University of Glasgow | School of Computing Science

# WebRTC: IETF Standards Update

September 2016

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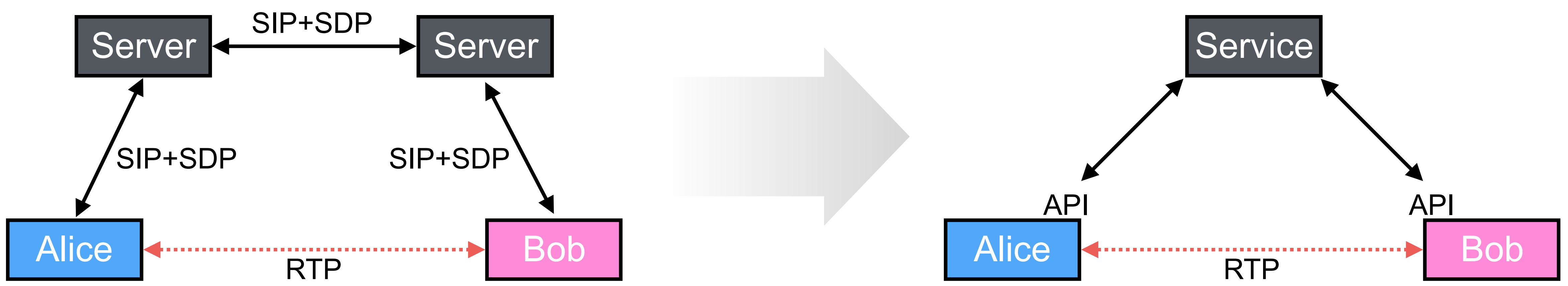
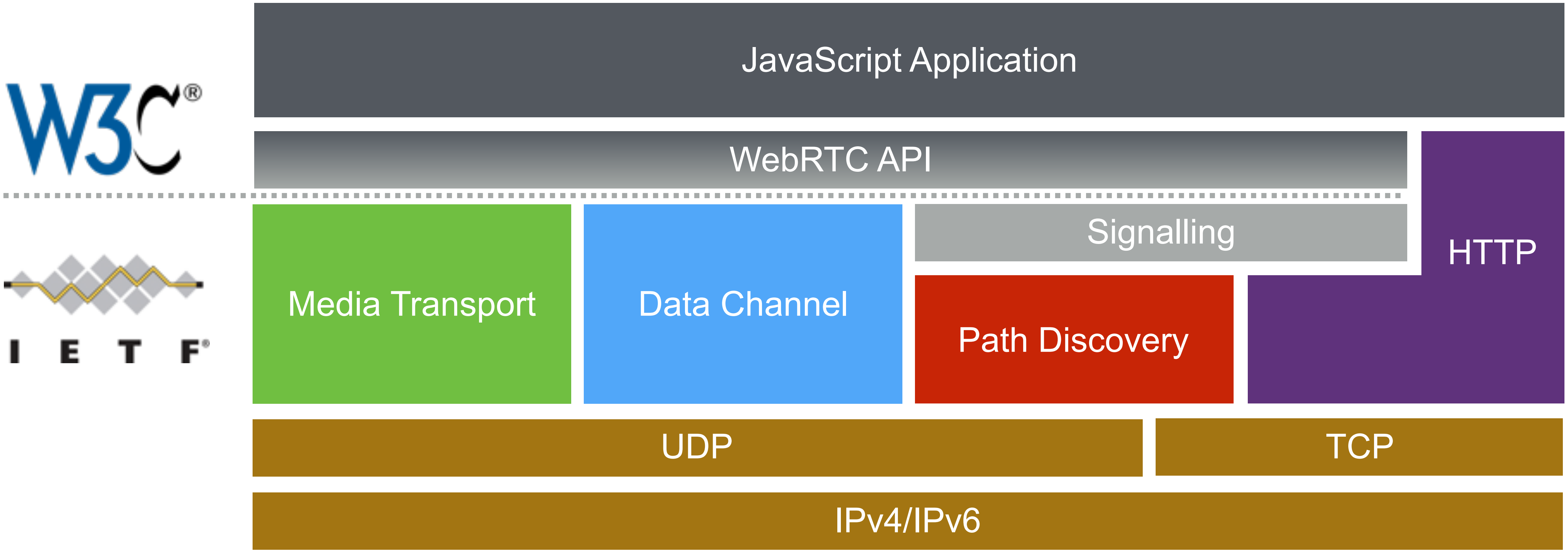
# WebRTC Goals



