DEVELOPMENT OF VIDEO CONFERENCE PLATFORM BASED ON WEBRTC

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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 and investigate the 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 andthen 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.

Index Terms - Videoconference, Web Conference, Video Chat, WebRTC

I. INTRODUCTION

Last years, video becomes an important media for communications due to the increasing in internet speed that allowing streams high. Previously, the video was captured and transmitted in analog form. The development in computers and digital integrated circuits was led to the digitalized of video, and the digital video leads to revolution in the communication and compression of video. Generally, the process of using the Internet to transmit content by encoding it into a number of decodable formats is called streaming. When the transmission is performed as content is created, the stream is called a "Live" stream [1].

Live VideoStreaming is a type of video streaming that transmits an electronic message through a local area network (LAN) or through the Internet in real time so that the video and/or audio from transmitter source can be heard and seen on the receiver side via personal computers, smart phones and mobile devices, etc. Real time media communication (audio, video) between different client devices, is includes one-way communication (streaming) or two-way communication (video/ audio chat or video conference).Live media video conferences send and receive video and audio between multiple endpoints. Particularly, the media streaming needs to setup of standalone streaming servers, instilling an appropriate standalone application in client side and support to streaming protocols that control the transferring the streamed packets. With respect to conferencing and chatting the need additionally to mediation of a session manager in-between the clients and the support of the corresponding session protocols. With respect to real time communicating by the web, until now HTTP is just the media streaming myth method. With respect to receipt media streaming over the web, it can be achieved only via installation of the suitable third-party software (browser plug-in) in order to receive and process the media streamed from the server. At last, the need to media players that provide plugins for browsers to permit audio and video streams to be run over the web[2, 3].

In order to combine video conferencing, which goes beyond a basic face-to-face setup with a flexible web browser-based groupware system, the video conferencing has to be run in a web browser context too. Compared to native groupware applications, this implies limitations regarding user interface (UI) layout (e.g. window transparency) and available video processing application programming interfaces (APIs). Rather than traditional method, in which the audio/video real-time communication within web browsers are only possible via plugins or third-party software, the Web Real-Time Communication (WebRTC)7 is a collection of communication protocols and APIs that support peer-to-peer (P2P) real-time communication among web browsers. With the P2P capabilities introduced by WebRTC, browsers now break away from the classic client-server model. The advantage of this shift is that the APIs defined by WebRTC are the same regardless of the underlying browser, operating system etc. and are available on many platforms, especially mobile device [4].

The work is aiming to develop a video conferencing website that allows users anywhere in the world to join real-time streaming video chat rooms without installing any software. This proposed environment is in security infrastructure that performs secure streaming to users for the prevention of security threats.

II. VIDEOCONFERENCING CONCEPT

A videoconference (which also called video teleconference) is several interactive telecommunication technologies that allow two or multiusers to interact through two-way audio and video transmissions together. Video conferencing uses video and/or audio telecommunications to allows peoples from different locations to meet together in one chatting area. The videoconference is simple as a conversation between two (point-to-point)in private offices or more people (multi-point) in large rooms at several sites. In addition to audio and video transmission through meeting activities, the videoconferencing is also allowing to share files such as documents [5].

Video conferences has two main types, point-to-point video conference which takes place between individuals in-between two separate sites, and multi-point video conference that make interaction between individuals at three or more different locations [5].

The main technology that has been used in a videoconferencing system is an audio/video streams compression in real time. The software or hardware that proceed compression is known as codec (coder/ decoder). The utilizing of audio modems in the transmission line was allow for the use of Plain Old Telephone System (POTS), in some low-speed applications includes video telephony, this due to fact it converts the digital pulses to/from analog waves in the audio spectrum range [6].

The general components needed for a videoconferencing system can be described in follows [7]:

1. Video input: the video input device such as webcam or video camera
2. Video output: the device that view the streaming media such as television, computer monitor, or projector
3. Audio input: the device that received audio such as CD/DVD player, microphones, or any other audio outlet source.
4. Audio output: this to output audio such as loudspeakers or telephone
5. Data transfer: digital or Analog telephone network, Internet or LAN
6. Processing device: the device that processesthe overall videoconference operation such as computers, smartphones, tablet, etc.

