

## Article 4

# Web Application: Tele-education using Web Real-Time Communication (WebRTC)

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### **Abstract**

*Distance learning can be described as the process of learning between lecturers and students from different locations over the Internet. This paper discusses about a web based video call application using Web Real-Time Communication (WebRTC). It allows lecturers and students to communicate in a real time through the web application for learning purposes. The web application enables users to perform a video call using web browser. However, sometimes the lecturers are not able to attend a class due to unforeseen circumstances. So, they need to find a suitable time to make a replacement class and gather all students to perform a video conference. With the availability of this web application, both students and lecturers can just make a replacement class through video conference. Besides, some students are usually busy with their activities and they do not have time to consult or make appointment with their lecturers. This video call application helps to address the problem by connecting both students and lecturers from different places through a web application. The objectives have been identified, which are to design and develop a web based video call application using WebRTC and to evaluate the performance and functionality of the web application. The result is gathered based on the test conducted, which shows that users will experience a smooth video streaming by using wireless LAN as compared to the broadband, which causes a sluggish video streaming.*

**Keywords:** Web Application, Video call, Real-Time Communication, WebRTC, Learning

### **Introduction**

Nowadays, online learning is a new method of gaining information through a web application. Lecturers can interact with their students for teaching and learning purposes on internet-based learning. Tele-education, also known as distance education, is a situation where students and instructors are physically far away from each other and a two-way communication is included between them (Shepherd & Amoroso, 1998). Video conferencing is a transmission of video and audio back and forth between two or more physically separated locations. This communication can be accomplished through the use of web cameras, microphones and speakers.

The WebRTC is a media engine with Java Script API that has a capability to send and receive a real-time media and data in a peer-to-peer method between web browsers (Loreto, Romano & Pietro, 2012). The WebRTC provides ability of establishing a real-time communication capability such as audio, video and data communication into web browsers without the needs to install additional software (Zeidan, Lehmann & Trick, 2014). Therefore, this paper discuss on developing a web based video calls application using WebRTC.

### **Related Works**

*A. Open Meetings as a Browsers-based Teleconferencing Tool for EFDA Laboratories*

In the past few years, video conference was introduced to worldwide organization as a medium to communicate between organizations from different locations. According to Santos, Castro, Santos, Gomes, Fernandes, Sousa and Varandas, (2011), Open Meetings is open source software that allows video conference via web browsers. Open Meetings does not require installation because it can integrate with current existing tools and customization of requirements.

*B. A video conferencing system based on WebRTC*

The WebRTC is a browser based video chatting that does not require any third parties software or plug in. According to Chiang, Chen, Tsai and Yuan (2014), the WebRTC is an open project that grants a web browser with real-time communication (RTC) capabilities. There are many types of application programming interface that can be used to develop video chat application. To obtain and exchange media stream browsers, two API were needed, which are Media Stream and RTC Peer Connection. Media Stream is a type of API that functions to get audio or video from media stream from end device such as Webcam. RTC Peer Connection play an important role in providing connection establishment between two end users.

*C. Using Web Conference System during Lesson in Higher Education*

According to this article, around 80% of students in Obuda University prefer education delivered in virtual way rather than classical way. Analysis has been made and the result is students seem more motivated in learning through virtual way. According to Kiss (2012), Big Blue Button is an open source web conference system used as consultation and presentation tools in education. The application allows instructors to deliver learning material and provide video conference between instructor and learners.

**Methodology**

The methodology involve in this project consist of two elements, which are WebRTC architecture and application development.

*A. WebRTC Architecture*

The WebRTC technology grants a real-time voice and video communication from web browser or other peer node without demanding any third party software or servers between two endpoints. It requires components of a typical VoIP media engine into a browser or any peer endpoint with a simple API that a web server can control. Now users can experience voice and video calls that support browser to browser applications without the needs to download anything (Ellis,2015). Figure1 shows an overview of WebRTC.

