WebRTC Performance Analytics

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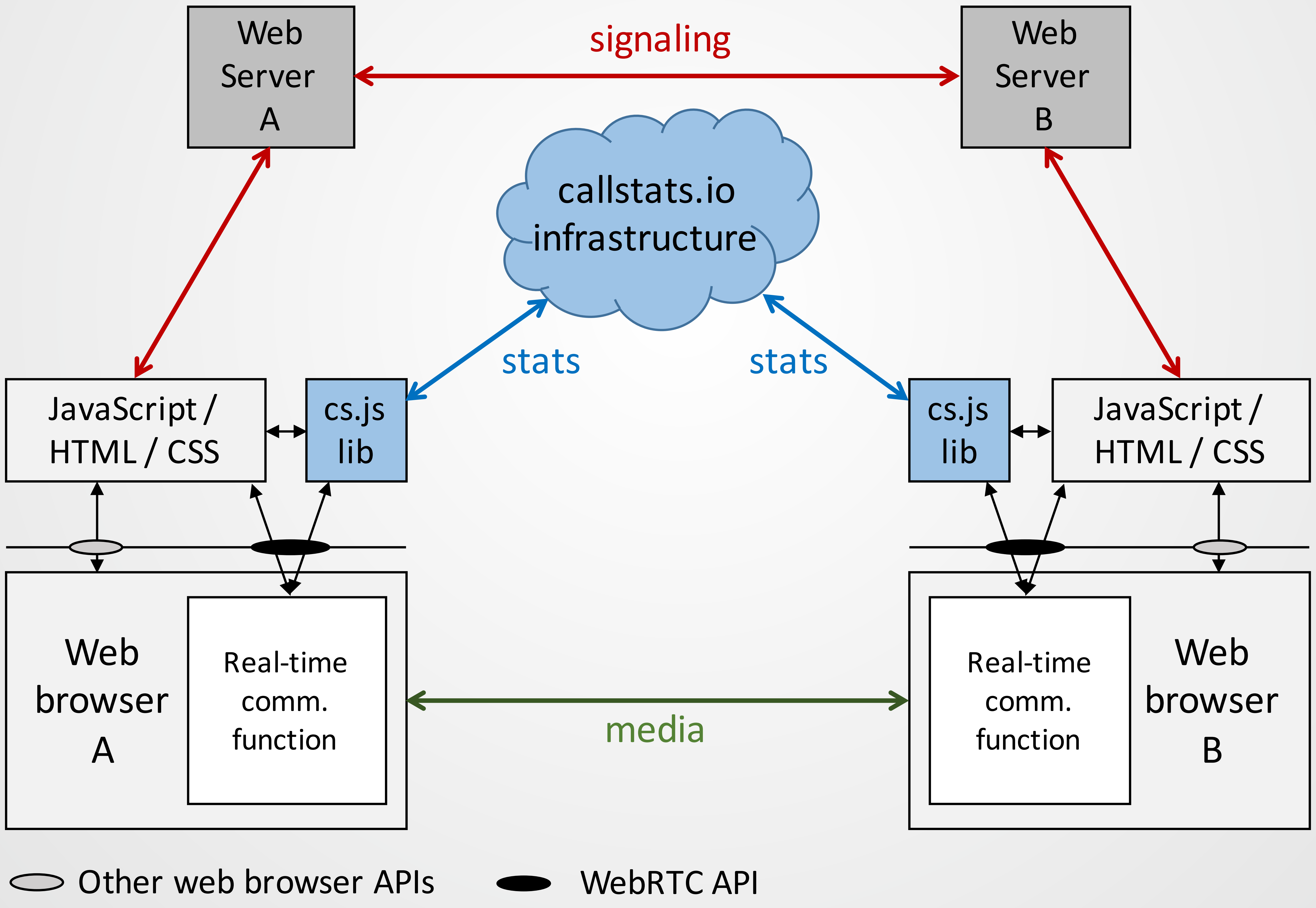
WebRTC Measurement Infrastructure

