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Overview: Real Time Protocols for Browser-based Applications draft-ietf-rtcweb-overview-17

Abstract

This document gives an overview and context of a protocol suite intended for use with real-time applications that can be deployed in browsers - "real time communication on the Web".

It intends to serve as a starting and coordination point to make sure all the parts that are needed to achieve this goal are findable, and that the parts that belong in the Internet protocol suite are fully specified and on the right publication track.

This document is an Applicability Statement - it does not itself specify any protocol, but specifies which other specifications WebRTC compliant implementations are supposed to follow.

This document is a work item of the RTCWEB working group.

Status of This Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

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This Internet-Draft will expire on August 21, 2017.

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1. Introduction

The Internet was, from very early in its lifetime, considered a possible vehicle for the deployment of real-time, interactive applications - with the most easily imaginable being audio conversations (aka "Internet telephony") and video conferencing.

The first attempts to build this were dependent on special networks, special hardware and custom-built software, often at very high prices or at low quality, placing great demands on the infrastructure.

As the available bandwidth has increased, and as processors an other hardware has become ever faster, the barriers to participation have decreased, and it has become possible to deliver a satisfactory experience on commonly available computing hardware.

Still, there are a number of barriers to the ability to communicate universally - one of these is that there is, as of yet, no single set of communication protocols that all agree should be made available for communication; another is the sheer lack of universal identification systems (such as is served by telephone numbers or email addresses in other communications systems).

Development of The Universal Solution has proved hard, however, for all the usual reasons.

The last few years have also seen a new platform rise for deployment of services: The browser-embedded application, or "Web application". It turns out that as long as the browser platform has the necessary interfaces, it is possible to deliver almost any kind of service on it.

Traditionally, these interfaces have been delivered by plugins, which had to be downloaded and installed separately from the browser; in the development of HTML5, application developers see much promise in the possibility of making those interfaces available in a standardized way within the browser.

This memo describes a set of building blocks that can be made accessible and controllable through a Javascript API in a browser, and which together form a sufficient set of functions to allow the use of interactive audio and video in applications that communicate directly between browsers across the Internet. The resulting protocol suite is intended to enable all the applications that are described as required scenarios in the use cases document [I-D.ietf-rtcweb-use-cases-and-requirements].

Other efforts, for instance the W3C WEBRTC, Web Applications and Device API working groups, focus on making standardized APIs and interfaces available, within or alongside the HTML5 effort, for those functions; this memo concentrates on specifying the protocols and subprotocols that are needed to specify the interactions that happen across the network.

This memo uses the term "WebRTC" (note the case used) to refer to the overall effort consisting of both IETF and W3C efforts.

2. Principles and Terminology

2.1. Goals of this document

The goal of the WebRTC protocol specification is to specify a set of protocols that, if all are implemented, will allow an implementation to communicate with another implementation using audio, video and data sent along the most direct possible path between the participants.

This document is intended to serve as the roadmap to the WebRTC specifications. It defines terms used by other parts of the WebRTC protocol specifications, lists references to other specifications that don't need further elaboration in the WebRTC context, and gives pointers to other documents that form part of the WebRTC suite.

By reading this document and the documents it refers to, it should be possible to have all information needed to implement an WebRTC compatible implementation.

2.2. Relationship between API and protocol

The total WebRTC effort consists of two major parts, each consisting of multiple documents:

- A protocol specification, done in the IETF

- A Javascript API specification, defined in a series of W3C documents [W3C.WD-webrtc-20120209][W3C.WD-mediacapture-streams-20120628]

Together, these two specifications aim to provide an environment where Javascript embedded in any page, when suitably authorized by its user, is able to set up communication using audio, video and auxiliary data, as long as the browser supports this specification. The browser environment does not constrain the types of application in which this functionality can be used.

The protocol specification does not assume that all implementations implement this API; it is not intended to be necessary for interoperation to know whether the entity one is communicating with is a browser or another device implementing this specification.

The goal of cooperation between the protocol specification and the API specification is that for all options and features of the protocol specification, it should be clear which API calls to make to exercise that option or feature; similarly, for any sequence of API calls, it should be clear which protocol options and features will be invoked. Both subject to constraints of the implementation, of course.

