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M. Sc. in Electrical And Electronics Engineering

SANABIL AHMED MAHMOOD

UNIVERSITY OF GAZİANTEP

GRADUATE SCHOOL OF

NATURAL & APPLIED SCIENCES

DEVELOPMENT OF VIDEO Conference PLATFORM BASED ON WEB RTC

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Supervisor

Prof. Dr. Ergun ERÇELEBİ

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Prof. Dr. A. Necmeddin YAZICI
Director

I certify that this thesis satisfies all the requirements as a thesis for the degree of Master of Science.

Prof. Dr. Ergün ERÇELEBİ
Head of Department

This is to certify that we have read this thesis and that in our consensus; it is fully adequate, in scope and quality, as a thesis for the degree of Master of Science.

Prof. Dr. Ergün ERÇELEBİ
Supervisor

Examining Committee Members:	Signature
Prof. Dr. Ergun ERÇELEBİ	
Prof. Dr. İlyas EKER	
Assoc.Prof. Dr.Sema KAYHAN	

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Sanabil Ahmed MAHMOOD

ABSTRACT

DEVELOPMENT OF VIDEO CONFERENCE PLATFORM BASED ON WEBRTC

MAHMOOD, Sanabil Ahmed
M. Sc. in Electrical and Electronics Engineering
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Videoconference is an internet-based which takes place in between two parties, which every participant is able to see a real time video of the another. Web Conference is a type of videoconference, which is a real-time online event based on audio or video communication. In this thesis, we have designed 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 the 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 proposed videoconference is designed to be multi-point videoconferencing that allows many users to enter videoconference room and makes conservation between all in same time.

The security point of view is based on protect videoconference page via used authorization that allows to access the website pages or video-conference page depending on authorizing level; The password has been encrypted by used an encryption technique that utilized hashed and salted algorithm to protect it from cracking by any types of attack.

The system has been tested in real operation for both network and internet, where the site has been hosted in ASP web server and then tested the overall system operating in term of performance, stress, security and ability of operation in different operation systems and internet browsers. The results show that the site worked accurately and able to run in all devices and operation systems, however the tested shows that the Mozilla Firefox is best internet browser that is highly combatable with proposed system and it is available for most operation systems. From test it can be conclude that the proposed system can be utilize as a dependable videoconferencing method which can be run in any device from any place without the need to instill specific program or limited to device or operation system and it give ability to start multi-point videoconferencing for large number of users.

Keywords: Videoconference, Web Conference, Video Chat, WebRTC.

ÖZET

WEBRTC dayalı video konferans platformunun geliştirilmesi

MAHMOOD, Sanabil Ahmed
Yüksek Lisans Tezi, Makine Mühendisliği
Tez Yöneticisi: Prof. Dr. Ergun ERÇELEBİ
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Video konferans, her katılımcının diğerinin gerçek zamanlı videosunu görebildiği iki taraf arasında yer alan internet tabanlı bir sitedir. Web Konferansı, sesli veya görüntülü iletişime dayalı gerçek zamanlı bir çevrimiçi etkinlik olan bir video konferans türüdür. Bu tezde sesli ve görüntülü iletişim sağlayan gerçek zamanlı video konferans sistemi tasarlanmış ve gerçekleştirilmiştir. Web sitesi, Web Gerçek Zamanlı Haberleşme (WebRTC) temel alınarak geliştirilmiştir. Site, bir programlama dili olarak C # kullanarak ASP.net altında tasarlanmış dinamik bir sitedir. Video konferansın dinamik içeriği, web sunucusunun JavaScript'i belirli eylemleri yürütmesini bildirdiği ve komut dosyasının web sunucusuna geri bildirim bilgisi verdiği JavaScript'dir. Güvenlik bakış açısı, yetkilendirme düzeyine bağlı olarak web sitesi sayfalarına veya video konferans sayfasına erişmeye izin veren kullanılmış yetkilendirme yoluyla video konferansı sayfasını korumaya dayanır. Parola, herhangi bir saldırı türünden çatlamadan korunmak için karma ve tuzlanmış algoritma kullanan bir şifreleme tekniği kullanılarak şifrelenmiştir. ASP web sunucusunda barındırılan sistem, hem ağ hem de internet için gerçek operasyonda test edilmiştir. Ayrıca, farklı işletim sistemlerinde ve internet tarayıcılarında performans, stres, güvenlik ve çalışma yeteneği açısından değerlendirilmiştir. Sonuçlar, video konferans sitesinin tüm cihazlarda ve işletim sistemlerinde doğru şekilde çalıştığını göstermektedir. Bunun yanında testler esnasında geliştirilen video konferans sitesinin Mozilla Firefox web tarayıcısıyla daha uyumlu olduğu ve daha iyi sonuçlar verdiği gözlemlenmiştir. Testlerden, önerilen video konferans sitesinin, herhangi bir cihaz veya işletim sisteminde, özel program

gerektirmeksizin herhangi bir noktada çalışabilen güvenilir bir video konferans yöntemi olarak kullanılabileceği görülmüştür.

Anahtar Kelimeler: Video konferans, Web Konferansı, Görüntülü Sohbet, WebRTC.

Dedicated to

"To Beloved parents, my brothers and friends"

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LIST OF SYMBOLS/ABREVIATIONS

ANSI	American National Standards Institute
ANSI C	C programming language published by the ANSI C
API	Application Programming Interface
ASP	Active Server Pages
CFML	ColdFusion Markup Language
HTML	Hypertext Markup Language
HTTP	Hypertext Transfer Protocol
IETF	Internet Engineering Task Force
IP	Internet Protocol
ISDN	Integrated Services Digital Network
JSP	Java Server Pages
LAN	Local Area Network
MCU	Multipoint Conferencing Unit
MPEG	Moving Picture Experts Group
PHP	Hypertext Preprocessor
POTS	Plain Old Telephone System
SHA	Secure Hash Algorithm
SQL	Structured Query Language
VCS	Video Conference Site
VOD	Video-On-Demand
W3C	World Wide Web Consortium
WebRTC	Web Real-Time Communications

CHAPTER 1

INTRODUCTION

1.1 General Introduction

Video is a significant media for entertainment and communications for several decades. First video was captured and transferred in analog form. The development of computers and digital integrated circuits developed the video digitization. Digital video allowed a revolution in the communication and compression of video [1].

The popularity and development of the internet from the mid of1990s driven video communication across best-effort packet networks. From when high bandwidth broadband connections are becomes usable to everyone, the needs for online video are frequently increasing. Video-on-demand systems such as BitTorrent and YouTube are highly popular, in which the user has the chances to choose if likes to play a video [2].

Video across best-effort packet networks is challenging by a range of factors including delay, unknown and time-varied bandwidth, and losses, in addition to several additional issues including the most effective way to relatively share the network sources over several flows and the most effective way to carry out one-to-several communication for favored content [1].

Realtime media communication (audio and/or video) amongst different user devices, involves 1-way communication (streaming) or 2-way communication (audio and/or video chat or video conference), had been, more or less static and uniform operation overriding by different platform-specific solutions. Mainly, the media streaming requires to install of separate streaming servers, instilling an proper separate

application in user side and support to streaming protocols that control the transmitting the streamed packets. With respect to chatting and conferencing the need further to a session manager mediation in-amongst the users and the support of the related session protocols. With respect to Realtime communication by the web, up to now HTTP is just the media streaming myth technique. With respect to receipt media streaming using the web, it is achievable just by installation of the appropriate third-party program (browser plug-in) to be able to process and receive the media streamed direct from the server. Finally, the need to media players that give plugins for web browser to allow audio/video streams to be run using the web [3, 4].

There are a number of formats for streaming vender for example, Adobe flash, Apple Quicktime, Microsoft Windows Media, and RealMedia RealNet-works) and every one of the formats are shown by a supporting player in that each vendor is offering a player that is usually free downloading from the Internet. However, HTML5 and other associated technologies of the web have changed drastically the dynamic website, because it platform independent and browser-friendly approach. This is because the HTML5 insert different extended functionalities to web browsers, which changes the method of transferred the data, the graphics has been processed by used hardware acceleration and the visualizations are displayed. This is usually accomplished through using many JavaScript libraries and custom programming-based JavaScript that permit the web-pages to gain access to different device features provides for customization and media access. In that context, the installing the flash player not required for video streaming anymore, because the HTML5 offers an element name “video”, which can be used for any such plug-in. Moreover, the images are able to be loaded in an element by use the tag name “canvas”, that represent the container that can be utilized to draw graphics on the fly by used JavaScript. The canvas has been supported from most popular mobile browsers and computers. The streams API and media Capture allow access through the web to a user’s camera and microphone device [4].

1.2 Video Streaming

Video Streaming is a sequence of "progressing images" that are transferred in compressed format over the Internet and watched it by the viewer as they are available. It's simply, a technique for transmitting data in a way that it can be processed as a continuous and steady stream known as streaming. If a web client is getting the video data as streams, then he/she doesn't have to waiting to download a large file before going to listening to the audio or watching the video. In streaming technique, the file is transferred to the end user in a (less or more) continued stream [5].

1.2 1 Video Streaming Over Dynamic Web Page

Dynamic web pages are usually the websites that are usually created at the time of reach by a user or modify due to the interaction with the user. Usually, the dynamic web pages is a essential part of Web 2.0 that allows data sharing over multiple websites. It would have two kinds of dynamic activity: Content and Code. dynamic content is fully displayed, and visible Dynamic code is hidden or behind the scenes [6].

1) **Dynamic content:** Dynamic content observed in plain view. Changeable contented is observed dynamically on the fly dependent on specified conditions, often by getting contented saved in a database. A website that have dynamic contented represents how its images, text, messages and other data are observed on the web page and more especially how its contented changes in any provided time. The contented of web page changes dependent on specified conditions, either by changeable user enter or by predefined rules. Such as, a website which includes a database of news articles are able to use a predefined rule that tells it to display all news articles for present date. This type of dynamic website can automatically display the latest news articles on any provided date. The other dynamic contented example is whenever a retail website which includes a media products database permits a user to enter a search request for the search phrase [7].

2) **Dynamic code:** The code will be created dynamically on the fly making utilize of dynamic programming language rather than used static, plain HTML. The dynamic webpage is created on the fly by combined together specific blocks of code, routines or procedures. A web page that is dynamically created will recall a range of bits of data from a database and place them together in a predefined format to supply the reader with a defined page. It communicates with users in a range of ways such as session factors, checking cookies knowing users' old history, server-side factors etc., or via making utilize of direct interaction (mouse overs, form elements, etc.). A site is capable of displaying the current state of a dialogue amongst users, observe a changing situation, or supply data in some way personalised to the needs of the individual user [7].

There is a variety of software systems, such as JavaServer Pages (JSP), ANSI C servlets, Python, Perl, PHP, Ruby, Active Server Pages ASP.NET, (ASP), ColdFusion (CFML) and YUMA, which are available to create dynamic sites and dynamic web systems. Sites could also have content that is gathered from one or more databases (such as MySQL, SQL or Access database). In this thesis, ASP.NET is used to build SVS Site; because Asp.net is a useful tool to build dynamic code page that can give reliability, flexibility, and open range of options for controlling. Also, it can contact to data base (SQL used as a database) and with best security; because it can be used to make page site for authorization to prevent person not allowed to enter that site, side by side to ability to encoding content like information's, files, and others by building server-side program controlling all of that.

1.3 Video Conferencing Basics

A videoconference (also known as video teleconference) is numerous interactive telecommunication techniques that permit two or multi people to interact via two-way audio and video sending together. Videoconferencing is being used telecommunications of audio and video to allow peoples meeting together at different sites. The videoconference is simple

as a conversation between two (point-to-point) in private offices or more people (multi-point) in large rooms at numerous sites. In addition to audio and video transmission via meeting activities, the videoconferencing is also allowing to share files such as documents [8].

Video conferences has two main types, point-to-point video conference which takes place in between people in between two standalone sites, and multi-point video conference which make conversation in between people at 3 or more separated locations. The main technology that 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 [9, 10].

The general components needed for a videoconferencing system can be described in follows [11]:

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 process the overall videoconference operation such as computers, smartphones, tablet, etc.

1.4 Literature Survey

Many investigators have studied the video conferencing and the technologies used to designed it.

1) Petrík and Skala (2006) [12], has tested a a variety of videoconference models and consulting this topic with the users used videoconferencing inside Czech Republic and global. The report contains a recommended branch of videoconference types recommended set of criteria for review of each videoconferencing service as well as the reviews of all models tested along with details of how they satisfy each single criterion, suggestion of a unique system for each certain kind of videoconferencing and the set of other possible videoconferencing products. There report assist to make faster determination of which unique system need to be used for appropriate type of videoconferencing for improve consultations of technical problems.

2) Chokshi (2011) [13], provides a report pointed the video chat application requirement specially for Facebook. The report includes in depth information of red5 streaming server, the application work flow. This project provides a program that could possibly be used by every user in his/her daily life. The video chat application tends to make user's life comfortably by offering a easy way of communication.

3) Rodríguez et al. (2013) [14], proposed a central video conferencing architecture dependent upon a MCU for WebRTC environments. that architecture helps the participation of multiple people in a video conferencing meeting making use of a single connection. that work reviewed how these types of architecture offers solutions to particular scenarios with requirements like screen adaptation, stream processing and composition for bandwidth and meeting recording. also, that work gives current problems that should found throughout the implementation of that MCU and the qualities need to be established in WebRTC libraries. The qualities like sessions recording, the multiple user's participation, transcoding, etc. They describe that elements are required for implementing capable applications of WebRTC and the variances can be found between implementations and standard at the time of did this work. And they also describe their first MCU implementation that considers these types of requirements. By

that implementation they described how WebRTC works, and this work is a good starting point for developer to implementing further qualities and testing central web video conferencing architectures.

4) Panagiotakis et al. (2013) [4], presented a common architecture that uses HTML5 and other new techniques recently come out with HTML 5. Particularly, the canvas element assists video display by graphics design and a camera on the fly. Moreover, the protocol of WebSockets determines a bidirectional connection in between a browser and server. Taking benefit of such techniques, they presented a couple cross-device, collective web applications. Each application makes use of the browser to create a communication with a webcam. The camera data is streamed through a WebSocket connection to the server and is in the end sent to the connected clients. The first application allows streaming live video to every connected user and the second application is a webchat in between a couple of users. This method is less complicated than other similar attempts (for example the WebRTC), because it only needs the support of WebSocket protocol and HTML 5 by the client peers. Meeting management is implemented by their service logic with no needed to any Meeting explanation protocol by the peers.

5) Kwok-Fai et al. (2014) [15], proposed an approach of P2P-MCU for multi-party video conferencing that will effectively operates with both PCs and ordinary smart phones. In this approach, a module of MC has been added in to the browser to transcode and mix the audio and video streams in realtime. Once the browser operates as the MCU node will leave the conference meeting without notice, the other candidate browser could take over the control instantaneously, and the on-going WebRTC conference is often easily restored with proposed MCU selection algorithm. The proposed P2P-MCU solution decreases 35% bandwidth consumptions and 64% CPU usages for every person when compared with the mesh-network solution.