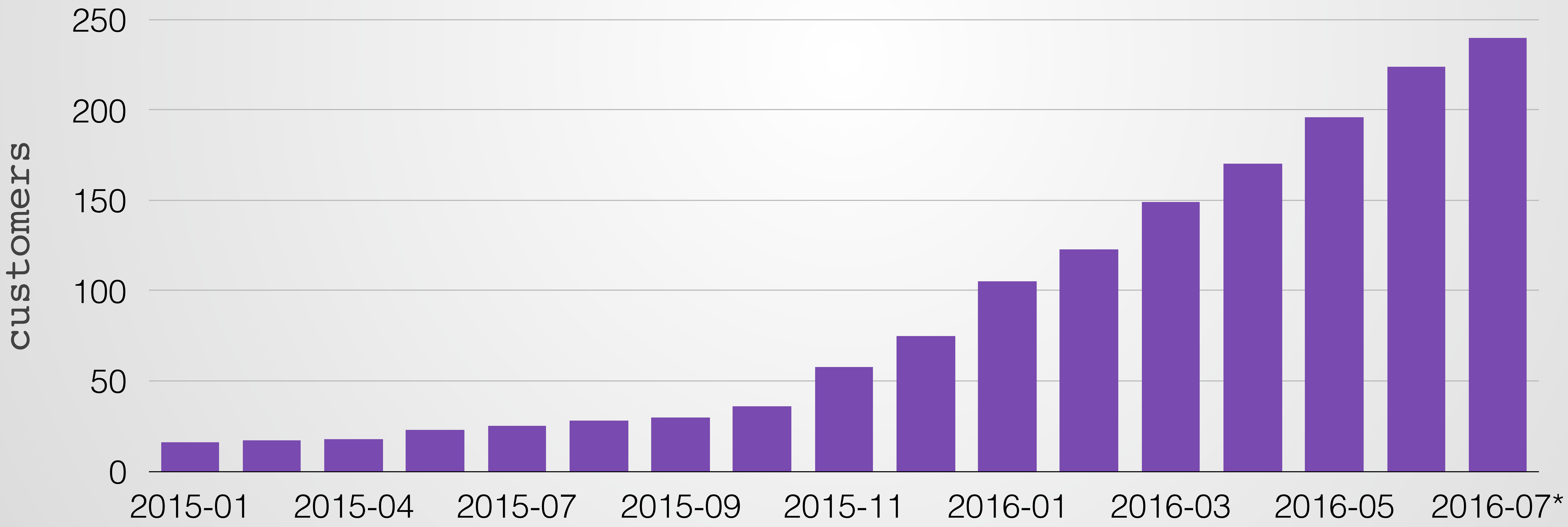


Infrastructure:
120-150 VPC

telemetry
data rate:
3-12 kbps

callstats.io

- the first cloud-based **monitoring** and **management service** for WebRTC (audio and video).



What to Measure?

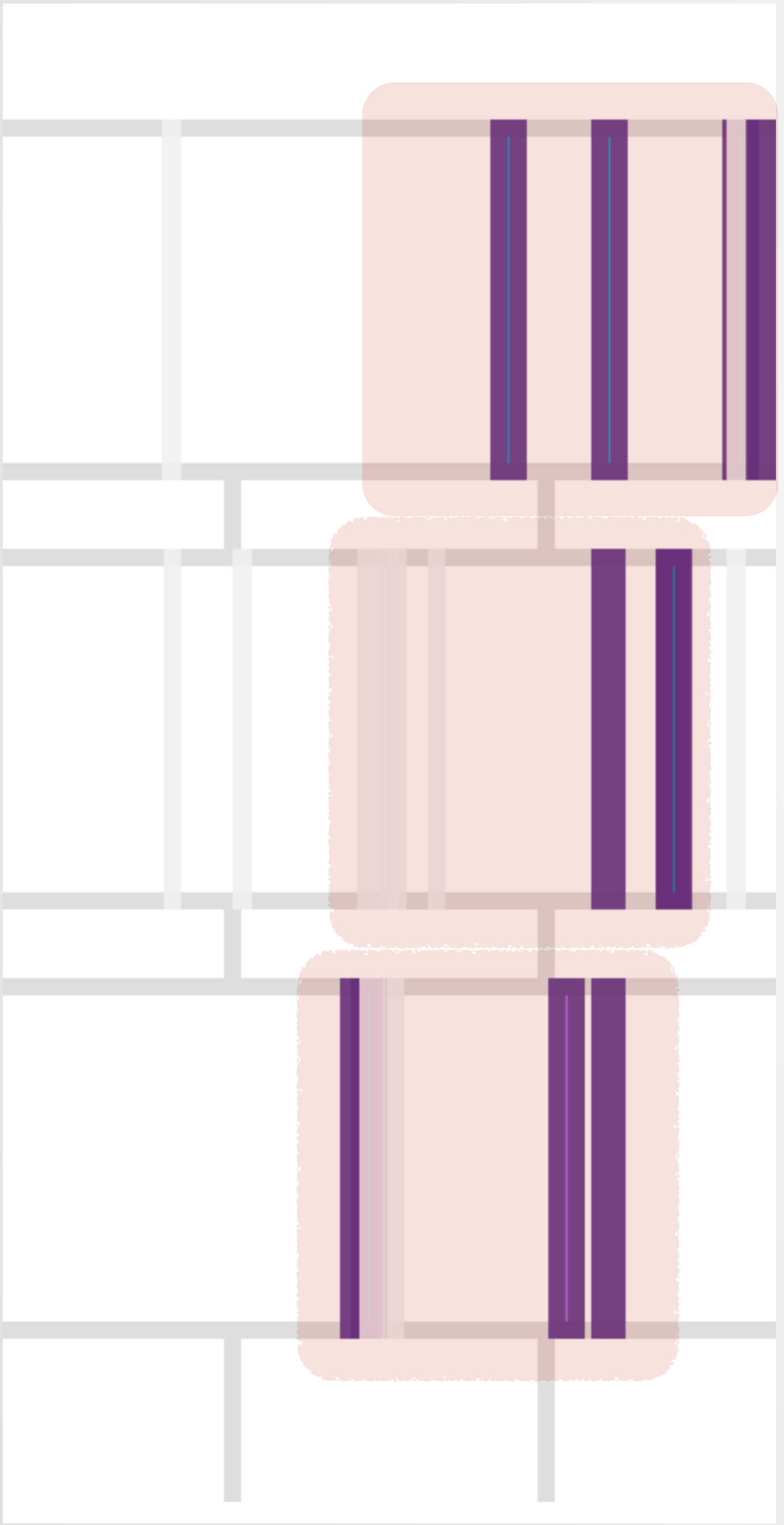
- Network metrics
  - bits per second, RTT, jitter, packet losses, …
- Multimedia pipeline metrics
  - playout delay, frames metrics, …
  - Quality Models from metrics
- Annoyances
  - resolution/frame rate changes, interface changes, …
  - failures (NATs, insufficient capacity for carrying media, …)
  - 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

