WebRTC Performance Analytics

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WebRTC Measurement Infrastructure

Web Server A

Web Server B

JavaScript / HTML / CSS

cs.js lib

Web browser A

Real-time comm. function

Web browser B

Real-time comm. function

signaling

stats

stats

callstats.io infrastructure

media

 Other web browser APIs

WebRTC API

Infrastructure:
120-150 VPC

telemetry
data rate:
3-12 kbps
• the first cloud-based **monitoring** and **management service** for WebRTC (audio and video).
What to Measure?

• Network metrics
  • bits per second, RTT, jitter, packet losses, ...

• Multimedia pipeline metrics
  • playout delay, frames metrics, ...
  • Quality Models from metrics

• Annoyances
  • resolution/frame rate changes, interface changes, ...
  • failures (NATs, insufficient capacity for carrying media, ...)
  • user feedback
Disruptions

**Disruption**: loss of connectivity when network interfaces change, low available capacity, or high delay

The light grey vertical lines show disruption, highlighted by the red bounding boxes.
Disruptions and user behaviour

User Behaviour: The user tries to correct for the disruption by turning on and off video
state of webrtc

(as we observe it)
Multiparty calls

- Number of participants:
  - 1 participant: 11%
  - 2 participants: 45%
  - >3 participants: 44%

IPv6?

- IPv4 only: 72%
- IPv6 only: 2%
- IPv4+v6: 26%
How many ICE candidates?

- Host
- Stun
- Turn
- V4
- V6
- Multi homed

10%
51%
35%
4%
24% of the calls goes through a TURN Relay
Types of Network Relays

- 76% No Relay
- 16% TURN/UDP
- 7% TURN/TCP
- 1% TURN/TLS

8% of the media goes over TCP
4% of the calls fail to set up

Note: traffic doubled in the same period
Failure Reasons

<1%  85%  <1%
Config or SDP error  Connectivity failure

Failed to get media access  Error during offer/answer (neg. failure)  Lost connection

Conference setup complete

14%  <1%
12% calls drop after setup
Dropped calls

- Connectivity lost: 49%
- High delay: 12%
- High loss: 17%
- Battery crash: 1%
Round Trip Times

95th percentile RTTs of each participant in each session.
Summary

- **Participants**: ~2 participants
- **Relays**: ~25% sessions need a TURN server
- **Setup time**: 80% sessions setup in <5s
- **Call Setup Failures**: ~4% of calls fail to setup
- **Reason for failure**: 85% due to NAT/FW
- **Call Drop**: 12% calls fail after setup