

Videoconference System for Rural Education: Issues, Challenges, and Solutions a Title is Fewest Possible Words

Kusprasapta Mutijarsa*, Yoanes Bandung, Luki B. Subekti

School of Electrical Engineering and Informatics, Institut Teknologi Bandung
10 Ganesha Street, Bandung, +622212345, Indonesia

Corresponding author, e-mail: kusprasapta.mutijarsa@itb.ac.id¹, bandung@stei.itb.ac.id²,
lukibsubekti@gmail.com³

Abstract

In this paper, we presented issues, challenges, and solutions of videoconference system for rural education. First, we discussed several issues on the implementation of videoconference system for education, particularly in rural area in Indonesia, which covered videoconference requirement, rural condition, and education needs. Second, we presented several challenges consisted of choosing videoconference technology, choosing compression method, system and application development, ensuring quality of services, and ensuring quality of experiences. Based on the issues and challenges, we proposed a solution of videoconference system which is specifically deployed in rural education. The solution was based on WebRTC technology implemented in Intel i5 core miniPC chosen to increase portability of the system. A STUN server was built on Javascript to facilitate communication between each client terminal. A simple and intuitive user interface was designed to facilitate the use of application by rural people. The system was deployed at two elementary schools in Cianjur, West Java, representing rural education in Indonesia. From the experiment, we obtained video sent data rate 82 kbit/s, video received data rate 245 kbit/s, average delay 316 ms and packet lost rate 1.32%. The experiment results showed that the audio and video quality can be accepted by users to implement distance learning.

Keywords: videoconference system, rural education, multimedia communication technology, quality of service (QoS)

Copyright © 2017 Universitas Ahmad Dahlan. All rights reserved.

1. Introduction

One of internet based services which continues to grow and becomes more popular for internet consumers is videoconference service. Nowadays, many available technologies enable people to access this service from any devices and platforms. The solutions vary based on the endpoint, those are dedicated hardware system, desktop application system, and web-based system. Advances in some technological areas have driven the development of videoconference technology. Those areas include high-performance computer system, larger communication bandwidth, advanced compression technology, improvement of solid-state storage and memory system, and advanced internet technology. High-performance computer used for personal computer has become common.

People can run some applications with high computing or graphical processing power simultaneously in a single desktop. Latest processor and GPU technology with high clock frequency and data bandwidth enable clients to run multiple multimedia streams (e.g. audio, video, text, and data) in a single instance. Multimedia data stream over networks can consume large portion of network bandwidth related to its data quality such as resolution and compression method. Larger resolution video with lossless compression method will consume more network bandwidth than one with low resolution and lossy compression method. Today technologies for communication like fiber optics, high-performance router, and parallel computing enable networks to provide higher data bandwidth for that need.

Advance compression technology also has important role for improvement of videoconferencing and real-time multimedia streaming services especially for multi-party audio/video conferencing [1]. Compression technology is used to create reversible conversion of data that contains fewer bits. Latest video compression technology like H.265/HEVC (High

Efficiency Video Coding) technology can provide high quality rating with low bitrate. This technology provides higher efficiency than H.264 and become the key element for wide deployment of 4K and 8K resolutions. At the H.265 released time, another video codec was also released by Google in 2013 which is called VP9 to compete with HEVC. According to Netflix VMAF quality benchmark which is held in 2016, HEVC was about 20% more efficient than VP9 at 360p, 720p, and 1080p resolutions while VP9 was still about 30% more efficient than H.264 at 720p resolution. According to [2], [3], HEVC also has smaller bit rates than VP9 in several measurement schemes. However, another comparison in 2014 using subjective comparison with synthetic imagery, VP9 efficiency could be close to HEVC in certain circumstances [4].

Storage and memory technology are also important components in videoconference system. Multimedia data stream should be processed by application on client nodes. In high traffic networks with high amount of multimedia bandwidth needs, this process can require large storage or memory space. Low latency and high throughput storage or memory are required to accelerate streaming process.

