config-class)# end
```

Example: Adding "User-Agent: SIP-GW-UA" to the User-Agent Field of the 200 Response SIP Messages

```
Device(config)# voice class sip-profiles 40
Device(config-class)# response 200 sip-header User-Agent add "User-Agent: SIP-GW-UA"
Device(config-class)# end
```

Example: Adding "User-Agent: SIP-GW-UA" to the User-Agent Field of the 200 Response SIP Messages in rule format

```
Device(config)# voice class sip-profiles 40
Device(config-class)# rule 1 response 200 sip-header User-Agent add "User-Agent: SIP-GW-UA"
Device(config-class)# end
```

Example: Adding "a=ixmap:0 ping" in M-Line number 4 of the INVITE SDP Request Messages

```
Device(config)# voice class sip-profiles 10
Device(config-class)#  request INVITE sdp-header mline-index 4 a=ixmap add "a=ixmap:0 ping"
Device(config-class)# end
```

Example: Modifying a SIP, SDP, or Peer Header

Example: Modifying SIP-Req-URI of the Header of the INVITE and RE-INVITE SIP Request Messages to include "user=phone"

```
Device(config)# voice class sip-profiles 30
Device(config-class)# request INVITE sip-header SIP-Req-URI modify "; SIP/2.0" ";user=phone
 SIP/2.0"
Device(config-class)# request RE-INVITE sip-header SIP-Req-URI modify "; SIP/2.0" ";user=phone
 SIP/2.0"
Device(config-class)# end
```

Example: Modifying SIP-Req-URI of the Header of the INVITE and RE-INVITE SIP Request Messages to include "user=phone" in rule format

```
Device(config)# voice class sip-profiles 30
```

```
Device(config-class)# rule 1 request INVITE sip-header SIP-Req-URI modify "; SIP/2.0"
";user=phone SIP/2.0"
Device(config-class)# rule 2 request RE-INVITE sip-header SIP-Req-URI modify "; SIP/2.0"
";user=phone SIP/2.0"
Device(config-class)# end
```

Modify the From Field of a SIP INVITE Request Messages to “gateway@gw-ip-address” Format

For example, modify 2222000020@10.13.24.7 to gateway@10.13.24.7

```
Device(config)# voice class sip-profiles 20
Device(config-class)# request INVITE sip-header From modify "(<.*:)(.*@)" "\1gateway@"
```

Modify the From Field of a SIP INVITE Request Messages to “gateway@gw-ip-address” Format in rule format

For example, modify 2222000020@10.13.24.7 to gateway@10.13.24.7

```
Device(config)# voice class sip-profiles 20
Device(config-class)# rule 1 request INVITE sip-header From modify "(<.*:)(.*@)" "\1gateway@"
```

Replace "CiscoSystems-SIP-GW-UserAgent" with "-" in the Originator Header of the SDP in INVITE Request Messages

```
Device(config)# voice class sip-profiles 10
Device(config-class)# request INVITE sdp-header Session-Owner modify
"CiscoSystems-SIP-GW-UserAgent" "-"
```

Replace "CiscoSystems-SIP-GW-UserAgent" with "-" in the Originator Header of the SDP in INVITE Request Messages in rule format

```
Device(config)# voice class sip-profiles 10
Device(config-class)# rule 1 request INVITE sdp-header Session-Owner modify
"CiscoSystems-SIP-GW-UserAgent" "-"
```

Convert "sip uri" to "tel uri" in Req-URI, From and To Headers of SIP INVITE Request Messages

For example, modify sip:2222000020@9.13.24.6:5060” to “tel:2222000020

```
Device(config)# voice class sip-profiles 40
Device(config-class)# request INVITE sip-header SIP-Req-URI modify "sip:(.*)@[^ ]+" "tel:\1"
Device(config-class)# request INVITE sip-header From modify "<sip:(.*)@.*>" "<tel:\1>"
Device(config-class)# request INVITE sip-header To modify "<sip:(.*)@.*>" "<tel:\1>"
```

Convert "sip uri" to "tel uri" in Req-URI, From and To Headers of SIP INVITE Request Messagesin rule format

For example, modify sip:2222000020@9.13.24.6:5060” to “tel:2222000020

```
Device(config)# voice class sip-profiles 40
Device(config-class)# rule 1 request INVITE sip-header SIP-Req-URI modify "sip:(.*)@[^ ]+"
 "tel:\1"
```

