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Group Meet – A Web Application for Video Conferencing

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ABSTRACT
Web Conference is a type of videoconference, which is a real-time online event based on audio or video communication. In this paper we have designed and investigated audio/video conferencing as a website Service. The website has been designed based on Web Real Time Communications (WebRTC). The site is dynamic site designed under ASP.net using C# as a programming language, the dynamic content of video conference is JavaScript in which the web server instructs the JavaScript to run certain actions and then the script will return feedback information to the web server. The process of authorization is done by allowing the access to website pages or videoconference page depending on authorized level; password encrypted using encryption technique which used hashed and salted algorithm to protect it from cracking by any types of attack. In addition, a Secure Sockets Layer (SSL) has been used to encrypt all connections between site and client, which provides complete protection for all data transfer operations between the server and the client. The system has been tested in real work for both network and internet and the result show it worked perfectly and the video streaming is based on internet speed and streaming bandwidth.

I. INTRODUCTION

Remaining years, video becomes a crucial media for communications due to the growing in net velocity that allows excessive video streams. Previously, the video was captured and transmitted in analog form. The development in computers and virtual included circuits led to the digitalization of video, and the virtual video ends in revolution within the communication and compression of video. Generally, the process of using the internet to send content material with the aid of encoding it into a number of decodable formats is known as streaming. whilst the transmission is achieved as content is created, the circulation is known as a "LIVE" stream.

Live Video Streaming is a type of video streaming that transmits an email correspondence through a neighborhood region network (LAN) or via the internet in actual time in order that the video and/or audio from transmitter supply may be heard and seen on the receiver aspect through non-public computers, clever phones and cellular gadgets, and so on. real time media conversation (audio, video) between one-of-a-kind client devices, includes one-way communication (streaming) or two-manner communication (video/ audio chat or video conference). stay media video meetings send and receive video and audio between a couple of endpoints. Mainly, the media streaming desires to setup of standalone streaming servers, instilling the correct standalone software in client facet and help to streaming protocols that manipulate the moving the streamed packets. With admire to conferencing and chatting the want additionally to mediation of a consultation manager in-among the clients and the assist of the corresponding session protocols. With appreciate to actual time communicating with the aid of the net, until now HTTP is simply the media streaming fantasy method. With respect to receipt media streaming over the internet, it can be accomplished only via installation of the best 0.33-birthday party software program (browser plug-in) to obtain and manner the media streamed from the server. Finally, the need to media players that provide plugins for browsers to permit audio and video streams to be run over the net.

To be able to combine video conferencing, which is going beyond a fundamental face-to-face setup with a bendy internet browser-based totally groupware system, the video conferencing has to be run in a web browser context too. in comparison to local groupware applications, this implies barriers concerning person interface (UI) layout (e.g., window transparency) and available video processing software programming interfaces

(APIs). rather than conventional approach, in which the audio/video real-time verbal exchange within web browsers are simplest possible thru plugins or 0.33-birthday party software, the internet real-Time communique (WebRTC)7 is a collection of communication protocols and APIs that guide peer-to-peer (P2P) actual-time verbal exchange amongst web browsers. With the P2P competencies delivered through WebRTC, browsers now run far away from the classic customer-server version. The advantage of this shift is that the APIs defined through WebRTC are identical regardless of the underlying browser, working device etc. and are to be had on many systems, particularly mobile device. The work is aiming to develop a video conferencing website that allows users anywhere in the international world to sign up for real-time streaming video chat rooms without installing any software program. These proposed surroundings are in security infrastructure that plays comfortable streaming to users for the prevention of safety threats.

VIDEO CONFERENCING CONCEPT

A video conference (which is also known as video teleconference) is many interactive telecommunication technologies that permit two or multiusers to interact via two-manner audio and video transmissions together. Video conferencing uses video and/or audio telecommunications to let people from exclusive places meet collectively in one chatting region. Video conference is easy as a communication between (factor-to-point) in personal offices or extra human beings (multi-point) in huge rooms at many websites. In addition to audio and video transmission through meeting activities, videoconferencing is also allowing to percentage files which includes files. Video meetings have two predominant types, factor-to-factor video conference which takes location between individuals in-among separate web sites, and multi-factor video convention that make interplay among individuals at three or greater various places. The predominant generation that has been used in a videoconferencing machine is audio/video streams compression in real time. The software or hardware that goes ahead compression is referred to as codec (coder/ decoder). The using of audio modems in the transmission line was allow for using plain old phone device (POTS), in a few low-pace programs includes video telephony, this because of truth it converts the virtual pulses to/from analog waves within the audio spectrum variety. The trendy additives needed for a videoconferencing gadget may be described as follows:

- Video enter: the video input tool such as webcam or video digicam
- Video output: the tool that views the streaming media including tv, laptop screen, or projector
- Audio input: the device that acquired audio consisting of CD/DVD player, microphones, or another audio outlet supply.
- Audio output: this to output audio such as loudspeakers or telephone five.
- Record switch: virtual or Analog phone community, net or LAN.
- Processing tool: the device that processes the overall videoconference operation such as computers, smartphones, pills, etc.

