OpenSIPS Workshop

Federated SIP
with OpenSIPS and RTPEngine

Who are you people?

- Eric Tamme
  - Principal Engineer
- OnSIP
  - Hosted PBX
  - Hosted SIP Platform
  - Developers of SIP.js

See: sipjs.com, or https://github.com/onsip/sip.js

Quick Outline

- VM Setup
- What is Federated SIP
  - Federated SIP Proxy (RFC3263)
  - REGEX based outbound translations
  - Registrar
  - Server Side NAT handling
  - Media Relay

Get the code!

- Project Github page
  - https://github.com/etamme/federated-sip
- Create a new Centos7, or Debian8 VM
  - Preferably create a VM with a public IP address (Digital Ocean)
- Install git
  - yum -y install git

    OR
  - apt-get -y install git
- From your VM Clone the repo
  - https://github.com/etamme/federated-sip.git

VM Setup

- cd federated-sip

  Run the install script as root

- scripts/centos7_install.sh

  OR
- scripts/debian8_install.sh
- DO NOT enter a domain or user, just press ENTER when prompted

What is the install script doing

- Installing dependencies
- Cloning repos
  - OpenSIPS
  - RTPEngine
  - Sqlite3-pcre
- Compiling projects
- Creating opensips database
- Building opensips config from federated-sip
- Starting rtpengine and opensips
- Adding a user and domain

What is the federated-sip project?

- Reference implementation for OpenSIPS
  - Federated SIP proxy
  - General purpose: registrar, server side nat handling
  - Multi protocol: UDP, TCP, TLS, WS … WSS?
  - RTP interop: RTP, SRTP, DTLS-SRTP

- My personal SIP service

What is federated SIP signalling?

- Core RFC3263 support “locating sip servers”
  - Section 4:
    - “in all cases, the problem boils down to resolution of a SIP or SIPS URI in DNS to determine the IP address, port, and transport of the host to which the request is to be sent.”

General process

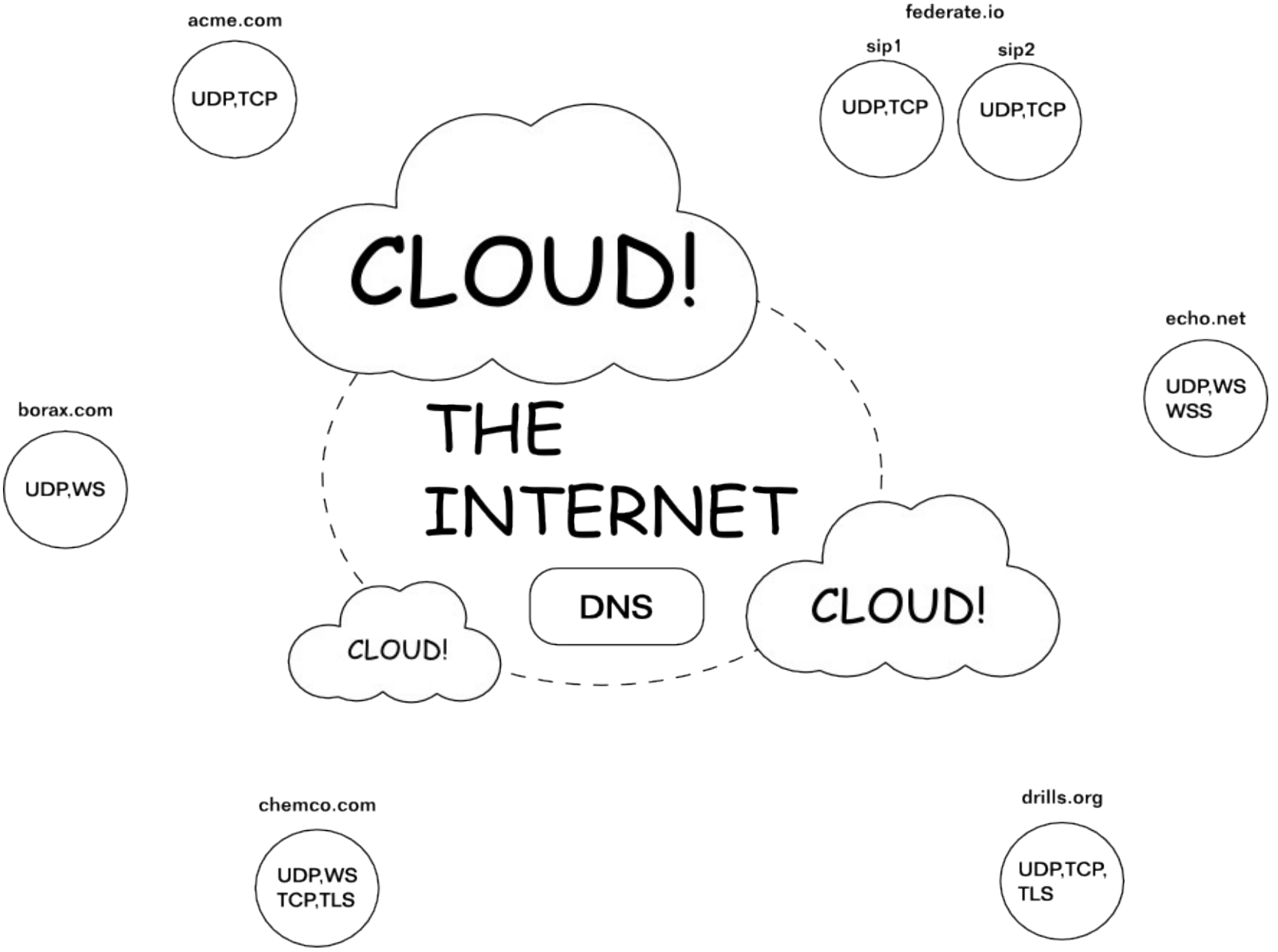
- NAPTR query to get transports in SRV records
- SRV query to get A records for given transports and ports
- A record lookup to get IP

