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Online Live Teaching Using WebRTC Protocol and Live Sharing Notes

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Abstract: This paper describes the Web Real Time Communication (WebRTC) is a peer-to-peer technology that provides web browsers with Real-Time Communications (RTC) capabilities. The WebRTC has been built within almost all modern browsers using which the browsers are able to talk to each other instead of web servers. With this peer to peer (P2P) technology as the foundation to the web application, the system is able to provide diverse learning capabilities and the learner can discover the expert to establish a direct connection. This teaching system brings out an open standard-based, user-friendly learning platform that allows multimedia streams such as audio and video, screen sharing, Whiteboard, noise cancellation, invite mail, analytics and reporting, attendance tracking teaching collaboration at any time with participants at anywhere to actively interact with the experts for the clear understanding of a particular concept. In this paper, we have proposed a WebSocket, Socket.IO and Peer JS to facilitate with the WebRTC in an easy and effective manner. This will not only effectively drive the development of network education, but also powerfully promote the popularization of continuing education and on-the-job training.

Index Terms - Web Real-Time Communication (RTC); Peer to Peer (P2P); WebSocket; Socket.IO; Peer JS.

I. INTRODUCTION

In recent education times, the online learning is certainly the most effective informative development. The advancements in the field of computer technology and the internet in network education has wide acceptance in the world. The internet has combined with education can able to make teaching and learning activities is very simple and while teachers teach over the internet then the students also study in the internet, the information flows and the knowledge is gained. Eventually, the offline activities will become the supplement of online activities. Nowadays, the students need not be constrained by the physical classroom they can use online learning tools to find material when and where best suites them.

But today's network education systems are primarily implemented by using document-based or web page-based User Interfaces (UI) so the only audio and video network education choices currently available to both teachers and students for the proprietary systems. However, those systems are required an additional plug-in such as the most commonly used plug-in as Adobe Flash or the system need a completely standalone application to be installed such as Skype, both are inconvenient for users and developers and also the setting time to those systems is quite high, even some of them required a registration fee.

The WebRTC happens to be a very good solution to these defects. It helps to combine different services such as presence of audio/video conferencing, instant messaging, and it does not require any installation of plug-in or to setup the software. It reduces the workload for the developers about system to deliver lecturers to the students to help them to take up online courses, interact with remote experts in an easy and effective manner so the Online live teaching using the WebRTC protocol and live sharing notes can provide an interactive and engaging learning experience for both teachers and students. WebRTC is a peer-to-peer open-source framework that is considered as a collection of standards, protocols and JavaScript. Also, it is supported by Opera, Mozilla Firefox and Google Chrome and to establish communication among various users and/or devices, WebRTC requires a kind of signalling mechanism or a support of protocols.

II. OVERVIEW

The WebRTC is to establish a peer-to-peer connectivity to other web browsers easily. By connecting via peer to peer the server will not be loaded and the users will not face any issues during conference. The traditional way of building such application from starting requires a number of frameworks and libraries that deal with the typical issues like packet loss and connection dropping. With Web Real Time Communication (WebRTC), all of this comes built-in into the browser by default and there is no need for plug-ins or the third-party libraries. It is open-sourced which simplifies development and it can be used for multiple tasks but real time peer-to-peer media streams such as audio and video communication is the primary benefit. In order to communicate with another person through web browser, each person's web browser must agree to begin the communication and to know how to locate one another, bypass the security and firewall protection, and transmit all multimedia communications in the real time. Once the user allows to use those devices then the WebRTC can create individual streams of the transmittable audio and video data from the data generated by the input devices. WebRTC promises to provide secured direct Peer to Peer (P2P) communication between users and free of plug-ins. The WebRTC assures a simplified, flexible, and cost-effective of real-time communication for users without dependence on service providers. One of the critical challenge faces with the plug-in as Flash, Silverlight, and Shockwave is the need for downloads each time when a connection is to be established. The Plug-ins can be problematic during execution; they increase bandwidth, latency, execution time, and speed.

III. DEFINITIONS

A. WebSocket (WS)

The WebSocket is a communication protocol based on the web and it allows for real time, bi directional communication between a web client and web server. WebSockets are commonly used for tasks such as real-time notifications, chat applications, and interactive web applications and the WebSocket enables persistent connection for efficient communication to the users.

