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Multimedia Transport Protocols for WebRTC

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What is WebRTC?

A framework for browser-based real-time conferencing

Includes network, audio, and video components used in voice and video chat

Accessed through Javascript API to support custom applications
(e.g., implement Google hangouts as native HTML5 application)

IETF®

W3C®

Signalling infrastructure

IETF

W3C

Real-time multimedia transport

NAT traversal

Peer-to-peer data

Signalling infrastructure

Identity provision

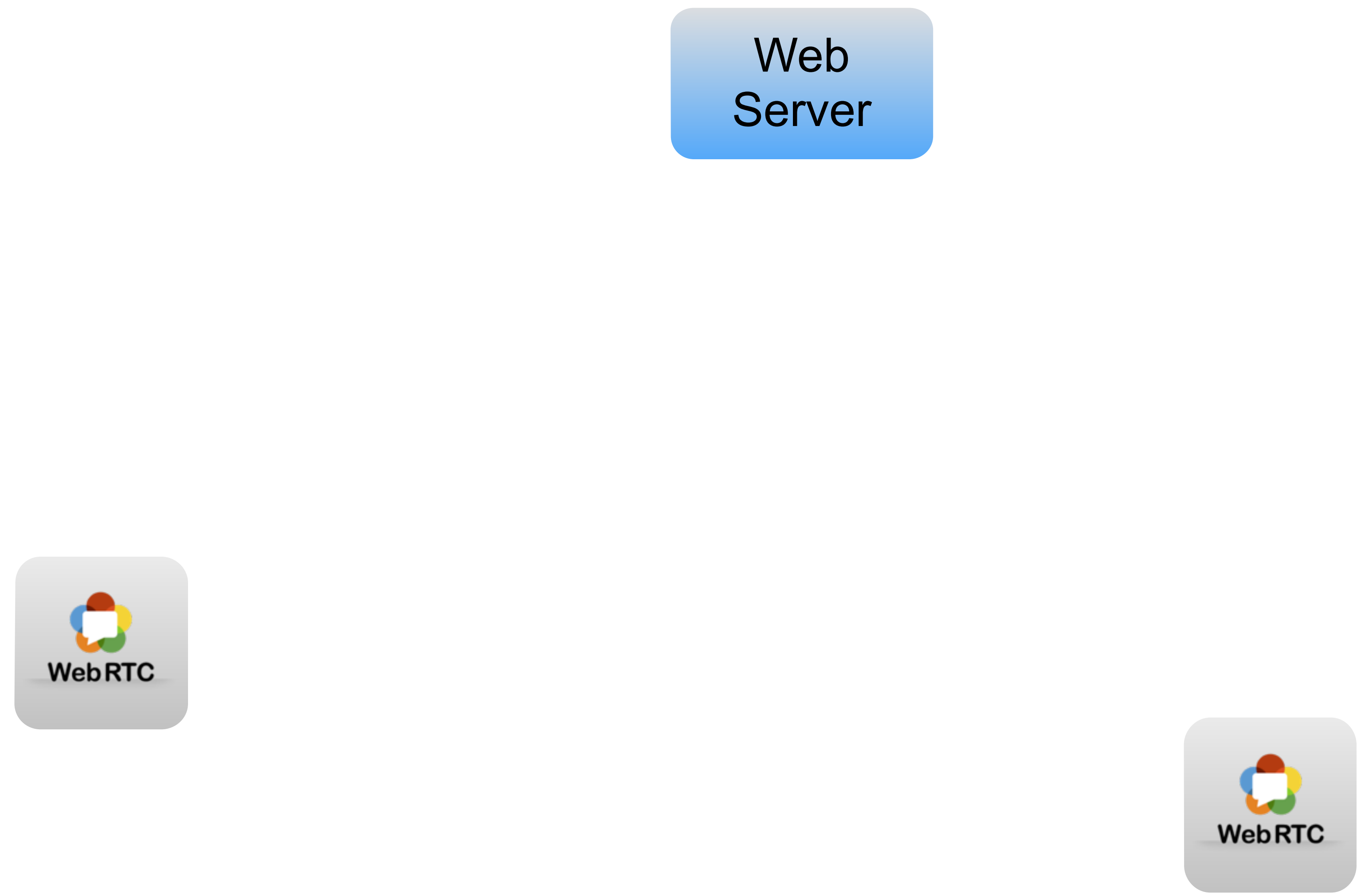
Browser extensions for media capture, encoding, and playout