“If no NAPTR records are found, the client constructs SRV queries for those transport protocols it supports, and does a query for each.”

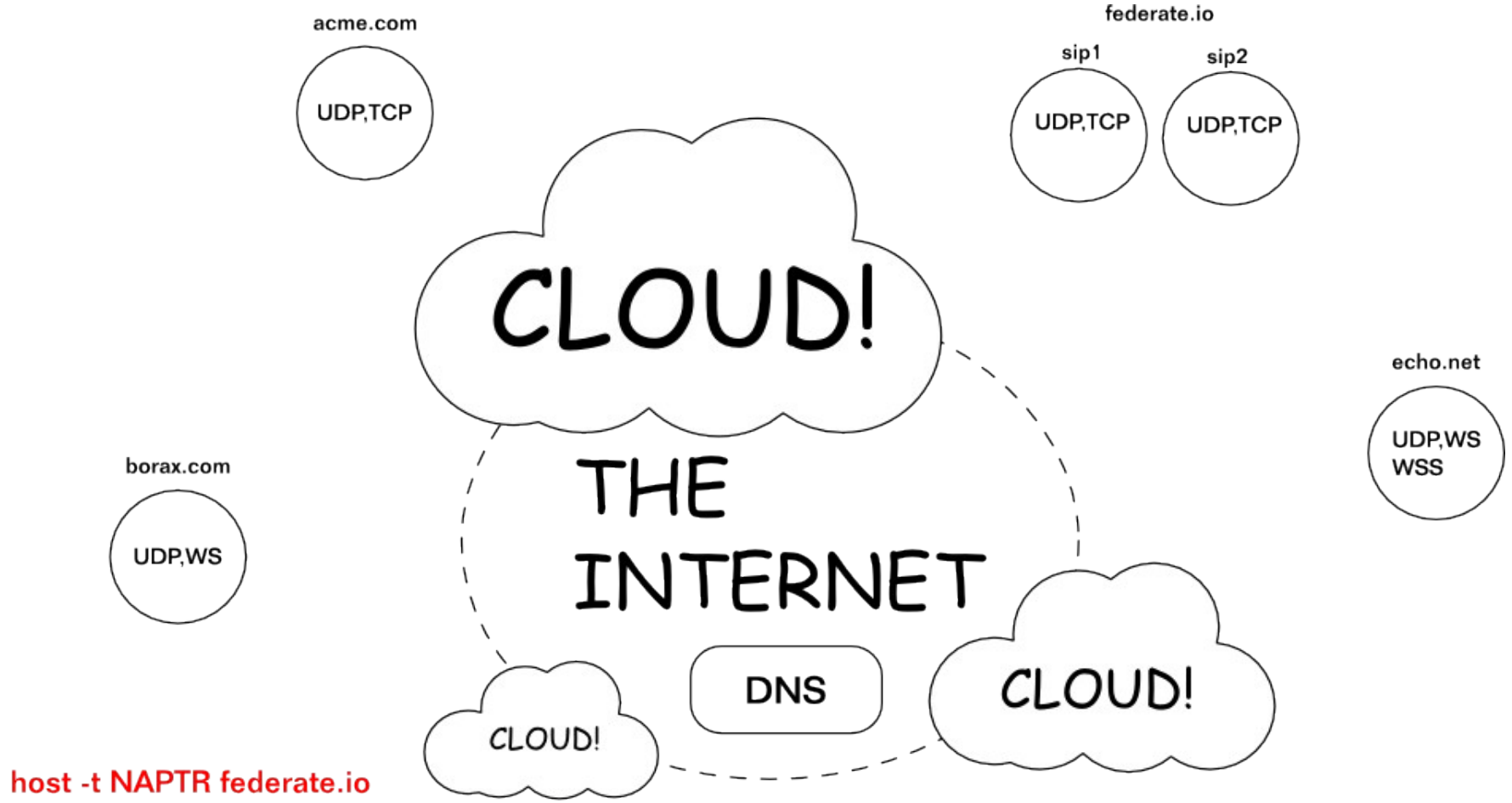
“gotchas”

