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VIDIFY: ONLINE VIDEO CALL APPLICATION WITH FACIAL EXPRESSION DETECTION

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Abstract:
This paper gives an overview of video conferencing software, which allows people to connect visually with one another and has a wide range of uses. 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 discusses the numerous types of enhancements that can be realised through the use of WebRTC's efficient protocols.

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. To help us overcome the disadvantages, we integrated WebRTC in our application. 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) demonstrates how videoconferencing may be integrated into a learning framework by emphasising the significance of institutions having a complete grasp of videoconferencing capabilities before to committing to the usage 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 for 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. 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.

III. PROPOSED SYSTEM

This project explains how we can 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 affect 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.Node.js is a suitable fit because client requests are processed in a loop, particularly the event loop. We would be implanting Face recognition on our web application using completely light weight JavaScript based tensor flow using face-api.

IV. Applications of Face Recognition

Several application designs are available to help in face recognition and some of the applications are:

1- Application number one is related to finding a face within a large database of faces. The system retrieves a potential list of faces from the database using this method. Crowd surveillance, video content indexing, and personal identification are among the most helpful applications: driver's licence and mug shot matching are examples of this sort of application (Waite et al., 1991).

2. Application number two is about the real time face recognition. Face recognition is used to instantly identify a person and give admission to a building or compound, eliminating security concerns. In this case the face that is to be compared will be against multiple samples of a person (Bartoli, 2006) (Wright, 2009).

V. Existing Algorithms

Face Recognition Based on Principal Component Analysis. PCA stands for Principal Component Analysis, and it is a face recognition technique. The main principle of PCA is to find a much lower-dimensional vector that best approximates a given data vector in some way; therefore, in face recognition it takes as input, an s-dimensional vector representation of each face in a training set of pictures, and finds a t-dimensional subspace whose basis vector is maximal corresponding to the original image, and the size of this new subspace is smaller than the original ($t \ll s$). If the original image elements are considered as random variables, then the principal components are along the eigenvectors corresponding to the larger eigen values of the correlation matrix, and error minimization is done in a least-squares sense (Qing Chen, Xiaoli Yang, Jiying Zhao, 2006).

VI. CONCLUSION

This review article investigates the WebRTC framework, which is based on a variety of processes that have long histories: RTP-based media interchange, offer/answer negotiation, NAT/firewall traversal, 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 p

VII. 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. REFERENCES

[1] https://ieeexplore.ieee.org/document/89817 2 4.
[2] https://www.researchgate.net/publication/317544143_Demonstration_of_a_P2P_assisted_video_streaming_with_WebRTC_a nd_Network_Coding
[3] http://www.cs.sjsu.edu/faculty/pollett/master s/Semesters/Spring18/yang Cha/CS298Report.pdf
[4] https://www.researchgate.net/publication/332415582_A_PeerToPeer_Architecture_For_RealTime_Communication_Using_W ebrtc
[5] https://ieeexplore.ieee.org/document/ 7912968 [6] https://www.ijitee.org/wpcontent/uploads/papers/v8i9S