SIP Auto Peer

draft-kinamdar-dispatch-sip-auto-peer-01

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Agenda

• Problem Space
• Solution Overview
• Asks
Bringing up service provider facing SIP trunks is a long process that could take several hours... sometimes days...

- Administrators usually uncover a large number of problems when the trunk is setup.
- Requires time-consuming interaction between the administrator and service provider to ensure everything is working.
- Deployment times increase significantly due to interoperability problems.
Magnitude of Problem

• A total of 6000 support cases opened with Cisco last year for its Enterprise SBC, CUBE

• 22% of these cases were directly related to ITSP interoperability

• Multiply this by the number of enterprise SBC vendors to get a rough estimate of the magnitude of the problem

• Still a significant number of enterprise networks are yet to migrate from TDM/Analog to SIP trunking ... the problem isn’t going away anytime soon!
Current Solutions

- SIP Connect Technical Recommendations
- Service Provider Documentation
Issues Still Persist...

- What happens if the ITSP forks calls? The SBC has to be configured to either pass through all responses or pass through only the first confirmed dialog.
- What happens for hairpinned calls? Does the ITSP expect the enterprise to send the first RTP packet? Can the enterprise network “hand-off” the call?
- How many re-INVITES with or without SDP change can be sent to the ITSP?
- Is the called party number in the Request-URI? To header field or the P-Called-Party ID?
- How is the user portion of the From header field formatted? Is it any number or specifically a number within the DID range allocated to the enterprise?
- Will the ITSP reject calls that have a video media line (along with an audio media line) or follow the offer/answer model and simply set the video media line to 0?
SIP Auto Peer

• SIP Auto Peer is a framework that converts answers to questions typically needed by an administrator to setup a SIP trunk into a machine workable format.

• Parameters are encoded according to the YANG model defined in the draft and communicated to the enterprise edge element.

• Parameters may be subsequently parsed by automaton on the edge element and converted to configuration.

• Parameters are a combination of both basic (e.g. registrar, call control IP) and nuanced (e.g. who sends the first RTP packet for a hairpinned call) parameters.

• Only some parameters are included in Version 00 and 01 of this draft...we hope to work with the larger community to define a robust/complete parameter set.
High-Level Overview

1. Capability Server Discovered or Manually Configured
2. HTTPS GET (Capability Set Request)
3. HTTPS Response (Capability Set)
4. Parse Capability Set & Configure Trunk

*** Body encoded in XML or JSON
Why HTTPS?

- HTTPS is the path of least disruption – no changes required to SIP stacks.
- Exchange of SIP traffic between enterprise and service provider networks usually requires SIP trunk registration.
- Exchange of the capability set after trunk registration is - well - pointless.
- Most SBCs also have an HTTP stack built-in.
XML or JSON?

XML

Pros:
- XML and configuration files are synonymous.
- SBCs usually have XML parsers built-in
- Strong schema constraint languages for XML

Cons:
- Bulky, verbose
- Requires translation of doc to structure

JSON

Pros:
- Compact and more readable
- Data formatted as a map

Cons:
- Can have issues with weakly structured data
Yang Model

- Parameters applicable to any vendor SBC for SIP trunk registration
- Significant media parameters
- Security specifications
- SIP extensions supported by SP
Example Capability Set

<peering-info xmlns="urn:ietf:params:xml:ns:yang:ietf-peering"
  xmlns:xxi="http://www.w3.org/2001/XMLSchema-instance"
  <variant>1.0</variant>
  <transport-info>
    <transport>TCP;TLS;UDP</transport>
    <registrar>registrar1.voip.example.com:5060</registrar>
    <registrar>registrar2.voip.example.com:5060</registrar>
    <callControl>callserver1.voip.example.com:5065</callControl>
    <dnsserver>8.8.8.8</dnsserver>
    <outboundProxy>0.0.0.0</outboundProxy>
  </transport-info>
  <call-specs>
    <earlyMedia>true</earlyMedia>
    <signalingWorking>false</signalingWorking>
    <supportedMethods>INVITE;OPTIONS;BYE;CANCEL;ACK;PRACK;SUBSCRIBE;NOTIFY</supportedMethods>
  </call-specs>
  <media>
    <media type="audio">
      <media-format>FMU;rate=8000;ptime=20</media-format>
      <media-format>G729;rate=8000;annexb</media-format>
      <media-format>G722;rate=8000;bitrate=56k,64k</media-format>
    </media>
    <fax>
      <protocol>pass-through</protocol>
      <protocol>138</protocol>
    </fax>
    <rtcp>
      <RTPtrigger>true</RTPtrigger>
      <rtcpFeedback>true</rtcpFeedback>
    </rtcp>
    <symmetricRTP>true</symmetricRTP>
    <tcp>
      <tcpFeedback>true</tcpFeedback>
    </tcp>
  </media>
</peering-info>

<dtmf>
  <payloadNumber>101</payloadNumber>
</dtmf>

<security>
  <signaling>
    <type>TLS</type>
  </signaling>
  <mediaSecurity>
    <keyManagement>SDES;DTLS-SRTP,version=1.2</keyManagement>
  </mediaSecurity>
</security>

<extensions>
  <timer>rel00:gin;path</timer>
</extensions>
The way forward...

• Objections / Challenges?

• Refining the capability set. Additional parameters?
  o URI Scheme of calls – TEL or SIP URIs?
  o Full or Compact form headers?
  o DID block allocated by ITSP?
  o Emergency Numbers?

• How should the draft be taken forward?
  • Mini Workgroup
  • SIP Core
  • AD Sponsored
Thank You