For the purpose of this document, we define the following terminology to talk about WebRTC things:

- A WebRTC browser (also called a WebRTC User Agent or WebRTC UA) is something that conforms to both the protocol specification and the Javascript API defined above.

- A WebRTC non-browser is something that conforms to the protocol specification, but does not claim to implement the Javascript API. This can also be called a "WebRTC device" or "WebRTC native application".

- A WebRTC endpoint is either a WebRTC browser or a WebRTC non-browser. It conforms to the protocol specification.

- A WebRTC-compatible endpoint is an endpoint that is able to successfully communicate with a WebRTC endpoint, but may fail to meet some requirements of a WebRTC endpoint. This may limit where in the network such an endpoint can be attached, or may limit the security guarantees that it offers to others. It is not constrained by this specification; when it is mentioned at all, it is to note the implications on WebRTC-compatible endpoints of the requirements placed on WebRTC endpoints.

- A WebRTC gateway is a WebRTC-compatible endpoint that mediates media traffic to non-WebRTC entities.

All WebRTC browsers are WebRTC endpoints, so any requirement on a WebRTC endpoint also applies to a WebRTC browser.

A WebRTC non-browser may be capable of hosting applications in a similar way to the way in which a browser can host Javascript applications, typically by offering APIs in other languages. For instance it may be implemented as a library that offers a C++ API intended to be loaded into applications. In this case, similar security considerations as for Javascript may be needed; however, since such APIs are not defined or referenced here, this document cannot give any specific rules for those interfaces.

WebRTC gateways are described in a separate document, [I-D.ietf-rtcweb-gateways].

2.3. On interoperability and innovation

The "Mission statement of the IETF" [RFC3935] states that "The benefit of a standard to the Internet is in interoperability - that multiple products implementing a standard are able to work together in order to deliver valuable functions to the Internet's users."

Communication on the Internet frequently occurs in two phases:

- Two parties communicate, through some mechanism, what functionality they both are able to support

- They use that shared communicative functionality to communicate, or, failing to find anything in common, give up on communication.

There are often many choices that can be made for communicative functionality; the history of the Internet is rife with the proposal, standardization, implementation, and success or failure of many types of options, in all sorts of protocols.

The goal of having a mandatory to implement function set is to prevent negotiation failure, not to preempt or prevent negotiation.

The presence of a mandatory to implement function set serves as a strong changer of the marketplace of deployment - in that it gives a guarantee that, as long as you conform to a specification, and the other party is willing to accept communication at the base level of that specification, you can communicate successfully.

The alternative - that of having no mandatory to implement - does not mean that you cannot communicate, it merely means that in order to be part of the communications partnership, you have to implement the standard "and then some" - that "and then some" usually being called a profile of some sort; in the version most antithetical to the Internet ethos, that "and then some" consists of having to use a specific vendor's product only.

2.4. Terminology

The following terms are used across the documents specifying the WebRTC suite, in the specific meanings given here. Not all terms are used in this document. Other terms are used in their commonly used meaning.

The list is in alphabetical order.

Agent: Undefined term. See "SDP Agent" and "ICE Agent".

API: Application Programming Interface - a specification of a set of calls and events, usually tied to a programming language or an abstract formal specification such as WebIDL, with its defined semantics.

Browser: Used synonymously with "Interactive User Agent" as defined in the HTML specification [W3C.WD-html5-20110525]. See also "WebRTC User Agent".

Data channel: An abstraction that allows data to be sent between WebRTC endpoints in the form of messages. Two endpoints can have multiple data channels between them.

ICE Agent: An implementation of the Interactive Connectivty Establishment (ICE) [I-D.ietf-ice-rfc5245bis] protocol. An ICE Agent may also be an SDP Agent, but there exist ICE Agents that do not use SDP (for instance those that use Jingle).

Interactive: Communication between multiple parties, where the expectation is that an action from one party can cause a reaction by another party, and the reaction can be observed by the first party, with the total time required for the action/reaction/ observation is on the order of no more than hundreds of milliseconds.

Media: Audio and video content. Not to be confused with "transmission media" such as wires.

Media path: The path that media data follows from one WebRTC endpoint to another.

Protocol: A specification of a set of data units, their representation, and rules for their transmission, with their defined semantics. A protocol is usually thought of as going between systems.

