GISC2020
Global ICT Standards Conference

뉴노멀시대 선도를 위한
ICT 표준의 역할

(W3C) WebRTC 표준화 현황 : 웹표준기술융합포럼

손성영 유프리즘

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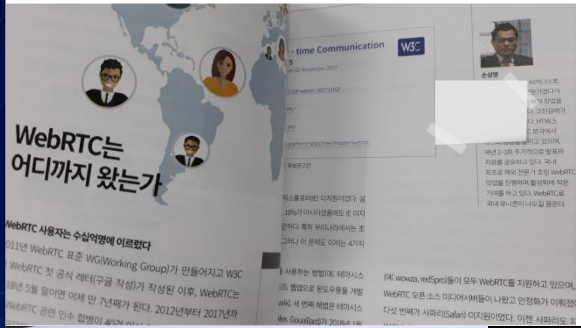
소개



WebRTC Visionist
(2014 ~)

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WebRTC 리뷰

October 6, 2010 - RTC Web Workshop

(Google Mountain View, CA Headquarters, building 40, Kiev room)

WebRTC 리뷰

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October 6, 2010 - RTC Web Workshop

회사명	인원	비고	회사명	인원	비고
Microsoft	1		**Logitech**	2	
Google	10		**Cisco**	2	
Skype	6	※ MS인수 - 2011.05.	**Nokia**	2	
BBN	1		**Opera**	1	
Mozilla	3		**W3C**	1	
Intel	3		**Yahoo**	1	
CDT	1		**Neustar**	1	
Linden Lab	1		**Octasic**	1	
GIPS	4		**Apple**	1	
IBM	1		**무소속**	1	
Ericsson	3		**합계**	47	

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WebRTC 리뷰

October 6, 2010 - RTC Web Workshop

Agenda

0830: Intro & Goals - Chris Blizzard

0900: Use Cases, Requirements - Joe Hildebrand

1000: Nat Traversal - Justin Uberti

1030: Media Encryption (What's the key exchange) - Eric Rescorla

1100: Same Orgin Security Issues for P2P Media - Adam Barth (in docs)

1130: Security & Privacy - Alissa Cooper

1300: Levels of APIs - Stefan Håkansson

1400: Notifications, System Tray, Persistence - 20 min (Joe H)

1420: Codecs - Jan Linden

15:00 Interoperability - Jonathan Rosenberg

16:00 Next steps and wrap Up - Philippe Le Hegaret, Jon Peterson

* Chairman

Harald Tveit Alvestrand

Born	29 June 1959 (age 61) Namsos, Norway
Occupation	computer scientist
Known for	i18n, IETF, and Linux work
Notable work	RFC 2277 (BCP 18)
Children	3
Website	www.alvestrand.no

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WebRTC 리뷰

"WebRTC fills a critical gap in the web platform
as you can communicate in real-time just by loading a web page"

2013, Google I/O
Justin Uberti
(Tech Lead on WebRTC, Google)

https://youtu.be/p2HzZkd2A40

WebRTC 리뷰

With WebRTC, for example, Web pages can talk to each other.
The Web as a programmable platform is going to allow real-time
collaboration, videoconferencing based on any Website,
not just your video conference Website.
It'll allow data conferencing as well -- people sharing artwork, sharing ideas, sharing ideas.”

2014 / 25th Anniversary, the Web still needs Works
Tim Berners-Lee

https://www.cnet.com/news/tim-berners-lee-on-its-25th-anniversary-the-web-still-needs-work-q-a/

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WebRTC 리뷰

2011.05.31 First Google release of WebRTC source code

Google release of WebRTC source code

This message: [Message body] [Respond] [More options]
Related messages: [Previous message]

From: Harald Alvestrand <hta@google.com>
Date: Wed, 1 Jun 2011 01:04:28 +0200
Message-ID: <BANLkTinXEJypAc2tVvAySzUGOtG9kLZNXA@mail.gmail.com>
To: public-webrtc@w3.org

Today, Google made available WebRTC, an open source software package for real-time voice and video on the web that we will be integrating in Chrome. This is our initial contribution to achieve the mission of making audio and video available in all browsers, through a uniform standard set of APIs. This initial release will provide the functionality we envision for WebRTC/RTCWEB, as detailed at https://sites.google.com/site/webrtc/. Working with the browser community and working groups like this, our goal is to expand the available APIs over the next few months for developers to create voice and video applications on the web.

The underlying components we're releasing are stable and the interfaces for this initial release are consistent with the discussions in this working group. We will continue to provide working implementations for consideration and feedback to collectively ensure stable standards are finalized. Google is committed to fully supporting these standards and we look forward to your input in the coming months.

Harald, speaking for Google.

Received on Tuesday, 31 May 2011 23:05:13 UTC

This message: [Message body]
Previous message: Harald Alvestrand: "Next teleconference: June 14, 1600 GMT"

Today, Google made available WebRTC, an open source software package for real-time voice and video on the web that we will be integrating in Chrome. This is our initial contribution to achieve the mission of making audio and video available in all browsers, through a uniform standard set of APIs.

WebRTC 리뷰

2011.10.27 Working Draft

W3C Working Draft

WebRTC 1.0: Real-time Communication Between Browsers

W3C Working Draft 27 October 2011

This version:
  http://www.w3.org/TR/2011/WD-webrtc-20111027/
Latest published version:
  http://www.w3.org/TR/webrtc/
Latest editor's draft:
  http://dev.w3.org/2011/webrtc/editor/webrtc.html
Previous version:
  none
Editors:
  Adam Bergkvist, Ericsson
  Daniel C. Burnett, Voxeo
  Cullen Jennings, Cisco
  Anant Narayanan, Mozilla

WebRTC 리뷰

Draft -> CR (Candidate Recommendation)

WebRTC 1.0: Real-time Communication Between Browsers

W3C Candidate Recommendation 02 November 2017

※ 의미 (Dan Burnnet say)

- 스펙 완성
- 스펙에 대한 테스팅이 시작
- WebRTC에서는 다음 버전에 대한 일을 시작할 수 있게 되었다는 것

WebRTC 리뷰

WebRTC 1.0: Real-time Communication Between Browsers

W3C Candidate Recommendation 24 September 2020

※ 올해안에 PR(Proposed Recommendation)?

WebRTC 리뷰

WebRTC 1.0: Real-time Communication Between Browsers

W3C Candidate Recommendation 24 September 2020

※ 올해안에 PR(Proposed Recommendation)?

* 9월 3일 CR 버전에 있던 문구
"This Candidate Recommendation is expected to advance to Proposed Recommendation no earlier than 24 September 2020."