B. Socket.IO

The Socket.IO is a JavaScript library that enables real-time, bidirectional communication between web clients and servers. It is often used in conjunction with WebRTC to facilitate signalling and other communication tasks in real-time communication applications. Socket.IO can be used for signalling by allowing clients to exchange messages with a signalling server.

C. Peer JS

Peer JS is a JavaScript library to communication from one client to other client. Peer JS can be compatible with all modern web browsers that support WebRTC and each peer has a unique ID that identifies it within the network. Peer JS also simplifies the process of setting up peer-to-peer connections in web applications, making it easier for developers to implement features like real-time messaging, file sharing, and video/audio calls.

D. Peer to peer (P2P)

The Peer to Peer is one of the decentralized communication model where each party can have the same capabilities and either the party can initiate the a communication session. The Peers can communicate with each other directly, without the need for intermediaries so that this allows for efficient data.

IV. OBJECTIVE

The main objective of online live teaching using the WebRTC protocol and live sharing notes is for the effective communication for both the teachers and the students, fostering the real time communication. Since WebRTC is a peer-to-peer technology so there is no need for handling of a centralized server due to decentralized communication among peers. The peer-to-peer learning system can help the students to acquire the necessary knowledge by consulting with the experts by making use of the WebRTC. It Provide accessibility to education regardless of geographical location, allowing students to attend classes from anywhere with an internet connection.

This online live leaving using WebRTC protocol and live sharing notes can facilitates the dynamic interactions such as file sharing, sharing of lecture notes, presentations, and other educational materials in real-time, allowing students to follow along and refer to course materials as the lesson progresses for the effective communication between various participants during conference. It also utilizes the interactive features such as screen sharing, whiteboarding, chat functionalities, attendance tracking to enhance student engagement and participation during live teaching sessions in the classroom. It provides a flexible learning as it can offer the learning schedules by recording live teaching sessions for the later playback, allowing students to review material at their own pace and convenience.

V. EXISTING SYSTEM

To implement a WebRTC protocol different developers had attempted to create or develop a signaling mechanism but most of the developers faced some issues. In this, the signaling mechanism has not yet specified by the Web Real Time Communication (WebRTC) so in order to allow the different developers to modify, reuse the existing protocols and permitted them to design the own signaling mechanism.

The existing system is based on the Multipoint Control Unit (MCU) and it is commonly used device to bridge the videoconferencing connections between the users and also it is a type of video conferencing hardware, used to connect with multiple video conferencing points simultaneously. However, this does not discuss any kind of signaling and it can ran the application using the Licode - Erizo over LAN (Local Area Network). This would not be possible without using a third party libraries. The Multipoint Control Unit is very expensive and it can be rented from service providers such as jitsi, kurento, twilio and the pentok during the conference. Here they used Node JS and Quality of Experience (QoE). The Node JS is a server platform and is a google's open source high-performance JavaScript engine and it can be the platform for building the network web application whereas the Quality of Experience (QoE) refers to the perceived quality of audio and video communication facilitated through web browsers and other compatible applications using real-time communication protocols.

DISADVANTAGES

Some of the disadvantages include in the existing system are shown below,

- It is very expensive.
- It can able to support a specific number of users.
- Need separate installations of additional plugins and software's.
- Lack of privacy.
- Pay-per-use model.
- Utilizes centralized servers.
- Difficult to set up.

VI. PROPOSED SYSTEM

The purpose of this system is to use the underlying Web Real Time Communication (RTC) technology to enable the interaction between various students and experts so that the knowledge can be imparted in an effective manner. In this there is no need of plugin approach offered by WebRTC is helpful since the users does not need to install the pug-ins like Adobe Flash and the third party software's such as Skype. All the users need to access the web application is to access it via the Uniform Resource Locator (URL) within the web browsers. We proposed to the online live teaching using WebRTC protocol and live sharing notes can use the WebSocket, Socket.IO, Peer JS and TCP protocol to work and run with the system. To detect the background noise during conference we used the noise cancellation technique and we ensure to increase the participants limit is up to the range of 500 members for free plan and also to increase the time limit is up to the range of 5 hours only for free plan.

