

Video Conferencing Software Development Kit (SDK) Using WEBRTC

Jeevitha P¹, Sowmya D², K N Abhiram³, Kandiga Akshay Kumar⁴, Chethan Kumar B⁵

¹Final Year Student, ²Professor, ³Final year Student, ⁴Final year Student, ⁵Final year Student Department of Computer Science and Engineering, Jawaharlal Nehru New College of Engineering Navule, Shimoga.

Abstract - The studies offered on this paper introduces video conferencing, that is a device orientated to offer its customers with an extra communicate availability through the convergence of traditional telephony and real-time multimedia in web browsers. The device consists of Internet software that provides a video conference room for multiple customers without the need to download additional software. The prevailing task intent is to recommend a unified communications device that stands out from the rest, particularly on the side of interacting with the telephony community directly from an internet browser as well as an animated video conference that allows for a switch in real-time media flows between these technologies. The device was realized using WebRTC (Web Real Time Communications) for real-time audio and video transmission, Node.js as web server and signaling.

Key Words: WebRTC, Video Conferencing System, Real Time Communication.

1. INTRODUCTION

Video conferencing is a conversation system among or extra contributors wherein audio, video, and records indicators are transmitted electronically to allow real-time conversation. Much extra expert and powerful than audio conferencing, everybody concerned can see the face visage and gestures which might be so vital to our conversation. Communication is a totally vital a part of existence and has hence developed from natural voice conversation to video conversation. With the arrival of the Internet, real-time video conversation has ended up a reality. We want to increase a video conferencing internet utility in the use of WebRTC that lets in each voice and video conversation, it may be one-to-one or among a couple of users. Due to the browser-primarily based totally nature of the utility, it's miles device-agnostic and might run on a huge variety of devices, making it platform-agnostic. We're going to create a totally easy utility, so one can permit us to transmit audio and video to the related device: an easy

video chat utility. WebRTC permits media devices (virtual camera and microphone) to transmit audio and video among connected devices.

2. RELATED WORK

A paper "Performance Evaluation of WebRTC-Based Video Conferencing: A Comprehensive Analysis" by Shyam Sunder Saini and Lalit Sen Sharma, published in the Journal of Advanced Zoology on December 14, 2023, delves into a comprehensive assessment of WebRTC-based video conferencing systems. The research investigates various performance metrics, including video quality, audio quality, latency, packet loss, and jitter, under diverse network conditions. The findings provide valuable insights into the strengths and limitations of WebRTC technology, aiding in the optimization and deployment of robust video conferencing solutions.

A paper "Video Conference Room Implementation with WebRTC and React" by Phuc Truong, published by Metropolia University of Applied Sciences on May 6, 2021, explores the development of a video conference room application utilizing WebRTC and React.js. The paper likely delves into the technical challenges and solutions involved in implementing real-time video and audio communication, as well as the integration of React.js for a seamless user interface.

A paper "Towards Seamless Authentication for Zoom-Based Online Teaching and Meeting" by S. Mahmood, U. Ijaz Bajwa, published by arXiv on May 2020 they suggest a basis for an unbroken authentication mechanism for zoom-primarily based totally completely instructions and conferences. This approach is primarily based totally mostly on image graph reaction non-uniformity based totally completely virtual camera authentication, which could authenticate the digital virtual camera of a tool carried out in a zoom assembly without requiring the assist of the contributors (e.g. The participant providing biometric records provide). The outcomes of a small-scale check validate the proposed approach.

A paper "A Computational Model to Translate and Analyze Voices from Real-Time Video Calling" by Aneek Barman Roy, published on arXiv in 2019, presents a novel approach to real-time language translation and voice analysis within video calls. The research likely explores techniques for accurate speech recognition, machine translation, and sentiment analysis, aiming to enhance cross-language communication and understanding in virtual environments.

2.1 METHODOLOGY

Video Conferencing the use of WebRTC presents a platform in which customers can create room themselves to have created actual time communicate with character to character in a conferencing manner. The User can percentage screen, chat and document the video conference.

The preliminary components of the internet site is the net portal. Creates Room with the aid of using coming into require. Invitee thought to pass the code and input required details, then simplest invitee will input withinside the video chat room. Room Creator and Invitee each could be in video chat room as a peer-to-peer actual time communication. Each peer can use different functions like Screen Share, Chat field and Recording.