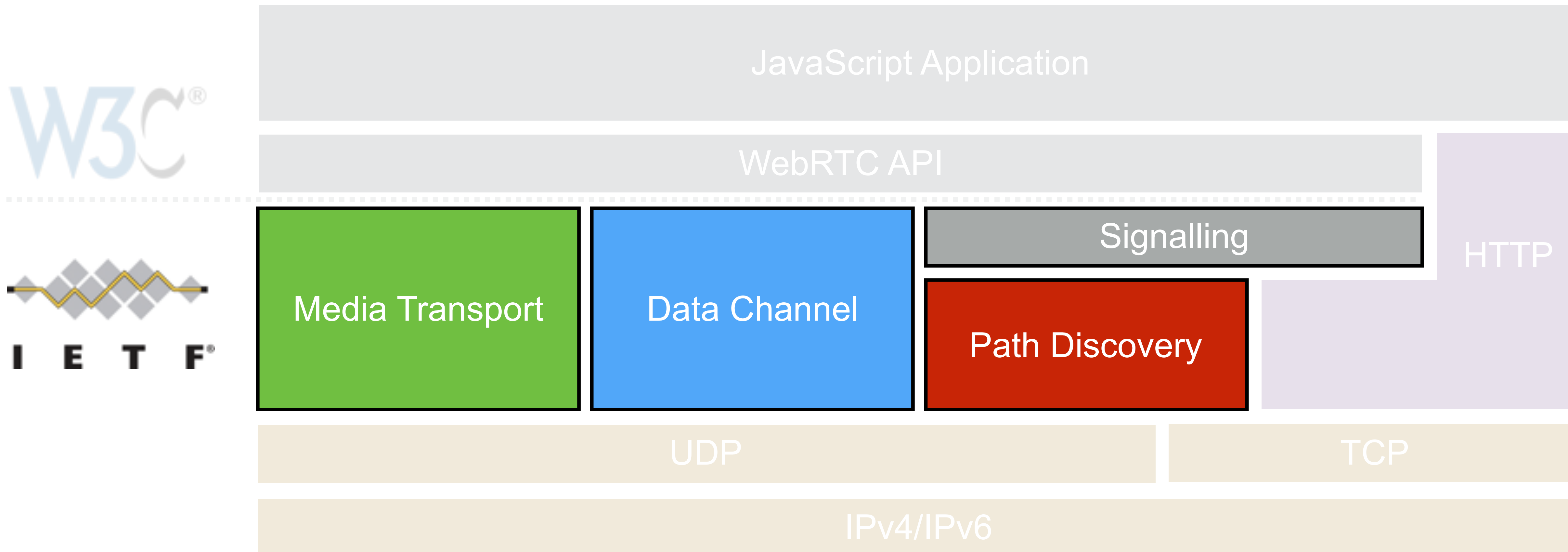

The SIP framework is overly complex and rigid – hinders innovation

Embed standard media stack (RTP, ICE, etc.) into browsers, expose a standard control API rather than a standard signalling protocol – innovate above that API

