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WEBRTC BASED VIDEO CONFERENCE APP

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ABSTRACT

This article describes how WebRTChas swiftly gained fame as a platform for videoconferencing, in part because so many browsers now support it. Real-time audio, data and video communication in web and applications is accomplished through the opensource WebRTC. It is supported by many browsers. Secondly, we'll create up a signaling server using Socket.IO and Node Js Development Services. A new standard termed WebRTC allows users to use Real Time Communication in web browsers. A internet browser that has support for WebRTC incorporates the essential innovation to fabricate a two-way video talk clientstraightforwardly in the program withoutrequiring the client to download any software.

Keywords: WebRTC, Peer To Peer, SDP, ICE Candidates.

I. INTRODUCTION

The most current and empowered innovation supporting virtual learning frameworks are WebRTC ("Web Ongoing Correspondences"). It is a publicly accessible platform that enables real time communication over Internet. It's a strategy for communicating media capabilities data .HTML 5 along with bunch of principles, conventions, and JavaScript APIs are utilized in the innovation. Media stream API, which is an extension of HTML5, is used to access the users microphone and webcam for live streaming without installing any third party software . Moreover, getUserMedia() is the way to access local input devices.It permits you to consolidate different administrations like Texting, sound/video conferencing without the requirement of any modules or set up tools.The advancement region and opportunities for WebRTC based internet learning frameworks are tremendous. To set up WebRTC flagging, the ICE structure expects you to give two sorts of servers, explained beneath.

The STUN (Session Traversal Utilities for NAT) and TURN (Traversal Using Relays around NAT) are protocols used to facilitate real-time communication between devices over the internet. STUN servers are used to find the public IP address of a device behind a NAT firewall by exchanging contact information. Once this information is exchanged, a direct connection can be established between the devices, and the STUN server is no longer needed. Connections established via STUN servers are inexpensive and efficient, but they may not always work due to different types of NAT devices being used by each device. In these cases, ICE candidates find the best path for data transmission, and a fallback option is used: a TURN server. TURN servers are used when a direct connection cannot be established, and they allow data to be sent via an intermediate server. The main difference between STUN and TURN servers is that the latter handles the entire communication session. TURN servers are more expensive to use, as the device initiating the communication must pay for the processing and bandwidth required to operate the server.

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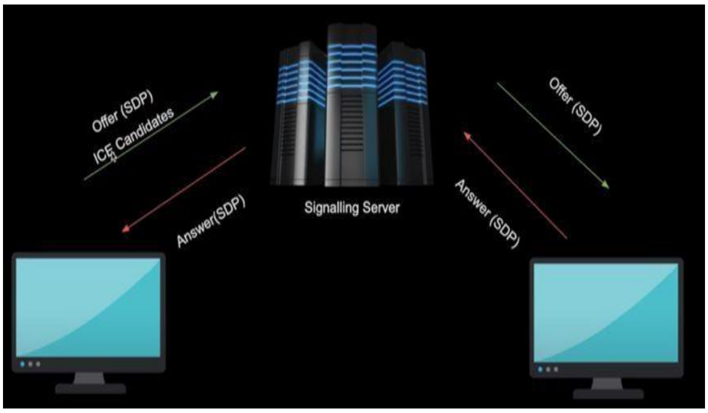

