**WebRTC** (**Web Real-Time Communication**) is a free, [open-source](https://en.wikipedia.org/wiki/Open-source_software) project that provides [web browsers](https://en.wikipedia.org/wiki/Web_browser) and [mobile applications](https://en.wikipedia.org/wiki/Mobile_application) with [real-time](https://en.wikipedia.org/wiki/Real-time_communication) communication (RTC) via simple [application programming interfaces](https://en.wikipedia.org/wiki/Application_programming_interface) (APIs). It allows audio and video communication to work inside web pages by allowing direct [peer-to-peer](https://en.wikipedia.org/wiki/Peer-to-peer) communication, eliminating the need to install [plugins](https://en.wikipedia.org/wiki/Plug-in_(computing)) or download native apps.[[3]](https://en.wikipedia.org/wiki/WebRTC#cite_note-revolutionizing-3) Supported by [Apple](https://en.wikipedia.org/wiki/Apple_Inc.), [Google](https://en.wikipedia.org/wiki/Google), [Microsoft](https://en.wikipedia.org/wiki/Microsoft), [Mozilla](https://en.wikipedia.org/wiki/Mozilla), and [Opera](https://en.wikipedia.org/wiki/Opera_Software), WebRTC is being standardized through the [World Wide Web Consortium](https://en.wikipedia.org/wiki/World_Wide_Web_Consortium) (W3C) and the [Internet Engineering Task Force](https://en.wikipedia.org/wiki/Internet_Engineering_Task_Force) (IETF).[[4]](https://en.wikipedia.org/wiki/WebRTC#cite_note-org-4)

Its mission is to "enable rich, high-quality RTC applications to be developed for the browser, mobile platforms, and [IoT](https://en.wikipedia.org/wiki/Internet_of_things) devices, and allow them all to communicate via a common set of [protocols](https://en.wikipedia.org/wiki/Communication_protocol)".

History[[edit](https://en.wikipedia.org/w/index.php?title=WebRTC&action=edit&section=1)]

In May 2010, Google bought [Global IP Solutions](https://en.wikipedia.org/wiki/Global_IP_Solutions) or GIPS, a [VoIP](https://en.wikipedia.org/wiki/Voice_over_IP) and [videoconferencing](https://en.wikipedia.org/wiki/Videoconferencing) software company that had developed many components required for RTC, such as codecs and echo cancellation techniques. Google open-sourced the GIPS technology and engaged with relevant standards bodies at the [IETF](https://en.wikipedia.org/wiki/IETF) and [W3C](https://en.wikipedia.org/wiki/W3C) to ensure industry consensus.[[5]](https://en.wikipedia.org/wiki/WebRTC#cite_note-5)[[6]](https://en.wikipedia.org/wiki/WebRTC#cite_note-6) In May 2011, [Google](https://en.wikipedia.org/wiki/Google) released an [open-source](https://en.wikipedia.org/wiki/Open-source_software) project for browser-based real-time communication known as WebRTC.[[7]](https://en.wikipedia.org/wiki/WebRTC#cite_note-code-7) This has been followed by ongoing work to standardize the relevant [protocols](https://en.wikipedia.org/wiki/Communication_protocol) in the [IETF](https://en.wikipedia.org/wiki/Internet_Engineering_Task_Force)[[8]](https://en.wikipedia.org/wiki/WebRTC#cite_note-rtcweb_charter-8) and browser APIs in the W3C.[[9]](https://en.wikipedia.org/wiki/WebRTC#cite_note-w3-9)