In general, there are two types of videoconference services, dedicated hardware-based service and software-based service. The hardware-based service mostly focuses on high quality and best user experience. As the shortcoming, the system can be costly and hard to be implemented. Another one can be called as software-based service. This service is more popular because it is commonly cheap and focused on multi-platform support. One of the latest technology that belongs to this system is WebRTC. WebRTC is a communication standard which lets web browsers to communicate in real time using a peer-to-peer architecture and enables web applications to provide videoconferencing services without need for proprietary plug-ins [5]. This technology is also supported by the development of HTML5 for browser application.

Figure 1 shows general WebRTC architectural models which are called WebRTC Trapezoid and WebRTC Triangle model. WebRTC Trapezoid model enables clients which are connected to different servers to run peer-to-peer communication. In the most common WebRTC implementation scenario, clients access a same web application from a same server. In that case, Trapezoid becomes a Triangle.

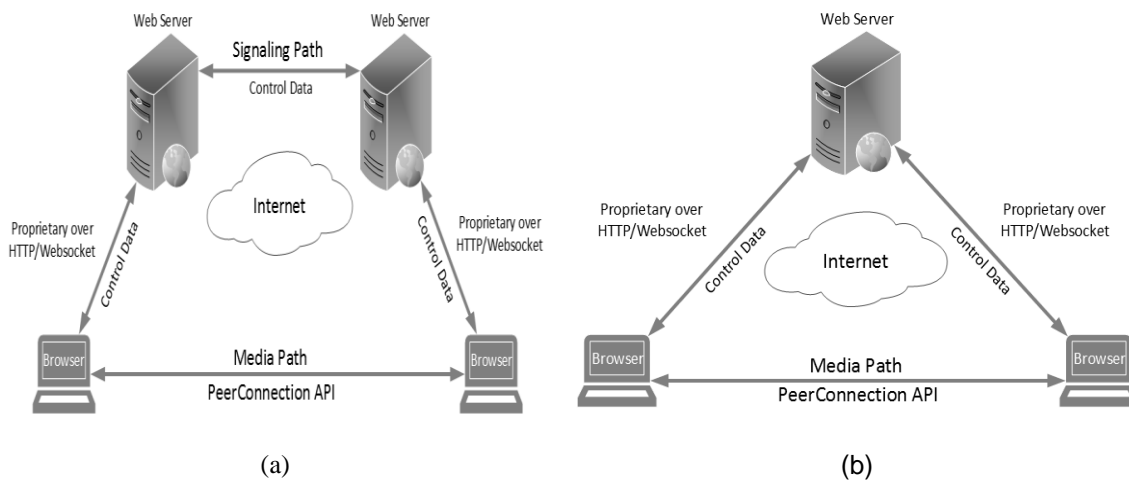


Figure 1. (a) WebRTC Trapezoid (b) WebRTC Triangle [6]

Videoconference service can be very beneficial in many fields such as business, education, interview, customer service, medical consultation, and so on. For example, videoconference service can be used by a teacher to present learning material from a remote location to some students in other locations. That implementation can overcome some related issues in learning executions like financial limitation, time management, and learning effectiveness. Moreover, implementation of videoconference service in education not only can

overcome those existing issues in education but also can create a possibility to increase value of a learning activity by enabling collaboration with outside parties such as industry people or specialists. As explained in [7], implementation of VRVS (Virtual Room Videoconference System) for educational activities in a university indicates that videoconference system should be used especially for some purposes including lectures and discussions with active participants, creation of conditions for lifelong learning, enhancement of employees' qualification, and provision of vocational consultations among specialists from different places. Another implementation of videoconference system based on WebRTC and HTML5 as multi-videoconference system for remote laboratory platform shows that it can engage students and teachers aligning with recent pedagogical trends in group dynamics and collaborative learning [8]. Videoconference system is also possible to be integrated with Learning Object [9] that can raise new learning opportunities to increase students readiness and engagement as a consequence of the combination of real-time communication and synchronized co-browsing [10]. Realizing the benefits of videoconference for education, it's necessary for us to understand any related issues and challenges about implementation of videoconference for education.