WebRTC 리뷰

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Candidate Recommendations

Audio Output Devices API
2017-10-03 Media Web API

Identifiers for WebRTC's Statistics API
2020-01-14 Media Web API

Identity for WebRTC 1.0
2018-09-27

Media Capture and Streams
2020-09-29 Media Web API

WebRTC 1.0: Real-time Communication Between Browsers
2020-09-24 Media Web API

WebRTC 리뷰

Working Drafts

"MediaStream Image Capture"
2017-06-21 Media Web API

Media Capture Depth Stream Extensions
2017-04-18 Media Web API

Media Capture from DOM Elements
2017-09-06 Media Web API

MediaStream Capture Scenarios
2012-03-06 Media Web API

MediaStream Recording
2017-06-21 Media Web API

MediaStreamTrack Content Hints
2020-07-28 Media Web API

Scalable Video Coding (SVC) Extension for WebRTC
2020-04-08 Media

Screen Capture
2019-11-19 Media Web API

WebRTC Next Version Use Cases
2018-12-11 Media

WebRTC Priority Control API
2020-09-22 Media Web API

WebRTC DSCP Control API
W3C First Public Working Draft, 3 July 2018

WebRTC 리뷰

RTC Accessibility User Requirements

W3C First Public Working Draft 19 March 2020

This version:
https://www.w3.org/TR/2020/WD-raur-20200319/

Latest published version:
https://www.w3.org/TR/raur/

Latest editor's draft:
https://w3c.github.io/apa/raur/

RTC 접근성이 장애인의 요구를 어떻게 지원할 수 있는지 설명

WebRTC 동향

Insertable Streams

WebRTC Insertable Media using Streams

Editor's Draft, 1 September 2020

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WebRTC 동향

Insertable Streams

WebRTC Insertable Media using Streams
Editor's Draft, 1 September 2020

Abstract

This API defines an API surface for manipulating the bits on MediaStreamTracks being sent via an RTCPeerConnection.

WebRTC 동향

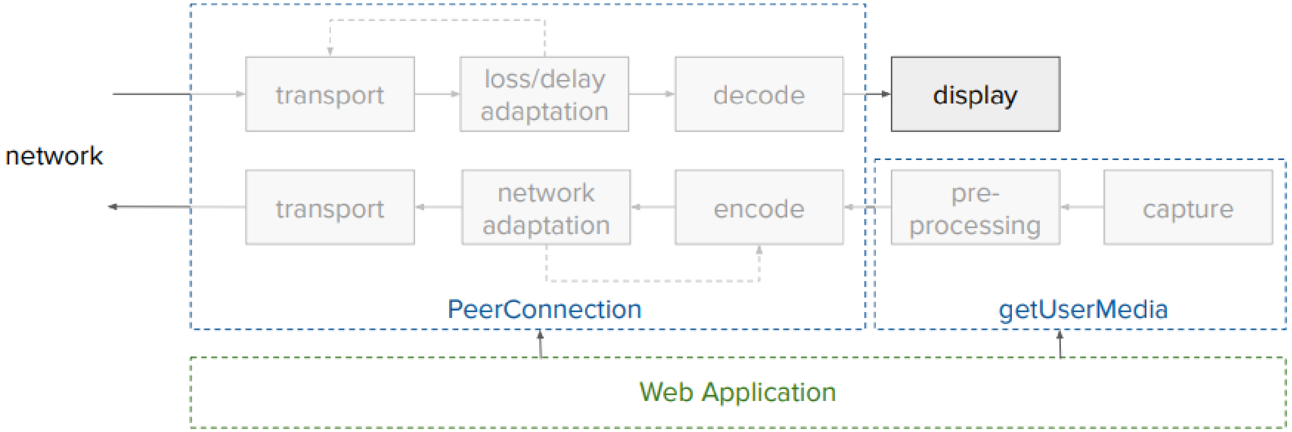
Insertable Streams

API that allows the insertion of user-defined processing steps in the pipeline that handles media in a WebRTC context.

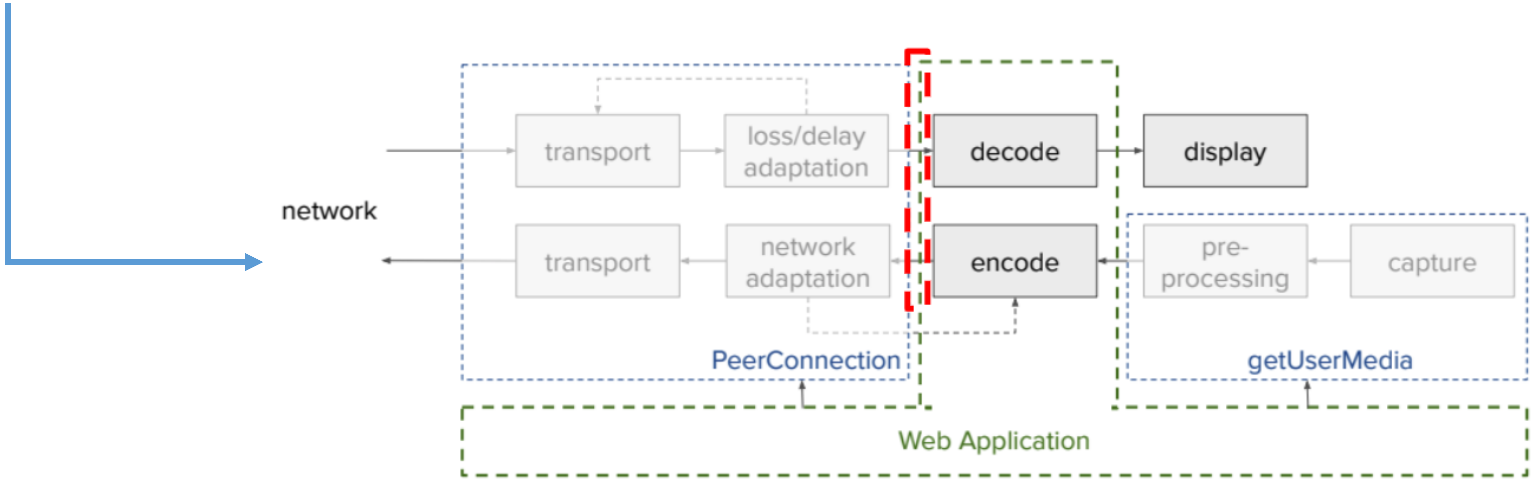
애플리케이션이 사용자 정의 데이터 처리를 삽입하도록 허용합니다.
특정 사용 사례는 중간 서버를 통해 RTCPeerConnections간에 전송되는 인코딩 된 데이터의 종단 간 암호화.

