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Talk Over: A Voice Chat Web Application based on WebRTC

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ABSTRACT
There are very less number of systems available for voice streaming and chatting and also they are often criticized for their low voice quality and latency. The technology is growing and the systems made from the conventional Technologies are not sufficient to meet the today's need. There is a need of compatible, and a simple complete voice chat system. We offer a web application for Voice Chatting and broadcasting for live debates and discussions in this study using JavaScript and the WebRTC API, and allows users to communicate in real-time through a web browser. This system will allows a user to create a private or public voice channel for private meetings and public broadcasting . It will allow users to join any public channel or private channel through invitation link. The results showed that the application was able to provide high-quality audio with low latency, making it suitable for use in various scenarios such as remote work and online education. Overall, the development of a voice chat application based on WebRTC provides a convenient and efficient solution for realtime communication over the web. The application can be easily accessed from any device with a web browser, making it a useful tool for a wide range of users.

Keywords
WebRTC , Voice broadcasting

1 INTRODUCTION
Talk Over allows users to talk In real time and access a single site where they may join various rooms based on their interests, connect with others, and Even create their own rooms. This paper aims to provide a System for Debates and Discussions for users which can be simplified by using WebRtc's data channel API to transfer data directly from one peer to another peer connection. The web application grants clients for a consistent casing correspondence and to run over a singular spot through which clients can join different rooms in view of their interests and can speak with others and besides they are allowed to make their own rooms.

WebRTC (Web Real-Time Communication)[12] is an open source project that provides web browsers with real time communication capabilities with the help of simple API. This allows developers to build applications that support audio and video communication between users, without the need for any other external dependencies. WebRTC has the potential to make a difference in the way we communicate and interact with each other online. With its ability to enable high-quality, low-latency audio and video communication, WebRTC can be used to create a wide range of applications, including voice and video conferencing, online gaming, and voice chat.[2][3][4]

In this research paper, authors explore the use of WebRTC for voice chat. Paper will first provide an overview of WebRTC, including its architecture and design, and discuss its key features and capabilities. After that it describes the challenges and limitations of using WebRTC for voice chat, and discuss potential solutions and future directions for the technology.

2 LITERATURE REVIEW
In recent years, there have been a number of research papers published on the WebRTC .One area of research in WebRTC is voice chat, which refers to the ability to hold voice conversations over the internet using WebRTC technology. [2][3]

WebRTC is a collection of APIs and protocols which provides real time audio & video communication between web servers.

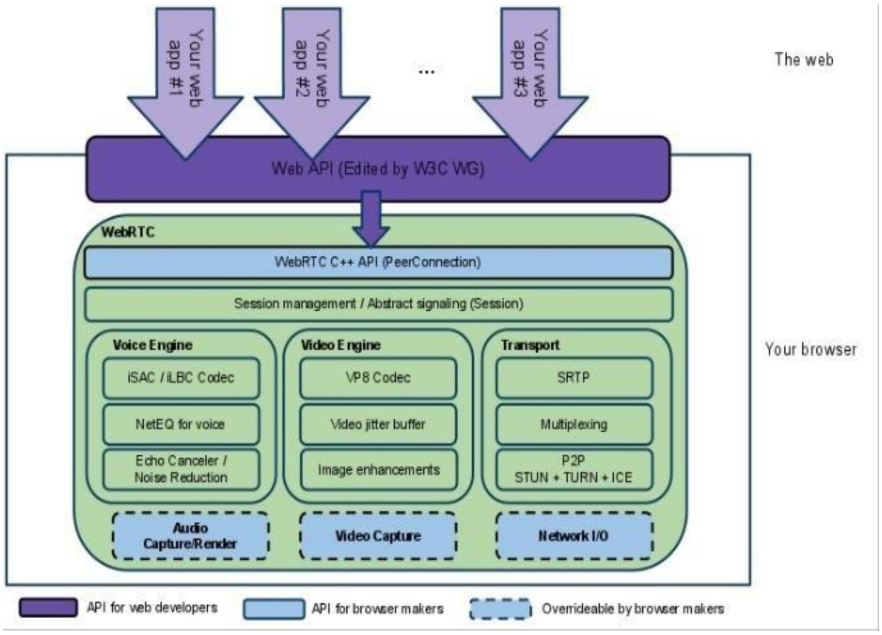

Fig. 1 WebRTC Architecture[1]

The core components of WebRTC include the following [2]:

- Media capture and streaming: WebRTC provides APIs for accessing the user's microphone and camera, and for capturing and encoding audio and video streams. This allows developers to build systems that can capture audio and video from the user's device and send it to other users in real time.
- Network connectivity: WebRTC includes protocols and mechanisms for establishing peer-to-peer connections between web browsers and mobile devices. This enables users to communicate directly with each other, without the need for a central server or other intermediary.
- Codecs: WebRTC uses a number of open-source codecs for encoding and decoding audio and video streams. These codecs are designed to provide high-quality audio and video at low bitrates, which is important for real-time communication applications.
- Security: WebRTC includes mechanisms for encrypting audio and video streams, as well as for authenticating users and protecting against common threats such as man-in-the-middle attacks.

