Cisco IOS SIP Configuration Guide

Dialpeer Configuration

Terminology

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- **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-nte
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…

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- **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

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- **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.**

Number Translation using Translation Profile

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- 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:

  http://www.cisco.com/en/US/tech/tk652/tk90/technologies_configuration_example09186a00803f818a.shtml

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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

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- 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 ..

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- **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>

sip-ua configurations contd ..

- **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 …

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- **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.**

Caller Identity and Privacy contd…

- **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