6) Pasha (2016) et al. [16], proposed a most useful centralized structure for the video conferencing to service the WebRTC via using MCU. that work is a

centralized and integrated structure of video conferencing that depends on a multiple communication unit for WebRTC environment. They fully discussed how that structure provides resolutions to certain contexts such as session recording, stream processing and composition for screen adaptation as well as bandwidth. A reasonable part of the paper focuses on the challenges that the MCU might face and on the features that are supposed to be fully supported from WebRTC libraries.

1.5 Aim of the Work

The work is aiming to develop a video conferencing website that makes it possible for users anyplace in the world to enter realtime streaming video chat rooms without installing any software. This proposed environment will be in security infrastructure that performs secure streaming to users for the prevention of security threats.

1.6 Thesis Layout

This thesis is arranged in four chapters besides this chapter:

- **Chapter Two:** reviews the video conference methods and the techniques used for designed it.
- **Chapter Three:** describes the design of the video conference system with all techniques and steps used in it.
- **Chapter Four:** details the results and outlines the main conclusions of the study provided in this thesis.
- **Chapter Five:** includes the conclusions, then the suggestions for future work.

CHAPTER 2

VIDEO STREAMING AND VIDEO CONFERENCING CONCEPT AND ITS TECHNIQUES

2.1 Streaming Video Principles

Streaming video data is transferred in compressed form through the Internet and will be shown by the viewer in realtime. With streaming media or streaming video, the web client doesn't need to waiting download the file to play it, rather than, the media is delivered as a continuous stream of data which allow to be played when it arrives. In order to play streaming video, the user will need a desired player, with a specific program which can uncompressed and play video data on user display and hear audio by the speakers [17].

Major technologies of streaming media and streaming video involve Microsoft Windows Media Technologies (which include its Theater Server and NetShow Services), and RealNetwork (RealSystem (G2) with RealSystem Flash), and VOD. Microsoft's method utilizes the basic compression algorithm for video called Moving Picture Experts Group (MPEG). Other methods use exclusive algorithms. Streaming video is often sent from prerecorded videos but could also be sent out as an important part of a live broadcast. Within the live broadcast, the video signal is converted to a compressed digital signal and sent from a specific web server that is able to do multi-cast, transmitting the same file to several users at the same time [18].

2.1.1 Video Streaming Methods

The strategy chosen are seeking to be depend on the state, the majority people are seeking to opt for HTTP streaming (progressive download or normal download) due to the fact

that it is the cheapest and easiest way to start. For commercial and advanced utilize when traffic is very large, there's a are seeking for streaming server which must be used. Each progressive download and real streaming get their own limitations and benefits. If they are seeking to reach viewers at slower connections and are seeking the high quality, the progressive download will be the best choice. Nevertheless, if there's capability to identify the viewers are seeking to get a fast-enough connection to view stream, the real stream could save on bandwidth via streaming the video. In case that no determining the right video are seeking to be served to, the progressive download are the safer choice because no problem what connection people get. For case of live streaming, the streaming server must be used, which can't be done over HTTP [19].

2.1.2 HTTP Streaming

This is the almost all basic and costly way to stream video from a website. The HTTP refer to Hyper Text Transfer Protocol, which is the same protocol utilized to deliver web pages. When using this fact, it is simple to setup and use in almost any type of website, with no needing to special hosting plans or additional software. The websites which is medium or small sized are far more likely to use this method compared with the more expensive streaming servers. When using this method, it does not need any special type of host or website, it just need host server which knows common video file types (almost all traditional hosting accounts carry this out) [20].

There are a number of limitations to take into account concerning HTTP streaming [21]:

- HTTP streaming is a great option for the modest traffic websites, i.e. lower than about a dozen people watching video content or communicate at the same time. For heavy traffic sites an additional serious streaming solution need to take in consideration.

- It cannot detect the speed of end user's connection automatically by used HTTP. If there is a need to make various versions for various speeds, it should create a

separate file for every speed.

- HTTP streaming is not as efficient as other methods and it will incur a heavier server load.

2.1.3 Video Servers Streaming

A streaming video or streaming media server is a specific application that run on an Internet server. This is usually referred to "true streaming", because the other methods just simulate streaming. There is some advantages true streaming can be described in follows [22]:

- The capability to deal with much larger traffic loads.
- The capability to recognize the speeds of users' connection and provide recommended files instantly.
- The capability to broadcast live events

.

There are a couple of ways to access to a streaming server [23]:

1. **Operate own server (via leasing or purchasing):** This is a dedicated step and is better over the needs of most websites. To operate the personal streaming server, it should be either buying a standalone server device or buying a software package of streaming server and set up it on a current web server. The streaming software is obtainable for all popular server platforms like Windows, Linux, etc. In follows a number of examples of software used to streaming media:
    - **RealNetworks Helix Universal Server**, supporting a wide range of formats, such as Windows Media, MPEG-4, Quicktime and RealMedia.
    - **Apple Quicktime Streaming Server,** supporting a limited format such as 3GPP and MPEG-4.

- **Macromedia Communication Server,** specialise in Flash-based interactive multimedia and video.

2. **Signup for a streaming hosted plan with an ISP:** In in much like to sign up for a website hosting plan, it could get a hosting plan that specializes in streaming media.

2.2 Video Conference Concept

Video conference is an internet-based conference (or a conference throughout distance) which takes place in between two parties, which every participant is able to see an image of the another, hence every party can listen and speak to other respondents in real time. The elements required to make this happen involve, A webcam a display, microphone and speakers, A software program which captures the video and voice stream from the microphone and webcam, encodes it, sends it to another participant, and at the same time decodes the digital voice and video stream which received in the other participant. The software program which connect every party together throughout a digital connection, controlling the exchange of video and voice between respondents. At both end of the connection, the voice and video traffic are merged and sent to every participant in the kind of a realtime audio and video image stream and an optional administration tool for the arranging of video conferencing times [24].

At a bit more expert level, it is furthermore likely to offer the ability to share contents from a device throughout a video call. The type and quality of contents which can be shared is dependent upon the data exchange rate throughout the call. Terminology utilized via video conferencing clients to identify the process of dialing to and active in a virtual meeting is called “joining a bridge”. A variety of virtual meeting rooms are given distinctive “bridge numbers,” and clients join up a video call via “dialing a bridge number. Video-enabled conferences take place in two different ways: First way is point-to-point and second way is multi-point [24].

2.3 The Videoconferencing Technology

The videoconferencing technology is based on a variety of determining factors that transfer audio and video through a network. The common keyword related with video conferencing is interactivity. This permits the real-time audio and visual contact amongst two or more persons at a variety of geographical locations. Concurrently, it likewise supports the utilize of a variety of further resources like pre-edited video clips and computer presentations [25].

The three main elements of the conference system are [25]:

1. The communication system and service that refers to the real communication system in between the sites.
2. The mechanism of connection, if it activated, permits the connection in between two or more places in same time. Some mechanisms could perhaps have directory services that facilitate authentication and authorization enabling manage over who can uses the services of videoconferencing.
3. The conferencing should contain the video signal and audio that is compressed and transmit to the receiving site/s.

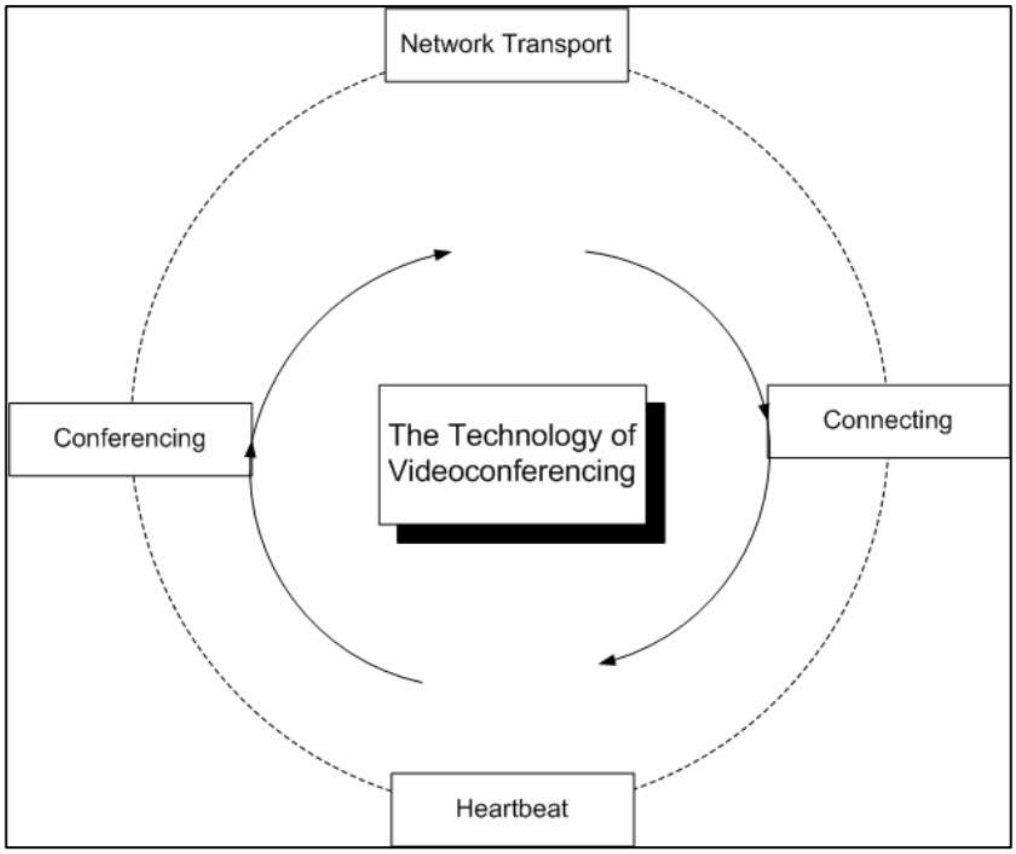


Figure 2.1. ASP.NET Managing Server-Side Code [25]

2.4 Types of Videoconferencing systems

There are several types of Videoconferencing systems, which can be described in follows [25]:

1. **Integrated Room Size**

   Multi-monitors are usually used to show video, and for the audio used multi microphones. This assists one to various interactive learning, usually controlled by the chairperson or teacher. Participation is usually irregular and specific with formal communication.

2. **Desktop**

   This usually supports one-to-one video conferencing, in which participants that doesn't have access to video conferencing devices may connect with each other by using desktop video conferencing program. It's an ideal tool for tutorial sessions and meetings.

3. **Portable Units**

   The video conference system is usually set in such a manner so that it can be transferred from one place to the second. Connection can also be established via a network connecting near-end (the local), the far-end (remote sites)).

2.5 Equipment of Videoconferencing

Any video conferencing setup needs some basic hardware components [26]:

a) **Audio**: Video conference rooms are usually equipped with some microphones set. These could either be standalone parts connected to the system or in built in systems.

b) **Cameras**: The camera used in conference are range from a simple one such as fixed camera of low resolution up to high quality camera of high resolution and has automatic pan and zoom. At least two cameras, are necessary for appropriate

communication between clients, which is, one camera at the first client and second at the other one.

c) **Codec:** this can be implemented in software, hardware, or a mix of both. Initially, the codec converts the analogue video signal to encodes (a compressed digital form) and after that decompresses it and converts the digital signal back to the original analogue video once receiving decodes (transferred information). A low bandwidth or slow codec leads to an unstable picture.

d) **Monitor**: In conference rooms a display screen is needed to show the people at conservation in real time.

.

2.6 Connecting Through Video Conferencing

Right after a connection is established data, it can then be transferred from one site to the another. Connection may occur in a several ways [24]:

1. **IP Video**: IP Video is considered as a stable solution available that can be connecting houses or institutions that is available with good enough bandwidth through their network. This platform utilizes the Internet protocol, thus, in its simplest form; no unique connection is needed except for the persistent network cabling.

2. **ISDN**: This technique is commonly used with high quality video conferencing conversation. Video is transferred by using actual phone line infrastructure offering guaranteed bandwidth for a point-to-point connection. Both ISDN and IP come with some points of disadvantages and advantages. Various connections presently, offer for multiple platforms which means the change from ISDN to IP is simpler to achieve.

The other connection systems normally include a Multi-Point Control Unit (MCU) that allows multiple sites to join in in a video conference in real time.

2.7 Types of Video Conferencing Contacting

Video-enabled conferences take place in two distinct ways: either point-to-point or with multi-point [12].

1- **Point-to-point videoconferencing:** In point-to-point, the most basic scenario is that one person or group is connect to another. The physical elements (i.e. camera and microphone) that allow the meeting to occur are usually integrated in to laptop, desktop computing or tablet, or it can be connected to a specific hardware solution for room.

2- **Multi-point videoconferencing:** In multi-point videoconferencing, 3 or more sites are linked together, where all members can hear and see each other, in addition to see any content becoming shared through the meeting. In this method, the digital data streams of content, video and voice are processed by an essential, independent software program. Pairing the people participant's voice and video traffic, the program resends a collective data stream directly to conference members in the kind of realtime video and audio imagery. People can join in a conference in and can choice video conservation that combine video images and audio of the conference to show on the screen or can choice "audio only" mode.

Based on the technical functionality of the video conferencing system that being used, images could be seen by members are either categorized as "continuous presence" or "active speaker". In this mode, the screen is just gives a video image of a person that is at conversing mode at any point over time. In other hand, the "continuous presence" mode is more advanced level that bridge separates the image on the screen towards a number of different locations. A person that conversing at any point over time is displayed in a large essential field, while the other conference members are displayed around the essential image. Thus, the "continuous presence" mode is allowing the conference members to interact and view all conference members in a 'virtual conference room.' The program that creates the "virtual conference room" along with the digital processing hardware, is termed a "video bridge", or to a short term "bridge", for short. An additional term for a bridge that

is often utilized in a video conferencing is called MUC or “multi-point control unit” [27].

While the point-to be able to-point videoconferencing is somewhat simple, the management and creation of multi-point videoconferences is usually complex. An MCU must be capable to be make, facilitate and control several simultaneous live videos of the conferencing meetings. An additional complexity is placed when various sites may be connected to be capable to the meeting over analogue or digital streams at various speeds, with various data transport and signaling protocols utilized to be able to facilitate the communicating. to be capable to link all those users within a virtual, common meeting, the MCU must therefore be able to recognize and translate among several various protocols (for example uses H.264 for communicating with IP and uses H.263 when communicating via ISDN). The MCU can also let those to linking the video bridge to be capable to do so best possible quality at the highest speed at what their specific system can support. Even though there are two individual processes occurring here called “transcoding.” It is significant to note that not all of the bridges offer such transcoding ability and failing to do this can significantly influence the experience and quality of video calls. In case there is no transcoding be offered and users dial through a bridge across a range of various connection speeds, it is feasible that the bridge can only be capable to support the video meeting by setting the connections in its lowest typical denominator. To explaine the downside effect of this, it can consider a meeting that occurs with most users connecting the bridge from the high-speed company network, but wherever one or two persons dial within the meeting from a low-bandwidth ISDN or DSL such as from home. In such a case the quality of video conference of the many high-speed network users is then downgraded to be corresponding to the lowest common denominator of the low speed -users, probably making the video call ineffective. While when there is a support for effective transcoding is by the MCU, the people have a high-speed network will continue to still able to enjoy HD video quality, in same time the users of low speed network can receive quality appropriate to their connection speed. In summary, the MCU must be designed well, it should have ability to allowing users to be able to call in appropriate resolution and data rate they need or need, it also should be simple and allowing people to focus on the meeting, and it must be seamless experience for all users [27].

