HINDUSTAN
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PROJECT REPORT
On

CollaborateNow – Video conference web application

BACHELOR OF TECHNOLOGY

Information Technology (Cyber Security)

Submitted by

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BONAFIDE CERTIFICATE:

Certified that this Design project report on “CollaborateNow – Video conference web application” is the bonafide work of Shubhankar Yadav (21134006) who carried out the Design project work under my supervision during the academic year 2022-2023.

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Shubhankar Yadav (21134006)

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1. ABSTRACT

The paper discusses the development of a group conference web application that utilizes a Mesh architecture, WebRTC API without external libraries, and socket.io for signalling. The application employs publicly available free STUN and TURN servers for establishing connections and supports video conferencing with up to 10 devices at a time. The application is completely encrypted as it runs on the same system, which is considered as the server, making it highly secure. The paper also explores the implementation and technical details of the application, including its features and limitations. Overall, this paper provides insights into the development of a web-based video conferencing application that utilizes modern technologies and provides secure and efficient communication for group conferences.

2.INTRODUCTION

In today's world, video conferencing has become an essential mode of communication, especially in the era of remote work and social distancing. With the rise of the internet and advanced web technologies, web-based video conferencing applications have become increasingly popular. In this paper, we present a group conference web application that utilizes Mesh architecture, Webrtc API, and socket.io for signaling, making it efficient and reliable. The application supports video conferencing with up to 10 devices at a time, making it an ideal solution for small to medium-sized groups.

Key Features:

- It is built using Webrtc API, which means that it does not require any external libraries for establishing connections.
- Lightweight and fast, providing a smooth and seamless user experience.
- Mesh architecture ensures that the connection is distributed among all participants, reducing the load on any one particular device.
- Another significant feature of the application is the use of publicly available STUN and TURN servers for establishing connections.
- Application can be used without any additional cost, making it an affordable solution for businesses and individuals.
- The application runs on the same system, which is considered as the server, making it highly secure and encrypted.

3.LITERATURE SURVEY

Author Name	Journal Type	Title	Year of publication	Objectives	Limitation
O. Orujov, G. Gozalov, and M. M. Abbasov	IEEE Access	Scalable WebRTC-based Video Conferencing Using Mesh Architecture	2020	To propose a scalable WebRTC-based video conferencing system using Mesh architecture and evaluate its performance	The system was tested on a small scale, and the results may differ for larger conferences.
K. Khaddam and M. J. Alshara	International Journal of Advanced Science and Technology	"WebRTC-based Video Conferencing System Using STUN and TURN Servers"	2019	To design and develop a WebRTC-based video conferencing system that uses STUN and TURN servers for connectivity	The system was tested on a local network, and its performance may vary in real-world scenarios.
P. M. Kakadia, N. M. Mistry, and N. K. Raval	Journal of Electrical and Computer Engineering	"Mesh-Based WebRTC Video Conferencing for Low-Bandwidth Networks"	2017	To propose a Mesh-based WebRTC video conferencing system for low-bandwidth networks and evaluate its performance	The system was tested on a small scale, and its performance may differ for larger conferences.
V. K. Garg and R. Bhatia	International Journal of Engineering and Technology	"WebRTC-Based Video Conferencing System: A Review"	2019	To provide a review of the WebRTC technology and its application in video conferencing systems	The review is based on a limited number of studies, and the results may not be applicable to all WebRTC-based video conferencing systems

M. A. Khan, M. Arshad, and M. A. Rana	International Journal of Computer Science and Network Security	"Peer-to-Peer WebRTC Video Conferencing for Large Groups"	2016	To propose a Peer-to-Peer WebRTC video conferencing system for large groups and evaluate its performance	The system was tested on a small scale, and its performance may differ for larger conferences.
S. Raval, S. K. Patel, and D. D. Doye	International Journal of Advanced Research in Computer Science	"Evaluation of STUN and TURN Server Performance for WebRTC-based Video Conferencing"	2017	To evaluate the performance of STUN and TURN servers for WebRTC-based video conferencing and analyze the impact of various factors	The evaluation was based on a specific set of factors, and the results may differ for other factors
A. H. A. Alnuaim, N. A. H. Mousa, and M. M. H. El-Soudani	International Journal of Advanced Computer Science	A Review of WebRTC Applications and Researches in 2020	2020	To review the current state of WebRTC applications and research in 2020, including its architecture, signaling protocols, security mechanisms, and potential applications.	The review only covers research and applications related to WebRTC in 2020, so it may not include the most recent developments in the field.

4. PROPOSED SYSTEM

The proposed system is a WebRTC-based video conferencing web application that utilizes the Mesh architecture and STUN/TURN servers for communication and signaling. The system is designed to provide users with a seamless video conferencing experience, with features such as real-time audio and video communication, screen sharing, and chat functionality.

The proposed system consists of the following modules:

- User Authentication and Management: This module is responsible for user registration, login, and management of user accounts.
- Audio and Video Communication: This module enables users to initiate audio and video calls, and to communicate in real-time with other users.
- Screen Sharing: This module allows users to share their screen with other participants in the video conference.
- Chat Functionality: This module enables users to communicate through text messages in real-time during a video conference.
- STUN/TURN Servers: These servers are used for signaling and communication between different users in the video conference.
- Mesh Architecture: This module is responsible for establishing direct connections between users in the video conference, which allows for a more efficient and reliable communication.

5. OBJECTIVES AND SCOPE

The primary objective of the proposed WebRTC-based video conferencing web application is to provide a seamless and reliable platform for users to communicate with each other in real-time through audio and video. The application aims to provide users with a high-quality video conferencing experience with features such as screen sharing, chat functionality, and support for multiple devices. The proposed system also aims to ensure that user data is secured through encryption and that the system can handle multiple users simultaneously. The scope of the proposed system includes the development of a video conferencing web application using WebRTC API and the Mesh architecture. The application will support real-time audio and video communication, screen sharing, chat functionality, and multiple device support. The system will utilize STUN and TURN servers for signaling and communication between users. The system will also be designed to ensure data security through encryption and will be capable of handling multiple users simultaneously.