Real-time media: Media where generation of content and display of content are intended to occur closely together in time (on the order of no more than hundreds of milliseconds). Real-time media can be used to support interactive communication.

SDP Agent: The protocol implementation involved in the SDP offer/answer exchange, as defined in [RFC3264] section 3.

Signaling: Communication that happens in order to establish, manage and control media paths and data paths.

Signaling Path: The communication channels used between entities participating in signaling to transfer signaling. There may be more entities in the signaling path than in the media path.

NOTE: Where common definitions exist for these terms, those definitions should be used to the greatest extent possible.

3. Architecture and Functionality groups

The model of real-time support for browser-based applications does not assume that the browser will contain all the functions that need to be performed in order to have a function such as a telephone or a video conferencing unit; the vision is that the browser will have the functions that are needed for a Web application, working in conjunction with its backend servers, to implement these functions.

This means that two vital interfaces need specification: The protocols that browsers use to talk to each other, without any intervening servers, and the APIs that are offered for a Javascript application to take advantage of the browser's functionality.

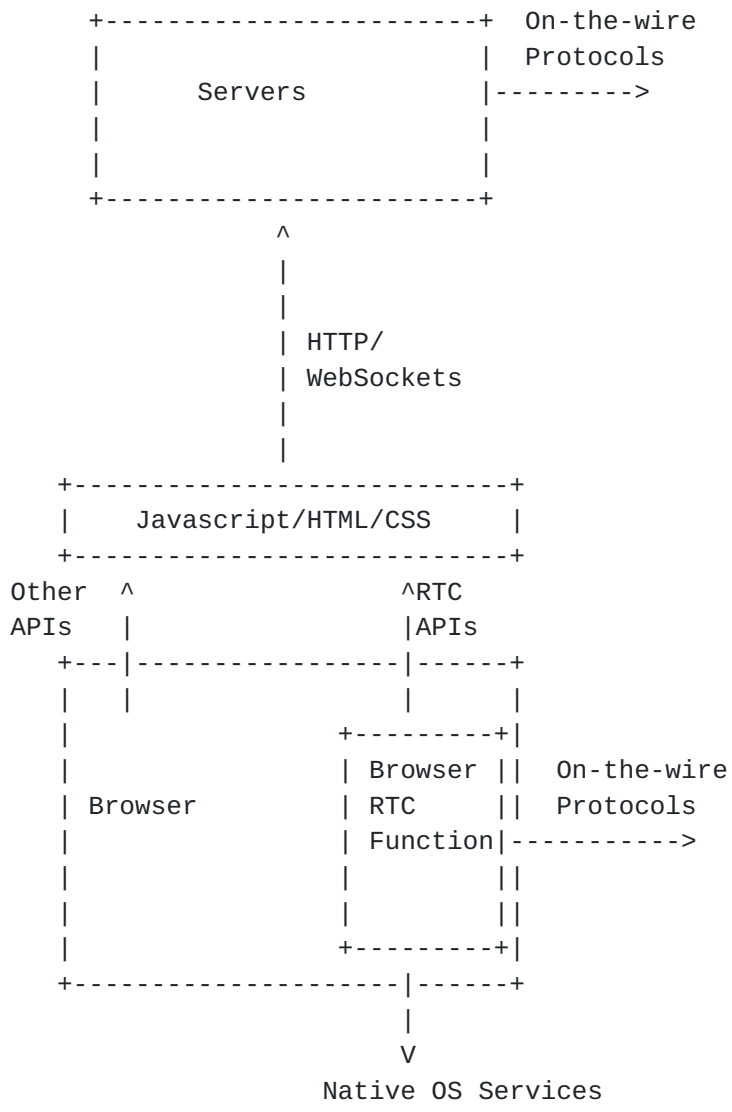

Figure 1: Browser Model

Note that HTTP and WebSockets are also offered to the Javascript application through browser APIs.

As for all protocol and API specifications, there is no restriction that the protocols can only be used to talk to another browser; since they are fully specified, any endpoint that implements the protocols

faithfully should be able to interoperate with the application running in the browser.

A commonly imagined model of deployment is the one depicted below.