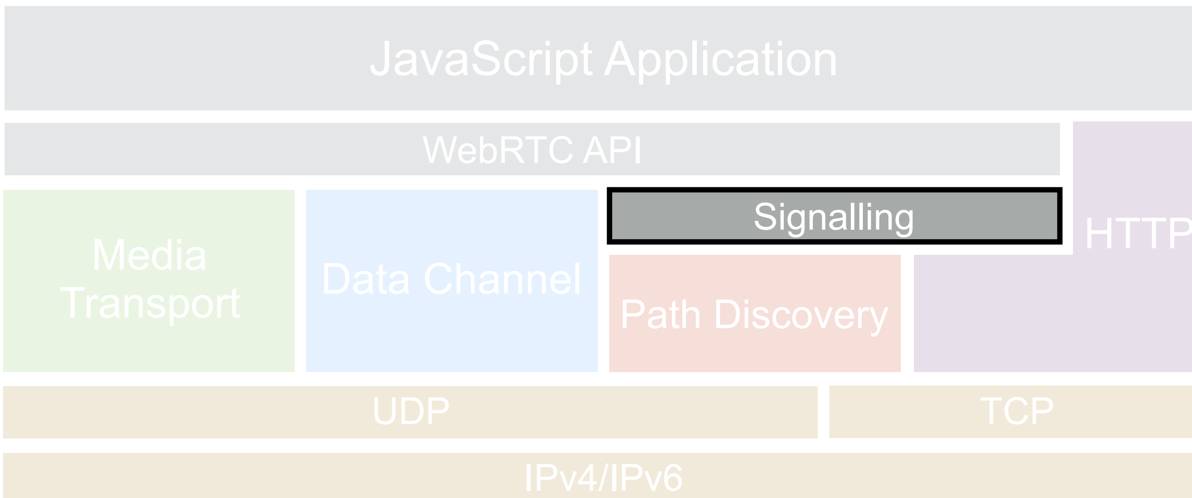
# WebRTC


JavaScript Application
WebRTC API
Media Transport
Data Channel
Signalling
Path Discovery
HTTP
UDP
TCP
IPv4/IPv6

