W3C WebRTC WG Meeting

April 26, 2018
8 AM Pacific Time

Chairs: Stefan Hakansson
Bernard Aboba
Harald Alvestrand

W3C WG IPR Policy

- This group abides by the W3C Patent Policy https://www.w3.org/Consortium/Patent-Policy/
- Only people and companies listed at https://www.w3.org/2004/01/pp-impl/47318/status are allowed to make substantive contributions to the WebRTC specs

Welcome!

- Welcome to the interim meeting of the W3C WebRTC WG!
- During this meeting, we hope to:
  - Discuss the implementation status of WebRTC 1.0.
  - Go over the status of WPT webrtc Issues and PRs
  - Go over principles for WPT test design
  - Provide an example test.
  - Discuss cross-browser testing using WPT.
  - Provide an update on advanced testing using the KITE framework.

June f2f (Stockholm, Sweden)

- Date: June 19-20, 2018 at the Google Stockholm offices.
- Remote participation will be supported.

About this Virtual Meeting

Information on the meeting:

- Meeting info:
  - https://www.w3.org/2011/04/webrtc/wiki/April_26_2018
- Link to latest drafts:
  - https://w3c.github.io/mediacapture-main/
  - https://w3c.github.io/webrtc-pc/
  - https://w3c.github.io/mediacapture-screen-share/
  - https://w3c.github.io/webrtc-stats/
- Link to Slides has been published on WG wiki
- Scribe? IRC http://irc.w3.org/ Channel: #webrtc
- The meeting is being recorded.
- WebEx info here

For Discussion Today

- WebRTC 1.0 Implementation Status (Dom)
  - Confluence tests
  - Adapter vs. non-adapter results (Bernard)
- WPT
  - wpt/webrtc test status
  - WPT Issues and PRs
  - Thoughts on Testing (Fippo)
  - Test principles (Fippo)
  - Example test (Fippo)
- Cross-browser testing (Lennart)
- Update on the KITE framework (Dr. Alex)

WebRTC 1.0 Implementation Status (Dom)

- Question: How can we measure implementation status of WebRTC 1.0 CR?
  - Web-platform-tests dashboard (https://wpt.fyi/webrtc) “does not contain useful metrics for evaluation or comparison of web platform features”
- Web confluence project:
  - Looks at properties and methods exposed by browsers: https://web-confluence.appspot.com/#!/
  - Caveat: no guarantee that a widely-supported API is interoperable in its details, or will remain part of the web platform.
  - Tool that extracts data from the confluence tracker: https://dontcallmedom.github.io/webrtc-impl-tracker/?webrtc
  - Issue: data on RTCIdentity* is incorrect, probably because these interfaces are only exposed in the non-default global

Confluence Tracker Tool

Interface	Member		Chrome	Edge	Firefox	Safari
RTCPeerConnection	createOffer	4/4	40.0.2214.93+	15.15063+	45.0+	11.0.3+
	createAnswer	4/4	40.0.2214.93+	15.15063+	45.0+	11.0.3+
	setLocalDescription	4/4	40.0.2214.93+	15.15063+	45.0+	11.0.3+
	localDescription	4/4	43.0.2357.65+	15.15063+	45.0+	11.0.3+
	currentLocalDescription	2/4			57.0+	11.0.3+
	pendingLocalDescription	2/4			57.0+	11.0.3+
	setRemoteDescription	4/4	40.0.2214.93+	15.15063+	45.0+	11.0.3+
	remoteDescription	4/4	43.0.2357.65+	15.15063+	45.0+	11.0.3+
	currentRemoteDescription	2/4			57.0+	11.0.3+
	pendingRemoteDescription	2/4			57.0+	11.0.3+
	addIceCandidate	4/4	40.0.2214.93+	15.15063+	45.0+	11.0.3+
	signalingState	4/4	43.0.2357.65+	15.15063+	45.0+	11.0.3+
	iceGatheringState	4/4	43.0.2357.65+	15.15063+	45.0+	11.0.3+
	iceConnectionState	4/4	43.0.2357.65+	15.15063+	45.0+	11.0.3+
	connectionState	1/4				11.0.3+
	canTrickleIceCandidates	2/4		15.15063+	47.0+	
	getDefaultIceServers	0/4				
	getConfiguration	3/4		15.15063+	45.0+	11.0.3+
	setConfiguration	2/4	58.0.3029.81+			11.0.3+
	close	4/4	40.0.2214.93+	15.15063+	45.0+	11.0.3+
	onnegotiationneeded	4/4	43.0.2357.65+	15.15063+	45.0+	11.0.3+
	onicecandidate	4/4	43.0.2357.65+	15.15063+	45.0+	11.0.3+
	onicecandidateerror	0/4				
	onsignalingstatechange	4/4	43.0.2357.65+	15.15063+	45.0+	11.0.3+
	oniceconnectionstatechange	4/4	43.0.2357.65+	15.15063+	45.0+	11.0.3+
	onicegatheringstatechange	4/4	59.0.3071.86+	15.15063+	53.0+	11.0.3+
	onconnectionstatechange	1/4				11.0.3+
	generateCertificate	2/4	49.0.2623.75+		45.0+	
	getSenders	3/4	64.0.3282.119+		45.0+	11.0.3+
	getReceivers	3/4	59.0.3071.86+		45.0+	11.0.3+
	getTransceivers	2/4			59.0+	11.0.3+
	addTrack	3/4	64.0.3282.119+		45.0+	11.0.3+
	removeTrack	3/4	64.0.3282.119+		45.0+	11.0.3+
	addTransceiver	2/4			59.0+	11.0.3+
	ontrack	3/4	64.0.3282.119+		46.0+	11.0.3+
	sctp	0/4				
	createDataChannel	3/4	40.0.2214.93+		45.0+	11.0.3+
	ondatachannel	3/4	43.0.2357.65+		45.0+	11.0.3+
	getStats	4/4	40.0.2214.93+	15.15063+	45.0+	11.0.3+
	onstatsended	0/4				
	setIdentityProvider	1/4			45.0+	
	getIdentityAssertion	1/4			45.0+	
	peerIdentity	1/4			45.0+	
	idpLoginUrl	1/4			45.0+	