Education in rural areas still has some drawbacks in many ways compared with those in urban areas. This condition happens in a number of developing countries especially in Indonesia. Innovation and research for rural education improvement in Indonesia is still very relevant and important because rural population is about 46.26% of Indonesia total population in 2015 according to Trading Economics measurement. There are some issues related with the drawbacks those are infrastructure or education facilities, number of qualified teachers, and economic and family factors. Moreover, this situation worsens because of urbanization. People from rural areas move to urban areas to get better education that makes deficiency of human resources in rural areas itself. Education is one of inequality dimension faced by rural children which brings out other inequalities and problems [11]. Improvement of rural education is expected to solve some issues related with poverty in rural areas.

Implementation of videoconference service in rural to support education can possibly face some significant challenges. It can happen because of some factors such as lack of reliable network infrastructures and inexperienced people for operating internet or computer related stuffs. The challenges can be categorized into some aspects: electrical resources, networking, user application system, human resources, and hardware system [12]. The challenges will affect some cases in the development and implementation of videoconference system including what compression and streaming technology is used, how user interface is designed, what hardware technology is used, and how quality of service can be guaranteed.

This research aims to investigate issues and challenges of the implementation of videoconference system, particularly for rural education in Indonesia. In addition, this research proposes a design and implementation of videoconference system that is suitable for rural education. The design and implementation of videoconference system in rural areas like in Indonesia become very interesting because their communication infrastructure as well as internet connectivity are very limited. This paper consists of three main topics including issues, challenges, and solutions of videoconferencing system for rural education. First, this paper will show some issues which are related to the implementation of videoconferencing system for rural education such as its requirement, network infrastructure, multimedia conference technology, and education purpose, also some indicators which are key to find QoS of the network including bandwidth, packet loss, and jitter [13] [14]. Second, this paper will explain some challenges in delivering videoconference system in rural areas, including choosing suitable videoconference technology, choosing encoding and compression methods, developing of system and application, measuring and ensuring methods for videoconference quality of services (QoS) and quality of experience (QoE). Last, solutions of videoconferencing system for rural education was proposed. This part will describe how to apply suitable videoconference technology for rural education where internet bandwidth is limited, how to design intuitive and user-friendly GUI for videoconferencing application which is specialized for rural education, and how to constructing dedicated hardware for videoconferencing which is specialized for rural education. For system validation, experiment is conducted by measuring some system parameters.

2. Issues on the Implementation of Videoconference System for Rural Education

Videoconference is a type of conference that is based on video and audio transmission in both directions between conference member which utilizes internet or other network types [15]. In general, videoconference may have some common requirements such as slides sharing, comfortable audio system, on the fly users additioning, and conference recording. Slides sharing feature is required as the implementation of videoconference may be for presenting work report, giving learning material, or other activities that need slides presentation. In videoconference system, microphone and speaker must be configured and positioned well or be integrated with echo cancelation to prevent audio feedback arising in local audio system or being transmitted to other parties. Videoconference system also needs to provide mechanism for new users to join conference or access slides on the fly. It can increase flexibility of the service. Video recording feature is also important so that user can share or distribute conference content to others or replay the material in the future. Table 1 shows network related requirements of videoconference application. Videoconference application requires very high network availability and stability that make implementation of videoconference in network with limited bandwidth becomes a challenging issue.

Hardware requirements of videoconference terminal is highly dependent with videoconference applications and multimedia standards which are used. Hardware system can be divided into three categories those are unit central, capturing device, and presentation device. Unit central is used as main processor unit that gets media stream from capturing devices, transcodes data, sends data through network, and receive media stream from network. Capturing devices consist of microphone and video camera while presentation devices consist of speaker and projector or display monitor. Unit central becomes source of main issues for determining hardware requirements of videoconference system because it is where most of data being processed. As mentioned in the introduction, there are dedicated hardware, desktop application, and web based videoconference system. Web based system utilizes browser with WebRTC standard so that hardware requirements mostly depend on requirement of the browser and audio/video codecs which is used. System requirements of some browsers are shown in Table 2. Desktop application based system mainly depend on specific application platform which is used to build the application and its audio/video codecs itself. This characteristic also applies to dedicated hardware based system.