- If there is a port, only do an A record lookup
- transport parameter on uri
  - SHOULD be used … not MUST
- Host portion is an IP, but no transport
  - SHOULD use UDP for SIP and TCP for SIPS

Locating SIP servers



Locating SIP servers: NAPTR




```
host -t NAPTR federate.io
federate.io NAPTR 10 100 "S" "SIP+D2U" "" _sip._udp.federate.io.
federate.io NAPTR 20 100 "S" "SIP+D2T" "" _sip._tcp.federate.io.
```

NAPTR Records

```
federate.io NAPTR 10 100 "S" "SIP+D2U" "" _sip._udp.federate.io.
federate.io NAPTR 20 100 "S" "SIP+D2T" "" _sip._tcp.federate.io.
```

Priority Weight

Flags:

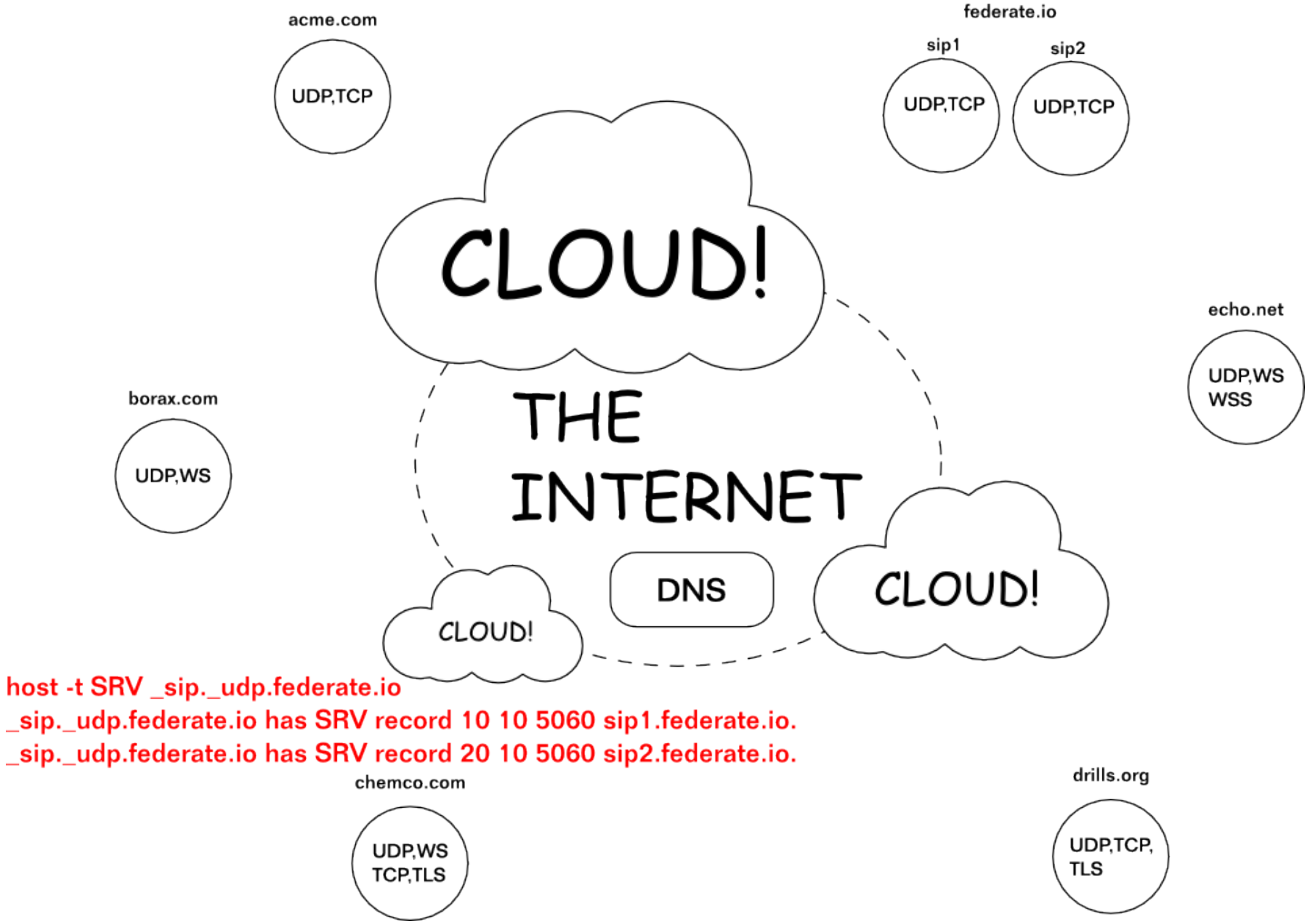
```
S:  SRV
A:  A
A6: AAAA
U:  Absolute URI
```