In May 2011, [Ericsson](https://en.wikipedia.org/wiki/Ericsson) Labs built the first implementation of WebRTC using a modified [WebKit](https://en.wikipedia.org/wiki/WebKit) library.[[10]](https://en.wikipedia.org/wiki/WebRTC#cite_note-eric-10) In October 2011, the [W3C](https://en.wikipedia.org/wiki/World_Wide_Web_Consortium) published its first draft for the spec.[[11]](https://en.wikipedia.org/wiki/WebRTC#cite_note-first-wd-11) WebRTC milestones include the first cross-browser video call (February 2013), first cross-browser data transfers (February 2014), and as of July 2014 [Google Hangouts](https://en.wikipedia.org/wiki/Google_Hangouts) was "kind of" using WebRTC.[[12]](https://en.wikipedia.org/wiki/WebRTC#cite_note-nowak-12)

The W3C draft API was based on preliminary work done in the [WHATWG](https://en.wikipedia.org/wiki/WHATWG).[[13]](https://en.wikipedia.org/wiki/WebRTC#cite_note-whatwg-13) It was referred to as the ConnectionPeer API, and a pre-standards concept implementation was created at [Ericsson](https://en.wikipedia.org/wiki/Ericsson) Labs.[[14]](https://en.wikipedia.org/wiki/WebRTC#cite_note-ericsson-14) The WebRTC Working Group expects this specification to evolve significantly based on:

* Outcomes of ongoing exchanges in the companion RTCWEB group at [IETF](https://en.wikipedia.org/wiki/IETF)[[15]](https://en.wikipedia.org/wiki/WebRTC#cite_note-rtcweb-15) to define the set of [protocols](https://en.wikipedia.org/wiki/Communications_protocol) that, together with this document, define [real-time communications](https://en.wikipedia.org/wiki/Real-time_communication) in web browsers. While no one signaling protocol is mandated, [SIP](https://en.wikipedia.org/wiki/Session_Initiation_Protocol) over [WebSockets](https://en.wikipedia.org/wiki/WebSocket) ([RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [7118](https://tools.ietf.org/html/rfc7118)) is often used partially due to the applicability of SIP to most of the envisaged communication scenarios as well as the availability of open-source software such as [JsSIP](https://en.wikipedia.org/wiki/JsSIP).
* [Privacy issues](https://en.wikipedia.org/wiki/Internet_privacy) that arise when exposing local capabilities and local streams
* Technical discussions within the group, on implementing data channels in particular[[16]](https://en.wikipedia.org/wiki/WebRTC#cite_note-data-protocol-00-16)
* Experience gained through early experimentation
* Feedback from other groups and individuals

In November 2017, the WebRTC 1.0 specification transitioned from Working Draft to Candidate Recommendation.[[17]](https://en.wikipedia.org/wiki/WebRTC#cite_note-first-cr-17)

## verview[[edit](https://en.wikipedia.org/w/index.php?title=WebRTC&action=edit&section=2)]

### Design**[**[**edit**](https://en.wikipedia.org/w/index.php?title=WebRTC&action=edit&section=3)**]**

Major components of WebRTC include several [JavaScript](https://en.wikipedia.org/wiki/JavaScript) [APIs](https://en.wikipedia.org/wiki/Application_programming_interface):

* getUserMedia acquires the audio and video media (e.g., by accessing a device's camera and microphone).[[18]](https://en.wikipedia.org/wiki/WebRTC#cite_note-c-18)
* RTCPeerConnection enables audio and video communication between peers. It performs [signal processing](https://en.wikipedia.org/wiki/Signal_processing), [codec](https://en.wikipedia.org/wiki/Codec) handling, peer-to-peer communication, security, and [bandwidth](https://en.wikipedia.org/wiki/Bandwidth_(computing)) management.[[19]](https://en.wikipedia.org/wiki/WebRTC#cite_note-rtcpeerconnection-19)
* RTCDataChannel allows bidirectional communication of arbitrary data between peers. It uses the same API as [WebSockets](https://en.wikipedia.org/wiki/WebSocket) and has very low [latency](https://en.wikipedia.org/wiki/Network_latency).[[20]](https://en.wikipedia.org/wiki/WebRTC#cite_note-rtcdatachannel-20)

The WebRTC API also includes a statistics function:

* getStats allows the web application to retrieve a set of statistics about WebRTC sessions. These statistics data are being described in a separate W3C document.[[21]](https://en.wikipedia.org/wiki/WebRTC#cite_note-stats-21)

The WebRTC API includes **no provisions for signaling**, that is discovering peers to connect to and determine how to establish connections among them. Applications use [Interactive Connectivity Establishment](https://en.wikipedia.org/wiki/Interactive_Connectivity_Establishment) for connections and somehow manage sessions, possibly relying on any of [Session Initiation Protocol](https://en.wikipedia.org/wiki/Session_Initiation_Protocol), [Extensible Messaging and Presence Protocol](https://en.wikipedia.org/wiki/Extensible_Messaging_and_Presence_Protocol), [Message Queuing Telemetry Transport](https://en.wikipedia.org/wiki/Message_Queuing_Telemetry_Transport), [Matrix (protocol)](https://en.wikipedia.org/wiki/Matrix_(protocol)), or another protocol. Signaling may depend on one or more servers.[[22]](https://en.wikipedia.org/wiki/WebRTC#cite_note-22)[[23]](https://en.wikipedia.org/wiki/WebRTC#cite_note-23)

[RFC 7874](https://tools.ietf.org/html/rfc7874) requires implementations to provide [PCMA](https://en.wikipedia.org/wiki/A-law_algorithm)/[PCMU](https://en.wikipedia.org/wiki/PCMU) ([RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [3551](https://tools.ietf.org/html/rfc3551)), Telephone Event as [DTMF](https://en.wikipedia.org/wiki/DTMF) ([RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [4733](https://tools.ietf.org/html/rfc4733)), and [Opus](https://en.wikipedia.org/wiki/Opus_(audio_format)) ([RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [6716](https://tools.ietf.org/html/rfc6716)) [audio codecs](https://en.wikipedia.org/wiki/Audio_codec) as minimum capabilities. The PeerConnection, data channel and media capture browser APIs are detailed in the W3C.

W3C is developing ORTC (Object Real-Time Communications) for WebRTC.[[24]](https://en.wikipedia.org/wiki/WebRTC#cite_note-ORTC-24)

### Examples**[**[**edit**](https://en.wikipedia.org/w/index.php?title=WebRTC&action=edit&section=4)**]**

Although initially developed for web browsers, WebRTC has applications for non-browser devices, including mobile platforms and [IoT devices](https://en.wikipedia.org/wiki/Internet_of_Things). Examples include browser-based [VoIP](https://en.wikipedia.org/wiki/VoIP) telephony, also called cloud phones or web phones, which allow calls to be made and received from within a web browser, replacing the requirement to download and install a softphone.[[25]](https://en.wikipedia.org/wiki/WebRTC#cite_note-25)

### Support**[**[**edit**](https://en.wikipedia.org/w/index.php?title=WebRTC&action=edit&section=5)**]**

WebRTC is supported by the following browsers:

* Desktop PC
  + [Microsoft Edge](https://en.wikipedia.org/wiki/Microsoft_Edge) 12+[[26]](https://en.wikipedia.org/wiki/WebRTC#cite_note-edge-26)
  + [Google Chrome](https://en.wikipedia.org/wiki/Google_Chrome) 28+
  + [Mozilla Firefox](https://en.wikipedia.org/wiki/Firefox) 22+[[27]](https://en.wikipedia.org/wiki/WebRTC#cite_note-Firefox22-27)
  + [Safari](https://en.wikipedia.org/wiki/Safari_(web_browser)) 11+[[28]](https://en.wikipedia.org/wiki/WebRTC#cite_note-Safari11-28)
  + [Opera](https://en.wikipedia.org/wiki/Opera_(web_browser)) 18+[[29]](https://en.wikipedia.org/wiki/WebRTC#cite_note-opera-18-29)
  + [Vivaldi](https://en.wikipedia.org/wiki/Vivaldi_(web_browser)) 1.9+
  + [Brave](https://en.wikipedia.org/wiki/Brave_(web_browser))
* [Android](https://en.wikipedia.org/wiki/Android_(operating_system))
  + Google Chrome 28+ (enabled by default since 29)
  + Mozilla Firefox 24+[[30]](https://en.wikipedia.org/wiki/WebRTC#cite_note-Firefox24-30)
  + Opera Mobile 12+
* [Chrome OS](https://en.wikipedia.org/wiki/Chrome_OS)
* [Firefox OS](https://en.wikipedia.org/wiki/Firefox_OS)
* [BlackBerry 10](https://en.wikipedia.org/wiki/BlackBerry_10)
* [iOS](https://en.wikipedia.org/wiki/IOS)
  + MobileSafari/WebKit ([iOS 11](https://en.wikipedia.org/wiki/IOS_11)+)
* [Tizen](https://en.wikipedia.org/wiki/Tizen) 3.0

[GStreamer](https://en.wikipedia.org/wiki/GStreamer) directly provides a free WebRTC implementation[[31]](https://en.wikipedia.org/wiki/WebRTC#cite_note-31)

## Concerns[[edit](https://en.wikipedia.org/w/index.php?title=WebRTC&action=edit&section=6)]

In January 2015, [TorrentFreak](https://en.wikipedia.org/wiki/TorrentFreak) reported a serious security flaw in browsers that support WebRTC, saying that it compromised the security of [VPN](https://en.wikipedia.org/wiki/Virtual_private_network) tunnels by exposing the true [IP address](https://en.wikipedia.org/wiki/IP_address) of a user.[[32]](https://en.wikipedia.org/wiki/WebRTC#cite_note-IPleak-32) The IP address read requests are not visible in the browser's developer console, and they are not blocked by most [ad blocking](https://en.wikipedia.org/wiki/Ad_blocking)/[privacy](https://en.wikipedia.org/wiki/Internet_privacy)/[security](https://en.wikipedia.org/wiki/Internet_security) add-ons, enabling online tracking by advertisers and other entities despite precautions[[33]](https://en.wikipedia.org/wiki/WebRTC#cite_note-webrtc-ips-33) (however the [uBlock Origin](https://en.wikipedia.org/wiki/UBlock_Origin) add-on can fix this problem). As of September 2019, this WebRTC flaw still surfaces on Firefox 69.x and still by default exposes the user's internal IP address to the web.[[34]](https://en.wikipedia.org/wiki/WebRTC#cite_note-34)

## See also[[edit](https://en.wikipedia.org/w/index.php?title=WebRTC&action=edit&section=7)]

* [Global IP Solutions](https://en.wikipedia.org/wiki/Global_IP_Solutions) (GIPS)
* [Real-time Transport Protocol](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol) (RTP)
* [Session Description Protocol](https://en.wikipedia.org/wiki/Session_Description_Protocol) (SDP)
* [WebRTC Gateway](https://en.wikipedia.org/wiki/WebRTC_Gateway)