During conference, If the registered participants has not yet joined in the classroom it goes an automatic call as the class was started but you forgot to join the classroom so you can join into the classroom which the expert had sent the link to your email. If the participants are visiting another tab's and pages the admin can give the alert that you are inactive in the class and if you do it for next time will dis qualified from the classroom. After finishing the classes, the recording notes and videos will be sending via their respective registered student's email. Additional functionalities are supported such as Whiteboard, messaging, file transfer, screen sharing and attendance tracking.

This system is a user friendly User Interface (UI) that can able to provide an easy understanding of the system and all the users required to establish connectivity with the other peer is the link to the website. The proposed method allows the use of voice, video and messaging to point out specific things you are concerned about or to show the person on the other end exactly what you are getting at, which demonstrates the learning capability of Web Real Time Communication (RTC).

ADVANTAGES

The advantages include in our proposed system is listed below,

- Real time communication with experts: The system allows for direct real time communication with the experts which is lacking in the existing system, using which students can acquire knowledge and proceed accordingly in an effective manner.
- Permits dynamic interactions: The system enables dynamic interactions between the participants through media data such as voice and video, and also supports advanced functionalities such as screen, file sharing and whiteboard for more active participation.
- Readily available for public usage: The system is readily available for active use by the end users since the system is hosted on a public server.
- Decentralized approach: The existing system involves maintaining a separate server for handling the media traffic and maintaining it is not cost effective but the proposed method eliminates

the need of a server. Since connectivity is directly established between the peers.

- Plugin free mechanism: The system does not requires any additional plugins and third-party software for its operations thereby enabling the users a hassle-free experience.
- Detecting background noise: It can detect background noise for ensuring a clear and distraction-free learning environment to users.

VII. WORKING

In the peer Learning, the system consists of the following components for its working.

A. STUN and TURN

The STUN server is primarily made use by clients to know their public IP address. Since most of the clients are located behind a NAT they are unable to determine their public IP, this is where the STUN server comes into play. STUN is a client-server protocol and can work across TCP and UDP connections. A STUN server can be defined as an entity that can handle the STUN requests made by the clients. The aim of STUN is to overcome issues associated with lack of standardized behavior in NATs. STUN servers generally reside on the public internet and have a simple task: check the IP: port address of an incoming request and send the obtained address back to the requesting clients as a response. The STUN servers are generally publicly available with offerings from Google as well. Most of the WebRTC calls make use of the STUN server to know and communicate the IP between the peers so that the peers discover each other and the connectivity is established through the signalling mechanism, in order to set up a direct link. STUN can also provide protocol encryption through TI_S (Transport Layer Security) which can guarantee message integrity and authentication.

The TURN server has been used when peer to peer communication fails. TURN can have advantage over STUN since it has the ability to traverse symmetric NATs. The TURN server acts a fallback mechanism when the STUN server fails since STUN servers are not reliable in nature especially when it comes to the handling of heavy media traffic and due to low bandwidth supported by them. However, TURN server has a disadvantage as when it comes to cost, maintenance, and huge bandwidth usage when HD video stream is being delivered.

B. Signalling

The Signalling is a crucial aspect of WebRTC, responsible for setting up and coordinating communication sessions between peers and the signalling is used to exchange session control messages between the peers using WebRTC connection. This signalling process needs a way to the clients to pass messages back and forth. The session exchange proceeds with one client making an offer and the other client send back the answer and the connection proceeds when the either of the clients accept the connection. The signalling mechanism is not implemented by the WebRTC APIs: it needs to be built separately. For the purpose of signalling Socket.IO is used and provides a good solution.

The Socket.IO is a JavaScript library that can provide a reliable and effective way to setup a real time connection between peers and this Socket.IO acts as the signalling medium and its fallback mechanism when the standard WebRTC peer to peer connectivity fails. Socket.IO can be added with a simple piece of middleware to enable this functionality so there is no need to setup own signalling exchange or deploy and scale new servers. The session termination and error handling in signalling also handles error conditions and session termination. If there are connectivity issues or if one of the peers decides to terminate the session, signalling messages are used to communicate this information to the other peer. The ICE is a framework used by WebRTC to establish a direct peer-to-peer connection even when both peers are behind NAT (Network Address Translation) or firewalls. During signalling, peers exchange ICE candidates, which are network endpoints (IP addresses and ports) where they can be reached.