Service:

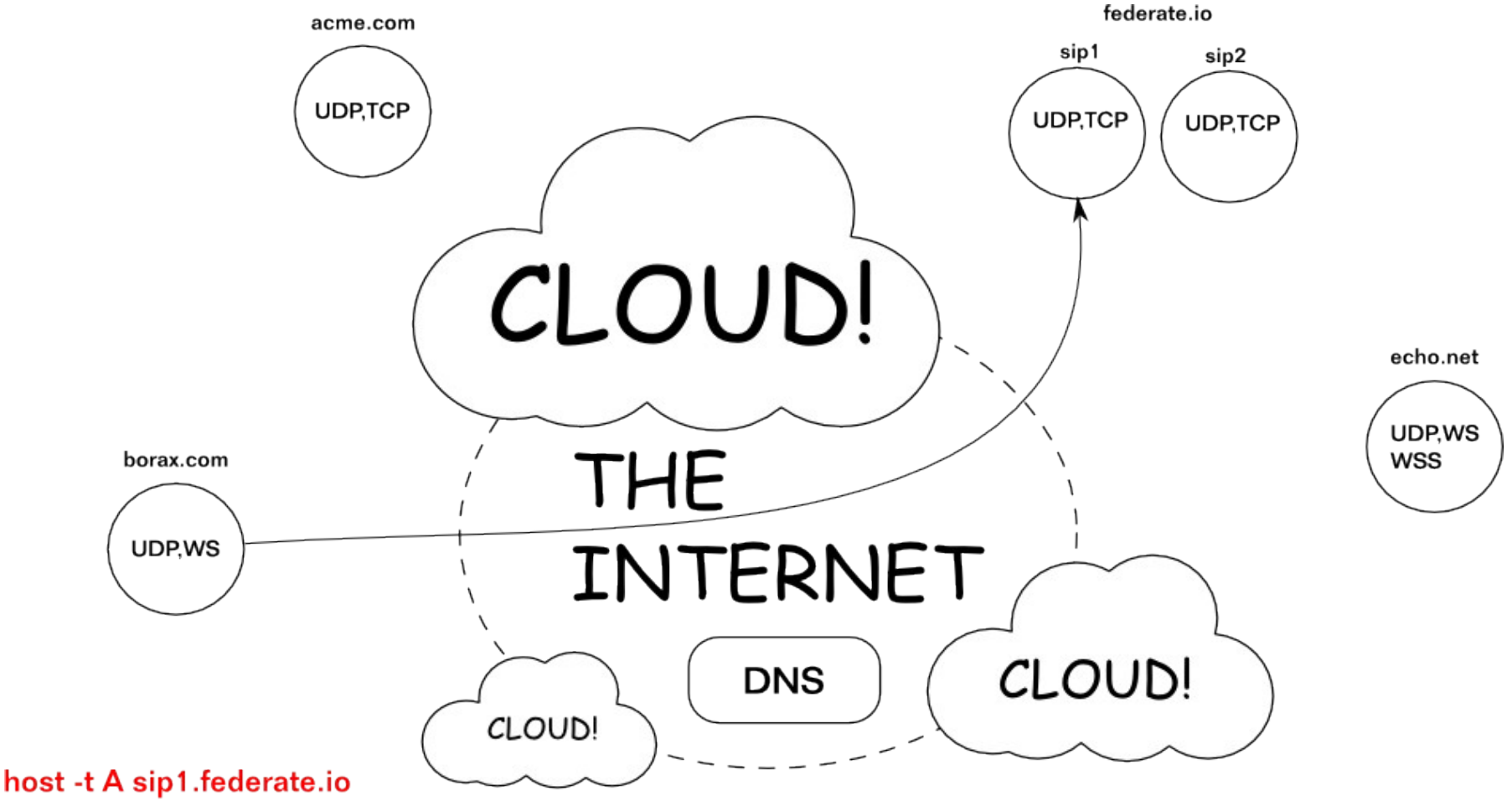
```
SIP+D2U:  sip udp
SIP+D2T:  sip tcp
SIPS+D2T: sip tcp
SIP+D2W:  sip ws
SIPS+D2W: sip ws
```

