WebRTC, Mobility, Cloud, and More...

IIT REAL-TIME COMMUNICATIONS

Conference & Expo ● Oct 17 - Oct 20, 2016 ● Chicago

ORTC/WebRTC Tutorial

IIT RTC Conference
Chicago, IL
October 17, 2016

What We Will Cover Today

- Tour of the Object Realtime Time Communications (ORTC) Model (120 minutes)
- ORTC and WebRTC 1.0 application development (45 minutes)
- Coming attractions: The WebRTC 1.0 object model (if time permits)

Other WebRTC/ORTC Sessions

http://www.rtc-conference.com/2016/wp-content/uploads/Schedule-IIT-RTC-Conf-2016.pdf

- ORTC Update
  - Tuesday, October 18: 9 AM – 10 AM (McCormick Ballroom West)
  - Will cover ORTC Status and ORTC Lib implementation for iOS, Android and UWP
- Panel: WebRTC in the Browser
  - Tuesday, October 18: 1 PM – 2 PM (McCormick Ballroom West)
  - Will cover status of Firefox, Chrome and Edge
- FreeSWITCH, Weathering the Storm of WebRTC
  - Tuesday, October 18: 4 PM – 4:30 PM (McCormick Ballroom West)

Goals of Today’s Tutorial

- To describe the philosophy behind the object approach to Realtime Communications
- To introduce the realtime object model described in the ORTC API
  - http://draft.ortc.org/
- To explain the differences between the current ORTC specification and the implementation in Microsoft Edge
  - http://internaut.com:8080/~baboba/ms/msortc-rs.html
- To demonstrate approaches to ORTC application development
  - ORTC native application
  - WebRTC 1.0 application running on ORTC
- To introduce the WebRTC 1.0 object model

Rules of the Road

- Please speak up if you cannot hear or see
- Raise your hand to ask questions
  - May not be able to call on you immediately
- Will poll the class from time to time
  - You are teaching me, too!
- Slides available here:
  - http://internaut.com:8080/~baboba/iit-tutorial/slides/

About the Presenter

- Principal Architect at Microsoft. Dual role:
  - Edge: RTC platform standards, partner relations
  - Skype: in media team supporting plugin-free Skype on all browsers
- Currently editor of the ORTC and WebRTC 1.0 specifications
  - That doesn’t mean I understand them completely!
  - Regularly find Issues in the specifications (and sometimes fix them)

Class Poll #0

- Question 1: Who is awake?
- Question 2: Of those who are awake, how many have ever used the Chrome browser? Firefox? Edge?
- Question 3: Of those who are awake, how many have ever coded in Javascript?
- Question 4: Of those who have coded in Javascript, how many have experience with the WebRTC 1.0 API? With ORTC?

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A Tour of the ORTC Object Model

Bernard Aboba, Microsoft Corporation
Robin Raymond, Optical Tone Ltd.

IIT RTC Conference
Chicago, IL
October 17, 2016

ORTC Philosophy

- ORTC *not* just for use in browsers.
  - Goal was to provide an API that could be useful for IoT, mobile, server *and* web development.
  - Desire for wide applicability implies need to support many usage scenarios.
- Philosophy: Less is More
  - Focus solely on the media plane.
  - Support for signaling protocols can be provided in add-on libraries (if needed).
  - Modular approach reduces CPU and memory demands, and improves battery life
  - (Imperfect) operating system analogy: micro-kernel architectures
- Result: flexibility and the ability to support advanced capabilities
  - Support for other APIs can be built on top of ORTC (e.g. WebRTC 1.0)
  - Wide range of signaling protocols can be supported: SIP, H.323, JSON signaling
  - Wide range of codecs can be fully supported: H.264/AVC, H.264/SVC, VP8, VP9/SVC, etc.
  - Support for advanced video technology: simulcast, scalable video coding, etc.

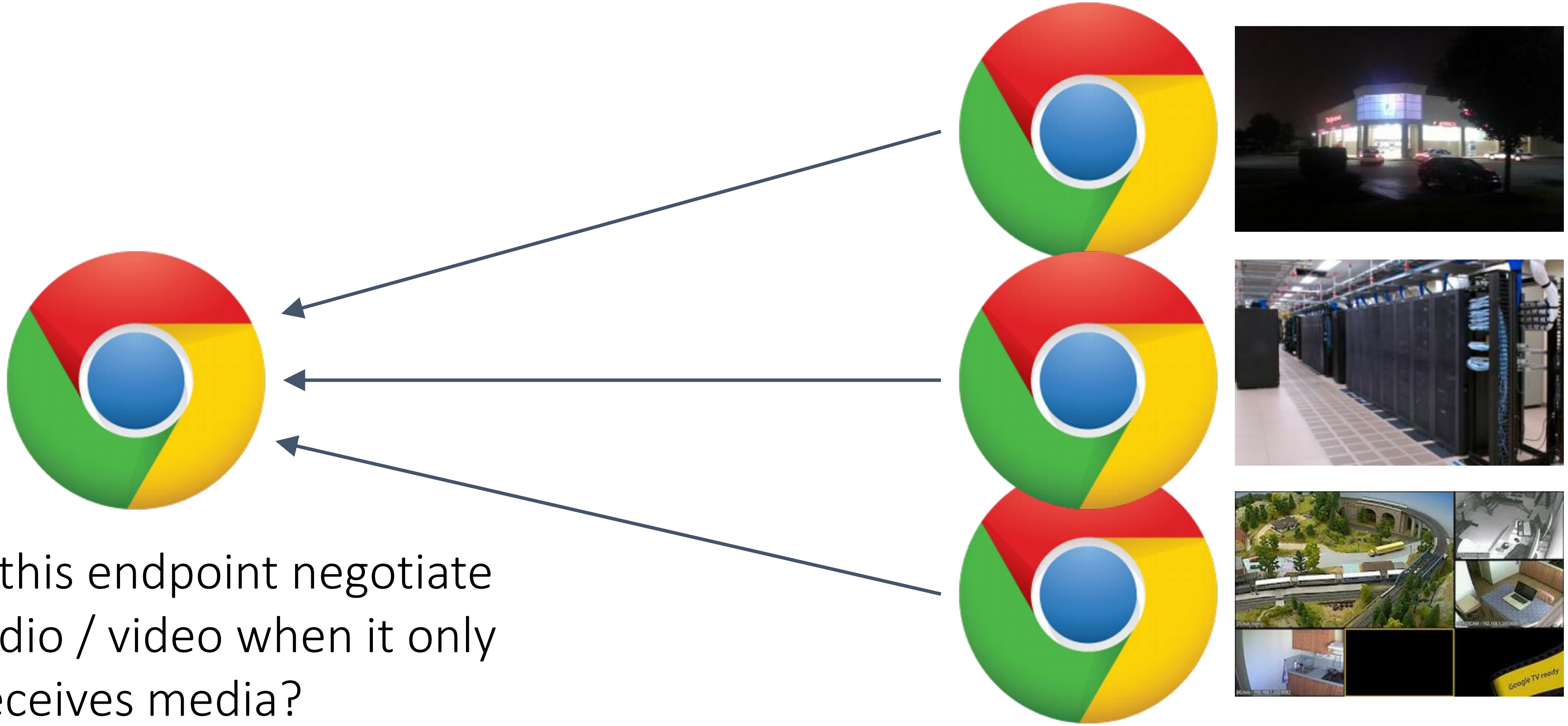
ORTC Implementations

- ORTC Lib
  - http://ortclib.org/
  - An open-source library for UWP, iOS and Android
  - Enhancements for mobile and IoT (seamless roaming, improved battery life, etc.)
  - Details covered during the “ORTC Update” session on Tuesday.
- Mediasoup
  - http://mediasoup.org/
  - Selective Forwarding Unit (SFU) for node.js
- Microsoft Edge:
  - https://developer.microsoft.com/en-us/microsoft-edge/testdrive/
  - Browser implementation supporting ORTC (as well as WebRTC 1.0 via the adapter.js library)
  - More on Edge during the “Browser Update” session.

ORTC Design Goals

- To provide a “**do as I say**” API without a dependency on the Session Description Protocol (SDP).
  - “Do as I say” means that the API either will do what has been requested, or an error will result.
  - SDP can be supported if needed via an add-on library, but can avoid the overhead of an SDP parser and Offer/Answer if your application does not need it.
- To support the functionality of WebRTC 1.0.
  - Enables the WebRTC 1.0 API to be emulated on top of ORTC.
- To support forking, simulcast (sending and receiving) and scalable video coding.
- To support a wide range of communications scenarios (such as asymmetric audio/video and negotiation-free adding/removing of media)

Asymmetric Audio / Video

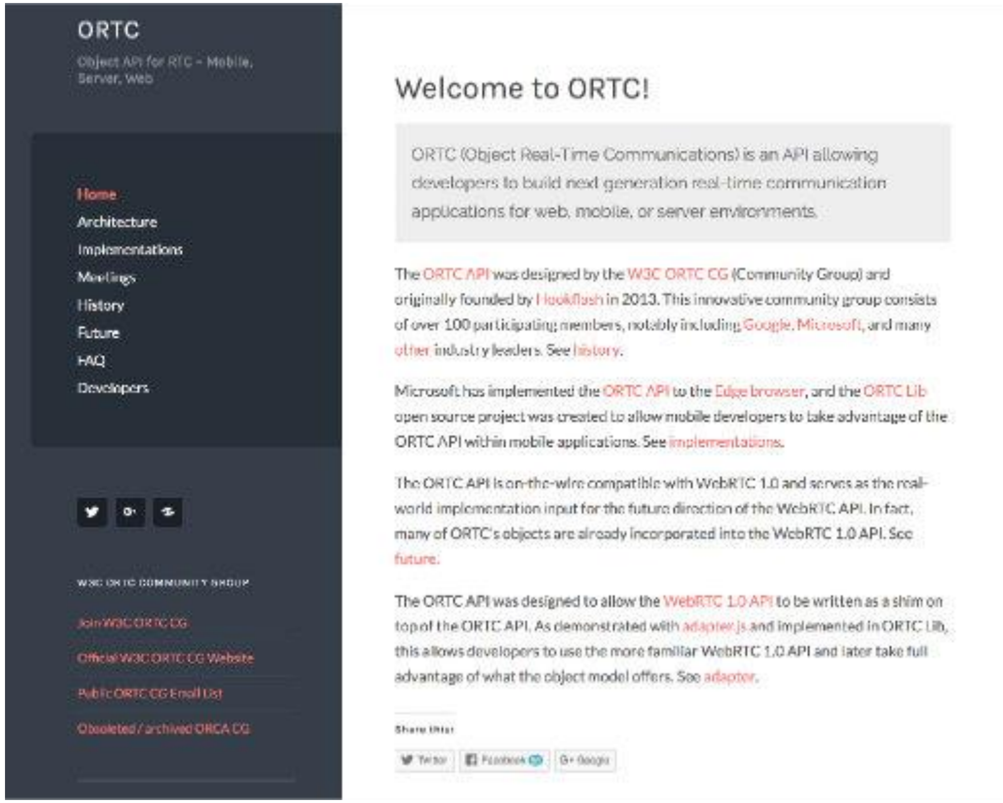


Why must this endpoint negotiate sending audio / video when it only receives media?