## References[[edit](https://en.wikipedia.org/w/index.php?title=WebRTC&action=edit&section=8)]

* 1. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-1) [*"WebRTC 1.0: Real-time Communication Between Browsers"*](https://www.w3.org/TR/webrtc/). World Wide Web Consortium. 27 September 2018*. Retrieved 25 March 2019*.
  2. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-2) [*"Src/webrtc - Git at Google"*](https://webrtc.googlesource.com/src/webrtc/).
  3. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-revolutionizing_3-0) [How WebRTC Is Revolutionizing Telephony](http://blogs.trilogy-lte.com/post/77427158750/how-webrtc-is-revolutionizing-telephony). Blogs.trilogy-lte.com (2014-02-21). Retrieved on 2014-04-11.
  4. ^ [Jump up to:***a***](https://en.wikipedia.org/wiki/WebRTC#cite_ref-org_4-0) [***b***](https://en.wikipedia.org/wiki/WebRTC#cite_ref-org_4-1) [*"WebRTC"*](https://archive.today/20180109223529/https:/webrtc.org/). WebRTC. Archived from [*the original*](https://webrtc.org/) on 9 January 2018*. Retrieved 6 February 2018*.
  5. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-5) [*"Are the WebRTC components from Google's acquisition of Global IP Solutions?"*](https://webrtc.org/faq/#are-the-webrtc-components-from-googles-acquisition-of-global-ip-solutions). WebRTC*. Retrieved 6 February 2018*.
  6. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-6) Wauters, Robin (18 May 2010). [*"Google makes $68.2 million cash offer for Global IP Solutions"*](https://techcrunch.com/2010/05/18/google-makes-68-2-million-cash-offer-for-global-ip-solutions). TechCrunch*. Retrieved 6 February 2018*.
  7. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-code_7-0) Harald Alvestrand (2011-05-31). [*"Google release of WebRTC source code"*](http://lists.w3.org/Archives/Public/public-webrtc/2011May/0022.html). public-webrtc@w3.org*. Retrieved 2012-09-12*.
  8. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-rtcweb_charter_8-0) [Charter of the Real-Time Communication in WEB-browsers (rtcweb) working group](http://datatracker.ietf.org/wg/rtcweb/charter/)
  9. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-w3_9-0) [*"WebRTC 1.0: Real-time Communication Between Browsers"*](http://www.w3.org/TR/webrtc/). W3.org*. Retrieved 2012-09-12*.
  10. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-eric_10-0) Stefan Håkansson; Stefan Ålund (26 May 2011). [*"Beyond HTML5: Experiment with Real-Time Communication in a Browser"*](https://www.ericsson.com/research-blog/beyond-html5-experiment-real-time-communication-browser). Ericsson Research blog*. Retrieved 6 February 2018*.
  11. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-first-wd_11-0) [*"WebRTC 1.0: Real-time Communication Between Browsers (W3C Working Draft 27 October 2011)"*](https://www.w3.org/TR/2011/WD-webrtc-20111027/). World Wide Web Consortium. 27 October 2011*. Retrieved 6 February 2018*.
  12. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-nowak_12-0) Nowak, Szymon. [*"WebRTC: So Much More Than Videoconferencing"*](https://szimek.github.io/presentation-meetjs-summit-2014-webrtc/#16). GitHub*. Retrieved 6 February 2018*.
  13. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-whatwg_13-0) [*"Introduction — HTML Standard"*](https://www.whatwg.org/specs/web-apps/current-work/multipage/introduction.html#history-1). Whatwg.org*. Retrieved 2012-09-12*.
  14. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-ericsson_14-0) [*"Beyond HTML5: Peer-to-Peer Conversational Video"*](http://www.ericsson.com/research-blog/context-aware-communication/beyond-html5-peer-peer-conversational-video/). Labs.ericsson.com*. Retrieved 2012-09-12*.
  15. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-rtcweb_15-0) [*"Rtcweb Status Pages"*](http://tools.ietf.org/wg/rtcweb/). Tools.ietf.org*. Retrieved 2012-09-12*.
  16. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-data-protocol-00_16-0) [*"draft-jesup-rtcweb-data-protocol-00 - WebRTC Data Channel Protocol"*](http://tools.ietf.org/html/draft-jesup-rtcweb-data-protocol-00). Tools.ietf.org*. Retrieved 2012-09-12*.