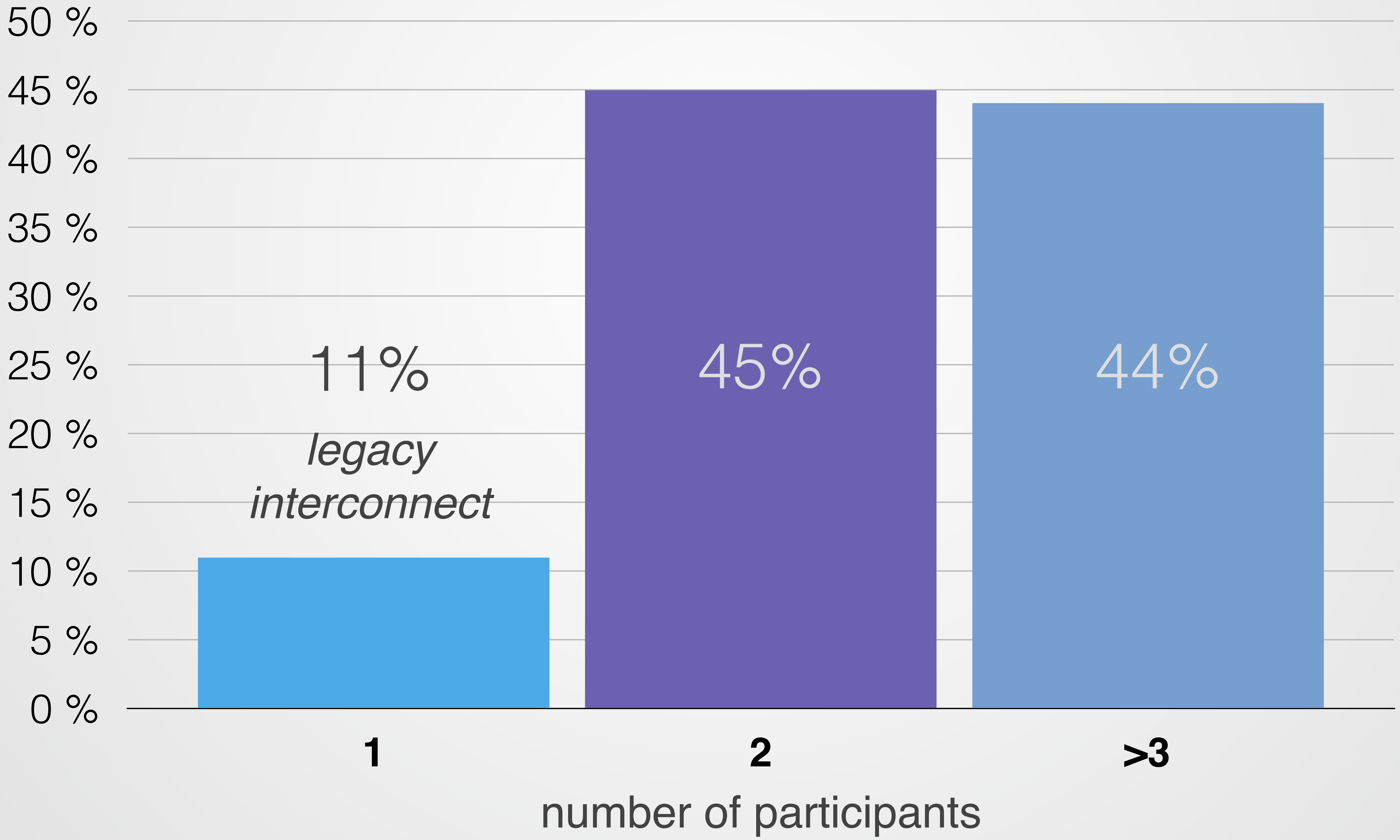


User Behaviour: *The user tries to correct for the disruption by turning on and off video*