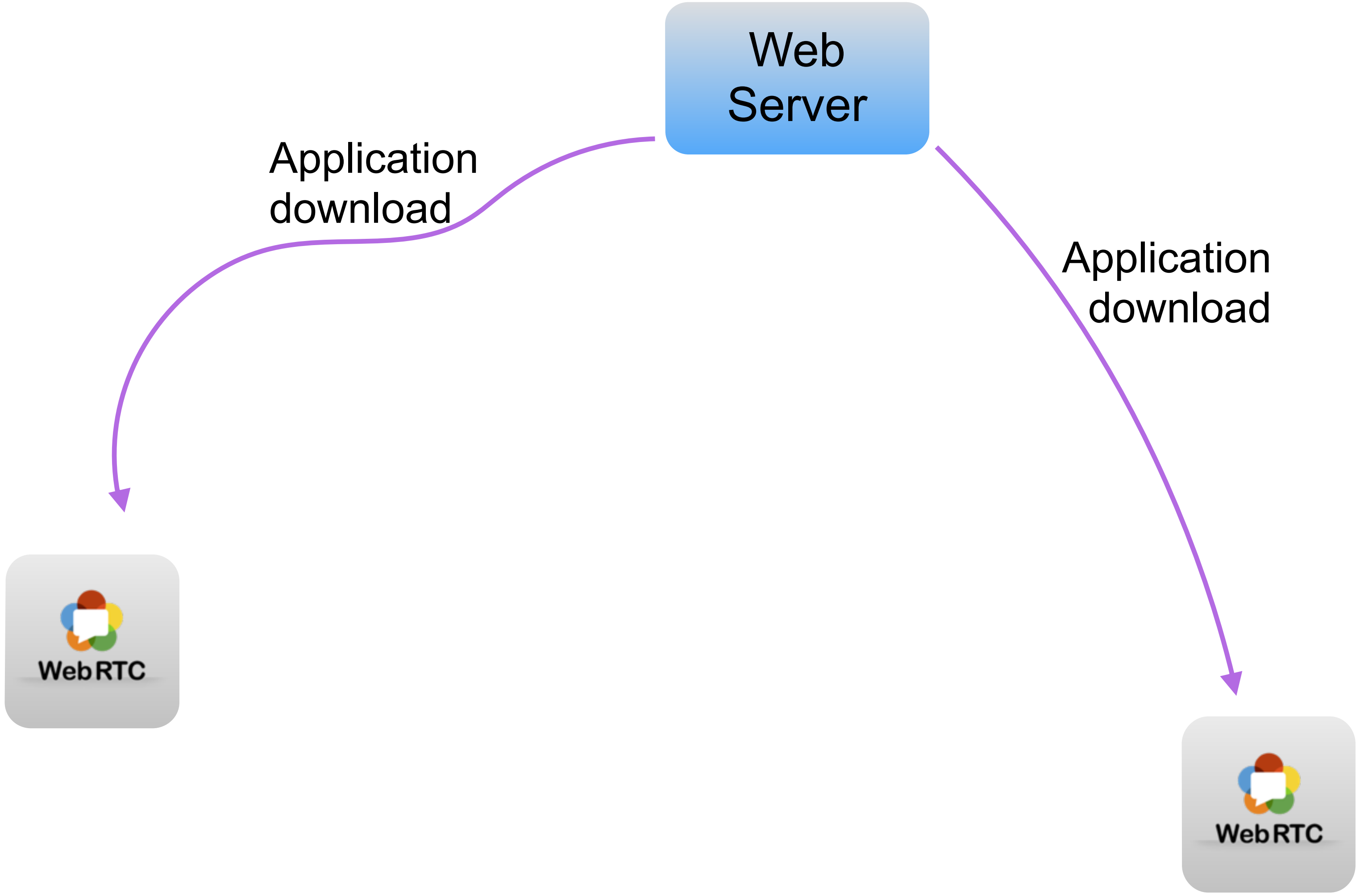
Real-time multimedia transport

NAT traversal

Peer-to-peer data

JavaScript APIs


Web Server
WebRTC
WebRTC


Web Server