```
Device(config-class)# rule 2 request INVITE sip-header From modify "<sip:(.*)@.*>" "<tel:\1>"
Device(config-class)# rule 3 request INVITE sip-header To modify "<sip:(.*)@.*>" "<tel:\1>"
```

Example: Change the Audio Attribute Ptime:20 to Ptime:30

Inbound ptime:

```
a=ptime:20
```

Outbound ptime:

```
a=ptime:30
```

```
Device(config)# voice class sip-profiles 103
Device(config-class)# request ANY sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"
```

Example: Modify Audio direction "Audio-Attribute"

Some service providers or customer equipment reply to delay offer invites and or re-invites that contain a=inactive with a=inactive, a=recvonly, or a=sendonly. This can create an issue when trying to transfer or retrieve a call from hold. The result is normally one-way audio after hold or resume or transfer or moh is not heard. To resolve this issue changing the audio attribute to Sendrecv prevents the provider from replaying back with a=inactive, a=recvonly, or a=sendonly.

Case 1:

```
Inbound Audio-Attribute

a=inactive

Outbound Audio-Attribute

a=sendrecv
```

Case 2:

```
Inbound Audio-Attribute

a=recvonly

Outbound Audio-Attribute

a=sendrecv
```

Case 3

```
Inbound Audio-Attribute

a=sendonly

Outbound Audio-Attribute

a=sendrecv
```

```
Device(config)# voice class sip-profiles 104
Device(config-class)# request any sdp-header Audio-Attribute modify "a=inactive" "a=sendrecv"
Device(config-class)# request any sdp-header Audio-Attribute modify "a=recvonly" "a=sendrecv"
Device(config-class)# request any sdp-header Audio-Attribute modify "a=sendonly" "a=sendrecv"

Device(config-class)# response any sdp-header Audio-Attribute modify "a=inactive" "a=sendrecv"
```

```
Device(config-class)# response any sdp-header Audio-Attribute modify "a=recvonly" "a=sendrecv"
Device(config-class)# response any sdp-header Audio-Attribute modify "a=sendonly" "a=sendrecv"
```

Example: Modifying Packetization Mode in a=fmtp line of M-line number 2 of the INVITE SDP Request Messages

```
Device(config)# voice class sip-profiles 10
Device(config-class)# request INVITE sdp-header mline-index 2 a=fmtp modify
"packetization-mode=1" "packetization-mode=0"
Device(config-class)# end
```

Example: Remove a SIP, SDP, or Peer Header

Remove Cisco-Guid SIP header from all Requests and Responses

```
Device(config)# voice class sip-profiles 20
Device(config-class)# request ANY sip-header Cisco-Guid remove
Device(config-class)# response ANY sip-header Cisco-Guid remove
Device(config-class)# end
```

Remove Server Header from 100 and 180 SIP Response Messages

```
Device(config)# voice class sip-profiles 20
Device(config-class)# response 100 sip-header Server remove
Device(config-class)# response 180 sip-header Server remove
Device(config-class)# end
```

Removing a SIP Profile rule in rule format configuration

SIP Profile configuration in rule format

```
Device(config)# voice class sip-profiles 10
Device(config-class)# rule 1 request any sdp-header Audio-Attribute modify "a=inactive"
"a=sendrecv"
Device(config-class)# rule 2 request any sdp-header Audio-Attribute modify "a=recvonly"
"a=sendrecv"
Device(config-class)# end
```

Removing the rule using rule tag

```
Device(config)# voice class sip-profiles 10
Device(config-class)# no rule 1
Device(config-class)# end
```

Once the rule is removed, the tag belonging to the removed rule remains vacant. The tags associated with the subsequent rules are unchanged.

The SIP Profile configuration after removing the rule

```
Device(config)# voice class sip-profiles 10
Device(config-class)# rule 2 request any sdp-header Audio-Attribute modify "a=recvonly"
"a=sendrecv"
Device(config-class)# end
```

Example: Removing "a=ixmap" in M-Line number 4 of the INVITE SDP Request Messages

```
Device(config)# voice class sip-profiles 10
Device(config-class)# request INVITE sdp-header mline-index 4 a=ixmap REMOVE
Device(config-class)# end
```

Example: Inserting SIP Profile Rules

Example: Inserting a SIP Profile Rule

Inserting a SIP profile rule to a SIP Profile