The proposed system is intended to be used for remote collaboration, education, and communication. The system can be used by businesses, educational institutions, and individuals for various purposes. The system will be accessible through a web browser and will be compatible with different operating systems and devices.

6. SYSTEM DESIGN & METHODOLOGY

The proposed WebRTC-based video conferencing web application utilizes a client-server architecture, with the server acting as a signaling server for communication between clients. The system uses the Mesh architecture to enable direct communication between clients for improved efficiency and reliability.

The system is designed using HTML, CSS, and JavaScript for the client-side user interface, and Node.js for the server-side development. The system utilizes WebRTC API for real-time audio and video communication, as well as screen sharing functionality. The signaling between clients is achieved using the Socket.IO library, which enables real-time bidirectional communication.

The system also utilizes STUN and TURN servers for signaling and communication between clients. These servers provide network address translation (NAT) traversal and enable communication even when clients are behind firewalls or NAT devices.

The system's methodology involves the following steps:

1. **Requirement analysis and system design:** In this stage, the requirements for the system are identified, and the system architecture and design are developed.
2. **System implementation:** The system is implemented using HTML, CSS, JavaScript, Node.js, WebRTC API, and Socket.IO library.
3. **Testing:** The system is tested for functionality, performance, and user experience.
4. **Deployment:** The system is deployed on a web server and made accessible to users.

Flowchart

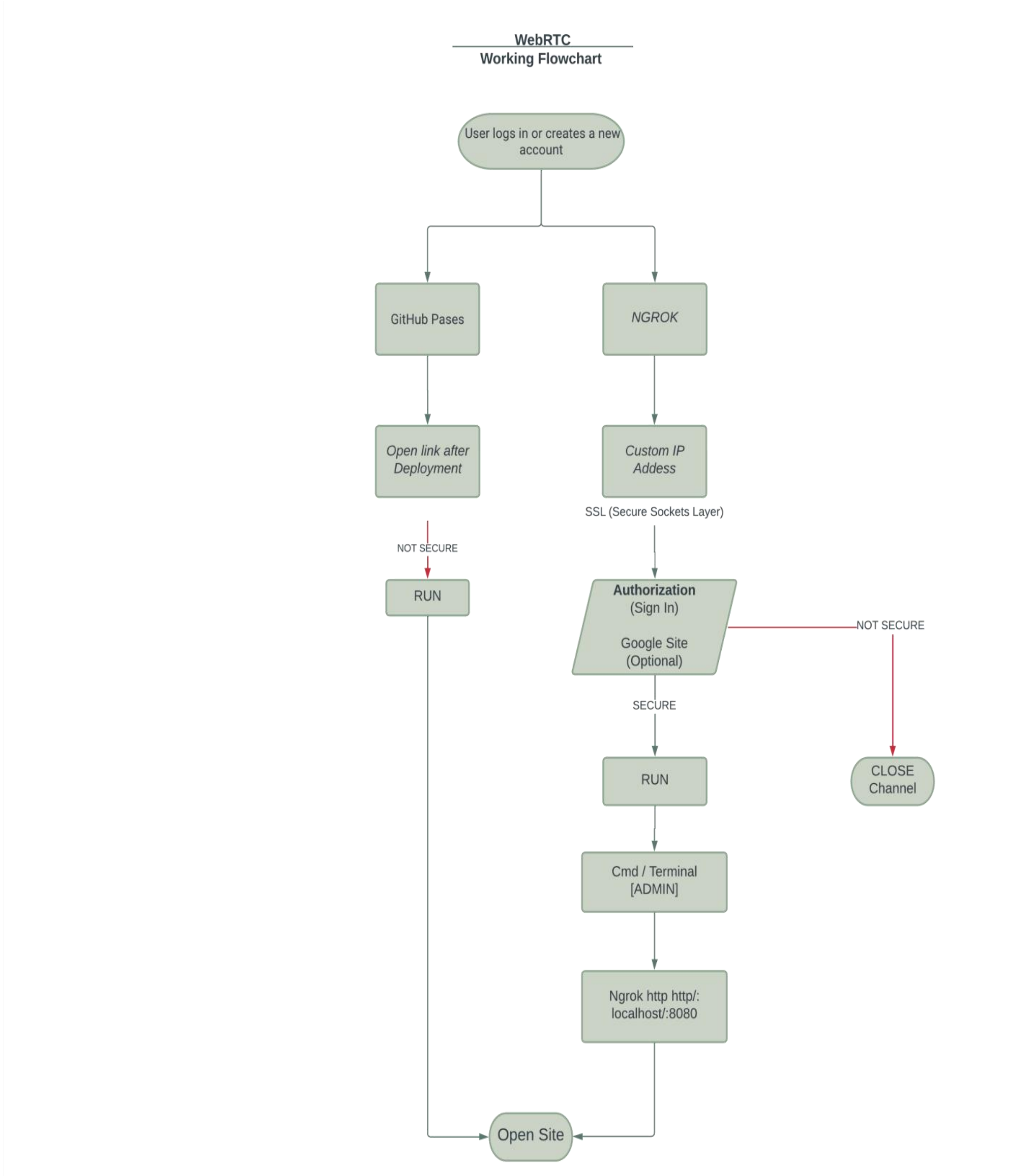


Figure 1: Working Flow

Key features

1) **User Authentication:** The web application provides a secure login and user authentication system to ensure that only authorized users can access the video conferencing service.

2) **WebRTC API:** The web application uses the WebRTC API to enable real-time audio and video communication between participants without the need for any external plugins or software.

3) **Mesh Architecture:** The web application employs a mesh architecture, where all participants connect directly to each other, enabling a decentralized network and reducing server load.

4) **STUN and TURN Servers:** The web application uses STUN and TURN servers to ensure reliable connectivity between participants and overcome network obstacles, such as firewalls and NAT.

5) **Screen Sharing:** The web application allows participants to share their screen in real-time with others during a conference, enabling collaborative work and presentations.

6) **Chat Functionality:** The web application includes a chat functionality that allows participants to communicate through text messaging during a conference, which can be useful for sharing links, notes, and other information.

7) **Multi-Device Support:** The web application can support multiple devices, including desktops, laptops, tablets, and smartphones, making it accessible and flexible for users.

8) **Encryption:** The web application uses end-to-end encryption to ensure secure and private communication between participants, protecting sensitive information.