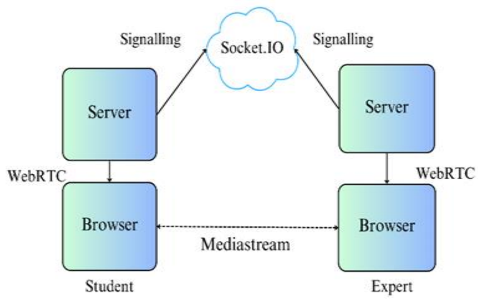

Figure 1: The architecture of Signalling

VIII. ARCHITECTURE

To start a live teaching session, initially the students has to be registered via their required details, if the user is an existing user then they can login into classroom and if the user is a new user then they can signup to the classroom during conference. The login request will go to the server while they logging in to the classroom. Once the teacher has created class and connected into the classroom then the link will be invited to the respective registered students mail id. So that the students can join in the classroom via the link. If the registered students has not yet joined in the class, then the admin gives alert to the student as you have registered to the conference, the class is commenced so you can join into the classroom. Once the users had joined in the classroom then they can able to connect peer to peer (P2P) via WebRTC (Web Real Time Communication) to the web browsers easily. We need a sever to establish a connection from one client to other client uses signalling server in the Web Real Time Communication (WebRTC). While going conference, the users can use webcam and multimedia streams such as audio and video to communicate with the experts. The experts can utilize whiteboard or a screen sharing feature to write and draw in real-time to deliver lectures and while teaching the notion will going on and at the same time the students can rectify their doubts with the experts. Once the classes was over, the notes and recorded videos has sent to the registered students email so that they can view documents. The attendance details will store in the database and the recorded videos and notes also stored in the database.

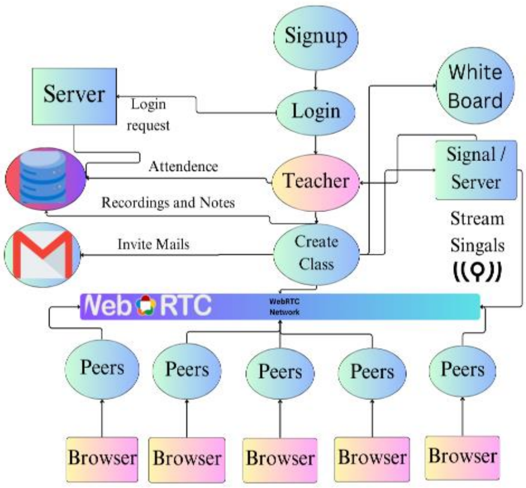

Figure 2: Architecture

IX. MODULE DESCRIPTION

In online live teaching using Web Real Time Communication (RTC) protocol and live sharing notes, we are using nine module such modules are listed below.

A) User Authntication and Authorization
B) Video Coferencing
C) Whitebord
D) Screen Sharing
E) Invite Mail
F) Attendance Tracking
G) Realtime chatting
H) Analytics and Reporting
I) Noise cancelation.

A) User Authentication and Authorization

User Authentication: In user authentication, both the experts and students can verify their required details such as username, password etc via authentication. It ensures that only authorized individuals or entities can gain access to protected information or functionalities. User Authorization: User authorization is the process of determining what actions or resources an authenticated user is allowed to access within a system or application. The authorization controls what activities the student and expert can perform or what data they can view or manipulate after they have been authenticated.

B) Video Conferencing

WebRTC (Web Real-Time Communication) technology is commonly used for implementing video conferencing applications directly within web browsers or mobile apps. The main purpose of this module is to allow multiple people to meet and collaborate face to face long distance by transmitting audio, video streaming, data sharing, peer to peer communication, text and presentation in real time through the internet.

C) Whiteboard

Implementing a whiteboard feature in a WebRTC-based application allows users to collaborate by drawing, annotating, or sharing visual content, explain and teach, write down ideas and many more things in real-time during video conferences or online meetings. It allows the users to export whiteboard content as images or PDF files for sharing or future reference and also it allows drawing tools such as pencil, eraser, shapes (e.g., lines, rectangles, circles), text input, and freehand drawing.