Application download
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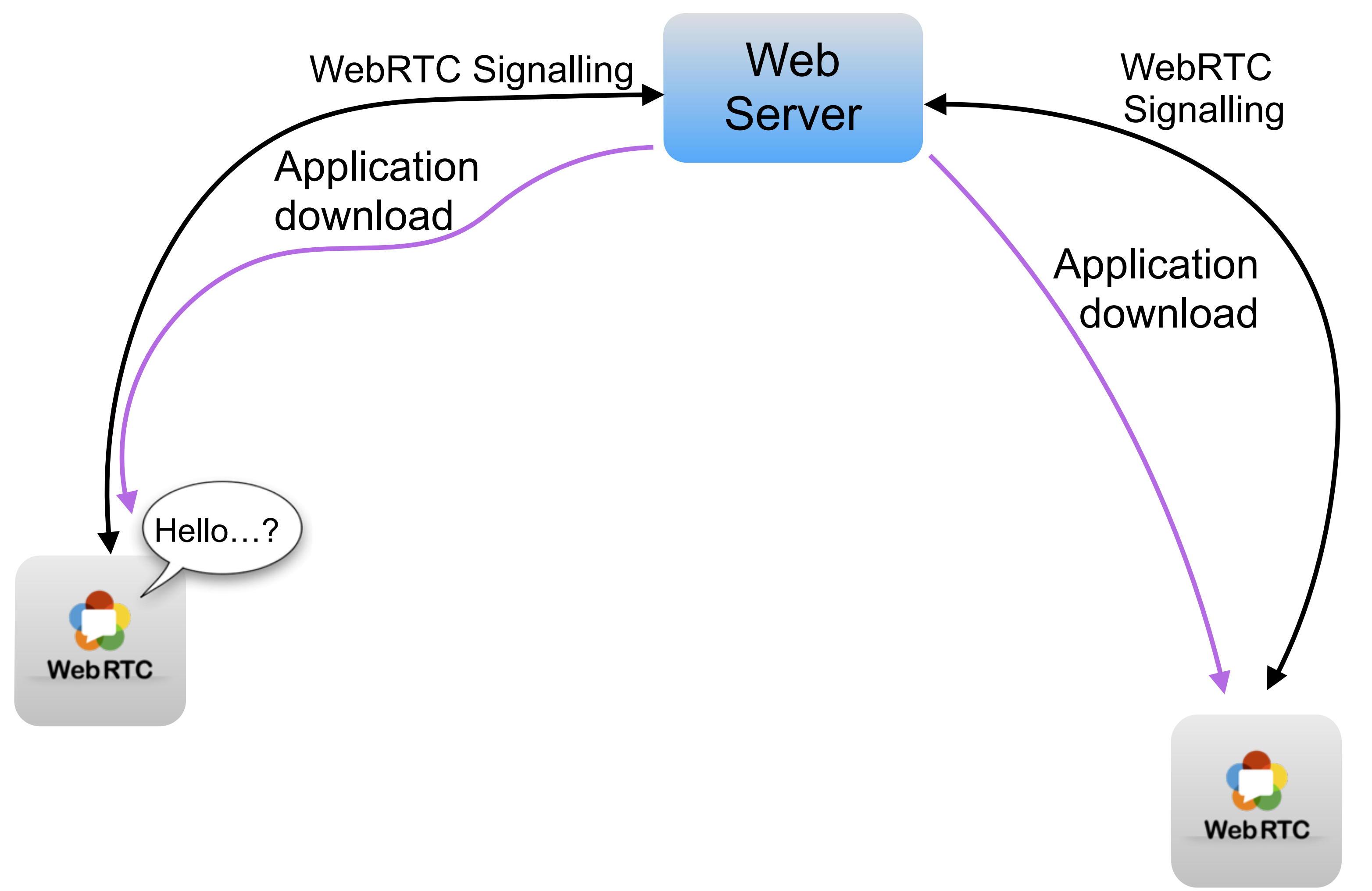
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Hello…?