2.8 Video Call Management and Protocol Conversion

The software platform of video conference must be ready to provide call signaling function and manage the maintenance and set-up dynamically of a large range video calls. The architecture of software needs to be able of reconfiguring itself and its resources in realtime, to ensure that these resources are utilized to their best ability. Furthermore, the software architecture needs to understand the policy that is related to each call (the importance and prioritization of a call), the requirements of bandwidth of each call being placed, and where the respondents of a call are geographically present. Via knowing this, the software platform can easily make use of local resources rather than call signaling and redirecting data streams to resources that are a distance. The software platform must also be capable to quickly detect any hardware failure or any communication loss throughout infrastructure links, to ensure that it can redirect trafc and reestablish calls using alternative resources, with no need to overly affecting video calls or their quality [24].

When systems on a variety of customer premises attempt to join the exact video call making use of devices that run different protocols (for example SIP, RTV or H.323), the video conferencing platform should first perform conversion process protocol for a popular language so the infrastructure is able to realize and process data efficiently. In other meaning, the software platform must offer intrinsic gateway functionality in between devices that conversation with different languages [24].

2.9 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, they 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 [28].

The WebRTC (figure 2.2) 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 JavaScript API. Web applications can use the standardized JavaScript API to access the functionality of WebRTC [29].

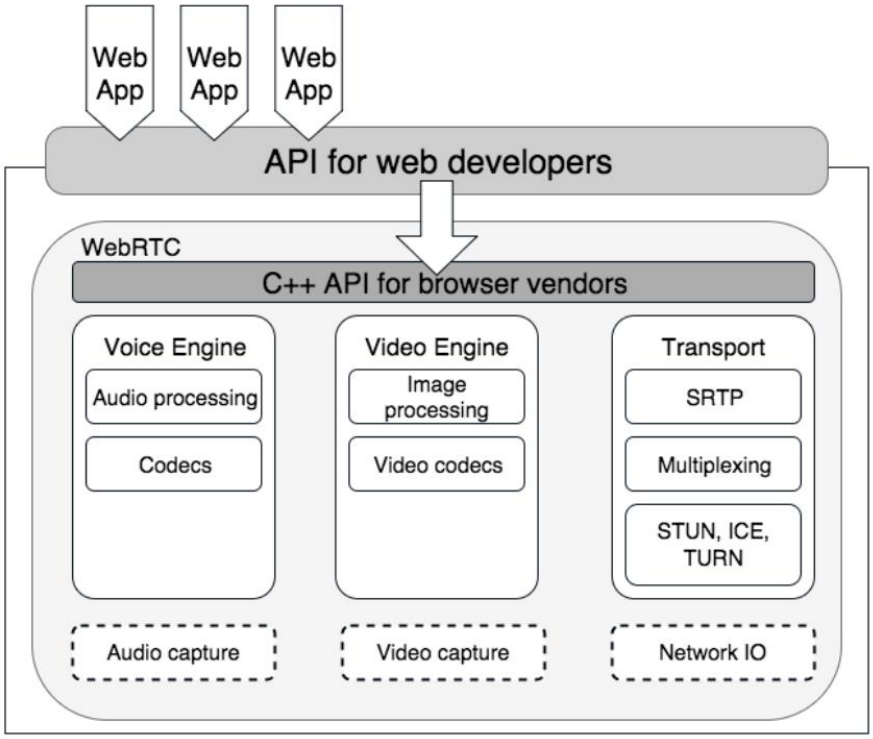


Figure 2.2. Simplified WebRTC Architecture, image inspired by [29].

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.org, an open-source project maintained by Google, Mozilla, Opera and others. There are other open-source implementations of WebRTC, such as Ericsson's OpenWebRTC [30].

CHAPTER 3

THE PROPOSED SYSTEM DESIGN AND ALGORITHMS

This chapter presents a procedure for designing Video Conference Website, named as VCS (Video Conference Site) which allows users securely to make conversation in real-time without installing any software.

3.1 The Proposed VCS Architecture

Basic network architecture of VCS application is a client/server-based system. There are two main implemented parts of this network design: video Streaming Server and Client Computer, as shown in Figure 3.1:

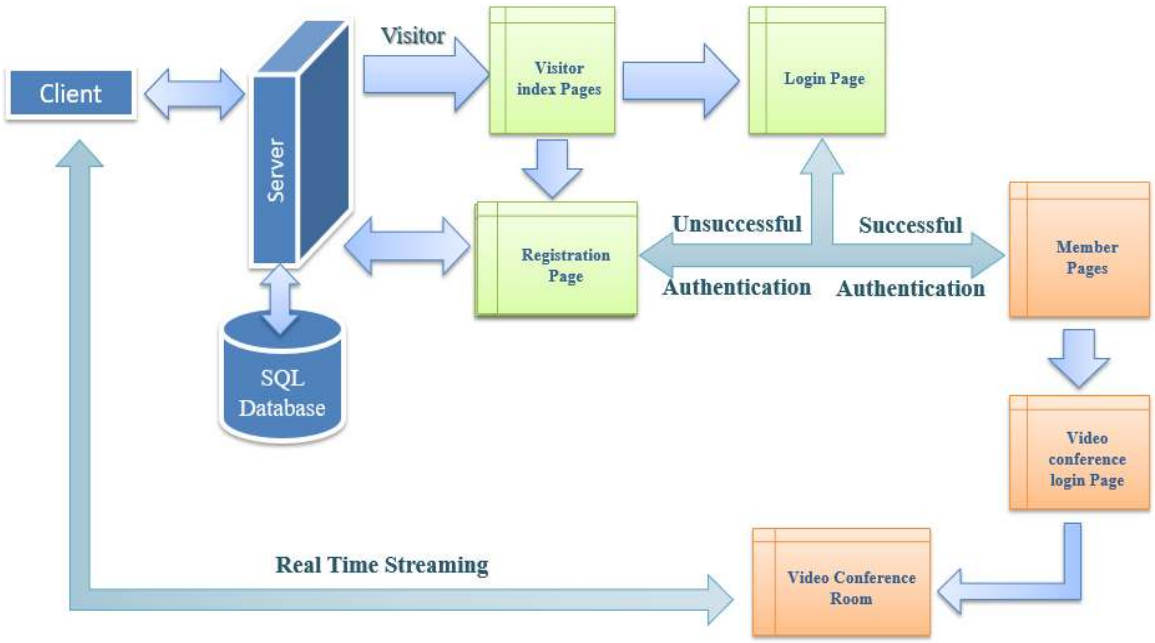


Figure 3.1. The Proposed VCS Architecture.

On the server side, there is a website designed with ASP.NET using visual C# .NET. The VCS has two levels 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 allows 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 have a unique Name and each member have unique login name. Figure 3.2 shows the videoconference page.

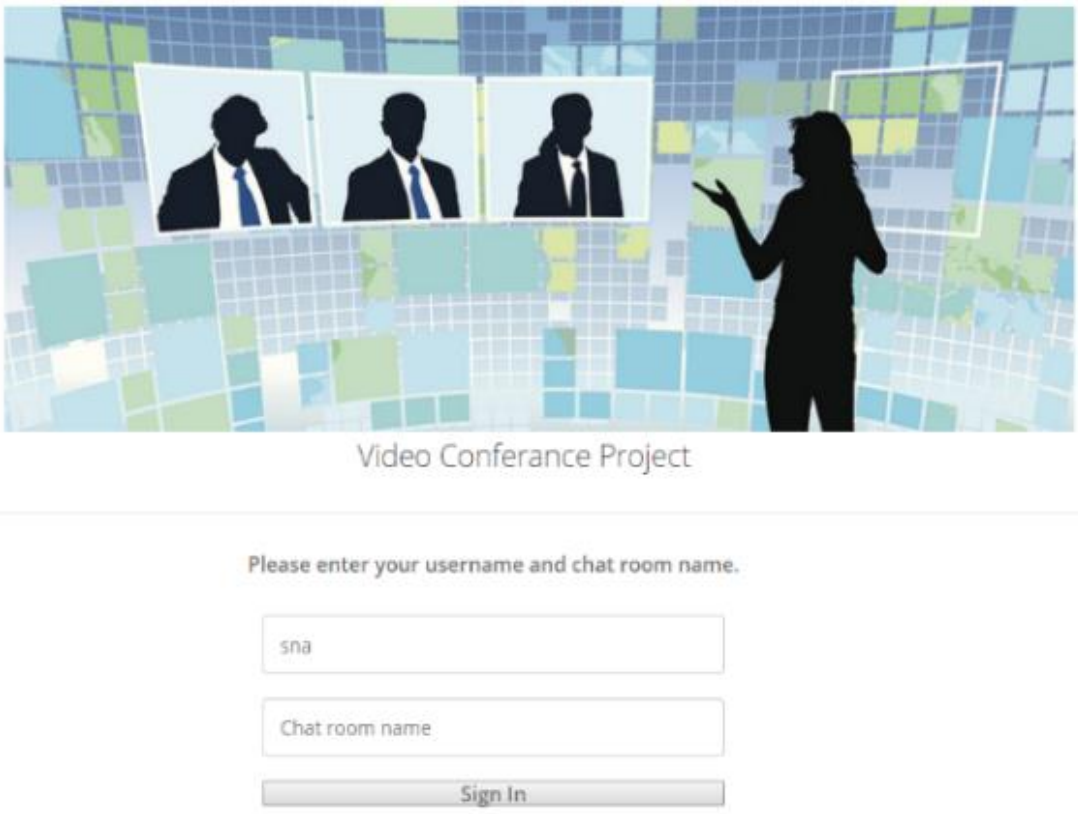


Figure 3.2. Videoconference page.

After login to chatroom, the members can start videoconference with each other. The chatting room structure (shown in figure 3.3) 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.

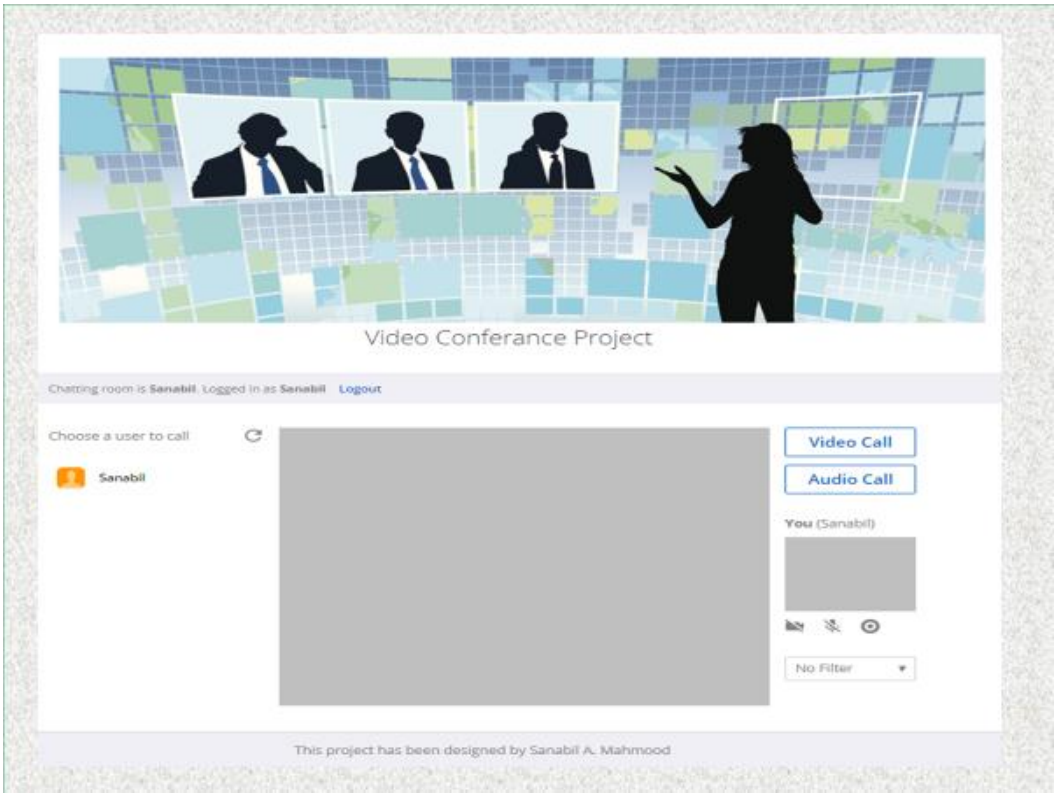


Figure 3.3. videoconference room.

The security of VCS can be described as follows:

1- **Code-behind Page:** The site is designed by ASP.NET by using C# language. The code behind files and raw aspx files are protected from being retrieved by the web server, which means that all main operations run in server-side and the user views just UI and cannot view the coded (.cs files) and cannot know what happens in the server.

2- **User Authentication and Authorization:** Site Authorization has been used to identify the user level (visitor or member). In visitor level, the user can access to the visitor's pages such as index, login, and register pages but cannot access the member pages but only the activated members have access to member pages and can enter videoconference room. In order to enter conference page, the user needs to enter authentication process to be redirected to the member pages.

3- **Password Encryption:** The password has been encrypted in both client side and server side in order to protect the transferred password from attacked by different attacks such as, man-in-the-middle, brute-Force and dictionary Attack, Rainbow Table, Lookup and Reverse Lookup Tables, SQL injection, etc., this is done by encrypted password by used SHA512 hashing algorithm.

The responsibilities of Video Server are to register or authorize users, to verify their identity in order, to allow them to make request for data they require, only member user that has the permission to access to video conference page and makes video chat.

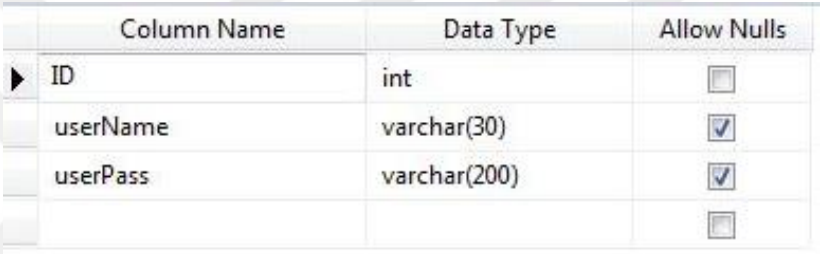
3.1.1 Server-Side Managed Code and Design

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

3.1.2 Proposed VCS Database

The proposed VCS database is designed by using SQL server 2018. The database consists of five tables:

1. **Administrator table:** This table will store login information of administrator. It consists of three rows; id: the row number, username: Administrator account name and AdminPass: account Administrator password (shown in Figure 3.4).



