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Multi-Client Real Time Communication using Browsers

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ABSTRACT: Real time communication has become the need of an hour. The increasing demand for real-time and instant messaging applications has led to the development of various real time applications. The motive of our project is to build a hassle-free medium for smooth communication for users all around the world.

We have developed a website where we would provide facilities like video conferencing, audio calling and group chat. As a result, users don't have to install any software or additional plug-ins required for real time communication. This eases the job of users. The accessibility, flexibility and scalability makes our project different from the current systems available.

KEYWORDS: VoIP, SIP, NAT, API, SIP Trapezoid.

I. INTRODUCTION

Real time communication is a direct path between the users who are communicating in real time; the link might contain intermediate nodes and the users can exchange information within no time. Real time communication has the potential to be leveraged in the following aspects: Live video conferences, VoIP (Voice over IP), Internet chatting, Mobile phones communication in conventional sense.

However real time communication can only be done through desktop or mobile applications. These applications need proper APKs and plug-ins and have to be compatible with the operating system and platform of the devices running these applications.

We have developed a real time multi-client communication system which can be used through browsers on any mobile or standalone devices. The user doesn't have to worry about the requirements related to their operating system or the platforms. Three basic functionalities include video conferencing, group chat and audio calling.

II.RELATED WORK

WebRTC:

According to the current scenario, a Web server is present in the path which facilitates the media flow. However, WebRTC is a further extension of the client-server connotation by introducing a peer-to-peer communication link between the browsers. This extension has drastically reduced the overhead on webservers. The direct media flow between the clients would enhance the current real time technology. The following WebRTC trapezoid represents the signalling and media flow:

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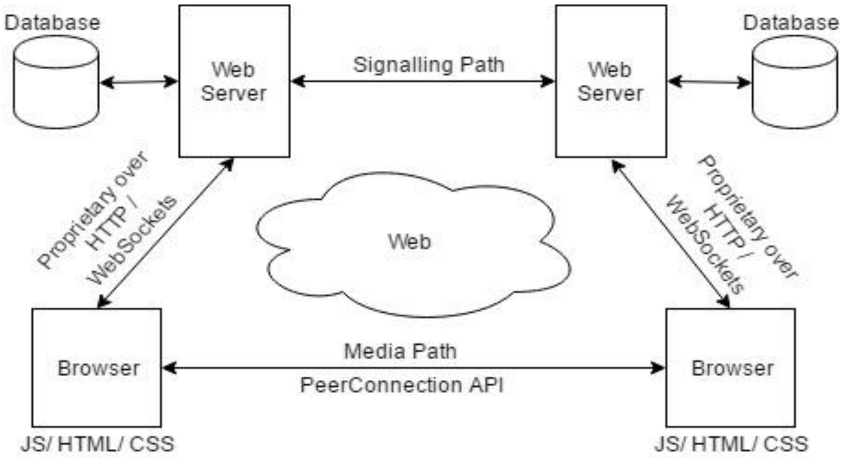

The WebRTC trapezoid has the browsers, web servers, database and internet connection. The user logs into the web application present in the browser. The WebRTC framework allows the users to use their own signalling mechanism. Signalling can happen over websockets, HTTP, SIP or Jingle. This flexibility renders high level usability to developers and clients.

As soon as signalling is established, the media directly starts flowing between the peers. This is one of the greatest advantage of WebRTC.

WebRTC in the Browser :

A WebRTC application relies on its standardized API for the proper execution of real time browser. There are 2 ways for an API to interact with a web browser. They are as follows : proactive way (eg. To determine browser capabilities) and a reactive way (eg. To receive browser-generated notifications) way.

The WebRTC API provides a wide range of functions. They are as follows: signalling and connection management, encryption and decryption negotiation techniques, media type selection, firewall and security support.