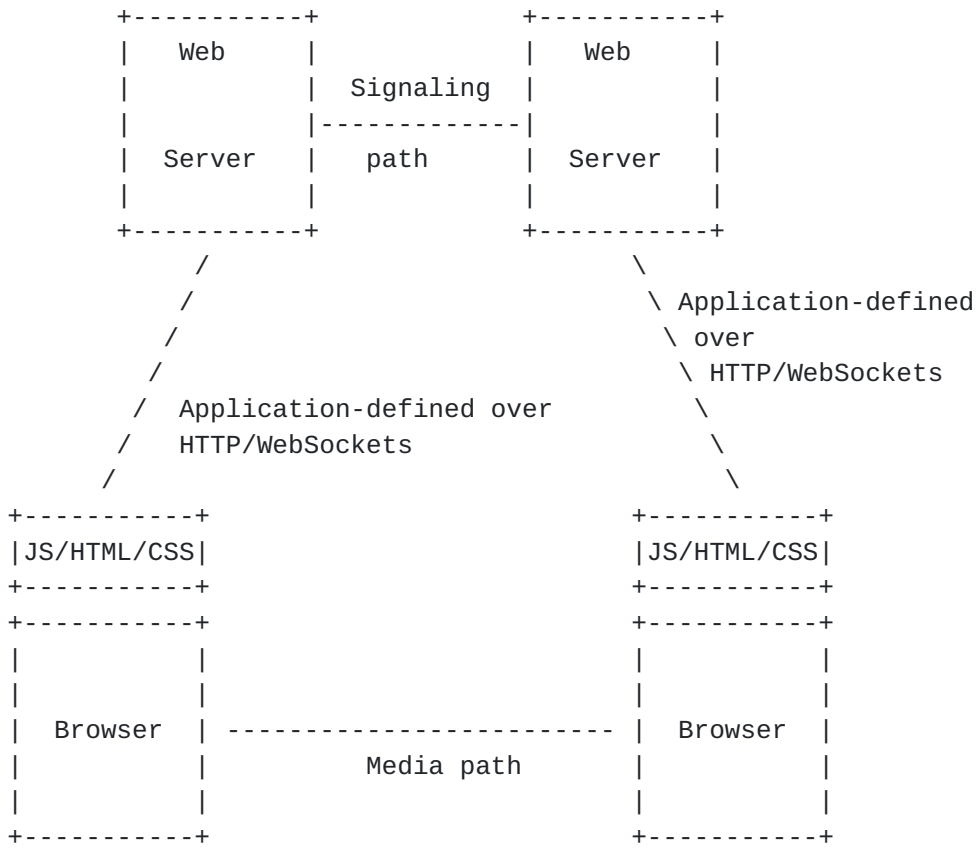

Figure 2: Browser RTC Trapezoid

On this drawing, the critical part to note is that the media path ("low path") goes directly between the browsers, so it has to be conformant to the specifications of the WebRTC protocol suite; the signaling path ("high path") goes via servers that can modify, translate or massage the signals as needed.

If the two Web servers are operated by different entities, the inter-server signaling mechanism needs to be agreed upon, either by standardization or by other means of agreement. Existing protocols (for example SIP [RFC3261] or XMPP [RFC6120]) could be used between servers, while either a standards-based or proprietary protocol could be used between the browser and the web server.

For example, if both operators' servers implement SIP, SIP could be used for communication between servers, along with either a

standardized signaling mechanism (e.g. SIP over WebSockets) or a proprietary signaling mechanism used between the application running in the browser and the web server. Similarly, if both operators' servers implement XMPP, XMPP could be used for communication between XMPP servers, with either a standardized signaling mechanism (e.g. XMPP over WebSockets or BOSH) or a proprietary signaling mechanism used between the application running in the browser and the web server.

The choice of protocols for client-server and inter-server signalling, and definition of the translation between them, is outside the scope of the WebRTC protocol suite described in the document.

The functionality groups that are needed in the browser can be specified, more or less from the bottom up, as:

- Data transport: TCP, UDP and the means to securely set up connections between entities, as well as the functions for deciding when to send data: Congestion management, bandwidth estimation and so on.

- Data framing: RTP, SCTP and other data formats that serve as containers, and their functions for data confidentiality and integrity.

- Data formats: Codec specifications, format specifications and functionality specifications for the data passed between systems. Audio and video codecs, as well as formats for data and document sharing, belong in this category. In order to make use of data formats, a way to describe them, a session description, is needed.