Important ORTC related sites

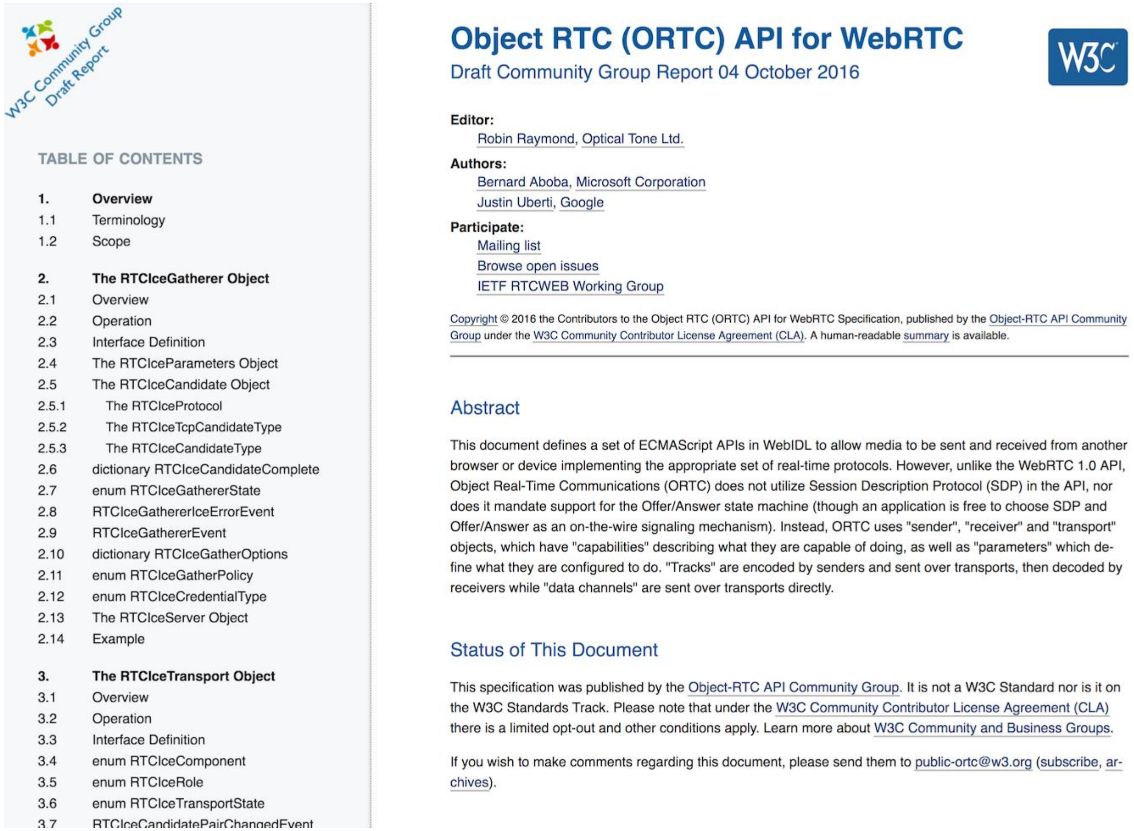
ortc.org

(portal to all things ORTC related)

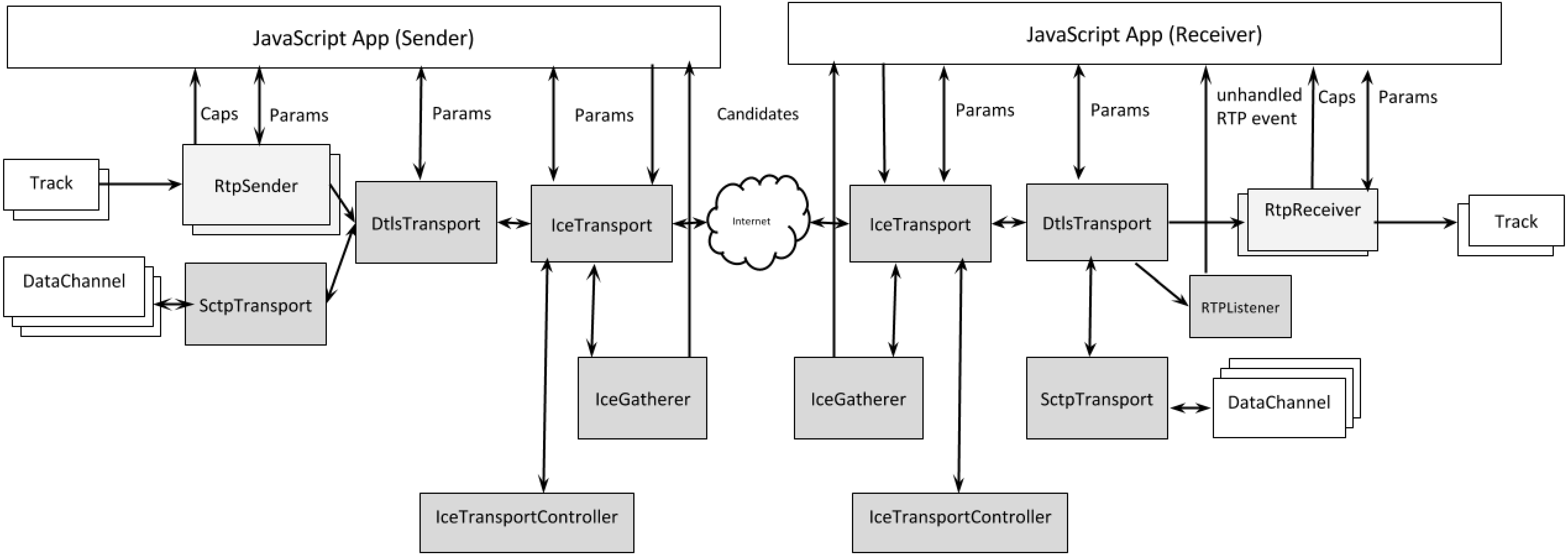


draft.ortc.org

(latest Object RTC API draft)



ORTC Object Model

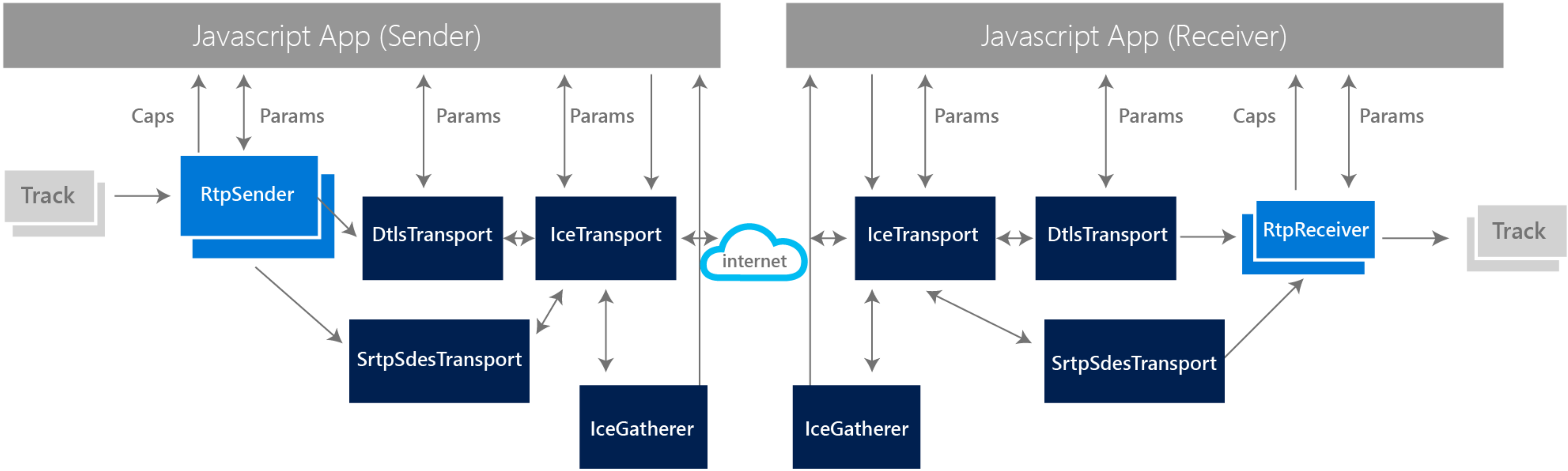


Important Objects in ORTC

- IceGatherer*
- IceTransport / IceTransportController
- DtlsTransport
- RtpSender / RtpReceiver
- RtpListener
- DataChannel
- SctpTransport

* all objects are prefixed with “RTC” in the ORTC API specification

Microsoft Edge ORTC Object Model



Differences

- IceGatherer
  - Edge: Only one IceTransport per IceGatherer (no forking)
- IceTransportController
  - Edge: Not supported, freezing handled via order of IceTransport creation.
- RtpListener
  - Edge: Not supported, unroutable packets dropped.
- DataChannel/SctpTransport
  - Edge: “Under Consideration”

Implemented in Edge but not in ORTC

- SrtpSdesTransport

Class Poll #1

- Question 1: Who is still awake?
- Question 2: Of those who are still awake: Is it important to support forking? Why?
- Question 3: Of those who are still awake: Is it important to support SRTP/SDES? Why?

What is in Microsoft Edge RTC?

