SURVEY ON REAL-TIME PEER TO PEER MULTIMEDIA COMMUNICATION APPLICATION

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Abstract: In the modern world and fast development of the internet, the connection among people is being very significant than ever, people are looking for new methods to do advance communication between them without any issue, real-time communication is one of this ways. Currently video conference system usually needs to install application software. Therefore software need to be developed for different operating systems (android, windows, and mac) and user data goes through servers. However, web-based video conference system is OS independent so it saves development cost. We propose web based peer to peer real time communication system using WebRTC which will allow users to communicate with high-speed data transmission over the communication channel using WebRTC technology, HTML5 and Node.js server. WebRTC (web real-time communication) is a technology which allows real time audio, video and data transmission through web browser using JavaScript APIs without any plugins. Additionally media stream is encrypted and a management mechanism will be developed to manage entry of conference room to achieve secure multiparty video conference.

Keywords – WebRTC, Web Conferencing, peer to peer, JavaScript API.

I. Introduction

Since the internet came into existence, web browsing is always known as a client-server architecture in which one peer is the client and the other is the server. If a multi-party connection use a client-server architecture, all streams would be transmitted to the other peer through the server [6]. The server will be a network traffic bottleneck. Therefore, we consider to use peer-to-peer architecture for media streaming without any server. P2P technology doesn't require support from internet router and network infrastructure making this extremely cost-effective and easy to deploy. Participant that tunes into a connection not only downloads a Stream but also upload it, increasing scalability.[9]It is clear that simple standalone applications Can be implemented with HTML5, but it is more Challenging when we start to signal diverse applications with High computation and communication needs.[4]:WebRTC enables web applications to establish a direct Communication channel between two browsers without relaying The data through a web server.it support browsers like Google chrome and Mozilla Firefox.[12]:It holds Peer Connection, Media Stream, and Data Channels components, API that can be combined to create P2P direct media Communication between peers. WebRTC promises to provide secured direct P2P Communication between users and free of plug-ins. It assures a simplified, flexible, and cost-effective means of real time Communication for users without dependence on service Providers. [11] Using WebRTC the website server sends the browser IDs to the visitor. Using this ID, browsers can then connect to a specific peer browser. WebRTC is a Standard of the World Wide Web Consortium (W3C) that Allows for real time communication between Internet browsers. Overlay network does not need to know any IP address with which to establish connections. Instead, the WebRTC API is used for connection management. According to the WebRTC Draft, this API is built into browser applications that can make use of real-time communication by calling the appropriate JavaScript functions in the API. Thus, P2P applications between two or three parties can be easily implemented with WebRTC. In [1] here, a system has introduced that afford multimedia transmission service such as video, audio, text, picture and screen sharing which identifies the user and detects any other users of the system, satisfying the basic requirements to be considered secure without complicated installation or setup actions within a web browser on a variety of devices and operating systems based on WebRTC.

WebRTC is a form of real-time communication technologies that have added standards of API (Application Programming Interface) that have made real-time multimedia transfer such as voice, and video (including codes) available to web browser without a plug-in that makes high-quality multimedia communication from peer- to- peer available to web developers without traditional plug-in components using some JavaScript codes [13].

II. Related work:

For real-time multimedia communication though it is video conferencing, chat or calling there are many applications available for this purpose in the market. As they are using these proprietary protocols for the transmission of multimedia streams, it requires external applications for the mobiles and the desktop to access services such as video conferences, phone calls, messages, file sharing etc. Where, with the help of WebRTC we can add real-time communication capabilities to the applications as it supports video, voice and data transmissions between peers allowing us to build a powerful voice and video communication solution. The WebRTC architecture provides end-to-end encrypted P2P communication with audio-visual content and data being transmitted directly. The proposed architecture is mainly focused on bypassing intermediary hardware servers and eliminating the security challenges like hijackers, thus making it different from other applications like Skype, zoom etc. which lacks the direct p2p ability, as well as a credible security feature found in WebRTC. Peer-to-peer also enables user's data to be encrypted, safe, and cannot be compromised [11]. A peer-to-peer connection can be credible because it is able to bypass all the problems associated with plug-ins. Factors such as latency, bandwidth, and memory utilization as well as support for anonymity are supported in WebRTC.As mentioned in [11] they have proposed a web-based peer-to-peer real-time communication system using the Mozilla Firefox together with the Scale Drone service which limits the application to the specific browser where it lacks the multiple browser support.