  17. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-first-cr_17-0) [*"WebRTC 1.0: Real-time Communication Between Browsers (W3C Candidate Recommendation 02 November 2017)"*](https://www.w3.org/TR/2017/CR-webrtc-20171102/). 2 November 2017.
  18. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-c_18-0) [*"Media Capture and Streams: getUserMedia"*](https://www.w3.org/TR/mediacapture-streams/#dom-mediadevices-getusermedia). W3C. 2013-09-03*. Retrieved 2014-01-15*.
  19. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-rtcpeerconnection_19-0) [*"WebRTC: RTCPeerConnection Interface"*](http://www.w3.org/TR/webrtc/#rtcpeerconnection-interface). W3C. 2013-09-10*. Retrieved 2014-01-15*.
  20. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-rtcdatachannel_20-0) [*"WebRTC: RTCDataChannel"*](http://www.w3.org/TR/webrtc/#rtcdatachannel). W3C. 2013-09-10*. Retrieved 2014-01-15*.
  21. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-stats_21-0) [*"Identifiers for WebRTC's Statistics API"*](https://w3c.github.io/webrtc-stats/webrtc-stats.html). W3C. 2014-09-29.
  22. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-22) Tsahi Levent-Levi (13 April 2020). [*"WebRTC Server: What is it exactly?"*](https://bloggeek.me/webrtc-server/). BlogGeek.me.
  23. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-23) Tsahi Levent-Levi (13 November 2014). [*"Matrix.org and WebRTC: An Interview with Matthew Hodgson"*](https://bloggeek.me/matrix-webrtc-interview/). BlogGeek.me.
  24. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-ORTC_24-0) [*"W3C ORTC (Object Real-time Communications) Community Group"*](http://www.w3.org/community/ortc/).
  25. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-25) [*"Catch the Babelfish: Irish telco devises a new kind of cloud phone"*](https://www.siliconrepublic.com/comms/babelfish-softphone-cloud-goldfish). November 2017.
  26. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-edge_26-0) [*"ORTC API is now available in Microsoft Edge"*](https://blogs.windows.com/msedgedev/2015/09/18/ortc-api-is-now-available-in-microsoft-edge/). Microsoft. 2015-09-18.
  27. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-Firefox22_27-0) [Firefox Notes - Desktop](https://www.mozilla.org/en-US/firefox/22.0/releasenotes/). Mozilla.org (2013-06-25). Retrieved on 2014-04-11.
  28. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-Safari11_28-0) [*"Safari 11.0"*](https://developer.apple.com/library/content/releasenotes/General/WhatsNewInSafari/Safari_11_0/Safari_11_0.html). Apple Inc*. Retrieved 6 June 2017*.
  29. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-opera-18_29-0) [Opera News](http://blogs.opera.com/news/2013/11/opera-18/). blogs.opera.com (2013-11-19). Retrieved on 2015-09-17.
  30. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-Firefox24_30-0) [Firefox Notes - Desktop](https://www.mozilla.org/en-US/mobile/24.0/releasenotes/). Mozilla.org (2013-09-17). Retrieved on 2014-08-04.
  31. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-31) [*"GStreamer 1.14 release notes"*](https://gstreamer.freedesktop.org/releases/1.14/). gstreamer.freedesktop.org*. Retrieved 2019-12-19*. since version 1.14
  32. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-IPleak_32-0) [Huge Security Flaw Leaks VPN Users’ Real IP-addresses](https://torrentfreak.com/huge-security-flaw-leaks-vpn-users-real-ip-addresses-150130/) TorrentFreak.com (2015-01-30). Retrieved on 2015-02-21.
  33. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-webrtc-ips_33-0) [STUN IP Address requests for WebRTC](https://github.com/diafygi/webrtc-ips) Retrieved on 2015-02-21.
  34. [**^**](https://en.wikipedia.org/wiki/WebRTC#cite_ref-34) Raymond Hill (26 March 2016). [*"Prevent WebRTC from leaking local IP address"*](https://github.com/gorhill/uBlock/wiki/Prevent-WebRTC-from-leaking-local-IP-address). uBlock Origin documentation*. Retrieved 1 Sep 2016*.