# WebRTC in IETF



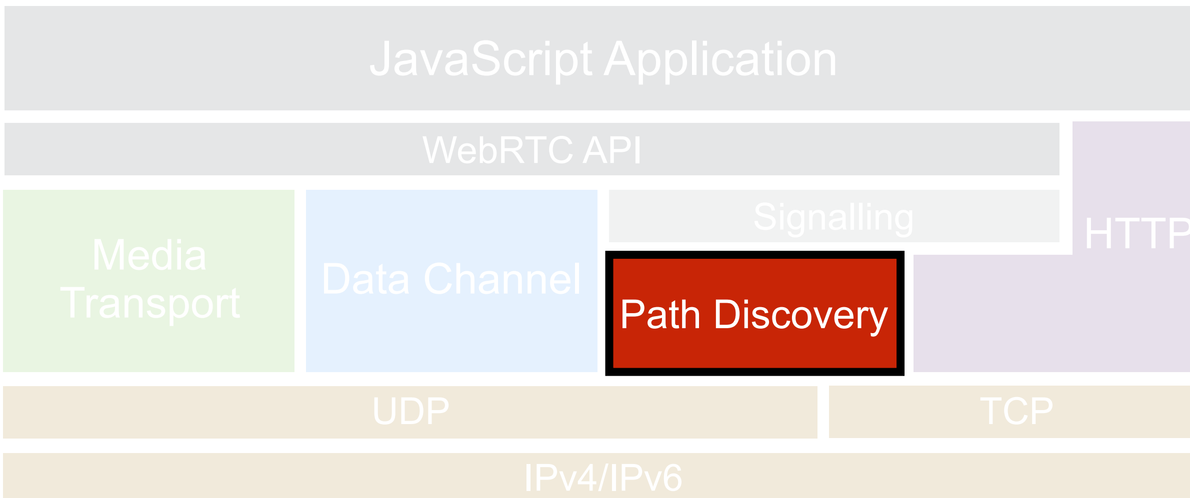
# WebRTC in IETF: Signalling




| Draft | Status |
|---|---|
| draft-ietf-rtcweb-use-cases-and-requirements | RFC 7478 |
| draft-ietf-rtcweb-overview | In progress |
| draft-ietf-rtcweb-security | In progress |
| draft-ietf-rtcweb-security-arch | In progress |
| draft-ietf-rtcweb-jsep | In progress |
| draft-ietf-rtcweb-sdp | In progress |
| draft-ietf-rtcweb-constraints-registry | In progress |
| draft-ietf-mmusic-sdp-bundle-negotiation | WG last call |
| draft-ietf-mmusic-msid | With RFC Editor |
| draft-ietf-mmusic-sdp-mux-attributes | IESG review |
| draft-ietf-mmusic-rid | In progress |
| draft-ietf-mmusic-sdp-simulcast | WG last call |
| draft-ietf-mmusic-mux-exclusive | With RFC Editor |
| draft-ietf-mmusic-4572-update | WG last call |
| draft-ietf-mmusic-dtls-sdp | WG last call |

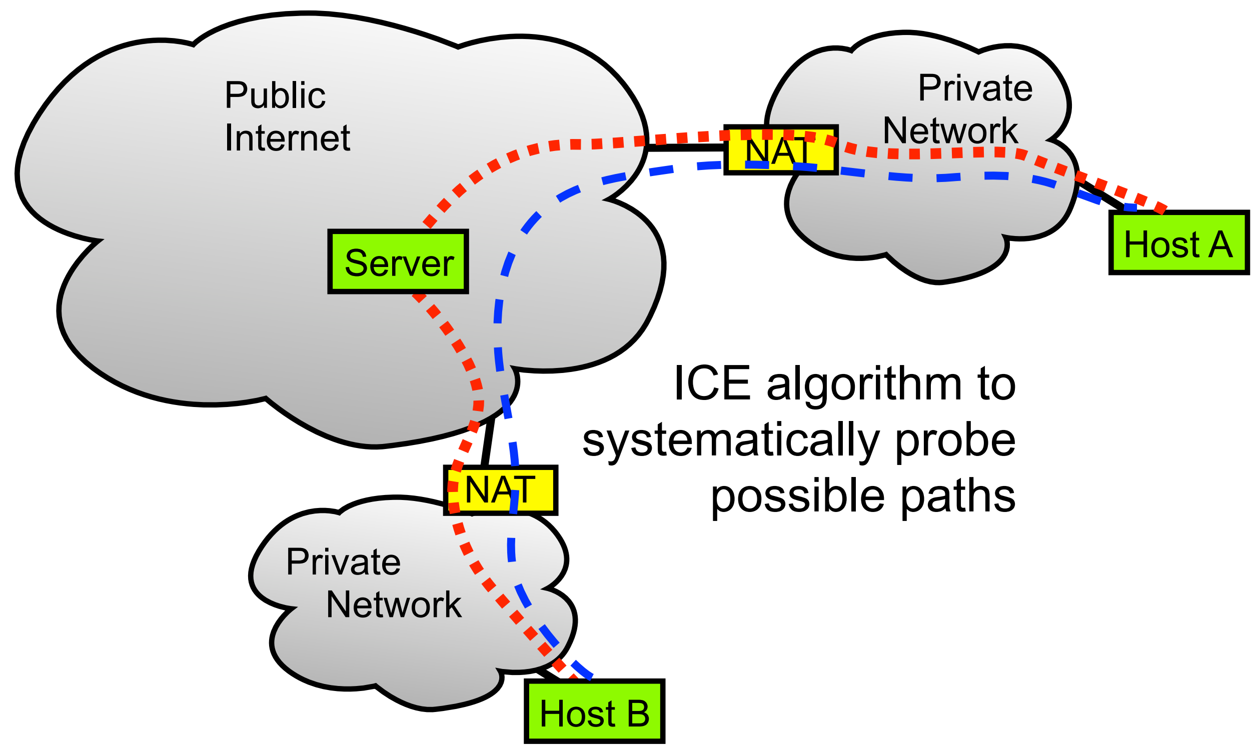
- JSEP and SDP exposed via API
- JSEP extracts SDP offer-answer out into reusable API component
    - SDP not easy to process with JavaScript
    - Extension and modification model poorly specified – simple applications are simple, but over-complicates other scenarios
    - An ORTC-like API might be cleaner?
- SDP BUNDLE extension groups WebRTC traffic on single port:
    - RTP, Data Channel, STUN, DTLS
    - Complexity in identifying m= lines when bundled → msid, rid
    - Complexity in handling bundled attributes, signalling multiplexed flows
- Major issues resolved, but details remain open...

