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-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…

- ## 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.**

Number Translation using Translation Profile

- 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#/ //**

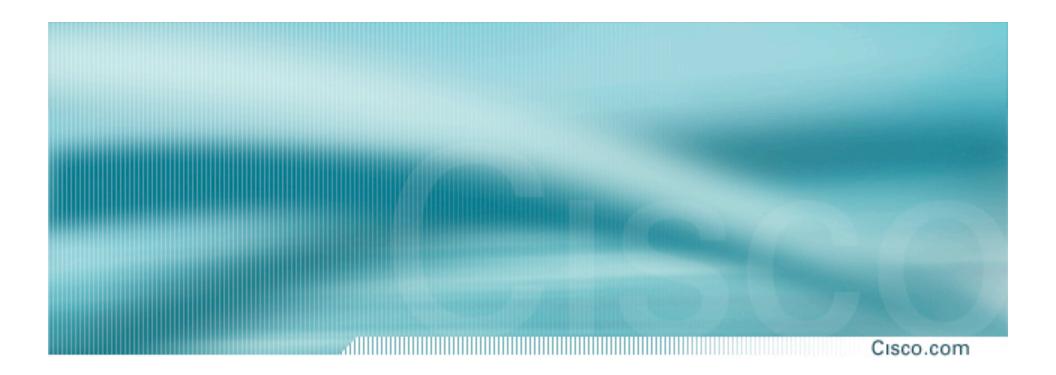**3. Expand 5 digit number to 10 digits**

> **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_example091
    86a00803f818a.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

- 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>

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 ...

- **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**