- No browsers currently support:
  - getDefaultIceServers
  - onicecandidateerror
  - RTCPeerConnection.sctp/RTCSctpTransport
  - Onstatsended
  - RTCPeerConnectionIceErrorEvent
  - RTCPeerConnectionIceEvent.url
  - RTCCertificate.getSupportedAlgorithms
  - RTCIceCandidate attributes (??)
  - RTCRtpTransceiver.setCodecPreferences
  - RTCDtlsTransport.onstatechange (->ondtlsstatechange)
  - RTCDataChannel.priority
- Only one browser supports
  - RTCRtpTransceiver.currentDirection
  - connectionState
  - onconnectionStateChange
  - Identity API
  - RTCDataChannel.maxPacketLifetime
  - Various RtpSender/RtpReceiver/DtlsTransport/IceTransport methods

Source: https://dontcallmedom.github.io/webrtc-impl-tracker/?webrtc

Confluence Tracker Tool (cont'd)

RTCSessionDescription	type	4/4	43.0.2357.65+	15.15063+	45.0+	11.0.3+
	sdp	4/4	43.0.2357.65+	15.15063+	45.0+	11.0.3+
	toJSON	4/4	43.0.2357.65+	15.15063+	45.0+	11.0.3+
RTCIceCandidate	candidate	4/4	43.0.2357.65+	15.15063+	45.0+	11.0.3+
	sdpMid	4/4	43.0.2357.65+	15.15063+	45.0+	11.0.3+
	sdpMLineIndex	4/4	43.0.2357.65+	15.15063+	45.0+	11.0.3+
	foundation	0/4				
	component	0/4				
	priority	0/4				
	ip	0/4				
	protocol	0/4				
	port	0/4				
	type	0/4				
	tcpType	0/4				
	relatedAddress	0/4				
	relatedPort	0/4				
	usernameFragment	0/4				
	toJSON	4/4	43.0.2357.65+	15.15063+	45.0+	11.0.3+
RTCPeerConnectionIceEvent	candidate	3/4	56.0.2924.76+	15.15063+	45.0+	
	url	0/4				
RTCPeerConnectionIceErrorEvent	hostCandidate	0/4				
	url	0/4				
	errorCode	0/4				
	errorText	0/4				
RTCCertificate	expires	2/4	49.0.2623.75+		45.0+	
	getSupportedAlgorithms	0/4				
	getFingerprints	1/4	61.0.3163.79+			
RTCRtpSender	track	4/4	64.0.3282.119+	13.10586+	45.0+	11.0.3+
	transport	1/4		13.10586+		
	rtcpTransport	1/4		13.10586+		
	getCapabilities	1/4		13.10586+		
	setParameters	1/4			46.0+	
	getParameters	2/4			46.0+	11.0.3+
	replaceTrack	3/4	65.0.3325.146+		45.0+	11.0.3+
	getStats	1/4			55.0+	
	dtmf	1/4			52.0+	
RTCRtpReceiver	track	4/4	59.0.3071.86+	13.10586+	45.0+	11.0.3+
	transport	1/4		13.10586+		
	rtcpTransport	1/4		13.10586+		
	getCapabilities	1/4		13.10586+		
	getParameters	1/4				11.0.3+
	getContributingSources	3/4	59.0.3071.86+	13.10586+	59.0+	
	getSynchronizationSources	1/4			59.0+	
	getStats	1/4			55.0+	

Confluence Tracker Tool (cont'd)

