Cisco IOS SIP Configuration Guide

Dialpeer Configuration
Terminology

• Call - A connection terminating on or passing through a gateway.

• Call Leg - The segment of a call associated with a particular signaling and transport technology, for example SIP or PSTN

• Service Provider - the implementation of the Interface for a particular protocol (signaling stack)

• Interface (voice-port) - A physical or logical connector that carries call legs. For example, an analog line or a T1/PRI span. The IP network is also modeled as an interface.

• Application (a.k.a. Session application) - accepts and creates call-legs, provides feature platform.
Dial Peer

- A dial-peer is the entity to which a call is connected. Includes VoIP, Pots etc.
- Incoming dial-peers point to an application to handle an incoming call.
- Outgoing dial-peers pick an interface, PSTN or SIP, to handle an outgoing call.
VoIP DialPeer

- Map phone numbers (E.164 addresses) or SIP URIs to IP addresses or DNS names
- Describe transport characteristics of the connection like: codec, vad, QoS, dtmf-relay type etc.
- Example:
  ```
  dial-peer voice 111 voip
destination-pattern 60154
incoming called number 1001
session protocol sipv2
session target dns:sipserver1.hawaii.edu
dtmf-relay rtp-npe
codec g711ulaw
  ```
URI Matching

• From 12.3(4)T onwards, a voip dialpeer can be matched based on a sip: uri

• A voice class uri needs to be configured:

  voice class uri SIP_1 sip

  user abc

  host sip.com
URI Matching contd...

- On the dialpeer, the voice class needs to be associated with from, to or request uri.

dial-peer voice 111 voip
  destination-pattern 60154
  incoming called number 1001
  incoming uri from SIP_1
  session protocol sipv2
  session target dns:sipserver1.hawaii.edu
  ....
VoIP Dialpeer Matching Rule

• Inbound dialpeer
  incoming uri request
  incoming uri to
  incoming uri from
  incoming called-number
  answer address
  destination-pattern

• Outbound dialpeer
  destination-uri
  destination-pattern
POTS Dialpeer

- Map phone numbers to voice ports.
- Destination-pattern is used to match an outbound dialpeer, incoming called-number is used to match an inbound dialpeer.
- Example:
  dial-peer voice 100 pots
  destination-pattern 9000
  port 1/0/0
- Voice ports further specify signaling properties
Order of Dialpeer matching

• All matched dialpeer are sorted based on preference. Higher preference is given to dialpeers with an exact pattern match.

• Two dialpeers with the same pattern match will be tried in the order they were configured.

• preference command can be used to break the tie between two dialpeers with same match characteristics.
Voice Translation Profiles introduce a scheme to translate numbers.

The translation rules replace a sub string of the input number if the number matches the match pattern, number plan, and type present in the rule.

Called, Calling and Redirect-Called numbers can be defined in a translation profile. Each type of call number in the profile can have different translation rules.

Translation profiles can be referenced on: Trunk Group, Source IP Group, Dial-Peer, Voice-Port, VoIP Incoming

The voice translation rules use characters similar to Regular Expression Syntax (regexp)
Configuring Translation Rule

- Syntax:
  Router(config)# voice translation-rule <num>
  Router(cfg-translation-rule)# rule precedence /match-pattern/ /replace-pattern/ [type {match-type replace-type} [plan {match-type replace-type}]]

- Examples:
  1. This example replaces any occurrence of the number "123" with "456".
     voice translation-rule 1
     rule 1 /123/ /456/
  2. Match 1# at the beginning and replace it with Null.
     voice translation-rule 2
     rule 2 /^1#/ //
  3. Expand 5 digit number to 10 digits
     voice translation-rule 3
     rule 3 /25555/ /91939&/
Configuring Translation Profile

- Once a translation rule has been configured, translation profile can be configured by:
  - voice translation-profile <name>
  - translate called <translation-rule num>
  - translate calling <translation-rule num>
  - translate redirect-called <translation-rule num>
- Dial-Peer configuration:
  - dial-peer voice <num> [pots|voip]
  - translation-profile [incoming | outgoing] <name>
- For more information on number translation:
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SIP Feature Configuration
Reliable Provisional Response