- API support: ORTC, native WebRTC 1.0 (in preview)
- Audio codecs: G.711, G.722, Opus, SILK, CN, DTMF, RED and external FEC (with burst loss resilience)
- Video codecs:
  - H.264/AVC (experimental support in about:flags)
  - H.264UC + ULPFECUC (ORTC only)
    - Implementation of RFC 6190 with support for simulcast, temporal scalability and MRST transport
    - Forward error correction with burst loss resilience
  - RTX (in preview)
  - VP8 (under development)
  - Feedback messages: PLI (experimental support in about:flags), REMB/Generic NACK (in preview)
- Statistics:
  - getStats() metrics
  - msGetStats() supplemental call statistics: support for session quality, burst loss and FEC performance metrics
- Audio/Video multiplexing (BUNDLE), RTP/RTCP mux
- IPv4 and IPv6 (with ICE “happy eyeballs” support)
- Support for DTLS 1.2, SRTP/SDES transports

Limitations of Edge RTC

- No data channel or screen sharing (under consideration)
- RTP/RTCP mux required with DTLS/SRTP
  - Non-mux supported for SRTP/SDES
- No forking support
- No IceGatherer state attribute or state change events
  - null event.candidate used to indicate end-of-candidates
- No unhandled RTP or SSRC conflict events
- No support for certificate management
  - RSA certificates generated by default
  - Can validate both RSA and ECDSA certificates

RTCIceGatherer

- Gathers host, reflexive, and relay ICE candidates
- Gathers for both UDP and TCP protocols
- Has a local ICE usernameFragment / password
- Listens for incoming ICE, SRTP, and DTLS packets
- Routes ICE packets to **RTCIceTransport**(s) based on remote ICE usernameFragments (i.e. supports forking)
- Routes SRTP / DTLS packets to **RTCIceTransport**(s) based on 5-tuple IP pairings for confirmed ICE paths

Interface Definition

WebIDL

```
[Constructor(RTCIceGatherOptions options)]
interface RTCIceGatherer : RTCStatsProvider {
    readonly attribute RTCIceComponent       component;
    readonly attribute RTCIceGathererState   state;
    void                                      close();
    void                                      gather(optional RTCIceGatherOptions options);
    RTCIceParameters                          getLocalParameters();
    sequence<RTCIceCandidate>                 getLocalCandidates();
    RTCIceGatherer                            createAssociatedGatherer();
             attribute EventHandler           onstatechange;
             attribute EventHandler           onerror;
             attribute EventHandler           onlocalcandidate;
};
```

Interface Definition (cont'd)

WebIDL

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

WebIDL

```
typedef (RTCIceCandidate or RTCIceCandidateComplete) RTCIceGatherCandidate;
```

WebIDL

```
dictionary RTCIceCandidate {
    required DOMString              foundation;
    required unsigned long          priority;
    required DOMString              ip;
    required RTCIceProtocol         protocol;
    required unsigned short         port;
    required RTCIceCandidateType    type;
             RTCIceTcpCandidateType tcpType;
             DOMString              relatedAddress;
             unsigned short         relatedPort;
};
```

RTCIceCandidateComplete is a dictionary signifying that all RTCIceCandidates are gathered.

WebIDL

```
dictionary RTCIceCandidateComplete {
    boolean complete = true;
};
```

Interface Definition (cont’d)

WebIDL

```
dictionary RTCIceGatherOptions {
    RTCIceGatherPolicy       gatherPolicy;
    sequence<RTCIceServer> iceServers;
};
```

WebIDL

```
enum RTCIceGatherPolicy {
    "all",
    "nohost",
    "relay"
};
```

WebIDL

```
enum RTCIceGathererState {
    "new",
    "gathering",
    "complete",
    "closed"
};
```

RTCIceGatherer Interface

- ICE Candidates obtained via onlocalcandidate EventHandler (one candidate at a time) or via getLocalCandidates() method (multiple candidates at once)
- getLocalParameters(): Retrieves local ICE usernameFragment / password
- Errors
  - error Events are never fatal
  - Can only enter “closed” state by calling close()
- IceGatherer state machine
  - gather() method: transition from “new” to “gathering”
  - close() method: transition to “closed” (terminal state)
  - Transition from “complete” to “gathering” if a new interface comes up

RTP/RTCP Multiplexing

- For RTP/RTCP multiplexing, only one IceGatherer is needed
- For RTP/RTCP non-mux, construct rtpIceGatherer, then create RTCP IceGatherer (which shares same RTCIceGatherOptions)
  - rtpIceGatherer = new RTCIceGatherer(gatherOptions);
  - rtcpIceGatherer = rtpIceGatherer.createAssociatedGatherer();
    - rtpIceGatherer.component === “RTP”
    - rtcpIceGatherer.component === “RTCP”

Microsoft Edge: Interface Definition

WebIDL

```
[Constructor(RTCIceGatherOptions options)]
interface RTCIceGatherer : RTCStatsProvider {
    readonly        attribute RTCIceComponent component;
    RTCIceParameters          getLocalParameters ();
    sequence<RTCIceCandidate> getLocalCandidates ();
    RTCIceGatherer            createAssociatedGatherer ();
                    attribute EventHandler?   onerror;
                    attribute EventHandler?   onlocalcandidate;
};
```

Differences

- Edge IceGatherer gathers upon construction (no gather() method)
- No close() method.
- No onstatechange EventHandler (or state attribute)
- No support for forking (IceGatherer can only be associated with a single IceTransport)
- When ICE gathering is complete, onlocalcandidate emits a null candidate (not RTCIceCandidateComplete)

When is ICE Gathering Complete?

- In ORTC, completion of local ICE candidate gathering can be determined by:
  - onlocalcandidate providing event.candidate === RTCIceCandidateComplete OR
  - RTCIceGatherer.state === “complete”
  - Calling IceTransport.addRemoteCandidate(RTCIceComplete) tells the IceTransport that the remote peer has completed gathering.
- In Edge, ICE candidate gathering completion can *only* be determined by onlocalcandidate providing event.candidate == null
  - Calling IceTransport.addRemoteCandidate(null) tells the IceTransport that the remote peer has completed gathering.
  - Why? Because emission of a null event.candidate is widely supported in other browsers.

RTCIceTransport

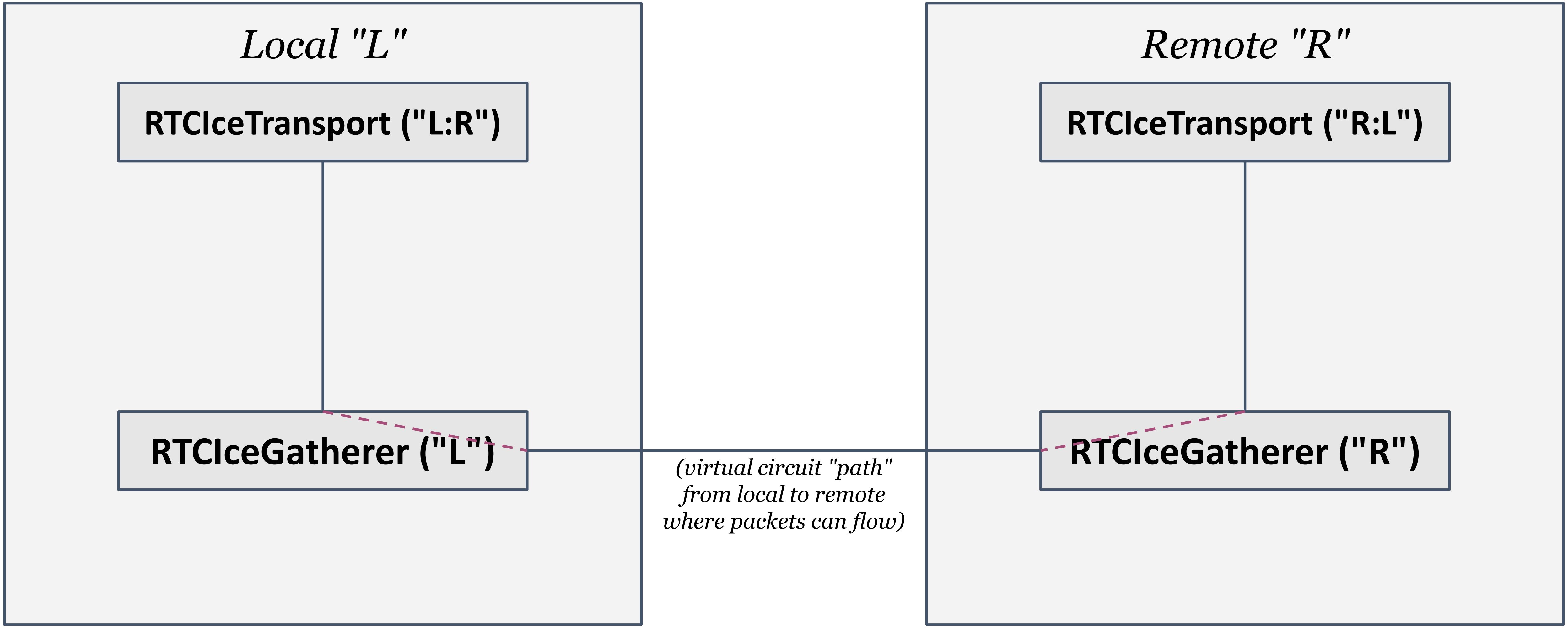
- Associated to a single local IceGatherer.
- Multiple IceTransport objects can share an IceGatherer (forking)
- Sends ICE connectivity tests to test communication paths between a local and remote peer
- Responds to incoming ICE connectivity checks with the specified remote username fragment
- Forms a virtual circuit over which DTLS and SRTP media packets can flow

Class Poll #2

- Question 1: Who is still awake?
- Question 1: For those still awake: Does it make sense to have separate objects for the IceGatherer and IceTransport?

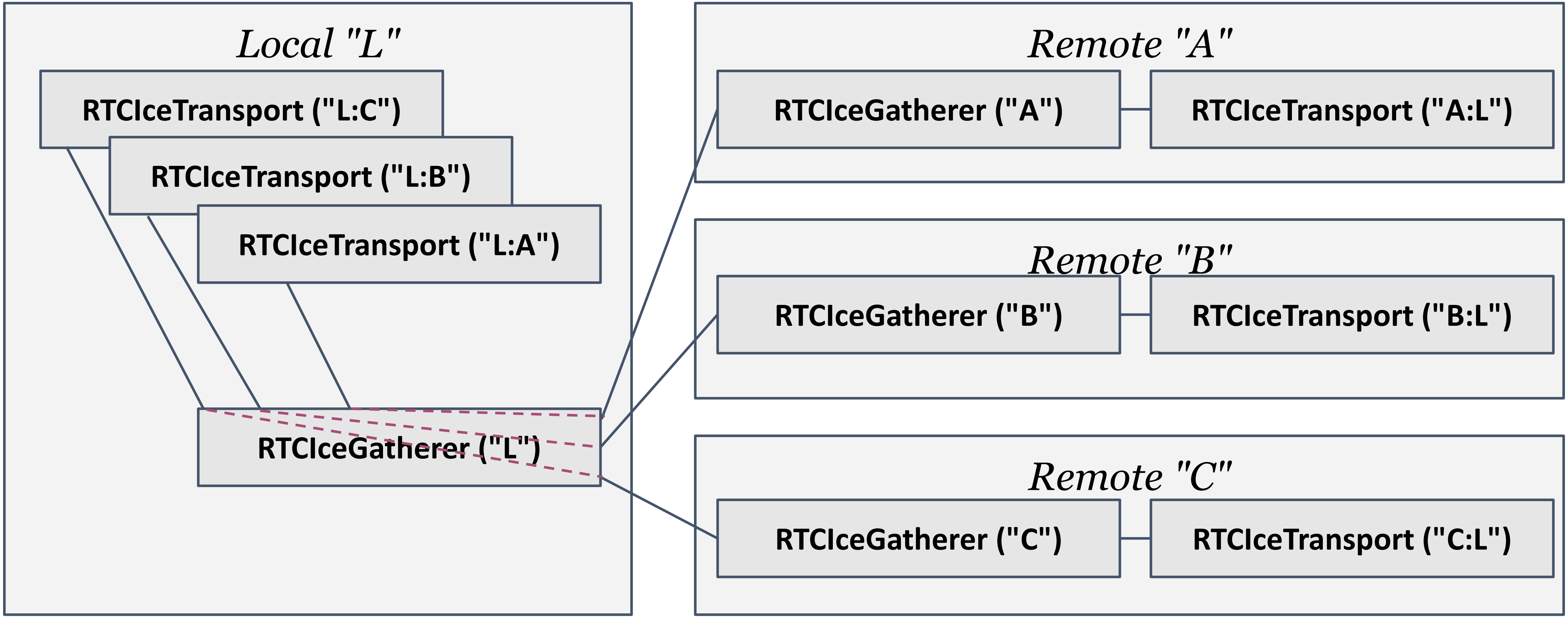
RTCIceGatherer/RTCIceTransport local / remote object relationships for "1-to-1" scenario

