Peer to Peer Multimedia Real-Time Communication System based on WebRTC Technology

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Research paper

Peer to Peer Multimedia Real-Time Communication System based on WebRTC Technology

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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. WebRTC (Web real-time communication) is a futuristic technology that makes real-time communication capabilities in audio, video and data transmission possible in real time communication through web browsers by using JavaScript APIs (Application Programming Interface) without the plug-in. In this paper, we proposed a web peer to peer real time communication system that allows users to communicate with high-speed data transmission over the communication channel using WebRTC technology, HTML5 and use Node.js server address. The result shows that the system is stable, fully functional, safe and can use in a practical network to transmit and receive multimedia data in real time between users.

Keyword: WebRTC, Nood.js server, HTML5, JavaScript APIs, multimedia.

1.1 Introduction:

The extreme development of modern technology in the transfer of multimedia and computer communications requires high-quality communication among people. To satisfy this growing need, real-time communication process was implemented [1].

Real-time communication (RTC) is a new standard and industry-wide effort that expand the web browsing model that allows access information area like social media, chat, video conferencing, and television over the internet, and unified communication, Users of these systems can view, record, remark, or video contents flow.

The open source project WebRTC enabled users of these systems to view video content or record, comment on or stream it to achieve real-time communication between web browsers [2][3]. 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 [4]. WebRTC is an open sourced standard from Google in 2011 for the multimedia web systems. At this time, it is included in Web browsers like Firefox, Chrome, opera, etc. In future, this technology will be included in all browsers without plug-in components, which can end the threat of virus while providing interactive communication [5][14].

In this paper, we introduce a system that afford multimedia transmission service such as video and audio, 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.

1.2Real-Time Communication with WebRTC

A. WebRTC Architecture

WebRTC follows the semantics client-server organizer with the concept of peer-to-peer communication among the browsers as shown in Figure (1). 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 it required by WebSockets or HTTP [2,7]. 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 [7]. Otherwise, a property signaling protocol can be used for this goal.

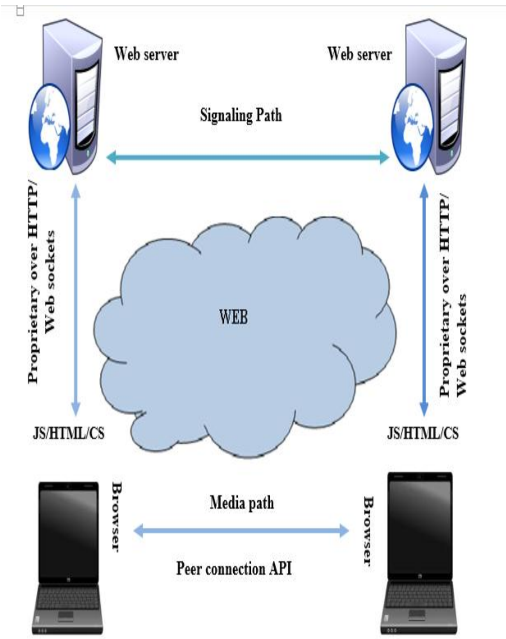


Figure (1) The WebRTC Trapezoid

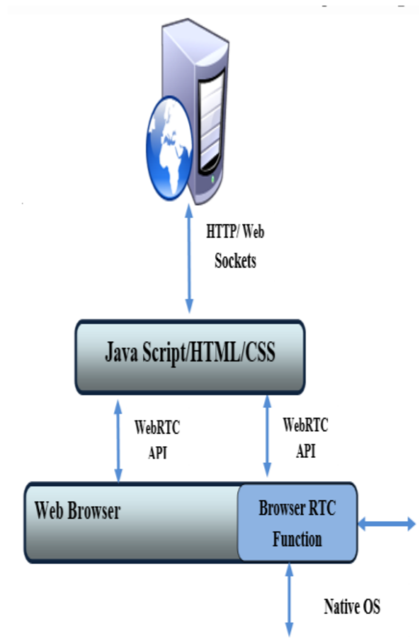


Figure (2) Real-Time communication in the browser.