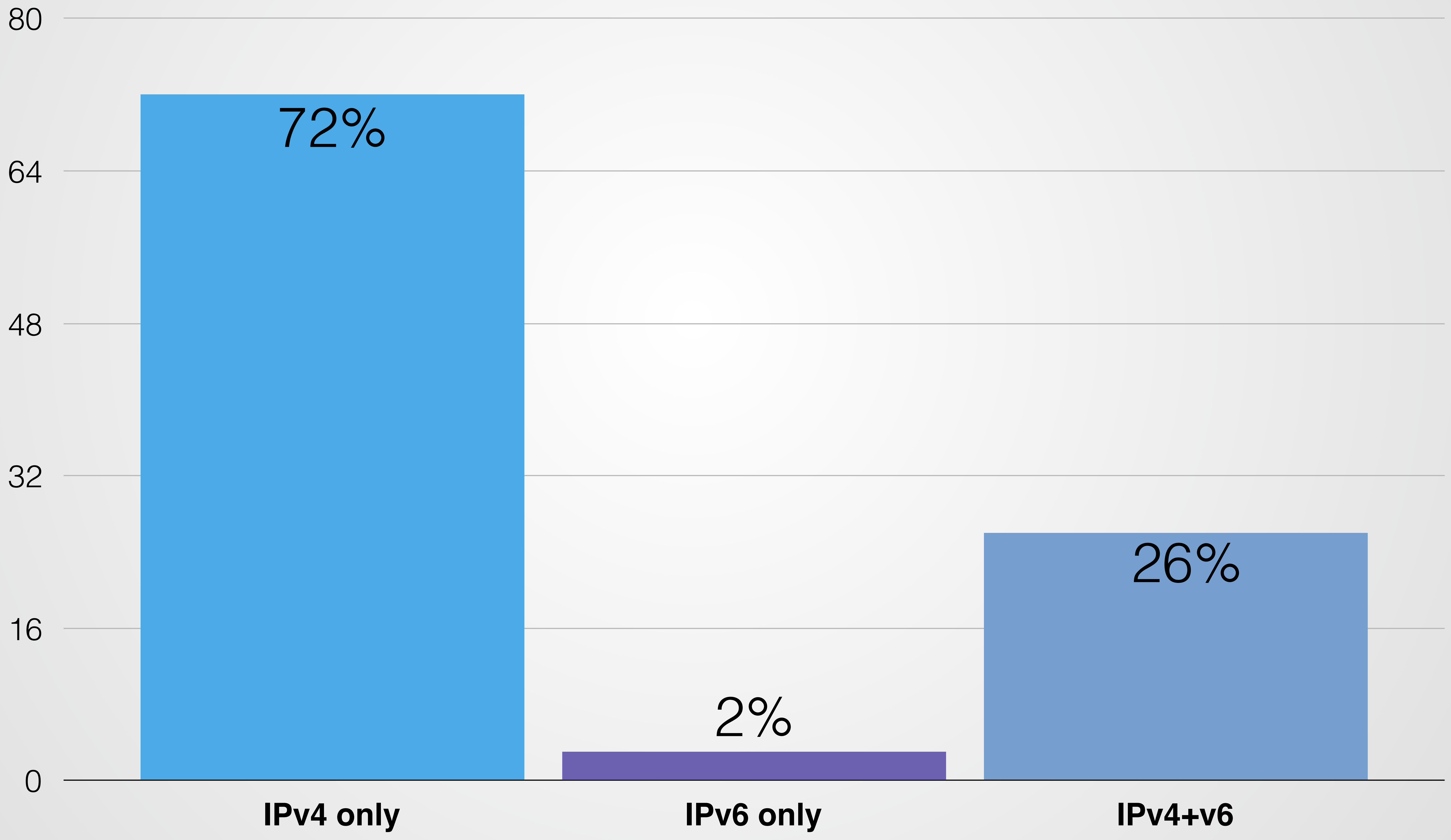
state of webrtc

(as we observe it)