D) Screen Sharing

Implementing screen sharing and live notes sharing in online live teaching using the WebRTC protocol involves integrating various features to facilitate real-time collaboration between teachers and students. Utilize the WebRTC protocol to enable screen sharing functionality. To establish WebRTC peer connections between the teacher's screen sharing stream and the students video streams using a signalling server and to transmit the screen sharing stream to all connected students in real-time. We implemented mechanisms to handle permissions and user consent for accessing screen sharing capabilities.

E) Invite Mail

The experts can invite mail to the students to join into the conference whenever the expert needs to be commenced classes to the students at that time the link will be going to the respective registered students email when the expert clicks the joining class. The expert can invite the class link to the students with the date, time, platform along with the meeting link and by using the link the students can join into the classroom via peer connection. This session will be facilitated by expert educators who are dedicated to providing an enriching learning experience for all participants.

F) Attendance Tracking

On the day of the session, the students and experts can join the meeting promptly at the scheduled start time using the provided Web Real Time Communication (RTC) meeting link. Upon joining the session, the student will be prompted to check in so that they can provide their name or any required information as requested to verify attendance and also at the end of the session, there may be a brief acknowledgment or final check-out process to confirm the student attendance before the session concludes.

G) Realtime Chatting

The Real-time chat is virtually any online communication that provides a real-time or live transmission of text messages from sender to receiver. The students and experts can share multimedia content such as images, documents, and links to enrich discussions and learning experiences via real-time chat and they can stay updated with real-time notifications for new messages, mentions, or replies and to ensure that you never miss out on important conversations. They can also join group chats dedicated to specific topics or modules, facilitating discussions and collaboration among participants with their shared interests.

H) Analytics and reporting

In online live teaching using WebRTC protocol and live sharing notes, the students and experts can get a benefit from reports as they quickly access and understand essential information and the course insights of student and expert can access analytics dashboards that provide an overview of course metrics, including enrollment statistics, completion rates, and learner feedback and the students and experts can monitor their progress and performance in real-time, allowing them to track the learning journey and identify areas for improvement.

I) Noise Cancellation

In this module, the online live teaching uses noise cancellation to detect the background noise and it can able to minimize the distractions such as hatter, traffic, or other disturbances that may disrupt your concentration during online lectures or study sessions so that we can get clear audio without the interference of ambient noise, ensuring that you can hear every word of your instructor's lecture or discussion with utmost clarity. The students can also improve concentration by eliminating external distractions, noise cancellation technology helps them to maintain focus and immerse themself fully in the learning materials, leading to better comprehension and retention of information.