Real Time Communication in the Browser:

Let us assume a real time video and audio call taking place between browsers. In such a situation, there is direct flow of media between the two browsers. The media flow is first established by negotiation and complex series of interaction. Following are the entities involved:

- The caller and callee browser and Javascript application in the form of APIs.
- Application provider i.e. Web server

III. PROPOSED SYSTEM

The proposed system is introduced to overcome the drawbacks encountered while using the existing desktop applications meant for real time communication. The system is capable of handling multiple clients at the same time for doing video and audio conferences and group chat.

We have used the WebRTC protocols and programming interfaces to implement our system. The APIs provided by the WebRTC are used for establishing peer-to-peer connections and media stream management.

We have used HTML for the web design and java script for the backend implementations. Php and MySQL database is used to keep track of the registered users and the newly joined users. The database even keeps a track of the ongoing conferences and group chat.

Functionalities provided:

- The users can get themselves registered on our site.
- The registered users can login to get access to the services.
- The logged in users can select if they want to do video conference, audio conference or group chat

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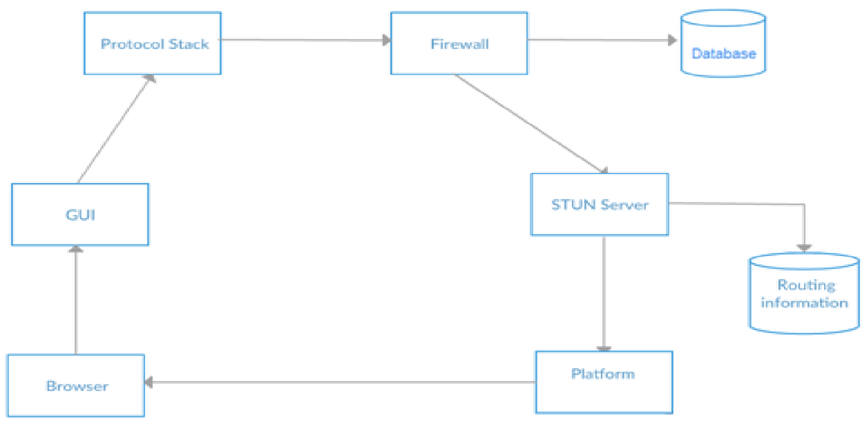
- The ongoing conferences are viewed on the site. The user can join the conferences if he/she wishes to by clicking on the join button.
- Else the user can start his/her own conference.
- The site asks users' permission to access the media devices like camera and microphone.
- The users can end or leave the conference by clicking on the end conference button.
- The users can logout if they want to.

Architectural description:



System contains of following components:

1. Browser
2. GUI
3. Protocol Stack
4. Firewall
5. Users
6. STUN browser
7. Routing information

Description for each component:

1. **Browser**
   The user would open the browser. The browser is platform independent and can be Google Chrome, Mozilla Firefox and Internet Explorer, etc.

2. **Graphical User Interface**
   It consists of text boxes for entering the login parameters, audio and video elements for viewing the streams of users who are communicating and buttons.

3. **Protocol Stack**
   The protocol stack is the framework which is responsible for transparent peer-to-peer connection of users, abstraction and encapsulation of data streams for secure delivery.

4. **Firewall**
   Firewall is a network security system that acts as an intermediary between the systems and the world wide web.

5. **Database**
   The information i.e. login credentials and data pertaining to ongoing conferences is stored in the table's database.

