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# RTC-Web Codec and Media Processing Requirements
draft-cbran-rtcweb-codec-00

## Abstract

This document outlines the codec and media processing requirements for RTC-Web client application and endpoint devices.

## Status of this Memo



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## Table of Contents

*1. Introduction

*2. Terminology

*3. Codec Requirements

*3.1. Audio Codec Requirements

*3.2. Video Codec Requirements

*4. RTC-Web Endpoint Device Requirements

*5. Legacy VoIP Interoperability

*6. IANA Considerations

*7. Security Considerations

*8. Acknowledgements

*9. References

*Authors' Addresses

## 1. Introduction

An integral part of the success and adoption of the Real-Time Communications Web (RTC-WEB) will be the interoperability between RTC-Web applications. This specification will focus on the media processing and codec requirements for RTC-Web client applications.
Media processing and codec requirements fit into a series of specifications have been created to address RTC-Web communications protocols, security requirements, data transmission and use cases. More information on the RTC-Web can be found here:
[TODO put links to supporting drafts here]

## 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 RTC-WEB client applications. To ensure a baseline level of interoperability between RTC-Web applications, a minimum set of required codes is specified below. While this section specifies the codecs that will be supported by all RTC-Web application implementations, it leaves the question of supporting additional codecs to the will of the implementer.

### 3.1. Audio Codec Requirements

RTC-WEB applications are REQUIRED to implement the following audio codecs.

*PCMA/PCMU - see section 4.5.14 of [RFC3551]

*Telephone-event - [RFC4734]

*Opus [draft-ietf-codec-opus]

Implementations of the PCMU and PMCA codecs are REQUIRED to support 1 channel with a rate of 8000 and a ptime of 20.
The following codecs are OPTIONAL for RTC-WEB application implementations.

*G729

*G722

*G722.1

*G723

*AMR

*AMR-WB

*iLBC

*L16

[Open Issue: minimum profile and identifying any additional mandatory to implement audio codecs.]

### 3.2. Video Codec Requirements

RTC-WEB applications are REQUIRED to implement the following video codecs.

*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

*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

The following video codecs are OPTIONAL for RTC-WEB application implementations.

*H.263

*H.264-AVC

*H.264-SVC

## 4. RTC-Web Endpoint Device Requirements

It is plausible that the dominant near-to-mid term RTC-Web usage model will be people using the RTC-Web functionality to communicate with each other via web browsers typically running within 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 RTC-Web compatible endpoints, it has been found that functionality such as automatic gain control, echo cancellation, and headset detection 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

*Echo cancellation, including acoustic echo cancelation

*Headset detection

*Auto-adjustments to gain control and echo cancelation algorithms based on if headset or internal speakers/microphone is being used

## 5. Legacy VoIP Interoperability

The codec requirements above will ensure, at a minimum, voice interoperability capabilities between RTC-Web client applications and legacy phone systems.
Video interoperability will be dependent upon the support, natively or through transcoding, of VP8 by the phone system vendors or the availability of an interoperable codec, such as H.264-AVC, from within the RTC-Web client application implementation.

## 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 particularly we would like to acknowledge, and say thanks for, work we incorporated from Harald Alvestrand.

## 9. References


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