II. LITERATURE SURVEY

The author of [1] studied a web application thatallows real-time communication but is susceptible to network problems that lower theend user's satisfaction. In order to examine theconversation quality of a WebRTC-based audio/video service, the authors of the study conduct subjective tests. To hinder communication between the parties, a specialised system was put in place to handle network disfigurement such delay, jitter, and packet loss. WebRTC streaming places the utmost premium on low-latency delivery. As a result, scaling and quality have limitations when WebRTC is used in isolation from other technologies. Scalability wasn't considered in the architecture of WebRTC. The bandwidth- intensive arrangement necessitates peer connections between all participatingbrowsers. To put it into perspective, Tsahi Levent-Levi, a WebRTC expert, advises refraining from using more than 50 concurrent peer connections. It's a popular misperception that WebRTC's bitrate restrictions result in its poor quality. High-bitrate encoding is still conceivable even though browser-based contribution is essentially connectivity- and camera-dependent on the resolution front. Yet this kind of procedure might not be the best for streaming for content distributors that wish to use a professional encoder and camera. Additionally, adaptive bit rate streaming support is limited with WebRTC. [2]According to the study, webRTC basically uses two types of servers that are STUN and TURN servers as signalling mechanisms and depends on ICE candidates, which are crucial for peer-to-peer network communication. A significant drawback is that ICE which provides the best path for communication will occasionallyconnect peers using a STUN server and UDP. A TURN server, that can pass across NATs, will be used by ICE in the event if the STUN server fails. However, TURN relay servers willbe necessary for highly constrained corporate networks. Since servers are pricy, you'll either have to buy your own or pay more, although ICE can typically join pairs with STUN at a large performance advantage that justifies theircost. This paper [3] introduces an Augmented Reality video calling system that combines augmented reality with a standard visual call toprovide a captivating and dynamic communication experience. Real-time video connection is established via the webrtc protocol, and augmented reality features are added to the vivid streaming video feed using an Augmented reality software development Kit, such as Google AR core. In order to assessits staging and viability of the suggested Augmented Reality video-call system, the authors deployed the system on an android mobile device and measured pertinent performance statistics, such as frame rate,CPUconsumption, and network utilisation. The trialfindings imply that the suggested solution is workable and steady.[4] The authors of this research reviewed how webRTC technologies are employed in telemedicine to deliver cutting-edge medical services. They have demonstrated a webRTC-based solution for online medical consultations that supports voice, video, and chat calls. The communication architecture and protocol of the application have been thoroughly explained. The program is tested and assessed on variousoperating systems as well as various network connections, including 3G, 4G, local, and DSL.In [5], In this study, the author reviewed research on a new video conference technology that permit end viewers to supervise video conference activities. This technology uses WebRTC like a replacement for applications that still employ flash technology.Additionally, there are some other features that make it simpler for consumers, like the ability to record screen sharing.[6] In this paper the authors have described Web ongoing correspondence (WebRTC) based administrations present the better approach for correspondence and joint effort. These kinds ofadministrations permit continuous correspondence with voice, video, informing, and information sharing through program. Their prosperity and reception rely upon various elements that impact client's experience and its quality. As WebRTC based administrations are developing, it is vital to recognize these impact factors (Uncertainties) and examine their effect on nature of involvement (QoE). In this paper, the emphasisis on clients' involvement in WebRTC video calling. The goal is to decide the most and leastpersuasive elements overall as indicated byanalyzed clients' viewpoints with regards to WebRTC video calling, and for the fruitfuladministration of QoE. With this in center, thispaper might act as a reason for additional research in the field to build the usage and reception of WebRTC innovation .[7] In the paper authors reviewed that WebRTC has swiftly gained popularity as a platform for video conferences, in part because so many browsers now support it. For real-time communications over UDP, an algorithm is used by WebRTC which is the Google Congestion Control algorithm to offer congestion control. The basic implementation, device, network characteristics and network topology can all have an impact on how well a WebRTC call performs. In this research, we carry out a complete performance evaluation ofWebRTC in both real wired and wireless networks as well as in simulated synthetic network settings. Our analysis reveals that when competing with cross traffic, streams have a little higher priority . In general, WebRTC functioned as expected , we saw significant instances where there was opportunity for development. These include thewireless space and the most recent addition of video codec support.[8]In this paper, Authors have hightlighted some specialised field of videoconference applications, the World Wide Web introduced an evolution. The most recent technologies enable browsers to start real-timeconnections. One of the free and open source

initiatives is WebRTC, which aims to give users the flexibility to engage in real-time communication by adhering to and redefining the standards. WebRTC is still a young project, hence it lacks some sophisticated videoconferencing features like media mixing,session recording, and network situation adaption. This makes an effort to analyse the drawbacks and difficulties that WebRTC facesand offers a Multipoint Control a unit based design or a traditional communications entity based architecture as a remedy.[9]In this paper the author has reviewed that visual conferencing with WebRTC — Access to information in areas like media networking, text, visual conferencing, television over the internet and coalesce communication is made possible by real-time communication , a new standard and industry-wide effort. It also expands the online browsing model. Cloud infrastructures is utilised that sustain the quality of services, users can view, record, annotate and alter media material flows. There are numerous primary protocols and codecs that are not easily compatible and scalable that are used to build multipoint video conferencing systems. Using plugin-free Java Script Application Programming Interfaces, WebRTC is a cutting-edge open technology that enables real-time communication capabilities in audio, video, and data transmission through web browsers . Using Mozilla Firefox and the Scale Drone service, It demonstrated that with the uses of HTML5 and a Node.js server address to connect users with high-speed data transmission through the communication channel is possible to create a peer to peer web based communication system. Our experiments show that WebRTC is an effective tool for scalability of live videoconferencing inside a web browser.

1. PROTOCOLS HTTP: hypertext transfer protocol

Web-sockets: Java script interface

SDP: Session Description protocol

ICE: Interactive Connectivity Establishment

TURN: Traversal Using Relay NAT **STUN**: Session Traversal Utilities for NAT**TCP**: Transmission control protocol

UDP: User datagram protocol

III. METHODOLOGY

Let us understand how our Video Calling App Works? It helps to first understand the steps involved in a WebRTC video chat app from theperspective of the caller that is peer1 and calleewhich is peer2.