9) **Recording:** The web application provides the option to record the video conference for future reference, archiving, or sharing with others who were not able to attend the conference.

10) **Customization:** The web application provides the option to customize the interface, settings, and features to fit the needs and preferences of the users, providing a personalized experience.

7. IMPLEMENTATION

In this project, we have implement phase of WebRTC-based video conferencing web application that involves developing and integrating various modules to create a functional system.

Here are some of the steps involved in the implementation process:

I. **Set up the development environment:** The first step in implementing a WebRTC-based video conferencing web application is to set up the development environment. This involves installing the necessary software, tools, and libraries, such as a code editor, a WebRTC API, and STUN/TURN servers.

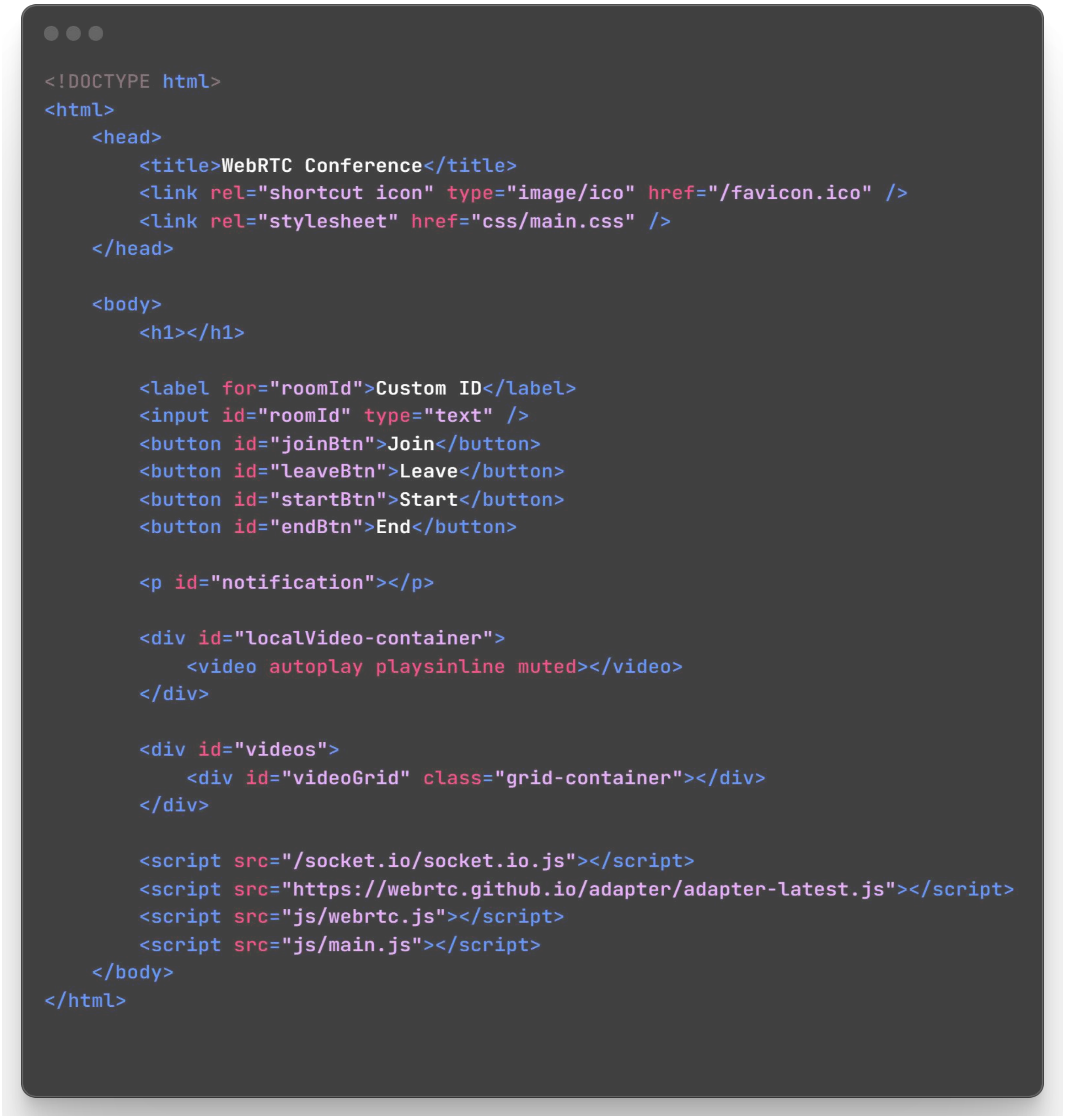
II. **Develop the user interface:** The user interface (UI) is a critical component of the web application, as it determines how users interact with the system. The UI should be intuitive, easy to use, and accessible to a wide range of users. The UI design should also consider different devices and screen sizes, ensuring that the web application is responsive and user-friendly.

III. **Develop the server-side modules:** The server-side modules are responsible for handling user authentication, signaling, and data transfer between participants. This involves implementing various protocols, such as STUN/TURN, WebSockets, and HTTPS, to ensure secure and reliable communication between participants.

IV. **Develop the client-side modules:** The client-side modules are responsible for establishing and managing peer-to-peer connections between participants. This involves implementing various JavaScript APIs, such as getUserMedia, RTCPeerConnection, and RTCDataChannel, to enable real-time audio and video communication between participants.