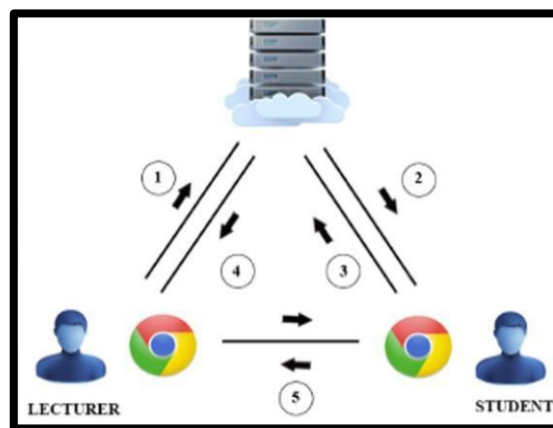


Figure1: Overview of Web Real-Time Communication

Figure1 illustrates the overview of how WebRTC works where it indicates that lecturers send message or data to students through the server and then students send a reply to the lecturer. Finally, the two peers communicate with each other. In WebRTC, the two types of API are required in order to acquire and exchange the media stream between browsers, which are Media Stream API and RTC Peer Connection API. Media Stream API is used to obtain audio and video media stream from local device, for example, the webcam and microphone, and to collect the input stream. Other than that, RTC Peer Connection is an API that gathers data from the output of Media Stream. RTC Peer Connection also provides connection establishment between the two users. The establishment of peer-to-peer connection requires Session Traversal Utilities for NAT (STUN) protocol and Traversal Using Relays around NAT (TURN).

### B. Application development

This section explains the information on the development of the web application such as context diagram, prototype development and software requirement. Data Flow Diagram is a method for modeling system that displays how the input data is processed through the sequence of functions to produce an output. Context diagram is also part of the Data Flow Diagram. Context diagram is a diagram that represents an overview of the web application. The main entity for the web application is STUDENT and LECTURER. Figure2 illustrates the context diagram for the web application.

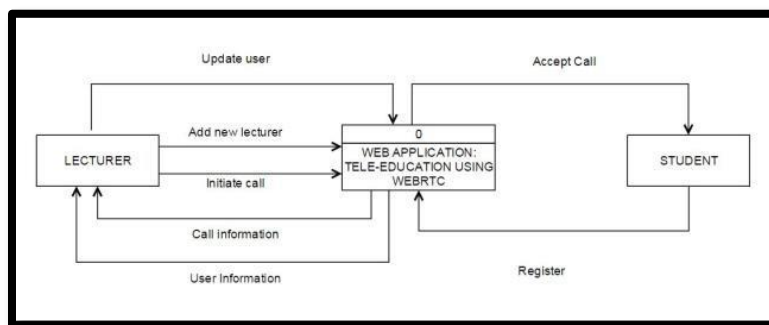


Figure 2: Context Diagram for Web Application

Besides, to develop the video call application, it follows several steps. The first step is to get the video and audio from device. This step requires Media Stream API, also known as “getUserMedia”. It is a JavaScript API that allows to synchronize stream of media such as camera and microphone. Its function is to request access to the local camera video stream using the get User Media call. If this stream is setup successfully, then it is displayed on the local browser page in a video element and then added to the “peer\_connection” object so that it can be sent to the remote peer's browser.

The second step is to create a user interface or video layout. The interface for the video call application is created using Cascading Style Sheet (CSS) language. The canvas size for the local and remote video is setup in this section. The size of remote video has been setup for 1024x768 pixels and the local video is 240x160 pixels. The button for 'dropacall' is also created in this section. This button functions to terminate the connection between two peers.

The third step is to stream video with RTC Peer Connection. RTC Peer Connection is the WebRTC API used for video calling. This section requires to setup a Session Traversal Utilities for NAT (STUN). In this project, Google's Public Stun Server is used to check port address for incoming request and allow peer to pass on another peer by signaling mechanism.

The fourth step requires to setup signaling server. The "call\_token" variable is a unique ID that links two users together. It is used to ensure that any signals passing through the signaling server are only exchanged between these two specific users. The "signaling\_server" is a variable that represents the Web Socket API connection to the signaling server to which both the local users and remote users will be connected.

The final step requires configuring specific port for server.js. Server.js is a JavaScript file that allows to run a web server in Node.js application. In order to provide functioned web socket server, it is compulsory to create basicNode.js application. In this JavaScript file contains a web socket library. This section also defines port80, which is the standard for HTTP port.

After video call has been successfully developed, the web application will be deploy into cloud hosting. In this project, Heroku Cloud Application Platform is used to host the video call application. Heroku is a cloud platform as a service (PaaS), which supports JavaScript programming language. The video call application requires Node.js application. Heroku cloud application platform allows user to store the Node.js application, which can be run in the server.

## Results and Analysis

The analysis is conducted by testing the performance and usability of the web application. Usability testing is conducted to determine the usefulness of the web application toward the potential users. The performance test is perform to measure the response time, round-trip time, the page load time and the smoothness of the video streaming based on different types of network.

### A. Usability testing

Usability test was a test conducted to evaluate the usability of the web application. This test is measured by having roughly 30 respondents. In addition, each respondent was given a set of questionnaire consisting of 13 questions to test the web application. The questionnaire was created using Google Forms and was published on the Internet. All the information collected from the respondents was saved into a spreadsheet file. The analysis was evaluated based on the feedbacks from the questionnaire given. Table1 shows questionnaire that was given to the respondents to evaluate the usability of the web application.

