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Multi Channeled Peer to Peer Communication

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Abstract— Audio and video conferencing system have been increasing gradually due to enhancement of work productivity and deduction of expenditure. We are implementing a web-RTC based audio and video conferencing system on virtual cloud. Using this system user can experience audio and video conferencing only through web browser. Web-RTC based applications are assumed to be based mostly on peer-to-peer communication, where an instance of the application is talking to another instance. This system will also provide us with point-to-point to multipoint-to-multipoint communication, from normal image quality to high-definition image quality, from still image transmission to moving image transmission, and from one-way transmission to two-way transmission.

***Keywords*—**web-RTC,cloud

I. INTRODUCTION

A live audio and video connection between two or more people at different geographical locations is known as audio and video conferencing. Audio and video conferencing provides a transmission of full motion video and high quality audio between various locations. Nowadays audio and video conferencing is massively used for business purposes. The clients don't need to visit the company at all.

The conventional audio and video conferencing application used to have many terminals and multipoint control units. Multipoint control unit (MCU) is basically a device which is used for setting up audio and video conferencing. So overall performance of audio and video conferencing used to dependent on the multipoint control unit (MCU).

We will be using virtual cloud based conferencing system so that we can avoid the use of multipoint control unit. Virtualized cloud enables flexible resource management in terms of performance and capability through the auto scale up and out function. In conventional video conferencing we need to install the applications on every device we need to connect through. The Web-RTC (Real-Time Communications) allows the web browser itself to be the video conferencing equipment without installing any plug-in. This means Web-RTC video conferencing is platform independent.

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This paper is organized as follows. In section II, we state the related work. The architecture and the methodology of the system has been shown in section III. In section IV we have discussed the protocols. Section V contains algorithms to be used to implement the proposed system. Finally, in section VI we provide our conclusions and the future scope of the proposed system.

II. RELATED WORK

[1] "Web Application for social Networking using RTC" by Nilesh Kumar Pandey and Doina Bein, Department of computer science from California state university, USA.

In this paper they used the concept of real time peer to peer communication (web-RTC) to present web application for interaction and networking between students and instructor.

Their web application server is a question and answer forum along with the networking sites for students to interact with other by posting their questions in twitter styles tweets and have other students or instructors to answer them. They also present that their system performs minimal functionality and in future it can be extended to include data transfer in real time which means files and documents at a time of video and audio chats.

The system can also be extended for encryption to encrypt the video/audio text communication channel to secure communication between two or more users or clients.

[2] "An Implementation of web-RTC based Audio/Video conferencing system on virtualized cloud", by Sunghyun Yoon, Taeheum Na, and Ho-Yong Ryu

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Audio and video conferencing significantly increasing in the different areas like schools, hospitals, defence, business etc.
The web-RTC based audio/video conferencing system become common to all in future.
In addition, in this paper they present a service which can be used in different video conferencing system using web-RTC so developers does not consider a different platform such as windows, android, ios etc., to adapt this service, which means video conferencing service developers can expect cost saving, which improve the satisfaction of clients.

[3] “Using web services for web-RTC signalling Interoperability”, by Baser daldal, Ibrahim Bilgin Dogac basaran, selin Meta from Netas Telecommunication Inc. Istanbul, Turkey.
This paper focus on exchanging of peer media description, not on protocols which are used between web-RTC client and gateway, and similarly between gateway and VoIP network.
For solution of all web-RTC gateways, the standardization work should contain all the details about the authentication, web service, URL, response format and REST.
They used web service technology to define a language and a vendor agnostic protocol between two different systems interacting with each other.
At a time when web-RTC gateway implements this solution, clients using web-RTC application are able to communicate with each other.

[4] “P2P Media streaming with HTML5 and web-RTC”, by Jukka K. Nurminen, Antony J. R. Meyn, Eetu Jalonen, Yrjo Raivio and Ra´ul Garc´ıa Marrero
Department of Computer Science and Engineering, Aalto University, Finland.
To perform peer to peer audio/video streaming between browsers without the use of additional plug-in, HTML5 is used.
Due to challenges in limitation in current browsers implementation along with performance issue required more study to understand how much their limitations are fundamental in browser technology.
In this paper their work doesn’t end by just implementing a peer to peer solution for media streaming. They also implemented a performance measures over various platforms including mobile devices. Mobile devices ha limited CPU and bandwidth capacities so they implemented a well optimized application for the search resources.

III. METHODOLOGY

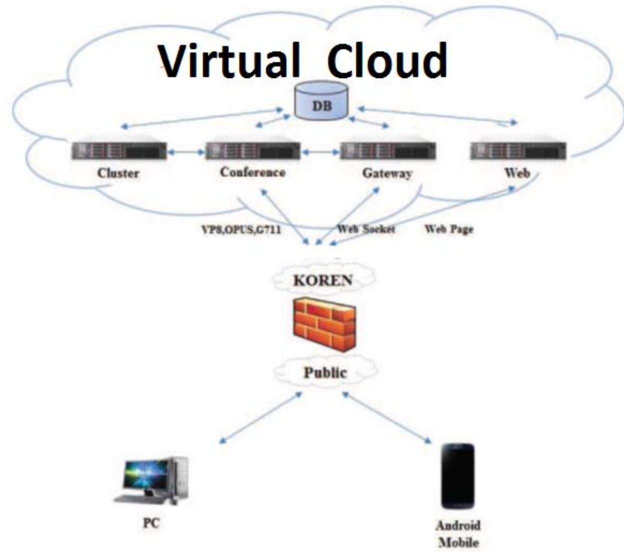

Figure 1. System Architecture

The working processes of the system is as follows:

1) Client login into the system with unique id and password.