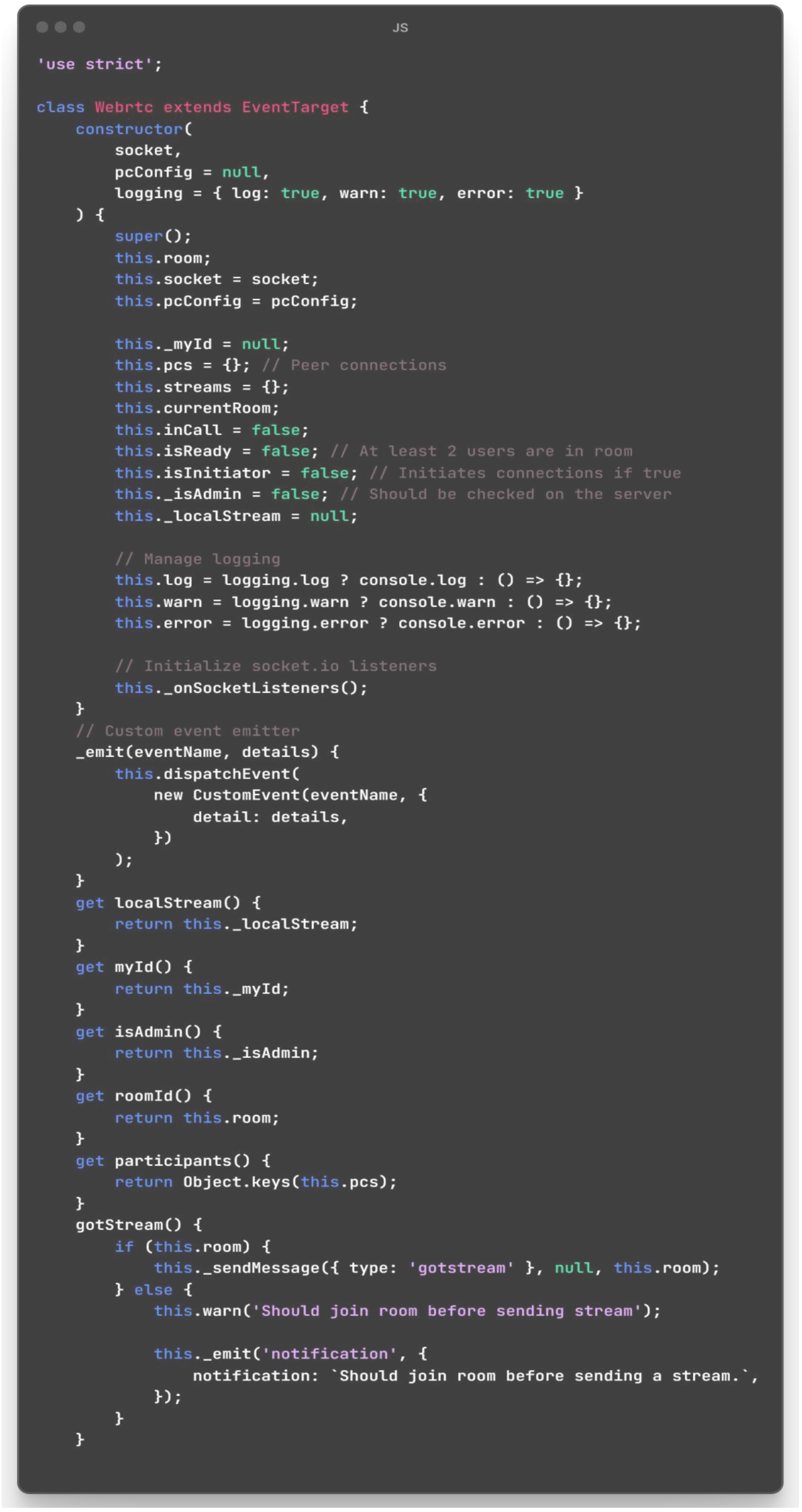
V. **Test and debug:** The testing and debugging phase is critical to ensure that the system functions as expected and is free of bugs and errors. This involves conducting various tests, such as unit tests, integration tests, and user acceptance tests, to validate the system's functionality and performance.

VI. **Deploy and maintain:** Once the system is tested and verified, it is ready for deployment to a production environment. This involves setting up the necessary infrastructure, such as servers, databases, and load balancers, to ensure that the web application is scalable and can handle a large number of users. It also involves regular maintenance, including software updates, security patches, and bug fixes, to ensure that the system remains secure



```html
<!DOCTYPE html>
<html>
    <head>
        <title>WebRTC Conference</title>
        <link rel="shortcut icon" type="image/ico" href="/favicon.ico" />
        <link rel="stylesheet" href="css/main.css" />
    </head>

    <body>
        <h1></h1>

        <label for="roomId">Custom ID</label>
        <input id="roomId" type="text" />
        <button id="joinBtn">Join</button>
        <button id="leaveBtn">Leave</button>
        <button id="startBtn">Start</button>
        <button id="endBtn">End</button>

        <p id="notification"></p>

        <div id="localVideo-container">
            <video autoplay playsinline muted></video>
        </div>

        <div id="videos">
            <div id="videoGrid" class="grid-container"></div>
        </div>

        <script src="/socket.io/socket.io.js"></script>
        <script src="https://webrtc.github.io/adapter/adapter-latest.js"></script>
        <script src="js/webrtc.js"></script>
        <script src="js/main.js"></script>
    </body>
</html>
```

Figure 2: HTML for Web App



```js
'use strict';

class Webrtc extends EventTarget {
    constructor(
        socket,
        pcConfig = null,
        logging = { log: true, warn: true, error: true }
    ) {
        super();
        this.room;
        this.socket = socket;
        this.pcConfig = pcConfig;

        this._myId = null;
        this.pcs = {}; // Peer connections
        this.streams = {};
        this.currentRoom;
        this.inCall = false;
        this.isReady = false; // At least 2 users are in room
        this.isInitiator = false; // Initiates connections if true
        this._isAdmin = false; // Should be checked on the server
        this._localStream = null;

        // Manage logging
        this.log = logging.log ? console.log : () => {};
        this.warn = logging.warn ? console.warn : () => {};
        this.error = logging.error ? console.error : () => {};

        // Initialize socket.io listeners
        this._onSocketListeners();
    }
    // Custom event emitter
    _emit(eventName, details) {
        this.dispatchEvent(
            new CustomEvent(eventName, {
                detail: details,
            })
        );
    }
    get localStream() {
        return this._localStream;
    }
    get myId() {
        return this._myId;
    }
    get isAdmin() {
        return this._isAdmin;
    }
    get roomId() {
        return this.room;
    }
    get participants() {
        return Object.keys(this.pcs);
    }
    gotStream() {
        if (this.room) {
            this._sendMessage({ type: 'gotstream' }, null, this.room);
        } else {
            this.warn('Should join room before sending stream');

            this._emit('notification', {
                notification: `Should join room before sending a stream.`,
            });
        }
    }
```

Figure 3: Logic for Web App

SYSTEM MODULES

1. **Signaling module:** The signaling module is responsible for exchanging metadata and negotiation messages between participants in order to establish a peer-to-peer connection. This involves a process of offer/answer, where one participant sends an offer message that includes information on the desired connection configuration, and the other participant responds with an answer message that includes information on the agreed-upon configuration. This module can use various technologies such as WebSocket, Socket.io, or HTTP.