• Gateway can be configured to send 18x response reliably as in RFC 3262.
• Global configuration is under voice-service voip; sip. It can also be configured on the voip dialpeer. Dialpeer configuration will take precedence over global configuration.
• To configure it:
  router# voice-service voip
  router(conf-voi-serv)#sip
  router(conf-serv-sip)# rel1xx [require|supported] 100rel
• Default mode is rel1xx supported 100rel
Codec configuration

- Codec can be configured on the voip dialpeer using codec <codec> cli.

  Example:
  
  router# conf t
  router(config)#dial-peer voice 6 voip
  router(config-dial-peer)#codec g711ulaw

- Codecs configured on the outbound dialpeer will be sent in sdp of INVITE. Default codec is G729
Codec Configuration contd..

- More than one codec can be configured using voice-class codec.

Example:

```
router# conf t
router(config)#voice class codec <num>
router(config-class)#codec preference 1 g711alaw
router(config-class)#codec preference 2 g711ulaw
On the dialpeer:
router(config)#dial-peer voice 6 voip
router(config)# voice-class codec <num>
```
Configuration under sip-ua

- Configurations specific to sip user agent are under sip-ua. Commonly used configs are message retry count, retry interval configs, configuring an outbound server.

- Configuring number of retries.
  
  router(config)# sip-ua
  
  router(config-sip-ua)# retry <message> <number>

- Signaling timer configuration.
  
  router(config)# sip-ua
  
  router(config-sip-ua)# timers <message> <timer-val>
• Configuring an outbound server

  router(config)# sip-ua
  router(config-sip-ua)# sip-server <server address>

  On the outbound voip dialpeer:
  router(config)# dial-peer voice 6 voip
  router(config)# session-target sip-server
sip-ua Configuration contd …

- Overriding default SIP-PSTN disconnect cause code
  
  `router(config)# sip-ua`
  
  `router(config)# set pstn-cause <num> sip-status <num>`
  
  `router(config)# set sip-status <num> pstn-status <num>`

  Range of sip-status is 400-699
  Range of pstn-status is 1-127
Caller identity and Privacy

- IOS SIP gateway uses Remote-Party-ID header that identifies the calling party and carries presentation and screening information.

- Implementation is based on draft-ietf-privacy-.02.txt, *SIP Extensions for Caller Identity and Privacy*.

- For PSTN-SIP call, information from octet3a is used to create presentation and screening parameters in Remote-Party-ID header.

- For SIP-PSTN, presentation and screening parameters in Remote-Party-ID header is used to create octet3a information in ISDN SETUP.
Caller Identity and Privacy contd..

- Additional CLI commands allow alternative calling information treatments for calls entering the SIP trunking gateway. Configurable treatment options for SIP-PSTN:
  - Calling name and number pass-through (default).
  - No calling name or number sent in the forwarded Setup message.
  - Calling name unconditionally set to the configured string in the forwarded Setup message.
  - Calling number unconditionally set to the configured string in the forwarded Setup message.
• Configurable treatment options for PSTN-SIP:
  • Calling name and number pass-through (default).
  • No calling name or number sent in the forwarded INVITE message.
  • Display-name of the From header unconditionally set to the configured
    string in the forwarded INVITE message.
  • User part of the From header unconditionally set to the configured string
    in the forwarded INVITE message.
  • Display-name of the Remote-Party-ID header unconditionally set to the
    configured string in the forwarded INVITE message.
  • User part of the Remote-Party-ID header unconditionally set to the
    configured string in the forwarded INVITE message.
• P-Asserted-Identity support will be available in a future release.
Addition SIP gateway features

- Call Transfer
- T.38 fax with fallback to fax-passthrough
- Buffered Calling-Name
- Registration
- Digest Authentication
- Call Redirection
- Ability to configure source address for signaling and media