RTCRtpTransceiver	mid	2/4			59.0+	11.0.3+
	sender	2/4			59.0+	11.0.3+
	receiver	2/4			59.0+	11.0.3+
	stopped	2/4			59.0+	11.0.3+
	direction	2/4			59.0+	11.0.3+
	currentDirection	1/4			59.0+	
	stop	2/4			59.0+	11.0.3+
	setCodecPreferences	0/4				
RTCDtlsTransport	transport	1/4		13.10586+		
	state	1/4		13.10586+		
	getRemoteCertificates	1/4		13.10586+		
	onstatechange	0/4				
	onerror	1/4		13.10586+		
RTCIceTransport	role	1/4		13.10586+		
	component	1/4		13.10586+		
	state	2/4		13.10586+		11.0.3+
	gatheringState	1/4				11.0.3+
	getLocalCandidates	0/4				
	getRemoteCandidates	1/4		13.10586+		
	getSelectedCandidatePair	0/4				
	getLocalParameters	0/4				
	getRemoteParameters	1/4		13.10586+		
	onstatechange	0/4				
	ongatheringstatechange	0/4				
	onselectedcandidatepairchange	0/4				
RTCTrackEvent	receiver	3/4	64.0.3282.119+		46.0+	11.0.3+
	track	3/4	64.0.3282.119+		46.0+	11.0.3+
	streams	3/4	64.0.3282.119+		46.0+	11.0.3+
	transceiver	2/4			59.0+	11.0.3+
RTCSctpTransport	transport	0/4				
	state	0/4				
	maxMessageSize	0/4				
	onstatechange	0/4				
RTCDataChannel	label	2/4	56.0.2924.76+			11.0.3+
	ordered	2/4	56.0.2924.76+			11.0.3+
	maxPacketLifeTime	1/4				11.0.3+
	maxRetransmits	2/4	56.0.2924.76+			11.0.3+
	protocol	2/4	56.0.2924.76+			11.0.3+
	negotiated	2/4	56.0.2924.76+			11.0.3+
	id	2/4	56.0.2924.76+			11.0.3+
	priority	0/4				
	readyState	2/4	56.0.2924.76+			11.0.3+
	bufferedAmount	2/4	56.0.2924.76+			11.0.3+
	bufferedAmountLowThreshold	2/4	56.0.2924.76+			11.0.3+
	onopen	2/4	56.0.2924.76+			11.0.3+

WebRTC 1.0 Implementation Status (Bernard)

- Question: How do results change with adapter.js?
  - http://bluebox.internaut.com:8080/~baboba/cluecon-tutorial/cap-dumper/
  - http://bluebox.internaut.com:8080/~baboba/cluecon-tutorial/cap-dumper/no-adapter.html
- Answer: adapter.js adjusts for:
  - Name changes (ondtlsstatechange -> onstatechange, onicestatechange -> onstatechange, getNominatedCandidatePair -> getSelectedCandidatePair, RTCDtmfSender -> RTCDTMFSender)
  - Object model incongruities:
    - RTCIceTransport.getLocalCandidates (shimmed from RTCIceGatherer.getLocalCandidates)
    - RTCIceTransport attributes (visible from adapter.js but not native WebRTC 1.0 API in Edge)
- Question: Should we (separately) track adapter.js results?

For Discussion Today

- WebRTC 1.0 Implementation Status (Dom)
  - Confluence tests
  - Adapter vs. non-adapter results (Bernard)
- **WPT**
  - wpt/webrtc test status
  - WPT Issues and PRs
  - Thoughts on Testing (Fippo)
  - Test principles (Fippo)
  - Example test (Fippo)
- Cross-browser testing (Lennart)
- Update on the KITE framework (Dr. Alex)

Web-platform-tests dashboard

Spec	chrome 66.0.3359.117 linux 4.4 @f6d0423be2 Apr 23 2018	edge 15 windows 10 @4a0df34066 Apr 19 2018	firefox 59.0.2 linux 4.4 @f6d0423be2 Apr 23 2018	safari 11.0 macos 10.12 @19aab25e57 Apr 21 2018
RTCCertificate.html	5 / 6	1 / 6	1 / 6	1 / 6
RTCConfiguration-bundlePolicy.html	11 / 16	1 / 16	8 / 16	14 / 16
RTCConfiguration-iceCandidatePoolSize.html	5 / 10	0 / 1	1 / 10	10 / 10
RTCConfiguration-iceServers.html	25 / 78	0 / 1	25 / 78	18 / 78
RTCConfiguration-iceTransportPolicy.html	7 / 17	4 / 17	11 / 17	17 / 17
RTCConfiguration-rtcpMuxPolicy.html	8 / 12	3 / 12	1 / 12	1 / 12
RTCDTMFSender-insertDTMF.https.html	4 / 8	1 / 8	0 / 1	1 / 8
RTCDTMFSender-ontonechange-long.https.html	2 / 2	1 / 2	0 / 1	1 / 2
RTCDTMFSender-ontonechange.https.html	4 / 14	3 / 14	0 / 1	1 / 14
RTCDataChannel-bufferedAmount.html	1 / 5	1 / 5	1 / 5	0 / 5
RTCDataChannel-id.html	3 / 3	1 / 3	1 / 3	1 / 3
RTCDataChannel-send.html	7 / 11	1 / 11	11 / 11	1 / 11
RTCDataChannelEvent-constructor.html	5 / 5	1 / 5	2 / 5	5 / 5
RTCDtlsTransport-getRemoteCertificates.html	1 / 2	1 / 2	1 / 2	1 / 2
RTCIceCandidate-constructor.html	2 / 18	1 / 18	4 / 18	8 / 18
RTCIceTransport.html	1 / 3	1 / 3	1 / 3	0 / 3
RTCPeerConnection-addIceCandidate.html	14 / 24	1 / 24	14 / 24	1 / 24
RTCPeerConnection-addTrack.https.html	4 / 9	1 / 9	0 / 1	4 / 9
RTCPeerConnection-addTransceiver.html	1 / 16	1 / 16	11 / 16	8 / 16
RTCPeerConnection-canTrickleIceCandidates.html	1 / 4	1 / 4	4 / 4	1 / 4
RTCPeerConnection-connectionState.html	1 / 3	1 / 3	1 / 3	1 / 3
RTCPeerConnection-constructor.html	15 / 24	3 / 24	22 / 24	19 / 24
RTCPeerConnection-createAnswer.html	2 / 4	1 / 4	4 / 4	4 / 4
RTCPeerConnection-createDataChannel.html	19 / 31	1 / 31	16 / 31	21 / 31
RTCPeerConnection-createOffer-offerToReceive.html	1 / 16	1 / 16	0 / 1	3 / 16
RTCPeerConnection-createOffer.html	3 / 9	1 / 8	8 / 9	5 / 9
RTCPeerConnection-generateCertificate.html	7 / 9	1 / 9	7 / 9	1 / 9
RTCPeerConnection-getDefaultIceServers.html	1 / 2	1 / 2	1 / 2	1 / 2
RTCPeerConnection-getIdentityAssertion.html	1 / 13	1 / 13	1 / 13	1 / 13
RTCPeerConnection-getStats.https.html	3 / 14	1 / 14	0 / 1	3 / 14
RTCPeerConnection-getTransceivers.html	1 / 2	1 / 2	2 / 2	2 / 2
RTCPeerConnection-iceConnectionState.html	2 / 3	1 / 3	2 / 3	2 / 3