III. WEB REAL-TIME COMMUNICATION (WRTC)

Web Real-Time Communication (WebRTC) is a framework that allows peer-to-peer communication between web browsers. The technologies in the WebRTC stack and its API:s are currently being standardized by the World Wide Web Consortium (W3C) and the Internet Engineering Task Force (IETF), and implemented by browser vendors such as Google, Ericsson and Mozilla. WebRTC allows browsers to stream audio, video and arbitrary data directly to one another without the need for a central server. This makes it possible to write and run real-time applications such as games and communication services directly in the browser; there is no need for plugins or platform-specific applications [8].

The WebRTC (figure 1) contains a Voice Engine, Video Engine, and tools for Transport and communication. This means that anything related to media encoding (converting audio and video from one format to another) and compression, as well as low-level networking is handled by the framework. Web browsers and other native applications can access the framework through its C++ API. Web applications cannot access this low-level API for security- and interoperability reasons, so web browsers need to provide another way for developers to use it. The standard way of doing this is through a JavaScriptAPI[1]. Web applications can use the standardized JavaScript API to access the functionality of WebRTC [9].

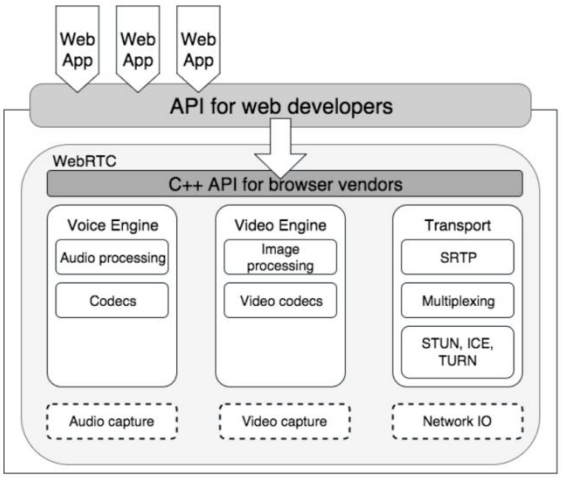


Figure 1: Simplified WebRTC Architecture [9].

Implementation details such as codecs, transport protocols and interoperability between web browsers are handled by the browser developers and WebRTC implementation. The architecture depicted here is provided by WebRTC site, an open-source project maintained by Google, Mozilla, Opera and others. There are other open-source implementations of WebRTC, such as Ericsson's OpenWebRTC[10].

IV. PROPOSED VIDEOCONFERENCE 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. Fig. 2 illustrates the VCS structure and operation.