- Connection management: Setting up connections, agreeing on data formats, changing data formats during the duration of a call; SIP and Jingle/XMPP belong in this category.

- Presentation and control: What needs to happen in order to ensure that interactions behave in a non-surprising manner. This can include floor control, screen layout, voice activated image switching and other such functions - where part of the system require the cooperation between parties. XCON and Cisco/ Tandberg's TIP were some attempts at specifying this kind of functionality; many applications have been built without standardized interfaces to these functions.

- Local system support functions: These are things that need not be specified uniformly, because each participant may choose to do these in a way of the participant's choosing, without affecting

the bits on the wire in a way that others have to be cognizant of. Examples in this category include echo cancellation (some forms of it), local authentication and authorization mechanisms, OS access control and the ability to do local recording of conversations.

Within each functionality group, it is important to preserve both freedom to innovate and the ability for global communication. Freedom to innovate is helped by doing the specification in terms of interfaces, not implementation; any implementation able to communicate according to the interfaces is a valid implementation. Ability to communicate globally is helped both by having core specifications be unencumbered by IPR issues and by having the formats and protocols be fully enough specified to allow for independent implementation.

One can think of the three first groups as forming a "media transport infrastructure", and of the three last groups as forming a "media service". In many contexts, it makes sense to use a common specification for the media transport infrastructure, which can be embedded in browsers and accessed using standard interfaces, and "let a thousand flowers bloom" in the "media service" layer; to achieve interoperable services, however, at least the first five of the six groups need to be specified.

4. Data transport

Data transport refers to the sending and receiving of data over the network interfaces, the choice of network-layer addresses at each end of the communication, and the interaction with any intermediate entities that handle the data, but do not modify it (such as TURN relays).

It includes necessary functions for congestion control: When not to send data.

WebRTC endpoints MUST implement the transport protocols described in [I-D.ietf-rtcweb-transports].

5. Data framing and securing

The format for media transport is RTP [RFC3550]. Implementation of SRTP [RFC3711] is REQUIRED for all implementations.

The detailed considerations for usage of functions from RTP and SRTP are given in [I-D.ietf-rtcweb-rtp-usage]. The security considerations for the WebRTC use case are in [I-D.ietf-rtcweb-security], and the resulting security functions are described in [I-D.ietf-rtcweb-security-arch].

Considerations for the transfer of data that is not in RTP format is described in [I-D.ietf-rtcweb-data-channel], and a supporting protocol for establishing individual data channels is described in [I-D.ietf-rtcweb-data-protocol]. WebRTC endpoints MUST implement these two specifications.

WebRTC endpoints MUST implement [I-D.ietf-rtcweb-rtp-usage], [I-D.ietf-rtcweb-security], [I-D.ietf-rtcweb-security-arch], and the requirements they include.

6. Data formats

The intent of this specification is to allow each communications event to use the data formats that are best suited for that particular instance, where a format is supported by both sides of the connection. However, a minimum standard is greatly helpful in order to ensure that communication can be achieved. This document specifies a minimum baseline that will be supported by all implementations of this specification, and leaves further codecs to be included at the will of the implementor.

WebRTC endpoints that support audio and/or video MUST implement the codecs and profiles required in [I-D.ietf-rtcweb-audio] and [I-D.ietf-rtcweb-video].

7. Connection management

The methods, mechanisms and requirements for setting up, negotiating and tearing down connections is a large subject, and one where it is desirable to have both interoperability and freedom to innovate.

The following principles apply:

1. The WebRTC media negotiations will be capable of representing the same SDP offer/answer semantics that are used in SIP [RFC3264], in such a way that it is possible to build a signaling gateway between SIP and the WebRTC media negotiation.

2. It will be possible to gateway between legacy SIP devices that support ICE and appropriate RTP / SDP mechanisms, codecs and security mechanisms without using a media gateway. A signaling gateway to convert between the signaling on the web side to the SIP signaling may be needed.

3. When a new codec is specified, and the SDP for the new codec is specified in the MMUSIC WG, no other standardization should be required for it to be possible to use that in the web browsers. Adding new codecs which might have new SDP parameters should not

change the APIs between the browser and Javascript application. As soon as the browsers support the new codecs, old applications written before the codecs were specified should automatically be able to use the new codecs where appropriate with no changes to the JS applications.