Table1: List of usability test question

No.	Question
1.	User can register to the web application.
2.	User can login into the web application.

3.	Popup error message if user enter wrong details.
4.	User can display list of calls.
5.	When user click call button, popup window will appear.
6.	Video call application will generate link for student.
7.	User can save generated link and date.
8.	User can search call's record by date.
9.	When user click go button, popup window will appear.
10.	User can perform video call by clicking go button.
11.	During video streaming, user can view another user.
12.	User can click drop call button to end communication.
13.	When user click logout button, the web application will terminate the session.

Table1 shows the questionnaire that was distributed over the Internet. Based on the questionnaire, the respondents need to select whether successful or unsuccessful on each of the given questionnaire. Figure 3 shows feedback from the questionnaire after giving the ball to the respondents. The percentage shows that 100% of the respondents have successfully completed the usability test based on the given questionnaire, except for question 11 and question 12.

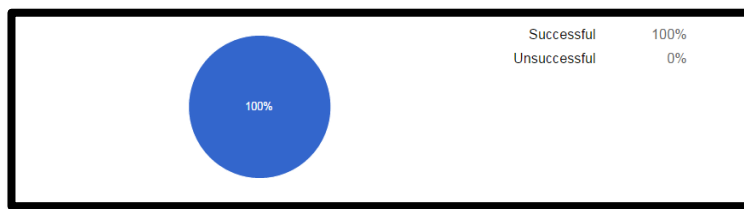


Figure 3: Usability testing –All question except Question11 and Question 12

Based on Figure 3, most of the respondents have successfully completed the test without any problem. However, several respondents were not able to complete question 11 and question 12. Figure 4 shows percentage for question11, which is during video streaming, the users are able to see another user on the video application.

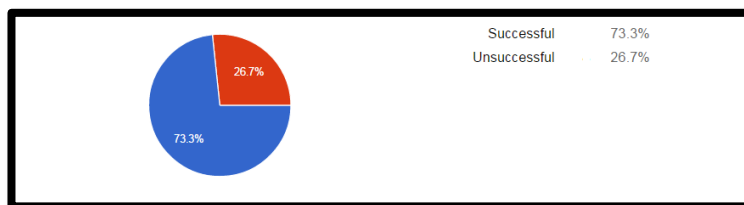


Figure 4: Usability testing for Question11

As seen on Figure 4, 73.3% of the respondents have successfully completed question 11 without any problems. Only 26.7% of the respondents are having problems with the video application. The problem might occur if the users use other than Google Chrome web browser. This is because, the WebRTC API can only support on the Google Chrome web browser. The video call interface will appear on the screen if the user uses other than Google Chrome web browser. However, the connection between two peers will not establish because the WebRTC cannots end and receive

data from different web browser. Other possible problem might occur if the users have slow Internet connection. The web browser cannot load the video application because it requires a stable Internet connection to perform real-time communication. Figure 5 shows the percentage for question 12. Question 12 require user to click drop call button to terminate connection during video streaming



Figure 5: Usability testing for Question 12

Based on Figure 5, only 26.7% of the respondents did not successfully complete the usability test for question 12. However, 73.3% of the respondents were able to complete the test without having any problems. This problem actually related to the question 11. If the video application did not appear on the web browser, automatically the users could not see any button appearing on the application. The user will not be able to click on the button because the video application did not load into the web browser.

#### *B. Performance Testing*

Web application response time was tested to measure how long does it takes for a web application to response back on user request. Figure 6 shows web application response time was tested using Blaze Meter testing tools.

