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A REVIEW OF MYFRAMES –VIDEO CONFERENCING WEB APPLICATION USING WEBRTC

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Abstract: This paper provides an overview of video conferencing software, which allows individuals to visually communicate with one another and has applications in a variety of industries. Users of existing virtual video conferencing software experience issues with internet connectivity, which affects audio and visual quality. Various browsers and operating systems are incompatible with many applications. In this application, security is also crucial. We have improved the shortcomings of previous programmes with the help of WebRTC. It uses a variety of protocols for audio and video encryption and authentication, including the Secure RTP protocol (SRTP). The Opus audio codec is used by WebRTC to provide high-fidelity voice. This article highlights the various types of improvements that can be achieved by utilising WebRTC's effective protocols.

***Index Terms* – Virtual Video Conferencing, WebRTC, Opus , SRTP.**

I. INTRODUCTION

In terms of virtual meetings, video conferencing is the most widely used technology, with applications in education, medical, military, and business meetings. Many people use video conferencing, which was first launched in the late 1990s. Video conferencing apps have seen an unparalleled increase in downloads during the epidemic era. The video conferencing market is anticipated to reach $11.56 billion by the end of 2027, according to Transparency Market Research. Video conferencing has numerous advantages, including time savings, virtual meetings, increased productivity, lower travel costs, and so on. We've created a video conferencing application with a variety of features to help with this. Many challenges have been identified in previous study on the usefulness of videoconferencing, including network issues, hardware, security, audio and visual quality, and so on. We implemented WebRTC for our application to help us overcome the disadvantages. Google launched WebRTC in 2011 to provide rich, high-quality RTC applications like audio and video teleconferencing, as well as peer-to-peer data sharing. The video and audio transmitted over the app are encrypted and authenticated using the Secure RTP protocol (SRTP). Interoperability with existing audio and video systems, including devices that use SIP, Jingle, XMPP, and the PSTN, is also included. It uses RTP Control Protocol (RTCP) and Secure Audio Video Profile with Feedback to support numerous media streams and adapt to network conditions (SAVPF).

Videoconferencing, whether accessed via web, desktop, or mobile, is one of the most widely utilised technologies in higher education for promoting learners' self-directed use of technology in a synchronous manner. Individuals' learning experiences can be influenced by employing various types of communication within and across different learning contexts, according to Lawson, Comber, Gage, and CullumHanshaw (2010). Coventry (1995) shows how videoconferencing can be integrated into a learning framework by focusing on the learner rather than the technology, and emphasising the importance of institutions having a thorough understanding of videoconferencing capabilities before committing to the use of videoconferencing technology.

II. LITERATURE SURVEY

Video conferencing will soon be more than a perk for businesses; it will be a need for those that want to stay competitive. Currently, 59 percent of employees use video communications at work on a regular or weekly basis, with 45 percent doing so on a daily or weekly basis. Almost half of the respondents say video consumption at work has increased since two years ago, while 27% say personal usage has declined in the same time period.

Some Video Conferencing Systems

Many video conferencing systems are available, including Skype, Q audio Conf, and Cisco WebEx Meeting.

i]Skype:

Skype is a software that allows you to make audio/video calls as well as send and receive text messages and files over the Internet. In 2003, the initial version of this software, which was designed for voice communication, was released. It has recently become popular since then as one of the first websites towards using VoIP technology.

ii]Q audio Conf:

QCONF takes use of WebRTC technology to provide low-cost international audio conferencing that is both safe and high-quality. Audio conferencing is offered via secure local access lines in over 50 countries across the world. Regardless of how many people join or how long you talk, you'll simply have to pay one set rate per conference call. Easy-to-use online and smartphone apps for scheduling and meeting management.

iii]Cisco WebEx Meeting:

Video conference Meeting with Cisco WebEx is a multi-purpose audio and video conferencing system for companies of all sizes. Cisco WebEx can host many meetings at the same time, allowing users to interact in real time. For large-scale promotional events and educational sessions, this web conferencing software is also highly recommended.

Iv]Google Meet:

Google Meet is a video-communication service created by Google. It's one of two applications that replace Google Hangouts, the other being Google Chat. Google Meet is tightly integrated with Google Suite, making it simple to schedule meetings using the Event Calendar.

III. Proposed System

This project explains how to solve the problem by creating a similar web application that uses WebRTC. Web Real-Time Communications (WebRTC) is an Application Programming Interface that allows web developers to integrate Real-Time Communication (RTC) capabilities into their web-based applications without the need for plugins. Peer-to-peer (P2P) architecture is superior to client-server design in terms of scalability and reliability, as single nodes failure does not effect the entire system. In addition, the WebRTC system consists of a web server and browser that run on many operating systems, as well as workstations, tablets, and mobile phones.

In the web-based system, WebRTC is used for real-time audio and video transmission, while Node.js is employed as a web and signalling server. WebRTC is a protocol that allows web apps and sites to capture and potentially broadcast audio and/or video material, as well as share arbitrary data across browsers, all without requiring the use of a third-party server.

WebRTC is a protocol that allows users to exchange data and conduct peer-to-peer teleconferencing without the need for third-party programmes or plug-ins. Using Node.js, a server is required to establish remote connectivity with two or more devices (users). In this instance, you'll need a server that can handle real-time communication. Node.js is a programming language designed for scalable, real-time applications.

Develop two-way connection apps with free data exchange using Web Sockets, which allows a client and a server to create a communication session. Because client requests are processed in a loop, notably the event loop, Node.js is a good fit. Because it serves requests in a "non-blocking" manner, it achieves low latency and great throughput along the way.

IV. Conclusion

The inquiry of the WebRTC framework is presented in this review paper, which relies on a number of mechanisms with extensive histories: offer/answer negotiation, NAT/firewall traversal, RTP-based media exchange, peer-to-peer data channels, and the web itself. When they're combined, they'll form an open ecosystem that will enable peer-to-peer apps far easier to build and far richer in media content.

We discovered the many ways for real-time transmission or communication via the web as a result of the review.Methods, on the other hand, have some limits that can be overcome by upgrading existing methods. In the future, we propose presenting an effective way for real-time communication over the internet.

When it comes to business requirements, the ability to create custom apps that match your specific demands, high reliability, and data security are all critical things to consider. Because Google has done all possible to make the development process as simple as possible, creating your own private WebRTC video chatting software will not result in substantial financial or time losses.

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VI. Author Contributions

All of the authors listed have contributed a significant, direct, and intellectual contribution to the work and have given their permission for it to be published.

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