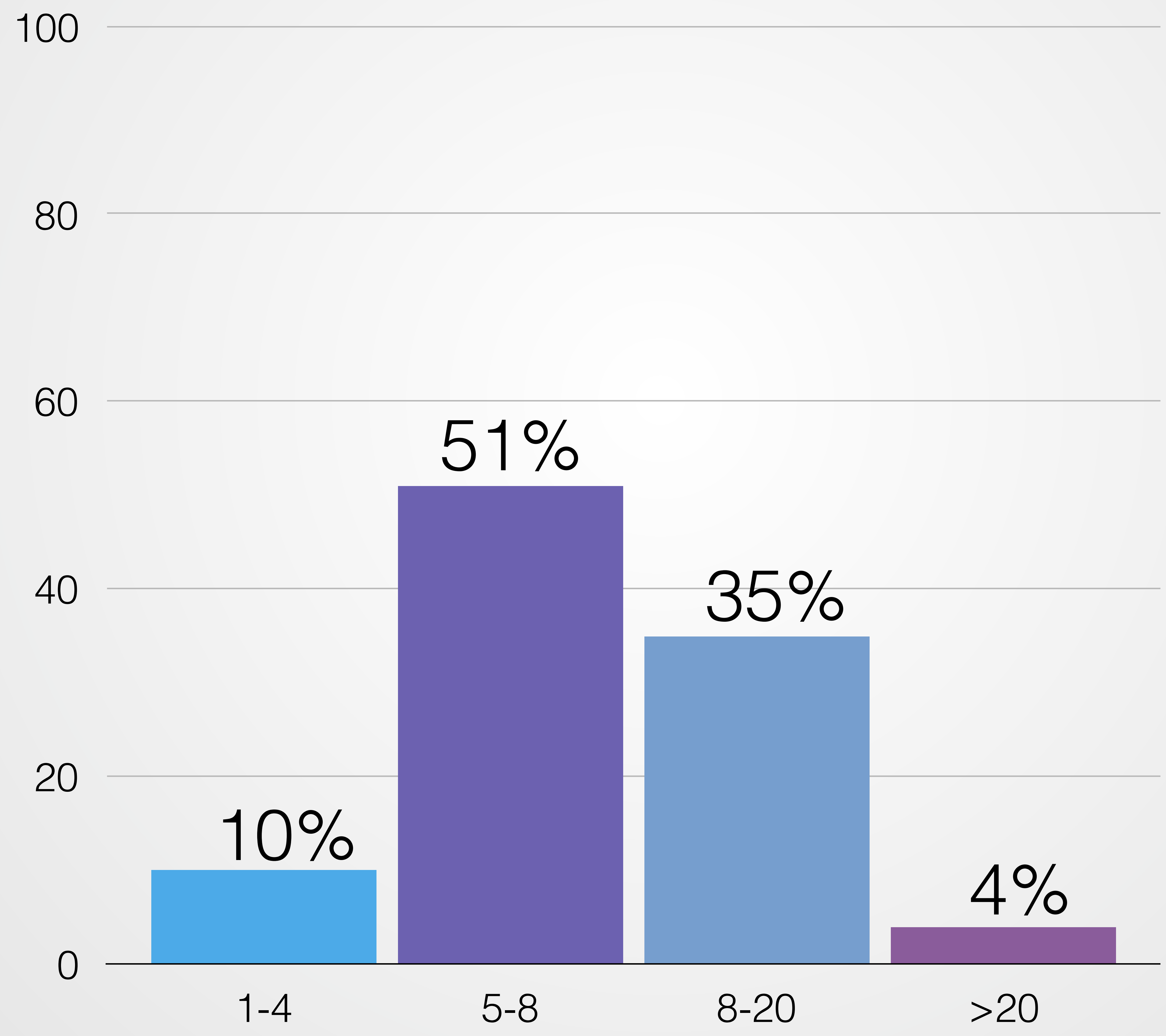
Multiparty calls



IPv6?



How many ICE candidates?



