WebRTC- An Abstract Overview

Priyank Lohariwal
January 31, 2021

1 Introduction

In the modern world and fast development of the internet, the connection among people is being very significant than ever, people are looking for new methods to do advance communication between them without any issue, real-time communication is one of this ways. Web Real-Time Communication (WebRTC) is a new standard that lets browsers communicate in real time using a peer-to-peer architecture. It is about secure, consent-based, audio/video peer-to-peer communication between HTML5 browsers. This is a disruptive evolution in the web applications world, since it enables, for the very first time, web developers to build real-time multimedia applications with no need for proprietary plug-ins.

2 WebRTC Architecture

WebRTC follows the semantics client-server organizer with the concept of peer-to-peer communication among the browsers as shown in Figure 1. The connection manages the media path to permit a direct flow between browsers. Network signals are transmitted during the Web Servers that help in modifying, interpreting or managing the signals, as it required by WebSockets or HTTP.

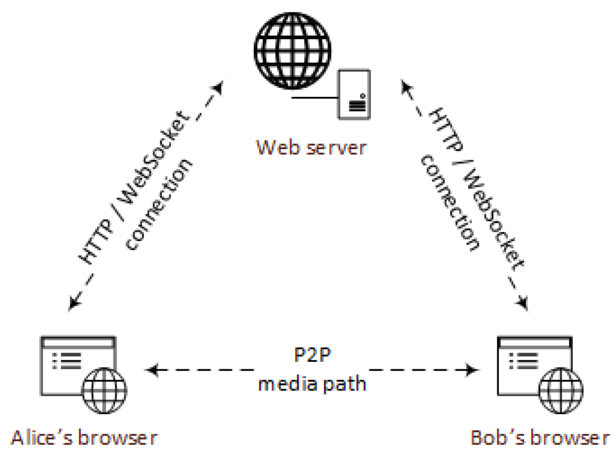

Figure 1: WebRTC Architecture

2.1 The Browser in WebRTC function

WebRTC application (usually written using HTML5, CSS, and JavaScript) is interacted with several web browser by the WebRTC API (Application Programming Interface), which allow proper use and management of the Real-time browser function. Using fundamentally WebSocket technology, signals can be exchanged between online participants. The WebRTC web application interacts with browsers that use both standards application of API and WebRTC application. Real-time imaging communication (Like a video, and audio call) among two browsers includes direct media streaming among the two browsers, within the media path parley and create an instance of multifaceted interaction through the following entities:

- The caller and caller browser are using JavaScript application (using JavaScript API).
- The JavaScript application for caller and application provider (typically a web server)
- The JavaScript application for called and caller browser (through the application browser JavaScript API).

2.2 WebRTC API

The W3C (World Wide Web Consortium) WebRTC API allows JavaScript application to take the benefit of the real-time capabilities of an uncommon browser's. The real-time browser function provides the necessary function for setting the video, audio and data channel required. Three basic concepts that been relied upon in the design of the API:

1. **MediaStream**
   MediaStream represents media streaming from a local media device like a microphone or a webcam. The web application must request user access to make and utilize a local stream through "GetUserMedia()" function.
2. **PeerConnection**
   RTCPeerConnection reads the output data from MediaStream and creates the connection between two users. To create the connection, a technique known as **signalling** is used.
3. **DataChannel**
   The RTCDataChannel API is a bi-directional data channel between two peers provides the possibility exchanging random data among them.

3 Advantages of WebRTC

- WebRTC enables the implementation of secure and high data transmission between clients as peer-to-peer connection in real-time.
- WebRTC requires no plugins, frameworks or applications. All we need is a WebRTC compatible browser.
- WebRTC helps achieving highest performance and lowest latency possible possible, owing to the direct conection between clients.

4 Challenges

- WebRTC works via UDP. Since UDP does not allow for reliable data transfer, the transfer of important documents may be vulnerable to this issue.
- WebRTC does not have any standard signaling protocol.