Paper [1] introduce a system that afford multimedia transmission service such as one-to-one video/audio calling and text. We will be improving this system by providing additional features like group video conferencing, screen sharing, text sharing, file sharing and by providing collaborative whiteboard.

As mentioned in [2], they have developed systems for a large number of peers using WebRTC. Timer based connection management is applied in the implementation, where connections expire if a party is idle for 10 seconds. To keep connections from expiring when, heartbeat messages are sent every 3 seconds if no other data has been sent. It is only for video conferencing with high diffusion delay.

The paper [3] presented a signaling mechanism via web sockets. The whole architecture is based on handshaking by only a stun server, making it a single point failure. We are using an alternative turn server to establish connection through NAT traversal.

[4] Describes, online video conferencing system which can be implemented in different operating systems. It uses a web page, client/server design in a browser. It only transfers text and binary data and also creates distributed table for routing layers. It also uses server components to create node ids.

Work in [5] describes, they have focused on messaging and chat sessions. They have differentiated between instant messaging and internet chat sessions.

[6] Implies, a point to point media connection with complex authentication system. They have created a MS SQL database which contains user id, name, password, email, phone number. Set the default resolution to 200*200.

In [7] system they have designed two protocols for p2p streaming. The first protocol is WebPeer and second one CodedWebPeer for sending data over a network. These protocols splits the data into splits the data into pieces, but instead of transmitting individual pieces, the protocol groups them into generations. Each generation goes into one RLNC encoder that generates the network coded packets3 from the original data. Only these coded packets can be requested by peers.

In the [8] application has been developed in pure Java thus it requires additional software installation process. And this application is developed only for windows operating system making it platform dependent. Our proposed architecture is designed in a way that it is overcoming the limitations by providing multiple browser support and OS independency.

In [9], the main use case is video-on-demand using peer to peer approach. This paper uses HTML5 API for video publishing. The Peer creates a torrent file containing the metadata of the video, hashes of each piece of the video, and uploads the torrent file to the Tracker web application starts downloading the pieces of the video file from the Peers using the Data Channel API. It also uses hashing algorithm which uses too much CPU and power.

In paper [10] they have used video streaming protocols into web applications with the use of WebRTC. They have used a main server and PeerJS sever to establish connection between peers. To establish a connection, the PeerJS server then uses XmlHttpRequests and WebSockets.

In [11], they have focused on the description of the video conference processing system. They used a video conference system using Scale drone. They have used thermal camera and NoIR camera using rassberry-pi for video conferencing.

[12] mentions, They have implemented p2p file sharing on a virtually created network. They have created a torrent file which is encoded in bencode. They have also suggested the use of NPN library for communication gap between RTC and P2P.

III. Proposed Methodology:

In this Section we present the methodology we used for P2P video conferencing system based on WebRTC starting with Overview of the WebRTC Domain.

A] Overview of the WebRTC Domain

WebRTC (Web Real-Time Communication) is a free, open-source project that provides web browsers and mobile applications with real-time communication (RTC) via simple application programming interfaces (APIs). It allows audio and video communication to work inside web pages by allowing direct peer-to-peer communication, eliminating the need to install plugins or download native apps. This communication is extremely fast with low latency as the connection is peer to peer and there's no server in the middle. This technology is developed by Google and currently almost all browsers support it except in iOS platform (Only Safari supports).

