ADVANCE WEB BASED APPLICATION FOR MULTIMEDIA USING P2P CONNECTION

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Abstract— *The Real-Time Communications system is to develop a Web Browser (social networking) site through which users can communicate with friends at different location, by using Web RTC (Real-Time Communications) technology, so that can be enables audio/video streaming and message sharing between browser to browser. As a set of standards, Web RTC provides any browser with the ability to share application data and perform teleconferencing peer to peer connection, without the need to install any plug-ins or third-party software or apps. Currently in development are the Network Stream API, which represents an audio or video stream, and the Peer Connection API, which allows two or more users to communicate browser to browser. Also under development is a Data Channel API that enables communication of other types of data for message chat, real-time gaming, file transfer, and so forth. The proposed system for virtual world approach to the construction and facilitation of communities to practice within the context of social innovation due to the limited computational power and network resources machines are often implemented using massage chat, Achieved by creating peer-to-peer connection. How ever with the help of new technologies it is possible to create a virtual world in a social network. A virtual world can have anything which a human being can do in his real life. Social network and virtual world both combine it creates a new phenomenon for the users to get live message passing connection between them.*

Keywords— P2P; WebRTC; Node.js; MongoDB; Message chat

1. INTRODUCTION

The Real-Time Communications system is to develop a Web Browser (social networking) site through which users can communicate with friends at different location, by using Web RTC (Real-Time Communications). The Webrtc technology using to developing the audio/video and message chat for browser to browser communication , Node.js as a server side response and request can be passing the client and server interact then Mongodb is a data base using to store the more number of data, so the data can be stored for different collection field . socket connection as input and output response /request passing for two user as communicate at peer to peer connection, RTC peer connection is two user directly communicate and message sharing but not responsible for server.

This project can be alleviated by using WebRTC, RTC DataChannel API to transfer data directly from one peer to another peer connection. In this article we will cover the basics of how to sing in set up and use adio/video & data channels, as well as the common use cases on the web today.

2. RELATED WORK

A. *SIP APIs for Voice and Video Communications on the Web*

Carol Davids, We propose a voice and video communication architecture on the web and compare it with other alternatives. Using a separate adaptor application has several advantages compared to modifying the web protocol and language or using the Flash Player plugin. In particular, the architecture is platform independent, can be easily implemented, and is flexible enough to accommodate new codecs, application protocols (e.g., SIP/RTP).

B. *Flash-based Audio and Video Communication in the Cloud*

Kundan Singh , We have described the various challenges of web-based audio and video communication, and presented in detail a generic API using our Flash application. We showed how to implement several web video communication scenarios. This paper presented both the advantages and disadvantages of using the Flash Player plugin. The proposed API can be used as a baseline for future extensions to HTML for video communication.

C. *Analysis of RTCWeb Data Channel Transport Option*

Hasan Mahmood , We proposed IETF and W3C are working together on extending the web architecture. In particular , RTC Web project aims to allow browsers to natively support interactive peer to peer communications and real time data collaboration. The fundamental goal of RTC Web is to specify a set of protocols that support transportation of non-media data, such as data sharing along the most direct possible path between RTC Web clients.

D. *WebRTC: APIs and RTCWEB Protocols ofthe HTML5 Real-Time Web*

Alan B. Johnston , WebRTC interoperating with SIP. The Web Server has a built-in SIP signaling gateway to allow the call setup information to be exchanged between the browser and the SIP client. The resulting media flow is directly between the browser and the SIP client, as the Peer Connection establishes a standard Real-time Transport Protocol (RTP)media session with the SIP User Agent.

3. SYSTEM ANALYSIS

A. *Existing System*

The existing system is that to establish a video chat connection two users in a network. Supports conference calls up to 25 people at a time. Also supports video chat between two people for free. Screen sharing and group video calling is available for Premium subscribers between a maximum of 10 people. Depending on strong he connection , there could be delays in conversation. An Additional plug-in is necessary to be installed. To use Skype from a desktop computer, you just plug in a headset, a specialized a regular phone.

i) Disadvantages of Existing System:

- Does not allow IM outside the Skype network.
- On-time communication depends from being logged on.
- Not only does the connection have something to do with quality
- Services are depending on software and hardware capabilities of user and recipient.
- Charge to use phone line services.

