WebRTC

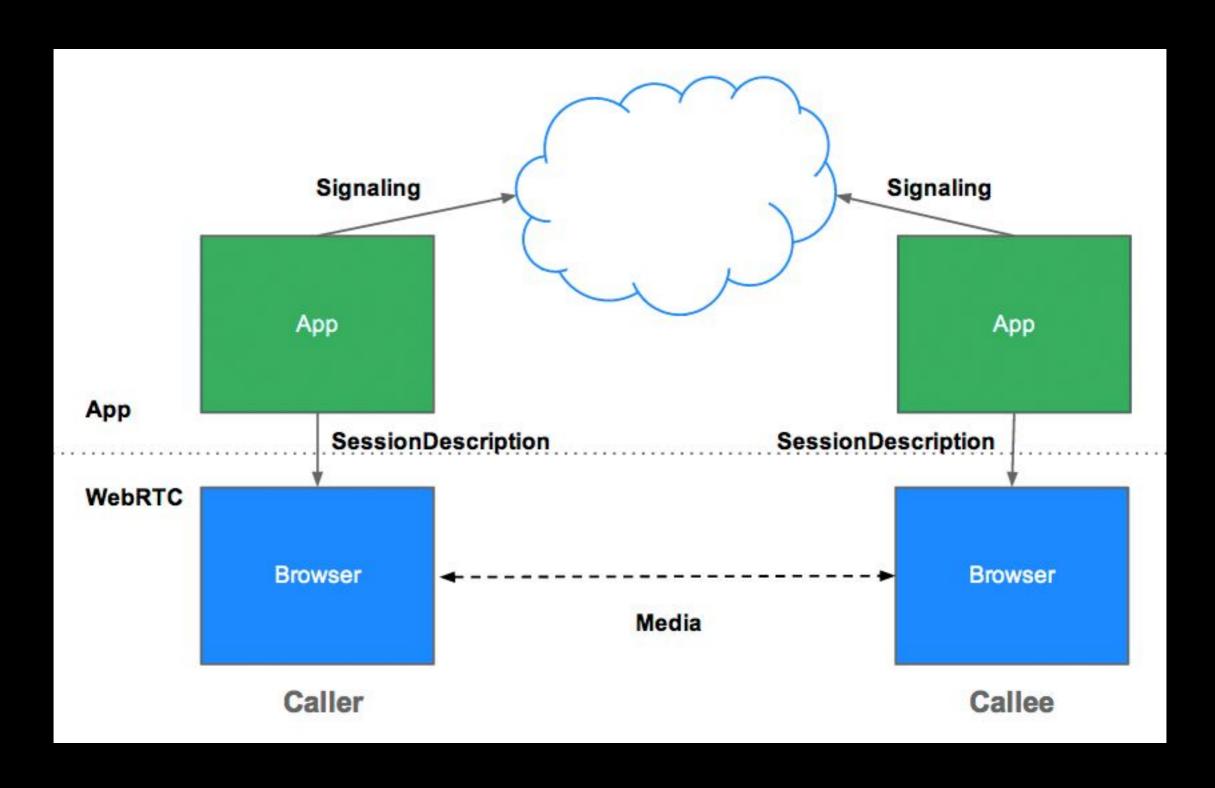
A Brief Overview



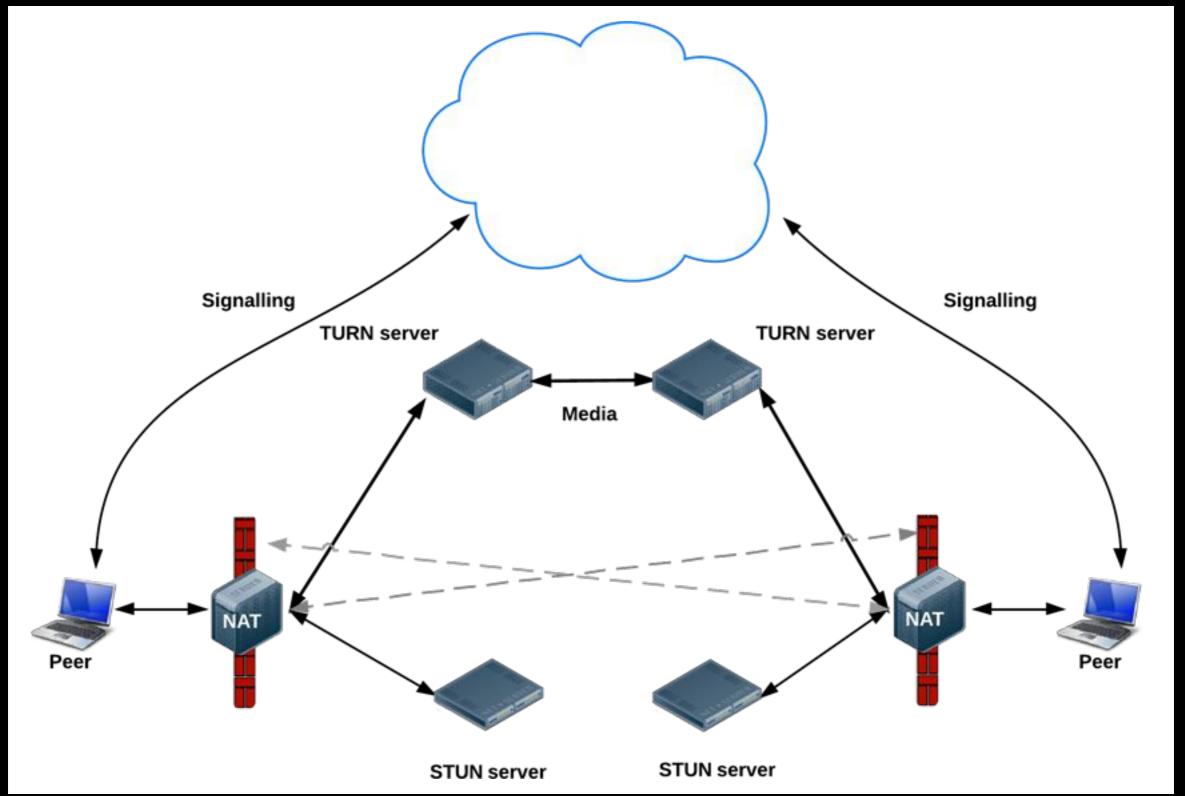
What is it?

- MediaStream (aka GetUserMedia)
- RTCPeerConnection
- RTCDataChannel

Ideal Diagram



Typical Diagram

