WebRTC 1.0 objects

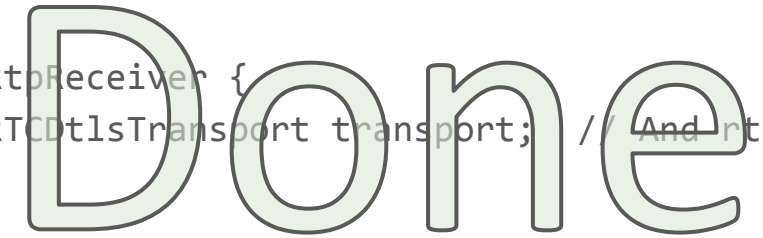
at 2015 f2f

Done: DtlsTransport

```
partial interface RTCRtpSender {
  readonly attribute RTCDtlsTransport transport;  // And rtcpTransport
}

partial interface RTCRtpReceiver {
  readonly attribute RTCDtlsTransport transport;  // And rtcpTransport
}

interface RTCDtlsTransport {
  readonly attribute RTCIceTransport transport;
  readonly attribute RTCDtlsTransportState state;
  sequence<ArrayBuffer> getRemoteCertificates();
  attribute EventHandler onstatechange;
}
```



Done: initial IceTransport

```
interface RTCIceTransport {
  readonly attribute RTCIceConnectionState state;
  RTCIceCandidatePair? getSelectedCandidatePair();
  attribute EventHandler onstatechange;
  attribute EventHandler onselectedcandidatepairchange
}

dictionary RTCIceCandidatePair {
  RTCIceCandidate local;
  RTCIceCandidate remote;
}
```

Done

Done: RtpSender/RtpReceiver parameters

```
partial interface RTCRtpSender {
  RTCRtpParameters getParameters();
  void setParameters(RTCRtpParameters parameters);
}


dictionary RTCRtpParameters {
  sequence<RTCRtpEncodingParameters> encodings;
}

dictionary RTCRtpEncodingParameters {
  boolean active;
  RTCPriorityType priority;  // high, medium, low, very-low
  unsigned long maxBitrate;
}
```

Done

Pending: RtpSender/RtpReceiver capabilities (PR 269)

```
partial interface RTCRtpSender {  // Same as RTCRtpReceiver
  static RTCRtpCapabilities getCapabilities(DOMString kind)
}

dictionary RTCRtpCapabilities {
  sequence<RTCRtpCodecCapability> codecs;
  sequence<RTCRtpHeaderExtensionCapability> headerExtensions;
}

dictionary RTCRtpCodecCapability {
  DOMString mimeType;
}

dictionary RTCRtpCodecCapability {
  DOMString uri;
}
```

Can we merge?

Pending: SctpTransport (PR 270)

```
partial interface RTCPeerConnection {
  readonly attribute RTCSctpTransport? sctp
}

interface RTCSctpTransport {
  readonly attribute RTCDtlsTransport transport;
  readonly attribute unsigned long maxMessageSize;
}
```

Can we merge?

Needs discussion: IceTransport more readonly info: (PR 280)

```
partial interface RTCIceTransport {
  readonly attribute RTCIceRole role;
  readonly attribute RTCIceComponent component;
  readonly attribute RTCIceGatheringState state;
  RTCIceParameters? getLocalParameters();
  RTCIceParameters? getRemoteParameters();
  sequence<RTCIceCandidate> getLocalCandidates();
  sequence<RTCIceCandidate> getRemoteCandidates();
  attribute EventHandler ongatheringstatechange;
}

dictionary RTCIceParameters {
  DOMString usernameFragment;
  DOMString password;
}
```

Discussion:

Worth Having?