B. Proposed System

A project known as Web RTC, for browser based real-time communication. Get User Media, which allows a web browser to access the camera and microphone and to capture media. RTC Peer Connection, which sets up audio/video calls. RTC Data Channels, which allow browsers to share data via peer-to-peer.

i) Advantages of Proposed System:

- Reduce the time.
- No additional plug-in required.
- WebRTC enables users to participate in a communications experience as delivered by any website without downloads, registration or general cost.
- Advanced voice and video quality.
- WebRTC makes it simple to connect web browsers and setup video calls.

4. TECHNOLOGY USED

A. Web RTC

Web Real-Time Communications (RTC), adds few functionality to the web browser. For the first time browsers will interact directly with other browsers, resulting in a number of architectures including a triangle and trapezoid model. The media capabilities of Web RTC are state of the art, with many new features. The underlying standards of Web RTC are being developed by the World Wide Web Consortium (W3C) and the Internet Engineering Task Force (IETF).

B. The Web Browsing Model

The basic model of web applications is shown in Figure 1 Transport of information between the browser and the web server is provided by the Hyper-Text Transport Protocol, HTTP which runs over Transmission Control Protocol, TCP or in some new Web RTC: APIs and RTCWEB Protocols of the Real-Time Web4implementations, over the Web Socket protocol. The content or application is carried in Hyper-Text Markup Language, HTML, which typically includes JavaScript and Cascading Style Sheets (CSS).

C. The Real-Time Communication Function in the Browser

Figure 2 shows the browser model and the role of the real-time communication function. The lighter block called “Browser RTC Function” is the focus . The unique nature and requirements of real-time communications means that adding and standardizing this block is non-trivial. The RTC function interacts with the web application using standard APIs. It communicates with the Operating System using the browser.

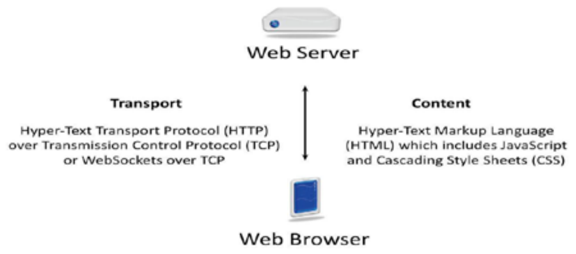

Fig 1 Web Browser Model

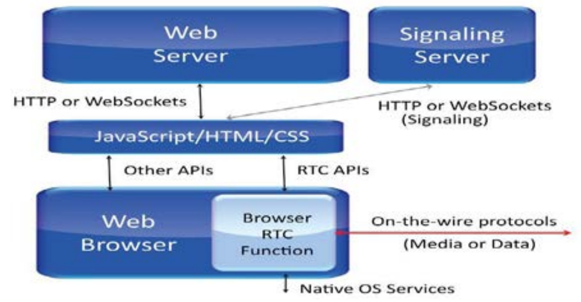

Fig.2 Real-Time Communication in the Browser

D. MongoDB

We begin our journey by getting to know the basic mechanics of working with MongoDB. Obviously this is core to understanding MongoDB, but it should also help us answer higher-level questions about where MongoDB fits.

i) Feature: MongoDB is intended to be a general-purpose database, so aside from creating, reading, updating, and deleting data, it provides an ever-growing list of unique features.

ii) Indexing: MongoDB supports generic secondary indexes, allowing a variety of fast queries, and provides unique, compound, geospatial, and full-text indexing capabilities as well.

iii) Special collection types: MongoDB supports time-to-live collections for data that should expire at a certain time, such as sessions. It also supports fixed-size collections, which are useful for holding recent data, such as logs.

vi) File storage: MongoDB supports an easy-to-use protocol for storing large files and file metadata. Some features common to relational databases are not present in MongoDB, notably joins and complex multi row transactions. Omitting these was an architectural decision to allow for greater scalability, as both of those features are difficult to provide efficiently in a distributed system.