1:1 local IceTransports to local IceGatherer * 1:1 local to remote IceGatherers * 1:1 local IceTransport to remote IceTransport



RTCIceGatherer/RTCIceTransport local / remote object relationships for "forked 1-to-many" scenario

N:1 local IceTransports to local IceGatherer * 1:N local to remote IceGatherers * 1:1 local IceTransport to remote IceTransport



Interface Definition

WebIDL

```
[Constructor(optional RTCIceGatherer gatherer)]
interface RTCIceTransport : RTCStatsProvider {
    readonly attribute RTCIceGatherer?       iceGatherer;
    readonly attribute RTCIceRole            role;
    readonly attribute RTCIceComponent       component;
    readonly attribute RTCIceTransportState state;
    sequence<RTCIceCandidate> getRemoteCandidates();
    RTCIceCandidatePair?      getSelectedCandidatePair();
    void                      start(RTCIceGatherer gatherer,
                                    RTCIceParameters remoteParameters,
                                    optional RTCIceRole role = "controlled");
    void                      stop();
    RTCIceParameters?         getRemoteParameters();
    RTCIceTransport           createAssociatedTransport();
    void                      addRemoteCandidate(RTCIceGatherCandidate remoteCandidate);
    void                      setRemoteCandidates(sequence<RTCIceCandidate> remoteCandidates);
                attribute EventHandler       onstatechange;
                attribute EventHandler       oncandidatepairchange;
};
```

Interface Definition (cont'd)

WebIDL

```
enum RTCIceComponent {
    "RTP",
    "RTCP"
};
```

WebIDL

```
enum RTCIceRole {
    "controlling",
    "controlled"
};
```

WebIDL

```
enum RTCIceTransportState {
    "new",
    "checking",
    "connected",
    "completed",
    "disconnected",
    "failed",
    "closed"
};
```

RTCIceTransport Interface

- IceTransport constructed from an (optional) IceGatherer object.
  - If an IceGatherer is not provided in the constructor, it can be provided when the start() method is called.
- Once start() is called, an IceTransport can respond to incoming ICE connectivity checks based on its configured role and can also initiate its own checks.
  - To perform an ICE restart, call iceTransport.start again with a new ICE gatherer. This flushes all remote candidates, so addRemoteCandidate() or setRemoteCandidates() needs to be called again.
- For RTP/RTCP non-mux, construct rtpIceTransport, then create RTCP IceTransport
  - var rtcpIceTransport = rtpIceTransport.createAssociatedTransport()
  - rtpIceTransport.component === “RTP”
  - rtcpIceTransport.component === “RTCP”

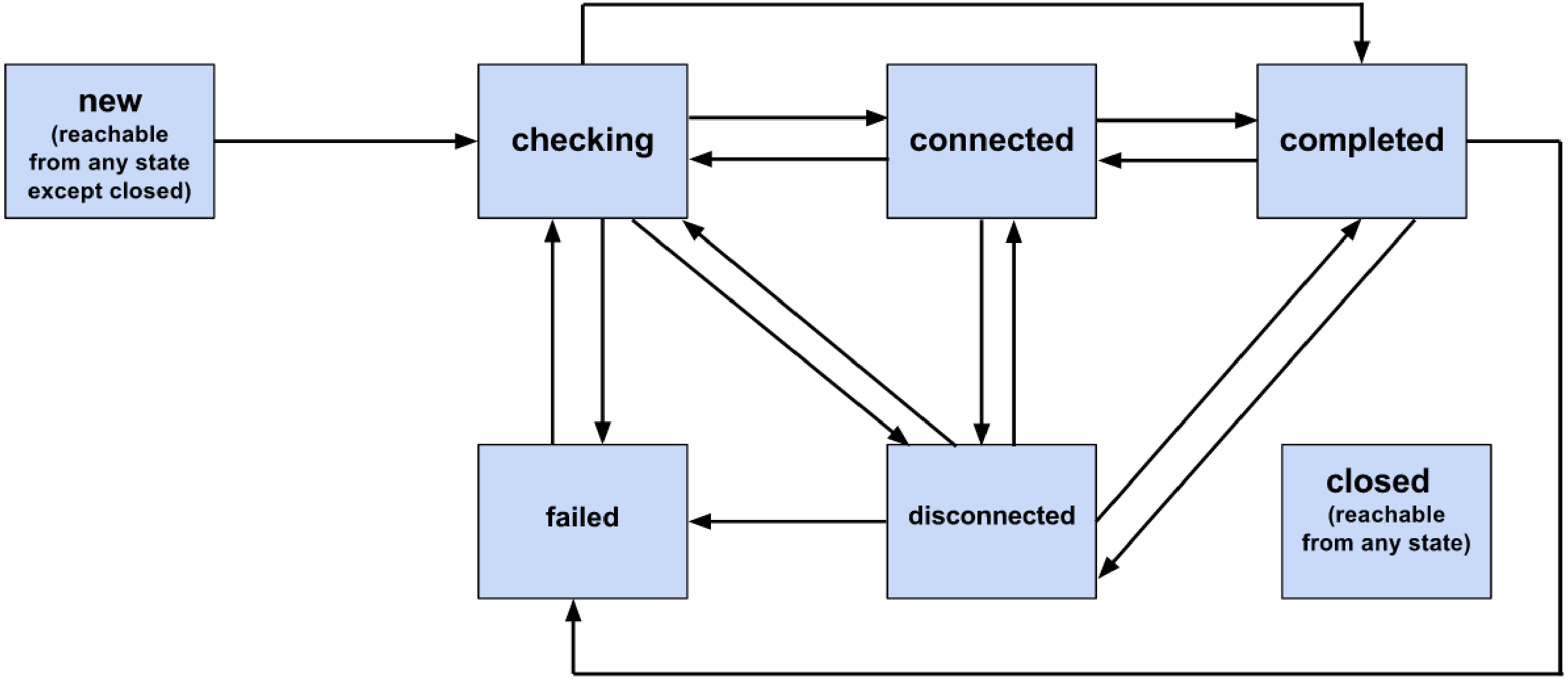
Microsoft Edge Interface Definition

```
[Constructor(optional RTCIceGatherer gatherer)]
interface RTCIceTransport : RTCStatsProvider {
    readonly        attribute RTCIceGatherer?      iceGatherer;
    readonly        attribute RTCIceRole           role;
    readonly        attribute RTCIceComponent      component;
    readonly        attribute RTCIceTransportState state;
    sequence<RTCIceCandidate> getRemoteCandidates ();
    RTCIceCandidatePair?      getNominatedCandidatePair ();
    void                      start (RTCIceGatherer gatherer, RTCIceParameters remoteParameters, optional
RTCIceRole role);
    void                      stop ();
    RTCIceParameters?         getRemoteParameters ();
    RTCIceTransport           createAssociatedTransport ();
    void                      addRemoteCandidate (RTCIceGatherCandidate remoteCandidate);
    void                      setRemoteCandidates (sequence<RTCIceCandidate> remoteCandidates);
                    attribute EventHandler?        onicestatechange;
};
```

Differences

- getNominatedCandidatePair instead of getSelectedCandidatePair()
- onicestatechange instead of onstatechange EventHandler
- No oncandidatepairchange EventHandler

RTCIceTransportState Transitions
http://draft.ortc.org/#rtcicetransportstate*



Forking

- The desire to support forking has had a major impact on the design of ORTC.
- Forking implies that several local IceTransport objects may utilize the same ufrag/password with multiple remote IceTransport objects
  - A local IceTransport is created for each single fork.
  - The need for multiple IceTransport objects to share a ufrag/password combination motivated the introduction of the IceGatherer object.
- Forking implies that several local DtlsTransport objects may utilize the same certificates/fingerprints with multiple remote DtlsTransport objects
  - The need to for multiple DtlsTransport objects to share certificates/fingerprints motivated inclusion of certificates in the DtlsTransport constructor.
- Use cases:
  - Multiple user agents (e.g. mobile device as well as a desk phone).
  - Peer-to-peer signaling over a broadcast medium (e.g. a chat room).
- Forking supported in ORTC Lib, but not in Edge implementation of ORTC.