host
stun
turn

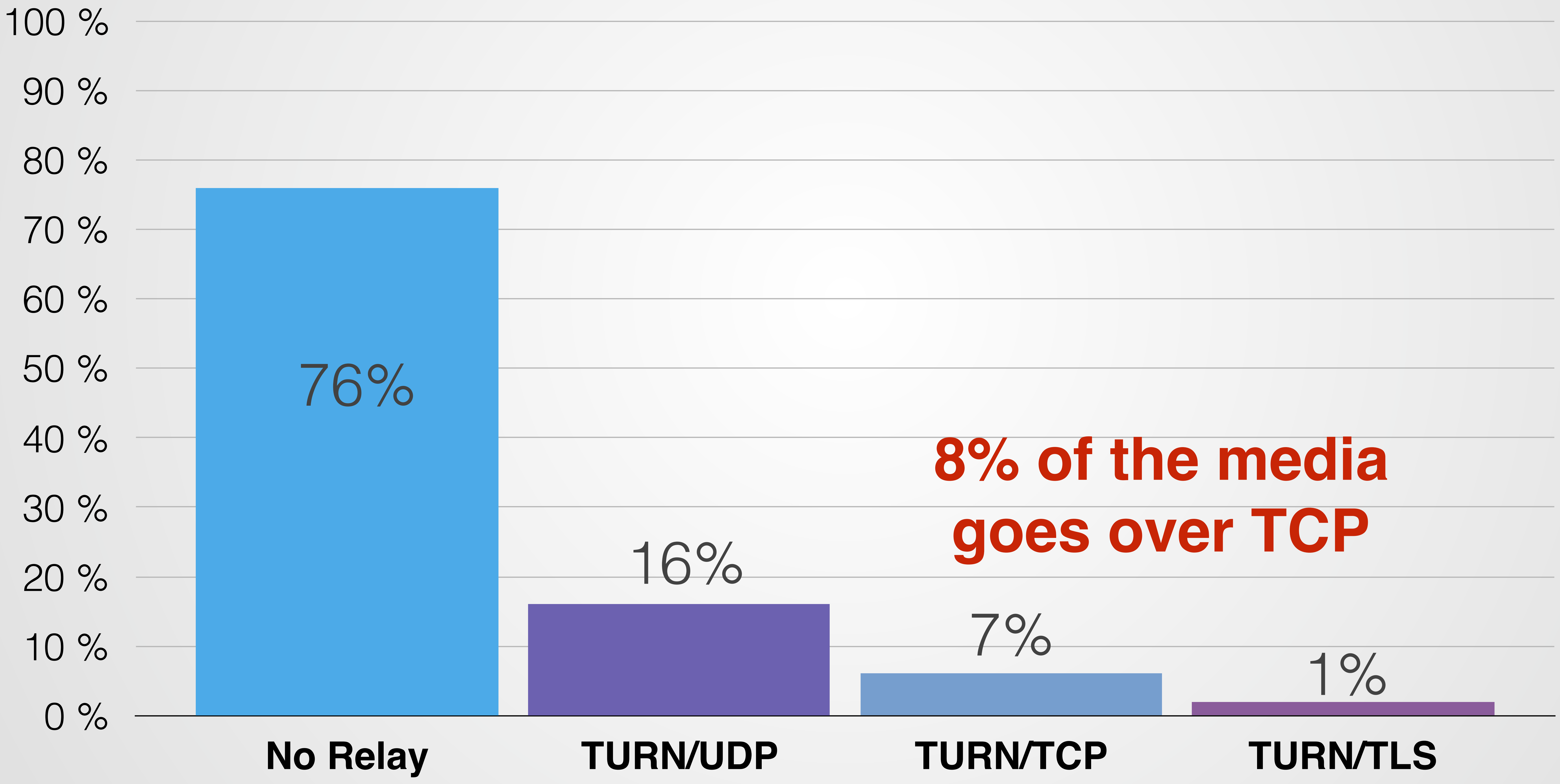
v4
v6

multi homed

24%

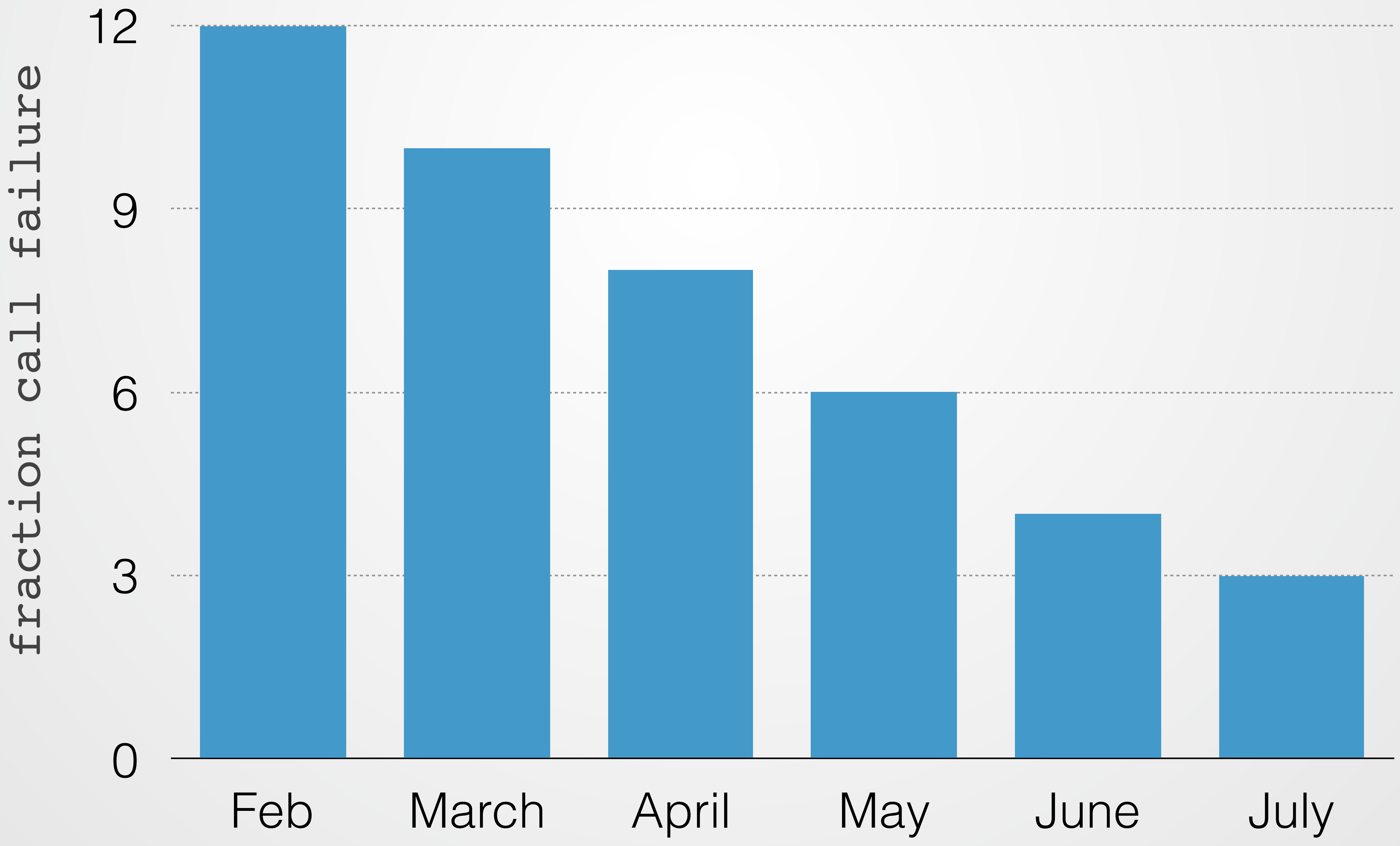
of the calls
goes through a
TURN Relay

Types of Network Relays