WebRTC
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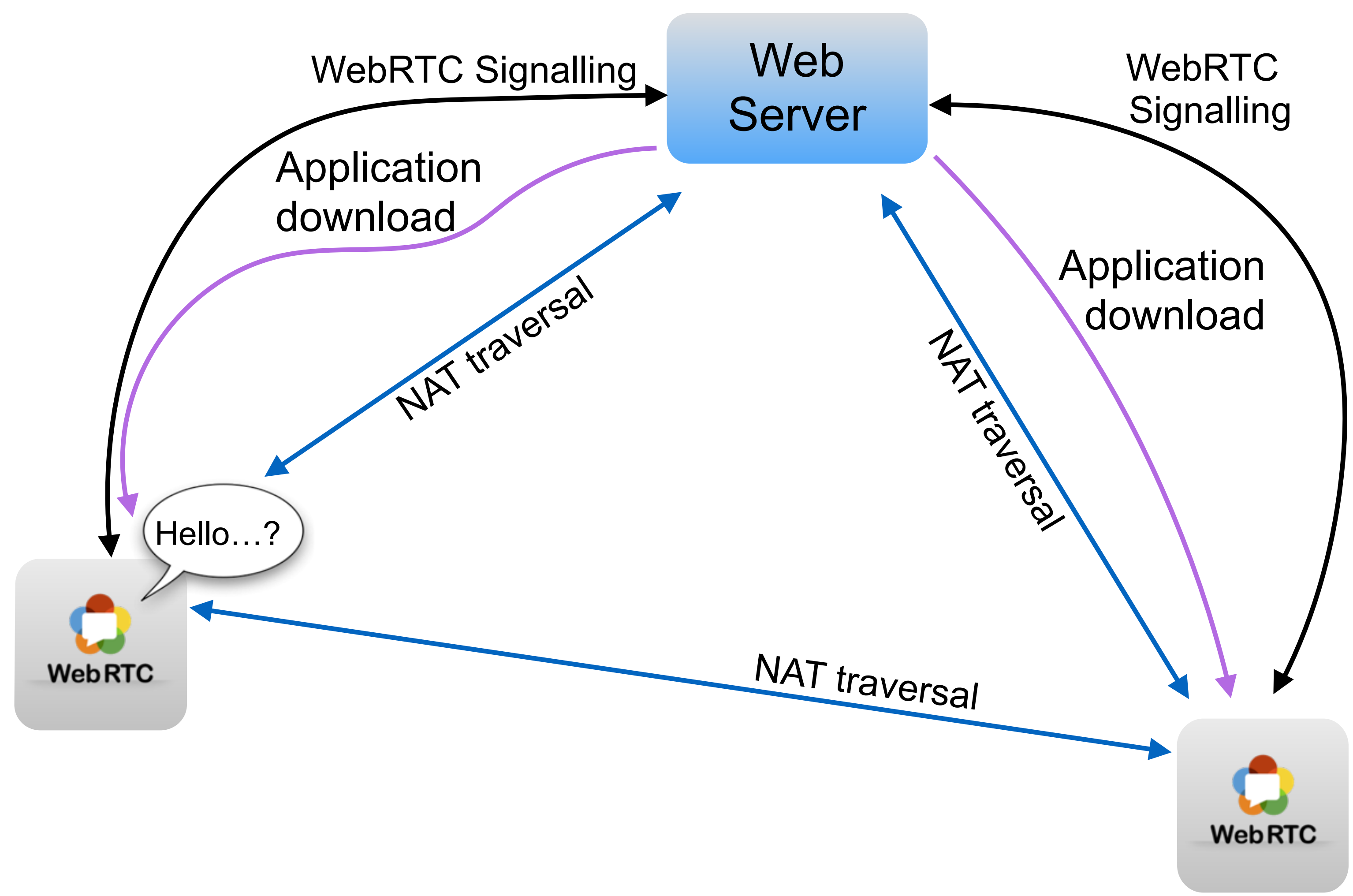
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