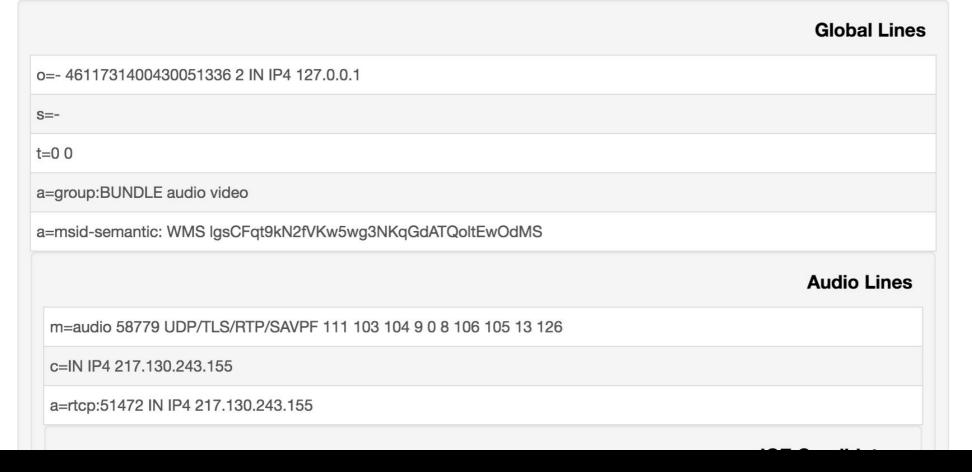


Detail of SDP



Anatomy of a WebRTC SDP

Behold the wonders and perils of a Session Description Protocol (SDP) generated by Chrome for WebRTC! See the source post by Antón Román for more background and commentary.