B. The Browser in WebRTC Function

WebRTC application (usually written using HTML5, CSS, and JavaScript) is interacted with several web browser by the WebRTC API (Application Programming Interface), which allow proper use and management of the Real-time browser function, as shown in Figure (2). Using fundamentally WebSocket technology, signals can be exchanged between online participants [6, 8].

The WebRTC web application interacts with browsers that use both standers application of API and WebRTC application, proactively (e.g. inquiry browser competency) and an interactive way (e.g. receiving a cross-browser notification) way [4, 3].

Real-time imaging communication (Like a video, and audio call) among two browsers includes direct media streaming among the two browsers, within the media path parley and create an instance of multifaceted interaction through the following entities [9]:

- The caller and caller browser are using JavaScript application (using JavaScript API).
- The JavaScript application for caller and application provider (typically a web server)
- The JavaScript application for called and caller browser (through the application browser JavaScript API).

C. WebRTC API

The W3C (World Wide Web Consortium) WebRTC API allows JavaScript application to take the benefit of the real-time capabilities of an uncommon browser’s. The real-time browser function as shown in Figure (2), which was implemented at the browser center, provides the necessary function for setting the video, audio, and data channel required. Three basic concepts that been relied upon in the design of the API (Application Programming Interface): MediaStream, PeerConnection, DataChannel [10].

1. MediaStream

MediaStream represents media streaming from a local media device like a microphone, webcam [11][4]. The web application must request user access to make and utilize a local stream through “GetUserMedia ()” function.

2. PeerConnection

RTC PeerConnection reads the output data from MediaStream and creates the connection between two users. To create the peer-to-peer connection, the STUN and TURN protocols are used by ICE (Interactive Connectivity Establishment) structure as its core of NAT (Network Address Translator) where these protocols are provided by Google [10].

3. DataChannel

The RTCDataChannel API is a bi-directional data channel between two peers provides the possibility of exchanging random data among them. Each RTCDataChannel provides the following [12]:

- Reliable or unreliable transport of messages.
- In-order or out-order transport of messages.

D. How to Use WebRTC:

The four procedure of setting up WebRTC is illustrated in figure (3) [13].

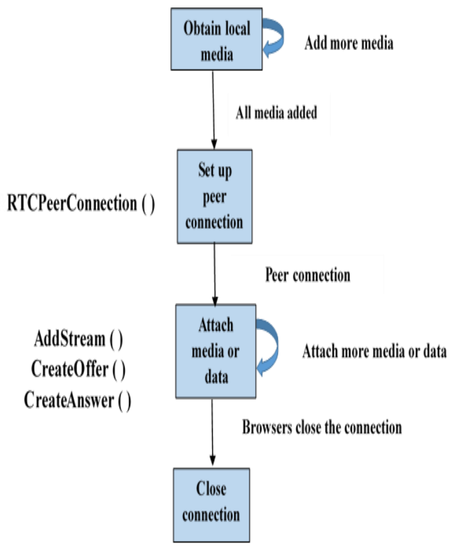


Figure (3) WebRTC Session Establishment, API View

1.3 System design

The system design to provide video, audio and chat communication with the mechanism of user identification and discovery of other users of the system without installation or setup procedures as illustrated in Figure (4). The proposed system is divided into several parts:

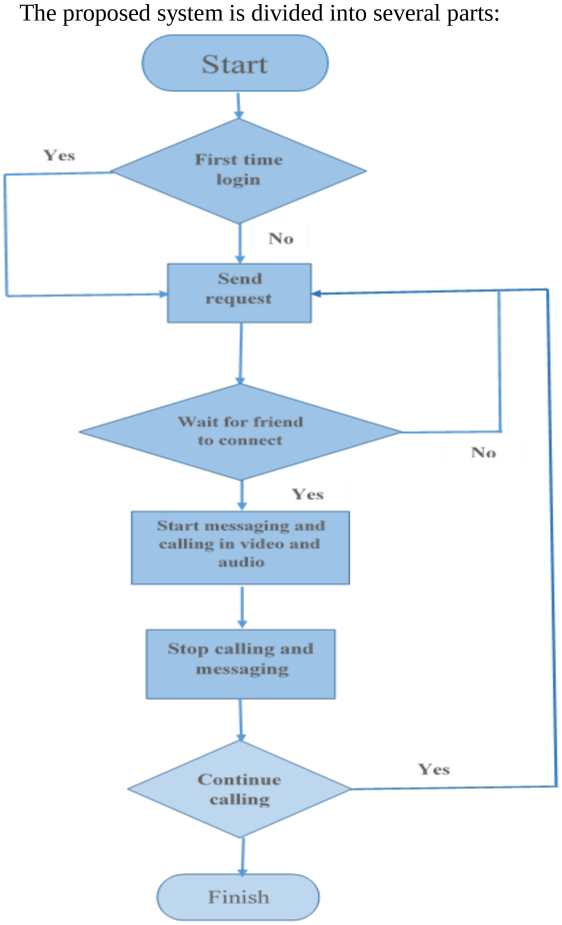


Figure (4) Flowchart of the proposed system

- All server-side and client-side information that transmitted is secure and encrypted using high secure protocols server which is used to distribute the content from different remote sources. Before the flow of data begins, the server application will verify user session and licenses for each demand. If the session has licenses for requested content and is valid, the flow will begin.
-
- The information contained in any request is noted by the server, and on that basis informs the user of incoming messages and calls. This server listens for calls or calls during the requested request creation process, information about the paths between the users and request parameters required for each call participant will be transferred by the server. Two servers connected together; Apache server for Website and Node.JS server for implementing WebRTC technology. Videos and images files that uploaded by a user to the server will be compressed while maintaining high resolution of uploaded files.
- Create a Database (using MySQL) consisting of many tables used to store information about users and their activities such as video calls, user logins, and sessions. All important information of users that stored in Database is encrypted by using high complex encrypt methods to protect Database and Website from outside attacks like XSS attack.
- Using a HTML5, CSS3, JavaScript and jQuery, JSON, AJAX Techniques on client-side and used PHP 5.5, PHP: PDO for SQL instructions and Node.JS on server-side.

1.4 Experimental Result

A. User interface

The Web page of Website which is named "Skyline" consist of Log in the panel for entering User Name and Password as referred by the first and second arrow and the third arrow refers to "Log in" button. In the case that user forgotten password than by click "Forget Password?" to get help to retrieve a password as referred by the fourth arrow. If the user is not subscribed, so, there is a need to go to "sign- up" web page of the website by clicking "Join us" link where the fifth arrow referring to it as shown in Figure (3).

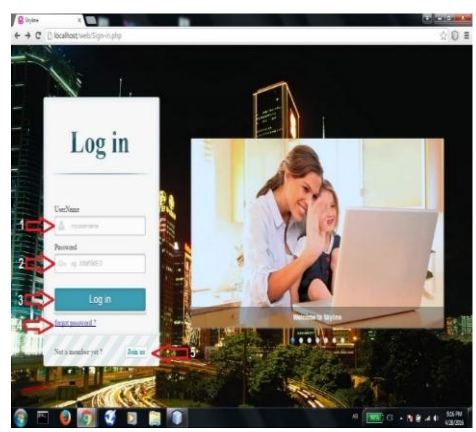


Figure (5) User interface of Website.

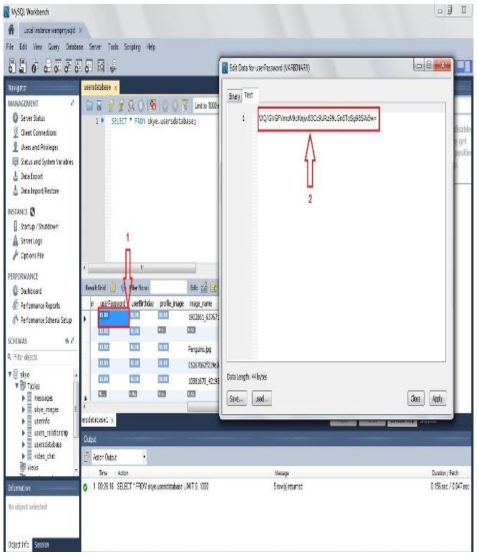


Figure (6) Encrypted information in the database.

B. Home Page of Website

After clicking “Log in” in “sign in” the user will redirect to the home page of Website as shown in figure (5).