Table 1. Requirement of Videoconference Application [15]

| Requirement | Value |
|-----------------------------|----------------------|
| Availability | Very high |
| Stability | Very high |
| Simultaneity | Almost |
| Start-up delay media stream | Few seconds |
| Delay within session | Not recognizable |
| Error | Not recognizable |
| Number of users | 2-50 |
| Transported media | Audio, Video, Slides |

Table 2. Hardware Requirements of Web Browsers [16]

| | IE 9 | Firefox | Chrome | Safari |
|-------------------------------|--------|-----------|-----------|--------------------------|
| Processor Windows | 1 GHz | Pentium 4 | Pentium 4 | 500 MHz Pentium-class |
| Processor Mac | - | Intel | Intel | Intel |
| Minimum RAM | - | - | 128 MB | 256 MB |
| Recommended RAM | 512 MB | 512 MB | - | - |
| Minimum Disk Space | - | - | 100 MB | unknown |
| Recommended Disk Space | 70 MB | 200 MB | - | - |
| Recommended Disk Space 64-bit | 120 MB | - | - | - |

For providing better quality of video presentation, hardware support to decode related media streams is needed. HEVC, one of latest video compression standard, has been supported by most hardware which is released in 2015. Current hardwares that support HEVC

are Intel 6th generation core processors, AMD 6th generation APUs, Nvidia GeForce GTX960/950 GPUs, Qualcomm Snapdragon 805/615/410/208, and other newer hardwares according to TechSpot publication. Unlike HEVC, VP9 still doesn't have hardware acceleration supports as many as HEVC. Currently hardware supports come from Intel 7th generation core processor for PC, latest ARM processors for most smartphone/mobile devices and some processors for other devices [17]. Without existence of hardware support or dedicated decoding block, decoding process of video/audio stream is still possibly done by software in playback process. VP9 codec is supported by most modern browser as WebM becomes standard file format for video file on the web.

Today, there are a number of communication related standards and protocols which are available for supporting videoconference application from network interface layer to application layer. Its utilization depends on the technology base that we used for building videoconference application. Videoconference based on WebRTC will require utilization of specific transport and application protocols. Media data for WebRTC application is carried by Secure Real-time Transport Protocol (SRTP) together with RTP Control Protocol (RTCP) information which is used to monitor transmission statistics associated with data streams. Datagram Transport Layer Security (DTLS) is used for SRTP key and association management [6]. Figure 2 shows the WebRTC protocol stack. Media streams can be sent between two browsers by establishing of a peer connection. The connection mechanism uses ICE protocol and requires Session Traversal Utilities for NAT (STUN) [18] and Traversal Using Relays around NAT (TURN) [19] servers to let UDP-based media streams traverse NAT boxes and firewalls.

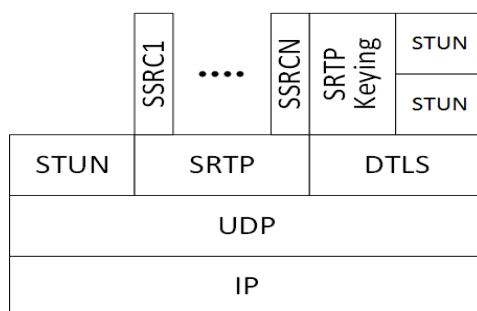


Figure 2. The WebRTC Protocol Stack [6].

2.2. Rural Condition

There are some challenging issues for videoconference implementation in rural areas. Network infrastructure which can provide reliable service is very limited. Cianjur, as the testbed area in our research, has some locations which are still unreachable by fiber optics and has limited bandwidth capacity to 2 Mbps. Another measurement scenario shows that bandwidth capacity that can achieved is only 1.05 Mbps [20]. The measurement also shows that end-to-end delay from Cianjur to our laboratory in Bandung is about 485 ms with packet loss rate about 16% while another measurement result varied from 8% to 90% [14]. Beside the lack of network related aspects, human capability for operating IT related system may become another concern.