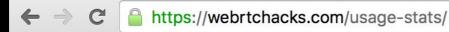


Global Lines

s=-

The s line contains a textual session name, which is not commonly used as you can see here.

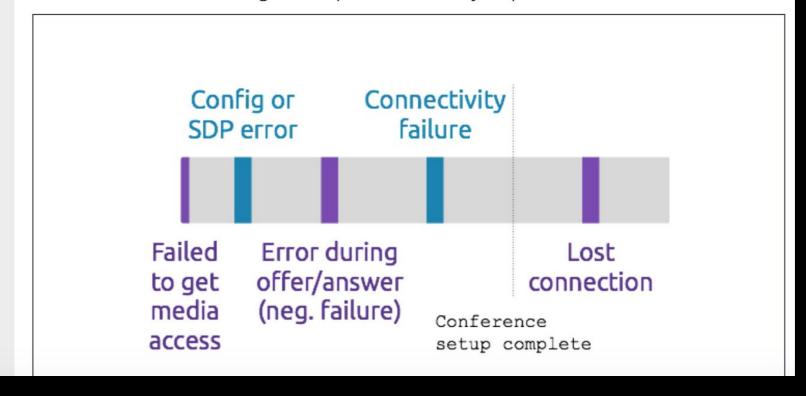
Pitfalls



Failures

The following are metrics as observed by callstats.io, these metrics are primarily gathered from the browser including Firefox and Chrome for Android.

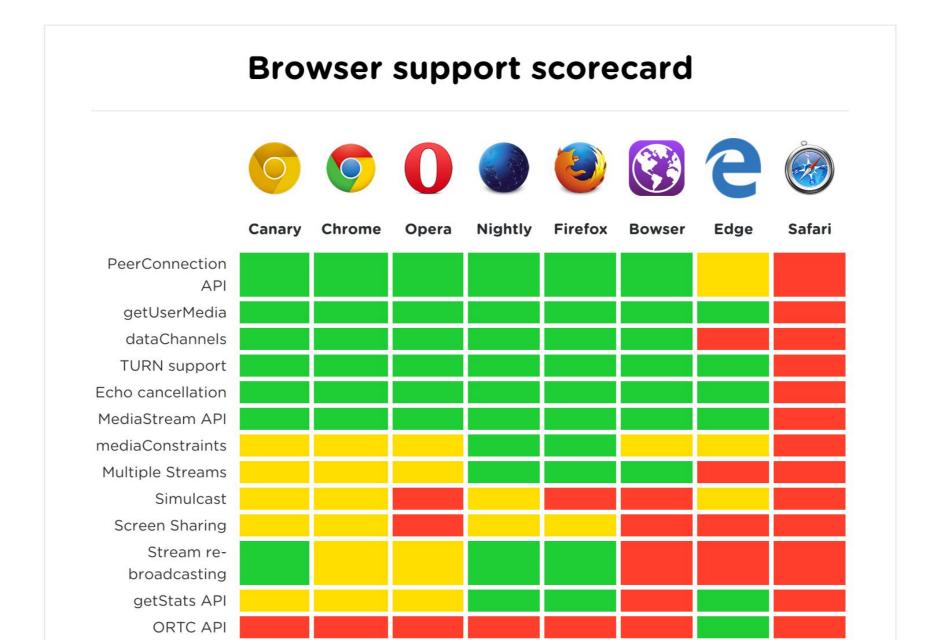
In the WebRTC services that we observe, the failure rate is on average 12%, i.e., 1 in 8 sessions are never set up. However, most failures (85%) come from the inability of an endpoint to traverse NATs or firewalls successfully. That problem can be usually fixed by deploying relay servers and we already observe that 22% of the conferences need some kind of TURN relay server. About 9% of the conferences required TCP, which may not be the best transport for media quality but is better than failing to establish the session. However, it would be remiss not to point out that not all our customers use a TURN/TCP or TURN/TLS. Ergo, the failure rate may still be indicative of not having the complete suite of relays in place.