# WebRTC in IETF: Path Discovery



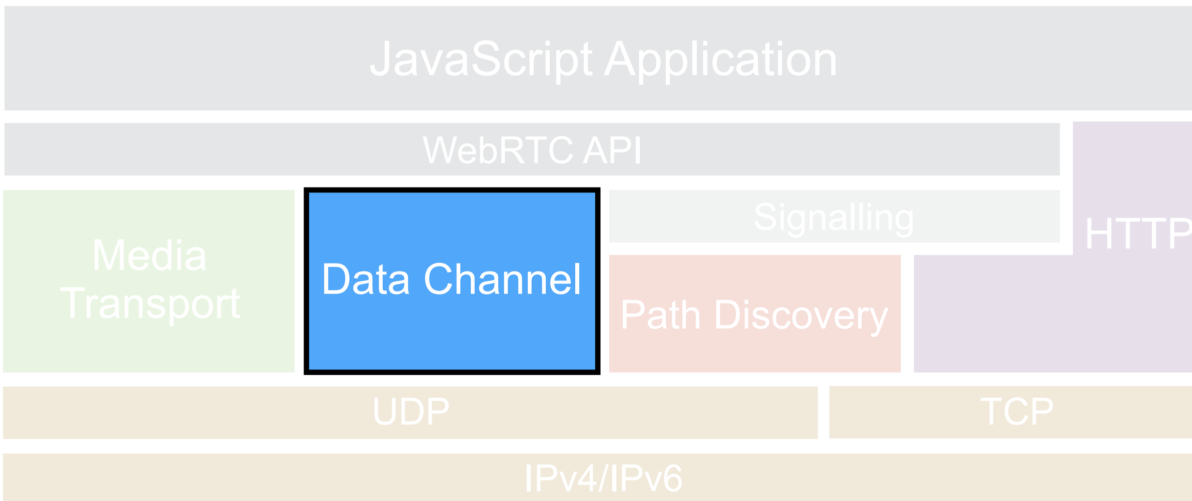

| Draft | Status |
|---|---|
| draft-ietf-rtcweb-transports | Approved |
| draft-ietf-rtcweb-ip-handling | In progress |
| draft-ietf-rtcweb-stun-consent-freshness | RFC 7675 |
| draft-ietf-mmusic-sctp-sdp | In progress |
| draft-ietf-ice-dualstack-fairness | IESG review |
| draft-ietf-ice-rfc5245bis | In progress |
| draft-ietf-ice-trickle | In progress |
| draft-ietf-rtcweb-alpn | With RFC Editor |
| draft-ietf-tsvwg-rtcweb-qos | With RFC Editor |
| draft-ietf-mmusic-ice-sip-sdp | In progress |
| draft-ietf-tram-stunbis | In progress |

- STUN and TURN to discover NAT bindings and relay traffic



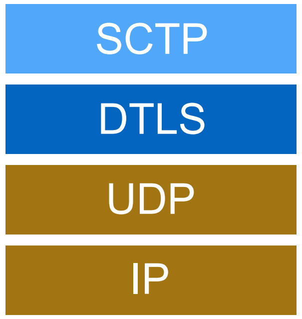

- Privacy concern around local IP address leak resolved
- Ongoing ICE revisions based on deployment experience with SIP