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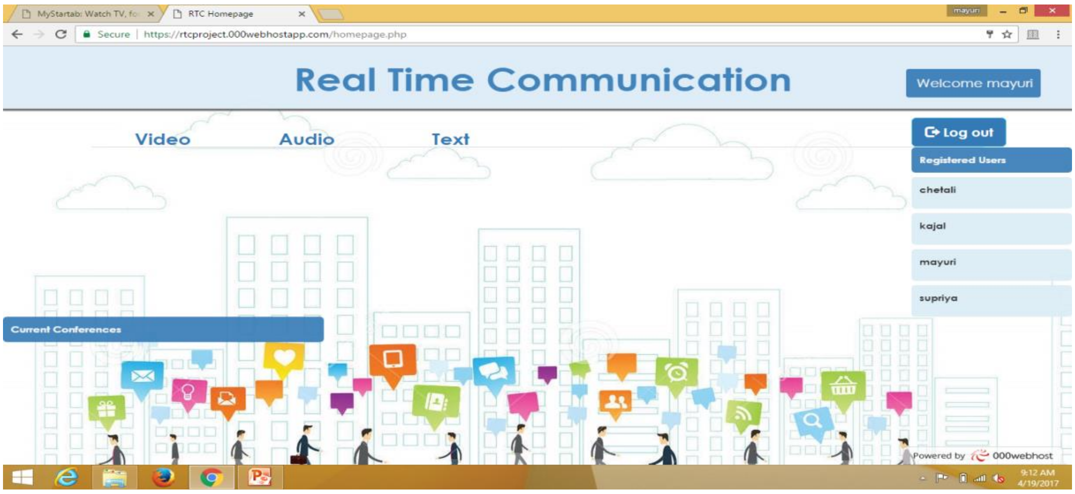
6. **STUN browser**
   When the user tries to connect to another user not in local network area, STUN server comes into picture. It maps the private address to public WAN address and helps in routing.

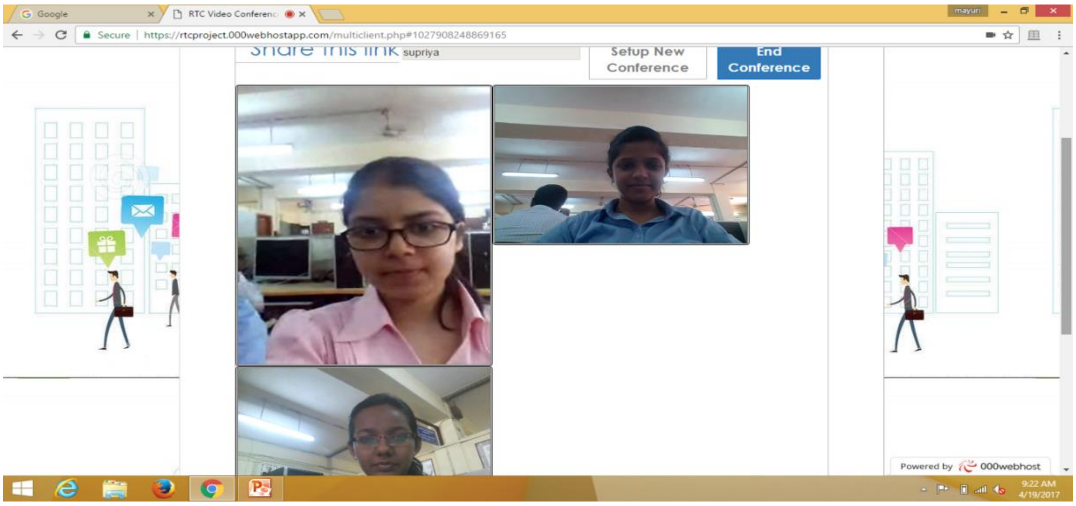
7. **Routing information**
   Routers stores the routing tables that contain the shortest delivery path for data transmission and also the number of hops to travel more to reach the destination.