Source: https://wpt.fyi/webrtc

Note: some red caused by permission timeouts. See: web-platform-tests/results-collection#125

Web-platform-tests dashboard (cont'd)

RTCPeerConnection-iceGatheringState.html	3 / 4	1 / 4	3 / 4	1 / 4
RTCPeerConnection-ondatachannel.html	3 / 4	1 / 4	2 / 4	1 / 4
RTCPeerConnection-onnegotiationneeded.html	4 / 8	1 / 8	8 / 8	5 / 8
RTCPeerConnection-ontrack.https.html	2 / 6	1 / 6	0 / 1	2 / 6
RTCPeerConnection-peerIdentity.html	1 / 7	1 / 7	1 / 7	1 / 7
RTCPeerConnection-removeTrack.https.html	5 / 13	1 / 13	0 / 1	3 / 13
RTCPeerConnection-setDescription-transceiver.html	1 / 6	1 / 6	5 / 6	1 / 6
RTCPeerConnection-setLocalDescription-answer.html	2 / 7	1 / 6	3 / 7	5 / 7
RTCPeerConnection-setLocalDescription-offer.html	3 / 8	1 / 6	4 / 8	4 / 8
RTCPeerConnection-setLocalDescription-pranswer.html	5 / 8	1 / 5	0 / 1	7 / 8
RTCPeerConnection-setLocalDescription-rollback.html	0 / 6	1 / 5	0 / 1	2 / 6
RTCPeerConnection-setLocalDescription.html	3 / 5	1 / 3	4 / 5	5 / 5
RTCPeerConnection-setRemoteDescription-answer.html	2 / 5	1 / 4	3 / 5	5 / 5
RTCPeerConnection-setRemoteDescription-offer.html	4 / 9	1 / 6	0 / 1	5 / 9
RTCPeerConnection-setRemoteDescription-pranswer.html	5 / 8	1 / 5	0 / 1	8 / 8
RTCPeerConnection-setRemoteDescription-replaceTrack.https.html	6 / 7	1 / 7	0 / 1	1 / 7
RTCPeerConnection-setRemoteDescription-rollback.html	0 / 5	1 / 4	0 / 1	1 / 5
RTCPeerConnection-setRemoteDescription-tracks.https.html	11 / 15	1 / 15	0 / 1	1 / 15
RTCPeerConnection-setRemoteDescription.html	1 / 7	1 / 5	0 / 1	6 / 7
RTCPeerConnection-track-stats.https.html	12 / 19	1 / 19	0 / 1	1 / 19
RTCPeerConnectionIceEvent-constructor.html	6 / 9	6 / 9	7 / 9	1 / 9
RTCRtpParameters-codecs.html	1 / 9	1 / 9	1 / 9	1 / 9
RTCRtpParameters-degradationPreference.html	1 / 3	1 / 3	1 / 3	1 / 3
RTCRtpParameters-encodings.html	1 / 19	1 / 19	1 / 19	1 / 19
RTCRtpParameters-headerExtensions.html	1 / 2	1 / 2	1 / 2	1 / 2
RTCRtpParameters-rtcp.html	1 / 3	1 / 3	1 / 3	1 / 3
RTCRtpParameters-transactionId.html	1 / 6	1 / 6	1 / 6	1 / 6
RTCRtpReceiver-getCapabilities.html	1 / 3	1 / 3	1 / 3	1 / 3
RTCRtpReceiver-getContributingSources.https.html	1 / 2	1 / 2	0 / 1	1 / 2
RTCRtpReceiver-getParameters.html	1 / 2	1 / 2	1 / 2	1 / 2
RTCRtpReceiver-getStats.https.html	1 / 3	1 / 3	0 / 1	1 / 3