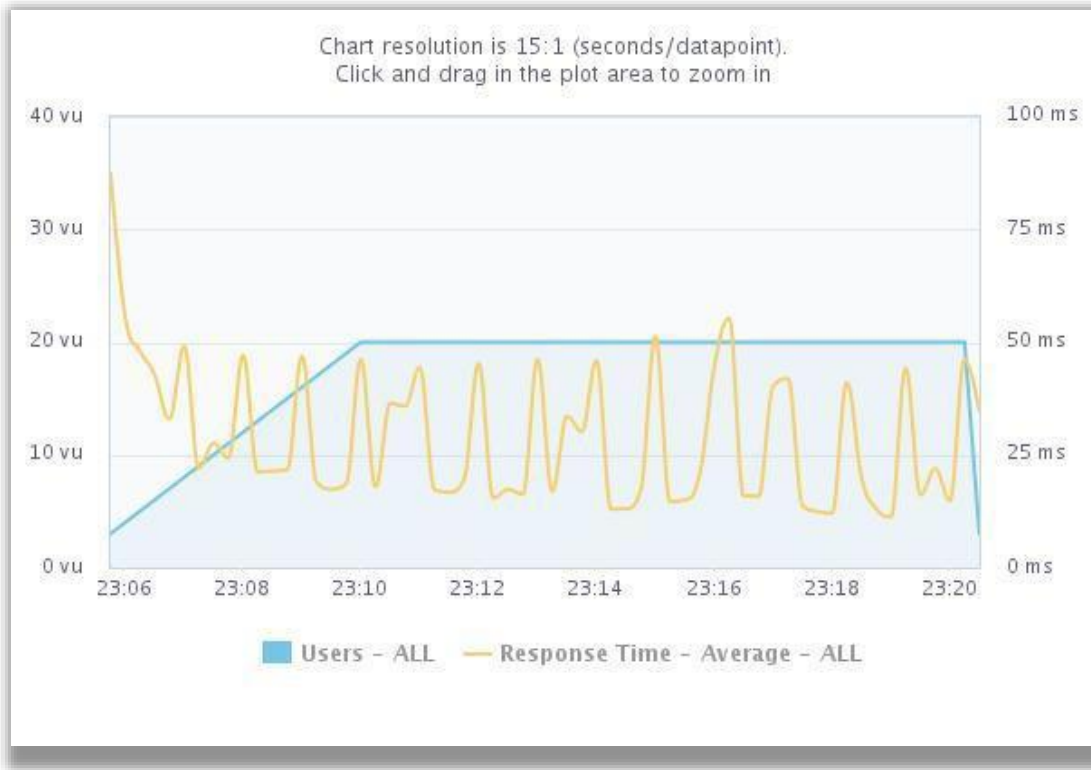


Figure 6: Web application response time

Based on the Figure 6, the graph shows the time taken for the application to response back on user’s request. In addition, the web application was tested based on 20 virtual users. The minimum response time was 14 milliseconds and the maximum response time was 51 milliseconds. Based on the results, the higher the time taken, the longer users must wait for the web application to process request.

The next section will discuss about testing of web application from different types of network by using wireless LAN and broadband. Four tests were carried out during the test in order to gather all the data and the result was included in this section.

*C. Scenario1: Comparing the amount of round trip time with different types of network*

Generally, round trip time is the amount of time it takes for a packet to reach its destination. A tool is used to capture the amount of time it takes from the moment a request is sent over network until a response is received. This test was conducted based on the different types of network by using wireless LAN and broadband. All the results were gathered in Table2. Figure 7 shows the amount of round trip time with different types of network.

Table 2: Result of the test for scenario1

Type of Network	Test 1	Test 2	Test 3	Test 4
Wireless LAN	120	110	50	190
Broadband	300	160	150	329

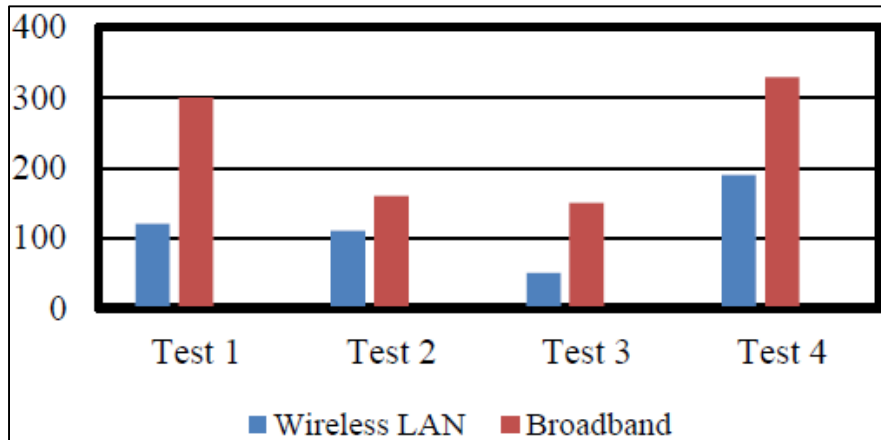


Figure 7: The total amount of round trip time in millisecond

Based on Figure 7, the amount of round trip time during test 1 for wireless LAN was 120 milliseconds, while for the broadband it took 300 milliseconds for the packet to reach its destination. For the second test, the amount of round trip time for wireless LAN was 110 milliseconds and 160 milliseconds according to the time taken for round trip time using broadband. The amount of round trip time reduces to 50 milliseconds for wireless LAN and 150 milliseconds for broadband during the third test. During the fourth test, round trip time was captured for wireless LAN was 190 milliseconds at the same time 329 milliseconds time taken for round trip time using broadband. Based on the results, the amount of round trip time taken for a packet to travel to a specific destination was longer by using a broadband compared to wireless LAN. As seen on Figure7, short period of time taken by a packet to travel to a specific destination and back again by using wireless LAN.