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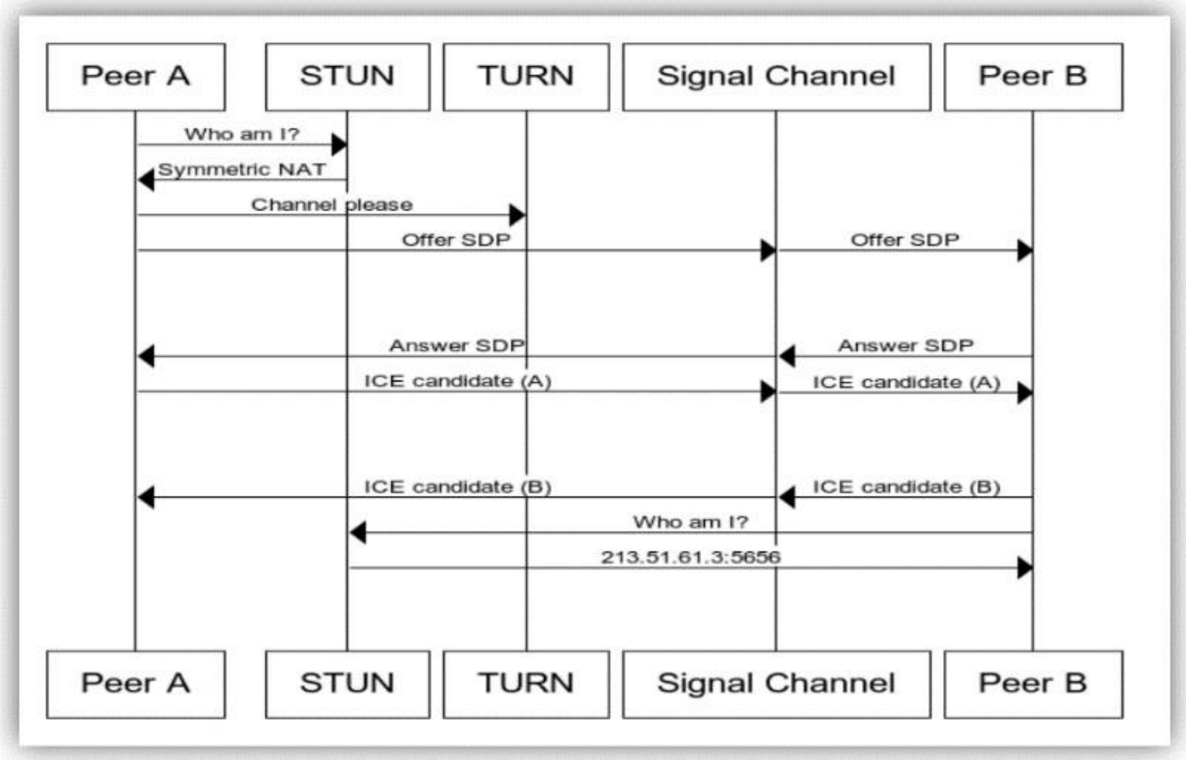

PEER TO PEER ARCHITECTURE

Caller (Peer 1)

Start a webcam feed

1. Create an 'RTCPeerConnection`connection
2. Call createOffer() and write the offer tothe database
3. Listen to the database for an answer
4. Share ICE candidates with other peer
5. Show remote video feed

Callee (Peer 2)

1. Start a webcam feed
2. Create an 'RTCPeerConnection`connection
3. Fetch database document with the offer.
4. Call createAnswer(), then write answer todatabase.
5. Share ICE candidates with other peer
6. Show remote video feed

IV. STEPS AND PROCEDURES

Step by step procedure upon how weimplemented our App are –

1. Display a Media-Stream video of yourself on your computer
2. Display a Media-Stream video of yourfriend on his computer
3. Create a Peer-Connection on your computer
4. Create a Peer-Connection on your friend's computer
5. Create an Offer on your computer
6. Add that Offer to the Peer-Connectionon your computer
7. Send that Offer to your friend'scomputer
8. Add that Offer to the Peer Connection onyour friend's computer
9. Generate ICE Candidates on your computer