Web-platform-tests dashboard (cont'd)

RTCRtpReceiver-getSynchronizationSources.https.html	1 / 2	1 / 2	0 / 1	1 / 2
RTCRtpSender-getCapabilities.html	1 / 3	1 / 3	1 / 3	1 / 3
RTCRtpSender-getStats.https.html	1 / 3	1 / 3	0 / 1	1 / 3
RTCRtpSender-replaceTrack.html	1 / 10	1 / 10	10 / 10	6 / 10
RTCRtpSender-setParameters.html	1 / 2	1 / 2	2 / 2	1 / 2
RTCRtpTransceiver-setCodecPreferences.html	1 / 10	1 / 10	1 / 10	1 / 10
RTCRtpTransceiver-setDirection.html	1 / 4	1 / 4	1 / 4	2 / 4
RTCSctpTransport-constructor.html	1 / 3	1 / 3	1 / 3	1 / 3
RTCSctpTransport-maxMessageSize.html	1 / 6	1 / 6	1 / 6	1 / 6
RTCTrackEvent-constructor.html	1 / 8	1 / 8	8 / 8	7 / 8
datachannel-emptystring.html	1 / 2	1 / 2	1 / 2	0 / 2
getstats.html	2 / 2	1 / 2	1 / 2	0 / 2
historical.html	7 / 16	7 / 16	6 / 16	10 / 16
interfaces.https.html	17 / 21	0 / 1	0 / 1	16 / 21
no-media-call.html	2 / 2	0 / 1	2 / 2	0 / 2
promises-call.html	2 / 2	1 / 2	2 / 2	0 / 2
simplecall.https.html	2 / 2	1 / 2	1 / 2	1 / 2

web-platform-tests/webrtc Status

https://github.com/w3c/web-platform-tests/pulls?q=is%3Aopen+is%3Apr+label%3Awebrtc

- “Test before commit” policy
  - In effect since TPAC (November 2017)
  - Only a few tests have been submitted, but..
  - No webrtc-pc PRs currently marked “Needs Test”
- Issue Status
  - 14 open issues, 9 open > 30 days
  - Several issues with major effect on “red” status (more later)
- PR Status
  - 40 PRs merged since 02 November 2017.
  - 11 open PRs, 6 open >30 days
- Question: Do we need process changes?
  - To improve frequency of PR submissions?
  - To improve PR review velocity?

WPT Ownership (Soares)

- Current owners of webrtc in WPT are volunteers
- Time to manage tests are limited
  - Keep track of spec changes
  - Update tests
  - Review PRs
  - Discussions on what should be the correct behavior
- Lack of time -> unmerged PRs
  - PRs submitted by non-owners are not reviewed by owners
  - PRs submitted by owners are rarely reviewed + no other owners to approve
- Need more owners for WPT
  - People who can commit time to manage tests in the long run

Test Helpers (Soares)

- Original plan - minimal helpers
- More and more test helpers added over time
- Defined as global variables and included via script tag
- Difficult to keep track of which tests use which helper
- Move test helpers to dedicated helper directory?
- ES Modules to the rescue?
  - Ok to use modern ES2015+ features in WPT?
  - (bonus) Can we use async/await in WPT?

web-platform-tests/webrtc Issues/PRs

https://github.com/w3c/web-platform-tests/pulls?q=is%3Aopen+is%3Apr+label%3Awebrtc

- Issue 7424: Need mock MediaStream data for some WebRTC tests
- Issue 9213: Parts of WebRTC require generating RTP to test
- Issue 836871: WebRTC Tests are leaking heavy resources
- Dependency Issues (more later)
  - Issue 9111/PR 9424: RTCIceTransport.html : dependency on SctpTransport
  - Issue 9110/PR 9424: RTCDtlsTransport-getRemoteCertificates.html : dependency on SctpTransport
  - PR 10566: addTrack: split up tests and reduce dependencies

Issue 7424: Need mock MediaStream data for some WebRTC tests

- Automated tests using `getUserMedia()` time out on a browser without support for command-line flags.
- To run these tests, a MediaStream is needed, obtained via:
  a. WebDriver APIs that bypass permissions and provide mock data from `getUserMedia()`
  b. Fake audio tracks from WebAudio.
  c. `canvas.captureStream()`
  d. `video.captureStream()`
  e. `new RTCPeerConnection().addTransceiver().receiver.track`
- Currently choice e is not widely supported and choices c, d are supported by browsers that support choice a.

Issue 9213: Parts of WebRTC require generating RTP to test

- Tests requiring RTP generation include:
  - Contributing sources: https://w3c.github.io/webrtc-pc/#dom-rtcrtpcontributingsource-audiolevel (depends on the mixer-to-client header extension defined in RFC 6465)
  - Simulcast tests (only in KITE)
- To test this would require a server (mixer or SFU)
  - Similar in concept to wptserve (HTTP server) or pywebsocket (WebSockets server)
  - Server controls what gets sent to the browser on the network.
  - Prerequisites: STUN/TURN, DTLS, etc.
- What (open source) mixers or SFUs can be used for these tests?