2.3. Education Needs

In Indonesia, especially in rural areas, some learning and educational aspects still need more improvement including facilities, learning materials, teachers, learning methods, and other aspects. Improvement of those aspects can possibly increase quality and effectiveness of a learning process. Information technology has been intensified as one of the solution for education improvement. However, utilization of information technology such as online learning can also raise challenging issues. Technology can diminish interaction between teacher and student in a learning activity which can affect the degree of student's interest, attention, or passion in learning. There are some options that we can take for building application for education such as gamification, micro podcast, online chats, and feedback.

3. Challenges

3.1. Choosing Videoconference Technology

As there are many available technology options for building videoconference service, it's challenging to determine proper technology which is suitable for rural education. Some aspects that need to be considered include network condition, learning activity requirement, human resource characteristic, and implementation budget. In previous discussion, we know that rural area may have unreliable or limited bandwidth capacity. This situation implies the need of videoconference technology that capable to adaptively change its configuration based on network condition. Related to learning activity, videoconference system should support multiple video inputs and changeable input stream on the run. Multiple video cameras are used that enable system to capture teacher or students image properly. Easiness of operation can be achieved by providing dedicated system which has intuitive user interface and auto-configuration feature with integrated peripherals and compact components. Unlike enterprise company, school in rural may have very limited budget that cost of hardware and software for building videoconference solution should be minimized. It can be achieved by compromising some values of service quality that require lower hardware or software requirements.

3.2. Choosing Compression Method

Data compression is very important when bandwidth capacity is unreliable and limited. Most recent video codec like HVEC or VP9 should be used as those are more efficient than other earliest codecs [3], [21]. However, codec implementation may be dependent to other technologies. For example, if videoconference application uses WebRTC technology on Chrome browser, we can not use H.264 as the codec because Chrome has dropped its support. VP9 doesn't have hardware block decoding supports as many as H.265 has from some vendors that cause VP9 software based decoding must be run for some hardware platforms. Preventing interoperability failure can be achieved by developing dedicated system with defined platform and technology which ensures each terminal in the network can decode incoming audio/video stream. Choosing most recent compression method with higher efficiency can possibly increase hardware requirement because of its computation needs.

3.3. System and Application Development

Building videoconference solution, especially as dedicated solution for rural education, requires comprehension in diverse technological aspects including hardware and software integration, computer network, and data communication. Dedicated videoconference solution needs to integrate capturing devices, microphones, and central unit as a compact system which is easy to be implemented and maintained. Moreover, videoconference solution intended for people in rural area can escalate the need of low complexity and easy-to-operate system. We need to design certain operating system or customized available operating system for central unit of videoconference system which is suitable for remote learning. System setup and configuration need to be as minimal as possible.

3.4. Ensuring Quality of Services (QoS)

The QoS of videoconference system may be affected by several network parameters such as bandwidth, delay (latency), jitter (delay variation), and packet drops (loss rate) [22]. Unlike data traffic, videoconference traffic which consists of voice and video traffic is delay and drop sensitive. In one-way communication, voice requires latency to be lower than 150 ms with loss rate below 1% while video requires latency to be lower than 400 ms with loss rate below 1% too [23]. QoS is required to manage utilization of network resources to maximize the end-user experience of a session.

3.5. Ensuring Quality of Experiences (QoE)

QoE value relies on end-user perception of the network's performance, instead of just to technical metrics of the network. Ensuring QoE becomes challenging as it implies the implementation of QoS in all network segments. QoE perception is equal to the impairment imposed by the worst-performing segment of the network. Classification of QoE metrics can be based on objectives methods or subjective methods which are time consuming [13].

4. Solution

4.1. Videoconference System

In this research, videoconference system focused for rural education has been built. The system is based on WebRTC technology. WebRTC is selected because of its capability to accept any changes of video resolution on the fly. WebRTC also enables each client terminal to stream multimedia data through peer-to-peer communication. The system is also designed to be able to run in local network (LAN) without internet connection by installing and activating STUN server in one of client terminal in the network. In the implementation for online learning in Cianjur, global STUN server is installed in our campus, ITB. Because of some network restrictions, VPN server is also implemented. Network architecture scheme for this videoconference system is shown in Figure 3.

