OpenSIPS and WebRTC

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Who we are

- VoIP and OpenSIPs software development and consultancy
- Based in UK
- Some of larger customers are
  - Localphone
    - Retail ITSP offering (VoIP accounts, apps, DIDs in UK, US, Europe, Worldwide)
    - Over 1,000,000 users
  - Magic Telecom
    - US Faclities based CLEC
  - Voxbeam
    - Wholesale, A-Z Termination, VoIP reseller
    - US CLEC
    - Terminate ~20,000,000 mins/week internationally
- We use
  - OpenSIPS
  - Asterisk
  - FreeSWITCH
  - RabbitMQ
  - Redis
  - Hadoop
  - Homer
  - Sangoma

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Workshop Aims

- Demonstrate new Websocket (secure) module
- Demonstrate how a browser can now be a SIP UA with OpenSIPs as the proxy
- Demonstrate how a browser can talk to the regular PSTN
- Live(!) and Exciting(!!) realtime OpenSIPs configuration!

What is WebRTC

- New standard ratified by W3C
- Defines how browser to browser media sessions can be established (Video and Audio)
- As of 2016
  - Good support in Chrome and Firefox
  - Android browser support
  - iOS browser support.

WebRTC = Standards

- Standardisation on Audio/Video encryption with DTLS/SRTP
- Standardisation on media NAT traversal using ICE

WebRTC != SIP

- Transport agnostic
- SIP can carry SDP transport
- In addition to WebRTC, browsers now support Websocket connections
- OpenSIPS now supports websocket connections

Result

Browser WebRTC

+ Browser Websocket support

+ OpenSIPS Websocket support

= audio and video calls from a browser, via OpenSIPS

WebRTC NAT

- Browser has no access to host IP.
- SIP request sent up websocket with no useful routing information.
- OpenSIPs must use fix_nated_register() to send requests and responses to IP and Port received (1,2)

```
REGISTER sip:146.185.161.38 SIP/2.0
Via: SIP/2.0/WSS lems5cahkshb.invalid;branch=z9hG4bK3199187
Max-Forwards: 70
To: <sip:peter@146.185.161.38>
From: <sip:peter@146.185.161.38>;tag=t0rk4v6e7v
Call-ID: 4gqa93oui7aib2pqc32obh
CSeq: 84 REGISTER
Contact: <sip:c339t4mq@lems5cahkshb.invalid;transport=ws>;reg-id=1;+sip.instance="
<urn:uuid:93370ae2-b982-48bb-86ca-1b0ba9e6ba8c>";expires=600
Allow: ACK,CANCEL,INVITE,MESSAGE,BYE,OPTIONS,INFO,NOTIFY,REFER
Supported: path, gruu, outbound
User-Agent: SIP.js/0.7.5
Content-Length: 0
```