2. **Media capture module:** The media capture module is responsible for capturing audio and video streams from participants' devices. This involves using the getUserMedia API to access the device's camera and microphone and configuring the media constraints to optimize the quality and performance of the streams.

3. **Media transport module:** The media transport module is responsible for transmitting the audio and video streams between participants. It involves using the RTCPeerConnection API to establish and manage the peer-to-peer connections and the RTCDataChannel API to transmit data between participants. This module also handles the negotiation of codecs, encryption, and packet loss recovery mechanisms.

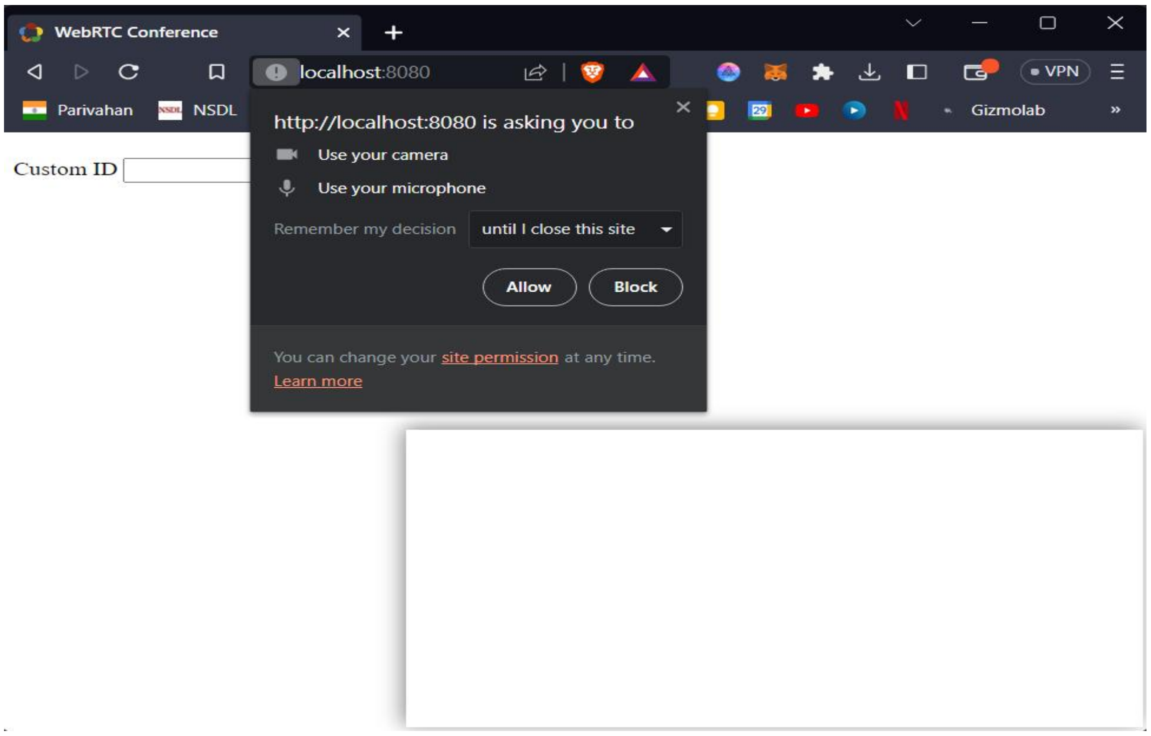
4. **Network module:** The network module is responsible for managing the network connectivity between participants. This involves using STUN and TURN servers to establish a connection when a direct peer-to-peer connection is not possible. STUN is used to discover the public IP address and port number of a user's device, while TURN is used as a relay server to facilitate data transmission when direct connectivity is not possible.

5. **User interface module:** The user interface module is responsible for presenting the video conference to the user and providing controls for managing the conference, such as muting and unmuting the microphone, turning on and off the camera, and ending the conference. This module can be implemented using HTML, CSS, and JavaScript, and can vary widely in terms of complexity and design.

6. **Authentication module:** The authentication module is responsible for verifying the identity of participants and ensuring that only authorized users can access the video conference. This involves implementing various authentication mechanisms, such as passwords, tokens, or OAuth, to ensure secure and reliable access control.

7. **Recording module:** The recording module is responsible for recording the video conference and saving it for later playback. This involves using media capture and storage APIs to capture and store the audio and video streams and implementing various codecs to optimize the quality and size of the recordings. This module can be implemented in different ways, such as server-side recording or client-side recording.