Video conferencing is a manner of verbal exchange among or more locations in which audio, video and information signs are transmitted electronically to allow simultaneous interactive verbal exchange. Far more non-public and effective than audio conferencing, every person worried can see the facial expressions and frame language which may be so vital to our communiqué. Communication is a completely crucial a part of lifestyles and has as a result developed from natural voice communicate to video communicate. With the arrival of the Internet, actual-time video verbal exchange has come to be a reality. We want to broaden a video conferencing net utility the use of WebRTC that lets in each voice and video communicate, it may be one-to-one or among a couple of customers. Due to the browser-primarily based totally nature of the utility, it's far device-agnostic and may run on a huge variety of devices, making it platform-agnostic. We're going to create a completely easy utility with a view to permit us to transmit audio and video to the linked device: an easy video chat utility. WebRTC permits media devices (virtual camera and microphone) to transmit audio and video amongst connected devices.

Detailed Workflow: -

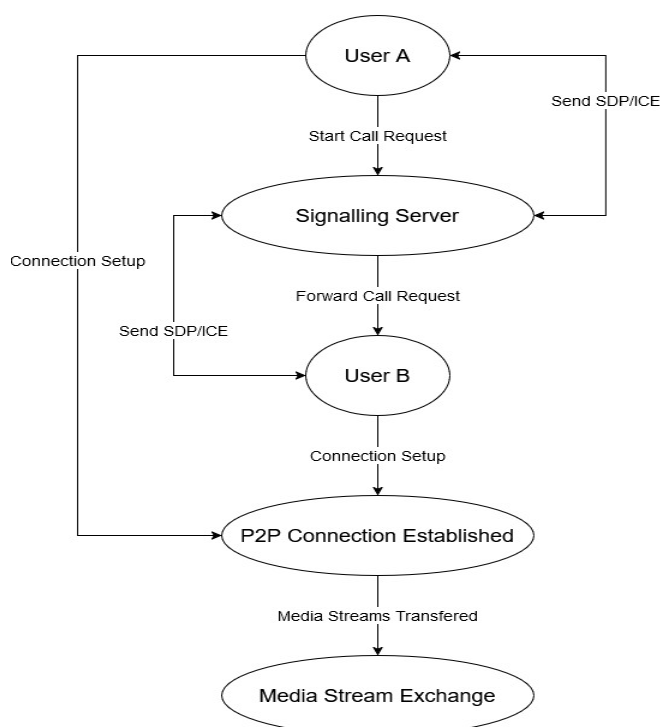


Figure 1. System Work Flow

1. User Registration and Connection: Users open the client application built with ReactJS. A WebSocket connection is established with the signaling server.
2. Initiating a Call: User A sends a request to start a call.
3. The signaling server relays this request to User B.
4. Exchange of Signaling Data: User A and User B exchange SDPs and ICE candidates via the signaling server to establish a peer-to-peer connection.
5. Media Stream Management: Users' audio and video streams are captured using the WebRTC getUserMedia API. Streams are added to the RTCPeerConnection and transmitted between peers.
6. Screen Sharing and Chat Features: Users can share their screens using the WebRTC getDisplayMedia API. Text messages are exchanged over WebRTC data channels for real-time communication.
7. Group Conferencing: The media server routes and mixes audio/video streams for all participants.

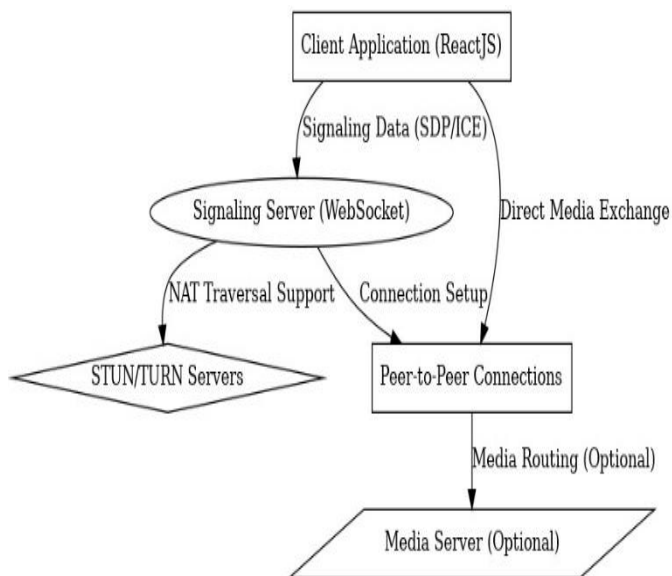


Figure 2. System Architecture

The system architecture for the video conferencing SDK using WebRTC and ReactJS is designed to ensure seamless communication, scalability, and user interaction.