2) Server verifies the entered data and opens the dashboard for the client.

3) We are using virtualized cloud as a base infrastructure, as it is possible to provide video conferencing regardless the failure of physical system.

4) The database server is responsible for storage of data generated during the whole process.

5) Cluster server used for clustering of all the servers for the load balancing during video conferencing

6) The conference server used for adding more number of clients for the video conferencing purpose.

7) The gateway server is responsible for account authentication of the web-RTC client, call management, and communication channel management including multicasting agent management, and video control such as mixing, separating video stream.

8) The web server is a kind of portal server that provides the user interface for conference service. It provides various features for conference service such as viewing user information access, bulletin boards, announcements, meeting room creation, Modify, delete, and so on.

IV. PROTOCOLS

Some of the many protocols used are as follows:

1. STUN (Session Traversal Utilities for NAT)
2. TURN (Traversal Using Relays around NAT)
3. SIP (Session initiation protocol)

Description of few important protocols/modules are:

- STUN (Session Traversal Utilities for NAT): STUN is a lightweight protocol that allows applications to discover the presence and types of NAT between them and the public Internet. STUN is a client-server protocol. A STUN client sends a request to a STUN server to discover its public IP and ports, and the STUN server returns a response.
- TURN (Traversal Using Relays around NAT): TURN is a protocol that assists in traversal of network address translators (NAT) for multimedia applications. It may be used with the Transmission Control Protocol (TCP) and User Datagram Protocol (UDP).
- SIP (Session initiation protocol): It is an application level signaling protocol used for voice over IP (VoIP) and video conferencing over IP network.

V. ALGORITHMS

The algorithms that are used for the implementation of the system are:

(i) (TFRC)
(ii) Google congestion control(GCC)

- TFRC algorithm: TFRC is designed to give the best performance for applications that use a fixed segment size and vary their sending rates in response to congestion. However, for applications that do not use a fixed segment size, such as video applications, the TFRC perhaps gives less performance because these applications vary their sending rates according to the needs of the application. The Google Congestion Control sender-side implements TFRC to estimate the sending rate based on the following equation:

$$X = \frac{s}{R\sqrt{\frac{2bp}{3}+(t\_RTO(3\sqrt{\frac{3bp}{8}})p(1+32p^2))}}$$

X: sending rate in byte/second
S: packet size in bytes
R: Round trip time in second
P: loss event rate between 0 and 1
t_RTO: TCP retransmission timeout values in a second
b: number of packets acknowledged by a single TCP

- GCC Algorithm: Web-RTC uses the Google Congestion Control (GCC) Algorithm for controlling congestion. GCC runs over the User Datagram Protocol (UDP) where the audio/video frames are encapsulated in Real Time Protocol (RTP) packets. The congestion control is applied only to the video streams since the audio streams bitrate are considered negligible

$$As(i) = \begin{cases} max\{S(i), As\,(i-1)(1-0.5p)\} & p > 0.10 \\ As(i-1) & 0.02 < p < 0.10 \\ 1.05(As(i-1) + 1\text{kbps}) & p < 0.02 \end{cases}$$

S(i) is the TCP throughput at time i that is used by TFRC, and p is the packet loss rate. The relationship between the packet loss rate and the estimation of the sending rate in above equation can be summarized as follows:

1) If p is larger than 10%, then the sending rate is decreased.
2) If p is less than 2%, then the sending rate is increased.
3) If p is between 2% and 10%, then the sender maintains the previous sending rate.

In Web-RTC, the receiver-side monitors both the video stream and changes in frame delay. According to the receiver side utilizes an overuse detector and estimates the receiving rate based on below equation. In this equation, Ar(i-1) is the previous receiving rate estimate, η is the receiving rate increase factor, R(i) is the current incoming rate, and α is the incoming rate decrease factor. α is a fixed value (normally chosen between 0.80 and 0.95) while the receiving rate estimate is constrained by the following condition: Ar(i) < 1.5 × R(i)

$$Ar(i) = \begin{cases} \eta Ar(i-1) & \text{Increase} \\ Ar(i-1) & \text{Hold} \\ \alpha\, R(i) & \text{Decrease} \end{cases}$$

VI. CONCLUSION ANF FUTURE SCOPE

Our system addresses one of the major problems in the business world – COMMUNICATION. With the dynamic increase in globalization, dependence on the platform on which the communication takes places, proves to be a huge barrier. Our system has successfully removed this barrier and we have achieved the motive of providing flawless P2P video communication over the net.

Future research will attempt for modifications in the signalling servers which can improve the efficiency of the video calls over lower bandwidths. With higher internet

speeds there is a possibility of increasing the number of users in the group chat. For security purposes, a module requesting username and password can be integrated.
Text chat in addition to video and audio can be incorporated.

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Mr. Kundan Pitroda is currently pursuing his Bachelor of Computer Engineering from Savitri Bai Phule Pune University. His interest in WebRTC made him do research work that he presents in this paper.

Mr. Prashant Yadav is currently pursuing his Bachelor of Computer Engineering from Savitri Bai Phule Pune University. He actively studied the protocols for real time communication as a part of course work and made his contribution to this research work.

Mr. Pratik Patil is currently pursuing his Bachelor of Computer Engineering from Savitri Bai Phule Pune University. Being involved in research web based real time communication he presents his work in this paper.

Mr. Suchit Kumar is currently pursuing his Bachelor of Computer Engineering from Savitri Bai Phule Pune University. His interest in internet based communication drives him to present his research on the same in this paper.

