## **ASAP**

### **Automatic SIP trunking And Peering**

**IETF 110** 

Monday, March 8, 2021

08:00 - 10:00 US Pacific Time

Session III, Room 1

Jean Mahoney Gonzalo Salgueiro

## **Note Well**

This is a reminder of IETF policies in effect on various topics such as patents or code of conduct. It is only meant to point you in the right direction. Exceptions may apply. The IETF's patent policy and the definition of an IETF "contribution" and "participation" are set forth in BCP 79; please read it carefully. As a reminder:

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Definitive information is in the documents listed below and other IETF BCPs. For advice, please talk to WG chairs or ADs:

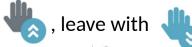
- <u>BCP 9</u> (Internet Standards Process)
- BCP 25 (Working Group processes)
- BCP 25 (Anti-Harassment Procedures)
- BCP 54 (Code of Conduct)
- BCP 78 (Copyright)
- BCP 79 (Patents, Participation)
- <a href="https://www.ietf.org/privacy-policy/">https://www.ietf.org/privacy-policy/</a> (Privacy Policy)



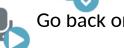


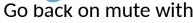
## Meeting Notes

- All decision will be confirmed on the mailing list.
- IETF 110 registration and a datatracker login required to attend
- Meeting is being recorded.
- Please use headphones when speaking.
- Remember to state full name when speaking.
- Bluesheets are tracked automatically by Meetecho.
- Microphone queue
  - O Enter session queue (audio/video) with 👢 , leave with 🦺



O Wait to be called. Enable audio by unmuting with Go back on mute with







Chairs will present slides.

#### **Resources:**

https://www.ietf.org/how/meetings/110/session-participant-guide/ https://www.ietf.org/media/documents/Documentation-Meetecho-IETF.pdf



## Administrivia

Jabber Room: asap@jabber.ietf.org

**Meetecho:** https://meetings.conf.meetecho.com/ietf110/?group=asap

Minutes: https://codimd.ietf.org/notes-ietf-110-asap

Minutes taker: Need 1 or 2

Jabber Scribe: Need 1

**Presentations:** <a href="https://datatracker.ietf.org/meeting/ietf110/session/asap">https://datatracker.ietf.org/meeting/ietf110/session/asap</a>

Mailing list: <a href="https://www.ietf.org/mailman/listinfo/asap">https://www.ietf.org/mailman/listinfo/asap</a>



## ASAP WG Agenda

- 1. Note Well, Note Takers, Agenda Bashing, Draft status (Chairs, 10 min)
- 2. Automatic Peering for SIP Trunks (K. Inamdar, S. Narayanan, 40 min) <a href="https://tools.ietf.org/html/draft-kinamdar-dispatch-sip-auto-peer">https://tools.ietf.org/html/draft-kinamdar-dispatch-sip-auto-peer</a>
- 3. Wrap-up and Next Steps (Chairs, 10 min)



## **Automatic Peering for SIP Trunks**

https://tools.ietf.org/html/draft-kinamdar-dispatch-sip-auto-peer

Kaustubh Inamdar, Sreekanth Narayanan, Cullen Jennings

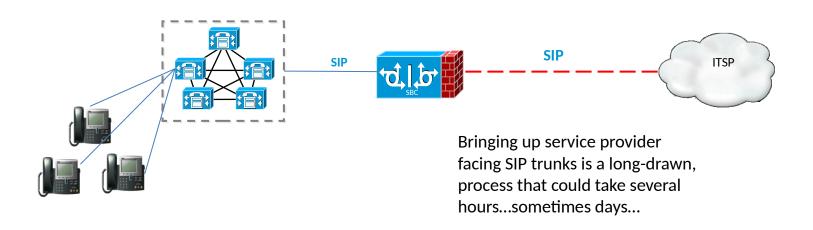


# Agenda

- Purpose of draft
- Story so far
- Open points of discussion



## Purpose of Draft



- Administrators usually uncover a number of problems when the trunk is setup.
- Requires a significant amount of interaction between the administrator and service provider to ensure everything works.
- Deployment times significantly increase because of interoperability problems.



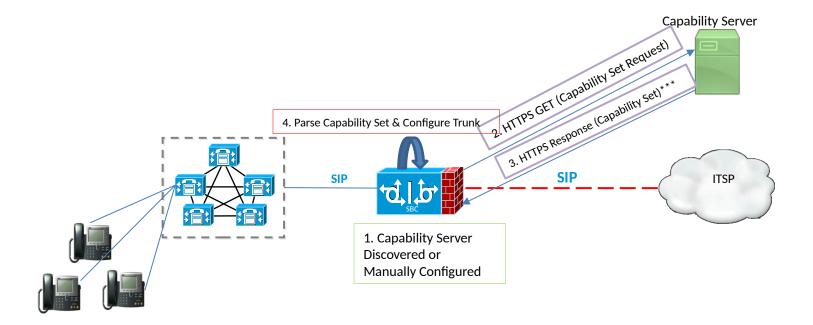
## Purpose of Draft (cont'd)

### How big is the problem?

- A total of 6000 support cases opened with Cisco in 2019 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...