Client terminal is implemented on miniPC with Intel i5 core processor. The miniPC is chosen to increase portability of the system. Client terminal is also supported with audio peripherals including wireless mic, sound mixer, and speaker. RCA video switch is included to support multiple video input. Output of the video switch is connected to RCA to USB adapter so that video stream can be processed by miniPC. While miniPC is equipped with Wi-Fi adapter, it's recommended to use wired connection for connecting to Internet to ensure reliability of the connection.

Figure 4 shows interconnection scheme of all peripherals and devices in the client terminal. Ubuntu is used as operating system of client terminal to minimize production cost and easiness of modification. Chromium browser is installed as the main platform to run videoconference application. The application is built on HTML5 and Javascript which enable video and WebRTC API access in web browser. NodeJS is also installed to execute STUN server application which is built on Javascript when clients run the videoconference service in local network.

Videoconference application is designed with simple and intuitive user interface as the target users are people in rural area which are inexperienced in utilization of videoconference service for online learning. The application has four pages, those are main, configuration, room list, and conference page. Main page presents some links to configuration and conference page. In configuration page, user can configure audio and video input devices, remote (STUN) server address, and user identity. When user runs videoconference service in local network, user must set remote address to one of client terminal address which is set to be central unit in the network. Every terminal can become the central unit because in startup process each terminal device will load STUN server application automatically. In room list page, user can choose to join available conference room. If there isn't any available room, user can create new one so that other users can join. In conference page, there are three main modules, those are video player, presentation viewer, and chat field. User can choose any video to be displayed on main video panel from any connected client terminal in the network by clicking on video thumbnail below the main video panel. Figure 5 shows main page of the application.