WebRTC 동향

Insertable Streams



TPAC 2019 ~



WebRTC 동향

Insertable Streams

- Implemented in Chrome
- Available in Chrome 83 (turn on --enable-experimental-web-platform-features)
- Available as an “origin trial” to any website that registers
- Used for Duo and Jitsi end-to-end additional frame encryption
- Experimented with for a number of other purposes

※ 출처 WebRTC WG virtual meeting 2020.06

WebRTC 동향

Insertable Streams

https://github.com/mganeko/webrtc_insertable_demo

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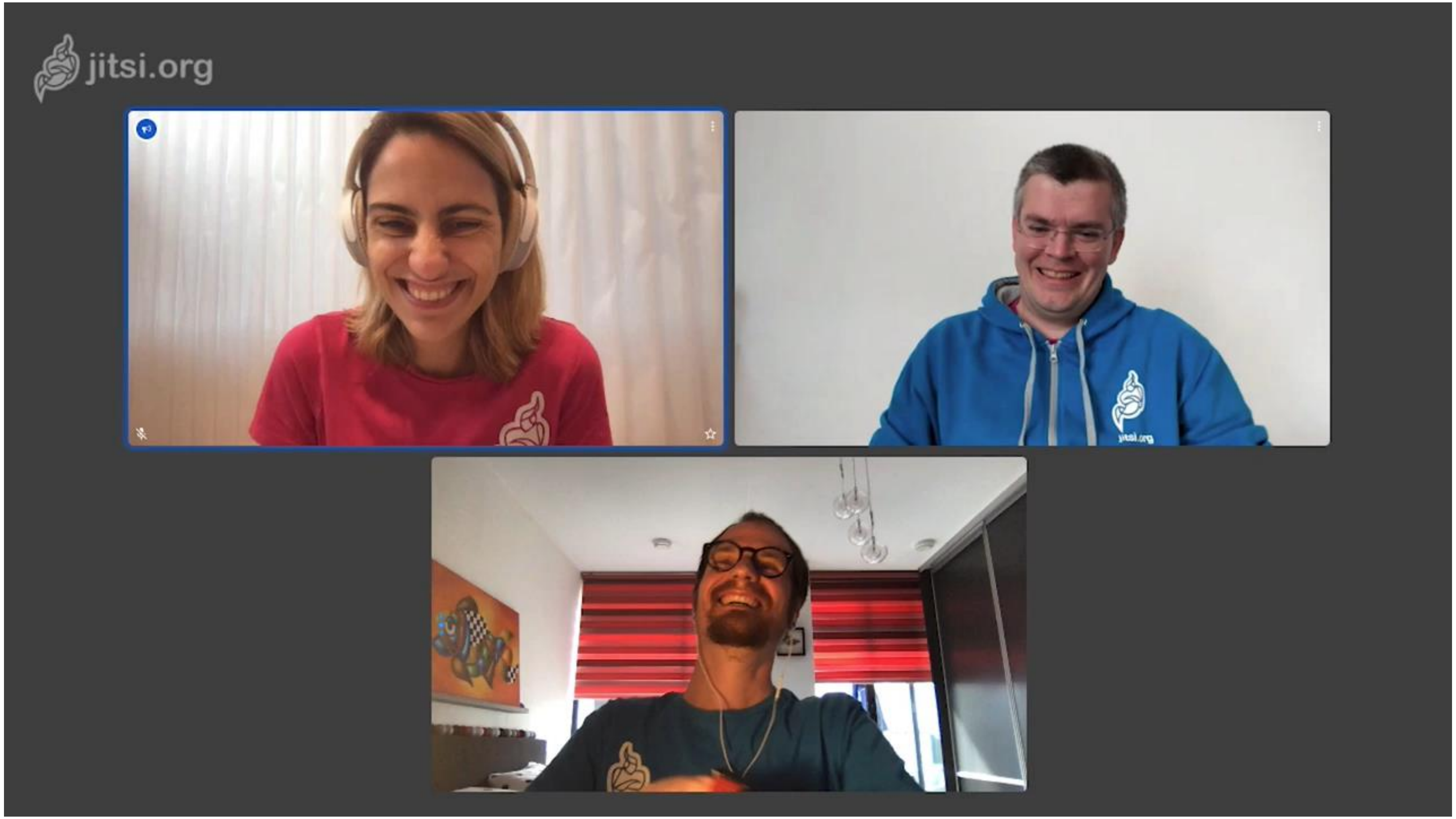
WebRTC 동향