The architecture consists of the following components:

- 1.Client Application: Built with ReactJS, it serves as the front-end interface where users can interact with the system. This includes functionalities like initiating calls, toggling video/audio, and screen sharing.
- 2.Signaling Server: A Node.js-based server that facilitates communication between clients. It uses WebSocket to exchange signaling messages such as session descriptions (SDPs) and ICE candidates.
- 3.STUN/TURN Servers: These servers help establish peer-to-peer connections by handling network traversal issues, especially in NAT or firewall-restricted environments.
- 4.Media Server (Optional): For group calls and scalability, a media server like Janus or Kurento is employed. It routes media streams efficiently to multiple participants.
- 5.Database (Optional): A database can be integrated to store user profiles, call logs, and other relevant metadata, enabling extended functionalities like user authentication and analysis.

3. EXPERIMENTAL RESULTS

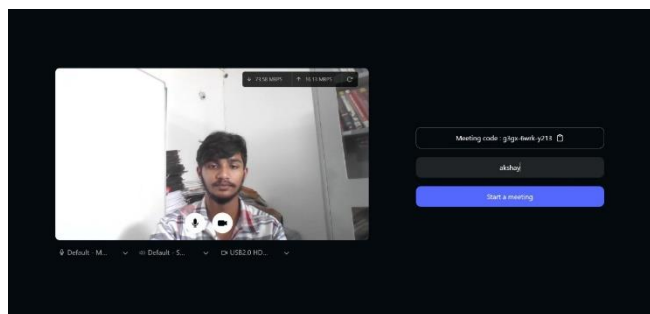


Fig -3.1: Main Page

As Shown in [Figure 3.1] Video Conferencing web application has a clean and easy interface. This justifies the idea of the project by making it clear about the functioning of the platform. This Image show procedure of making video chat room.

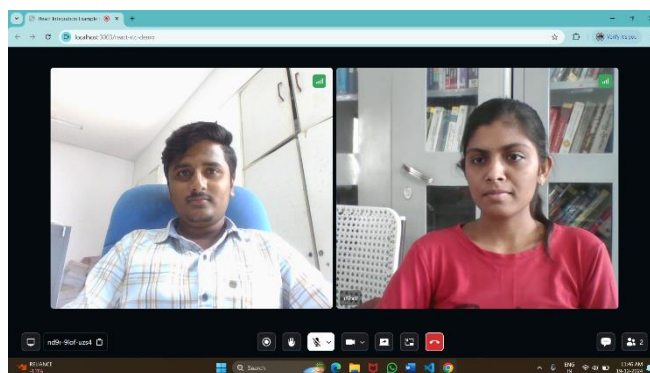


Fig -3.2: Room Page

As Shown in [Figure 3.2] A virtual space for real-time communication and collaboration.