*D. Scenario2: Comparing the frame rate with different types of network*

This test was performed to measure the frame rate of the video streaming from different types of network. The unit for frame rate is fps, which stands for frame per second. Technically, the smoothness of the video streaming can be analyzed from the frame rate of the video. All the recorded results are shown in Table 3. Figure 8 shows the frame rate for video streaming with different types of network. The higher the frames, the smoother the video will be (Tan & Chou, 2012).



Table3: Results of the test for scenario2

Type of Network	Test1	Test2	Test3	Test4
Wireless LAN	40	34	41	47
Broadband	35	33	40	35

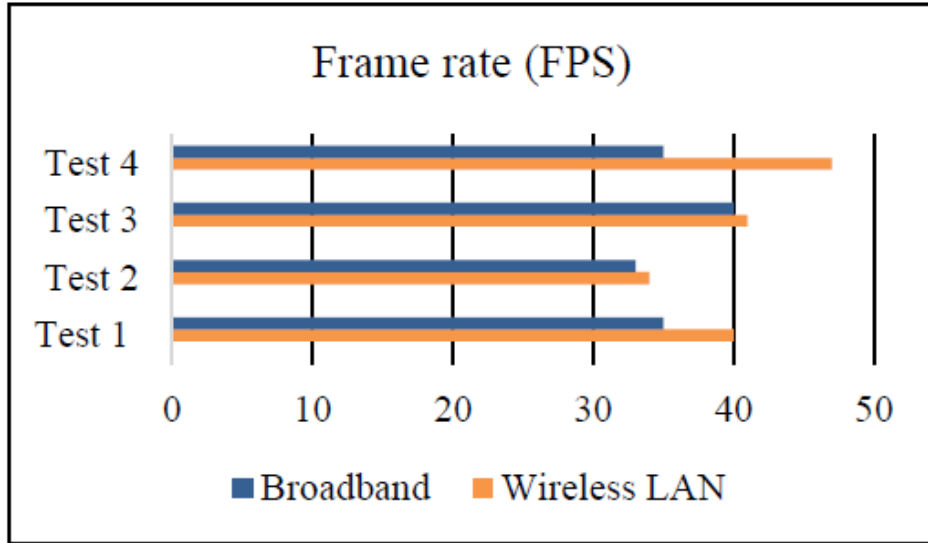


Figure 8: Frame rate with different types of network

Based on Figure 8, it shows the frame rate of video streaming by using different types of network. Four tests were conducted to measure the frame rate by using wireless LAN and broadband. The first test showed that the frame rate for video streaming was 40 fps using wireless LAN, while the frame rate for video streaming using broadband was 35 fps. During the second test, the result for frame rate using wireless LAN was 34 fps and for broadband was 33 fps. During this stage, both users might experience a sluggish video streaming as the frame rate was decreased. On the third test, both users have a smooth video streaming because the frame rate was 40 fps for wireless LAN and 41 fps for broadband. The final test showed that wireless LAN users have a smooth video streaming as the frame rate was around 47 fps, while the broadband users may experience a slow video streaming since the frame rate was 35 fps. Based on the test conducted, the results have been analyzed. If the frame rate of video was higher, then the user will experience a smooth video streaming. To have a smooth video streaming, it is recommended for the users to use wireless LAN as a network connection since the frame rate was higher.

## **Conclusion**

The WebRTC stands for web real-time communication. It is a media engine with JavaScript API that has been used in the development of the video call application. In the video call application, two APIs have been used, which are the Media Stream API and RTC Peer Connection. The Media Stream API played a role in order to obtain the media data from the local device. The media data consists of device's camera and microphone. The RTC Peer Connection is also JavaScript code use to establish connection between two endpoints. Besides, the interface for the video call application was created using Cascading Style Sheet language (CSS).

Two experiments have been conducted to test the performance and usability of the WebRTC. Usability testing was conducted to determine the usefulness of the web application towards the potential users. The performance test was performed to measure the response and round-trip time, and to test the smoothness of the video streaming based on different types of network.

Based on the results, the amount of round trip time taken for a packet to travel to a specific destination was longer by using broadband compared to wireless LAN. While for testing the smoothness of the video streaming, if the frame rate of video was higher, then the users will experience a smooth video streaming. To have a smooth video streaming, it is recommended for the users to use wireless LAN as a network connection since the captured frame rate was higher.

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