Browser Support

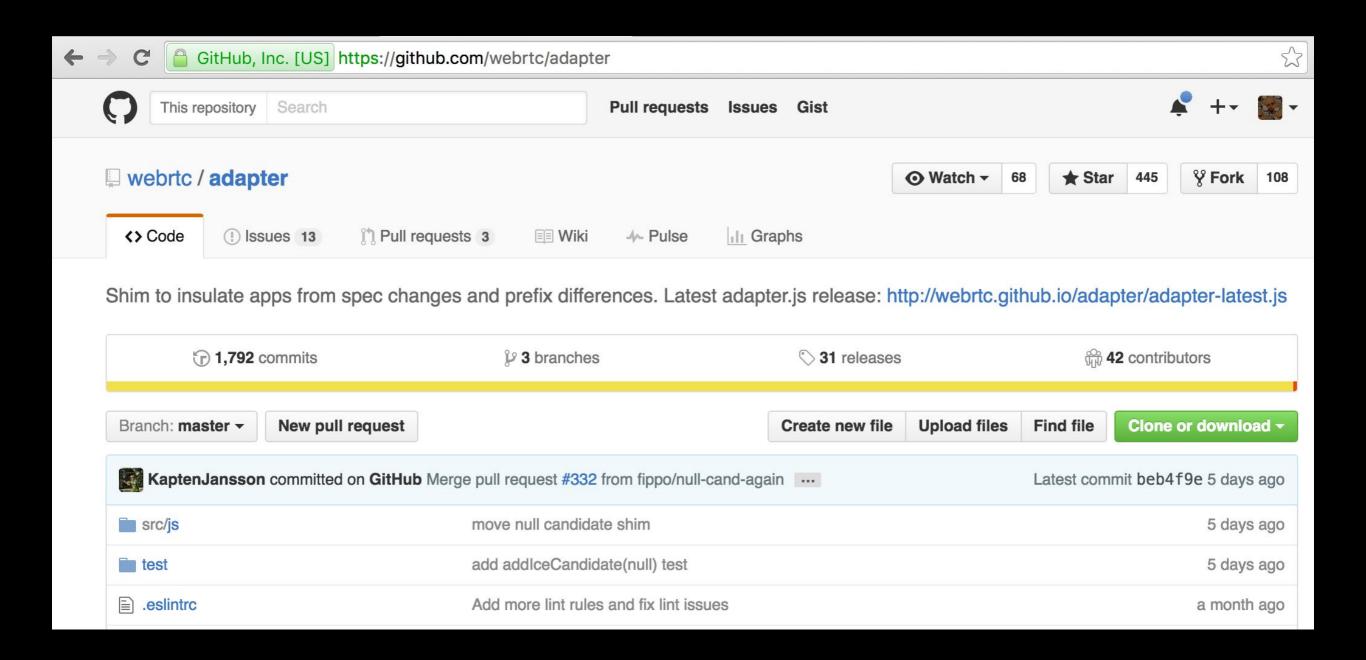


Is WebRTC ready yet?