Insertable Streams

PoC: end-to-end encryption in Jitsi Meet



WebRTC 동향

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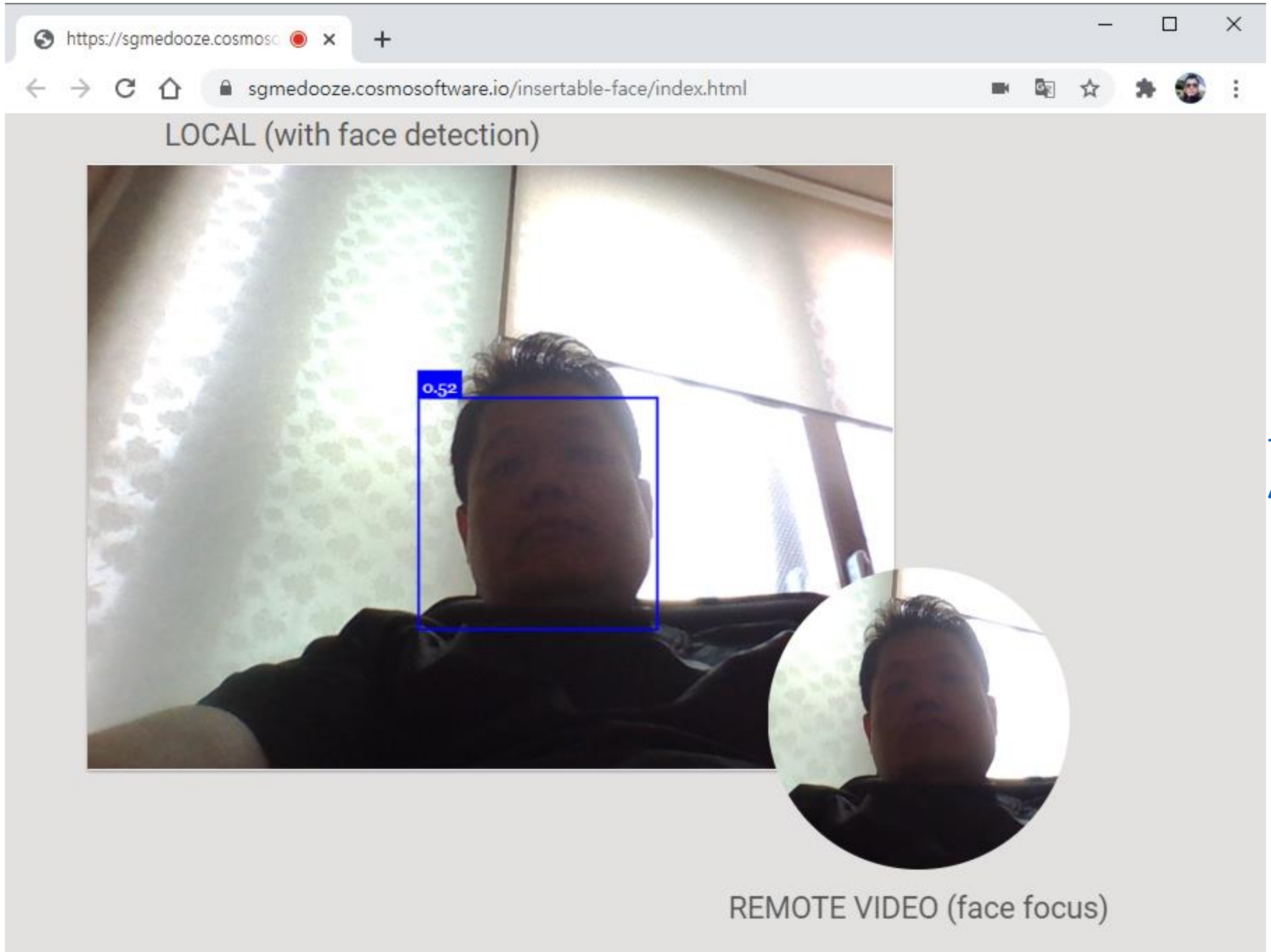
Insertable Streams



[https://sgmedooze.**cosmosoftware.io**/insertable-face/index.html](https://sgmedooze.cosmosoftware.io/insertable-face/index.html)

WebRTC 동향

Insertable Streams

For
WebRTC Next Version Use Cases

Funny Hats

: 인코딩 전과 디코딩 후 캡처 된 미디어를 조작하여
Funny hats
Background removal or blurring
In-browser compositing
Voice effects
Stress detection

WebRTC 동향

Insertable Streams

For
WebRTC Next Version Use Cases

Machine Learning
: 추론 스트림과 교육 스트림
새를 관찰하는 웹 사이트가 있다고 가정할 때, 자신의 장치를 통해 비디오와 오디오에 정보를 넣어 실시간으로 사용자에게 보냄.

Virtual Reality Gaming
: 단말의 미디어(비디오, 오디오)와 데이터를 동기화

WebRTC 동향

WebRTC-SVC

Scalable Video Coding (SVC) Extension for WebRTC
W3C Working Draft 08 April 2020

Before : 2019.10.22 First Public Draft

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WebRTC 동향

WebRTC-SVC

Chrome Platform Status

All features Releases Samples Stats

WebRTC Scalable Video Coding extensions

Web RTC

This extension defines a standard method for picking between possible Scalable Video Coding (SVC) configurations on an outgoing WebRTC video track.

Status in Chromium

Blink components: Blink>WebRTC>Video

In development (tracking bug)

Consensus & Standardization

After a feature ships in Chrome, the values listed here are not guaranteed to be up to date.

Firefox:	No signal
Edge:	No signal
Safari:	No signal
Web Developers:	No signals

WebRTC 동향

Content-Hints

MediaStreamTrack Content Hints

W3C Working Draft 28 July 2020

This version:

https://www.w3.org/TR/2020/WD-mst-content-hint-20200728/

WebRTC 동향

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Content-Hints

§ 4.1 Audio Content Hints

Audio content hints are only applicable when the `MediaStreamTrack` contains an audio track.

Audio content hints	
`""`	No hint has been provided, the implementation should make its best-informed guess on how to handle contained audio data. This may be inferred from how the track was opened or by doing content analysis.
`speech`	The track should be treated as if it contains speech data. Consuming this signal it may be appropriate to apply noise suppression or boost intelligibility of the incoming signal.
`speechRecognition`	The track should be treated as if it contains data for the purpose of speech recognition by a machine. Consuming this signal it may be appropriate to boost intelligibility of the incoming signal for transcription and turn off audio-processing components that are used for human consumption.
`music`	The track should be treated as if it contains music data. Generally this might imply tuning or turning off audio-processing components that are used to process speech data to prevent the audio from being distorted.

WebRTC 동향

Content-Hints

§ 4.2 Video Content Hints

Video content hints are only applicable when the `MediaStreamTrack` contains a video track.

Video content hints	
`""`	No hint has been provided, the implementation should make its best-informed guess on how contained video content should be treated. This can for example be inferred from how the track was opened or by doing content analysis.
`motion`	The track should be treated as if it contains video where motion is important. This is normally webcam video, movies or video games. Quantization artefacts and downscaling are acceptible in order to preserve motion as well as possible while still retaining target bitrates. During low bitrates when compromises have to be made, more effort is spent on preserving frame rate than edge quality and details.
`detail`	The track should be treated as if video details are extra important. This is generally applicable to presentations or web pages with text content, painting or line art. This setting would normally optimize for detail in the resulting individual frames rather than smooth playback. Artefacts from quantization or downscaling that make small text or line art unintelligible should be avoided.
`text`	The track should be treated as if video details are extra important, and that significant sharp edges and areas of consistent color can occur frequently. This is generally applicable to presentations or web pages with text content. This setting would normally optimize for detail in the resulting individual frames rather than smooth playback, and may take advantage of encoder tools that optimize for text rendering. Artefacts from quantization or downscaling that make small text or line art unintelligible should be avoided.

WebRTC 동향

WebCodec + WebTransport

WebCodecs

Draft Community Group Report, 29 September 2020

This version:
https://wicg.github.io/web-codecs/

Issue Tracking:
GitHub

Editors:
Chris Cunningham (Google Inc.)
Paul Adenot (Mozilla)

Participate:
Git Repository.
File an issue.