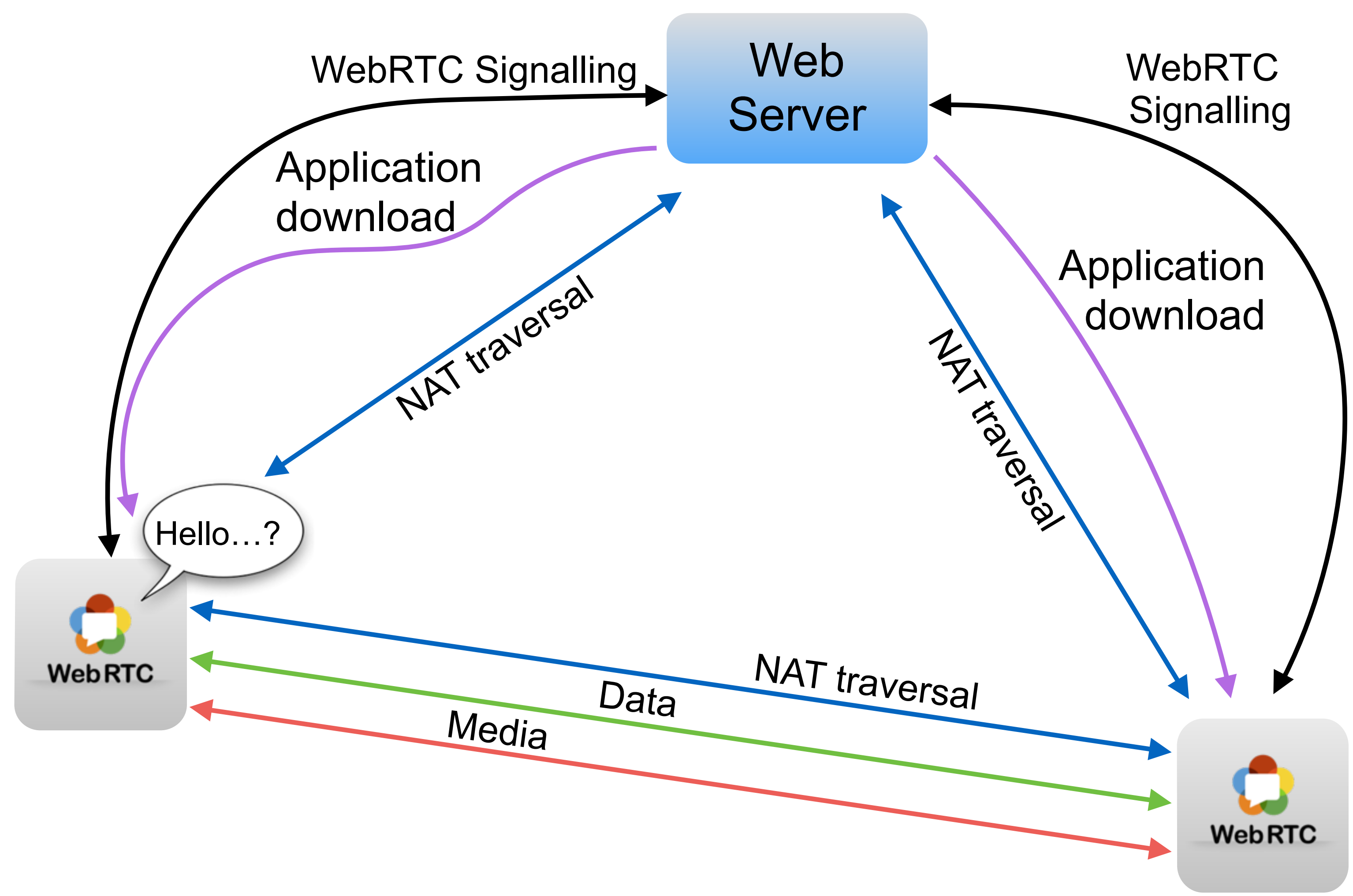
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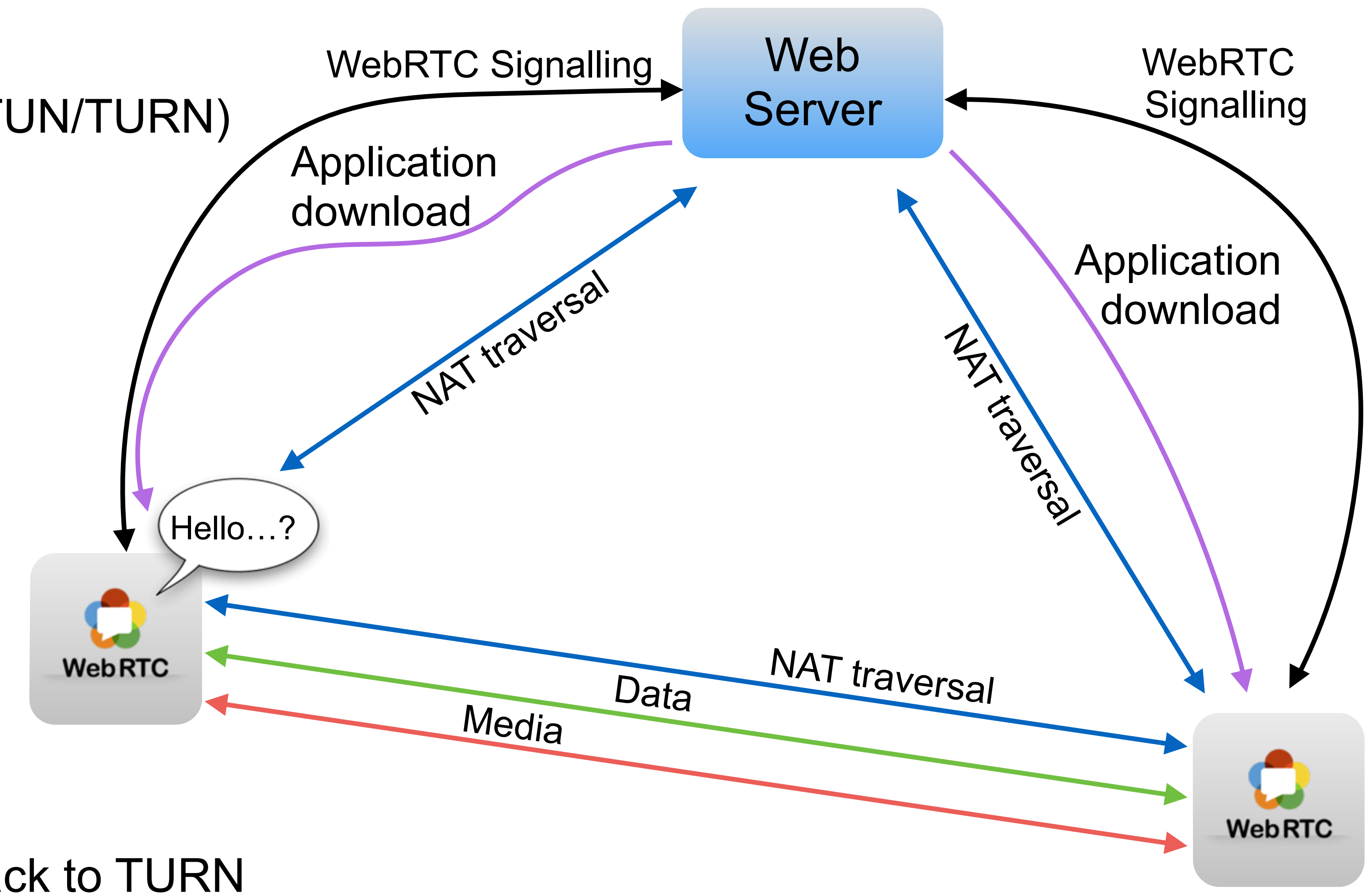
Hello…?
WebRTC
WebRTC
NAT traversal
Data
Media