Needs discussion: RtpSender more readonly info (PR 273)

```
dictionary RTCRtpParameters {
  // ...
  sequence<RTCRtpHeaderExtensionParameters> headerExtensions;
}

dictionary RTCRtpHeaderExtensionParameters {
  DOMString uri;
  unsigned short id;
  boolean encrypted;
}

dictionary RTCRtpEncodingParameters {
  unsigned long ssrc;
  RTCRtxParameters rtx;    // dictionary { unsigned long ssrc; }
  RTCFecParameters fec;    // dictionary { unsigned long ssrc; }
  RTCRtcpParaemeters rtcp; // dictionary { DOMString cname; boolean reducedSize; }
}
```

Discussion:

Worth Having?

Needs discussion: RtpSender codec selection (PR 258)

```
dictionary RTCRtpEncodingParameters {
  unsigned short payloadType;
}

dictionary RTCRtpParameters {
  // ...
  sequence<RTCRtpCodecParameters> codecs;  // These are all read-only
}

dictionary RTCRtpCodecParameters {
  // These are all read-only.
  DOMString mimeType;
  unsigned long clockRate;
  unsigned short channels;
  DOMString sdpFmtpLine;
}
```

Discussion:

Worth Having?

RtpSender codec selection example

```
// Pick dalaa if it's available
var params = sender.getParameters();
for (codec of params.codecs) {
  if (codec.mimeType == "video/dalaa") {
    params.encodings[0].payloadType = codec.payloadType;
    sender.setParameters(params);
  }
}
```

Needs discussion: RtpSender more parameters (PR 273)

```
dictionary RTCRtpEncodingParameters {
  double resolutionScale;  // 1 == full, 2 == half, 4 == quarter
  double framerateScale;   // 1 == full, 2 == half, 4 == quarter
  double framerateBias;    // 1.0 == framerate, 0.0 == resolution, 0.5 == default
}
```

Discussion:

Worth Having?

Needs Discussion: PeerConnection warmup (PR 271)

```
partial interface PeerConnection {
  RtpSender createRtpSender(DOMString kind);
}
```

For more details, this deserves its own slide deck:
[PeerConnection ICE/DTLS warmup]

Needs Discussion

PeerConnection warmup example

```
// Offer side
var sender = pc.createRtpSender("audio");  // Adds sendrecv m-line in createOffer
// ... otherwise normal offer/answer/SLD/SRD ...
// Wait for the "I really answered" bit in signalling
sender.replaceTrack(track);
// Hookup pc.getReceivers()[0].track

// Answer side
var sender = pc.createRtpSender("audio");  // Uses existing m-line in createAnswer
// ... otherwise normal offer/answer/SLD/SRD ...
// Wait for the user to really answer
// Send the "I really answered" bit
sender.replaceTrack(track);
// Hookup pc.getReceivers()[0].track
```

PeerConnection warmup questions

- What's the "track ID"/MSID/MID? A random ID
- Do we do anything with track-specific hardware codecs? No
- Does this create an m-line with sendrecv, sendonly, recvonly, or inactive? sendrecv, just like addTrack
- Can we rename replaceTrack to setTrack? replaceTrack doesn't make sense when there isn't one yet.
- Should replaceTrack cause renegotiation? If so, I vote we add another method: setTrackWithoutRenegotiation and use that for ICE/DTLS warmup.
- Do we need to worry about changing from sendonly/inactive to sendrecv/recvonly? I don't think we do.

Needs Discussion: Replace offerToReceiveX (PR 279)

```
partial interface PeerConnection {
  // Note: We need to change createRtpSender to create a
  // sendonly m-line unless an RtpReceiver is also created
  // This changes the warmup example.
  RtpSender createRtpSender(DOMString kind);
  RtpReceiver createRtpReceiver(DOMString kind);
}
```

For more details, this deserves its own slide deck

[Remaining issues with RtpSenders/RtpReceviers and SDP]