Version History:
https://github.com/wicg/web-codecs/commits

WebTransport

Draft Community Group Report, 24 September 2020

This version:
https://wicg.github.io/web-transport/

Issue Tracking:
GitHub
Inline In Spec

Editors:
Peter Thatcher (Google)
Bernard Aboba (Microsoft Corporation)
Robin Raymond (Optical Tone Ltd.)

W3C Community Group Draft Report

WebRTC 동향

WebCodec + WebTransport

WebCodecs Proposal

Media and Real Time Communications

pthatcher 1 Jun '19

WebCodecs

API that allows web applications to encode and decode audio and video

Many Web APIs use media codecs internally to support APIs for particular uses:

- HTMLMediaElement and Media Source Extensions
- WebAudio (decodeAudioData)
- MediaRecorder
- WebRTC

But there's no general way to flexibly configure and use these media codecs. Because of this, many web applications have resorted to implementing media codecs in JavaScript or WebAssembly, despite the disadvantages:

- Increased bandwidth to download codecs already in the browser.
- Reduced performance
- Reduced power efficiency

It's great for:

- Live streaming
- Cloud gaming
- Media file editing and transcoding

See the explainer 207 for more info.

Jun 2019
1 / 19
Jun 2019
Sep 2019

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WebCodec + WebTransport

WebCodecs in Chrome M86: Limitations

Supported

- `AudioDecoder` can decode AAC, FLAC, MP3, Opus, and Vorbis.
- `VideoDecoder` can decode VP8, VP9, H.264 and AV1 (AV1 not available on Android).
- `VideoEncoder` can encode VP8 and VP9.
- `VideoTrackReader`.
- `ImageDecoder` can decode BMP, GIF, JPG, PNG, ICO, AVIF, and WebP.

Unsupported

- `AudioEncoder` is not yet implemented.
- `AudioDecoder` can decode xHE-AAC on Android, but it requires ADTS headers. We do not expect to require ADTS headers in the future.
- `VideoEncoder` can encode H.264, but it is limited:
  - H.264 encoding is not available on all platforms.
  - The output format is Annex B. We expect the format to be AVC in the future.
- `new VideoFrame(pixelFormat, planes, frameInit)` is not available.
- `ImageDecoder` can't decode SVG. We don't expect SVG to ever be supported.

WebRTC 동향

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RTCQuicTransport

QUIC

- RTCQuicTransport는 QUIC 프로토콜을 사용하여 원격 피어와 임의 데이터를 교환할 수 있는 새로운 웹 플랫폼 API
- P2P 사용 사례를위한 것이므로 ICE를 통해 P2P 연결을 설정하기 위해 독립형 RTCIceTransport API와 함께 사용
- 데이터는 신뢰할 수 있고 순서대로 전송
- 일반 양방향 데이터 전송이므로 게임, 파일 전송, 미디어 전송, 메시징 등에 사용할 수 있음

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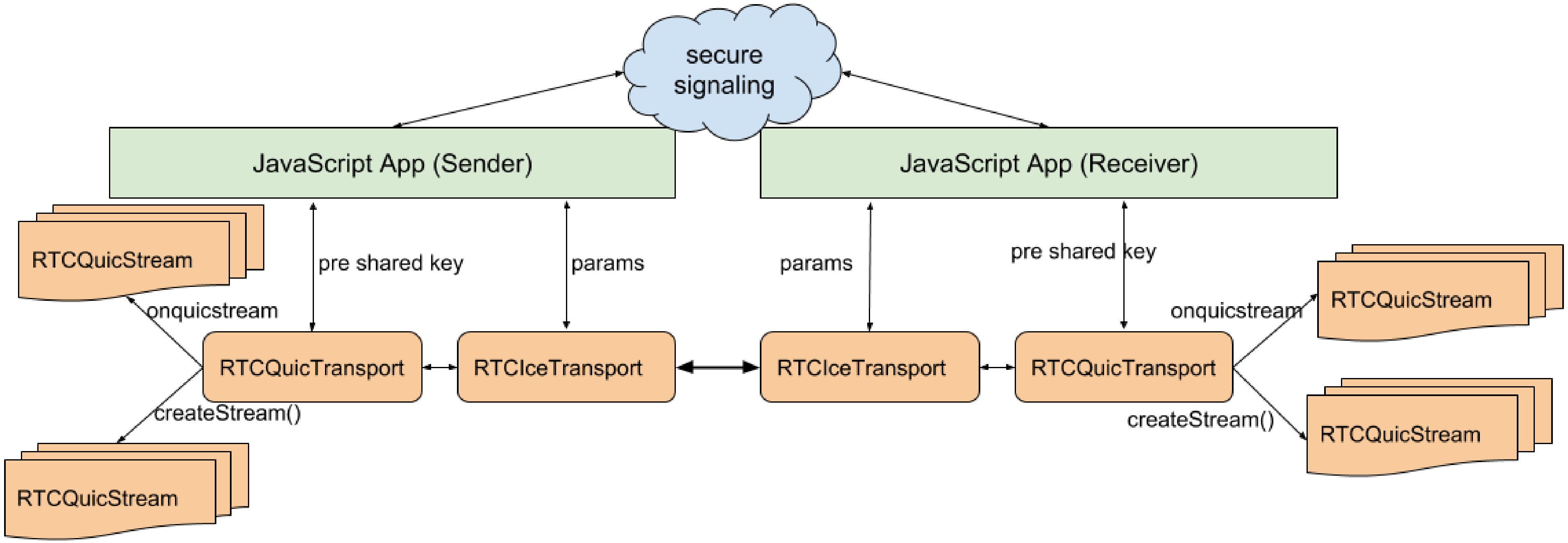
WebRTC 동향

