

# RtpTransport Reminder

Memory Tool for W3C

## Purpose of This Document

This document is a tool to help us quickly remember what is going on with RtpTransport. There are often large gaps of time between discussions about RtpTransport, and it's easy to forget what the current state is. Any time when it's needed, such as before a discussion, we may read this document to get up-to-speed faster than reading the entire spec of a bunch of Github issues. Hopefully this will allow us to make good use of our time in discussion and make good progress.

## Rough Consensus So Far

- We want some kind of packet-level API for sending and receiving RTP and RTCP.
- Packets must always be encrypted using SRTP/SRTCP.
- Packets must be congestion controlled (as much as RTP/RTCP usually is).

## Documents

### Explainer

<https://github.com/w3c/webrtc-rtptransport/blob/main/explainer.md>

Summary: The goal is for the web app to be able to do custom things like custom payloads/codecs, custom packetization, custom FEC, custom NACK/RTX, custom metadata, custom RTCP messages, etc.

There are examples, but they are *speculative* and *immature*. They are speculative in the sense that they do not represent any decisions or consensus, just ideas of what could be. They are immature in the sense that they probably have mistakes.

### Spec

<https://github.com/w3c/webrtc-rtptransport/blob/main/index.bs>

Summary: Just a skeleton so far.

But what will hopefully be added soon is: the web app can send and receive RtpPackets and RtcpPackets. The RtpPackets have a payload along with RTP metadata such as timestamp, header extensions, etc. The RtcpPackets can be compound and contain many payloads, where each has a payload type and subtype.

## Issues

If you think of a new issue, file in Github it at <https://github.com/w3c/webrtc-rtptransport/issues> (check to make sure there isn't an existing one)

### Customizing piecemeal

<https://github.com/w3c/webrtc-rtptransport/issues/9>

Summary: Can we customize one thing (RTP payload) while keeping other things unchanged (NACK/RTX)?

### SDP Bumper Lanes

<https://github.com/w3c/webrtc-rtptransport/issues/10>

Summary: How much can the web app go beyond the bounds of what is negotiated in SDP?

### Arbitrary RTCP

<https://github.com/w3c/webrtc-rtptransport/issues/11>

Summary: Can a web app send any arbitrary bytes in RTCP messages?

### Arbitrary RTP Header Extensions

<https://github.com/w3c/webrtc-rtptransport/issues/12>

Summary: Can a web app send any arbitrary bytes in RTP header extensions? If so, is it OK that they are not encrypted (or is cryptex required)?

## SRTP/Cryptex

<https://github.com/w3c/webrtc-rtptransport/issues/7>

Summary: How do you turn it on (to encrypt header extensions and CSRCs)?

## Workers

<https://github.com/w3c/webrtc-rtptransport/issues/13>

Summary: What is transferable between workers?

## WHATWG Streams or not

<https://github.com/w3c/webrtc-rtptransport/issues/8>

Summary: Should we use WHATWG streams?