Web Real-Time Communication(WebRTC)

Internet actual-Time conversation (WebRTC) is a framework that lets in peer-to-peer communication between net browsers. The technology in the WebRTC stack and its API: s is presently being standardized via the international huge net Consortium (W3C) and the net Engineering venture force (IETF), and carried out by browser companies together with Google, Ericsson and Mozilla. WebRTC allows browsers to circulation audio, video and arbitrary information directly to each other without the want for an imperative server. This makes it possible to write and run actual-time applications such as video games and communique offerings directly within the browser; there is no need for plugins or platform-particular applications.

The WebRTC has a Voice Engine, Video Engine, and equipment for transport and verbal exchange. This approach that anything associated with media encoding (converting audio and video from one layout to every other) and compression, as well as low-level networking is treated with the aid of the framework. net browsers and different native packages can get admission to the framework through its C++ API. web applications cannot access this low-stage API for safety- and interoperability motives, so web browsers want to offer some other way for developers to use it. The usual manner of doing this is through a JavaScript API. Internet packages can use the standardized JavaScript API to get right of entry to the capability of WebRTC.

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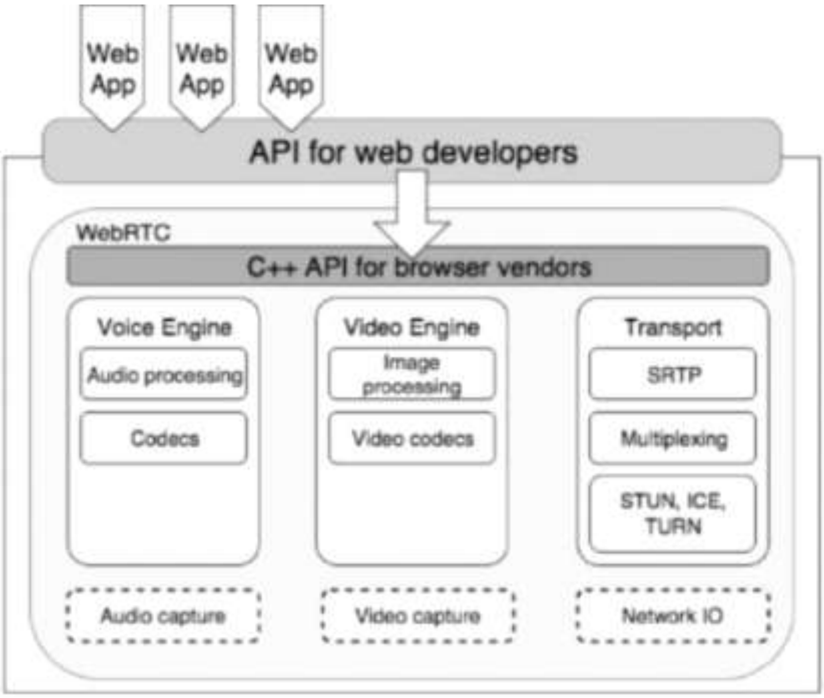
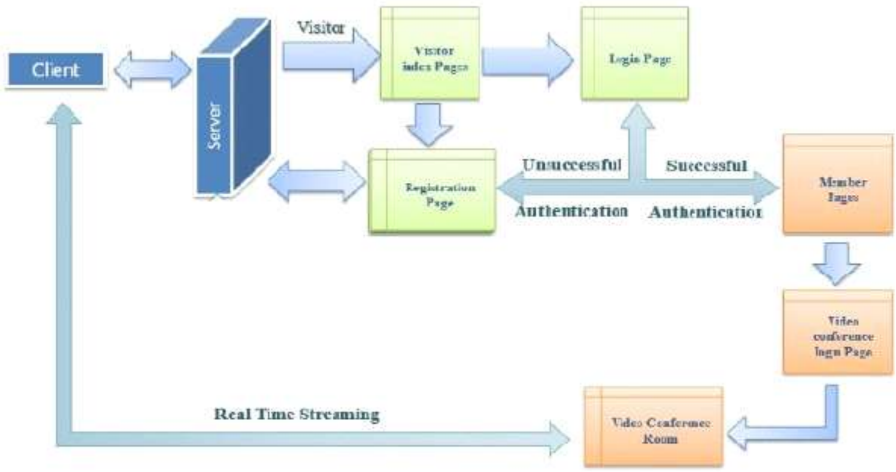