E. Node.js:

Before we talk about all the technical stuff, let’s take a moment and talk about you and your relationship with JavaScript. This chapter is here to allow you to estimate if reading this document any further makes sense for you. If you are like me, you started with HTML ”development” long ago, by writing HTML documents. You came along this funny thing called JavaScript.

5. ARCHITECTURE DIAGRAM

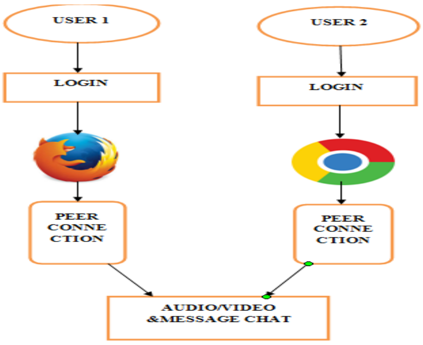

Fig 3 Audio/video and message Architecture

This architecture is peer to peer connecting to the message passing for two different user then. The socket connection as input and output value can be passing the one browser to another browser because WebRTC allows for peer-to-peer video, audio, and data channels. Being able to exchange data directly between two browsers, without any sort of intermediary web socket server, is very useful. The Data Channel carries the same advantages of WebRTC video and audio: it's fully peer-to-peer and encrypted. This means Data Channels are useful for things like text chat applications for P2P message exchanges.

A. P2P Connection

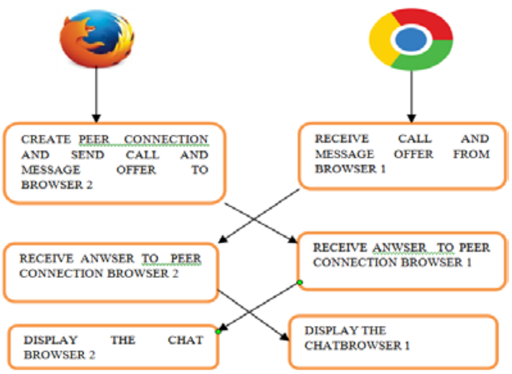

Fig 4 P2P Connection

6. VIDEO AND MESSAGE CHAT

WebRTC is an open-source project enabling plugin-free, Real Time Communications (RTC) in the browser. It includes the fundamental building blocks for high-quality communications such as network, audio, and video components used in voice and video chat applications. Data channel using to implementing message chat . when implemented in a browser, can be accessed through a JavaScript API, enabling developers to easily implement their own RTC web app. WebRTC aims to give the development community access to open, high-quality, real-time communications technology. WebRTC is being defined and developed to offer real-time peer-to-peer communications to the web browser for vice and video & data sharing , taking advantage of real-time protocols and codec instead of defining new one. WebRTC will open the door for a new wave of video & voice, and data sharing web applications. The WebRTC project is incredibly important as it marks the first time that a powerful real-time communications (RTC) standard has been open sourced for public consumption.

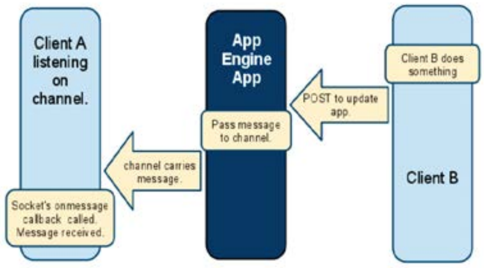

Fig 5 video and message chat

7. CONCLUSION

The implemented, developed and enhanced open source solution represents a voice/video and message chat application that can be used for P2P, live web broadcasting. Other additions can be enabled in the This project has designed an application which satisfies customer request. It also provides a good user interface and a good way of developing better communication. Thus by using this project, conversation can be implemented between loved ones, friends and others too. Also, this can be used for business purposes project.

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