2.1 Key features and capabilities:

WebRTC has a number of key features and capabilities that make it well-suited for voice chat applications.[3][14][15] These include:

- High-quality audio: WebRTC uses advanced codecs and algorithms to provide high-quality at low bitrates, which is important for real-time communication applications.
- latency: WebRTC is designed to minimize the delay (or latency) between when a user speaks and when the other users hear them. This is critical for voice chat applications, where even small delays can make the conversation difficult or impossible to follow.
- Peer-to-peer connectivity: WebRTC enables users to communicate directly with each other, without the need for a central server or other intermediary. This can reduce the cost and complexity of voice chat applications, as well as improve their performance and reliability.
- Cross-platform compatibility: WebRTC is supported by a wide range of web browsers and mobile devices, which means that voice chat applications built using WebRTC can be used by a large and diverse audience.

2.2 Challenges and limitations[13][14]:

Despite its many strengths, WebRTC is not without its challenges and limitations when it comes to voice chat applications. WebRTC is a complex technology, with many different components and APIs that developers must understand and use correctly in order to build effective system.

3 PROPOSED SYSTEM

The Talk Over Voice chat system is made with the web development technologies of HTML, CSS, JavaScript and MERN stack. WebRTC is used to implement the real time communication system.

The objective is to create a web server to server real time communication system that provides system for users to communicate with a high speed data transmission over the communication medium based on WebRTC (Web Real-time communication) technology without the need of installation of any plug-ins or third-party softwares or apps. The system is designed to allow audio, with user identification and finding of other system users without the need for installation of any system.[5][7]

When the user comes on the platform it first authenticates itself by entering the phone number or email id and validates it by entering the One Time Password generated on that phone number/email by the mechanism of the system. The system will generate a token which is valid for a particular time period and the user will need to complete its profile .In the provided time. If the user is unable to complete its profile in the given time then the session will expire and the user Will have to retry again. After successful authentication the user will enter into the main page of the web app where the user has two options, either To

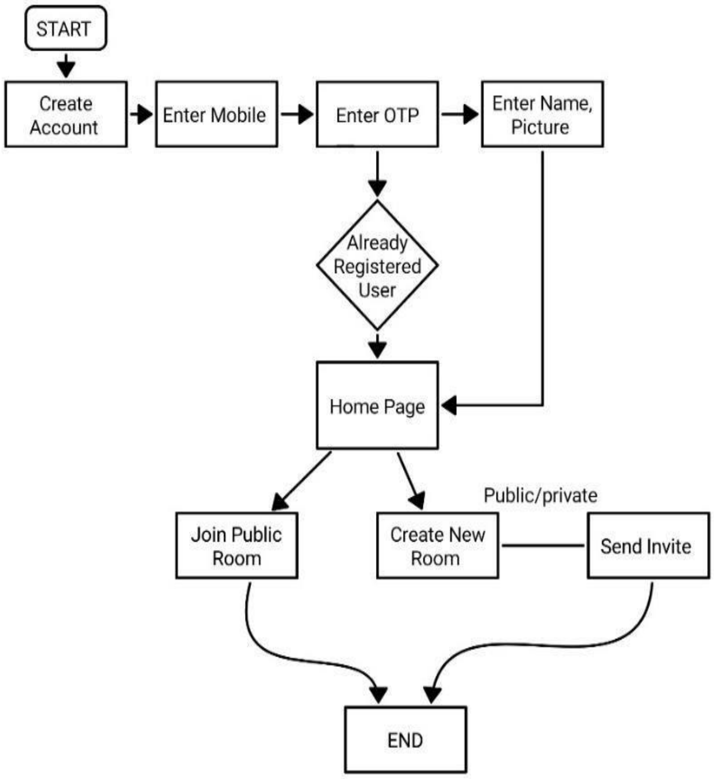

Fig. 2 Flowchart Diagram

create its own room or to enter in existing created room by the other users.[8][9]

User Interfaces : The user interface (UI) of the voice chat application based on WebRTC presented in this research paper was designed to be intuitive and user-friendly. The main UI consists of a simple chat window with a list of connected users on the left, and a chat area on the right.

At first, the user has to register himself on the web application. There is auto login facility. If the user is the first time login to the system, user need to register himself, the information of user will be stored as cookie and system will use cookie to maintain the login status of user so that user doesn't need to enter the credentials again next time using system. To initiate a voice chat, the user can select any discussion group according to its choice. The discussion group also contains a description over it. The user can also create a public channel of its own choice.The user can mute /unmute himself only.