B] System architecture

WebRTC follows the semantics client-server organizer with the concept of peer-to-peer communication among the browsers. The connection manages the media path to permit a direct flow between browsers. Network signals are transmitted during the Web Servers that help in modifying, interpreting or managing the signals, as is required by Web Sockets or HTTP. It was noted that the signals between the browser and server are not uniform in WebRTC, where they are part of the application. Web servers can communicate using the standard signaling protocol such as SIP (Session Initiation Protocol) or Jingle. So, basically a property signaling protocol can be used for this goal and establish the peer to peer connections between two or multiple peers.

IV. Conclusion:

In this research, we have implemented a practical experience in the analysis and design of a system architecture for real-time Communication. The architecture was implemented using WebRTC with its inherent features and other technologies in bringing the benefit and experience of a more flexible, speedy and cost effective real-time communication of messaging, video/audio conferencing and file sharing to all Internet users. WebRTC technology will be available through user's browsers to minimize installation and use of plugins in supporting communication, it will also improve the security of multimedia content and help developers to create better real-time video communication solutions. Apart from improving user experience, quality of Service, it will also reduce cost of communication, and provide better security of user data and information.

V. Reference:

[1] Zinah Tareq Nayyef, Sarah Faris Amer, Zena Hussain, "Peer to Peer Multimedia Real-Time Communication System based on WebRTC Technology", International Journal of Engineering & Technology, February 2019, pg. 126-130

[2] Khalid Ibn Zinnah Apu, Nafiz Mahmud, Firoz Hasan, Sabbir Hossain Sagar, "Peer to Peer video conferencing system based on WeRTC", International Conference on Electrical, Computer and Communication Engineering, February 2017, pg. 557-561, IEEE-2017

[3] Cui Jian, Zhuying Lin, "Research and Implementation of WebRTC Signaling via WebSocket-based for Real-time Multimedia Communications", 5th International Conference on Computer Sciences and Automation Engineering, February 2016, pg. 374-380, Atlantis Press-2016

[4] Christian Vogt, Max Jonas Werner, Thomas C. Schmidt, "Leveraging WebRTC for P2P Content Distribution in Web Browsers", 2013 21st IEEE International Conference on Network Protocol, October 2013, pg.1-2, IEEE-2014

[5] D.P.Olshefski, D. Saha, Zon-Yin Shae, R.B. Jennings, E.M. Nahum, "A study of Internet instant messaging and chat protocols", IEEE Network, vol. 20, August 2006, pg.16-21, IEEE

[6] Chun-Hung Richard Lin, Huan Zhang, Jain-Shing Liu, Shi-Huang Chen, "Implementation of Secure Web Conferencing", 2020 2nd International Conference on Computer Communication and the Internet, June 2020, pg.12-17, IEEE-2020

[7] Patrik J. Braun, Marton Sipos, Peter Ekler, "On the Performance Boost for Peer To Peer WebRTC-based Video Streaming with Network Coding", IEEE ICC 2017 Communications Software, Services, and Multimedia Applications Symposium, March 2018, pg.56-68

[8] Ajay Kulkarni, Saurabh Kulkarni, "An Open Source P2P Encrypted VoIP Application", (IJACSA) International Journal

[9] Jukka K. Nurminen, Antony J. R. Meyn, Eetu Jalonen, Yrjo Raivio and Ra´ul Garc´ia Marrero, "P2P Media Streaming with HTML5 and WebRTC"

[10] Florian Rhinow, Pablo Porto Veloso, Carlos Puyelo, Stephen Barrett, Eamonn O Nuallain, "P2P Live Video Streaming in WebRTC"

[11] George Suciu, Stefan Stefanescu, Cristian Beceanu, Marian Ceaparu, "WebRTC role in real-time communication and video Conferencing", IEEE Xplore

[12] Rajesh Dukiya, Shubhankar Sharma, Prof. Sriniwas Koppu Madusudhana Rao, "Reliable File Sharing in Distributed Operating System using Web RTC"