

# Echo in WebRTC; Why?

Echo is a distortion of voice that occurs either when you receive your own voice; or something unusual sound!

*Echo is one of the biggest issues in current WebRTC implementations!*

# Why?

Echo occurs if input output audio devices are placed close together!

*In old days; usage of the “headsets” was the only “suggested” solution to overcome the echo!*

[www.WebRTC-Experiment.com](http://www.WebRTC-Experiment.com)

# Why?

*Sometimes,*

44.1 kHz and non-441.1 kHz sample rate mismatches  
causes echo!

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# Solution?

The "**ambient noise reduction**" which can be enabled on the built-in mic on Mac appears to cause a very small amount of echo.

# Solution?

Ensure both **capture** and **render** devices are set to either **44.1** or **48** kHz. You can do this through the "**Audio MIDI Setup**" application

# Solution?

Disable "**ambient noise reduction**" on Mac.

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# Solution?

Make sure there is distance between input and output audio devices.

*(Old guys suggested that) always prefer headphones over built-in speakers!*

# Solution?

Suggest users to install third party software(s) to mimic echo.

E.g. Toshiba Echo Cancellation Utility

etc.

*Acoustic Echo Cancellation (AEC) is a preferred solution!*



# Solution?

High volume can produce sound noise on some audio devices. The value may not be significant if audio device volume is controlled externally.

```
HTMLAudioElement.volume = 0.9;
```

```
HTMLAudioElement.play();
```

*Acoustic Echo is what happens when sound from your speakers enters your microphone.*

*You may not hear it, but the people you are talking to will find it extremely irritating*

# Solution?

Don't use "autoplay:true" for local streams.

Manually set "muted:false" and "volume:0":

```
HTMLVideoElement.volume = 0;
```

```
HTMLVideoElement.muted = 0;
```

# Solution?

The **GainNode** is a simple element that lets us control the volume of the audio that's coming into it.

```
var context = new webkitAudioContext(),  
var sineWave = context.createOscillator();  
var gainNode = context.createGainNode();  
sineWave.connect(gainNode);  
gainNode.connect(context.destination);  
sineWave.noteOn(0);  
gainNode.gain.value = 0.9;
```

Try [this demo](#) locally to control volume using **GainNode**.

# Solution?

*No concrete solution!*

Browser vendors landed some patches; however, still echo issues occur on Firefox, Chrome and Opera!

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# Solution?

*Patches (getUserMedia constraints API):*

googEchoCancellation

googAutoGainControl

googNoiseSuppression

googHighpassFilter

Ref: [mediaconstraintsinterface.h](#)

googNoiseReduction

googLeakyBucket

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# Acoustic Echo Cancellation (AEC)

*Acoustic Echo Cancellation (AEC) solves echo/noise issues by analyzing the sound being played out from your speakers and removes it from the sound captured by your microphone.*

Current AEC implementation in chromium uses less CPU and improves voice quality.

# AEC & mobile platforms?

*Echo Controller works with 8k/16k sampling rate however there are special processes in chromium to make Echo Controller work for iSAC codec i.e. 32k sampling audio; and opus codec i.e. 48k.*

AEC may not work for mobile platforms with complex implementations; only AECM could work which has performance issues.

AECM provides poor quality; causes very loud background noise. However, unfortunately, it seems that 40% of android devices are unable to handle AEC.

# AEC & mobile platforms?

AECM is unable to handle double talk.

AEC may fail to handle distorted input due to linear algorithms.

AECM is limited to 8k/16k sampling rate; so skip opus usage on android devices until AEC gets improved in chromium!

Should mobile users compromise?

*Hmm, unfortunately!*



# Conclusion?

- It is said that echo occurs when sound from the speakers is picked up by the audio input devices i.e. microphones. You need to place both devices away from each other.
- People usually suggest to mute participants' audio while they're idle. Remember, voice-activity-detection is "already" enabled by default on chrome. RTP packets are streamed accordingly. So, there are less chances of echo in such scenario.
- You either need to install echo cancellation software according to your platform and devices; or try to use microphone/headphone.

# References?

- [WebRTC Eco Cancellation](#)
- [WebRTC improvement optimized aecacoustic echo cancellation](#)
- [mediaconstraintsinterface.h](#)
- <https://github.com/muaz-khan/WebRTC-Experiment/issues/95#issuecomment-24562690>
- <https://code.google.com/p/webrtc/issues/detail?id=2580>

# Thanks!

**Muaz Khan**

[www.WebRTC-Experiment.com](http://www.WebRTC-Experiment.com)

[muazkh@gmail.com](mailto:muazkh@gmail.com)