Column Name	Data Type	Allow Nulls
ID	int	☐
userName	varchar(30)	☑
userPass	varchar(200)	☑
		☐

Figure 3.4. Administrator Table Design

AdminPass size has been set to 200 because the password saved is encrypted password with to big size.

2. **UserRegister Table:** This table will store user registration information (as shown in Figure 3.5). It consists of 7 rows: ID: row number, Name: member account name, Pass: member account password, InstituteName: name of Institute that the member works in, InstituteID: the ID number of member institute page, JobTitle: the job title of member in his institute, Email: user registered email.

Name	Pass	InstituteName	InstituteID	JobTitle	Email
Mohammed	OGU458oF6GwNYIXJPBR9lzS42cRyr9lxU/w3jVAedUo6j1Fvxyg...	Baghdad Unive...	BG1234	Engineer	mohammed@...
Halla	BFk05O/MKhGXSncKiDPD8fnZohnJuYJFBQqnKoV8V4k0KYdxo...	Baghdad Unive...	Ha1234	Teacher	Halla@gmail.c...
Sanabil	+uuYcTEct3gcudx+apz0S7zxf5an/mgsKSju1t6Ar+lpiqjqN2W9...	GAZ?ANTEP U...	Sn1234	MSc. Student	Sanablhadid75...

Figure 3.5. User Register Table Design

UserPass size has been set to 200 to be able to store long size of hash of the password that is encrypted in server-password encryption.

3.1.3 User Authentication and Authorization

In VCS, the visitor can access to the website basic information and to register to be member or login if he already member, the authorized client can have access to videoconference page and make video conversation. When being a member of the server, then the member will need providing some of it is private information to the server like names, addresses, Identity card number, etc. Server will store all the information, maintain a complete log of all the registered users waiting from the site manger to activate their accounts once checking their certain information, and every time they attempt to sign in, they are validated by their certain existing information such as username and password. In the proposed VCS, the security has two side of encryptions, one on the client side to encrypt password and send it to server, then the other decrypts password and encrypts it again in server**.**

A. Password Encryption Algorithm

The proposed VCS system uses a SHA512 hashed with salted algorithm in order to encrypt password. The algorithm scheme includes the following steps to encrypt the password:

The Password Encryption Algorithm

Input:	Password (Text)
Output:	Encrypted Password Hash Value (Text)

Step 1: Start.

Step 2: Using the encryption algorithm (SHA512). This operation gets an input string and produces a one-way hash by using the “SHA512" hash algorithm.

Step 3: Generating a truly random number in range between 8 and 24 by used a Class library RNG Crypto Service Provider. This will be used to identify the random salt length.

Step 4: Used the variable length to generates a random salt.

Step 5: The user input “password” is then encrypted by using the random salt with SHA512 hashing algorithm., the SHA512 algorithm runs 1000 (or more) passes through the hashed output to offer a higher level of security.

Step 6: Generating the password and verifying functions.

Step 7: End.

In summary, the encrypt function performs little steps. It first changes the user input string "password" to bytes. Then, it inserts the unencrypted salt bytes at the start of the user bytes. Then, it uses the SHA512 hash algorithm, that can run 1000 passes over the array of (salt+input) byte to create a strong hash. Then, it appends this salt to the hashed encrypted (input+salt) byte array as well as places the first byte to keep the salt length. Finally, it then changes this to the Base64 encoded string and provides it to the user

CHAPTER 4

THE PROPOSED SYSTEM IMPLEMENTATION AND RESULTS

This chapter reviews the site operation and testing the security achieved by the proposed VCS (video conference site) website.

4.1 The Proposed VCS Site Overview

The proposed VCS is designed with visual studio 2017 using C# language. The site contains many folders and items as shown in figure (4.1).

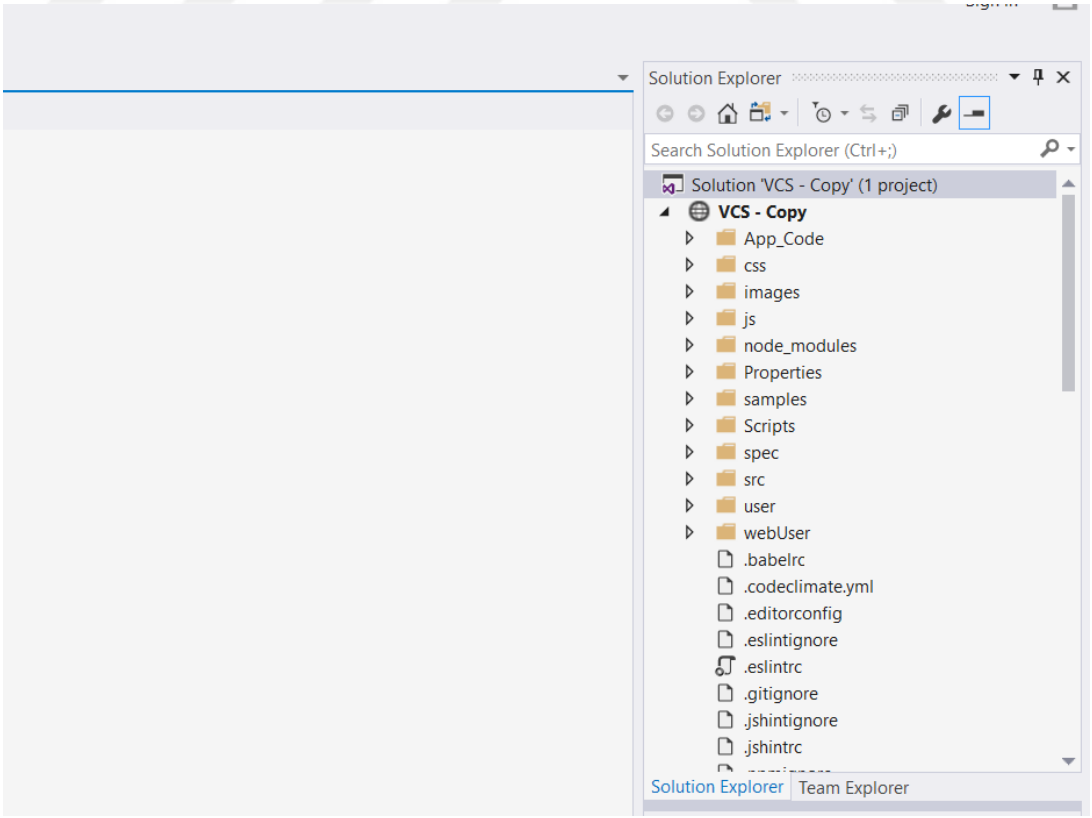


Figure 4.1. The Proposed VCS Designed by Visual Studio

The folder App_code contains the classes, which are the codes that controls most sites operating, it contains five classes (as shown in figure 4.2):

1- Hash: Contains the algorithms controls server-side password encryption.

2- OperateData: control the connection with SQL database.

3- OperateMethod: control the operation of uploading, converting and playing video files.

4- js: java script codes for controls the videoconference room.

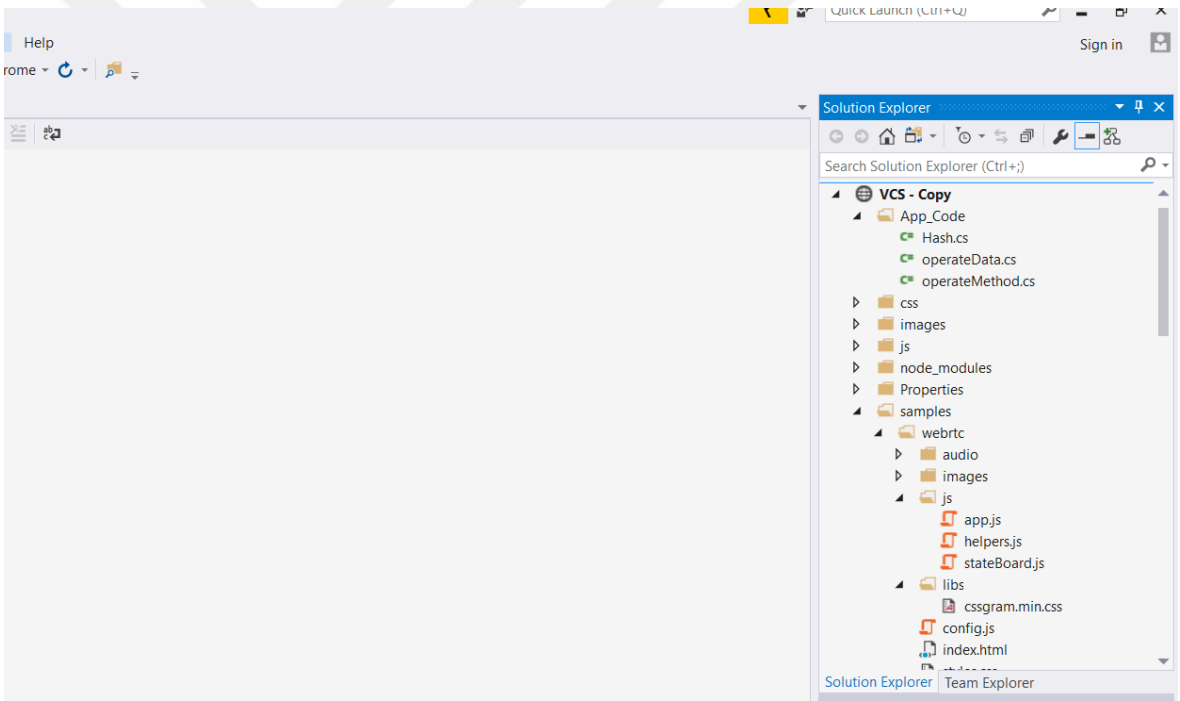


Figure 4.2. Application Code

The site consisted of three master pages which represent the three level of Authorizing (Visitor and Member) as follows:

1- MasterPage: This is the master page of visitor, it makes pages that connected to it allowable for each one. It gives an idea about the proposed

VCS site and contains obtainable video that can be viewed by visitors, also it contains links to membership library, login page, register page and contact information.

2- UserMasterPage: This is a master page of the member, it makes the pages that connected to it and allowable just for members and administrator only. All pages of the members are coded to prevent any one to enter the page without authentication.

The *css* folder contains the style page responsible on the site appearance (as shown in figure 4.3). An *images/img* folder contains the site images used for UI. A *js* folder contains java script code files that control the UI like menus, images slider, etc. A *manage* folder contains administrator pages. The *user* folder contains member upload pages.

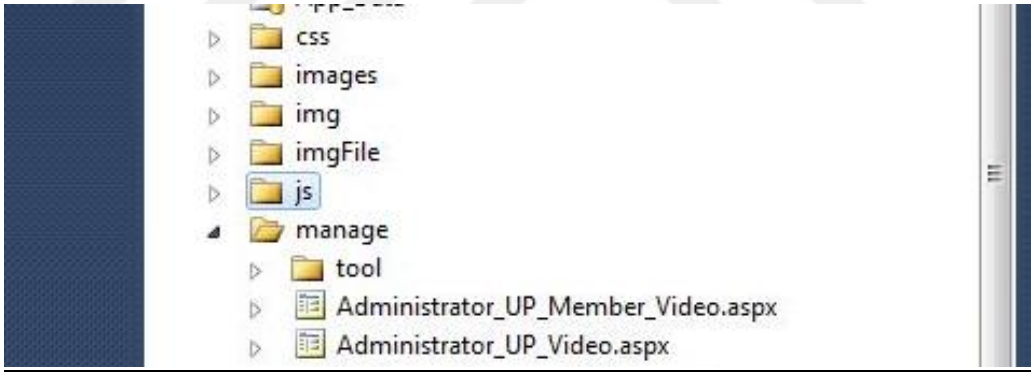


Figure 4.3. Main Folders of Proposed VCS Site

4.2 The Proposed VCS Database Structure

The proposed VCS database is designed by using SQL server 2018. The database consists of five tables:

1. **UserRegister Table:** This table will store user registration information (as shown in figure 4.5). It consistes of 9 rows: ID: row number, Name: member account name, Pass: member account passward, InstituteName:

name of Institute that member works in it, InstituteID: the ID number of member institute page, JobTitle: the job title of member in his institute, email.

Name	Pass	InstituteName	InstituteID	JobTitle	Email
Mohammed	OGU458oF6GwNYIXJPBR9IzS42cRyr9IxU/w3jVAedUo6j1Fvxyg...	Baghdad Unive...	BG1234	Engineer	mohammed@...
Halla	BFk05O/MKhGXSncKiDPD8fnZohnJuYJFBQqnKoV8V4k0KYdxo...	Baghdad Unive...	Ha1234	Teacher	Halla@gmail.c...
Sanabil	+uuYcTEct3gcudx+apz0S7zxf5an/mgsKSju1t6Ar+IpiqjqN2W9...	GAZ?ANTEP U...	Sn1234	MSc. Student	Sanablhadid75...

Figure 4.4. UserRegister Table Design

UserPass size has been set to 200 to be able to story long size hashed (about) of the password that encrypted in server-password encryption.

4.3 The proposed VCS Site Navigation

The proposed VCS site consists of three levels (Visitor, Member and administrator).

4.3.1 Visitors Level

This level is allowable for everyone, it is the default web address of proposed VCS site and the path for the other levels, it contains information about site and the pages to registers and login and contain obtainable videos library that available for everyone. This level consists of four pages:

1- **Index(Home) page:** it's the window of the proposed VCS site and the main URL address, it contains a links for login and registration pages (as shown in figure 4.5).

Figure 4.5. Visitor Home Page

2- **Register Page:** this page is an electronic form that user must full it to have account in the proposed VCS site (as shown in figure 4.6).