# Purpose of the Draft: High-Level Overview



\*\*\* Body encoded in XML or JSON



## **Example Capability Set**

```
<peering-info xmlns="urn:ietf:params:xml:ns:yang:ietf-peering"</pre>
 xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance"
 xsi:schemaLocation="urn:ietf:params:xml:ns:yang:ietf-peering ietf-peering.xsd">
 <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>
   <registrarRealm>voip.example.com</registrarRealm>
   <callControl>callServer1.voip.example.com:5060</callControl>
   <callControl>192.168.12.25:5065</callControl>
   <dns>8.8.8</dns>
   <dns>208.67.222.222</dns>
   <outboundProxy>0.0.0/outboundProxy>
 </transport-info>
 <call-specs>
   <earlyMedia>true</earlyMedia>
   <signalingForking>false</signalingForking>
   <supportedMethods>INVITE;OPTIONS;BYE;CANCEL;ACK;PRACK;SUBSCRIBE;NOTIFY;REGISTER
   <numRange>
                                                                                   <rtp>
     <type>range</type>
                                                                                    <RTPTrigger>true
     <count>20</count>
                                                                                    <symmetricRTP>true</symmetricRTP>
     <value>19725455000</value>
                                                                                   </rtp>
   </numRange>
   <numRange>
                                                                                    <symmetricRTCP>true</symmetricRTCP>
     <type>block</type>
                                                                                    <RTCPFeedback>true</RTCPFeedback>
     <count>2</count>
                                                                                   </rtcp>
                                                                                 </media>
     <value>1972546000
     <value>1972546001
                                                                                   <payloadNumber>101</payloadNumber>
   </numRange>
                                                                                   <iteration>0</iteration>
 </call-specs>
                                                                                 </dtmf>
 <media>
                                                                                 <security>
   <mediaTypeAudio>
                                                                                   <signaling>
     <mediaFormat>PCMU:rate=8000:ptime=20</mediaFormat>
                                                                                    <type>TLS</type>
                                                                                    <version>1.0;1.2
     <mediaFormat> G729;rate=8000;annexb=ves</mediaFormat>
                                                                                   </signaling>
     <mediaFormat>G722;rate=8000;bitrate=56k,64k</mediaFormat>
                                                                                   <mediaSecurity>
   </mediaTypeAudio>
                                                                                    <keyManagement>SDES;DTLS-SRTP,version=1.2</keyManagement>
                                                                                    </mediaSecurity>
      cprotocol>pass-through
                                                                                   <certLocation>https://sipserviceprovider.com/certificateList.pem</certLocation>
     col>t38
                                                                                   </security>
   </fax>
                                                                                 <extensions>timer;rel100;gin;path</extensions>
                                                                               </pering-response>
```

## Story So Far

- Versions 1&2 were published before presenting at IETF 106.
- Consensus at IETF 106 to form a mini-working group.
- Charter for proposed workgroup reviewed in April 2020 and approved in the following months.
- ASAP workgroup formed in August 2020 and now has two cochairs assigned (Gonzalo Salgueiro and Jean Mahoney)



## **Version History**

- Versions -1 & -2:
  - Initial draft followed by name change and formatting changes
- Version -3:
  - Added certLocation field to the capability set.
  - Field is populated with a URL that provides a single PEM encoded file that contains all certificates in the chain of trust.



# Version History (cont'd)

### Version -4:

- Added the numRange field to the capability set.
- numRange specifies the Direct Inward Dial (DID) number range allocated to the enterprise network by the SIP service provider.
- Can be communicated as a list of values or a reference.



# Version History (cont'd)

### Version -5:

- Clarifying HTTP versions supported by this draft (v1.1 and above).
- Clarifying the scope of work not to include low level details (e.g., IP addressing scheme, access network infrastructure)
- Added a paragraph to clarify how this workflow operates when intermediary service providers are present between the enterprise and peering service provider.



## Open Issues

- Inclusion of provision for discovery of capability set servers in service provider networks (e.g., via WebFinger).
- Are there any other parameters that need to be added to the capability set?
- Are there any modifications/clarifications that need to be carried out/provided?
- Are we good to have this draft adopted by the ASAP working group?



## Call for Adoption

Adopt as WG item?

**Automatic Peering for SIP Trunks** 

(https://tools.ietf.org/html/draft-kinamdar-dispatch-sip-auto-peer)

[[Decision to be confirmed on-list]]



# Thank You!