X. IMPLEMENTATION

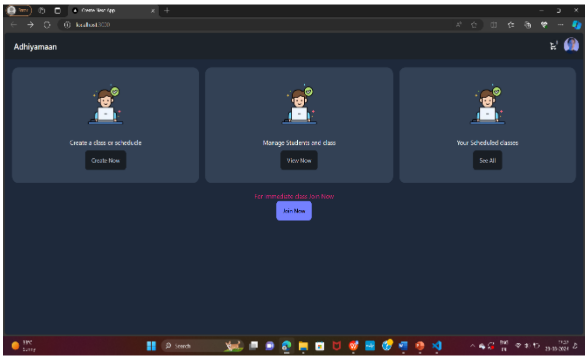

Figure 3: Homepage

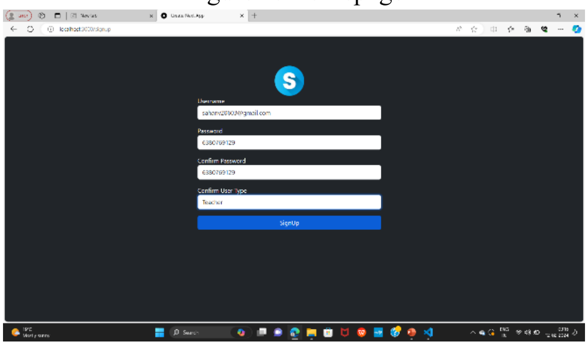

Figure 4: Signup Page

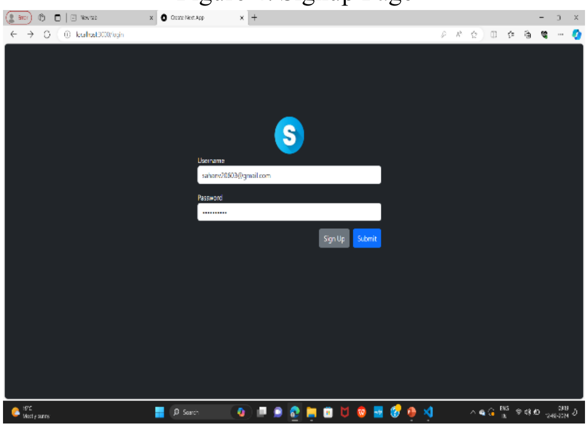

Figure 5: Sign In Page

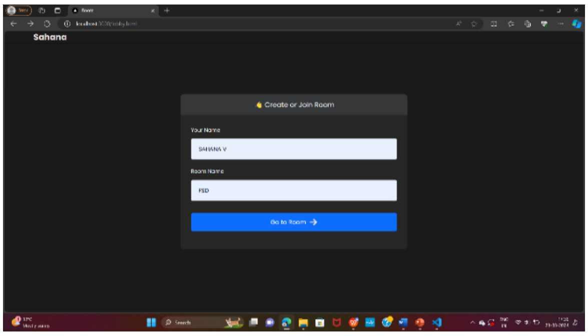

Figure 6: Create a class

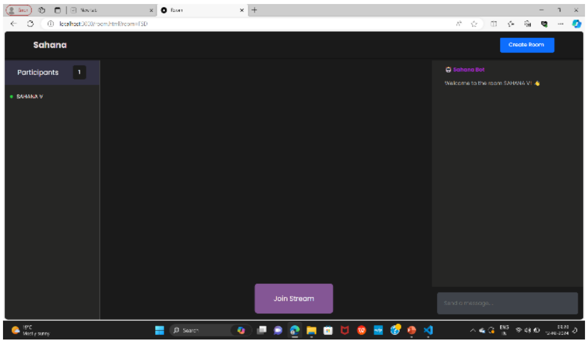

Figure 7: Joining into a class

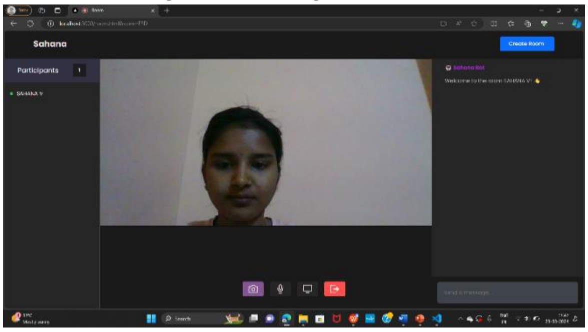

Figure 8: Video Conferencing

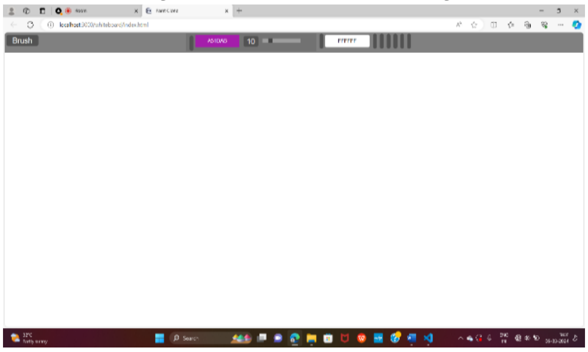

Figure 9: Whiteboard

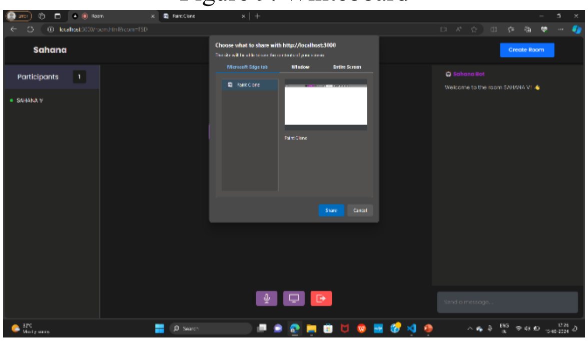

Figure 10: Screen Sharing

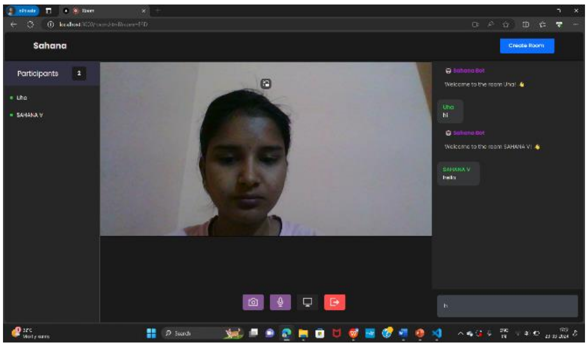

Figure 11: Real Time Chatting

XI. CONCLUSION

The Web Real-Time Communication (WebRTC) was developed as a new standard for facilitating the RTC between users through different web browsers without any extra installation. However, there is a major challenge for implementing WebRTC, that is, the WebRTC does not specify the communication standard between different browsers, which prevents the WebRTC from functioning correctly due to the incompatibility problem with multi-browser connection. The online live teaching using the WebRTC protocol and live sharing of notes, offers a modern and efficient approach to the education and it is possible for one-to-many connections. This paper utilized WebSocket protocol as a server to fully communicate between two different browsers and Socket.IO is used in WebRTC signalling mechanism for the unlimited peers during conference. By using the underlying WebRTC technology, a peer-to-peer learning system is designed and implemented. Such a system fosters collaboration and interaction, increasing engagement in and out of the classroom and increases accessibility and reach. The main goal is to make learning more affordable and cost effective which is possible through the plugin-free nature of WebRTC. The system can also be scaled to serve multiple users simultaneously. It is also possible to include a virtual whiteboard where the participants can draw diagrams and pictures which can be communicated in real time between them which can further help improve the learning capability of the system.

REFERENCE