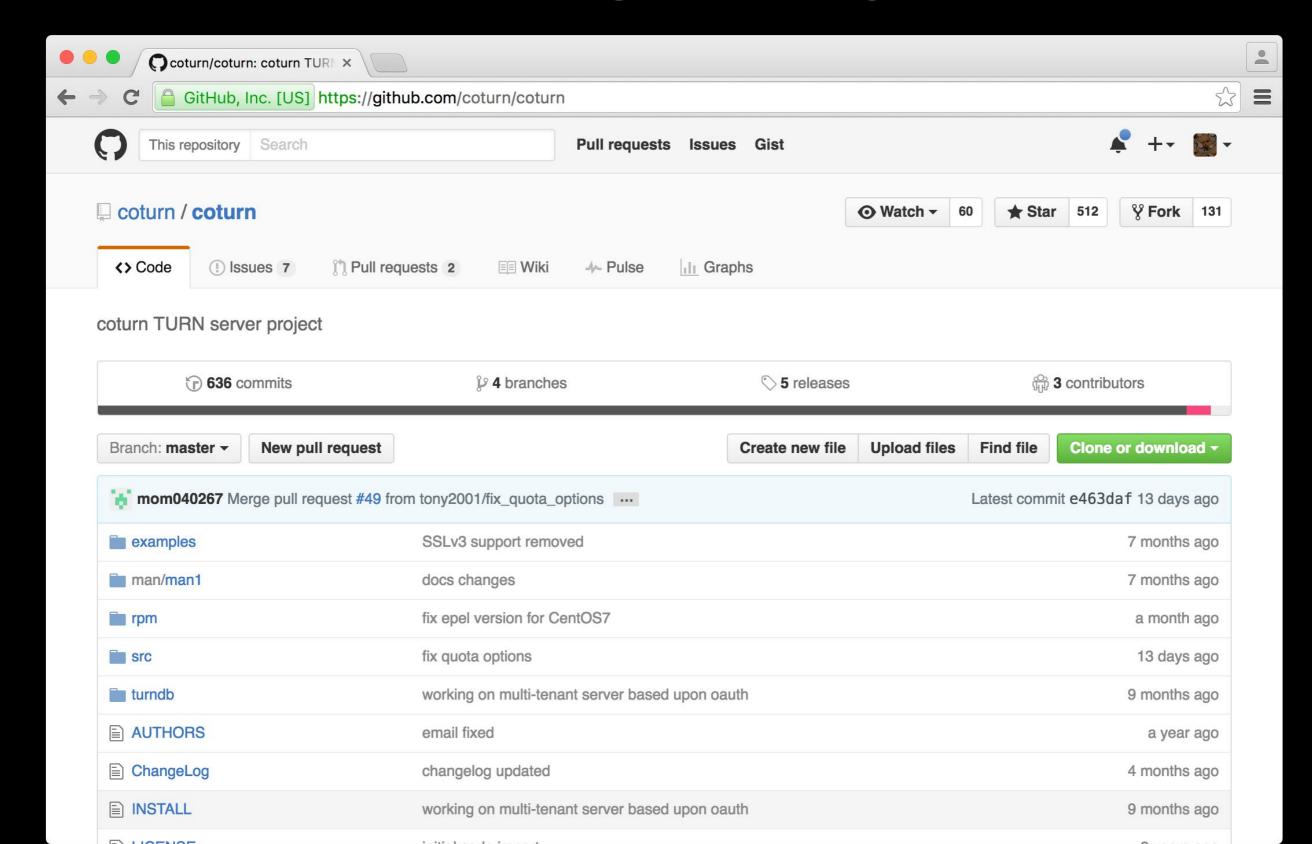


from &vet

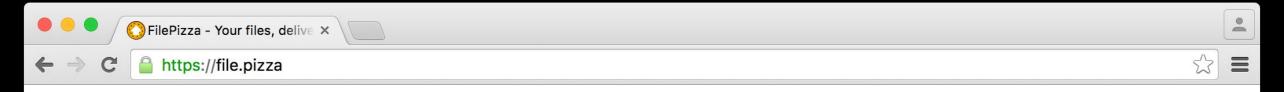
adapter.js shim



coturn TURN Server



Example





FilePizza

Free peer-to-peer file transfers in your browser. We never store anything. Files only served fresh.

select a file

Cooked up by Alex Kern & Neeraj Baid while eating Sliver @ UC Berkeley · FAQ · Fork us