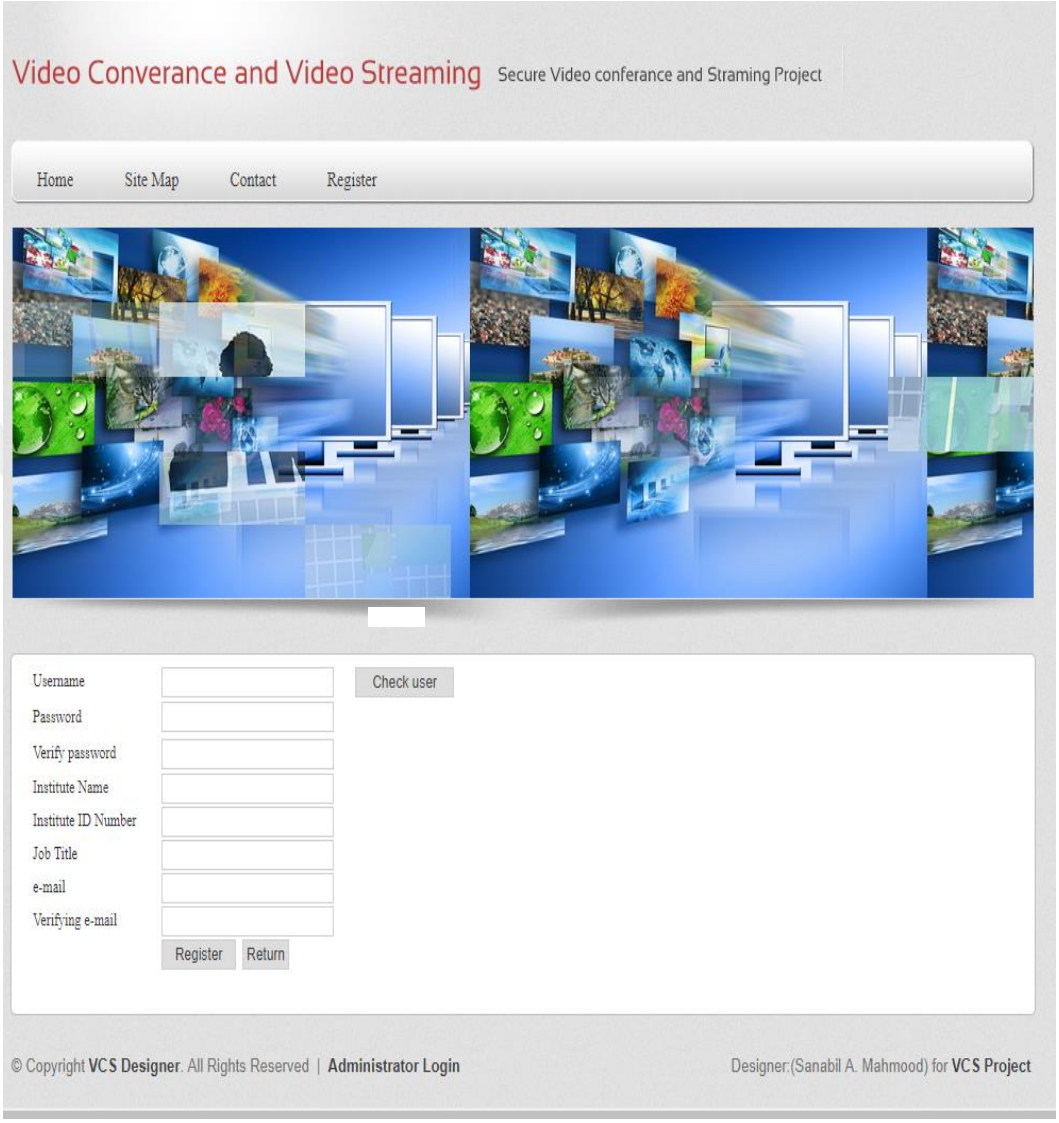


Figure 4.6. Proposed VCS Register page

3- **Member Login Page:** this page is the portal page to member pages that verified if the client is allowed or not allowed to enter member pages (as shown in figure 4.7). It includs Reverse Turing Test, which applied before authentication to prevent automated attacks on passwords.

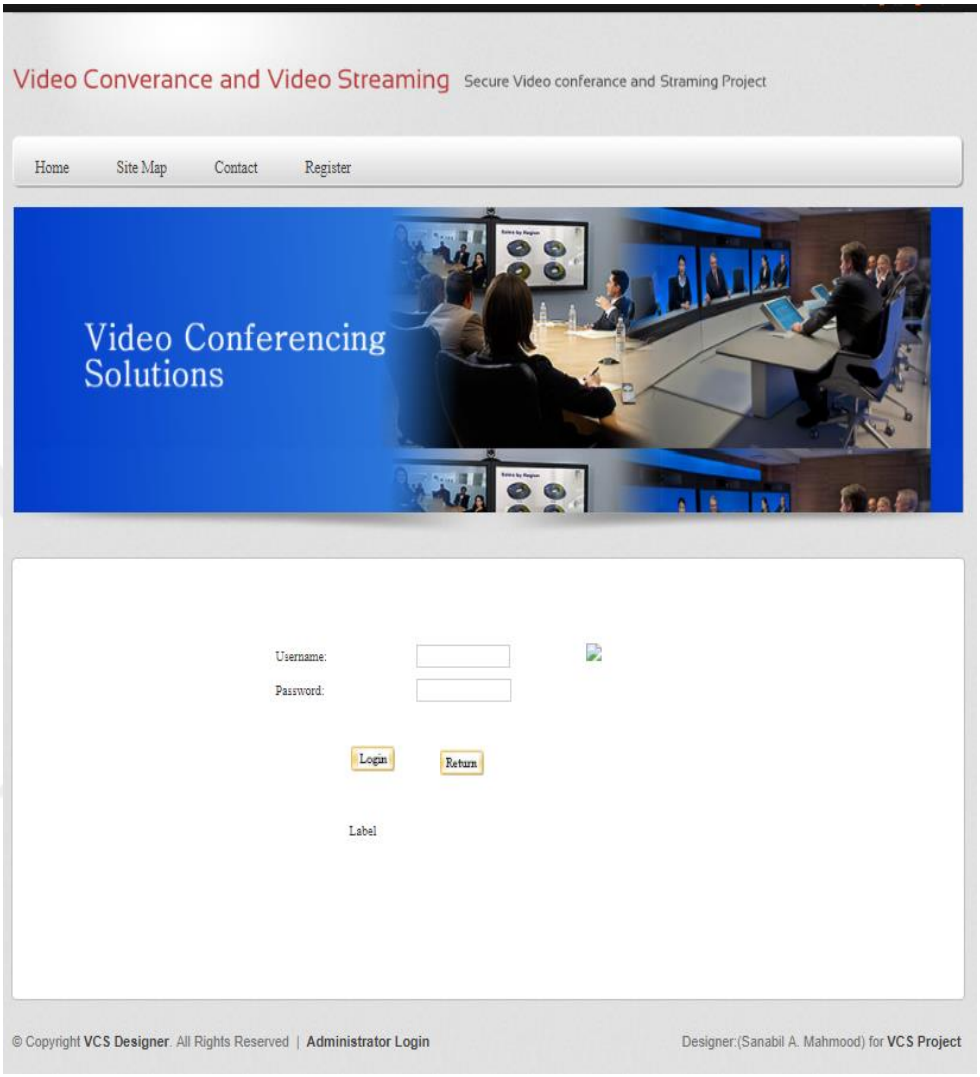


Figure 4.7. proposed VCS Member Login page

4- **Contact Page:** this page has information of contact with site administrator like e-mails, telephone numbers, or work address.

4.3.2 Member Level

This level is allowable just for the activated members after passing authentication process. It contains main web pages of proposed VCS site and can access to videoconference page. This video conferanse page shown in figure 4.8.

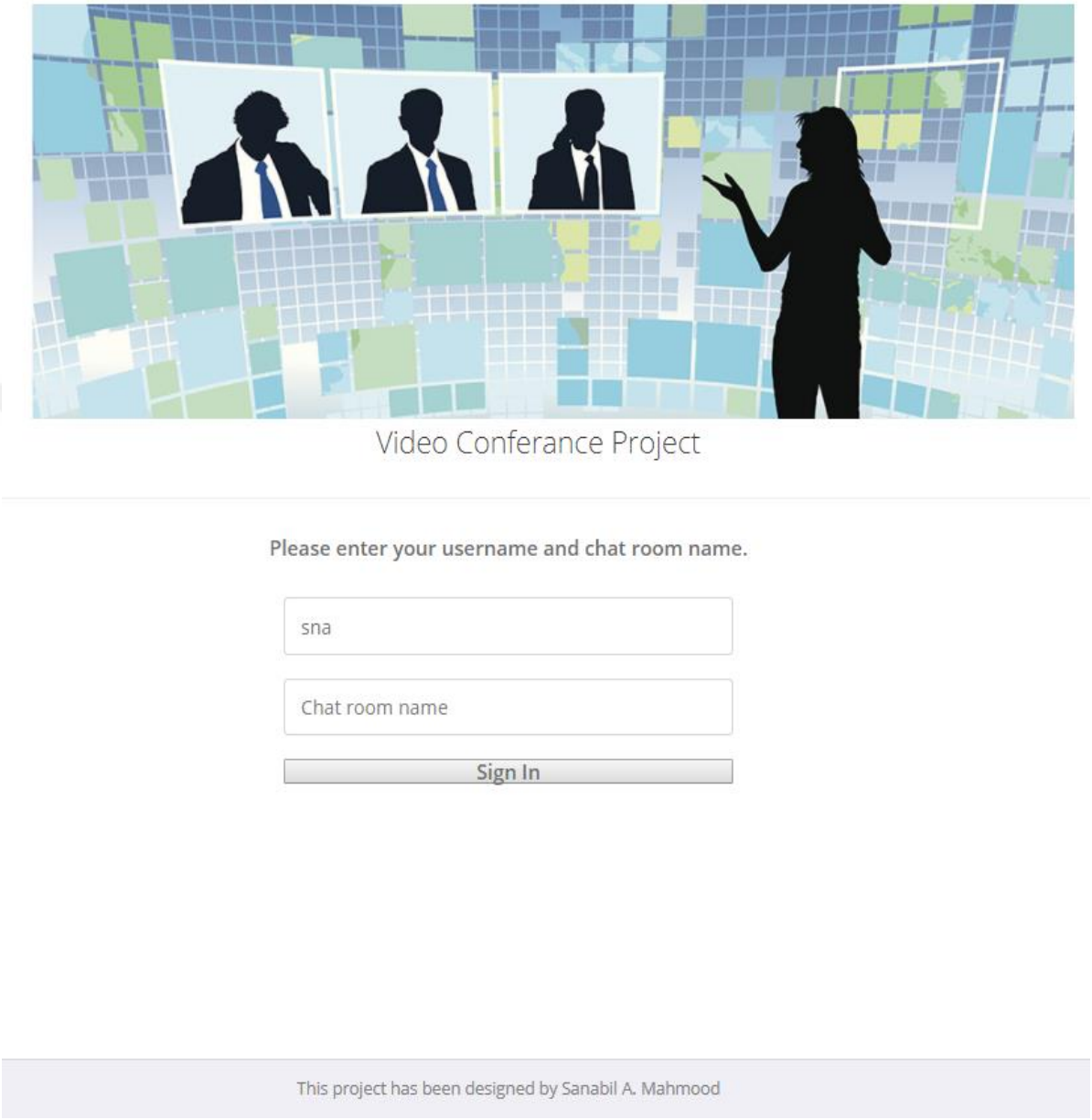


Figure 4.8. videoconference page Login page

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

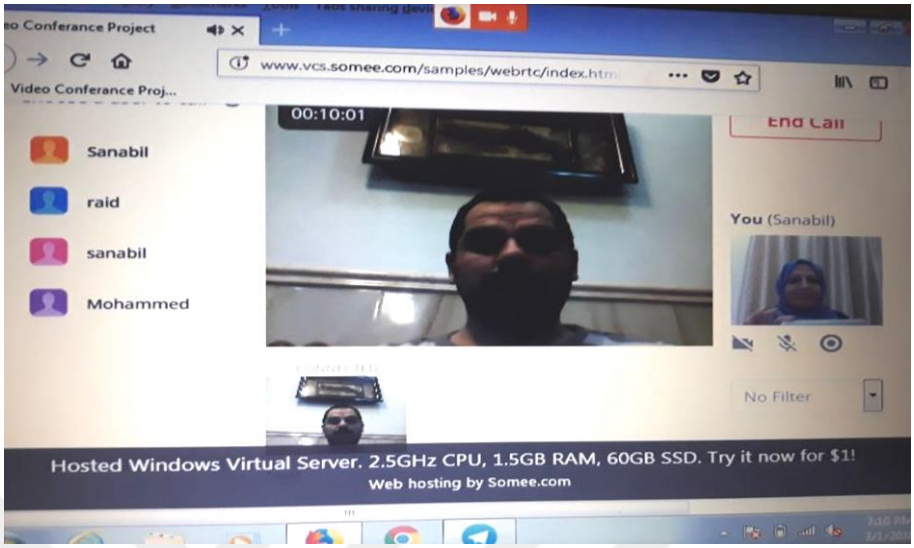
member side so no one can know what the member key is, thus VCS offer privacy to me member. Figure 4.9 shows the encrypted key saved in SQL database

Name	Pass
Mohammed	OGU458oF6GwNYIXJPBR9lzS42cRyr9lxU/w3jVAedUo6j1Fvxyg...
Halla	BFk05O/MKhGXSncKiDPD8fnZohnJuYJFBQqnKoV8V4k0KYdxo...
Sanabil	+uuYcTEct3gcudx+apz0S7zxf5an/mgsKSju1t6Ar+lpiqjqN2W9...

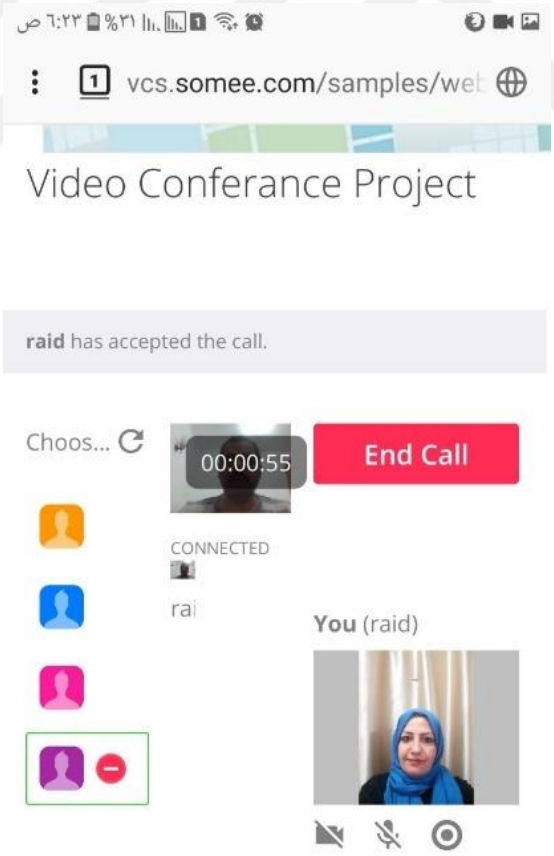
Figure 4.9. VCS SQL database registration information

4.4 The Proposed VCS Operation Test

For videoconference operation test, the VCS has been tested to work in different operation systems and devices. First, we test connection between computers have windows operation system with different version (windows 7, windows 8 and windows 10), then test the operation between MAC computer operate under MAC OSX operation system. The other test in between Android smartphones and between apple iPhone and between the two. Also, we test the connection between different devices in same times. The other test is to check the combability with common internet browsers (Internet Explorer, Safari, Google Chrome, Mozilla Firefox and smartphone default browsers). The test results are shown in table 4.1. Figure 4.10 shows the online calling between two peoples in video conference room



(a)



(b)

Figure 4.10. The video conference room when connection between (a) computer run by windows 7 operation system, (b) android phone**.**

Table 4.1: The ability of VCS to run in different devices, operation systems and internet browsers.

