Call me...maybe

A Guide to SIP Integration in the Cloud Era

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For App Developers, SIP is a difficult protocol to use

Quotes about SIP in Developer Forums:

- "SIP is Dead…Long Live WebRTC!"
- "HTTP has its problems, but implementing it is a walk in the park compared to many other telecom protocols. Try SIP and its dozens of add-on RFCs"
- "I wouldn't call [SIP] exactly simple or easy protocol to follow and implemented. Thus many alternatives or self-made solutions are more likely to be chosen. It does a lot, but at the same time it is increasingly complex protocol."

Suppose you're an app developer…

- Your background is developing services for companies & apps for consumers
- REST over HTTPS is your preferred protocol
- Server authentication: firewall ACLs & TLS certificates
- Domain names route traffic
- Availability zones…not data centers
- Software updates are automatic; rollbacks instant
- You rely on modules from 57 different sources and active APIs

Contrast that with typical voice service providers…

- Your work is providing voice services to companies and consumers
- SIP over UDP is your preferred protocol
- Authentication is done in SIP and with firewall ACLs
- Domain names are used as names, and to route traffic
- You have carefully designed data centers to minimize latency
- Software updates are done quarterly at most
- You have at most four key vendors involved in any call

"Call me...maybe?"

App developers can struggle to even get a basic call to complete!

- If the App developer cannot integrate with a Voice service provider over SIP, they can't use the phone network
- What does the Voice Service Provider and the App developer need to both understand about each other?

SIP Stack Options

Domain Names

SIP Transport: UDP, TCP, TLS

TLS Certificates & Certificate Authorities

Telephone Number Formatting

Packet Capture for Trouble-shooting

SRTP Crypto Negotiation

HTTP POST Mindset

Contact & Via Headers

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There's just not enough time in this session to cover all Nine big issues!

Scan to download a full developer guide

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Telephone Number Format

- One key struggle: the format for telephone numbers
- Developers are familiar with how they personally dial phones, but those are rarely just right for SIP trunking providers
- There are some software libraries that will attempt to standardize numbers, but they all fail
- And many Service Providers limit the telephone numbers that can be used as the calling party number (From) when making calls. (Call Screening is important for Call Authentication!)

Telephone Number Format

E.164 format with leading plus:

```
INVITE sip:+12293160013@example.com;user=tel SIP/2.0
From: <sip:+12293160000@example.com>;tag=abcdef
To: <sip:+12293160013@example.com>
```

"National" 10-digit format:

```
INVITE sip:2293160013@example.com;user=tel SIP/2.0
From: <sip:2293160000@example.com>;tag=abcdef
To: <sip:2293160013@example.com>
```

Telephone Number Format

SIP "Address of Record" format that expects the endpoint to just figure out what it means because it doesn't clarify that the digits represent a telephone number:

```
INVITE sip:2293160013@example.com SIP/2.0
```

Historical format for calling International numbers from the US:

```
INVITE sip:011442293160013@example.com SIP/2.0
```

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Domain Names

- Developers use domain names for *Internet domains*
- Domains are used in lots of different ways in SIP…and knowing which one is relevant can be a trick!
- Some Voice service providers try to live in a "Domain Name Free" world, where they just use IP addresses. But app developers don't operate like this! (Nor does Microsoft)

Domain Name for SIP Server Location

- A Domain name like "Vwave.net" can be used to actually find the right IP address of a destination server
- But…it's not just DNS A records like with *most applications*. It uses DNS SRV and NAPTR, as explained in RFC 3263
- For SIP traffic going from the Voice Provider toward a Cloud-hosted app, using the right domain name in the right way requires a mindset shift from ordinary web/API development

Domain Name Confusion In Action

Suppose the App Developer sends this INVITE to the VoIP Carrier SBC…

```
INVITE sip:+12293160013@VoIPCo.com;user=tel SIP/2.0

From: <sip:+12293160000@VoIPCo.com>;tag=abcdef

To: <sip:+12293160013@VoIPCo.com>

Contact: <sip:101@ec2-44-226-107-251.us-west-2.compute.amazonaws.com>

Via: SIP/2.0/UDP 10.11.228.67:5060;branch=z9hG4bK10_16a83292baa1de54e0b7843_I
```

Domain Name Complexity In Action

VoIPCo.com is the domain name of the SIP Service Provider for SIP Header Purposes

```
INVITE sip:+12293160013@VoIPCo.com;user=tel SIP/2.0
From: <sip:+12293160000@VoIPCo.com>;tag=abcdef
To: <sip:+12293160013@VoIPCo.com>
Contact: <sip:101@ec2-44-226-107-251.us-west-2.compute.amazonaws.com>
Via: SIP/2.0/UDP 10.11.228.67:5060;branch=z9hG4bK10_16a83292baa1de54e0b7843_I
```

Domain name for the reserved IP address for the domain name of the specific server that sent this call

Internal IP address of the Kubernetes Pod where the container was running

Common Gotcha! App developers need to know how the Contact and Via headers work for routing subsequent requests (like PRACK) and responses (like 200)

Domain Name Complexity In Action

But none of those are the domain name used to actually route the SIP message across the Internet..

App Developer Infrastructure

sip-trunk-app123.VoIPCo.net

Voice Service Provider

```
INVITE sip:+12293160013@VoIPCo.com;user=tel SIP/2.0
From: <sip:+12293160000@VoIPCo.com>;tag=abcdef
To: <sip:+12293160013@VoIPCo.com>
Contact: <sip:101@ec2-44-226-107-251.us-west-2.compute.amazonaws.com>
Via: SIP/2.0/UDP 10.11.228.67:5060;branch=z9hG4bK10_16a83292baa1de54e0b7843_I
```

Domain Name Complexity In Action

And yet another domain name is used to send a call from the VoIP Service Provider to the App.