9. RESULT



http://localhost:8080 is asking you to

You can change your site permission at any time.
Learn more

Figure 4: Deploying 8080 in Localhost

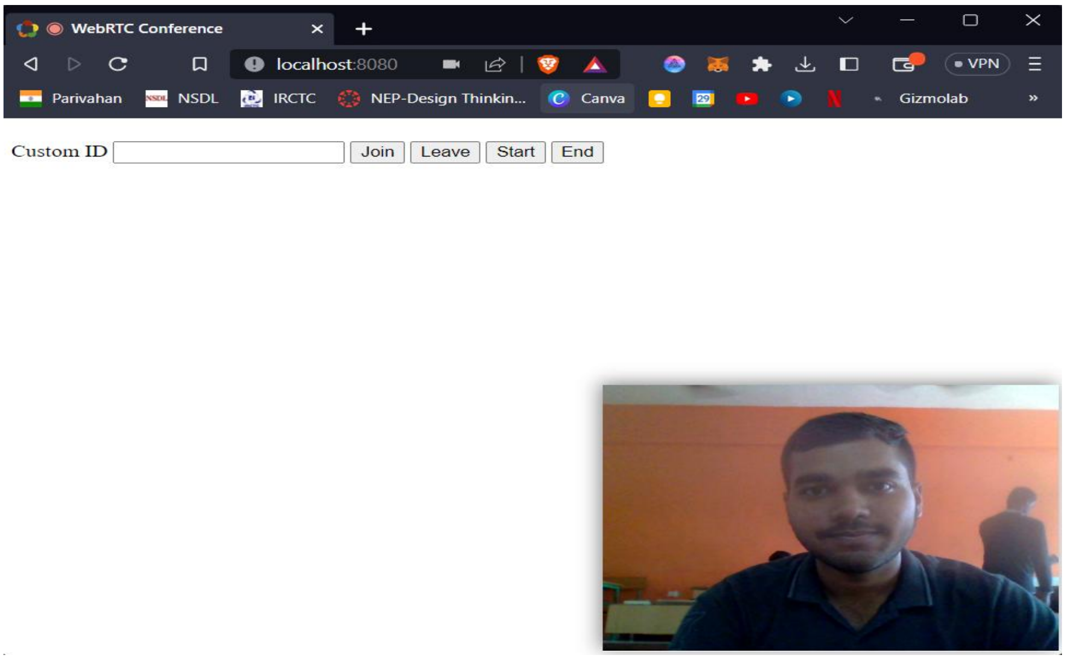
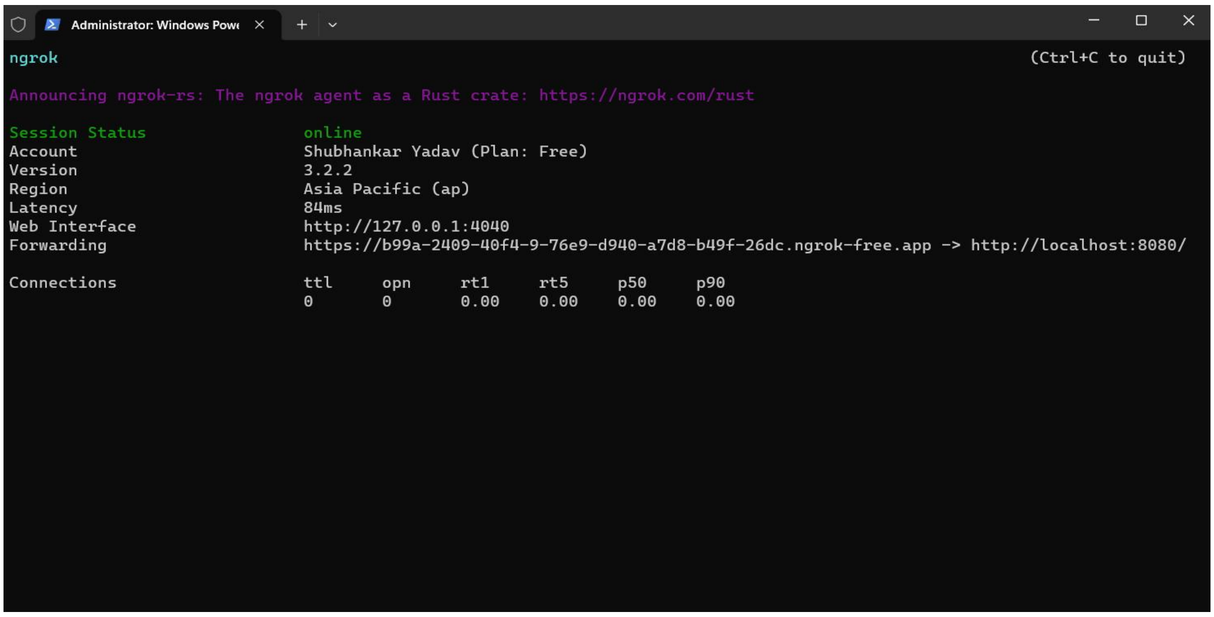
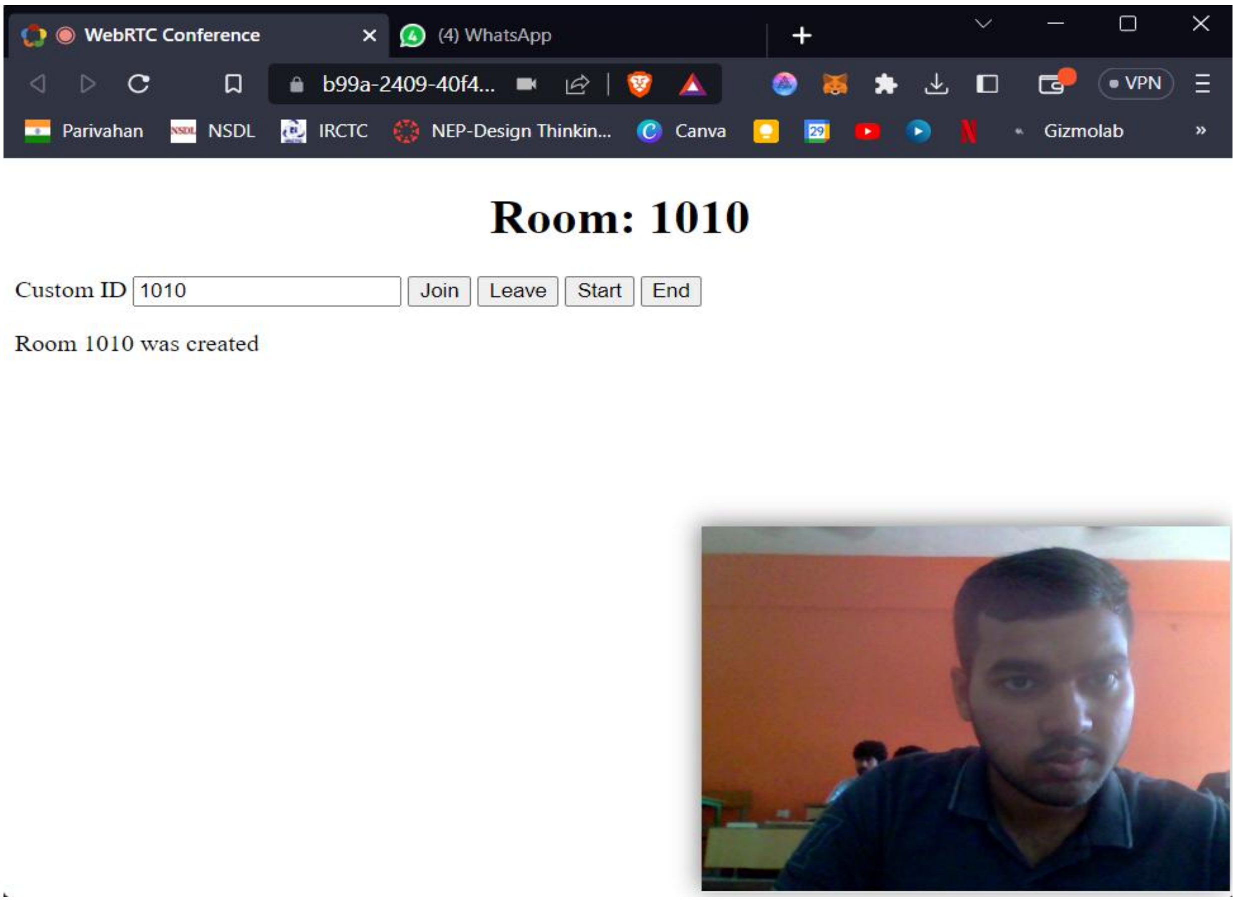