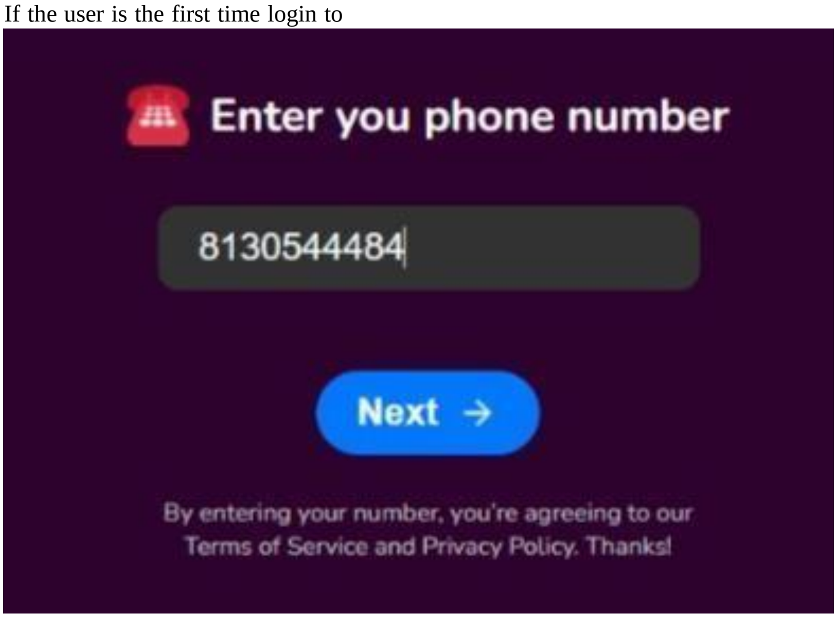

Fig. 3 Login Page

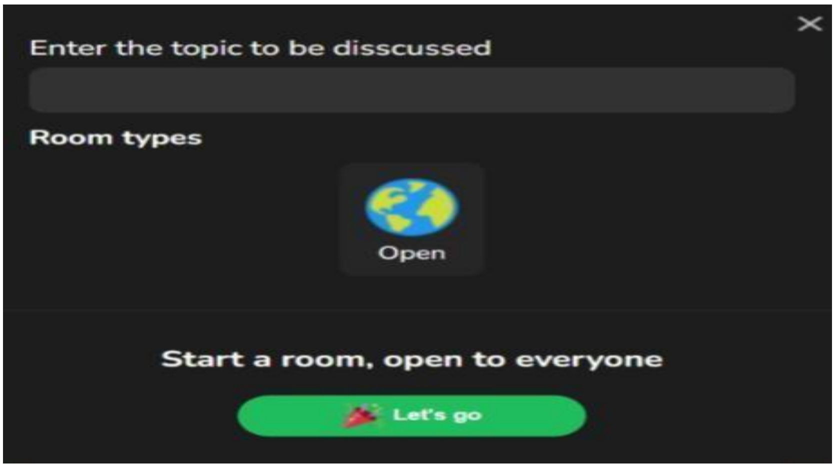

Fig. 4 Creating a Public Channel with a topic of discussion

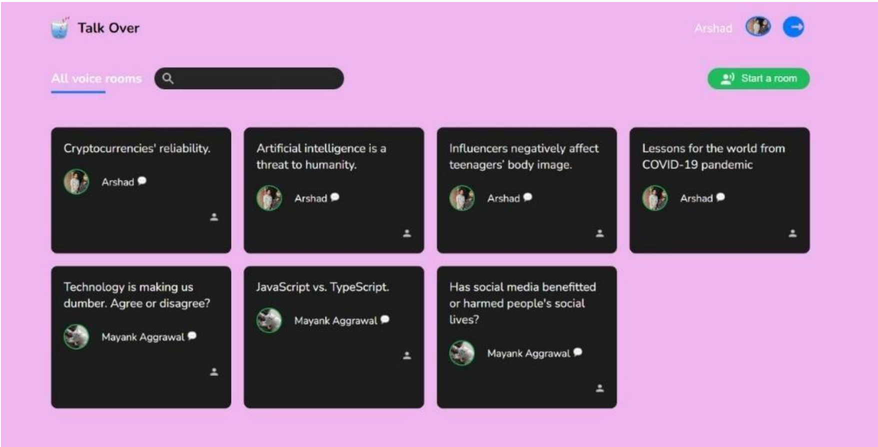

Fig. 5 Home page consisting public voice channels

Overall, the UI of the voice chat application was designed to be easy to use and to provide users with the necessary controls to customize their audio settings for optimal performance.

4 CONCLUSION

In conclusion, the use of WebRTC technology in voice chat applications offers a number of benefits[6], including improved performance, lower latency, and better security compared to traditional chat technologies. The ability to establish peer-to-peer connections directly between users' devices allows for high-quality audio and video communication without the need for centralized servers, resulting in faster and more reliable connections.

Additionally, WebRTC uses end-to-end encryption to protect the privacy of users' communication, making it a secure choice for businesses and individuals alike.

Overall, the adoption of WebRTC in voice chat applications has the potential to greatly enhance the user experience and drive the proliferation of real-time communication across a variety of industries and applications. As the technology continues to evolve and improve, it is likely that we will see an even wider adoption of WebRTC in voice chat and other real-time communication applications in the upcoming time.

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