Application provider

- JavaScript application delivered from web server
- Identity provider
- NAT traversal infrastructure (STUN/TURN)

Signalling

- Application protocol defined via Javascript API
- Offer-answer exchange of SDP (JSEP)



Media and data transport

- NAT traversal: STUN with fallback to TURN
- Media transport using secure RTP over UDP
- Secure peer-to-peer data using SCTP over DTLS/UDP

Benefits

Standard infrastructure requirements
Standard browser API
Flexible application support
Modern media codecs and transport protocols
Peer-to-peer data channel

Challenges

API complexity vs completeness/control
NAT traversal performance
Security and identity
Media transport and congestion control

Benefits

Standard infrastructure requirements
Standard browser API
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Media transport and congestion control

What is the problem?
Why not TCP?
What are the challenges?
Directions and solutions

Media transport and congestion control

What is the problem?
Why not TCP?
What are the challenges?
Directions and solutions

Real-time media transport that is safely deployable and high-performance

Baseline RTP media transport is well defined

- but will be deployed at very large scale
- using modern high-rate video codecs
- with no professional network support

Concern about potential network congestion

- applications can use significant bandwidth, are trivial to deploy, and difficult to control
- no appropriate congestion control algorithms

Media transport and congestion control

What is the problem?

Why not TCP?