```
Device(config)#voice class sip-profiles 1
 Device(config-class)#rule 1 request INVITE sip-header Contact Modify "(.*)" "\1;temp=xyz"
 Device(config-class)#rule 2 request INVITE sip-header Supported Add "Supported: "
 Device(config-class)#rule before 2 request INVITE sip-header To Modify "(.*)" "\1;temp=abc"
```

The SIP Profile after inserting the new rule

```
Device(config)#voice class sip-profiles 1
 Device(config-class)#rule 1 request INVITE sip-header Contact Modify "(.*)" "\1;temp=xyz"
 Device(config-class)#rule 2 request INVITE sip-header To Modify "(.*)" "\1;temp=abc"
 Device(config-class)#rule 3 request INVITE sip-header Supported Add "Supported: "
```

Example: Upgrading and Downgrading SIP Profiles automatically

Upgrading SIP Profiles to rule-format

The following is a snippet from **show running-config** command showing the SIP profiles in non-rule format:

```
Device#show running-config | section profiles 1
voice class sip-profiles 1
 request INVITE sip-header Contact Modify "(.*)" "\1;temp=xyz"
 request INVITE sip-header Supported Add "Supported: "
```

Execute the following command in EXEC (#) mode to upgrade the SIP Profiles to rule-format:

```
Device#voice sip sip-profiles upgrade
```

The following is a snippet from **show running-config** command showing the SIP profiles after upgrading to rule-format:

```
Device#show running-config | section profiles 1

voice class sip-profiles 1
 rule 1 request INVITE sip-header Contact Modify "(.*)" "\1;temp=xyz"
 rule 2 request INVITE sip-header Supported Add "Supported: "
```

Downgrading SIP Profiles to non-rule format

The following is a snippet from **show running-config** command showing SIP profiles in rule-format:

```
Device#show running-config | section profiles 1

voice class sip-profiles 1
 rule 1 request INVITE sip-header Contact Modify "(.*)" "\1;temp=xyz"
 rule 2 request INVITE sip-header Supported Add "Supported: "
```

Execute the following command in EXEC(#) mode to downgrade SIP Profiles to non-rule format:

```
Device# voice sip sip-profiles downgrade
```

The following is a snippet from **show running-config** command showing SIP profiles after downgrading to non-rule format:

```
Device#show running-config | section profiles 1

voice class sip-profiles 1
 request INVITE sip-header Contact Modify "(.*)" "\1;temp=xyz"
 request INVITE sip-header Supported Add "Supported: "
```

Example: Modifying Diversion Headers

Example: Modify Diversion Headers from Three-Digit Extensions to Ten Digits.

Most North American service providers require a ten digit diversion header. Prior to Call manager 8.6, Call manager would only send the extension in the diversion header. A SIP profile can be used to make the diversion header ten digits.

Call manager version 8.6 and above has the field "Redirecting Party Transformation CSS" which lets you expand the diversion header on the call manager.

The SIP profile will look for a diversion header containing "<sip:5..." , where ... stands for the three-digit extension and then concatenates 9789365 with these three digits.

Original Diversion Header:

```
Diversion:<sip:5100@161.44.77.193>;privacy=off;reason=unconditional;counter=1;screen=no
```

Modified Diversion Header:

```
Diversion: <sip:9789365100@10.86.176.19>;privacy=off;reason=unconditional;counter=1;screen=no
```

```
Device(config)# voice class sip-profiles 101
Device(config-class)# request Invite sip-header Diversion modify "<sip:5(...)@"
"<sip:9789365\1@"
Device(config-class)# end
```

Example: Create a Diversion header depending on the area code in the From field

Most service providers require a redirected call to have a diversion header that contains a full 10 digit number that is associated with a SIP trunk group. Sometimes, a SIP trunk may cover several different area codes, states, and geographic locations. In this scenario, the service provider may require a specific number to be placed in the diversion header depending on the calling party number.

In the below example, if the From field has an area code of 978 "<sip:978", the SIP profile leaves the From field as is and adds a diversion header.