Device	Operation System	Internet Browser		
		Default System Browser	Google Chrome	Mozilla Firefox
Standard Computer	Windows 7	Yes	No	Yes
	Windows 8	Yes	No	Yes
	Windows 10	Yes	No	Yes
Mac Computer	OSX	Yes	No	Yes
iPhone/ iPad	Android 5 and up	No	No	Yes
Android Phones/Pads	OX 10	No	No	Yes

From table 4.1, the test results show that the videoconference can run in any device with any operation system so it OS free, 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 ad 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 is 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.

4.5 The Proposed VCS Performance Test

This test used to test the performance of VCS site and it consist of two part of tests, one tested the whole site performance included site load test and its render, second test video site test included video streaming and http streaming loading test, video encryption and decryption test, video conferencing test.

. 4.5.1 Site Performance Test

This test used to verify performance of site by calculating its speed and loading through applying multiple requests and calculating load Time and rendering time. This test done by used special websites that tested the site performance.

1. **Pingdom**

   Pingdom is a company founded by Sam Nurmi. Pingdom has a narrow focus and highly strong. That focus is placed on covering the uptime screening needs of 90% of the organizations in the world which keeping the better uptime monitoring services that are available. The pingdom site included site test page that using to test the site performance and its speed. Figure 4.11 shows the Pingdom page.

   URL site link: https://www.pingdom.com/

Figure 4.11. Pingdom web site

Test Result:

By putted the site address in website URL field and then pressed the "Test Now" button, site start its test and after finishing the test it will appear Tested results. The tested result appeared that the VCS site performance in high level which its grade about (78/100) and loading time about (768ms) for 38 request, and the VCS site be faster than 93% of all tested website as shown in figure (4.12).

Figure 4.12 Pingdom VCS site Tested results

2. WebPagetest

WebPagetest is an open source website project that is mainly being supported and developed by Google as a part of work to make the web at a higher speed. WebPagetest is a tool that became primarily developed by AOL for internally usage and became an open source in 2008 under the "BSD" license. The program is under active improvement on Google code and can also be packaged up regularly and obtainable to download if one like to run own instance. The online version of webpagetest.org can run via the WPO Foundation for the advantage of the performance community with many

individuals and companies offering the testing system around the globe. Figure 4.13 shows the WebPagetest site page.

URL site link: http://www.mavitunasecurity.com/

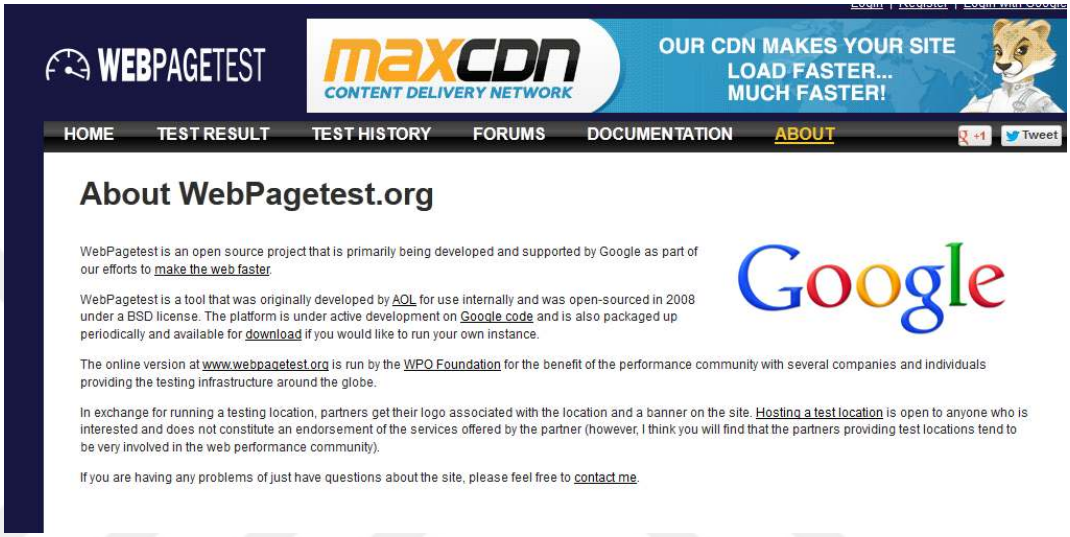


Figure 4.13 WebPagetest site page

Test Result:

By putted the site address in website URL field and then pressed the “start test” button, site start its test and after finishing the test it will appear Tests results. The test results appear that the VCS site Speed scores be (59/100) and the fully loaded time for 40 Request about (8 second). Figure 4.14 shown test results.

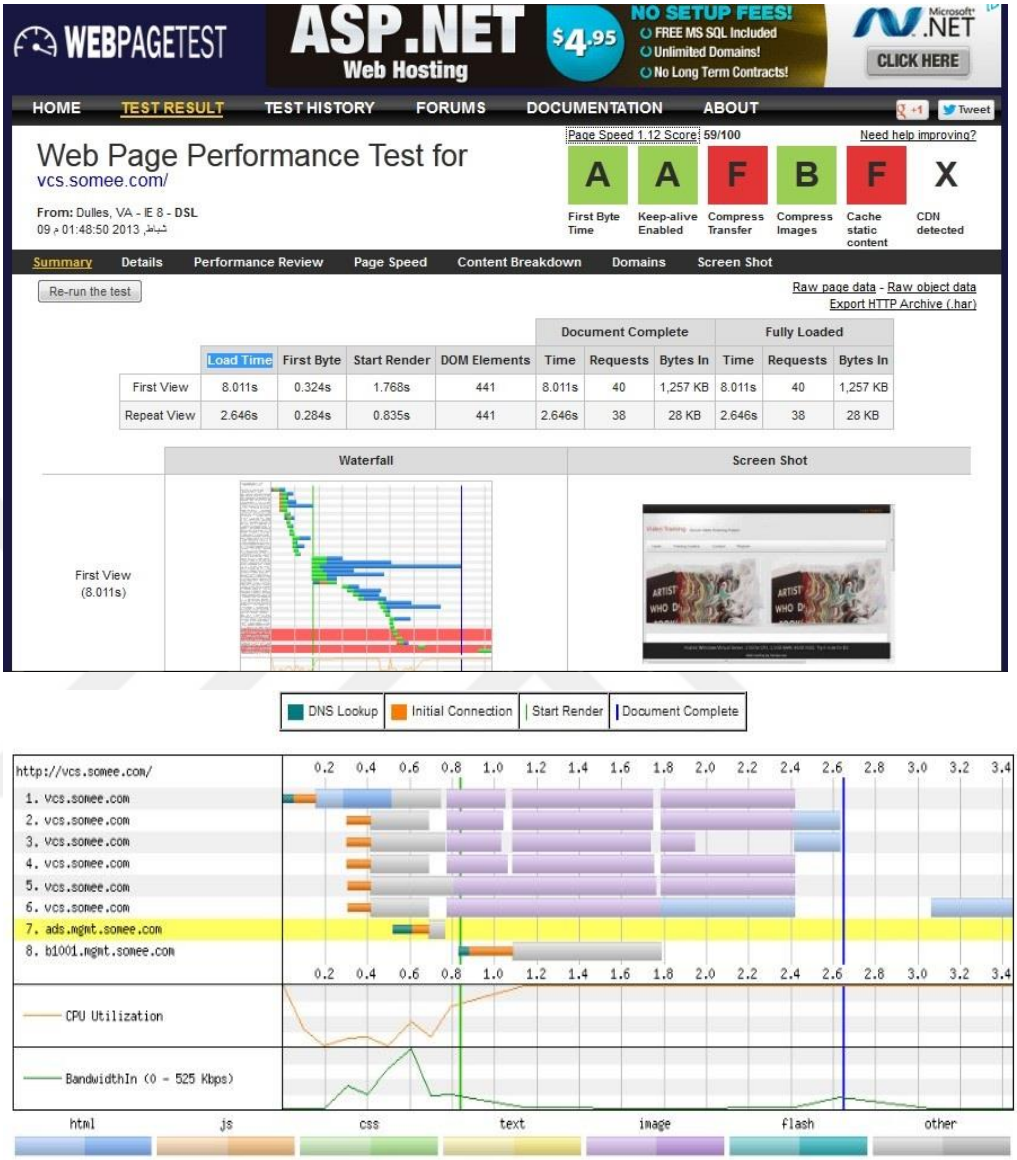


Figure 4.14. WebPagetest VCS site Test results

The test appears that the VCS site had a good speed score and good result for (21 type) but there is (5) points that need to improve to get full performance result that whole focus on compress and decreases sites images to get full performance . Figure 4.15 shown the points need to optimized VCS site

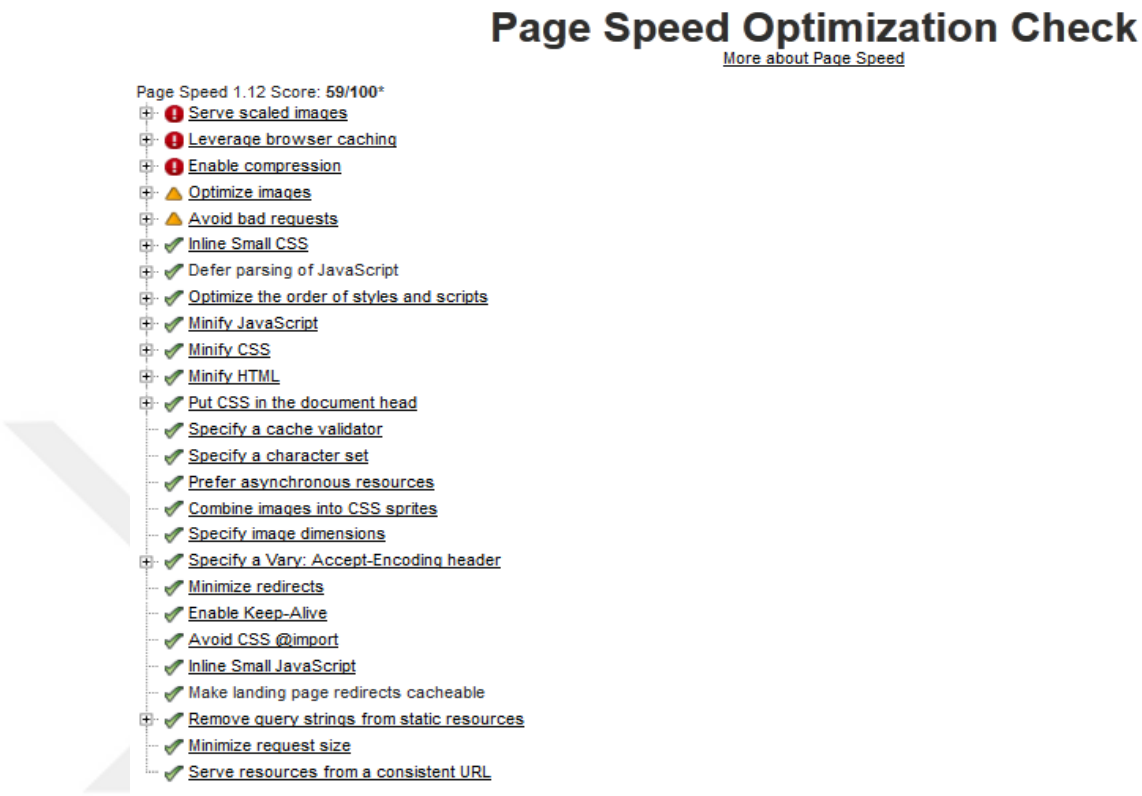


Figure 4.15. WebPagetest page peed optimization Test results.

4.5.2 Site Stress Test

This test was used to find the ability of site in erder to work at high stress conditions(large traffic), because the majority of web applications and websites run correctly and smoothly as long as only one there is one or few users are visiting the site at same times. However, the problems start when a large number of users (i.e thousands or millions) of users tried to access to the web application or website at the same time.

For that test, A special program is used to calculate the web stress names "Webserver Stress Tool", in which it is able to simulate a variety of load patterns for VCS webserver that could help to determine the problems in webserver setup. With gradually increasing the loads (that called "ramp tests") it can be able to determine the amount of load of VCS server can work on before of serious problems. Enterprise version is used for testing the VCS site to simulate 1000 users. Figure 4.16 shows the

Webserver Stress Tool.

URL site link: http://www.paessler.com/

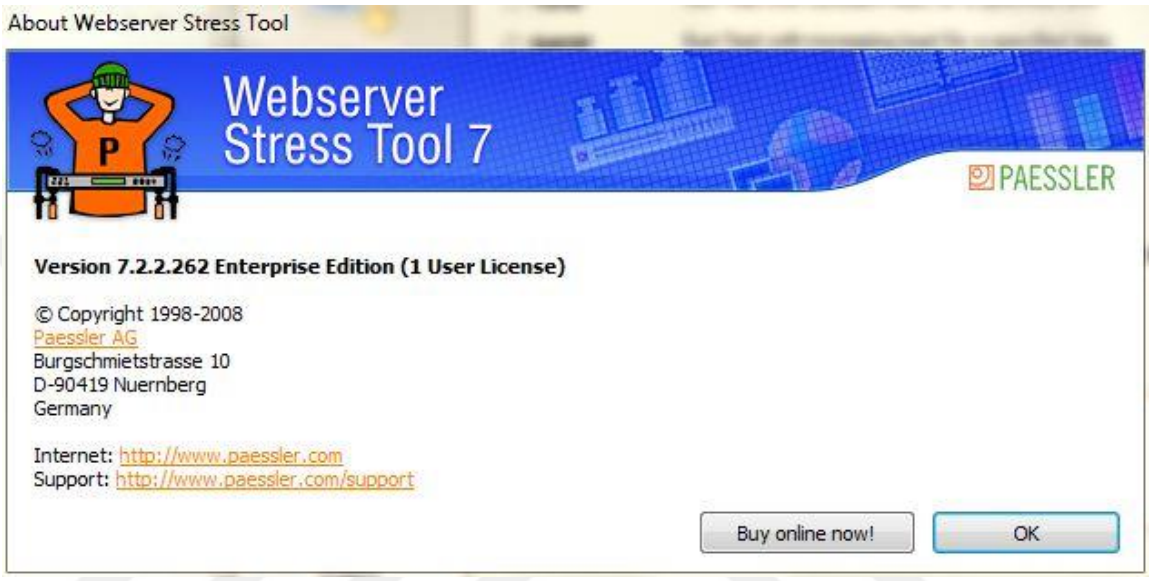


Figure 4.16. Webserver Stress Tool

Test Result:

The parameters were selected to simulate (1000 users) run 1 click in 6 second time with an estimated load of (~ 10,000 pageviews/minute) or (~600,000 pageviews/hour). As shown in figure 4.17.