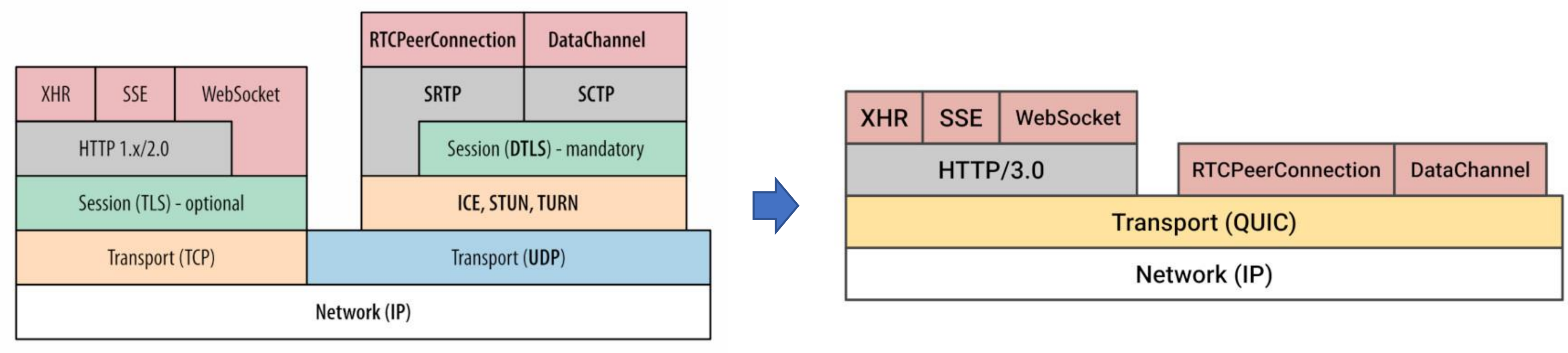
RTCQuicTransport



https://developers.google.com/web/updates/2019/01/rtcquictransport-api#rtcquictransport

WebRTC 동향

RTCQuicTransport



https://bloggeek.me/who-needs-quic-in-webrtc

WebRTC Next

RTCQuicTransport

QUIC API for Peer-to-peer Connections

Draft Community Group Report 21 January 2020

Latest editor's draft:
https://w3c.github.io/webrtc-quic/

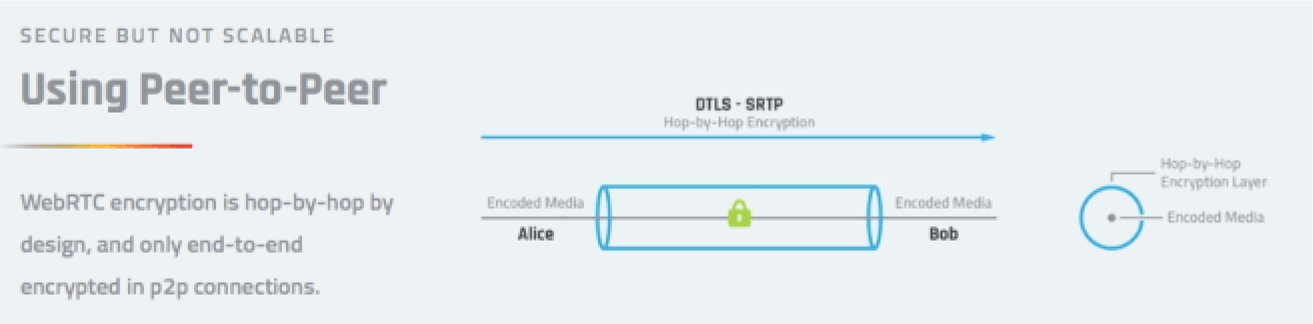
Editors:
Peter Thatcher (Google)
Bernard Aboba (Microsoft Corporation)
Robin Raymond (Optical Tone Ltd.)

W3C Community Group Draft Report

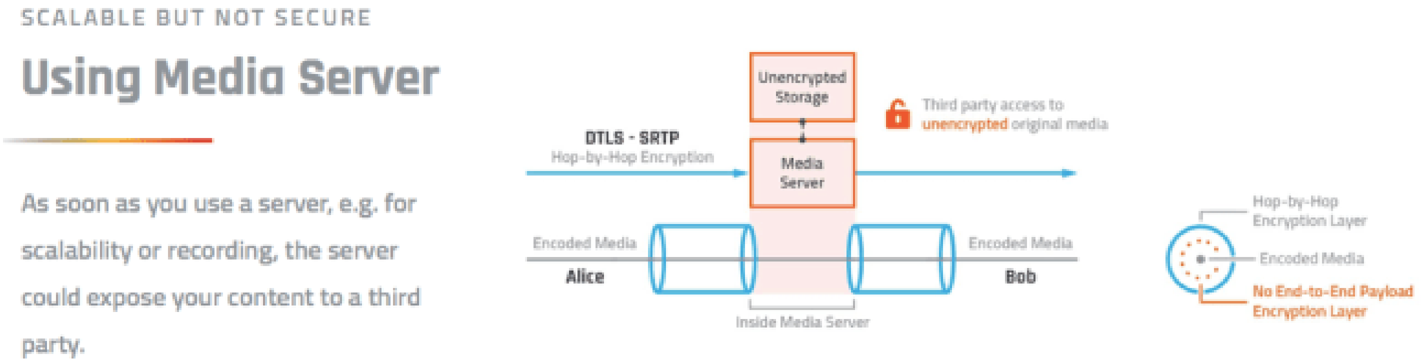
WebRTC 동향

sFrame (Secure Frame)

- E2EE (End to End Encryption)
- Google 과 CosmoSoftware 공동개발 하여 Duo 에서 사용중인 것
- PERC 의 향상 버전