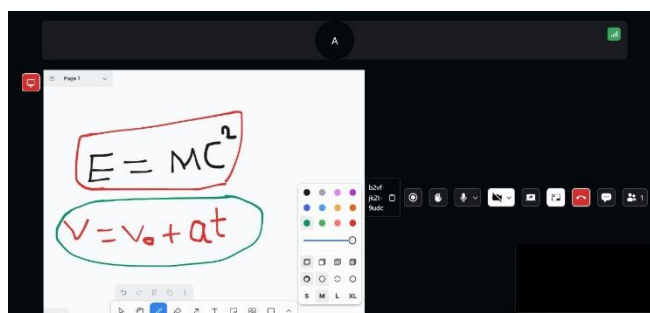


Fig-3.3: WhiteBoard

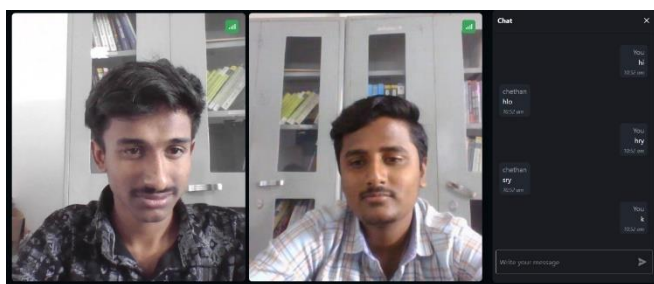


Fig -3.4: Chatbot

Page

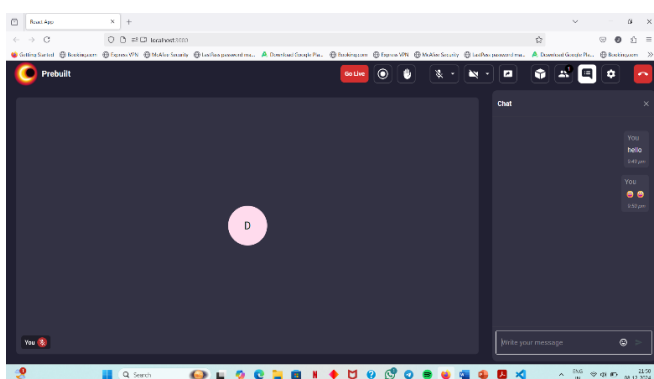


Fig -3.5: Chatroom

Page

As Shown in [Figure 3.5] A video call interface with a single participant's video stream visible.

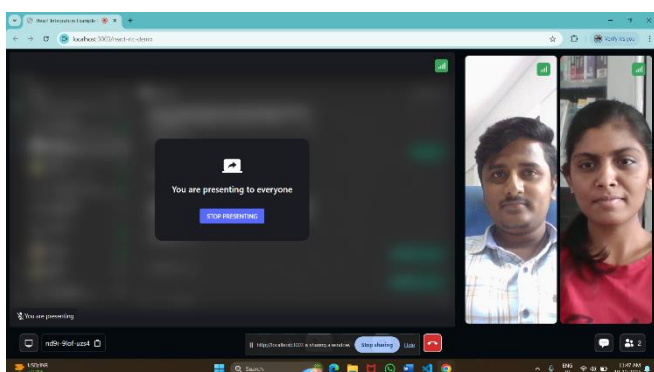


Fig -3.6: Screen Share Option

As Shown in [Figure 3.6] The image depicts a video conferencing interface where a user is presenting their screen to participants, with two video feeds of attendees visible on the side.

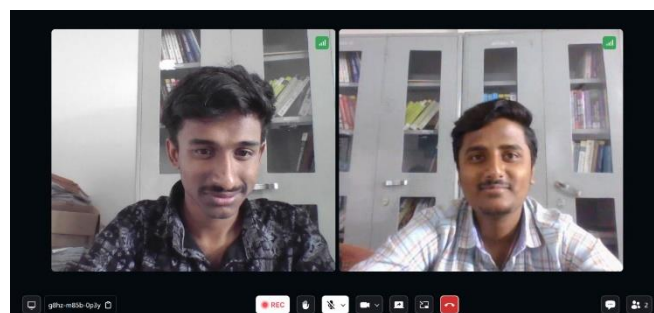


Fig-3.7: Record

Interface

As Shown in [Figure 3.7] The image shows a video conferencing session with two participants visible in separate video feeds, and the interface displays controls for recording and call management.

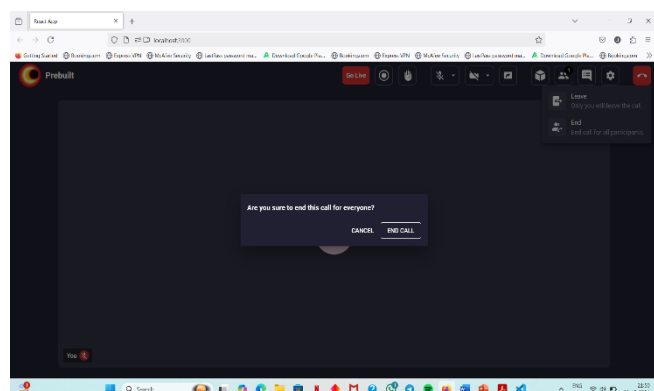


Fig -5.7: End Call

As Shown in [Figure 3.8] A user is presented with a confirmation dialog to end a call for all participants. This system is prompting the user to confirm ending a group video call

4. CONCLUSIONS

The proposed technique allows users to create a room and have real-time conversations one-on-one or in groups. It does not require a sign-up or login process, offering a dynamic, well-service-oriented, and user-friendly website. This flexible and highly efficient web application provides a time-saving tool where screen sharing, whiteboarding, video recording, and screen recording can be easily accomplished within the video chat room.

5. FUTURE SCOPE

The future scope for the above system includes integrating a TURN server to enhance peer connections. Another function that can be added is implementing a file sharing feature. As education, industry, and various sectors increasingly rely on online platforms, this feature can significantly improve effective teaching and collaboration across different fields.

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