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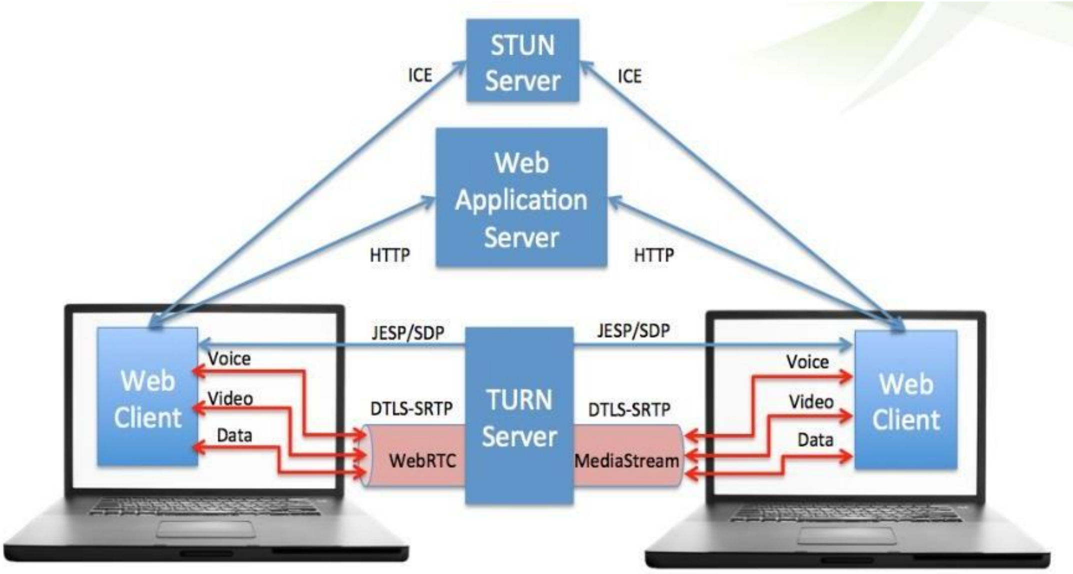
10. Send those ICE Candidates to yourfriend's computer
11. Add ICE Candidates to the Peer-Connection on your friend's computer
12. Create an Answer on your friend'scomputer
13. Add that Answer to the Peer-Connectionon your friend's computer
14. Send that Answer to your computer 15.Add that Answer to the Peer-Connection on your computer.
16. Generate ICE Candidates on yourfriend's computer
17. Send those ICE Candidates to yourcomputer
18. Add ICE Candidates to the Peer-Connection on your computer
19. Display a Media-Stream video of yourfriend on your computer
20. Display a Media-Stream video of yourself on your friend's computer



V. RESULTS

We successfully implemented the video calling App wherein more than two people can currently video call together using the web app.As a result we were able to achieve these manyresults as listed beneath:

1. Switch audio on/off
2. Switch video on/off
3. Create your own meeting url
4. Join an existing meeting with more thantwo people
5. Adding own name to profile with whichyou join the meeting
6. Update whenever somebody joins or leavesthe meeting
7. Copy URL in-meeting
8. Chat-box
9. Screenshare
10. Collaborative whiteboard
11. Leave meeting

VI. CONCLUSION

Using WebRTC technology ,WEBRTC is apowerful technology for real-time communication and has been widely used in video conferencing applications. One of the key advantages of WebRTC is its ability to provide a low-latency ,high-quality video and audio experience to users ,even in low- bandwidth environments, a video conferencingsolution was created. To create a successful video conferencing app using WebRTC, itsimportant to consider the user experience from start to finish. This includes designing an intuitive user interface ,providing features thatusers need , such as screen sharing, and recording , and ensuring that the video and audio quality is consistently high. For this study, WebRTC was chosen since it's easy to use and does not need any third party or other software loaded; so you need is a browser that supports it. A browser that works with a variety of operating systems powers the video conferencing system. The goal of this research is to lessen the effort and challenges associated with communicating while also creating a video. Calls like video-audio, file sharing, desktop sharing, and recording are all possible during a video conference. Depending on who goes, the format and attendance will change. Users will have the option to create and join rooms. They can transmit and receive audio and video, as well as build and join rooms.

They can share presentations and recorded videos that they have locally produced as well as receive audio-video broadcasts from other users. Additionally, it has cross-platform compatibility, the ability to communicate with one another either through personal or open texting. Computers, tablets, and smartphones can all use our software. In conclusion, WebRTC is an excellent technology for building real-time communication apps, and it can be a great choice for building a video conferencing app. By following best practices and focusing on user experience you can create a video conferencing app that delivers a seamless and high-quality experience for users.

VII. FUTURE SCOPE

The following are the future scope for theproject.

1. Developing new features and functionality.
2. Improving Video and Audio quality.
3. Deploy our Application over Heroku sothat our fellow Community can use it.
4. Individual Attendance system with photo using Student login.
5. To extend Support Components for evenmore versatile functionality.
6. Integrating with other technologies.
7. Augmented and virtual realityintegration.
8. Enhancing Security and Privacy features.
9. Optimizing for Specific Industries.
10. Integration with Artificial Intelligence.

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