IceGatherer/IceTransport Example

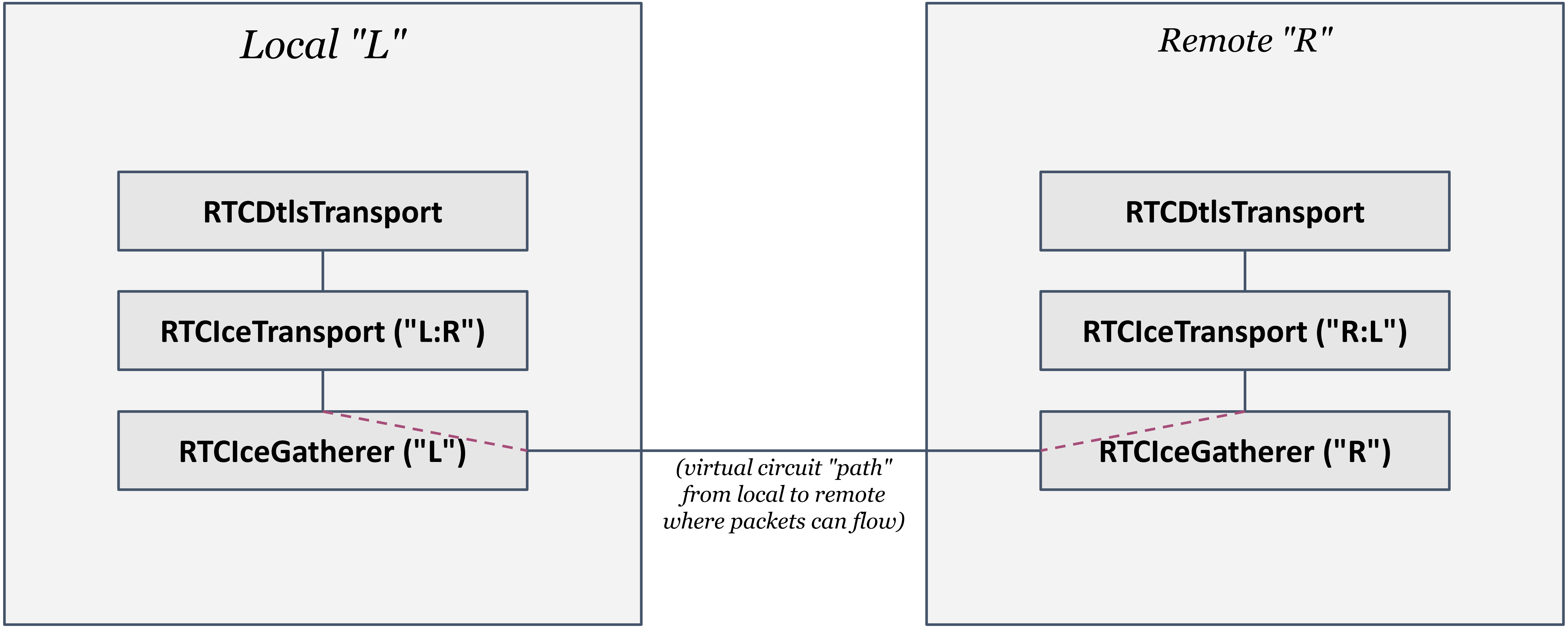
http://draft.ortc.org/#rtcicegatherer-example*

RTCDtlsTransport

- Supports forking via RTCCertificate interface (to enable forked DTLS transports to reuse the same certificate/fingerprint)
- Derives SRTP keys via DTLS/SRTP exchange
- Encrypts/decrypts data channel packets
- Associated to a single RTCIceTransport
- Sends/receives packets over virtual RTCIceTransport circuit path from local to remote party
- Requires fingerprint validation of DTLS certificate to prevent man-in-the-middle attacks

RTCIceGatherer / RTCIceTransport / RTCDtlsTransport local/remote object relationships

1:1 local DtlsTransports to local IceTransport * 1:1 local DtlsTransport to remote DtlsTransport



Interface Definition

WebIDL

```
typedef (octet[]) ArrayBuffer;
```

WebIDL

```
[Constructor(RTCIceTransport transport, sequence<RTCCertificate> certificates)]
interface RTCDtlsTransport : RTCStatsProvider {
    readonly attribute FrozenArray<RTCCertificate> certificates;
    readonly attribute RTCIceTransport             transport;
    readonly attribute RTCDtlsTransportState       state;
    RTCDtlsParameters       getLocalParameters();
    RTCDtlsParameters?      getRemoteParameters();
    sequence<ArrayBuffer>   getRemoteCertificates();
    void                    start(RTCDtlsParameters remoteParameters);
    void                    stop();
             attribute EventHandler                onstatechange;
             attribute EventHandler                onerror;
};
```

Interface Definition (cont'd)

WebIDL

```
dictionary RTCDtlsParameters {
    RTCDtlsRole                     role = "auto";
    sequence<RTCDtlsFingerprint> fingerprints;
};
```

WebIDL

```
dictionary RTCDtlsFingerprint {
    DOMString algorithm;
    DOMString value;
};
```

WebIDL

```
enum RTCDtlsTransportState {
    "new",
    "connecting",
    "connected",
    "closed",
    "failed"
};
```

RTCDtlsTransport Interface

- DtlsTransport constructed from an IceTransport and a sequence of certificates
  - Enables multiple DtlsTransport objects to be constructed with the same certificate(s) to enable forking.
- A newly constructed DtlsTransport MUST listen and respond to incoming DTLS packets before start() is called.
  - After the DTLS handshake completes (but before the remote fingerprint is verified) incoming media packets may be received.
- start() provides the remoteParameters needed to verify the remote fingerprint.
  - Incoming media MUST NOT be rendered prior to completion of remote fingerprint verification, to avoid man-in-the-middle attacks.

Microsoft Edge: Interface Definition

WebIDL

```
[Constructor(RTCIceTransport transport)]
interface RTCDtlsTransport : RTCStatsProvider {
    readonly          attribute RTCIceTransport        transport;
    readonly          attribute RTCDtlsTransportState state;
    RTCDtlsParameters  getLocalParameters ();
    RTCDtlsParameters? getRemoteParameters ();
    void               start (RTCDtlsParameters remoteParameters);
    void               stop ();
                      attribute EventHandler?          ondtlsstatechange;
                      attribute EventHandler?          onerror;
};
```

Differences

- No certificates passed in the constructor
- No getRemoteCertificates() method
- ondtlsstatechange instead of onstatechange EventHandler
- No “failed” state (transition to “closed” on receipt of a DTLS Alert)

RTCDtlsRole

WebIDL

```
enum RTCDtlsRole {
    "auto",
    "client",
    "server"
};
```

Enumeration description	
auto	The DTLS role is determined based on the resolved ICE role: the `controlled` role acts as the DTLS client, the `controlling` role acts as the DTLS server. Since `RTCDtlsRole` is initialized to `auto` on construction of an `RTCDtlsTransport` object, `transport.getLocalParameters().RTCDtlsRole` will have an initial value of `auto`.
client	The DTLS client role. A transition to `client` will occur if `start(remoteParameters)` is called with `remoteParameters.RTCDtlsRole` having a value of `server`. If `RTCDtlsRole` had previously had a value of `server` (e.g. due to the `RTCDtlsTransport` having previously received packets from a DTLS client), then the DTLS session is reset prior to transitioning to the `client` role.
server	The DTLS server role. If `RTCDtlsRole` has a value of `auto` and the `RTCDtlsTransport` receives a DTLS client_hello packet, `RTCDtlsRole` will transition to `server`, even before `start()` is called. A transition from `auto` to `server` will also occur if `start(remoteParameters)` is called with `remoteParameters.RTCDtlsRole` having a value of `client`.

RTCCertificate

WebIDL

```
interface RTCCertificate {
    readonly attribute DOMTimeStamp        expires;
    readonly attribute RTCDtlsFingerprint fingerprint;
    AlgorithmIdentifier      getAlgorithm();
    static Promise<RTCCertificate> generateCertificate(AlgorithmIdentifier keygenAlgorithm);
};
```

Notes

- RFC 4572-Update now allows multiple fingerprints per certificate (one of which needs to use the signature hash algorithm).
- WebRTC 1.0 updated to enable <FrozenArray> RTCDtlsFingerprint fingerprints

Why Mandate Certificates in the Constructor?

- With certificates supplied in the constructor:
  - Asynchronous RTCCertificate.generateCertificate() method used to generate a certificate.
  - After certificate(s) have been generated, DtlsTransport is constructed using the sequence of certificates.
  - Since certificates are pre-constructed, DtlsTransport constructor can return quickly and getLocalParameters() can be synchronous.

What if Certificates are Not Provided?

- Without certificates supplied in the constructor, a choice emerges:
  - DtlsTransport could create its own certificate(s) during construction.
    - This seems to work for Edge because it only runs on PCs (and envisages moving to generating ECDSA certificates by default in the future)
    - On a low cost mobile processor or in an IoT application, the UI thread could be blocked (> 20 ms), particularly if an RSA-2048 certificate is needed.
  - Alternatively, DtlsTransport constructor could return immediately with a null certificates attribute
    - Certificates/fingerprints guaranteed to be available when getLocalParameters() returns (would need to be async)

Class Poll #3

- Question 1: Should certificates be passed in the DtlsTransport constructor?
- Question 2: Do you care about SRTP/SDES support in Edge?

dtlsTransport Example

http://draft.ortc.org/#rtcicetransport-example1*

RTCSrtpSdesTransport (Edge Only)

- Negotiates SRTP keying via SDES/SRTP
- Associated to a single RTCIceTransport
- Sends/receives packets over virtual RTCIceTransport circuit path from local to remote party
- Supports non-multiplexed as well as multiplexed RTP/RTCP
- Support likely to be phased out at some point

Microsoft Edge: Interface Definition

```
[Constructor(RTCIceTransport transport, RTCSrtpSdesParameters
encryptParameters, RTCSrtpSdesParameters decryptParameters)]
interface RTCSrtpSdesTransport {
    readonly        attribute RTCIceTransport transport;
    static sequence<RTCSrtpSdesParameters> getLocalParameters ();
                    attribute EventHandler?   onerror;
};
```

Interface Definition (cont'd)

WebIDL

```
dictionary RTCSrtpSdesParameters {
            unsigned short                 tag;
            DOMString                      cryptoSuite;
            sequence<RTCSrtpKeyParam>  keyParams;
            sequence<DOMString>            sessionParams;
};
```