Implementation details such as formats, delivery protocols and interoperability among net browsers are handled by the browser builders and WebRTC implementation. The architecture depicted right here is given by using WebRTC web site, an open-source task maintained by way of Google, Mozilla, Opera and others. There are different open-supply implementations of WebRTC, inclusive of Ericsson's Open Web RTC[10].

Proposed video conference site structure

The proposed videoconference site (VCS)is ASP.net website programed by use C# and JavaScript. The site is a client-server-based system.



The proposed videoconference device has two major components:

- Video Streaming Server
- Client tool

The VCS has two levels of security, the visitor stage which lets all consumers input the site and view allowable content material and permits him to check in to be member and then can login to member level. The member level allows only for member users to view the full content and may enter videoconference page.

The VCS transmits audio and video and audio streams between consumers if you want to make video conferencing. The VCS streaming server lets in users to start video and audio conservation. The video convention web page allows members to enter the conservation room, in which each videoconference room has a particular call and every member has a unique login call.

After logging in to chatroom, the members can begin video conferencing with every other. The chat room shape described in follows:

- The room Chatting contributors: this is the right aspect of chatting room have been member that enter chatting room are regarded in it.
- Video box: it is the location where you can view video chat with called member, in case of multiple contributors there will many chatting video fields for all customers that streaming video from all at identical time. In bottom there may be small field that show the person video from its digital camera, he can prevent displaying video by way of clicking on video icon or prevent audio through clicking on microphone icon.
- Video/Audio call button: there are buttons to starting chat: video call, to starting video call. To begin video or audio assembly the consumer wants to choose one or multiple customers in video conference room then click on video or audio button, the other customers want to just accept call with the intention to begin meeting

II. CONCLUSION

In this paper, we present our redeveloped real-time web browser-based video conference. The proposed video conferencing system architecture is based on a WebRTC environment that supports single or multiple participants in a video conferencing session using a single connection. Compared with existing commercial video conference software, the proposed system is web browser-based and it is a cross-platform application which can be run on different devices such as desktop computers, etc. The test results show that the video conference system has worked perfectly. The tests approved that the quality and streaming speed of video conference is highly dependable on speed of internet of clients and streaming bandwidth of the server, this means it is independent of the number of conference members.

REFERENCES

[1]. J. G. Apostolopoulos, W. Tan, S. J. Wee, “Video Streaming: Concepts, Algorithms, and Systems,” Mobile and Media Systems Laboratory, HP Laboratories Palo Alto, Hewlett-Packard Company, September 2002.

[2]. Fundamentals of Streaming Media Systems,” Pearson Education, Inc., link: https://www.pearsonhighered.com/assets/samplechapter/0/1/3/0/0130670383.pdf

[3]. L. Coventry, “Video Conferencing in Higher Education,” SIMA Video Conferencing Outline, Institute for Computer Based Learning, Heriot Watt University, Edinburgh.

[4]. M. Wenzel and C. Meinel, “Full-Body WebRTC Video Conferencing in a Web-Based Real-Time Collaboration System,” Proceedings of the 2016 IEEE 20th International Conference on Computer Supported Cooperative Work in Design, 2016.

[5]. L. A. Tomei, “Lexicon of Online and Distance Learning,” R&L Education, 2009, ISBN:9781607092858.

[6]. W. Liu, K. Zhang, C. Locatis, and M. Ackerman, “Internet-Based Videoconferencing Coder/Decoders and Tools for Telemedicine,” Telemed J E Health 17(5), Jun 2011, pp.358–362.

[7]. Forrester, “An Introduction to the Basics of Video Conferencing,” Preparing for Uneven Corporate Adoption of Video Communications, polycom, May 2011.

[8]. S. Loreto and S. P. Romano, “Real-Time Communication with WebRTC: Peer-to-Peer in the Browser,” O'Reilly Media, Inc., 2014, ISBN: 9781449371852.

[9]. A. Hallberg, “A protocol for decentralized video conferencing with WebRTC,” Solving the scalability problems of conferencing services for the web, Degree Project in Information and Communication Technology, School of Information and Communication Technology, KTH ROYAL Institute of Technology.