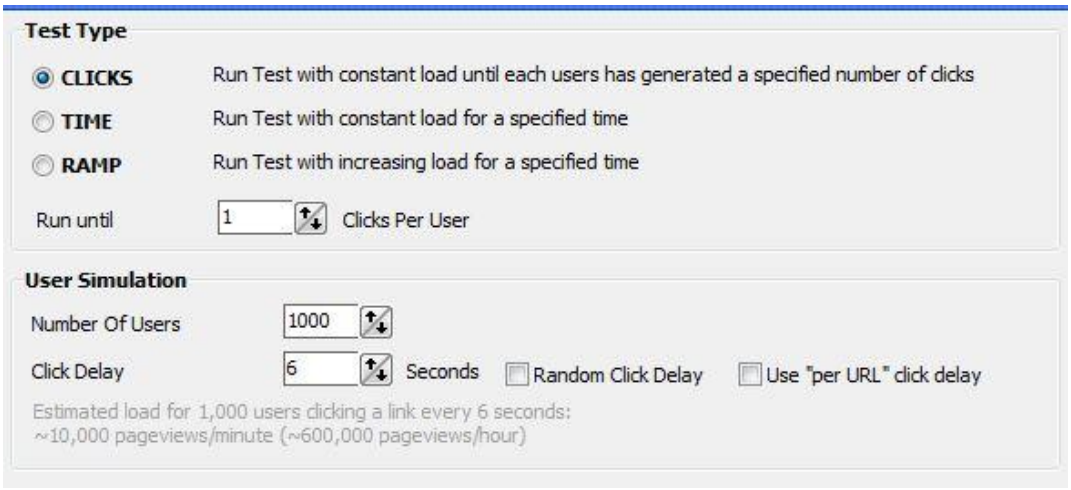


Figure 4.17 Webserver Stress Tool parameters selected for VCS stress Test

The test result show that the VCS site can run smoothly for that very stressed traffic and took just (30%) errors for that very stressed load, this is due to the VCS site using 1Mb/sec bandwidth from host server and can reduce errors for high traffic level by increasing the site bandwidth and host computational resourced. The figures (4.18, 4.19, 4.20, 4.21, 4.22) shows the webserver stress tool test result

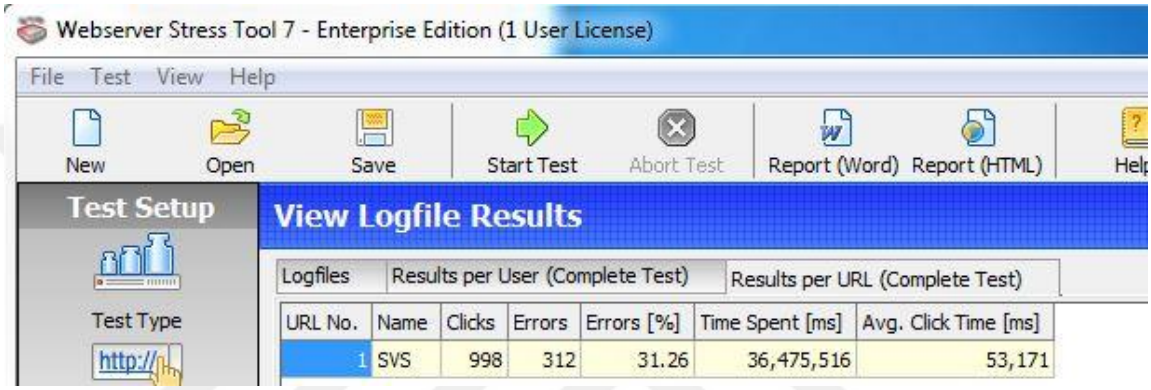


Figure 4.18 Webserver Stress Tool Test result

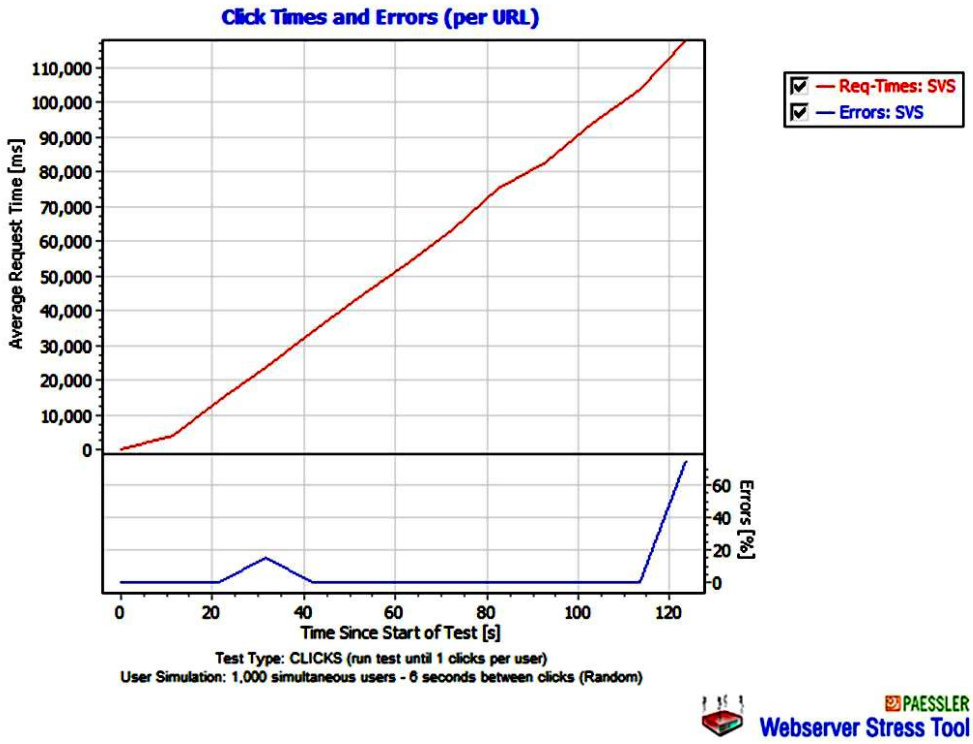


Figure 4.19. Webserver Stress Test, "Click times and errors"

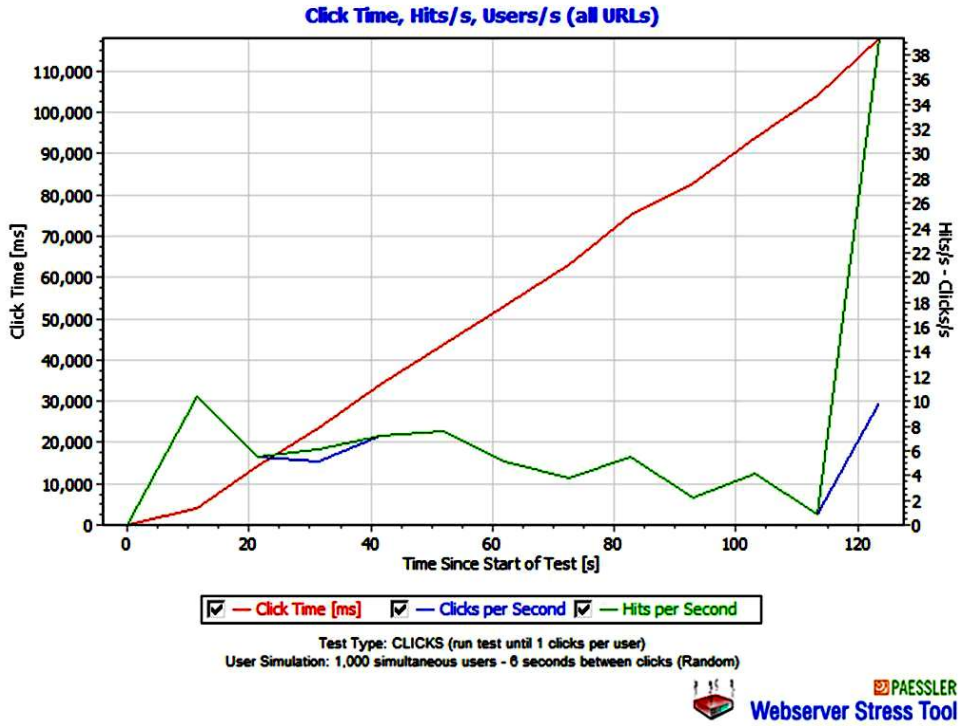


Figure 4.20. Webserver Stress Test, "Click times hits/s, users/s"

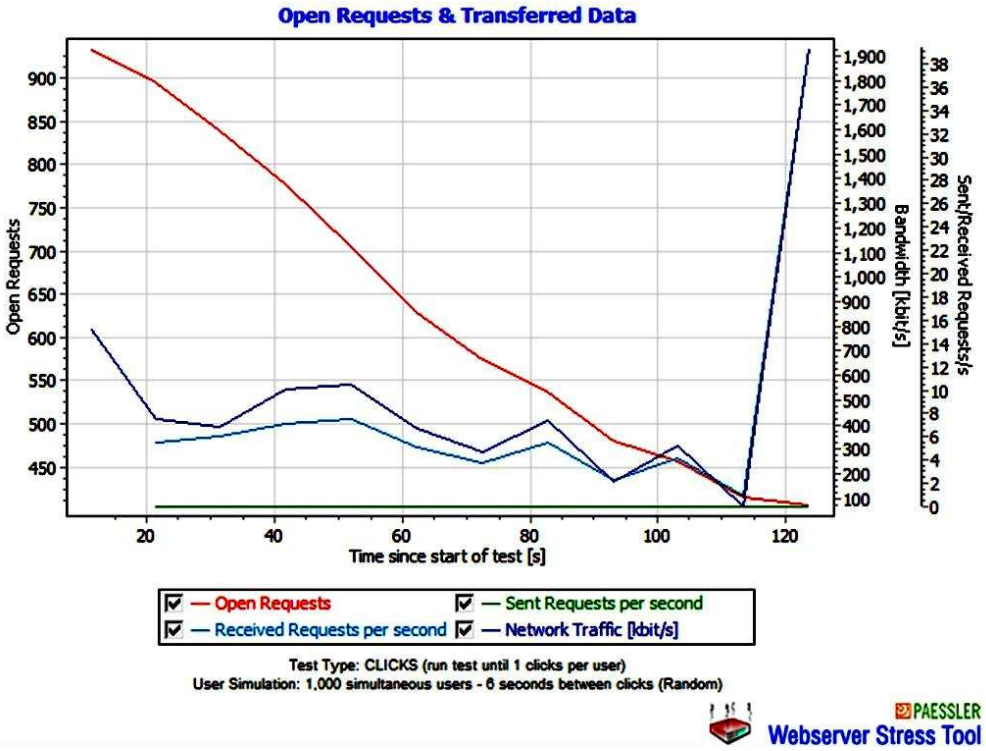


Figure 4.21 Webserver Stress Test, "Open request & transferred Data"

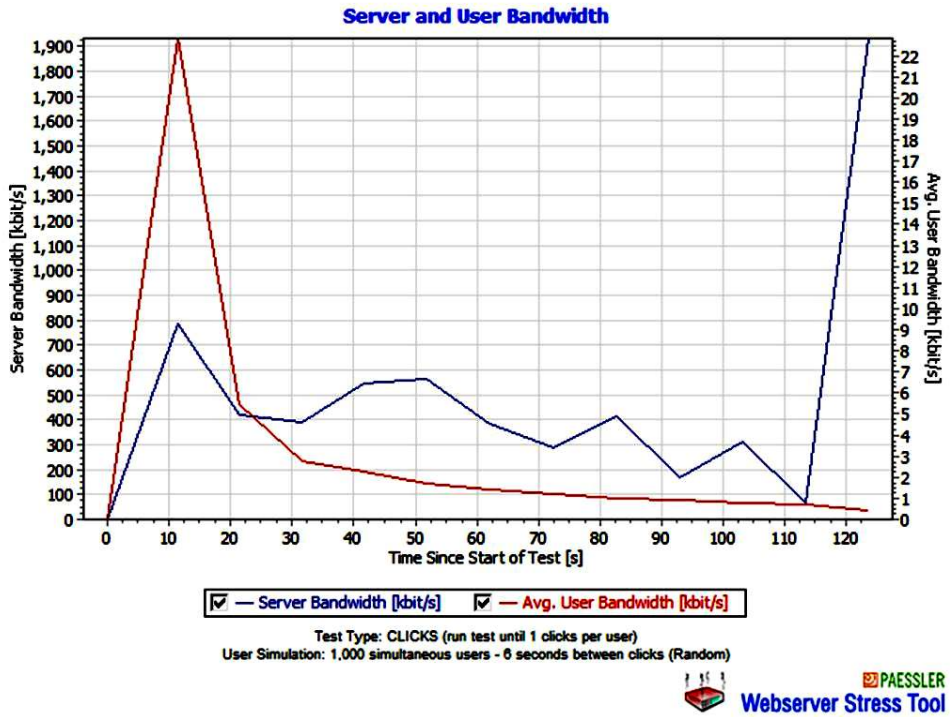


Figure 4.22. Webserver Stress Test

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4.6 The Proposed VCS Security Test

The proposed VCS site was tested with different types of password encryption, database and site attacks using attacking programs, Web Application Security Testing and password cracked site. All these attacks failed to success to broken proposed VCS site security. These programs and sites are not scientific tests but it is the commonly used from most hackers and attackers

4.6.1 Password Test

The proposed VCS hashed password was tested with various types of hash cracking websites and programs. A weak password of the value “San1” has been selected and tested, the results show that no one of the password cracking sites and programs can crackthat password as shown in table 4.2.

Table 4.2: An example for Password value after encryption

Username	Password Value	Password Hash Value
Sanabel	San1	upzCzAYKaIwUvvGDWg++IBH88lcTkgwlUxjsUGqfNyLAE8/Bn5LHbA7yBuv8WgOtp9hrz8wzqEka86UdI/+Ya04nQ8XoVg==

4.6.2 Site Security Test

This test was carried out to verify the security of site "on line" and find the attacking possibility and site breaching using different web-attacking methods. There are many sites can be used to test the security of website online by putting URL address of target site and then the site gives a report about the target site.

A. Websecurify

Websecurify is a highly user friendly and open-source tool that can automatically discovers web application weaknesses by making use of fuzzing and advanced discovery technologies (as shown in figure 4.23). It is able to create basic reports once run.The tool is multilingual and extensile with the add-on support. URL site link: http://www.websecurify.com/

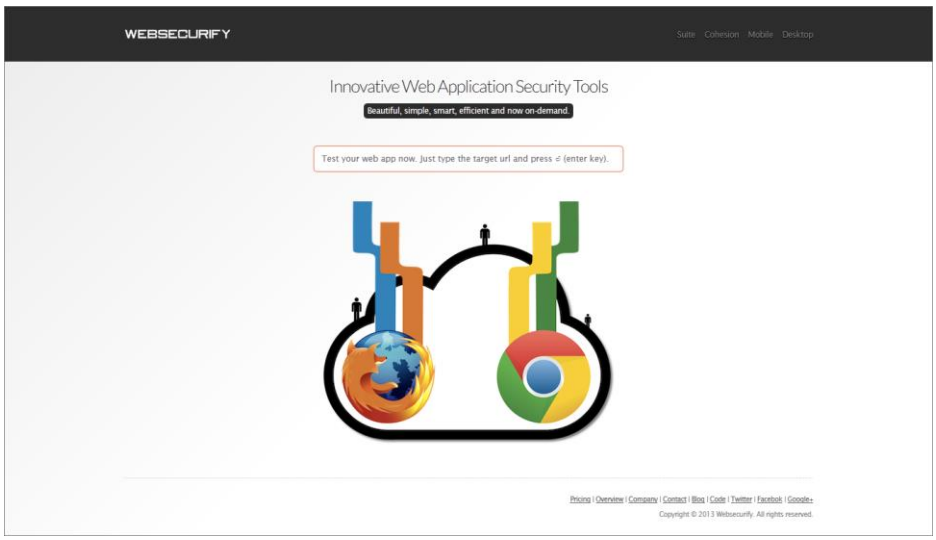

Figure 4.23. Websecurify Home Page

The site contains text box to enter site needed to test its security. The sit need to instill plug in for internet browser before (as shown in figure 4.24). After instilling plug in it then entering site page that need to be tested, after enter site link the warning page appears to be warn about the site maybe damaged after attacking.