Replacement

Locating SIP servers: SRV



Locating SIP servers: A



Thats it!

- *for initial requests
- RFC3263 specifies that the same host must be used throughout the duration of the transaction.
- Responses route back following Via headers

Sequential request routing

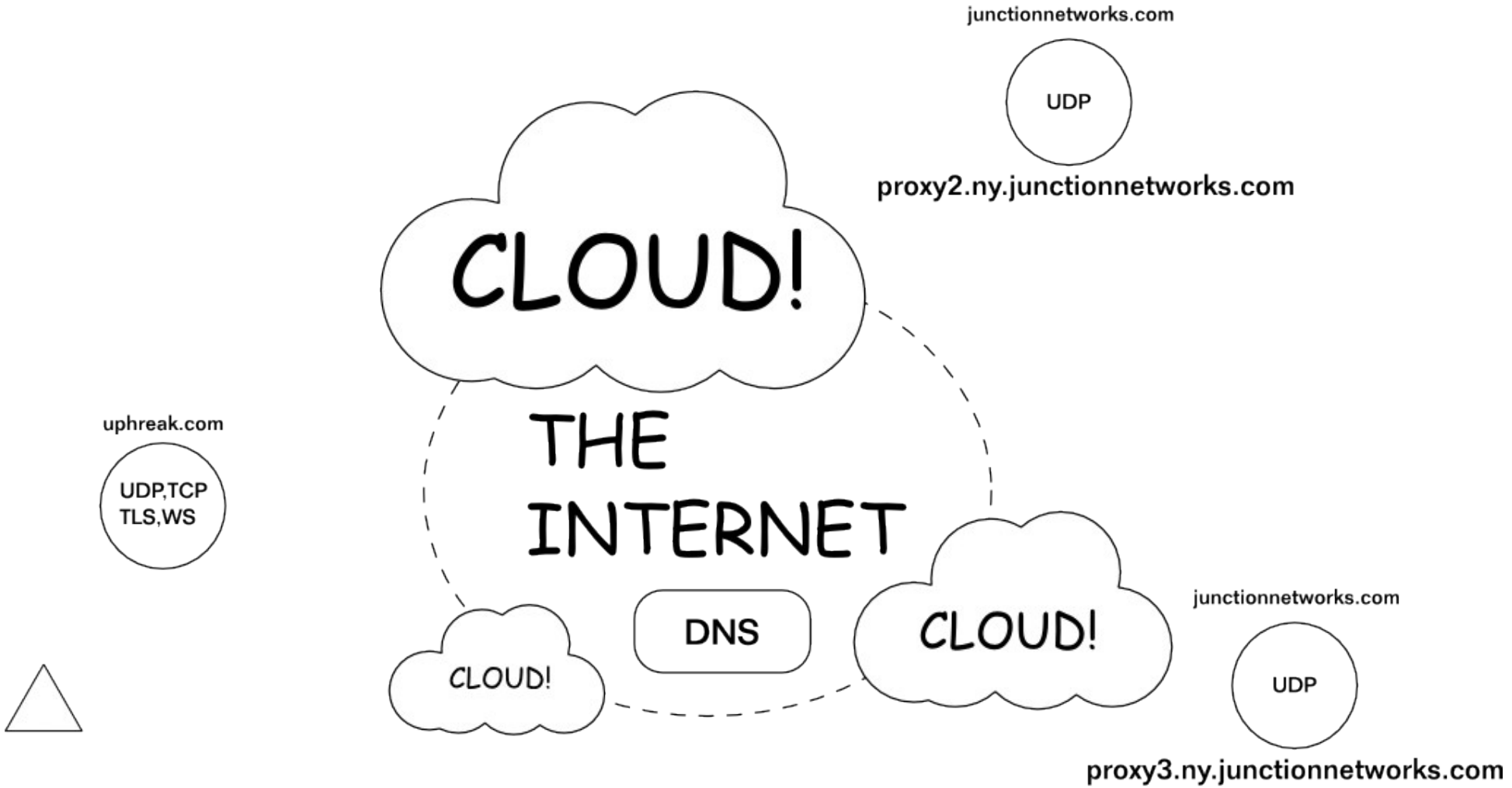
- Initial requests are only half the picture
- Record Route

```
# record route all requests b/c it doesn't hurt, and some
# UA are buggy and throw away route sets
if (!is_method("REGISTER")) {
  record_route();
}
```