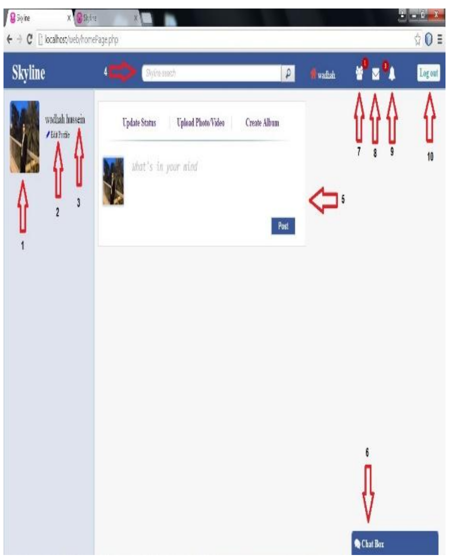


Figure (7) Homepage of user “wadhahhussaain”.

The arrows in Figure (5) indicate in order:-

1. Edit image2. Edit profile link
3. Username 4. Search field that can use to find a new friend.
5. Postbox 6. Chatbox
7. New friend requests notification
8. New messages notifications
9. Post notifications

C. Settings Web page

Edit profile in “home page” will redirect the user to another web page named “Settings” which enable the user to change basic information like username, password, image and another information as shown in Figure (6). After clicking “Confirm” the information will send to the server to ensure that it complies with the conditions of Website. One of the website features is the ability to compress files like profile image which selected by the user.

Figure(8)Settings of the web page.

D. Frind Requests & Chat Box

If the user “wadhahhussain’ click on a friend name “ahmedali” the chat box between them will appear and they can send messages to each other as shown in Figure (6). Messages with gray color from user “A” and messages in pink color from user “B” on the perspective of each user as shown in Figure (7) and Figure (8).

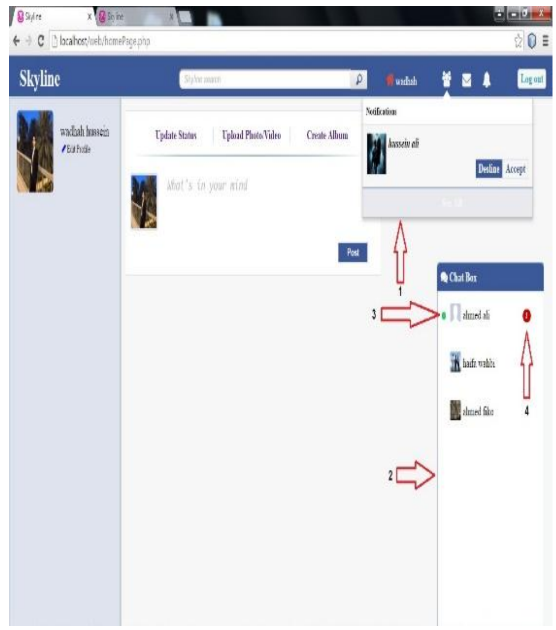
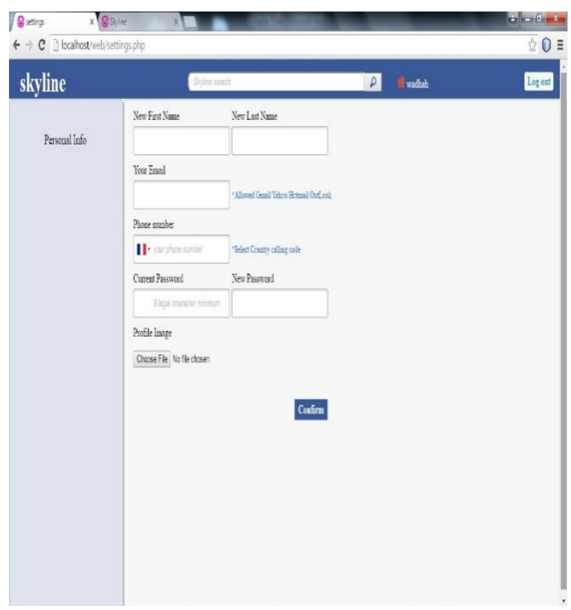


Figure (9) Friend request & Chatbox.