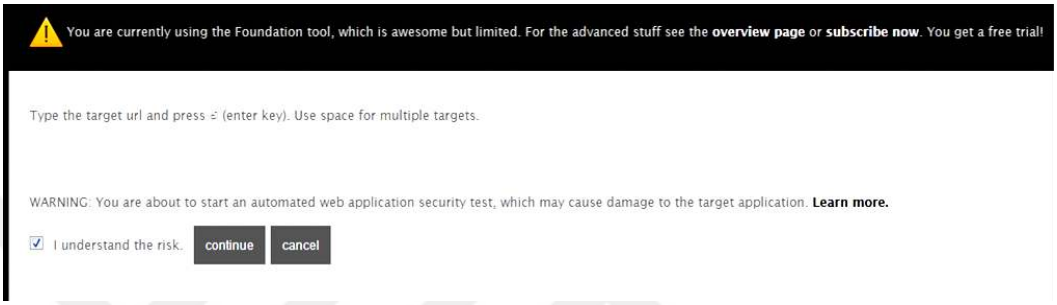

Figure 4.24. Websecurify Instillation

Test Result:

After imported the proposed VCS URL address http://VCS.somee.com, the plugin launched site test; and after running many tests on page. The results appeared that there isn't any gap for this website but there is one notice described as follows:

- **Full Path Disclosure (FPD):** Full Path Disclosure means being capable to see the full path of something located on the server. It is often, but not always, the position of the script itself. The FPD bug is performed by injecting unexpected characters to specific parameters of a web-page. The script does not expect the inserted character and yields an error message that has information of the error, along with the operational path of the targeted script. FPD weaknesses are commonly observed as low risk threats, frequently ignored by web-masters as almost nothing to features of or worry about the scripting language. The test alarm from PD as shown in figure 4.25.

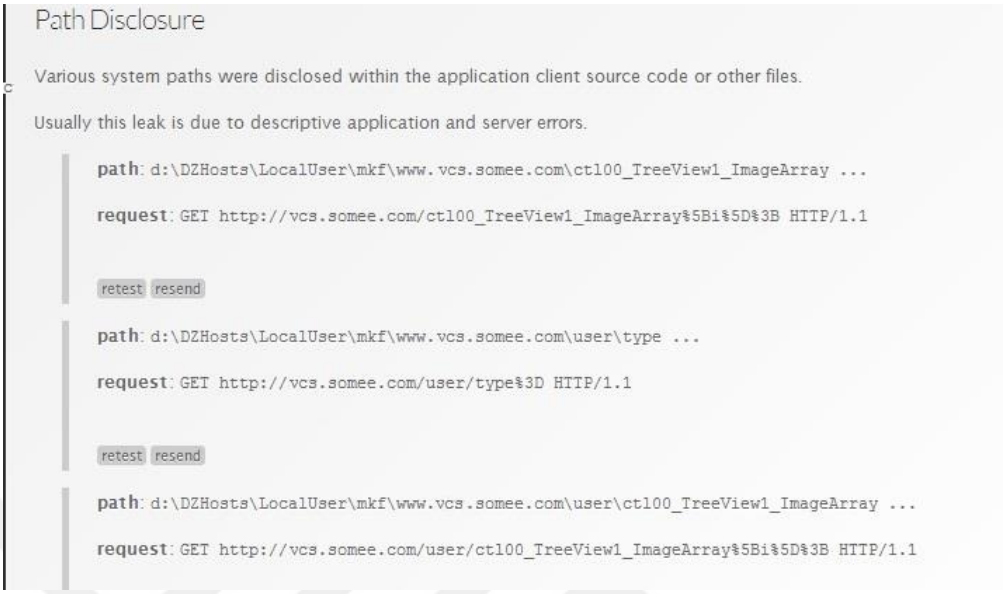

Figure 4.25. Websecurify *Path Disclosure* attention

B. Exploit-Me

This is a security testing tools package of Firefox web application that designed to be simple-to-use and lightweight. Instead of making use of a proxy like many of the security testing tools, this tool is integrating directly into Firefox. It is a highly strong test and it consists of 3 add-ons: SQL Inject Me, that used for testing SQL injection vulnerabilities**,** XSS-Me**,** that used for testing reflected XSS vulnerabilities and Access-Me**,** that used for testing access vulnerabilities.

Test Result:

After instilled these three plug-ins in Firefox and browse the proposed VCS site then run every plugin, the result was as in follows:

XSS-Me Test:

The XSS Test Results shows that No XSS vulnerabilities found, as shown in figure 4.26.

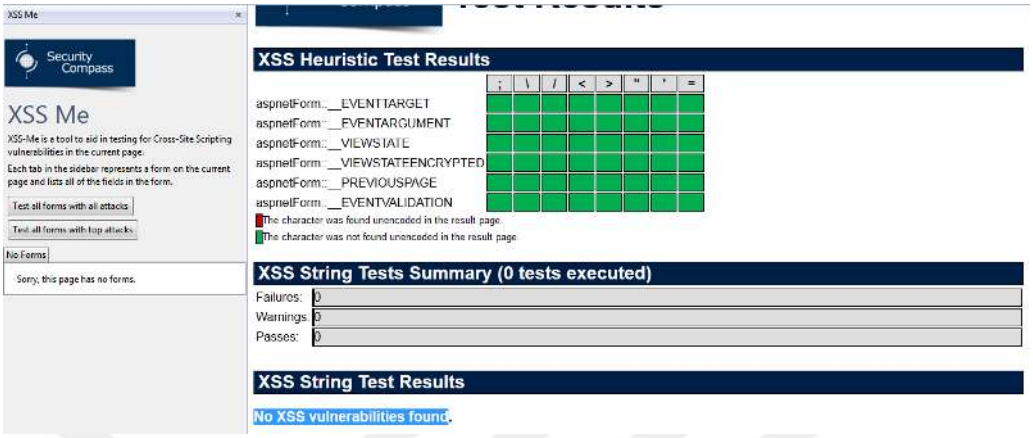

Figure 4.26. XSS-Me Heuristic Test Results

C. SQL Inject Me Test:

The test shows that the VCS site is very good and there isn't warning for weakness in site to be attacks by SQL injection. As shown in figure 4.27

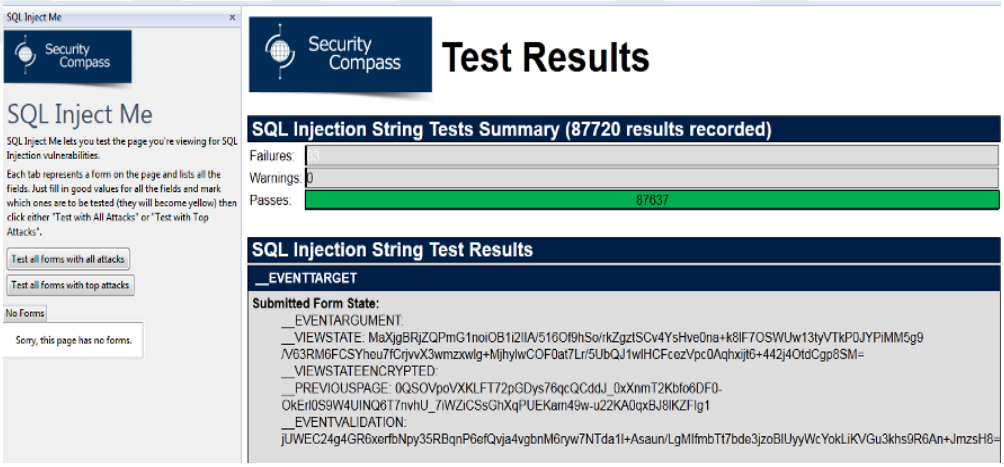

Figure 4.27. SQL Inject Me Test Results

CHAPTER 5

CONCLUSIONS AND RECOMMENDATIONS FOR FUTURE WORK

5.1 Conclusions

This work mainly emphasizes developing an infrastructure of an audio/video conferencing as a website Service that performs single or multiple participants in a video conferencing session using a single connection. The architecture of website video conferencing is based on a WebRTC environments. The proposed model that named (VCS) is designed to be browser-based video conferencing, thus no need of additional software downloads and any one can enter conservation room when log-in to account by used any web browsers from any device at any place of world. The VCS is an ASP.net website designed with three level of authentication with several security techniques in order to secure the conferencing between users. The security used includes an authentication level that prevent non-authorized users to pass to web-conference page.

The experimental result approved that it installation Platform independent, has simple user interface, Secure video streaming with three level of authorization (Gust, Member and admin) , can streaming video as well as run video conference and makes Live video chat with friends. The results show the system work perfectly and the quality and speed of video conference is highly dependable on internet speed of client and streaming bandwidth of the server.

5.2 Suggestions for Future Work

1- Investigate the computational requirement and the internet bandwidth that required from server to operate smoothly.

2- Investigate the ability to using Microsoft Silverlight or Adobe Flash in design the videoconference chat room

REFERENCE

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

2- Koli S. M., Purandare R., Kshirsagar S. P., Gohokar V. V. 2011. A Survey on Video Transmission Using Wireless Technology. Advances in Networks and Communications: First International Conference on Computer Science and Information Technology, CCSIT 2011, Springer, DOI 10.1007/978-3-642-17878-8.

3- Coventry L. Video Conferencing in Higher Education. 2001. SIMA Video Conferencing Outline, Institute for Computer Based Learning, Heriot Watt University, Edinburgh.

4- Panagiotakis S., Kapetanakis K., Malamos A. G. 2013. Architecture for Real Time Communications over the Web. *International Journal of Web Engineering*, 2(1): 1-8, DOI: 10.5923/j.web.20130201.01.

5- Patil J. 2012. Secure Media Streaming for Android. M.Sc. Thesis, The Faculty of the Department of Computer Science, San Jose State University.

6- Macías, J. A., Macías J. A. 2003. Dynamic web page authoring by example using ontology-based domain knowledge. IUI '03 Proceedings of the 8th international conference on Intelligent user interfaces table of contents, ACM New York, NY, USA, pp.133-140.

7- Madsen M. 2015. Static Analysis of Dynamic Languages. PhD Thesis, Faculty of Science and Technology, Aarhus University.

8- Dutta S. 2015. Mastering Commercial Applications. Vikas Publishing House, ISBN: 9789325980327.

9- Tomei L. A. 2009. Lexicon of Online and Distance Learning. R&L Education, ISBN: 9781607092858.

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

11- Forrester. 2011. An Introduction to the Basics of Video Conferencing. Preparing for Uneven Corporate Adoption of Video Communications, polycom.

12- Petrík S., Skala. 2006. Report on Videoconferencing Systems. Technical Report No. DCSE/TR-2006-04, Department of Computer Science and Engineering, University of West Bohemia in Pilsen, Czech Republic.

13- Chokshi S. 2011. Video Chat Application for Facebook. MSc. Thesis, The Faculty of the Department of Computer Science, San José State University.

14- Rodriguez Arriba P., J. C., Trajkovska I., Salvachua J. 2013. Advanced videoconferencing services based on webrtc. In Proceeding of IADIS multi conference on computer science and information systems, pp.1-8.

15- Kwok-Fai N., Ching M. Y., Liu Y., Cai T., Li L., Chou W. 2014. A P2P-MCU Approach to Multi-Party Video Conference with WebRTC. *International Journal of Future Computer and Communication*, **3(5)**, pp.319-324.

16- Pasha M., Shahzad F., Ahmad A. 2016. Analysis of Challenges Faced by WebRTC Videoconferencing and A Remedial Architecture. *International Journal of Computer Science and Information Security (IJCSIS)*, **14(10)**, pp.698-705.

17- Mohan S., Vani V. 2016. Multi-Core Computer Vision and Image Processing for Intelligent Applications Advances in Computational Intelligence and Robotics. IGI Global, ISBN: 9781522508908.

18- Rayburn D. 2016. The Early History of The Streaming Media Industry and The Battle Between Microsoft & Real. Retrieve from Streamingmediablog.com link: https://www.streamingmediablog.com/2016/03/history-of-the-streaming-media-industry.html.

19- Panoiu M., Rat C. L, Panoiu C. 2014. Video Streaming Technologies Using ActiveX and LabVIEW. International Conference on Applied Sciences (ICAS2014), IOP Conf. Series: Materials Science and Engineering, 85(2015), pp.1-6, doi:10.1088/1757-899X/85/1/012023.

20- Schaeffler J. 2013. Digital Video Recorders: DVRs Changing TV and Advertising Forever. Taylor & Francis, ISBN: 9781136030901.

21- Bechqito Y. 2009. High Definition Video Streaming Using H.264 Video Compression. MSc. Thesis, Information Technology, Multimedia Communications, Metropolia.

22- Radha R., Kumar J. U. A. 2012. A Survey on Video Streaming Methods in Multihop Radio Networks. International Journal of Computational Engineering Research (ijceronline.com) **2(6)**, pp.208-212.

23- Video and Media Streaming Servers. MediaCollege.com, Link: https://www.mediacollege.com/video/streaming/server.html.

24- Polycom. 2013. An Introduction to the Basics of Video Conferencing. White Paper, Polycom, Inc.

25- User Services & IT Services. 2013. Videoconferencing Guide. The Technology and Pedagogy of Two-way communication over Geographical distance Version 3.0, University of Malta.

26- Tandberg. 2005. Key Considerations for Maximizing Your Video Architecture: Moving Beyond Point-to-Point Conferencing. Tandberg White Paper.

27- Akia. 2017. Types of video conferencing systems. Meeteasy.com, Link: http://www.meeteasy.com.hk/2017/02/25/types-of-video-conferencing-systems.

28- User Services & IT Services. 2013. Videoconferencing Guide. The Technology and Pedagogy of Two-way communication over Geographical distance Version 3.0, University of Malta.

29- Loreto S., Romano S. P. 2014. Real-Time Communication with WebRTC: Peer-to-Peer in the Browser. O'Reilly Media, Inc., 1st Edition, ISBN: 9781449371852.

30- Hallberg A. 2016. 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.