[1] Natkal Moaid Edan, Ali-Al-Sherbaz Scott Turner, "Design and Evaluation of Browser-to-Browser video conferencing in WebRTC" – 2022.

[2] Global Information Infrastructure and Networking symposium- 2022.

[3] T. Sandholm, B. Magnusson, and B. A. Johnsson, “An on-demand WebRTC and IoT device tunneling service for hospitals”.

[4] S. Holmer, M. Shemer, and M. Paniconi, "Handling Packet Loss in WebRTC," 2013 IEEE Int'l. Conf. Image Processing, Melbourne, VIC, 2013, pp. 60-64, DOI: 10.1109/ ICIP.2013.6738383.

[5] R. L Barnes and M. Thomson, "Browser-to-Browser Security Assurances for WebRTC," IEEE Internet Computing, vol. 18, no. 6, Dec. 2014, pp. 11-17, DOI: 10.1109,'MIC.2014.106.

[6] Boni Garcia, Francisco Gortazar, Luis Lopez-Fernandez, Micael Gallego, Miguel Paris, "WebRTC Testing: Challenges and Practical Solutions", IEEE Communications Standard Magazine, vol. l, no. 2, July 2017, pp. 36-42, DOI:IO.1109/MCOMSTD.2017.1700005

[7] S. Loreto and S. P. Romano, "Real-Time Communications in the Web: Issues, Achievements, and Ongoing Standardization Efforts," IEEE Internet Computing, vol. 16, no. 5, October 2012, pp. 68-73, DOI: 10.1109/MIC.2012.115.

[8] I. T. Management, “WebRTC in the enterprise” 2016.

[9] J. Jang-Jaccard, S. Nepal, B. Celler and B. Yan, “WebRTC based video conferencing service for telehealth,” Computing, vol 98, no 1-2, pp 169-193, 2016.

[10] M. Phankokkruad and P. Jaturawat, “An evaluation of technical study and performance for Real-Time Face detection using Web Real-Time communication.” No.14act, pp. 162-166, 2015.

[11] L. O. D. Nedberg, “Quality of Experience of WebRTC based video communication Erik Fosser” Norwegian University of Science and Technology, 2016.

[12] C. Cola and H. Valean, “On multi-user web conference using WebRTC,” in 18th international conference on system theory, Control and Computing, ICSTCC, pp. 430-433, 2014.

[13] Suciu G., Anwar M.,Virtualized video conferencing for eLearning, 14th international scientific conference eLearning and software for education Bucharest, April 1920, 2018.

[14] J. Nurminen, A. Meyn, and E. Jalonen, “P2P media streaming with HTML5 and WebRTC,” in IEEE International, no. 4, pp. 1–2, 2013.