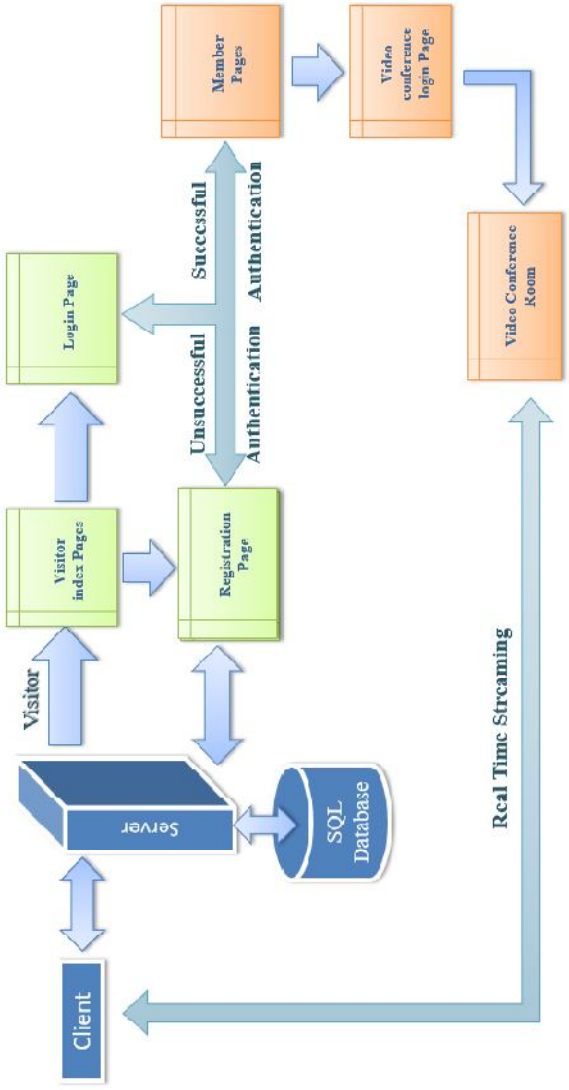


Figure 2: VCS structure and operation.

The proposed videoconference system (shown in Figure 3.1) has two main components:

- Video Streaming Server
- Client device

The VCS has two levelsof security, the visitor level which allows all user to enter the site and view allowable content and allow him to register to be member and then can login to member level. The member level, allow just for member users to view the full content and can enter videoconference page.

The VCS transmits audio and video and audio streams between Client in order to make video conferencing. The VCS streaming server allows users to start video and audio conservation. The video conference page allows member to enter conservation room, where each videoconference room has a unique name and each member have unique login name. Figure 3 shows the videoconference page.

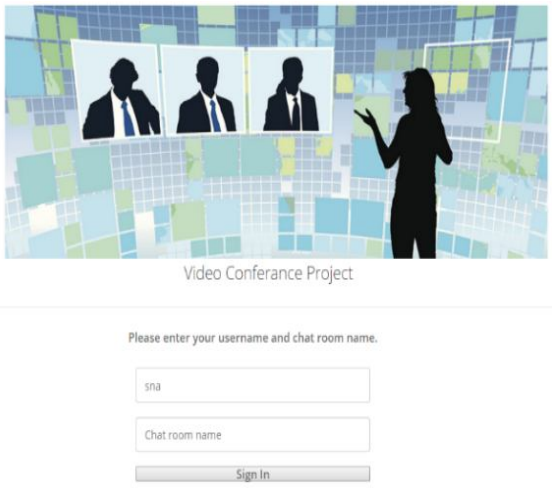


Figure 3: Videoconference page.

After login to chatroom, the members can start videoconference with each other. The chatting room structure (shown in figure 4)described in follows:

- The room Chatting members: this is the left side of chatting room were member that enter chatting room are appeared in it.
- Video box: It is the area when can view video chat with called member, in case of multiple members there will many chatting video box for all users that streaming video from all at same time. In right side there is small box that show the user video from its camera, he can stop showing video by clicking on video icon or stop audio by clicking on microphone icon.
- Video/Audio call button: there are two buttons to starting chat: video call, to starting video call and Audio call. To start video or audio meeting the user need to select one or multiple users in video conference room then click on video or audio button, the other users need to accept call in order to start meeting.

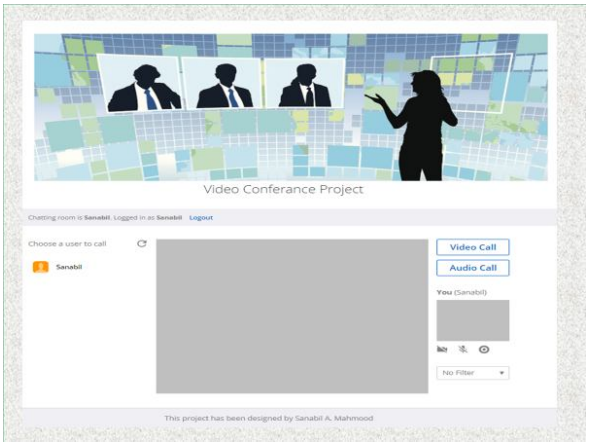


Figure 4: videoconference room.