※ source : Dr. Alex 's Blog



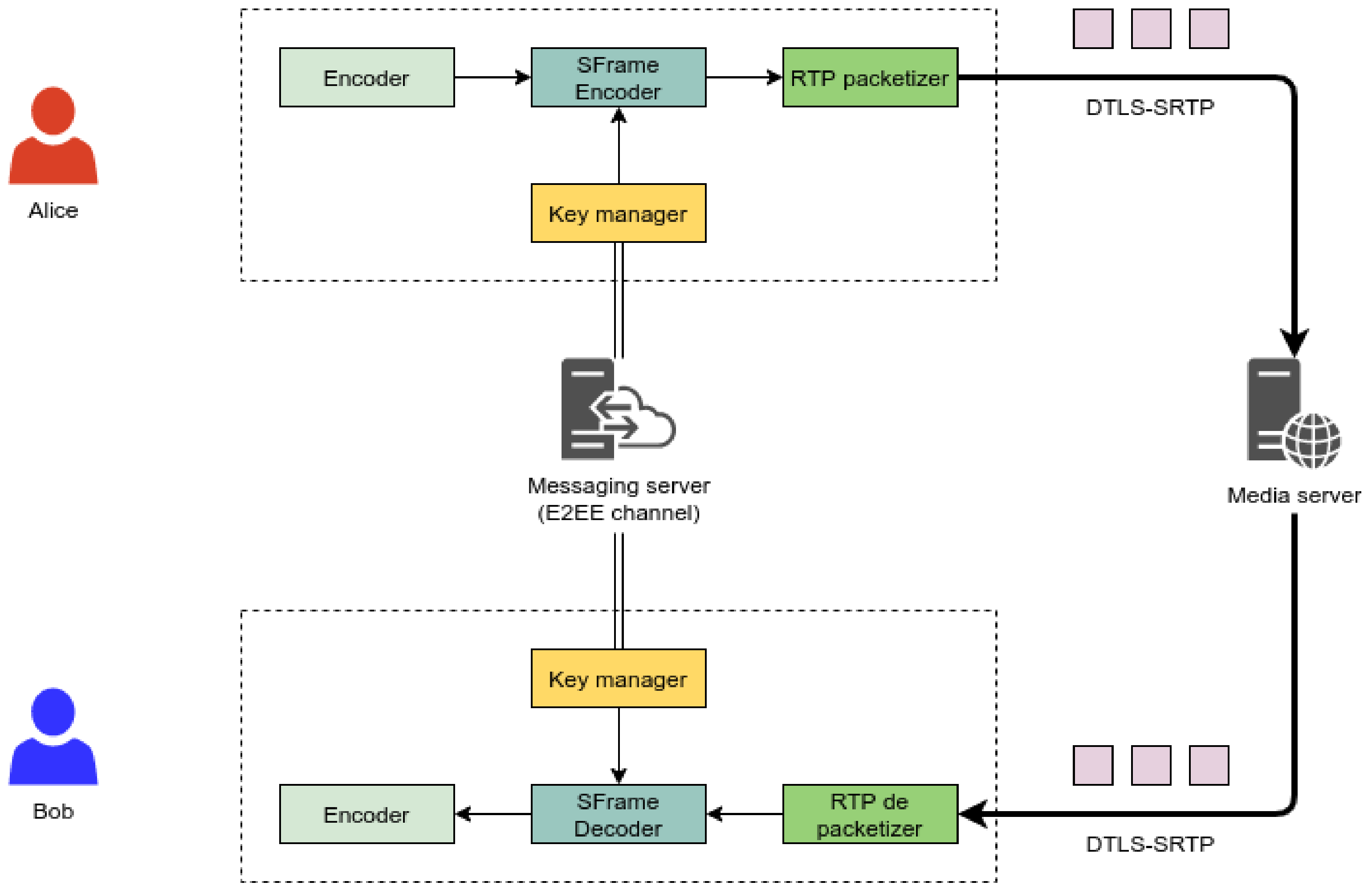
WebRTC 동향

sFrame (Secure Frame)

Alexandre Gouaillard

WebRTC 동향

sFrame (Secure Frame)



meetecho.com
Lorenzo
Janus

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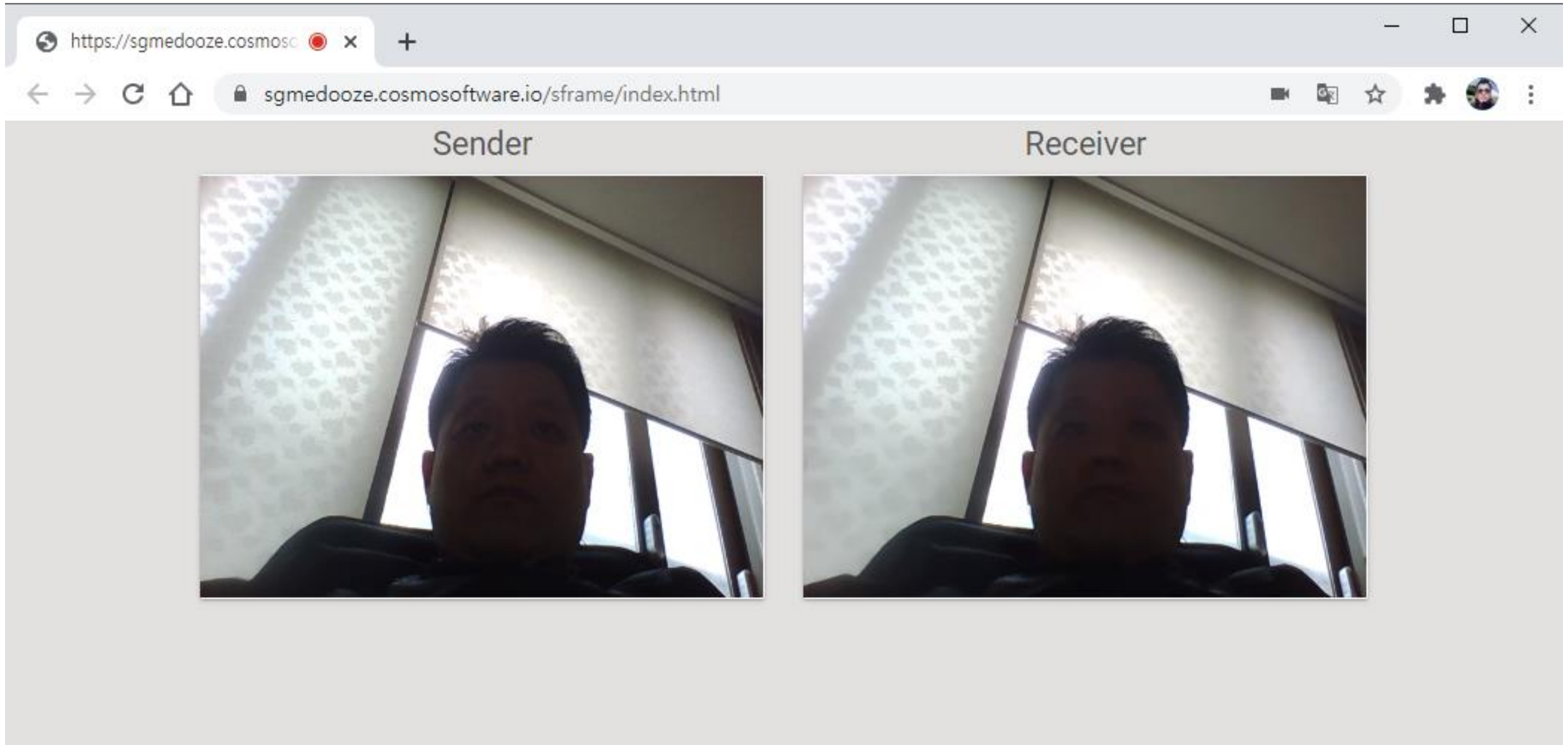
WebRTC 동향

sFrame (Secure Frame)