Issue 836871: WebRTC Tests Are Leaking Resources

- PeerConnections need to be closed with pc.close().
- Tracks need to be stopped with track.stop().
- Resources may not be garbage collected if we navigate to another test page.
  - Leakage causes issues on Travis as well as in browsers.
- What is the best approach to clean up?
  - More on this later.

Thoughts on Testing (Fippo)

- Writing good tests is hard.
- Writing good tests that pass in all browsers is even harder
  - Not to mention time-consuming
- Reviewing is hard too!
  - Currently not happening :-(
  - Chrome export adding quite a few new tests, but no review required (more later…)

Thoughts on Testing (cont’d)

- Not all tests will pass in all browsers
  - Suggestion: Explain as part of code review why something fails
- Needs attention from vendor and an upstream bug submission
  - Requires time commitment, nobody likes being blocked
- Can highlight areas for improvement!
  - Most browsers pass, please fix yours and we are done here
  - Requires a reduction in “false negatives” so as to demonstrate credibility.

WPT Test Principles (Fippo)

- Clean up after yourself
- Be thoughtful about dependencies
  - Where to “draw the line”
- Beware: Automatic upstreaming without review

Clean up after yourself

- t.add_cleanup(() => { … })
- Travis-CI has trouble with ~20 open peerconnections, tests become flaky
- Not stopping getUserMedia may lead to issues with resolution or aspect ratio being locked
- Has to be done in each test
  - Code review should look for that

Be Thoughtful About Dependencies

- Dependencies on addTransceiver to get a MediaStream(Track)
  - Working around lack of fake devices on travis
    - Fixed for Chrome in mid-march, Firefox still TBD (soon)
    - https://wpt.fyi/mediacapture-streams looks pretty good!
- Dependencies in helpers are hard to spot
- If there are dependencies, check their existence as a sync test at the beginning of the file?
  - ‘addTransceiver’ in RTCPeerConnection.prototype
- Chrome does not support addTransceiver
  - Should tests not relating to addTransceiver depend on it?
- No browser implements pc.sctp
  - Should tests not relating to data channel or SctpTransport depend on sctp?

Where to “Draw the Line”

- RTCPeerConnection() does not work in Edge. RTCPeerConnection(null) does. RTCPeerConnection({}) crashes even?

Beware: Automatic Upstreaming Without Review

- Two examples:
  - replaceTrack
    - Test did not work in Firefox even though Firefox supports replaceTrack.
    - Used addTrack(track) without any streams. Trivial fix.
    - spec issue found and fixed. Review is good...
  - dtmf helper function did not work in Firefox
    - Relied on canInsertDtmf, not implemented in Firefox
    - Trying to workaround Chrome issue
    - Helpers should not be tests

WPT Example Test (Fippo)

- PR: https://github.com/w3c/web-platform-tests/pull/10566/
- function addTrackFromGetUserMedia()
  - Test setup, like beforeEach() from karma/jasmine
    - Jasmine: “To help a test suite DRY up any duplicated setup and teardown code, Jasmine provides the global beforeEach and afterEach functions. As the name implies, the beforeEach function is called once before each spec in the describe in which it is called, and the afterEach function is called once after each spec.”
  - Separate setup and assertions

WPT Example Test (Fippo)

- Cleanup, release resources
  - closes RTCPeerConnection (important on travis!)
  - stops all MediaStreamTracks (important with real HW)
    - Cleanup should not go into setup?
    - Alternative: creating wrapper for getUserMedia and RTCPeerConnection which are responsible for cleanup. See getUserMedia beforeEach/afterEach in adapter tests

WPT Example Test (Fippo)

- Before:
  - addTrack with a single track argument and no mediaStream should succeed
- After: addTrack… (with a single MediaStreamTrack)
  - returns an RTCRtpSender
  - … whose track is set to the MediaStreamTrack
  - creates RTCRtpSender that is in getSenders()
  - creates RTCRtpReceiver
  - creates RTCRtpTransceiver

WPT Example Test (Fippo)

- Before: Chrome failed test
  - Does not implement transceiver model
  - Or getTransceivers
- After: Chrome passes 3/5 assertions
  - Test assertions show up in HTML result
- What assertions need to be repeated for no/single/multiple MediaStreams?
  - Three very similar tests
  - Coverage vs maintenance

WPT Example Test (Fippo)

- make getUserMedia release tracks and stop RTCPeerConnection magically?
  - see crbug; wraps getUserMedia + RTCPeerConnection,
  - ```
    webrtc_test(async t => {
        const pc = new RTCPeerConnection();
        const stream = await
    navigator.mediaDevices.getUserMedia({audio: true});
        const sender = pc.addTrack(stream.getTracks()[0],
    stream);
        assert_true(sender instanceof RTCRtpSender,
          'Expect sender to be instance of RTCRtpSender');
    });
    ```
  - Less boilerplate.
  - No explicit cleanup is bad?
    ```

WPT Example Alternative 1 (Jan-Ivar)

- Avoid wrappers that alter semantics, e.g. creating an abstraction that cleanup isn't needed. Instead:
- wrap getUserMedia and RTCPeerConnection to assert:

```
webrtc_test(async t => {
  const pc = new RTCPeerConnection();
  const stream = await navigator.mediaDevices.getUserMedia({audio: true});
  const sender = pc.addTrack(stream.getTracks()[0], stream);
  assert_true(sender instanceof RTCRtpSender,
              'Expect sender to be instance of RTCRtpSender');
  pc.close();
  stream.getTracks().forEach(track => track.stop());
});
```

- Forgetting to clean up will be asserted by wrapper (& cleaned?)
- Pro: Promotes writing “correct” API code. No surprises.
- Con: noise (or magic?) if assert hits, so not really “correct”.

WPT Example Alternative 2 (Jan-Ivar)

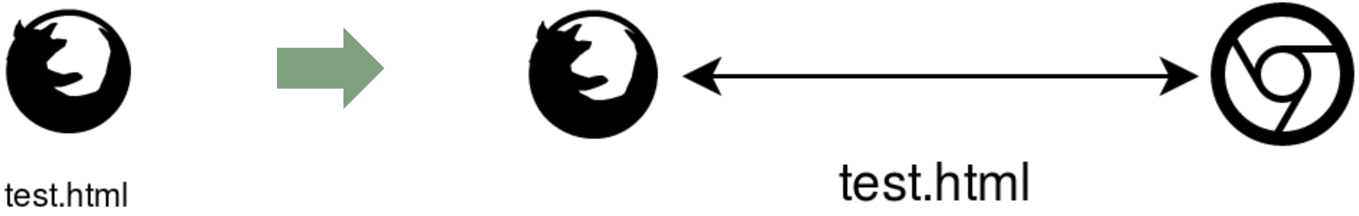
- No wrappers. Rely on well-named helpers:

```
webrtc_test_with_stream_and_pcs(async (t, {stream, pc1, pc2}, {audio:
true}) => {
  const sender = pc1.addTrack(stream.getTracks()[0], stream);
  assert_true(sender instanceof RTCRtpSender,
              'Expect sender to be instance of RTCRtpSender');
});
```

- No magic. Helpers clean up (using `try{}finally{}`)
- Could add optional flag to run tests with wrappers that assert tests don't leak.

Cross-Browser Testing (Lennart)

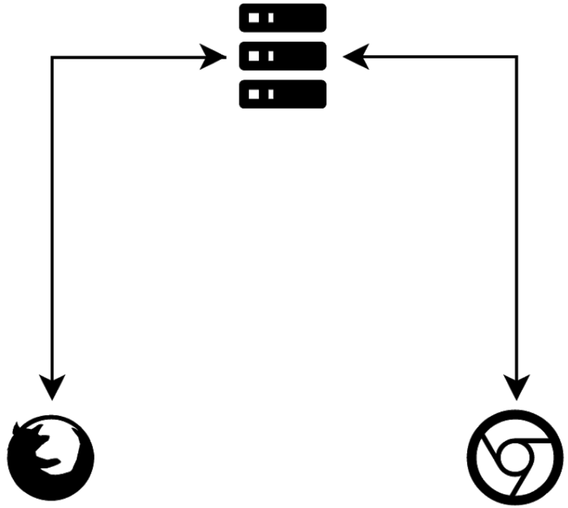
Goal: Reuse WPT tests for cross-browser (conformance) testing.



Cross-Browser Testing (Lennart)

Architecture

ws-url: ws://signaling/<uuid4>/<test-index>/<role>



test.html
?crossBrowser
&url=ws://signaling/<uuid4>/<test-index>
&role=1

test.html
?crossBrowser
&url=ws://signaling/<uuid4>/<test-index>
&role=0

Cross-Browser Testing (Lennart)

Example

```
async_test(t => {
  const pc1 = new RTCPeerConnection();
  const pc2 = new RTCPeerConnection();
  t.add_cleanup(() => pc1.close());
  t.add_cleanup(() => pc2.close());

  exchangeIceCandidates(pc1, pc2);

  // Create a channel on pc1
  const dc = pc1.createDataChannel('onopen');
  dc.onopen = t.step_func(() => {
    assert_equals(dc.readyState, 'open', '[...]');
    t.done();
  });

  doSignalingHandshake(pc1, pc2)
  .catch(t.step_func(err =>
    assert_unreached(`Unexpected promise rejection:
${err}`)));
}, 'In-band channel: Open event should be fired [...]');
```

```
cross_browser_test(async (t, signaling, offering) => {
  const pc = new RTCPeerConnection();
  t.add_cleanup(() => pc.close());

  // Create an in-band negotiated channel
  if (offering) {
    const dc = pc.createDataChannel('onopen');

    // Wait for the channel to open
    dc.addEventListener('open', t.step_func(() => {
      assert_equals(dc.readyState, 'open', '[...]');
      t.done();
    }), { once: true });
  } else {
    t.done();
  }