WebIDL

```
dictionary RTCSrtpKeyParam {
            DOMString      keyMethod;
            DOMString      keySalt;
            DOMString      lifetime;
            unsigned short mkiValue;
            unsigned short mkiLength;
};
```

srtpSdesTransport Example

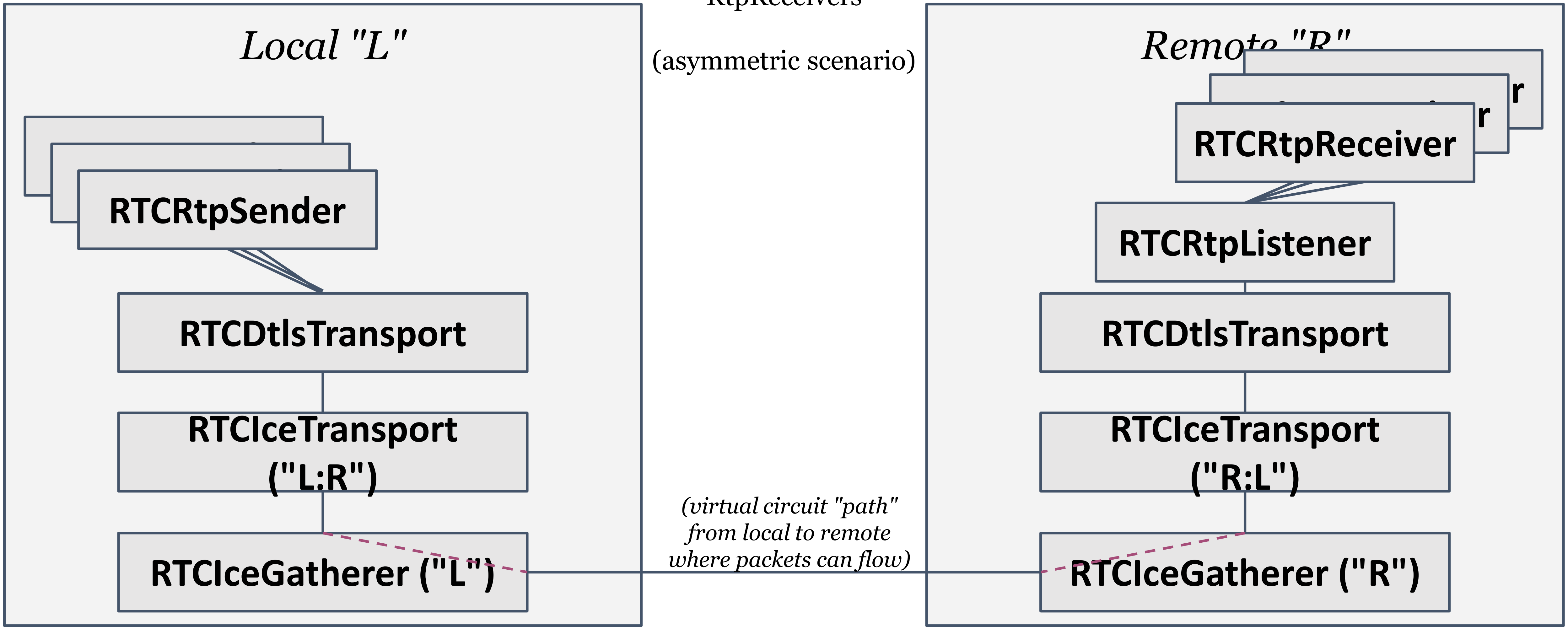
http://internaut.com:8080/~baboba/ms/msortc-rs.html#rtcsrtpsdestransport-example*

RTCRtpSender / RTCRtpReceiver

- Many senders / receivers can be associated to a single DtlsTransport (BUNDLE)
- Sender takes a single MediaStreamTrack
- Receiver emits a single MediaStreamTrack
- Media "kind" is typically "audio" or "video“ (but could also be “depth”)
- Capabilities describe what objects can be configured to do.
- Parameters describe how to encode/decode media on the wire.

RTCIceGatherer / RTCIceTransport / RTCDtlsTransport RTCRtpSender / Receiver / RTCRtpReceiver local/remote object relationships

N:1 local local RtpSender to DtlsTransports * 1:1 remote DtlsTransport to remote RtpListener * 1:N remote RtpListener to remote RtpReceivers



RtpSender Interface Definition

```
[Constructor(MediaStreamTrack track, RTCDtlsTransport transport, optional
RTCDtlsTransport rtcpTransport)]
interface RTCRtpSender : RTCStatsProvider {
    readonly attribute MediaStreamTrack  track;
    readonly attribute RTCDtlsTransport  transport;
    readonly attribute RTCDtlsTransport? rtcpTransport;
    void                 setTransport(RTCDtlsTransport transport,
                         optional RTCDtlsTransport rtcpTransport);
    Promise<void>        setTrack(MediaStreamTrack track);
    static RTCRtpCapabilities getCapabilities(DOMString kind);
    Promise<void>        send(RTCRtpParameters parameters);
    void                 stop();
             attribute EventHandler      onssrcconflict;
};
```

RTCRtpSender Interface

- send(*parameters*) method is now asynchronous to allow extended time for determining error conditions.
  - When send was synchronous, the onerror EventHandler could fire *after* send had returned (when a problem was discovered later).
- setTrack() method provided to allow seamless switching between input tracks (e.g. switching between a front and backward-facing camera).
- setTransport() method enables seamless response to a DTLS or ICE connection failure.
  - Construct new DtlsTransport(s) and then call setTransport() to send with them.
- No known implementation of the onssrcconflict EventHandler

Class Poll #4

- Question 1: Should send() be synchronous or asynchronous?
- Question 2: Does an SSRC conflict event make sense?

Microsoft Edge: RtpSender Interface Definition

WebIDL

```
typedef (RTCDtlsTransport or RTCSrtpSdesTransport) RTCTransport;
```

WebIDL

```
[Constructor(MediaStreamTrack track, RTCTransport transport)]
interface RTCRtpSender : RTCStatsProvider {
    readonly        attribute MediaStreamTrack track;
    readonly        attribute RTCTransport     transport;
    void                      setTransport (RTCTransport transport);
    void                      setTrack (MediaStreamTrack track);
    static RTCRtpCapabilities getCapabilities (optional DOMString kind);
    void                      send (RTCRtpParameters parameters);
    void                      stop ();
                    attribute EventHandler?    onerror;
};
```

Differences

- Uses RTCTransport instead of RTCDtlsTransport (for extensibility)
- Only supports non-mux RTP/RTCP with an RTCSrtpSdesTransport
- setTransport, setTrack and send are synchronous, rather than async
- onerror EventHandler
- No onssrcconflict EventHandler

RtpReceiver Interface Definition

WebIDL

```
[Constructor(DOMString kind, RTCDtlsTransport transport, optional RTCDtlsTransport rtcpTransport)]
interface RTCRtpReceiver : RTCStatsProvider {
    readonly attribute MediaStreamTrack  track;
    readonly attribute RTCDtlsTransport  transport;
    readonly attribute RTCDtlsTransport? rtcpTransport;
    void                                 setTransport(RTCDtlsTransport transport,
                                                      optional RTCDtlsTransport rtcpTransport);
    static RTCRtpCapabilities            getCapabilities(DOMString kind);
    Promise<void>                        receive(RTCRtpParameters parameters);
    sequence<RTCRtpContributingSource>   getContributingSources();
    void                                 stop();
};
```

RTCRtpReceiver Interface

- receive(*parameters*) method is asynchronous to allow extended time for determining error conditions.
  - When receive was synchronous, the onerror EventHandler could fire *after* receive had returned to indicate a problem discovered later.
- setTransport() method enables seamless response to a DTLS or ICE connection failure.
  - Construct new DtlsTransport(s) and then call setTransport() to send with them.
- getContributingSources() method used for:
  - Determining contributing sources (in a mixed stream): Supported in Edge
  - Determining audio levels (in a mixed stream or P2P): No known implementations

Microsoft Edge: RtpReceiver Interface Definition

WebIDL

```
[Constructor(RTCTransport transport, DOMString kind)]
interface RTCRtpReceiver : RTCStatsProvider {
    readonly        attribute MediaStreamTrack? track;
    readonly        attribute RTCTransport      transport;
    void                                  setTransport (RTCTransport transport);
    static RTCRtpCapabilities             getCapabilities (optional DOMString kind);
    sequence<RTCRtpContributingSource>    getContributingSources ();
    void                                  receive (RTCRtpParameters parameters);
    void                                  stop ();
                    attribute EventHandler?     onerror;
};
```

Differences

- Uses RTCTransport instead of RTCDtlsTransport
- Only supports non-mux RTP/RTCP with an RTCSrtpSdesTransport
- setTransport and receive are synchronous, rather than async
- onerror EventHandler

RTCRtpCapabilities

WebIDL

```
dictionary RTCRtpCapabilities {
    sequence<RTCRtpCodecCapability> codecs;
    sequence<RTCRtpHeaderExtension> headerExtensions;
    sequence<DOMString>              fecMechanisms;
};
```

WebIDL

```
dictionary RTCRtpCodecCapability {
    DOMString                     name;
    DOMString                     mimeType;
    DOMString                     kind;
    unsigned long                 clockRate;
    payloadtype                   preferredPayloadType;
    unsigned long                 maxptime;
    unsigned long                 ptime;
    unsigned long                 numChannels;
    sequence<RTCRtcpFeedback>     rtcpFeedback;
    Dictionary                    parameters;
    Dictionary                    options;
    unsigned short                maxTemporalLayers = 0;
    unsigned short                maxSpatialLayers = 0;
    boolean                       svcMultiStreamSupport;
};
```

RTCRtpCapabilities

- Indicates what codecs, header extensions and FEC mechanisms are supported by RtpSenders and RtpReceivers.
  - For each codec, information is provided, including name/mimeType, clockRate, maxptime/ptime/numChannels (for audio), rtcpFeedback, etc.
    - Parameters and options dictionaries provided for codec-specific capabilities.
  - codecs attribute includes more than just media codecs
    - RED, Retransmission (RTX), DTMF, CN and FEC mechanisms (e.g. “ulpfec”, “flexfec”, etc.) are included as well.
    - One RTX entry for each codec that can be retransmitted.

Demo

- Object Inventory and Capability dumper:
  - http://internaut.com:8080/~baboba/iit-tutorial/cap-dumper/
- Tells you what objects a browser supports
- Dumps RtpSender/RtpReceiver.getCapabilities(*kind*) for “audio” and “video”

RTCRtpParameters

WebIDL

```
dictionary RTCRtpParameters {
             DOMString                                  muxId = "";
    required sequence<RTCRtpCodecParameters>            codecs;
             sequence<RTCRtpHeaderExtensionParameters>  headerExtensions;
             sequence<RTCRtpEncodingParameters>         encodings;
             RTCRtcpParameters                          rtcp;
             RTCDegradationPreference                   degradationPreference = "balanced";
};
```

WebIDL

```
dictionary RTCRtpCodecParameters {
    required DOMString                   name;
             DOMString                   mimeType;
    required payloadtype                 payloadType;
             unsigned long               clockRate;
             unsigned long               maxptime;
             unsigned long               ptime;
             unsigned long               numChannels;
             sequence<RTCRtcpFeedback>   rtcpFeedback;
             Dictionary                  parameters;
};
```

RTCRtpParameters

- Indicates what codecs, header extensions, encodings, rtcp settings, etc. are to be used for sending and receiving.
  - Codec parameters resemble codec capabilities.
    - Parameters dictionaries are provided for codec-specific settings.
  - codecs attribute includes more than just media codecs
    - RED, Retransmission (RTX), DTMF, CN and FEC mechanisms (e.g. “ulpfec”, “flexfec”, etc.) are included as well.
    - An RTX entry is provided for each codec that supports retransmission.

Capabilities Exchange

- RTCRtpCapabilities designed for “capabilities exchange”
  - Peers exchange sender/receiver capabilities (including preferred Payload Types)
  - Capabilities used to compute RTCRtpParameters passed to send(*parameters*) and receive(*parameters*)
  - Intersection of codecs, header extensions, feedback messages can be computed without specific knowledge
    - sending codecs, header extensions and feedback messages computed from intersection of local sender capabilities and remote receiver capabilities.
    - receiving codecs, header extensions and feedback messages computed from intersection of local receiver capabilities and remote sender capabilities.