App Developer Infrastructure

sip.AppDeveloper.net

Voice Service Provider

```
INVITE sip:+12293160000@sip.AppDeveloper.net;user=tel SIP/2.0
From: <sip:+12293160000@sip.AppDeveloper.net>;tag=abcdef
To: <sip:+12293160013@sip.AppDeveloper.net>
Contact: <sip:216.128.202.68>
Via: SIP/2.0/UDP 216.128.202.68:5060;branch=z9hG4bK10_3292baa1de54e0b7843_I
```

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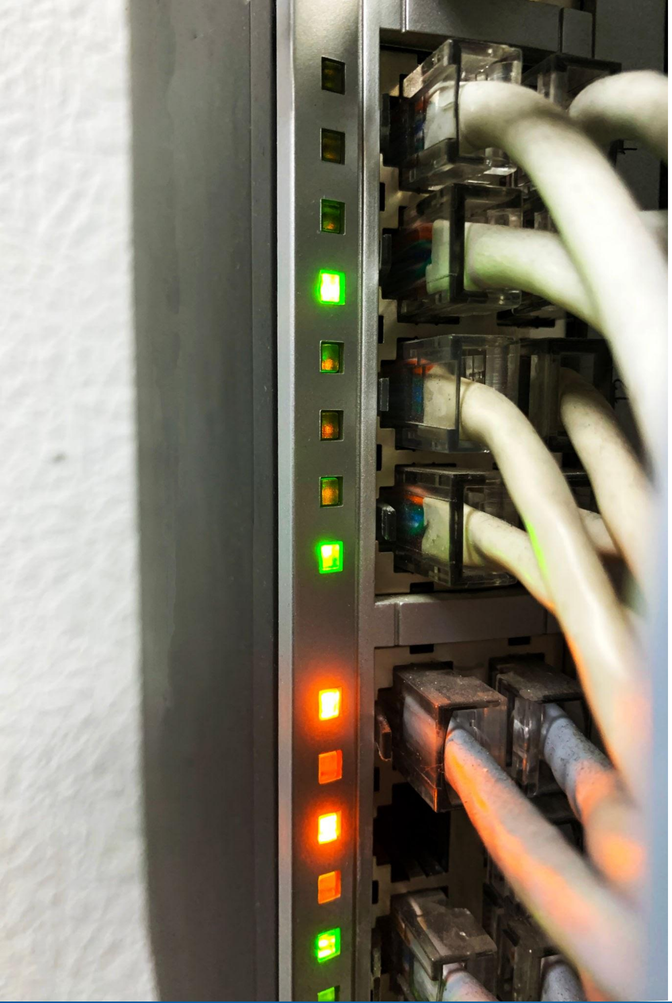
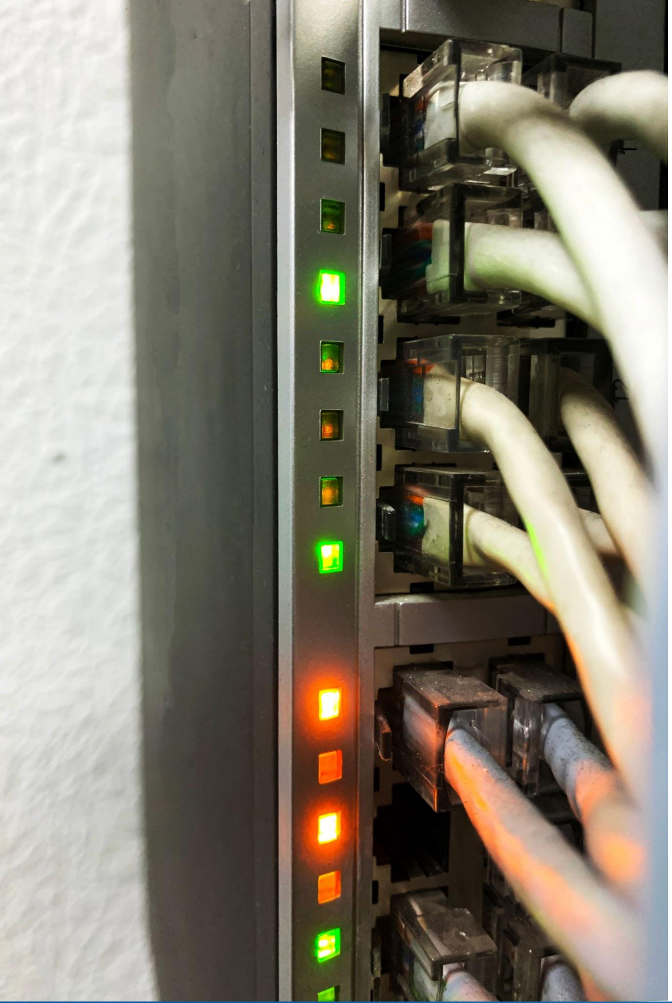


Packet Captures

- The SIP world of Voice service providers will often ask: "show me a SIP capture" or "send me a capture of the RTP"
- This comes from a mindset of unencrypted traffic, and readily available packet captures
- …But modern application development does not lend itself to packet capture *per se*



Packet Captures

- Traffic can be routed to a compute instance (what legacy types would call a "server") through several layers of application firewalls and load balancers
- "Security First" requirements may strictly prohibit decrypting or logging the data that could have sensitive information, for purposes of HIPAA or NIST 800-53 compliance
- Result: App developers may never be able to provide the packet captures of the SIP and RTP traffic that VoIP engineers are accustomed to getting

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Developers & VoIP Providers need to work together on all of these challenges.

Thank You!

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https://www.ecg.co/contact

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