- Loose Route

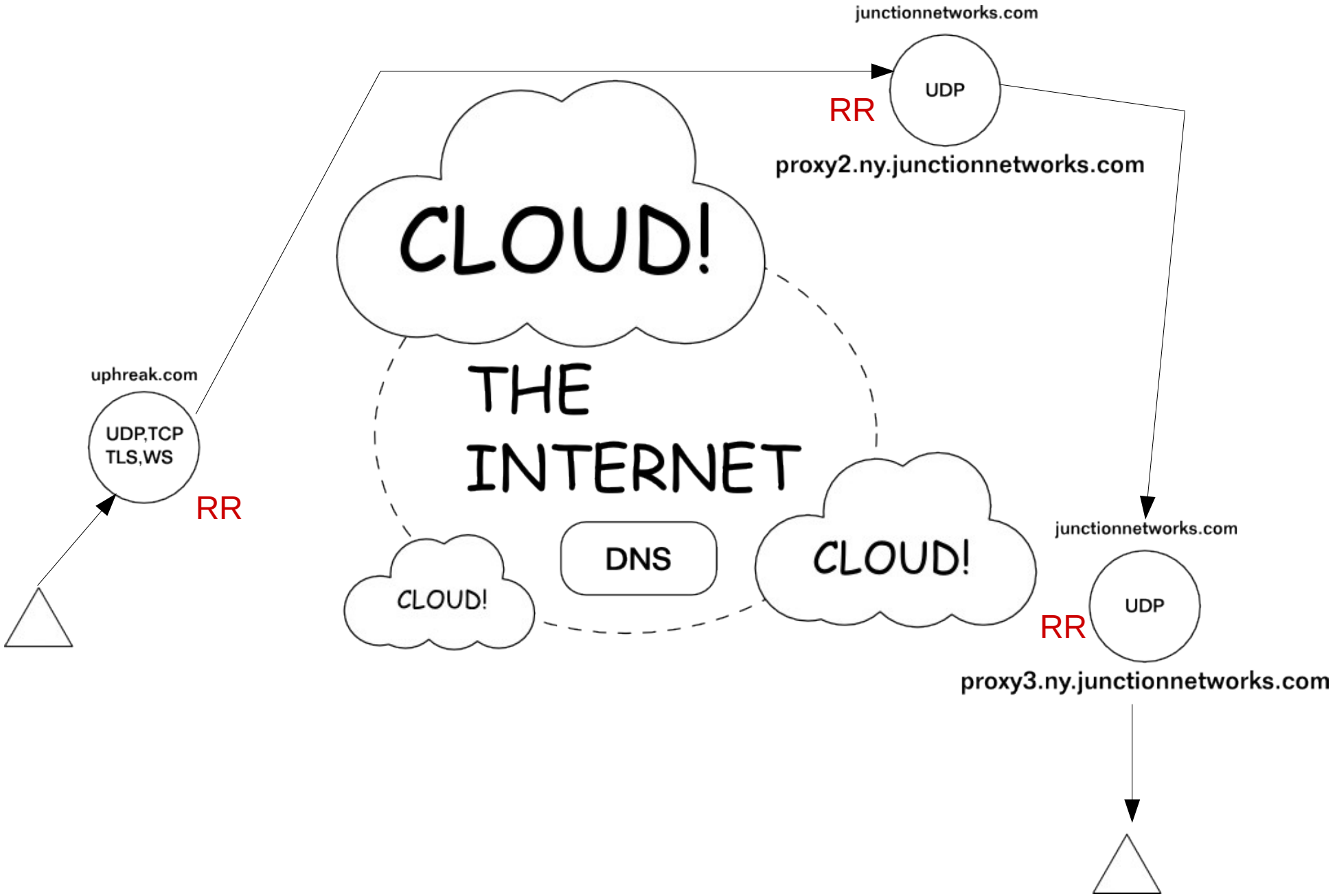
```
# sequential requests should take the path built up from record-routing
if (loose_route()) {
  xlog("L_INFO","$var(prefix) loose routing\n");
}
```