Capabilities Exchange (cont'd)

- Determining codec parameters is trickier.
- Several important sender codec parameters can be determined from receiver codec capabilities. Examples:
  - Opus: receiver useInbandFec capability copied to sender useInbandFec parameter
  - H.264/AVC: receiver profileLevelId capability copied to sender profileLevelId parameter
  - VP8/VP9: receiver maxFr/maxFs capability is copied to sender maxFr/maxFs parameter

Class Poll #5

- Question 1: Does “capabilities exchange” signaling make sense?

RtpSender/RtpReceiver Example
http://internaut.com:8080/~baboba/ms/msortc-rs.html#rtcrtpreceiver-example*

RTCRtpEncodingParameters

WebIDL

```
dictionary RTCRtpEncodingParameters {
    unsigned long          ssrc;
    payloadtype            codecPayloadType;
    RTCRtpFecParameters    fec;
    RTCRtpRtxParameters    rtx;
    RTCPriorityType        priority;
    unsigned long          maxBitrate;
    double                 resolutionScale;
    double                 framerateScale;
    unsigned long          maxFramerate;
    boolean                active = true;
    DOMString              encodingId;
    sequence<DOMString>    dependencyEncodingIds;
};
```

Attribute	Type	Receiver/Sender
ssrc	unsigned long	Receiver/Sender
codecPayloadType	payloadType	Receiver/Sender
fec	RTCRtpFecParameters	Receiver/Sender
rtx	RTCRtpRtxParameters	Receiver/Sender
priority	RTCPriorityType	Sender
maxBitrate	unsigned long	Sender
resolutionScale	double	Sender
framerateScale	double	Sender
maxFramerate	unsigned long	Sender
active	boolean	Receiver/Sender
encodingId	DOMString	Receiver/Sender
dependencyEncodingIds	sequence<DOMString>	Receiver/Sender

Encoding Parameters (receiver/sender)

- *ssrc* (if provided) specifies the specific SSRC to be used for the encoding. If omitted when sending, the browser chooses.
- *codecPayloadType* refers to a codec included in the codecs sequence. If omitted, the first codec in the sequence is implied.
- *fec* indicates whether this encoding is protected with forward error correction (and if so, what ssrc is used for FEC).
- *rtx* indicates whether this encoding is retransmitted (and if so, what ssrc is used for retransmission).

Encoding Parameters (receiver/sender)

- *active* indicates whether this encoding is being sent or received.
  - Can be used to implement “hold” scenarios.
- *encodingId* indicates the RID to be used for this encoding.
- *dependencyEncodingIds* indicate the other encodings that this encoding depends on (for scalable video coding only)

Encoding Parameters (sender only)

- *priority* indicates how QoS marking is to be applied for this encoding.
  - No implementations of this parameter.
- *maxBitrate* indicates the maximum bitrate which this encoding can use.
- *resolutionScale* indicates the how much the video width/height should be decreased, compared with sender.track.
  - 2.0 means send at half width/height.
  - Used in simulcast and scalable video coding.

Encoding Parameters (sender cont'd)

- *framerateScale* indicates the how much the framerate should be decreased, compared with sender.track.
  - 2.0 means send at half the framerate.
  - Primarily for temporal scalability.
  - No implementations of this parameter.
- *maxFramerate* indicates the maximum framerate to be used for this encoding.
  - Usage scenario: simulcast screen sharing.
  - No implementations of this parameter.

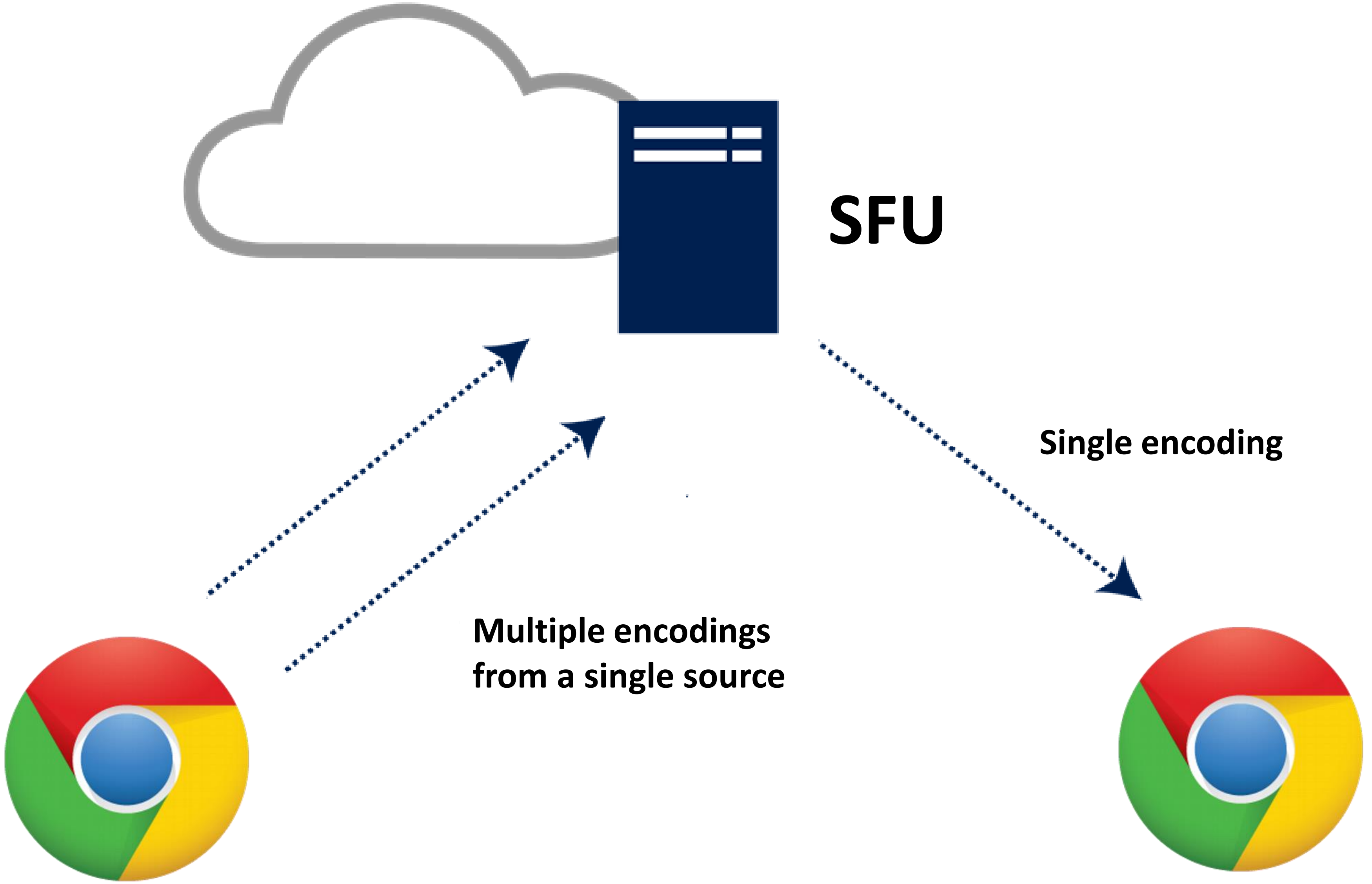
Class Poll #6

- Question 1: Who is still awake?
- Question 2: Of those still awake, how many know what simulcast and scalable video coding is?

Simulcast

- Simulcast involves sending (and possibly receiving) multiple streams that differ in characteristics such as resolution or frame rate.
- Typical scenario:
  - Browser sends multiple streams (e.g. a high resolution and a low resolution stream) to a Selective Forwarding Unit (SFU).
  - SFU selects which stream to forward to each conference participant.
    - Mobile device receives the low resolution stream
    - PC receives the high resolution stream
- Both WebRTC 1.0 and ORTC object models support sending multiple streams.
- ORTC also supports receiving multiple streams (and outputting a single track).