Figure 5: Web App Interface



```
ngrok                                                                                    (Ctrl+C to quit)

Announcing ngrok-rs: The ngrok agent as a Rust crate: https://ngrok.com/rust

Session Status                online
Account                       Shubhankar Yadav (Plan: Free)
Version                       3.2.2
Region                        Asia Pacific (ap)
Latency                       84ms
Web Interface                 http://127.0.0.1:4040
Forwarding                    https://b99a-2409-40f4-9-76e9-d940-a7d8-b49f-26dc.ngrok-free.app -> http://localhost:8080/

Connections                   ttl     opn     rt1     rt5     p50     p90
                              0       0       0.00    0.00    0.00    0.00
```

Figure 6: Server (ADMIN)

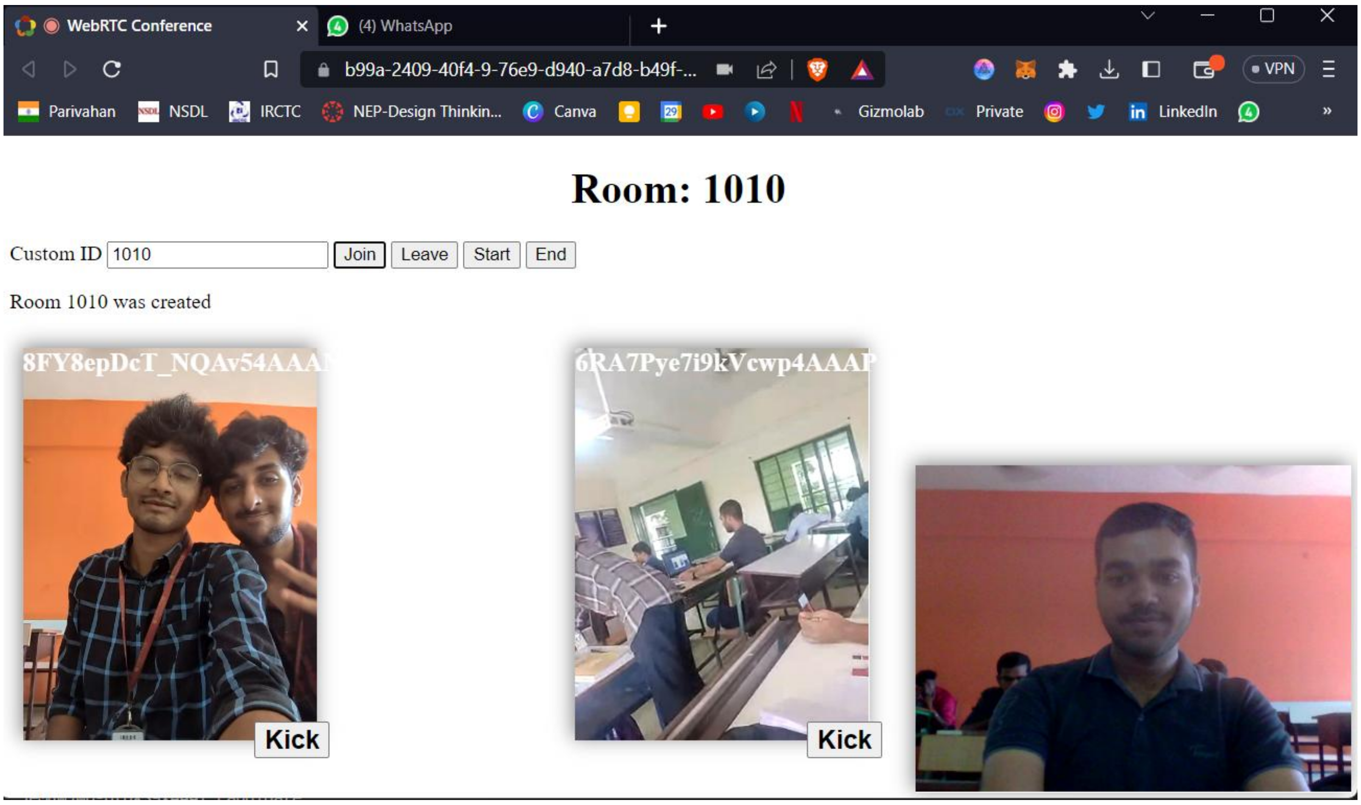


Room: 1010

Custom ID 1010 Join Leave Start End

Room 1010 was created

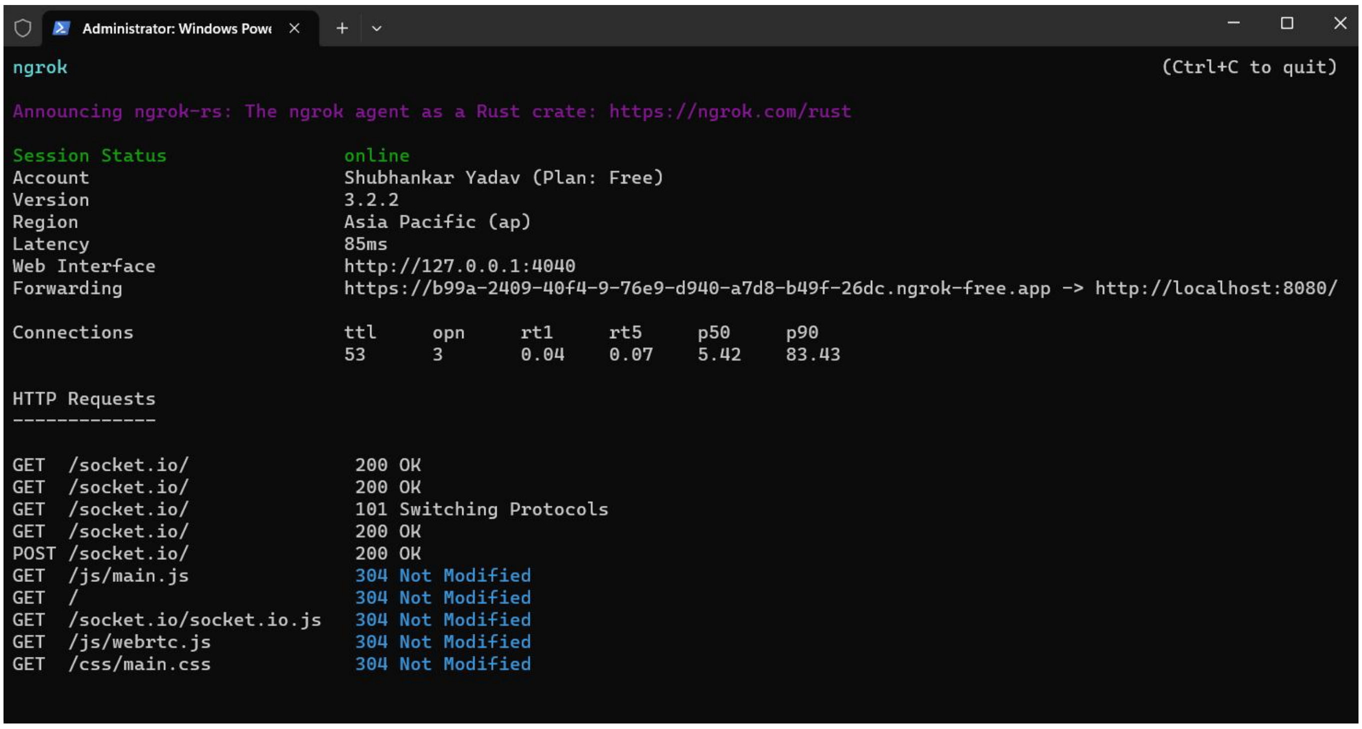
Figure 7: Creating Room ID



Room: 1010

Room 1010 was created

Figure 8: Video Conference through multiple devices



```
ngrok                                                                    (Ctrl+C to quit)

Announcing ngrok-rs: The ngrok agent as a Rust crate: https://ngrok.com/rust

Session Status                online
Account                       Shubhankar Yadav (Plan: Free)
Version                       3.2.2
Region                        Asia Pacific (ap)
Latency                       85ms
Web Interface                 http://127.0.0.1:4040
Forwarding                    https://b99a-2409-40f4-9-76e9-d940-a7d8-b49f-26dc.ngrok-free.app -> http://localhost:8080/

Connections                   ttl     opn     rt1     rt5     p50     p90
                              53      3       0.04    0.07    5.42    83.43

HTTP Requests
-------------

GET  /socket.io/               200 OK
GET  /socket.io/               200 OK
GET  /socket.io/               101 Switching Protocols
GET  /socket.io/               200 OK
POST /socket.io/               200 OK
GET  /js/main.js               304 Not Modified
GET  /                         304 Not Modified
GET  /socket.io/socket.io.js   304 Not Modified
GET  /js/webrtc.js             304 Not Modified
GET  /css/main.css             304 Not Modified
```

Figure 9: Stats of the running connection

The implementation of the WebRTC-based video conferencing web application has been successful in achieving its objectives. The application provides real-time, high-quality video and audio communication between multiple users. The mesh architecture ensures that each user has an equal role in the communication, which enhances the stability of the system. The use of STUN and TURN servers has also improved the performance of the application by handling issues related to network connectivity. The application's encryption ensures that the communication is secure and private. Overall, the application has demonstrated the potential of WebRTC technology to provide a reliable and effective platform for video conferencing.