Real world example



```
U 2015/07/28 16:53:09.568040 199.7.175.103:5060 -> 50.198.204.238:5066
INVITE sip:eric.tamme@10.1.10.26:5066;transport=udp;aor=eric.tamme%40junctionnetworks.com SIP/2.0.
Record-Route: <sip:199.7.175.103;lr;ftag=DSskm2PEh-IfTdATjmF4Yuyapue8yyrv;did=5d5.604bdf12;ns=1;pr=1>.
Record-Route: <sip:199.7.173.102;lr;ftag=DSskm2PEh-IfTdATjmF4Yuyapue8yyrv;did=5d5.c7bb8835>.
Record-Route: <sip:107.170.192.145;lr;ftag=DSskm2PEh-IfTdATjmF4Yuyapue8yyrv;did=5d5.ce8aab34>.
Via: SIP/2.0/UDP 199.7.175.103:5060;branch=z9hG4bK0f9d.dd184a3.0.
Via: SIP/2.0/UDP 199.7.173.102:5060;branch=z9hG4bK0f9d.889da69.2.
Via: SIP/2.0/UDP 107.170.192.145:5060;branch=z9hG4bK0f9d.136affc1.0.
Via: SIP/2.0/UDP 10.1.10.29:38718;branch=z9hG4bKPjsqf0FITZ1BWPyZvUjJdy2sqiX5bgBk38.
Max-Forwards: 67.
From: "Eric Tamme" <sip:eric@uphreak.com>;tag=DSskm2PEh-IfTdATjmF4Yuyapue8yyrv.
To: sip:eric.tamme@junctionnetworks.com.
Contact: "Eric Tamme" <sip:eric@50.198.204.238:38718;nat=yes>.
Call-ID: TDwXjq9RQS7PZXFQlFAYcHh4noLcjbfa.
CSeq: 26703 INVITE.
Allow: INVITE, ACK, BYE, CANCEL, UPDATE, INFO, SUBSCRIBE, NOTIFY, REFER, MESSAGE, OPTIONS.
Supported: replaces, timer, norefersub.
Session-Expires: 1800.
Min-SE: 90.
User-Agent: Bria Android 3.2.4.
Content-Type: application/sdp.
Content-Length: 361.
.
v=0.
o=- 3647091188 3647091188 IN IP4 107.170.192.145.
s=cpc_med.
c=IN IP4 199.7.175.93.
t=0 0.
m=audio 62320 RTP/AVP 111 9 0 101.
a=rtpmap:111 OPUS/48000/2.
a=fmtp:111 maxplaybackrate=32000;useinbandfec=1.
a=rtpmap:9 G722/8000.
a=rtpmap:0 PCMU/8000.
a=rtpmap:101 telephone-event/8000.
a=fmtp:101 0-16.
a=ptime:20.
a=sendrecv.
a=rtcp:62321 IN IP4 199.7.175.93.
```