  // Bind candidate events and do offer/answer exchange
  signaling.exchangeCandidates(t, pc);
  await signaling.exchangeDescriptions(pc, offering);
  await t.done_promise;
}, 'In-band channel: Open event should be fired [...]');
```

Cross-Browser Testing (Lennart)

Next Possible Steps

- Modify tests to be compatible
- Update wpt.py to run these automatically
- Push results to wpt.fyi

Questions?

Code: lgrahl/web-platform-tests/webrtc-cross-browser

KITE Update (Dr. Alex)

- Slides

For extra credit

Name those birds!

Thank you

Special thanks to:

W3C/MIT for WebEx

WG Participants, Editors & Chairs

The bird

Overflow slides

WPT WebRTC Issues

14 Open ✓ 6 Closed — Author ▾ Labels ▾ Projects ▾ Milestones ▾ Assignee ▾ Sort ▾

- RTCRtpHeaderExtensionParameters missing members are not tested `type:missing-coverage` `webrtc` `wg-webrtc`
  #10253 opened 25 days ago by foolip
- RTCDataChannel-bufferedAmount.html are flawed `webrtc` `wg-webrtc` — 6
  #10126 opened on Mar 21 by nils-ohlmeier
- Need to modify test to deal with setDirection being removed `webrtc` — 1
  #9438 opened on Feb 8 by alvestrand
- RTCRtpSender.get/setParameters() test coverage `webrtc` — 3
  #9395 opened on Feb 5 by Orphis
- Parts of WebRTC require generating RTP to test `infra` `priority:backlog` `type:untestable` `webrtc` `wg-webrtc` — 3
  #9213 opened on Jan 27 by foolip
- [WebRTC] missing EventTarget from webrtc.idl on idl interface test `webrtc`
  #9121 opened on Jan 22 by murillo128
- [WebRTC] RTCIceTransport.html : dependency on SctpTransport `webrtc`
  #9111 opened on Jan 19 by aboba
- [WebRTC] RTCDtlsTransport-getRemoteCertificates.html : dependency on SctpTransport `webrtc`
  #9110 opened on Jan 19 by aboba
- RTCRtpContributingSource's audioLevel member should be double (asserted to be byte) `type:missing-coverage` `webrtc` `wg-webrtc`
  #9108 opened on Jan 19 by foolip

WPT WebRTC Issues (cont'd)

- [WebRTC] Update IDL for RTCIceCandidate and RTCIceCandidateInit : ufrag->usernameFragment `webrtc`
  #9048 opened on Jan 16 by murillo128
- RTCStats-helper.js's validateStatsReport assumes video track `webrtc`
  #9010 opened on Jan 12 by henbos
- Get rid of web-platform.test references `clear-site-data` `content-security-policy` `fetch` `html` `longtask-timing` `navigation-timing` `service-workers` `type:cleanup` `webdriver` `webrtc`
  #8420 opened on Nov 24, 2017 by gsnedders 0 of 19
- Need mock MediaStream data for some WebRTC tests `infra` `mediacapture-streams` `priority:roadmap` `testdriver.js` `type:untestable` `webrtc`
  #7424 opened on Sep 20, 2017 by foolip 8
- webrtc/RTCRtpTransceiver-setDirection.html expects currentDirection to be 'recvonly' when it should be 'inactive' `webrtc`
  #7055 opened on Aug 30, 2017 by docfaraday 2

WPT WebRTC PRs

11 Open ✓ 172 Closed | Author | Labels | Projects | Milestones | Reviews | Assignee | Sort

- addTrack: split up tests and reduce dependencies ✓ webrtc wg-webrtc — 2
  #10566 opened 2 days ago by fippo • Review required
- More Tests for WebRTC Data Channels ✗ webrtc wg-webrtc
  #10468 opened 11 days ago by lgrahl • Review required
- adds tests for the missing members of RTCRtpHeaderExtensionParameters ✓ webrtc wg-webrtc — 2
  #10282 opened 21 days ago by kritisingh1 • Review required
- Bring RTCCertificate interface up to date with Candidate Recommendation ✓ webrtc wg-webrtc — 1
  #10271 opened 22 days ago by aboba • Review required
- Updating tests to test no stream ids and multiple stream ids. ✓ chromium-export do not merge yet webrtc wg-webrtc — 2
  #10225 opened 27 days ago by chromium-wpt-export-bot • Approved
- WebRTC: Add test for null ICE candidate ✗ webrtc wg-webrtc — 5
  #9517 opened on Feb 14 by alvestrand • Review required
- Rename RTCIceCandidate ufrag field to usernameFragment ✓ webrtc wg-webrtc — 1
  #9434 opened on Feb 7 by soareschen • Review required
- Rewrite RTC ICE and DTLS transport tests with alternative dependencies ✓ webrtc wg-webrtc — 17
  #9424 opened on Feb 7 by soareschen • Review required
- Close peer connection and tracks after end of each test ✓ webrtc wg-webrtc — 14
  #8542 opened on Dec 1, 2017 by soareschen • Changes requested
- Add coverage report and tools for WebRTC tests ✓ infra webrtc wg-webrtc — 2
  #8051 opened on Nov 3, 2017 by soareschen • Review required
- Test requirements from JSEP on initial offer for 1 data channel ✗ stale:awaiting-review webrtc wg-webrtc — 11
  #2301 opened on Nov 3, 2015 by dontcallmedom • Review required

- 40 PRs merged since 02 November 2017.
- 11 open PRs
  - 6 open >30 days