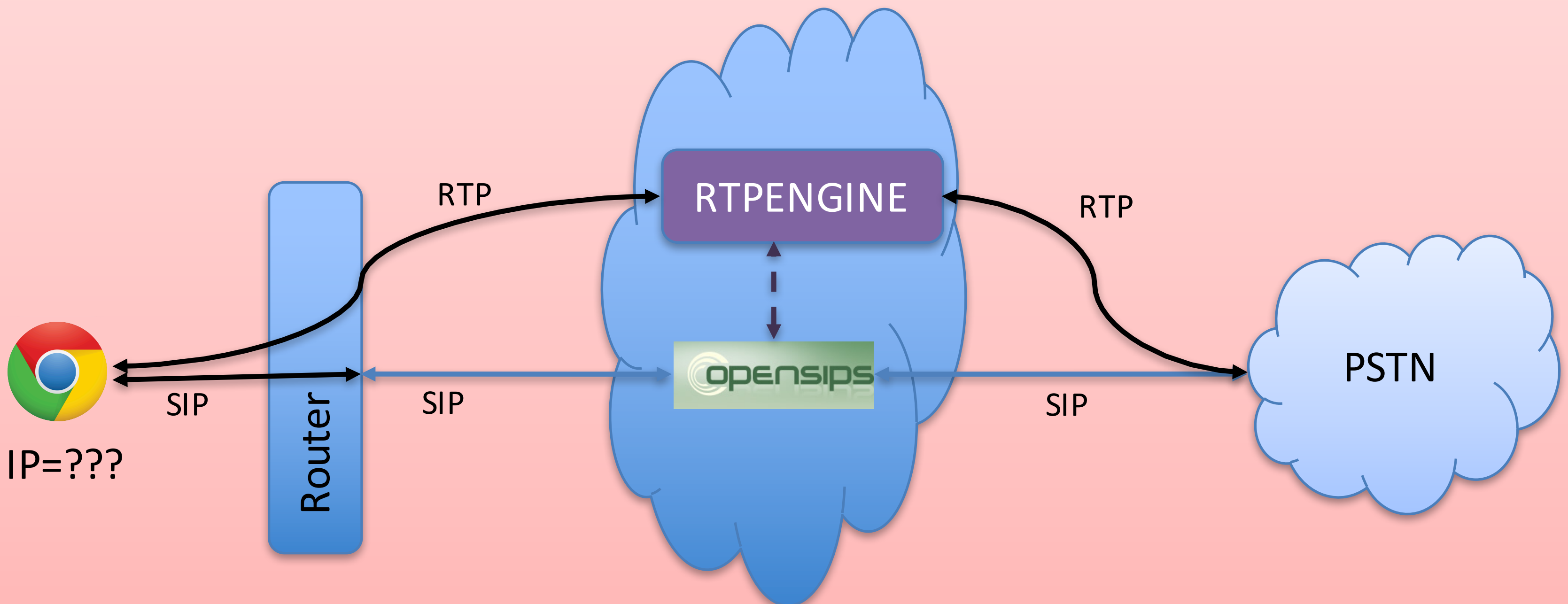
WebRTC NAT

```
INVITE sip:447968145898@146.185.161.38 SIP/2.0
Via: SIP/2.0/WSS lems5cahkshb.invalid;branch=z9hG4bK3904058
Max-Forwards: 70
To: <sip:447968145898@146.185.161.38>
From: <sip:peter@146.185.161.38>;tag=0d8gvshgqo
Call-ID: bg1j6kpo9v8m3shcn3fv
CSeq: 8302 INVITE
Contact: <sip:c339t4mq@lems5cahkshb.invalid;transport=ws;ob>
Allow: ACK,CANCEL,INVITE,MESSAGE,BYE,OPTIONS,INFO,NOTIFY,REFER
Content-Type: application/sdp
Supported: outbound
User-Agent: SIP.js/0.7.5
Content-Length: 5281
```

```
SIP/2.0 200 OK
Via: SIP/2.0/WSS lems5cahkshb.invalid;received=188.39.51.3;branch=z9hG4bK6957597
Record-Route: <sip:146.185.161.38;r2=on;lr>,
<sip:146.185.161.38:443;transport=wss;r2=on;lr>
To: <sip:447968145898@146.185.161.38>;tag=3671427251-152698
From: <sip:peter@146.185.161.38>;tag=dn5lmhchir
Call-ID: bg1j6ilillhb6486pihr
CSeq: 1070 INVITE
Allow: CANCEL, ACK, INVITE, BYE, OPTIONS, REGISTER, NOTIFY, INFO, REFER, SUBSCRIBE,
PRACK, UPDATE, MESSAGE, PUBLISH
Contact: <sip:146.185.161.38:5070;did=d06.3a57c1e>
Content-Type: application/sdp
Accept: application/sdp
Content-Length: 581
```

- Same problem with INVITE and 200OK reply
- OpenSIPS must use fix_nated_contact() so UAS and UAC insert correct IP for routing SIP packet.

Relay media to PSTN



- RTPEngine converts "ICE" SDP to regular SDP and vice versa
- Traditional PSTN gateways can talk to WebRTC media
- Result is Browser to PSTN calls

Conclusions

- WebRTC enables browser to be used as a SIP UA
- Easier for end users: No software to download, no proxy, username, transport, STUN, password, SSL certificate info needed
- WebRTC standard delivers secure voice and audio
- ICE negotiation removes any real need for NAT handling for media
- Easy to bolt onto an existing OpenSIPS registrar using WS and WSS modules.
- Easy to integrate into existing web apps. e.g. PRESENCE support, MESSAGE support.

Code samples

The code used in the presentation can be found at

- **https://github.com/petekelly/opensips-summit-2016**

Thank You…

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