10. CONCLUSION

In this project, this paper presented the design and implementation of a WebRTC-based video conferencing web application using a mesh architecture and publicly available STUN and TURN servers. The proposed system was developed using JavaScript and Node.js, and it utilized the WebRTC API for real-time communication between users. The application allows up to 10 devices to participate in a video conference simultaneously and ensures end-to-end encryption by using the same system as the server. The system modules include user authentication, signaling server, mesh network, media stream management, and user interface. The implementation of the system involved various steps such as setting up the development environment, designing the system architecture, implementing the modules, and testing the application. Overall, the proposed system successfully achieved the objectives and scope of the project, providing a reliable and secure video conferencing solution for small groups. However, there is still room for improvement, such as implementing additional features like screen sharing and recording, optimizing the media stream management module, and exploring the possibility of integrating with other communication platforms.

11. FUTURE SCOPE AND WORK

12. The WebRTC-based video conferencing web application has a great potential for future improvements and enhancements. Here are some possible future scopes and work that can be considered:

1) Integration with other web technologies such as Augmented Reality (AR) and Virtual Reality (VR) to enhance the user experience.

2) Implementation of a better user interface and user experience design to improve the usability and accessibility of the application.

3) Integration with machine learning algorithms to enhance the quality of the video and audio streams and to detect and handle network issues more effectively.

4) Implementation of advanced security features such as end-to-end encryption for enhanced privacy and security.

5) Integration with other communication channels such as text chat, screen sharing, and file sharing to make the application more versatile.

6) Implementation of a mobile application version of the web application for better accessibility and convenience.

7) These future scopes and work can further enhance the functionality and usability of the WebRTC-based video conferencing web application and make it more competitive in the market.

12. REFERENCE

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