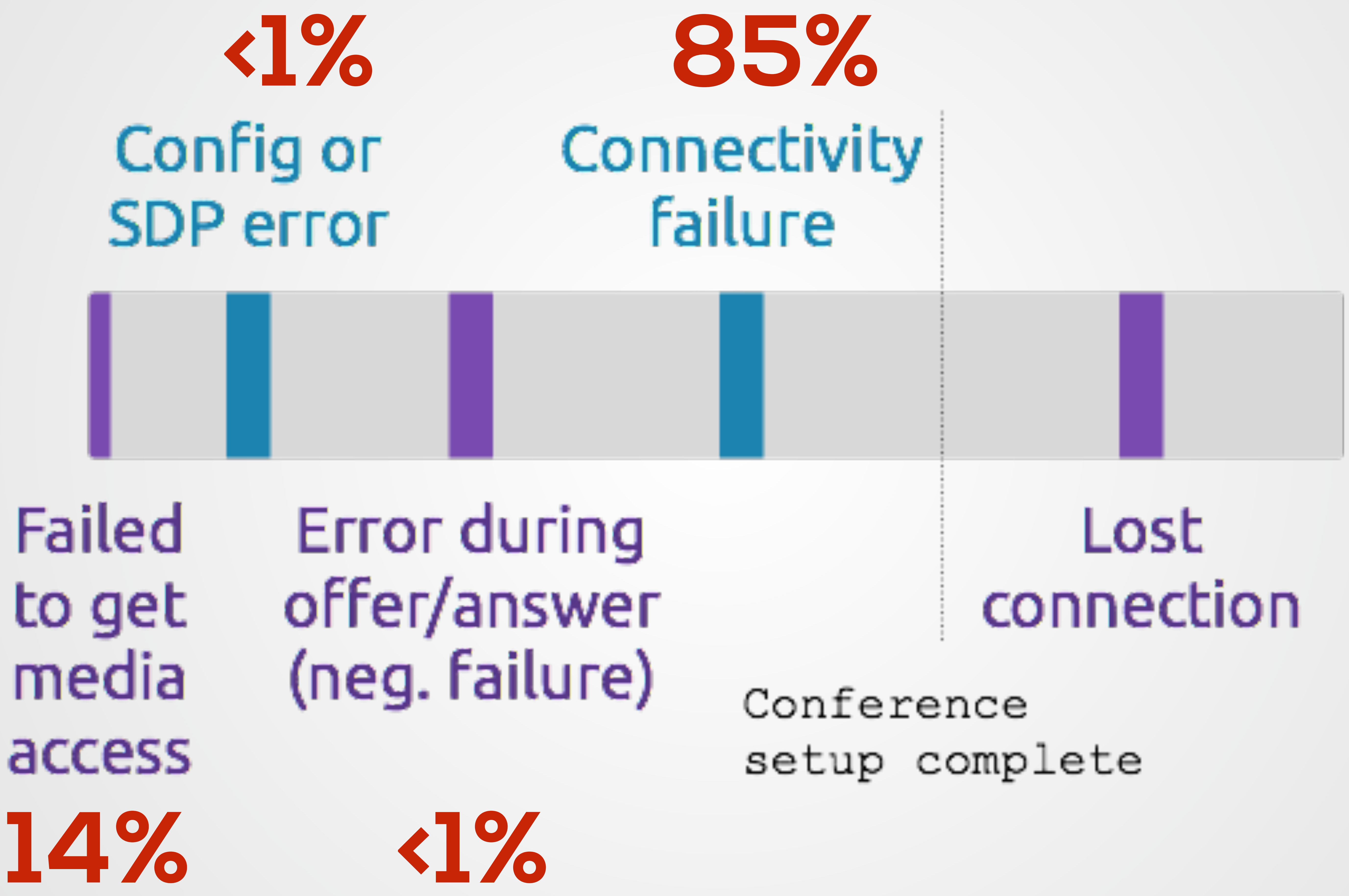
4%

of the calls fail to set up



Note: traffic doubled in the same period

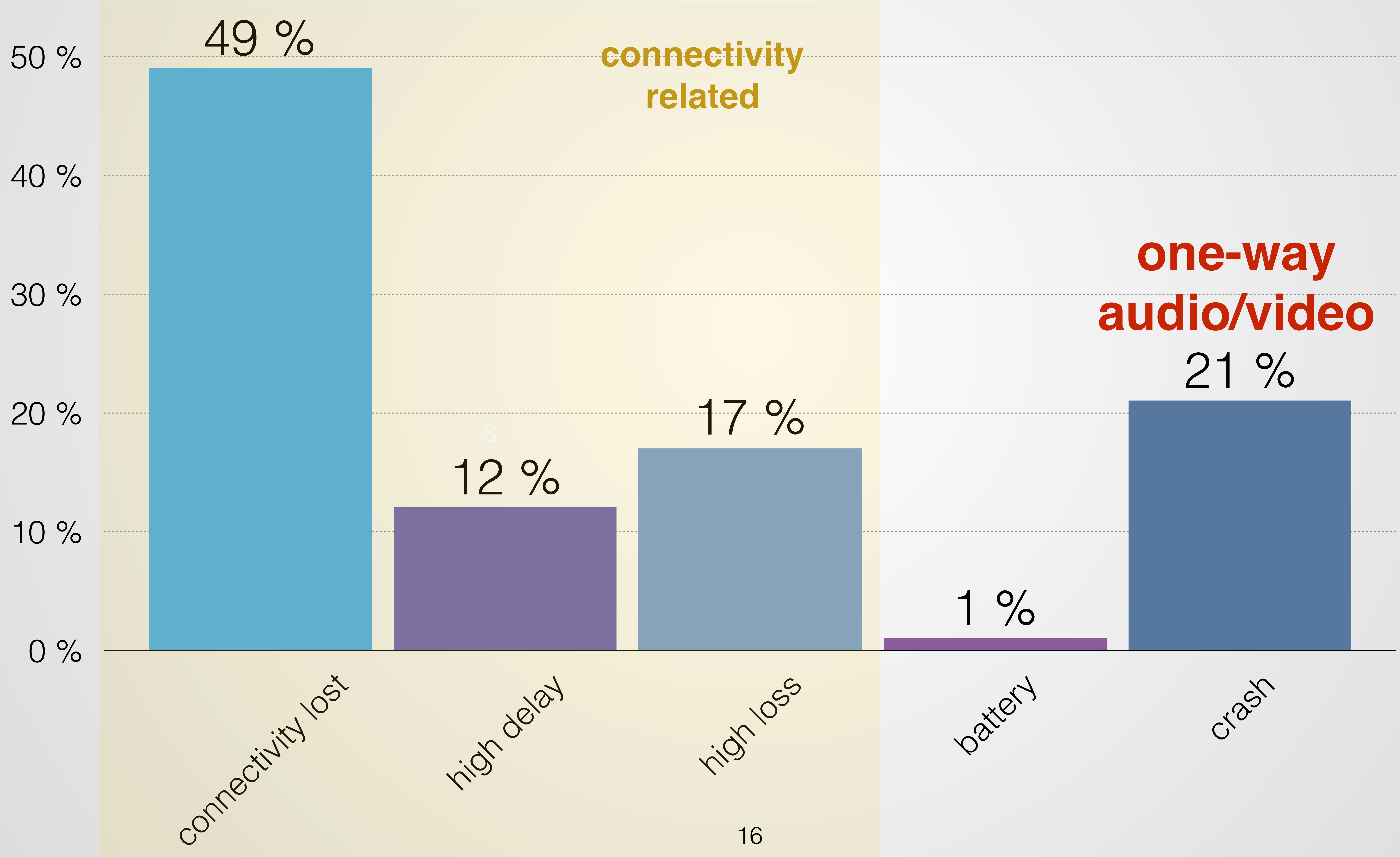
Failure Reasons