IV. RESULTS

1. **Homepage**



2. **Video Conferencing**



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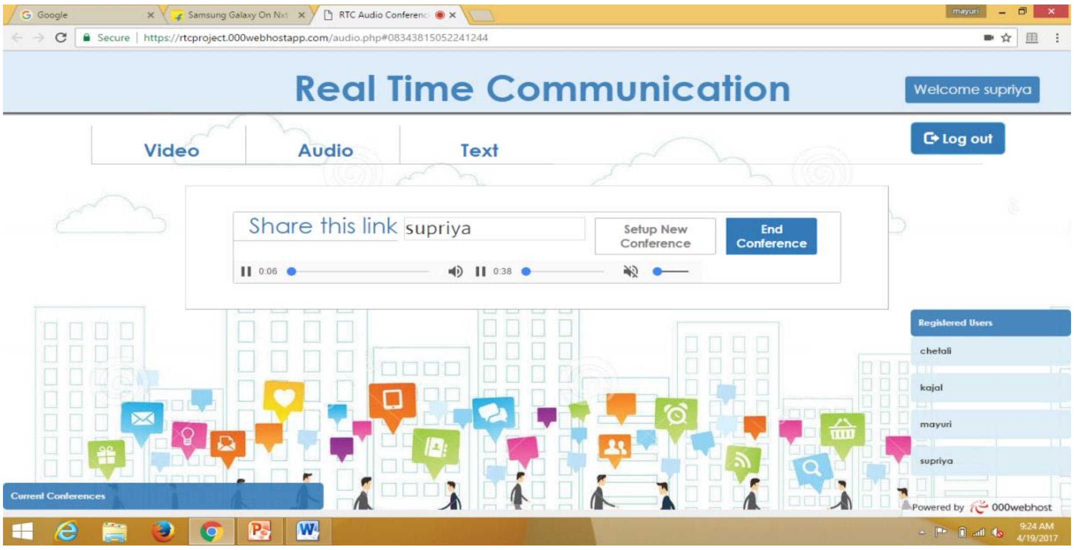
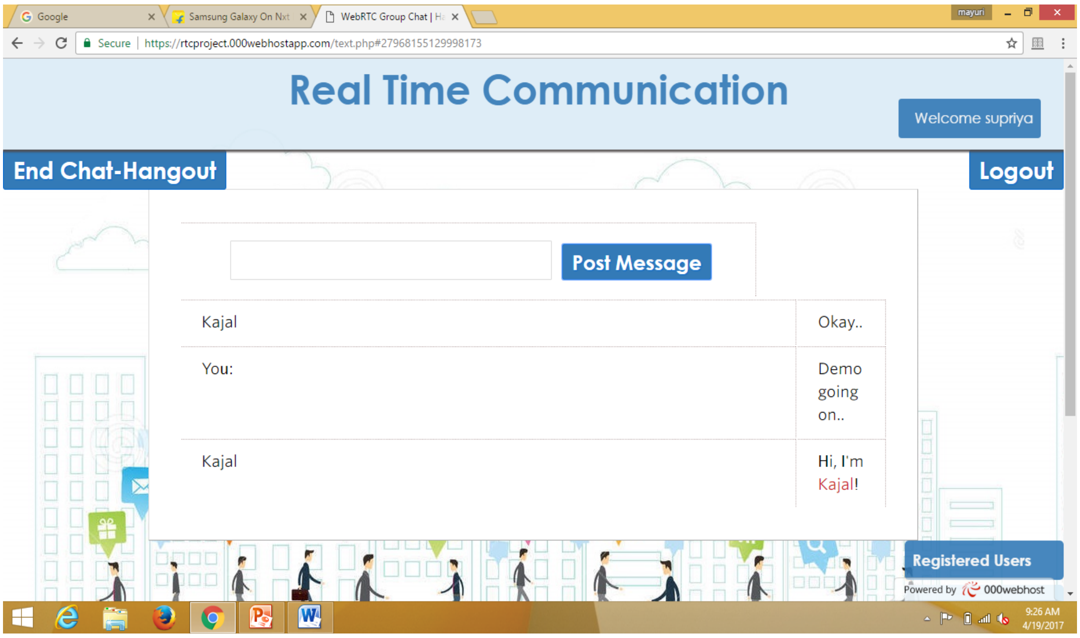
3. Audio Calling



4. Group Chat



V. CONCLUSION

We have successfully designed a website which serves the purpose of real time communication. Our website would enable people to communicate with their friends, relatives and colleagues all over the globe. The objective of our website is to present users an easy access to services of real time communication. Our website has in-built APIs which handle the media capture of local and remote users. Moreover, users need not worry about their device's storage space or compatibility with underlying system as they have to just access the website. The simplicity, uniqueness of our website makes it different from current desktop applications.

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