# WebRTC in IETF: Data Channel



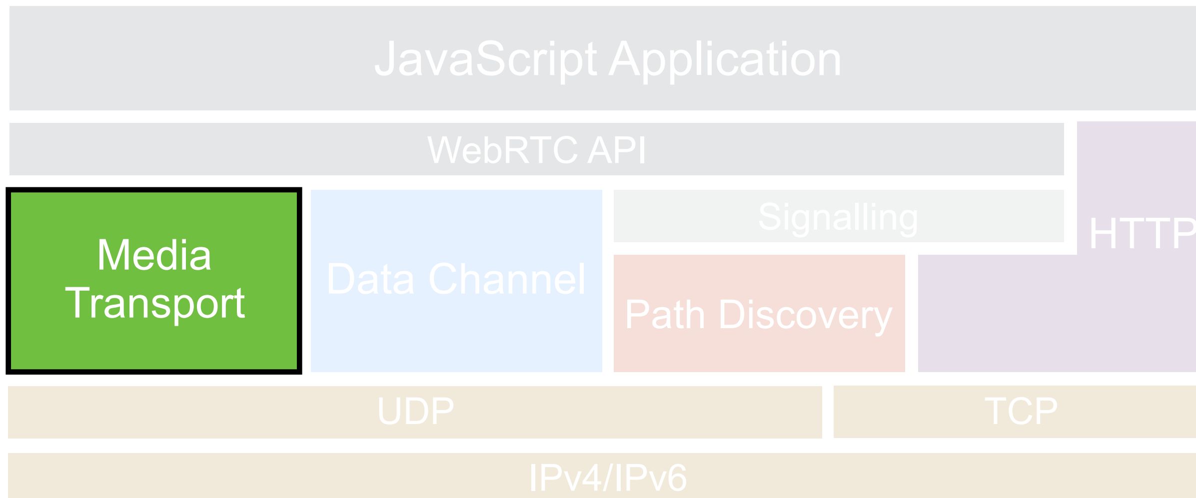

| Draft | Status |
|---|---|
| draft-ietf-rtcweb-data-channel | With RFC Editor |
| draft-ietf-rtcweb-data-protocol | With RFC Editor |
| draft-ietf-tsvwg-sctp-dtls-encaps | With RFC Editor |
| draft-ietf-tsvwg-sctp-ndata | In progress |

- Direct peer-to-peer data between browsers; no server involvement
- SCTP in secure UDP tunnel:




- UDP tunnel ensures deployability but prevents SCTP multihoming

# WebRTC in IETF: Media Transport

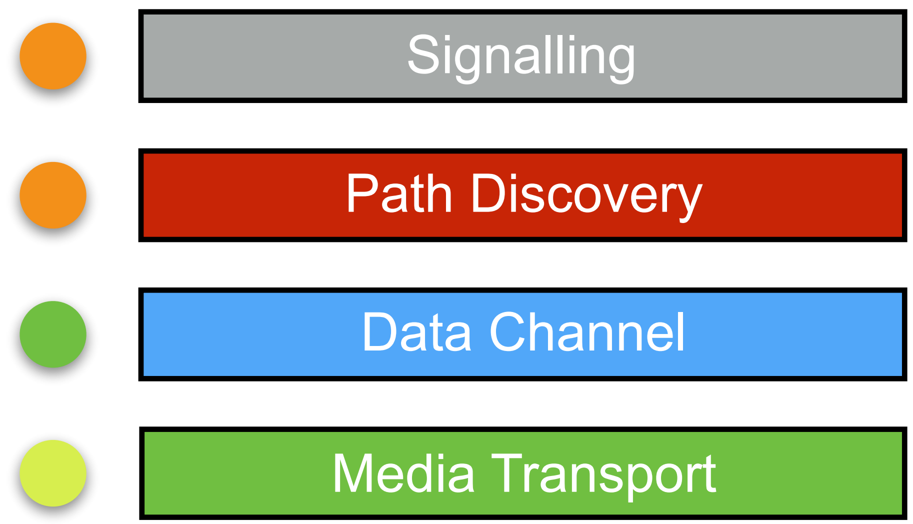



| Draft | Status |
|---|---|
| draft-ietf-rtcweb-rtp-usage | With RFC Editor |
| draft-ietf-rtcweb-audio | RFC 7874 |
| draft-ietf-rtcweb-audio-codecs-for-interop | RFC 7875 |
| draft-ietf-rtcweb-video | RFC 7742 |
| draft-ietf-rtcweb-fec | In progress |
| draft-ietf-avtcore-rtp-circuit-breakers | With RFC Editor |
| draft-ietf-avtcore-rtp-multi-stream | With RFC Editor |
| draft-ietf-avtcore-rtp-multi-stream-optimisation | With RFC Editor |
| draft-ietf-avtcore-multi-media-rtp-session | With RFC Editor |
| draft-ietf-avtext-rid | IESG review |
| draft-ietf-avtext-sdes-hdr-ext | RFC 7941 |
| draft-ietf-payload-flexible-fec-scheme | In progress |
| draft-ietf-avtcore-rfc5761-update | IESG review |

- Audio and video codecs
  - Opus, G.711, and DTMF digits required; AMR recommended
  - H.264 and VP8 required
  - Support for other codecs optional
- Modern RTP and RTCP stack
  - Bundled media on a single UDP port
  - Multiparty multimedia group conferencing – details around multiparty RTP sessions with different media types clarified
  - Secure RTP with DTLS-SRTP handshake
  - Detailed reception quality feedback, with NACK, retransmission, and FEC possible
  - Circuit breaker and congestion control for safe deployment on constrained paths

# WebRTC in IETF: Status Summary

- Media transport and data channel essentially complete
- Path discovery and signalling protocols near completion – resolving details




- Why are the standards taking so long?
  - IPR around choice of mandatory to implement codec
  - Decoupling SDP offer/answer from SIP to form JSEP, and complexity of resulting API interactions
  - Complexity of bundled media: signalling and feature interaction; corner cases around use of RTP and RTCP with multiple simultaneous media types; demultiplexing and QoS with several protocols on a single port
  - Revisions to STUN, TURN, and ICE

# Challenges and Future Directions

- How might WebRTC evolve in future?
  - Quality of service support
  - Congestion control
  - ECN and ensuring low latency
  - Multicast and IPTV
  - Relation to new path layer protocols

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**Differential QoS on a single UDP flow**

Applications set different DSCP code points for the different media types and the data channel, and for different flow priorities

- RFC 7657 and draft-ietf-tsvwg-rtcweb-qos-18

Do QoS-marked flows traverse the network?

- Forwarding behaviour for some DSCP values is implementation defined – unclear what's typical
- DSCP field can be re-written or zeroed at network boundaries
- Networks can discard packets with certain DSCP values due to security or business concerns

Unclear whether QoS support offers any benefits for interdomain use – or indeed, whether it hurts media quality

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**RTP Circuit Breaker**

New algorithm – does it work in the wide range of scenarios where WebRTC is deployed?

**Congestion control for interactive media**

Algorithms under development: Google Congestion Control, NADA, SCReAM

- Evaluation at an early stage – unclear any of these are stable in all desired scenarios, or with different types of cross traffic

Generic feedback mechanism under development

- Early work – unclear RTCP feedback can meet the timeliness requirements with reasonable overhead

**Initial WebRTC deployments will have evolving congestion control – does this matter?**

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**Explicit Congestion Notification**

Desire to move away from loss as congestion signal

- High latency → must fill queue to trigger loss
- Disruptive to user experience

Use of ECN with AQM allows smaller queues

- Requires support from network (CoDel, PIE, …)
- Requires support from circuit breaker
- Requires support from congestion controller
- Incrementally deployable