The particular choices made for WebRTC, and their implications for the API offered by a browser implementing WebRTC, are described in [I-D.ietf-rtcweb-jsep].

WebRTC browsers MUST implement [I-D.ietf-rtcweb-jsep].

WebRTC endpoints MUST implement the functions described in that document that relate to the network layer (for example Bundle, RTCP-mux and Trickle ICE), but do not need to support the API functionality described there.

8. Presentation and control

The most important part of control is the user's control over the browser's interaction with input/output devices and communications channels. It is important that the user have some way of figuring out where his audio, video or texting is being sent, for what purported reason, and what guarantees are made by the parties that form part of this control channel. This is largely a local function between the browser, the underlying operating system and the user interface; this is specified in the peer connection API [W3C.WD-webrtc-20120209], and the media capture API [W3C.WD-mediacapture-streams-20120628].

WebRTC browsers MUST implement these two specifications.

9. Local system support functions

These are characterized by the fact that the quality of these functions strongly influence the user experience, but the exact algorithm does not need coordination. In some cases (for instance echo cancellation, as described below), the overall system definition may need to specify that the overall system needs to have some characteristics for which these facilities are useful, without requiring them to be implemented a certain way.

Local functions include echo cancellation, volume control, camera management including focus, zoom, pan/tilt controls (if available), and more.

Certain parts of the system SHOULD conform to certain properties, for instance:

- Echo cancellation should be good enough to achieve the suppression of acoustical feedback loops below a perceptually noticeable level.

- Privacy concerns MUST be satisfied; for instance, if remote control of camera is offered, the APIs should be available to let the local participant figure out who's controlling the camera, and possibly decide to revoke the permission for camera usage.

- Automatic gain control, if present, should normalize a speaking voice into a reasonable dB range.

The requirements on WebRTC systems with regard to audio processing are found in [I-D.ietf-rtcweb-audio]; the proposed API for control of local devices are found in [W3C.WD-mediacapture-streams-20120628].

WebRTC endpoints MUST implement the processing functions in [I-D.ietf-rtcweb-audio]. (Together with the requirement in Section 6, this means that WebRTC endpoints MUST implement the whole document.)

10. IANA Considerations

This document makes no request of IANA.

Note to RFC Editor: this section may be removed on publication as an RFC.

11. Security Considerations

Security of the web-enabled real time communications comes in several pieces:

- Security of the components: The browsers, and other servers involved. The most target-rich environment here is probably the browser; the aim here should be that the introduction of these components introduces no additional vulnerability.

- Security of the communication channels: It should be easy for a participant to reassure himself of the security of his communication - by verifying the crypto parameters of the links he himself participates in, and to get reassurances from the other parties to the communication that they promise that appropriate measures are taken.

- Security of the partners' identity: verifying that the participants are who they say they are (when positive

identification is appropriate), or that their identity cannot be uncovered (when anonymity is a goal of the application).

The security analysis, and the requirements derived from that analysis, is contained in [I-D.ietf-rtcweb-security].

It is also important to read the security sections of [W3C.WD-mediacapture-streams-20120628] and [W3C.WD-webrtc-20120209].

12. Acknowledgements

The number of people who have taken part in the discussions surrounding this draft are too numerous to list, or even to identify. The ones below have made special, identifiable contributions; this does not mean that others' contributions are less important.

Thanks to Cary Bran, Cullen Jennings, Colin Perkins, Magnus Westerlund and Joerg Ott, who offered technical contributions on various versions of the draft.

Thanks to Jonathan Rosenberg, Matthew Kaufman and others at Skype for the ASCII drawings in section 1.

Thanks to Bjoern Hoehrmann, Colin Perkins, Colton Shields, Eric Rescorla, Heath Matlock, Henry Sinnreich, Justin Uberti, Keith Drage, Magnus Westerlund, Olle E. Johansson and Simon Leinen for document review.

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13.2. Informative References

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Appendix A. Change log

This section may be deleted by the RFC Editor when preparing for publication.

A.1. Changes from draft-alvestrand-dispatch-rtcweb-datagram-00 to -01

Added section "On interoperability and innovation"

Added data confidentiality and integrity to the "data framing" layer

Added congestion management requirements in the "data transport" layer section

Changed need for non-media data from "question: do we need this?" to "Open issue: How do we do this?"