Needs Discussion

Example of createRtpReceiver()

```
var sender = pc.addTrack(audioTrack);  // audio sendrecv
var sender = pc.createRtpSender();  // video sendonly
sender.replaceTrack(videoTrack);
var receiver2 = pc.createRtpReceiver("audio");  // audio recvonly
var receiver3 = pc.createRtpReceiver("audio");  // audio recvonly
// Generates like {offerToRecevieAudio: 3, offerToReceiveVideo: 0}
//   1 audio sendrecv line
//   1 video sendonly line
//   2 audio recvonly lines
pc.createOffer();
```

Needs Discussion: PC.connectionState (PR 291)

```
partial interface PeerConnection {
  // disconnected, connecting, connected, failed
  readonly attribute PeerConnectionState connectionState;
  attribute EventHandler onconnedctionstatechange;
}

pc.onconnectionstatechange = function() {
  if (pc.connectionstate == "failed") {
    // Uh-oh!
  } else if (pc.connectionstate == "connected") {
    // Great!
  }
}
```

Needs Discussion

More slides: https://docs.google.com/presentation/d/1vHTlEof4dV12iAtZ7SrchrtVK1o239pbrPi9fAutnb4/edit#slide=id.

Needs Discussion: PC.onwarning/onfatalerror (PR 292)

```
partial interface PeerConnection {
  // The PeerConnection can't continue.
  attribute EventHandler onfatalerror;
  // The PeerConnection can continue.
  attribute EventHandler onwarning;
}

pc.onwarning = function(evt) {
  console.log(evt.message);
}

pc.onfatalerror = function(evt) {
  console.log(evt.message);
  goToBrokedUI();
}
```

Needs Discussion

Did you notice a problem?

While discussing RtpSender/RtpReceiver, did you notice that it's very complicated to reason about when they are created, how they map to SDP, and what state they are in. With our current model of things, I found that things get hairy rather quickly and there are lots of remaining issues not address in the spec. I have documented many of those and created a proposal to fix them here:

Remaining issues with RtpSenders/RtpReceviers and SDP and a proposal to resolve them

Proposal: Explicit RTCSdpMediaSection objects

```
partial interface PeerConnection {
  sequence<RTCSdpMediaSection> getSdpMediaSections();

  // Replaces addTrack
  RTCSdpMediaSection addMedia(
    MediaStreamTrack track,
    RTCSdpMediaSectionInit dict);

  // Replaces createRtpSender/createRtpReceiver
  RTCSdpMediaSection addMedia(
    DOMString kind, RTCSdpMediaSectionInit dict);

  // Replaces onaddtrack
  attribute EventHandler? onaddmedia;
}
```

```
dictionary RTCSdpMediaSectionInit {
  RTCSdpMediaDirection direction = "sendrecv";
  array<MediaStream> streams = [];
}

interface RTCSdpMediaSection {
  readonly attribute DOMString kind;
  readonly attribute DOMString mid;
  readonly attribute RTCSdpMediaSectionState localState;
  readonly attribute RTCSdpMediaDirection localDirection;
  readonly attribute RTCSdpMediaSectionState remoteState;
  readonly attribute RTCSdpMediaDirection remoteDirection;

  readonly attribute RtpSender? sender;
  readonly attribute RtpReceiver? receiver;

  // Replaces RtpReceiver.track.stop()
  void close();
}
```

Examples of RTCSdpMediaSection objects

```
// Replaces addTrack, offerToReceiveX, createRtpSender, and createRtpReceiver
pc.addMedia(track);
pc.addMedia("audio", {direction: "recvonly"});
pc.addMedia("audio", {direction: "recvonly"});
pc.addMedia("video", {direction: "sendonly"});
// A normal audio, 2 recvonly audio, and 1 sendonly "warmup" video
pc.createOffer();

// On answer side, replace track.stop()
pc.onmedia = function(evt) {
  if (evt.media.direction == "sendrecv") {
    evt.media.close();
  }
}
pc.addTrack(track, {direction: "sendonly});
// reject incoming audio, send back 1 audio, leave one audio inactive, receive the video warmup
pc.setRemoteDescription(offer);
```

Needs lots of Discussion