12%

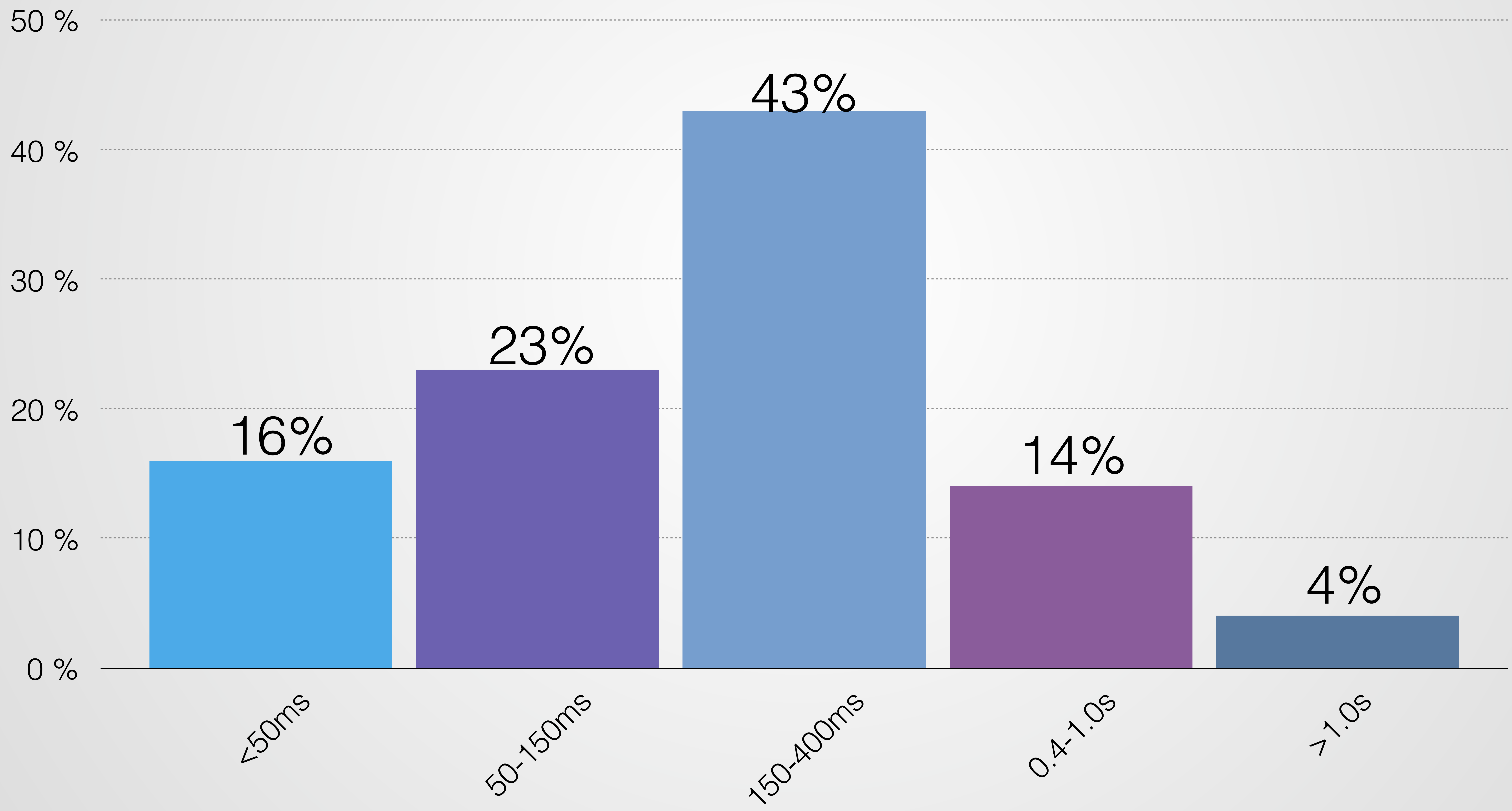
calls drop
after setup

Dropped calls



Round Trip Times

***95percentile** 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