Typical Simulcast Scenario

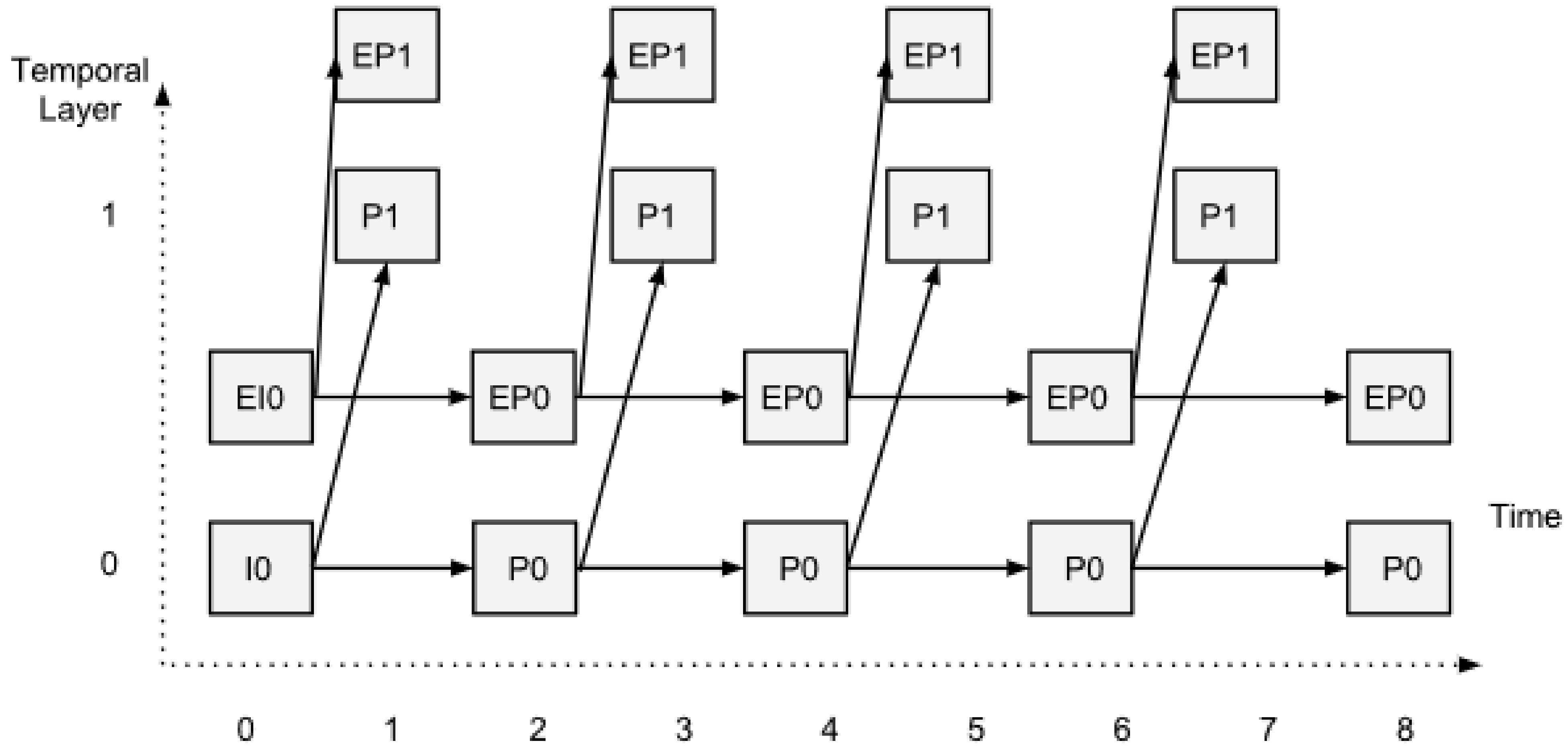


Scalable Video Coding

- Codecs implemented in WebRTC support scalable video coding (SVC):
  - VP8: temporal scalability
  - VP9: temporal and spatial scalability
  - H.264/SVC: temporal scalability
- SVC entering the mainstream
  - Utilized in products by Google (Hangouts), Microsoft (Skype), Vidyo, Polycom, Avaya/RADVISION, LifeSize, etc.
- ORTC makes it possible to enable SVC in the RtpSender/encoder
  - Developer can specify the number of layers but typically just wants the browser to “do the right thing” (send as many layers as conditions permit).
  - As long as the decoder supports SVC, no need to configure the RtpReceiver.

Simulcast + Scalable Video Coding

(2 layers spatial simulcast + 2 layers temporal scalability)



2-layer simulcast + 2 layers temporal scalability

```
var encodings = [{
  // Low resolution base layer (half the input framerate, half the input resolution)
  encodingId: "0",
  resolutionScale: 2.0,
  framerateScale: 2.0
}, {
  // High resolution Base layer (half the input framerate, full input resolution)
  encodingId: "E0",
  resolutionScale: 1.0,
  framerateScale: 2.0
}, {
  // Temporal enhancement to the low resolution base layer (full input framerate, half resolution)
  encodingId: "1",
  dependencyEncodingIds: ["0"],
  resolutionScale: 2.0,
  framerateScale: 1.0
}, {
  // Temporal enhancement to the high resolution base layer (full input framerate and resolution)
  encodingId: "E1",
  dependencyEncodingIds: ["E0"],
  resolutionScale: 1.0,
  framerateScale: 1.0
}];
```

RTCRtpListener

- Routes incoming media packets to correct RTCRtpReceiver
- Emits "unknown" media event when no RTCRtpReceiver attached that handles the incoming media
- RTCRtpListener routing engine exists whether it is created explicitly or not if an RTCRtpReceiver is used

Interface Definition

WebIDL

```
[Constructor(RTCDtlsTransport transport)]
interface RTCRtpListener {
    readonly attribute RTCDtlsTransport transport;
             attribute EventHandler     onunhandledrtp;
};
```

WebIDL

```
[Constructor(DOMString type, RTCRtpUnhandledEventInit eventInitDict)]
interface RTCRtpUnhandledEvent : Event {
    readonly attribute DOMString     muxId;
    readonly attribute DOMString     rid;
    readonly attribute payloadtype   payloadType;
    readonly attribute unsigned long ssrc;
};
```

Class Poll #7

- Question 1: Is the RtpListener a good idea?
- Question 2: Do you want to hear about the DataChannel and IceTransportController objects or take a break?

RTP Matching Rules (draft-ietf-rtcweb-jsep Section 7)

Construct a table mapping MID to RtpReceiver for each RtpReceiver configured to receive from this transport.

Construct a table mapping SSRC to RtpReceiver for each RtpReceiver configured to receive from this transport and for each SSRC that RtpReceiver is configured to receive. Some of the SSRCs may be presesnt in the m= section corresponding to that RtpReceiver in the remote description.

Construct a table mapping payload type to RtpReceiver for each RtpReceiver configured to receive from this transport and for each payload type that RtpReceiver is configured to receive. The payload types of a given RtpReceiver are found in the m= section corresponding to that RtpReceiver in the local description. If any payload type could map to more than one RtpReceiver, map to the RtpReceiver whose m= section appears earliest in the local description.

RTP Matching Rules (cont'd)

For each RTP packet received, the following steps MUST be followed to route the packet:

If the packet has a MID and that MID is not in the table mapping MID to RtpReceiver, drop the packet and stop.

If the packet has a MID and that MID is in the table mapping MID to RtpReceiver, update the SSRC mapping table to include an entry mapping the packet's SSRC to the RtpReceiver.

If the packet's SSRC is in the SSRC mapping table, route the packet to the mapped RtpReceiver and stop.

If the packet's payload type is in the payload type table, update the the SSRC mapping table to include an entry mapping the packet's SSRC to the RtpReceiver. Deliver the packet to the RtpReceiver and stop.

Otherwise, fire an rtpunhandledevent.

Class Poll #8

- Question 1: Can multiple receivers be set up to receive RTP packets with the same payload types?
- Question 2: Does the Payload Type table make sense?

RTCDataChannel

- Very similar to data channel API in WebRTC 1.0 (intentional)
- Represents a bi-directional data channel between two peers.
- Enables sending data over an RTCDataTransport.
- In future, can be extended to support additional transports (e.g. QUIC in addition to SCTP over DTLS)

Interface Definition

WebIDL

```
[Constructor(RTCDataTransport transport, RTCDataChannelParameters parameters)]
interface RTCDataChannel : EventTarget {
    readonly attribute RTCDataTransport     transport;
    readonly attribute RTCDataChannelState readyState;
    readonly attribute unsigned long        bufferedAmount;
             attribute unsigned long        bufferedAmountLowThreshold;
             attribute DOMString            binaryType;
    RTCDataChannelParameters getParameters();
    void                     close();
             attribute EventHandler         onopen;
             attribute EventHandler         onbufferedamountlow;
             attribute EventHandler         onerror;
             attribute EventHandler         onclose;
             attribute EventHandler         onmessage;
    void                     send(USVString data);
    void                     send(Blob data);
    void                     send(ArrayBuffer data);
    void                     send(ArrayBufferView data);
};
```

Interface Definition (cont'd)

WebIDL

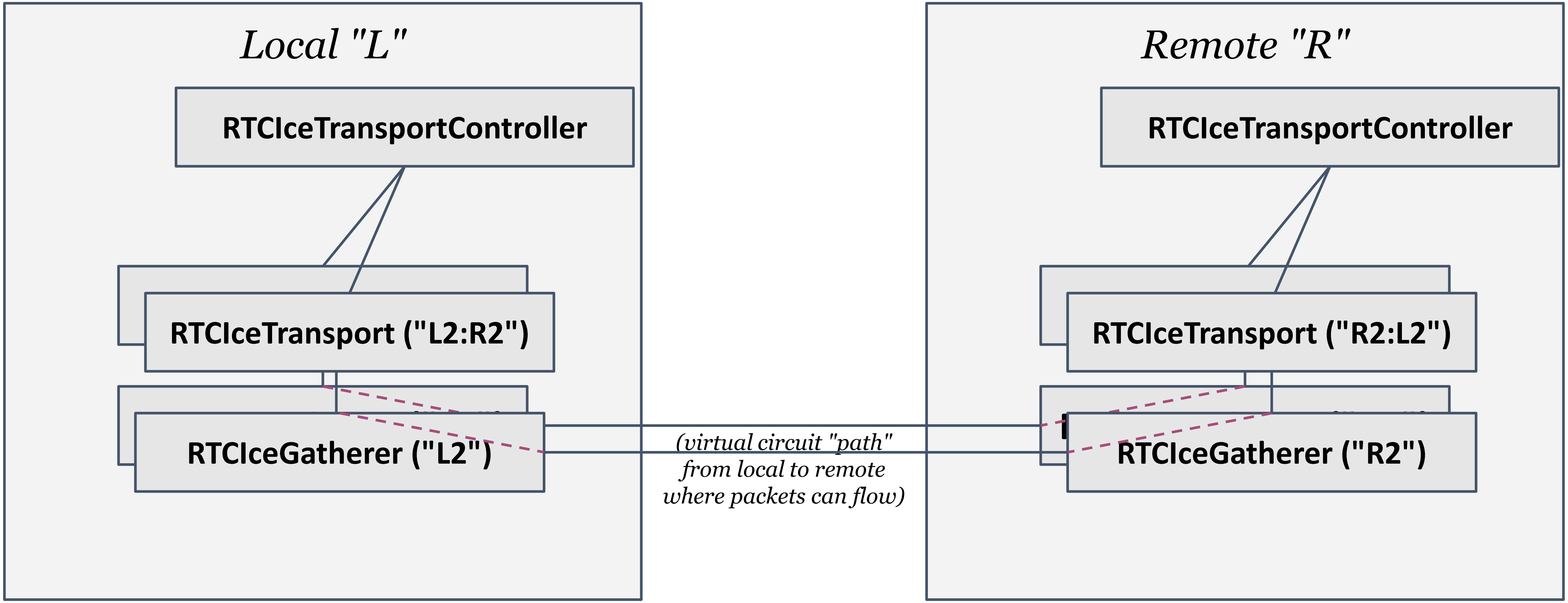
```
[Constructor(RTCDtlsTransport transport, optional unsigned short port)]
interface RTCSctpTransport : RTCDataTransport {
    readonly attribute RTCDtlsTransport         transport;
    readonly attribute RTCSctpTransportState    state;
    readonly attribute unsigned short           port;
    static RTCSctpCapabilities getCapabilities();
    void                   start(RTCSctpCapabilities remoteCaps);
    void                   stop();
             attribute EventHandler             ondatachannel;
             attribute EventHandler             onstatechange;
};
```

RTCIceTransportController

- Coordinates ICE freezing across IceTransports
- Manages collective bandwidth across IceTransports
- Contains 1 or more relationships to IceTransports
- Used mostly for when audio and video tracks are streamed non-muxed on different virtual IceTransport circuits

RTCIceTransport / RTCIceTransportController local/remote object relationships

RTPIceTransportController does not route packets. It's not in the media pipeline.
It coordinates the RTCIceTransports ICE freezing.



Interface Definition

WebIDL

```
[Constructor()]
interface RTCIceTransportController {
    void                           addTransport(RTCIceTransport transport,
                                                optional unsigned long index);
    sequence<RTCIceTransport> getTransports();
};
```

Questions?