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WebRTC Codec and Media Processing Requirements  
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## [Abstract](#)

This document outlines the codec and media processing requirements for WebRTC client application and endpoint devices.

## [Status of this Memo](#)

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## **[1. Introduction](#)**

An integral part of the success and adoption of the Web Real Time Communications (WebRTC) will be the voice and video interoperability between WebRTC applications. This specification will outline the media processing and codec requirements for WebRTC client implementations.

## **[2. Terminology](#)**

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC 2119](#) [RFC2119].

## **[3. Codec Requirements](#)**

This section covers the audio and video codec requirements for WebRTC client applications. To ensure a baseline level of interoperability between WebRTC clients, a minimum set of required codecs are specified below. While this section specifies the codecs that will be mandated for all WebRTC client implementations, it leaves the question of supporting additional codecs to the will of the implementer.

### **[3.1. Audio Codec Requirements](#)**

WebRTC clients are REQUIRED to implement the following audio codecs.

\*PCMA/PCMU - 1 channel with a rate of 8000 Hz and a ptime of 20 - see section 4.5.14 of [\[RFC3551\]](#)

\*Telephone Event - [\[RFC4734\]](#)

\*Opus [draft-ietf-codec-opus]

### **3.2. Video Codec Requirements**

If the MPEG-LA issues an intent to offer H.264 baseline profile on a royalty free basis for use in browsers before March 15, 2012, then the REQUIRED video codecs will be H.264 baseline. If this does not happen by that the date, then the REQUIRED video codec will be VP8 [\[I-D.webm\]](#). The following feature list applies to all required video codecs. Required video codecs:

- \*MUST support at least 10 frames per second (fps) and SHOULD support 30 fps
- \*If VP8, then MUST support a the bilinear and none reconstruction filters
- \*OPTIONALLY offer support for additional color spaces
- \*MUST support a minimum resolution of 320x240
- \*SHOULD support resolutions of 1280x720, 720x480, 1024x768, 800x600, 640x480, 640 x 360 , 320x240

### **4. WebRTC Client Requirements**

It is plausible that the dominant near to mid-term WebRTC usage model will be people using the interactive audio and video capabilities to communicate with each other via web browsers running on a notebook computer that has built-in microphone and speakers. The notebook-as-communication-device paradigm presents challenging echo cancellation and audio gain problems, the specific remedy of which will not be mandated here. However, while no specific algorithm or standard will be required by WebRTC compatible clients, functionality such as automatic gain control, echo cancellation, headset detection and passing call control events to connected devices will improve the user experience and should be implemented by the endpoint device. To address the problems outlined above, suitable implementations of the functionality listed below SHOULD be available within an RTC-Web endpoint device.

- \*Automatic gain control
- \*Ability to override automatic gain control to manually set gain
- \*Auto-adjustments to gain control and echo cancellation algorithms based on if a headset or internal speakers/microphone is being used
- \*Echo cancellation, including acoustic echo cancellation

\*Headset detection

\*Call control event notification to connected devices such as headsets

## **5. Legacy VoIP Interoperability**

The codec requirements above will ensure, at a minimum, voice interoperability capabilities between WebRTC client applications and legacy phone systems.

Video interoperability will be dependent upon the MPEG-LA decision regarding H.264 baseline.

## **6. IANA Considerations**

This document makes no request of IANA.

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

## **7. Security Considerations**

The codec requirements have no additional security considerations other than those captured in [\[I-D.ekr-security-considerations-for-rtc-web\]](#).

## **8. Acknowledgements**

This draft incorporates ideas and text from various other drafts. In particular we would like to acknowledge, and say thanks for, work we incorporated from Harald Alvestrand.

## **9. References**

<b>[RFC2119]</b>	<a href="#">Bradner, S.</a> , " <a href="#">Key words for use in RFCs to Indicate Requirement Levels</a> ", BCP 14, RFC 2119, March 1997.
<b>[RFC3551]</b>	Schulzrinne, H. and S. Casner, " <a href="#">RTP Profile for Audio and Video Conferences with Minimal Control</a> ", STD 65, RFC 3551, July 2003.
<b>[RFC4734]</b>	Schulzrinne, H. and T. Taylor, " <a href="#">Definition of Events for Modem, Fax, and Text Telephony Signals</a> ", RFC 4734, December 2006.
<b>[I-D.ekr-security-considerations-for-rtc-web]</b>	<a href="#">Rescorla, E.K.</a> , "Security Considerations for RTC-Web", May 2011.
<b>[I-D.webm]</b>	Google, Inc., , "VP8 Data Format and Decoding Guide", July 2010.

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