## Further reading[[edit](https://en.wikipedia.org/w/index.php?title=WebRTC&action=edit&section=9)]

* Proust, S., ed. (May 2016). [*Additional WebRTC Audio Codecs for Interoperability*](https://tools.ietf.org/html/rfc7875). [*IETF*](https://en.wikipedia.org/wiki/Internet_Engineering_Task_Force). [*doi*](https://en.wikipedia.org/wiki/Doi_(identifier)):[*10.17487/RFC7875*](https://doi.org/10.17487%2FRFC7875). [*RFC*](https://en.wikipedia.org/wiki/RFC_(identifier)) [*7875*](https://tools.ietf.org/html/rfc7875)*. Retrieved 2016-10-12*.
* Valin, J. M.; Bran, C. (May 2016). [*WebRTC Audio Codec and Processing Requirements*](https://tools.ietf.org/html/rfc7874). [*IETF*](https://en.wikipedia.org/wiki/Internet_Engineering_Task_Force). [*doi*](https://en.wikipedia.org/wiki/Doi_(identifier)):[*10.17487/RFC7874*](https://doi.org/10.17487%2FRFC7874). [*RFC*](https://en.wikipedia.org/wiki/RFC_(identifier)) [*7874*](https://tools.ietf.org/html/rfc7874)*. Retrieved 2016-10-12*.
* Roach, A. B. (March 2016). [*WebRTC Video Processing and Codec Requirements*](https://tools.ietf.org/html/rfc7742). [*IETF*](https://en.wikipedia.org/wiki/Internet_Engineering_Task_Force). [*doi*](https://en.wikipedia.org/wiki/Doi_(identifier)):[*10.17487/RFC7742*](https://doi.org/10.17487%2FRFC7742). [*RFC*](https://en.wikipedia.org/wiki/RFC_(identifier)) [*7742*](https://tools.ietf.org/html/rfc7742)*. Retrieved 2016-10-12*.
* Perumal, M.; Wing, D.; Ravindranath, R.; Reddy, T.; Thomson, M. (October 2015). [*Session Traversal Utilities for NAT (STUN) Usage for Consent Freshness*](https://tools.ietf.org/html/rfc7675). [*IETF*](https://en.wikipedia.org/wiki/Internet_Engineering_Task_Force). [*doi*](https://en.wikipedia.org/wiki/Doi_(identifier)):[*10.17487/RFC7675*](https://doi.org/10.17487%2FRFC7675). [*RFC*](https://en.wikipedia.org/wiki/RFC_(identifier)) [*7675*](https://tools.ietf.org/html/rfc7675)*. Retrieved 2016-10-12*.
* Holmberg, C.; Hakansson, S.; Eriksson, G. (March 2015). [*Web Real-Time Communication Use Cases and Requirements*](https://tools.ietf.org/html/rfc7478). [*IETF*](https://en.wikipedia.org/wiki/Internet_Engineering_Task_Force). [*doi*](https://en.wikipedia.org/wiki/Doi_(identifier)):[*10.17487/RFC7478*](https://doi.org/10.17487%2FRFC7478). [*RFC*](https://en.wikipedia.org/wiki/RFC_(identifier)) [*7478*](https://tools.ietf.org/html/rfc7478)*. Retrieved 2016-10-12*.

## External links[[edit](https://en.wikipedia.org/w/index.php?title=WebRTC&action=edit&section=10)]

* [Official website](https://webrtc.org/) [Edit this at Wikidata](https://www.wikidata.org/wiki/Q1089715#P856)
* [W3C Web Real-Time Communications Working Group](https://www.w3.org/2011/04/webrtc/)
* [IETF Real-Time Communication in WEB-browsers (rtcweb) Working Group](https://tools.ietf.org/wg/rtcweb/)
* [Video chat demo app based on WebRTC](https://github.com/webrtc/apprtc)
* [Open source standalone implementation of WebRTC data channels](https://github.com/paullouisageneau/libdatachannel)