

Implementation of Audio/Video Conferencing Using WebRTC

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Abstract

Audio/Video conferencing system have been increasing gradually and the most important reason for this increase is Covid19 pandemic .To keep the organizations running in this hard time most of them have chosen the internet based video and audio conferencing. In this paper, we have implemented a web-RTC-based audio/video conferencing system integrated inside a web-application. Through this application, user can communicate using video and audio conferencing with only web browser.

Keywords: Web-based Applications, JavaScript, Web Framework, Socket.io, WebRTC, ExpressJs, MongoOB, NodeJs, BcryptJs, JWT

Introduction

Social media applications is being widely used around the globe and based on the user requirement different application provide different features and in this particular project we have picked the most commonly used features that is video and audio conferencing and to implement this in our project we have used one of the free conferencing library available out there. WebRTC gives us the power that is needed for our application to provide seamless video and audio calling. For developing the application we have used MERN stack, below is the graphical representation of how MERN stack works.





Frameworks and Technologies Used

In this section, we will refer the technologies used for this project. These are the modern technologies commonly used around the globe to develop high level web applications. Below are the technologies that powers this project.

• WebRTC :

Web-based real-time communication

You can add real-time communication features to your application using WebRTC, which is based on an open standard. It allows developers to create sophisticated speech and video communication systems by allowing video, voice, and generic data to be transmitted between peers. All modern browsers, as well as native clients for all major platforms, support the technology. WebRTC technologies are implemented as an open web standard and are available in all major browsers as ordinary JavaScript APIs.

What can WebRTC be used for?

WebRTC has a wide range of applications, from simple online apps that use the camera or microphone to more complicated video-calling and screen-sharing applications.

Flow of the application

In most cases, a WebRTC application will go through a standard application process.

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• NodeJS :

Node.js (or Node) is a JavaScript runtime environment. This technology can be used to develop non-browser JavaScript applications. APIs for working with files, databases, and HTTP requests, among other things, are available in the Node.js framework.

• ReactJS :

React is a Facebook-developed open source toolkit for creating user interfaces. It is non-intrusive and can be used in conjunction with any other library or framework. React creates its own virtual DOM object, which reduces access to the browser's DOM and improves performance. React introduces the JSX format for content rendering, which is a JavaScript syntactic extension that looks like XML. It is suggested, but not required, to use JSX.

• Redux :

A Predictable State Container for JS Apps. It helps developers create apps that behave consistently, run in a range of environments (client, server, and native), and are easy to test.

The Redux DevTools make it easy to see when, why, and how an application's state changed. Redux's architecture lets you to track changes, use "time-travel debugging," and even transmit complete fault reports to a server. Redux works with any UI layer and has a large ecosystem of extensions to help you fulfil your individual needs.

• Iframe :

Inline Frame is a type of frame that is used to display information The " iframe " tag designates a rectangular area within a document where the browser can show a distinct document, complete with scrollbars and borders. An inline frame is a type of HTML document that allows you to embed another document within it.

Value of attributes: It has only one value URL, which is the URL of the content that is displayed in the iframe. The following are two different sorts of URL links:

It refers to a different website. Absolute URL: It refers to a different website. It points to other files on the same web page using a relative URL.

• Material UI

Material Design, a new suite of user interface components from Google, may become a viable alternative to Bootstrap. Material Design is a cross-device design that includes a set of nice-looking UI components7. Material Design is a cohesive system for creating digital experiences that integrates philosophy, resources, and tools.



SYSTEM DESIGN AND IMPLEMENTATION



Application Architecture

When a client hits the create room button, the server receives an event and sends the connection prepared event to the client. Following the establishment of the connection prepared event, the client initiates the room by delivering the connection initiator event to the server. When the other side, or the second client, hits the join room button, an event is sent to the server, and the server sends the client initiator details obtained from client 1 to client 2, establishing a peer-to-peer connection for audio/video and generic data exchange between the two clients. When the number of clients grows, the same procedure is utilised to build a link between them all.



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CONCLUSION

This article designs and implements an online video/audio conferencing web application utilising WebRTC. The benefit of using the WebRTC library is that it improves the usability and accessibility of setting up real-time audio and video. WebRTC peer-to-peer nature may potentially result in network and infrastructure cost reductions. Iframe is used to display coding IDE in this application that helps the users to collaborate and code together making this application one destination where you can connect with different users collaborate with them and simultaneously code. This application can be used in teaching , organizational meetings , interview purpose etc.

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REFERENCES

• Zinah Nayyef Sarah Amer and Hussain "Peer to Peer Multimedia Real-Time Communication System based on WebRTC Technology" International Journal for the History of Engineering & amp; Technology vol. 2 no. 9 pp. 125-130 2019

• Akshay Kashyap, Graduate Student Member, IEEE, and Benny Bing, Senior Member, IEEE "Efficient HD Video Streaming Over the Internet" 2019

• Alam Rahmatulloh, Irfan Darmawan, Irfan Darmawan, IEEE "Performance Analysis of Data Transmission on WebSocket for Real-time Communication" 2019

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