Response to ECN-CE mark should be less aggressive than response to packet loss

IETF L4S and TCP Prague experiments use ECT(1) with radically different congestion control: potentially much lower latency, but disruptive change

- Congestion response: $^1/_{\sqrt{p}} \rightarrow {}^1/_p$
- Not interoperable: dual queue AQM required

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**Support for IP Multicast in WebRTC**

Two approaches to video streaming:
- HTTP adaptive streaming – browser native format
- Multicast IPTV – designed for managed networks

WebRTC media stack is very similar to the multicast IPTV media stack:
- Missing MPEG-2 codec and payload format
- Missing source-specific multicast support
- Missing rapid channel change extensions

Incremental additions → not complex

Should WebRTC support multicast, so browsers can act as native IPTV clients?
- Better scaling for live streams
- Lower latency

Longer term: media interworking and interoperability?
- Different delivery modes need different encoding
- Hand-off between devices and delivery modes is difficult and non-scalable

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**Substrate protocols and the path layer**

Biggest challenge with WebRTC was making bundled media work

- Significant impact on RTP, congestion control, QoS
- Extremely complex signalling

New work in IETF: SPUD prototype and PLUS BoF

- Common UDP-based substrate layer on which new transport protocols can be run
- A secure path layer, with scope for edge-network communication

Can/should WebRTC migrate to run over this layer?

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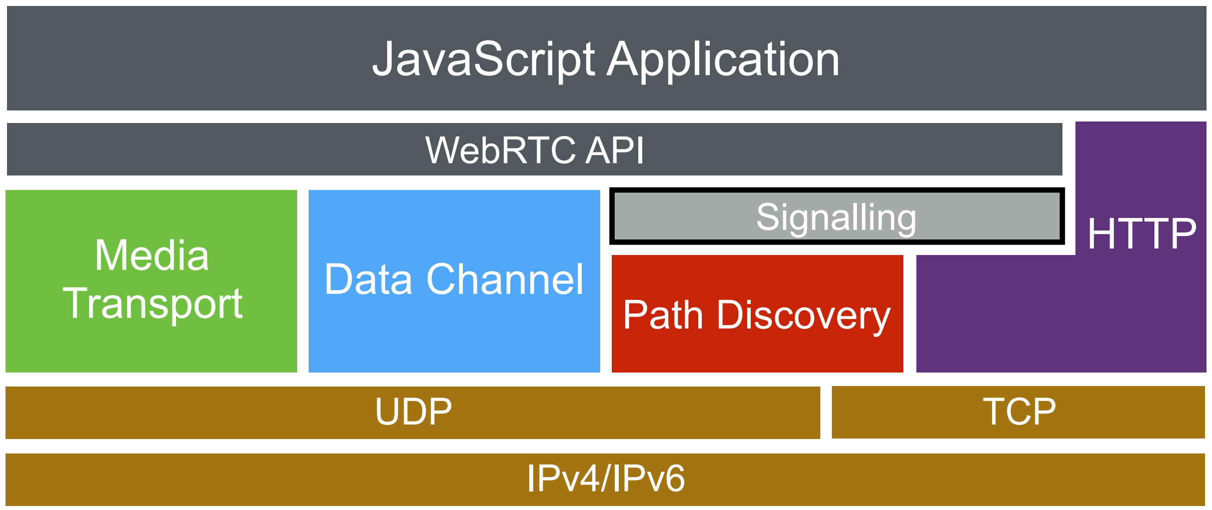
- A transport-oriented viewpoint – what else?
  - Signalling APIs – ORTC *vs.* SDP-based approaches
  - Simplified JavaScript libraries
  - Monitoring and management tools and interfaces

# Conclusions




- WebRTC provide a good baseline – a flexible, evolvable, framework
- Core IETF standards essentially done
- Clear path to evolve the network with lower latency, more adaptive media

Interesting challenges remain, but WebRTC is ready for deployment