https://sgmedooze.cosmosoftware.io/sframe/index.html

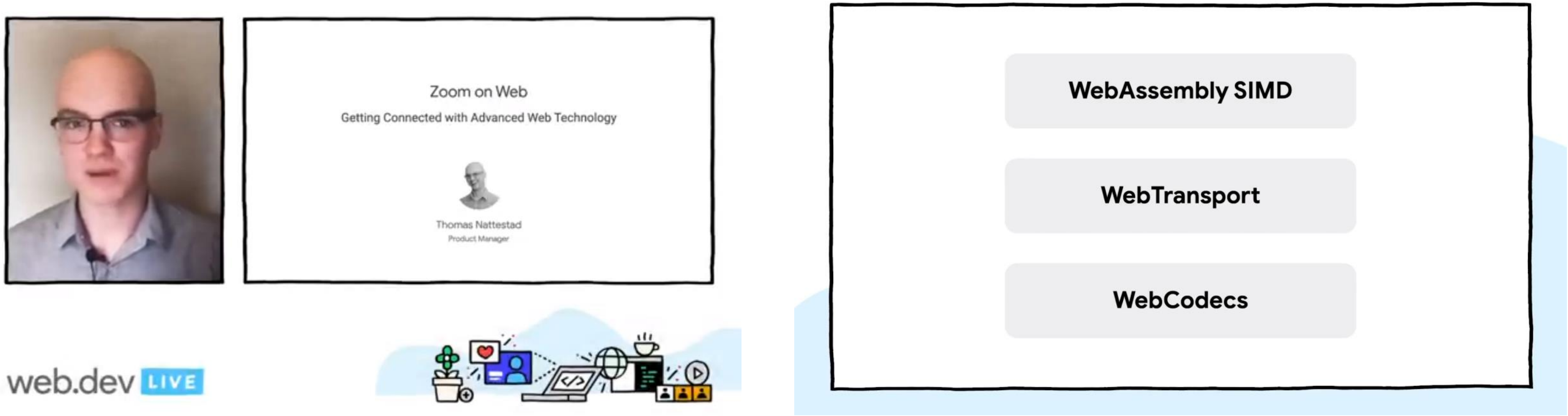
WebRTC 동향

sFrame (Secure Frame)

SFrame.js
Pure javascript library implementing the SFrame end to end encryption based on webcrypto.

https://github.com/medooze/sframe

zoom on web



https://youtu.be/uuKBqHDnJ2k

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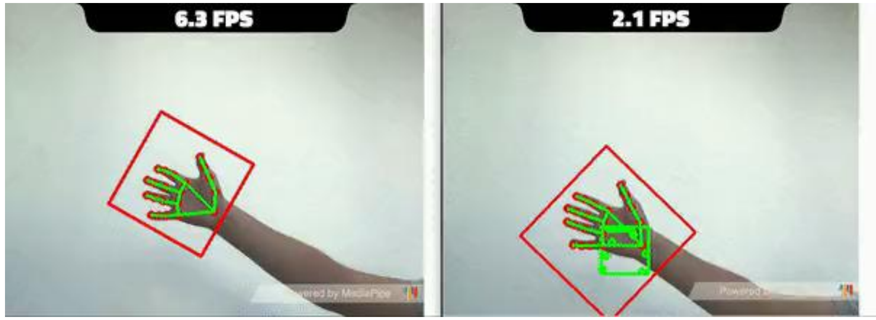
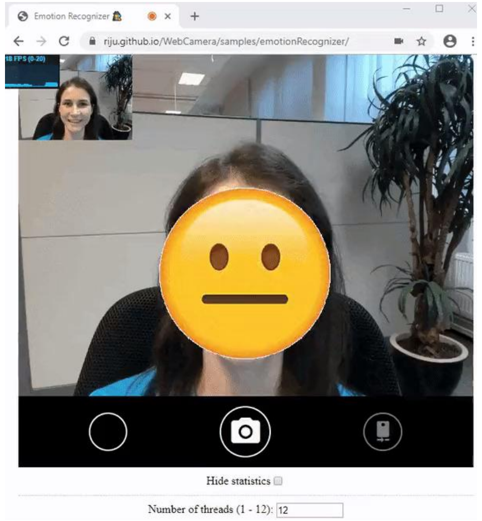
zoom on web

WebAssembly SIMD (Single Instruction Multiple Data)

- 오디오 / 비디오 코덱 및 이미지 프로세서와 같은 컴퓨팅 집약적 인 애플리케이션은 일반적으로 웹 브라우저 외부에서 SIMD를 활용.
- Zoom 내의 예는 사용자의 실제 배경을 가상 배경으로 대체.



https://v8.dev/features/simd



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WebTransport

- WebTransport는 WebSocket과 몇 가지 유사점이있는 WICG 제안이지만 WebSocket 내의 단일 스트림에 비해 여러 개의 취소 가능한 스트림을 지원
- WebTransport는 신뢰할 수없는 데이터 그램과 신뢰할 수있는 스트림 기반 메커니즘을 통해 양방향 전송을 제공

WebTransport vs. other networking APIs

WebTransport	WebSockets	WebRTC
Reliable & Unreliable	Reliable	Reliable & Unreliable
In development	Widely available	Widely available
Low level API	Low Level API	High level complete solution
Client - Server	Client - Server	Client-Server & Peer-to-Peer
Multiple, cancellable streams	Single stream	Multiple Streams

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WebCodecs

- WebCodecs API는 개발자가 브라우저에서 이미 사용 가능한 코덱을 활용하여 스트림 및 신호를 인코딩하거나 레코딩 할 수있는 직접 액세스를 제공

- 이전에는 개발자가 코덱을 제어하려면 JavaScript 또는 WebAssembly 코드베이스의 일부로 코덱을 제공해야 했었음

Use Case -> Capability Mapping

Use Case	WebCodecs Capabilities
Video Editing Tools	Frame-by-frame faster than real-time decode/ encode
Cloud Gaming/Inbound Low Latency Live Streams	Rendering of decoded frames without delay without WebRTC
Outbound Live streaming	Containerless stream, encoded without output buffers without WebRTC
Real Time Communications (Video conferencing)	Efficient transforms of decoded video frames & ability to apply additional effects

정리

- 표준 문서 완료는 안되었지만, 2017년말 이후 기본 규격은 완성된 상태
- 요구사항 대응을 위한 확장 API가 추가되고 정리되는 상태
- Media Capture and Streams : 많은 기능이 브라우저에서 75% 동작
- 새로운 접근에 관심을
  - WebTransport, WebCodecs
  - QUIC Protocol
  - WASM, DataChannel in Workers
  - Low Level
- 다기능으로 인해 웹클라이언트 개발의 복잡도 증가될 것
- 웹기반 기능 구현 시 브라우저의 리소스 효율성 감안
- 실감나는 비대면 소통

감사합니다.

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