Real world example



```
U 2015/07/28 16:53:12.128224 50.198.204.238:1024 -> 199.7.175.103:5060
SIP/2.0 200 OK.
Via: SIP/2.0/UDP 199.7.175.103:5060;branch=z9hG4bK0f9d.ed184a3.0.
Via: SIP/2.0/UDP 199.7.173.102:5060;branch=z9hG4bK0f9d.889da69.0.
Via: SIP/2.0/UDP 107.170.192.145:5060;branch=z9hG4bK0f9d.136affc1.0.
Via: SIP/2.0/UDP 10.1.10.29:38718;;branch=z9hG4bKPjsqf0FITZ1BWPyZvUjJdy2sqiX5bgBk38.
Record-Route: <sip:199.7.175.103;lr;ftag=DSskm2PEh-IfTdATjmF4Yuyapue8yyrv;did=5d5.704bdf12;ns=1;pr=1>.
Record-Route: <sip:199.7.173.102;lr;ftag=DSskm2PEh-IfTdATjmF4Yuyapue8yyrv;did=5d5.c7bb8835>.
Record-Route: <sip:107.170.192.145;lr;ftag=DSskm2PEh-IfTdATjmF4Yuyapue8yyrv;did=5d5.ce8aab34>.
From: "Eric Tamme" <sip:eric@uphreak.com>;tag=DSskm2PEh-IfTdATjmF4Yuyapue8yyrv.
To: <sip:eric.tamme@junctionnetworks.com>;tag=1839602824.
Call-ID: TDwXjq9RQS7PZXFQlFAYcHh4noLcjbfa.
CSeq: 26703 INVITE.
Contact: <sip:eric.tamme@10.1.10.20:5060>.
Supported: replaces, path, timer, eventlist.
User-Agent: Grandstream GXV3275 1.0.3.30.
Session-Expires: 1800;refresher=uac.
Require: timer.
Allow: INVITE, ACK, OPTIONS, CANCEL, BYE, SUBSCRIBE, NOTIFY, INFO, REFER, UPDATE, MESSAGE.
Content-Type: application/sdp.
Content-Length:   237.
.
v=0.
o=eric.tamme 8000 8000 IN IP4 10.1.10.20.
s=SIP Call.
c=IN IP4 10.1.10.20.
t=0 0.
m=audio 5004 RTP/AVP 9 0 101.
a=sendrecv.
a=rtpmap:9 G722/8000.
a=ptime:20.
a=rtpmap:0 PCMU/8000.
a=rtpmap:101 telephone-event/8000.
a=fmtp:101 0-15.
```

Sequential request routing

```
U 2015/07/28 13:01:39.279860 50.198.204.238:38718 -> 107.170.192.145:5060
BYE sip:eric.tamme*50.198.204.238!1024_n@199.7.175.103;gr SIP/2.0.
Via: SIP/2.0/UDP 10.1.10.29:38718;rport;branch=z9hG4bKPjkTV7NVDDzF74SoHi8y3V6gel2trc.Mb7.
Max-Forwards: 70.
From: "Eric Tamme" <sip:eric@uphreak.com>;tag=Yizu0F5VPgLxBLunhDYfWxd4m5MKAcOR.
To: sip:eric.tamme@junctionnetworks.com;tag=1878718727.
Call-ID: TDwXjq9RQS7PZXFQlFAYcHh4noLcjbfa.
CSeq: 17180 BYE.
Route: <sip:107.170.192.145;lr;ftag=Yizu0F5VPgLxBLunhDYfWxd4m5MKAcOR;did=869.5a9ffc25>.
Route: <sip:199.7.173.102;lr;ftag=Yizu0F5VPgLxBLunhDYfWxd4m5MKAcOR;did=869.0b81b316>.
Route: <sip:199.7.175.103;lr;ftag=Yizu0F5VPgLxBLunhDYfWxd4m5MKAcOR;did=869.9a2f9f17;ns=1;pr=1>.
User-Agent: Bria Android 3.2.4.
Content-Length:  0.
.
```

REGEX based translations

- Match on
  - REGEX user portion
  - From domain
  - From user
  - User agent
- Translate
  - Strip digits
  - Add prefix
  - Replace user
  - Replace domain
  - Replace port
  - Strip port
  - Add header
  - Multiple matches with priority weights

Example translation

```
insert into translations
(
 id,from_domain,match_regex,
 tran_user,  tran_domain
)
Values
(
1,
'proxy123.uphreak.com',
'^1000$','eric',
'workshop.uphreak.com'
);
```

Example translation

```
insert into translations
(
 id,from_domain,match_regex,
 tran_domain,
 tran_strip
)
Values
(
1,
'proxy123.uphreak.com',
'^\+18[0678]{2}[0-9]{7}$',
'tf.arctele.com',
1
);
```

Code review

- Server side nat handling + Registrar
- RTPEngine media relaying and interop

Thank You

- OpenSIPS Team, and community
- Jarrod Baumann for federated-sip contributions

- Please checkout and use:

  https://github.com/etamme/federated-sip

  more users == better

- Contact

  sip:eric.tamme@onsip.com

  email:eric.tamme@onsip.com