Strengthened disclaimer that listed codecs are placeholders, not decisions.

More details on why the "local system support functions" section is there.

A.2. Changes from draft-alvestrand-dispatch-01 to draft-alvestrand-rtcweb-overview-00

Added section on "Relationship between API and protocol"

Added terminology section

Mentioned congestion management as part of the "data transport" layer in the layer list

A.3. Changes from draft-alvestrand-rtcweb-00 to -01

Removed most technical content, and replaced with pointers to drafts as requested and identified by the RTCWEB WG chairs.

Added content to acknowledgments section.

Added change log.

Spell-checked document.

A.4. Changes from draft-alvestrand-rtcweb-overview-01 to draft-ietf-rtcweb-overview-00

Changed draft name and document date.

Removed unused references

A.5. Changes from -00 to -01 of draft-ietf-rtcweb-overview

Added architecture figures to section 2.

Changed the description of "echo cancellation" under "local system support functions".

Added a few more definitions.

A.6. Changes from -01 to -02 of draft-ietf-rtcweb-overview

Added pointers to use cases, security and rtp-usage drafts (now WG drafts).

Changed description of SRTP from mandatory-to-use to mandatory-to-implement.

Added the "3 principles of negotiation" to the connection management section.

Added an explicit statement that ICE is required for both NAT and consent-to-receive.

A.7. Changes from -02 to -03 of draft-ietf-rtcweb-overview

Added references to a number of new drafts.

Expanded the description text under the "trapezoid" drawing with some more text discussed on the list.

Changed the "Connection management" sentence from "will be done using SDP offer/answer" to "will be capable of representing SDP offer/answer" - this seems more consistent with JSEP.

Added "security mechanisms" to the things a non-gatewayed SIP devices must support in order to not need a media gateway.

Added a definition for "browser".

A.8. Changes from -03 to -04 of draft-ietf-rtcweb-overview

Made introduction more normative.

Several wording changes in response to review comments from EKR

Added an appendix to hold references and notes that are not yet in a separate document.

A.9. Changes from -04 to -05 of draft-ietf-rtcweb-overview

Minor grammatical fixes. This is mainly a "keepalive" refresh.

A.10. Changes from -05 to -06

Clarifications in response to Last Call review comments. Inserted reference to draft-ietf-rtcweb-audio.

A.11. Changes from -06 to -07

Added a reference to the "unified plan" draft, and updated some references.

Otherwise, it's a "keepalive" draft.

A.12. Changes from -07 to -08

Removed the appendix that detailed transports, and replaced it with a reference to draft-ietf-rtcweb-transports. Removed now-unused references.

A.13. Changes from -08 to -09

Added text to the Abstract indicating that the intended status is an Applicability Statement.

A.14. Changes from -09 to -10

Defined "WebRTC Browser" and "WebRTC device" as things that do, or don't, conform to the API.

Updated reference to data-protocol draft

Updated data formats to reference -rtcweb-audio- and not the expired -cbran draft.

Deleted references to -unified-plan

Deleted reference to -generic-idp (draft expired)

Added notes on which referenced documents WebRTC browsers or devices MUST conform to.

Added pointer to the security section of the API drafts.

A.15. Changes from -10 to -11

Added "WebRTC Gateway" as a third class of device, and referenced the doc describing them.

Made a number of text clarifications in response to document reviews.

A.16. Changes from -11 to -12

Refined entity definitions to define "WebRTC endpoint" and "WebRTC-compatible endpoint".

Changed remaining usage of the term "RTCWEB" to "WebRTC", including in the page header.

A.17. Changes from -12 to -13

Changed "WebRTC device" to be "WebRTC non-browser", per decision at IETF 91. This led to the need for "WebRTC endpoint" as the common label for both, and the usage of that term in the rest of the document.

Added words about WebRTC APIs in languages other than Javascript.

Referenced draft-ietf-rtcweb-video for video codecs to support.

A.18. Changes from -13 to -14

None. This is a "keepalive" update.

A.19. Changes from -14 to -15

Changed "gateways" reference to point to the WG document.

A.20. Changes from -15 to -16

None. This is a "keepalive" publication.

A.21. Changes from -16 to -17

Addressed review comments by Olle E. Johansson and Magnus Westerlund

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