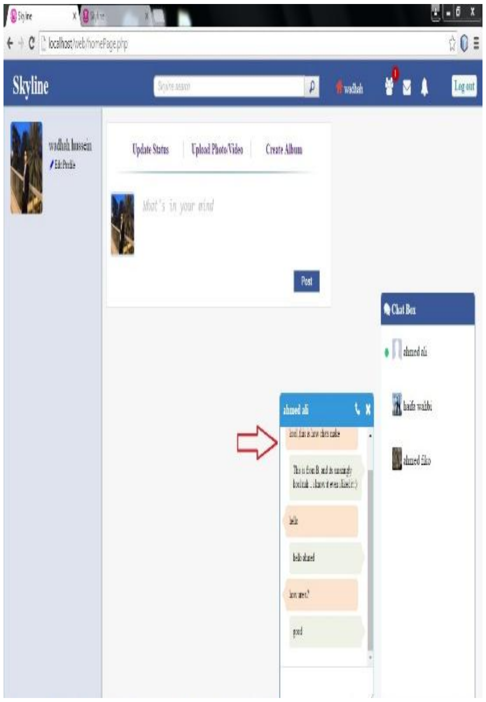


Figure (10) user “Wadhah Hussain”.

Figure (11) user “ahmedali”.

If user “Wadhah Hussain” click on call link where first arrow referring as shown in Figure (10) and second arrow refers to “Contact Call” box and then he waiting for “ahmedali” to accept the offer as shown in Figure (11).if the user accept the call the Website will open video/audio chat window for each user as shown in Figure (12) where first arrow refers to receiving video/ audio call and second arrow refers to sending video/audio call while third arrow refers to room name which both users communicate through it.

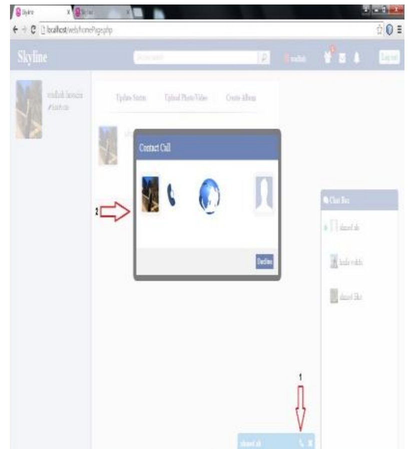


Figure (12) Contact call.

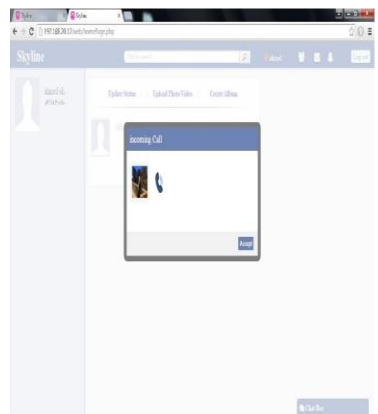


Figure (13) Incoming call.

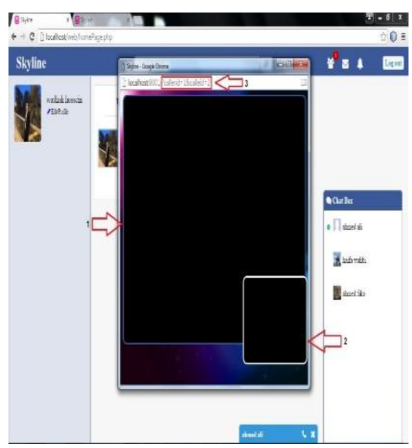


Figure (14) Video/Audio call of user ‘wadhah Hussain”/ Image black deliberately.

1.5 Conclusion

The use of WebRTC technology enabled the implementation of secure and high data transmission between users as peer-to-peer or peer-to-group connection in real-time communication, therefore, anyone can create their own Webpage or application such as real-time sharing files, real-time communication environment as messaging chat or video/audio conferencing this opened the way for programmers and developers to enter the actual job market and compete with social media owners. Using WebRTC technology allows us to create a webpage with the most powerful features, allowing each user to connect to another via Text messages, video/audio call by using simple JavaScript APIs and Node JS. The server associated with Google STUN and TURN server.

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