```
Device(config)# voice class sip-profiles 102
Device(config-class)# request INVITE sip-header From modify "From:(.*)<sip:978(.*)@(.*)"
"From:\1<sip:978\2@\3\x0ADiversion:
<sip:9789365000@10.86.176.19:5060;privacy=off;reason=unconditional;counter=1;screen=no"
```

The below diversion header is added. There was no diversion header before this was added:

```
Diversion: <sip:9789365000@10.86.176.19:5060;transport=udp>"
```

Example: Sample SIP Profile Application on SIP Invite Message

The SIP profile configured is below:

The SIP INVITE message before the SIP profile has been applied is show below:

```
voice class sip-profiles 1
  request INVITE sdp-header Audio-Bandwidth-Info add "b=AS:1600‟
  request ANY sip-header Cisco-Guid remove
  request INVITE sdp-header Session-Owner modify "CiscoSystems-SIP-GW-UserAgent" "-‟
```

```
INVITE sip:2222000020@9.13.40.250:5060 SIP/2.0
Via: SIP/2.0/UDP 9.13.40.249:5060;branch=z9hG4bK1A203F
From: "sipp " <sip:1111000010@9.13.40.249>;tag=F11AE0-1D8D
To: <sip:2222000020@9.13.40.250>
Date: Mon, 29 Oct 2007 19:02:04 GMT
Call-ID: 4561B116-858811DC-804DEF2E-4CF2D71B@9.13.40.249
Cisco-Guid: 1163870326-2240287196-2152197934-1290983195
Content-Length: 290

v=0
o=CiscoSystemsSIP-GW-UserAgent 6906 8069 IN IP4 9.13.40.249
s=SIP Call
c=IN IP4 9.13.40.249
t=0 0
m=audio 17070 RTP/AVP 0
c=IN IP4 9.13.40.249
a=rtpmap:0 PCMU/8000
a=ptime:20
```

The SIP INVITE message after the SIP profile has been applied is shown below:

- The Cisco-Guid has been removed.
- CiscoSystemsSIP-GW-UserAgent has been replaced with -.
- The Audio-Bandwidth SDP header has been added with the value b=AS:1600.

```
INVITE sip:2222000020@9.13.40.250:5060 SIP/2.0
Via: SIP/2.0/UDP 9.13.40.249:5060;branch=z9hG4bK1A203F
```

```
From: "sipp " <sip:1111000010@9.13.40.249>;tag=F11AE0-1D8D
To: <sip:2222000020@9.13.40.250>
Date: Mon, 29 Oct 2007 19:02:04 GMT
Call-ID: 4561B116-858811DC-804DEF2E-4CF2D71B@9.13.40.249
Content-Length: 279

v=0
o=- 6906 8069 IN IP4 9.13.40.249
s=SIP Call
c=IN IP4 9.13.40.249
t=0 0
m=audio 17070 RTP/AVP 0
c=IN IP4 9.13.40.249
a=rtpmap:0 PCMU/8000
a=ptime:20
b=AS:1600
```

Example: Sample SIP Profile for Non-Standard SIP Headers

It is possible to add, copy, modify or delete any SIP header.

```
voice class sip-profiles 1
 request INVITE sip-header X-Cisco-Recording-Participant copy "sip:(.*)@" u01
 request INVITE sip-header X-Cisco-Recording-Participant modify "sip:sipp@" "sip:1000@"
 request INVITE sip-header My-Info add "My-Info: MF Call"
 request INVITE sip-header My-Info remove
```

Example: Copy User-to-User Information from REFER Message

When a call is transferred using REFER, user-to-user content from the originating message is not automatically copied to the triggered INVITE. This example illustrates how SIP profiles can be used to capture this information and pass it to the outbound INVITE, where it is added as a new header.

SIP profile 1210 applied to the incoming dial-peer copies the user-to-user information to a temporary header (x-user). This header is passed to the outbound leg where SIP profile 1211 extracts this information and uses it to create the new INVITE User-to-user header before removing the temporary x-user header.

To ensure that the x-user header is passed to the INVITE message, either use a sip-copylist that is applied to both dial-peers or enable unsupported header pass-through.

```
voice class sip-profiles 1210
request REFER sip-header Refer-To copy "Refer-To:.*User-to-User=(.*)>" u03
request REFER sip-header x-user add "x-user: TEST
request REFER sip-header x-user modify "x-user: (.*)" "x-user: \u03"

voice class sip-profiles 1211
request INVITE sip-header x-user copy "x-user: (.*)" u05
request INVITE sip-header User-to-User add "User-to-User: TEST"
request INVITE sip-header User-to-User modify "User-to-User: (.*)" "User-to-User: \u05"
request INVITE sip-header x-user remove"
```