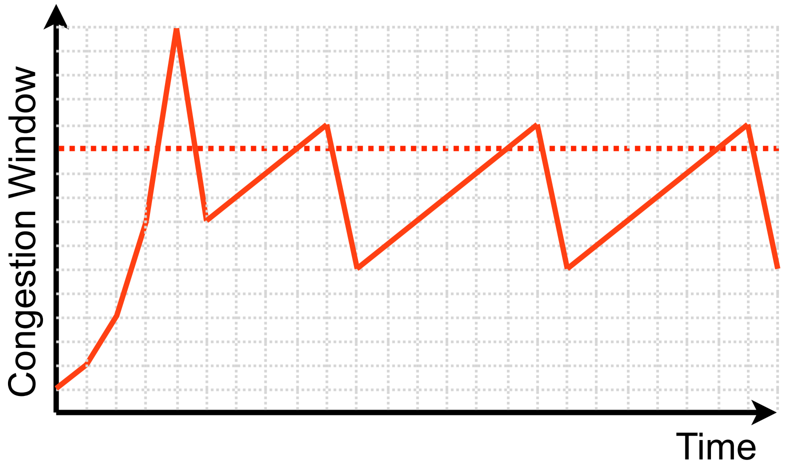
What are the challenges?

Directions and solutions

Video streaming uses TCP – but high latency, and poorly suited for interactive real-time applications

TCP congestion control algorithm causes latency

- Loss-driven – relies on queue overflow
- Needs buffer to smooth abrupt changes in rate and match codec output



Retransmission with head-of-line blocking on loss

- Further latency

Media transport and congestion control

What is the problem?
Why not TCP?
What are the challenges?
Directions and solutions

Need an alternative to TCP congestion control
- suitable for interactive multimedia

Avoid TCP-induced latency
- for media flows
- due to TCP cross traffic

Latency is critical; maximising throughput less so
- media traffic has rate bounds

Media transport and congestion control

What is the problem?

Why not TCP?

What are the challenges?

Directions and solutions

Exploring three directions

- RTP circuit breaker
- Media congestion control algorithms
- Active queue management

RTP circuit breaker

How to stop errant media flows?

RTP circuit breaker

How to stop errant media flows?

Build on RTP reception quality feedback

- Media quality unusable
- Media and/or feedback timeout
- Congestion: 10× TCP throughput

$$T = \frac{s}{R\sqrt{\frac{2p}{3}} + (t_{RTO}(3\sqrt{\frac{3p}{8}})p(1+32p^2))}$$

Protects network on multi-second timescale

- Three reporting intervals
- Cease transmission or back-off 10×

Works in parallel with congestion control

- Last resort to protect network
- Does not address latency concerns

V. Singh, S. McQuistin, M. Ellis, and C. S. Perkins, Circuit Breakers for Multimedia Congestion Control, Proc. 20th Intl. Packet Video Workshop, San Jose, CA, USA, December 2013. DOI:10.1109/PV.2013.6691439

Z. Sarker, V. Singh, and C. S. Perkins, An Evaluation of RTP Circuit Breaker Performance on LTE Networks, Proc. IEEE Infocom Workshop on Communication and Networking Techniques for Contemporary Video, Toronto, Canada, April 2014.

RTP circuit breaker

congestion control

How to adapt media to network capacity?

congestion control

How to adapt media to network capacity?

L. S. Brakmo, S. W. O'Malley, and L. L. Peterson. TCP Vegas: New techniques for congestion detection and avoidance. Proc. ACM SIGCOMM Conference, London, UK, August 1994

congestion control

Use delay as congestion signal

- increased delay/inter-packet spacing → congestion
- Avoid standing queues in routers and packet loss
- Evolution of ideas in TCP Vegas

Google proposal draft-alvestrand-rmcat-congestion-02

- Congestion signal: filtered inter-arrival time
- Deployed in Chrome

Cisco proposal draft-zhu-rmcat-nada-03

- Congestion signal: filtered one-way delay
- Natively incorporates ECN feedback

Active discussion in IETF RMCAT working group

- Google proposal has stability issues
- Cisco proposal requires synchronised clocks
- Both could develop into reasonable protocols

Avoid standing queues in routers

Active queue management (AQM)

Avoid standing queues in routers

Can we separate media traffic from TCP?

- Delay-based congestion control will lose to TCP
- Can switch to loss-based mode, but will lose on latency

AQM gives media traffic segregated queue

- CoDel, PIE, etc.
- Latency benefit, irrespective of cross traffic
- Deployment concerns

K. Nichols and V. Jacobson. Controlling queue delay. ACM Queue, 10(5), May 2012

WebRTC deployment is starting

- Chrome and Firefox
- Increasing developer interest

Implications for network operators

- Increasing peer-to-peer UDP media and data flows
- Protected via RTP circuit breaker
- Evolving congestion control story

Implications for research

- Interactive multimedia congestion control an open issue
- Need to understand network characteristics
- Need to understand performance of RTP circuit breaker
- Need to understand performance of AQM