V. RESULT AND DISCUSSION

The VCS site has been uploaded to ASP host server in order to test the operation in real working conditions.

For security side, the authentication process appears that no one can access to member page without enter as an authorized member. The key is encrypted in client and member side so no one can know what the member key is, thus VCS offer privacy to me member. Figure 5 shows the SQL database registration information.

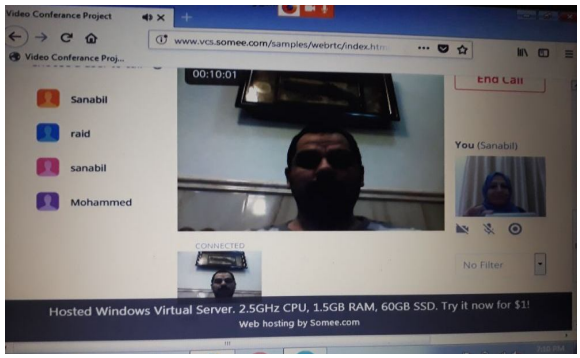
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Figure 5: VCS SQL database registration information.

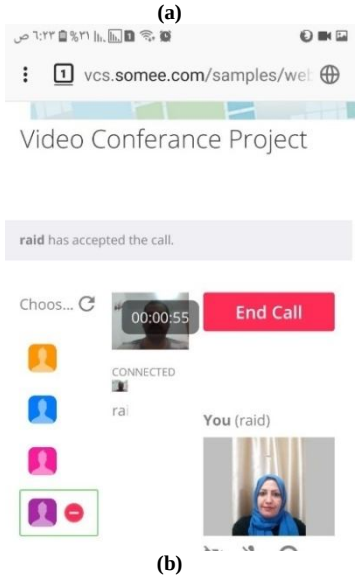
For videoconference operation test, the VCS has been tested to work in different operation systems and devices. First, we have tested connection between computers having windows operation system with different version (windows 7, windows 8 and windows 10). Then, we tested the operation of VCS on the other computer that operate with different operation system (MAC OSX operation system).After that we test the operation of VCS on smartphone devices with different operation system (Android and iOS)Also, we Have evaluated the connection between different devices at same times. The other test is to check the compatibility with common internet browsers (Internet Explorer, Safari, Google Chrome, Mozilla Firefox and smartphone default browsers). The test results are shown in table 1. Figure 6 shows the online calling between two peoples in video conference room

Device	Operation System	Internet Browser		
		Default Browser	Google Chrome	Mozilla Firefox
Computer	Windows 7	Yes	No	Yes
	Windows 8	Yes	No	Yes
	Windows 10	Yes	No	Yes
Mac Computer	OSX	Yes	No	Yes
Android phone	Android 7	No	No	Yes
iPhone	OX 10	No	No	Yes

Table 1: The ability of VCS to run in different devices, operation systems and internet browsers.



(a)



(b)

Figure 5: The video conference room when connection between (a) computer run by windows 7 operation system, (b) android phone.

From table 1, the test results show that the videoconference can run in any device with any operation system so it independent on operation system, that's mean any one from any were can uses the site to contact the other just need contact device (computer tablet, or smartphone), internet, webcam and microphone with any specification. The results show not all browsers can run videoconference directly this due to many browsers disable java scripts and the user need to enable it doing some security setting to make it run, the test shows that the best compatible internet browser is Mozilla Firefox that is available for all operation system where the VCS run perfectly with it.

The videoconference speed test shows that the video resolution and response time are depending on user internet speed and the server streaming bandwidth, so as increasing in internet speed the video chat more smother and low delay, and more streaming bandwidth in server more users can contact in high speed.

CONCLUSION

In this paper, we present our redeveloped real-time web browser-based video conference. The proposed video conferencing system architecture based on a WebRTC environments 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, smartphone, tablets, etc. The video streaming has been securing by two level of authorization Gust and Member to allow only to authorized people to enter videoconference room and makes Live video chat with friends. 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.

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