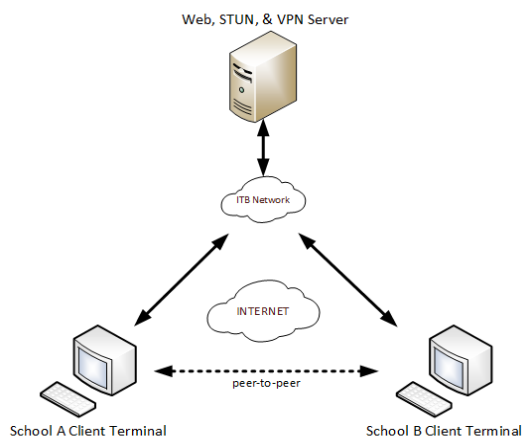


Figure 3. Videoconference System Network Implementation

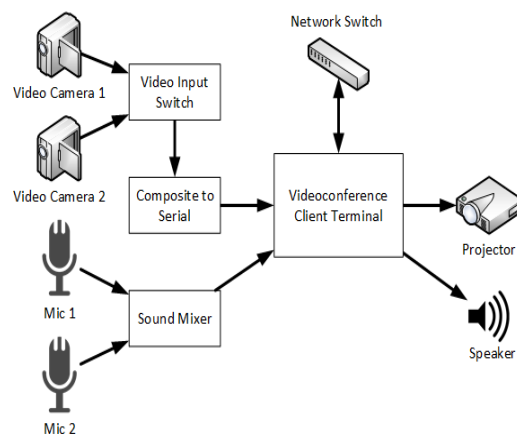


Figure 4. Client Terminal and Peripherals

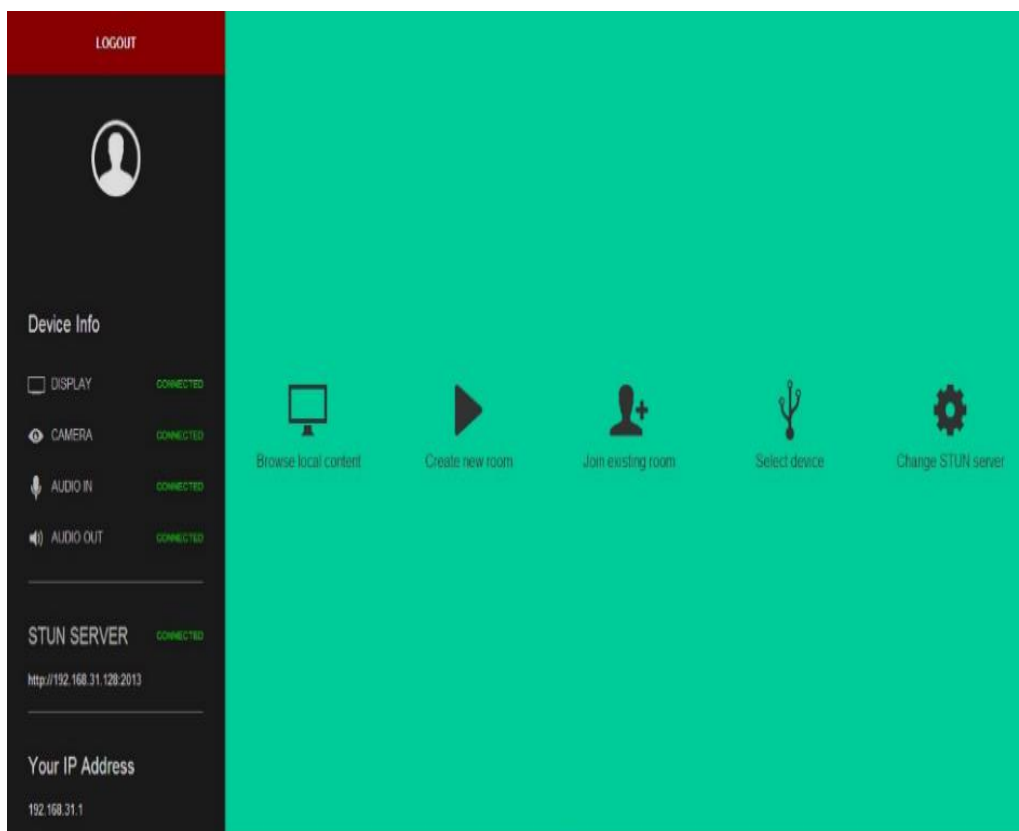


Figure 5. Main Page of Videoconference Application

4.2. Experimental Results

Experiment was conducted for a session of online learning at two schools in Cianjur, West Java. QoS parameters data is collected using WebRTC internal tool in Chrome browser. The tool can show some parameters including delay, jitter, packet loss, and transfer rate for certain period of time. Data presented in this paper is taken from one of the schools with up-to-2Mbps of connection speed from ISP. The data is focused on QoS parameters for video which typically consumes more network resources.

Using this videoconference system can reach 82 kbit/s of video sent data rate and 245 kbit/s of video received data rate. Both received and sent video resolution are 640 x 480 pixels. WebRTC provides mechanism for estimating end-to-end bandwidth between receiver and sender. The estimation comes from certain algorithm by detecting packet loss and analyzing congestion [24]. This estimation is used as limit of how much data can be sent in certain session to ensure that all data can be transmitted in current network condition. From the graphics above, we can see that average data rate doesn't exceed average bandwidth estimation.

Data rate adjustment is achieved by frame dropping or packet dropping while the resolution is maintained. We can see from fluctuation in the graph of frame rate and packet rate. Frame rate can be decreased up to 10 fps while the original frame rate from capturing device is 30 fps. Graph of packet rate also shows congruity with graph of bandwidth estimation. Packet loss is happened only at a certain time during 12 minutes periode of measuring time. As 300 packets loss for 22,867 packets transmitted in that periode, it resulted 1.32% of packet lost rate (PLR). Delay of received video is also tolerable for real-time communication which is about 316 ms in average. Comparing this experiment results with our other experiments using videoconference system based on Linnphone [25], we figure out that our WebRTC based system can provide better quality of service. WebRTC based solution can provide lower bandwidth consumption with higher and maintained video resolution as the result of several supporting features and mechanism in WebRTC technology which has been mentioned previously.

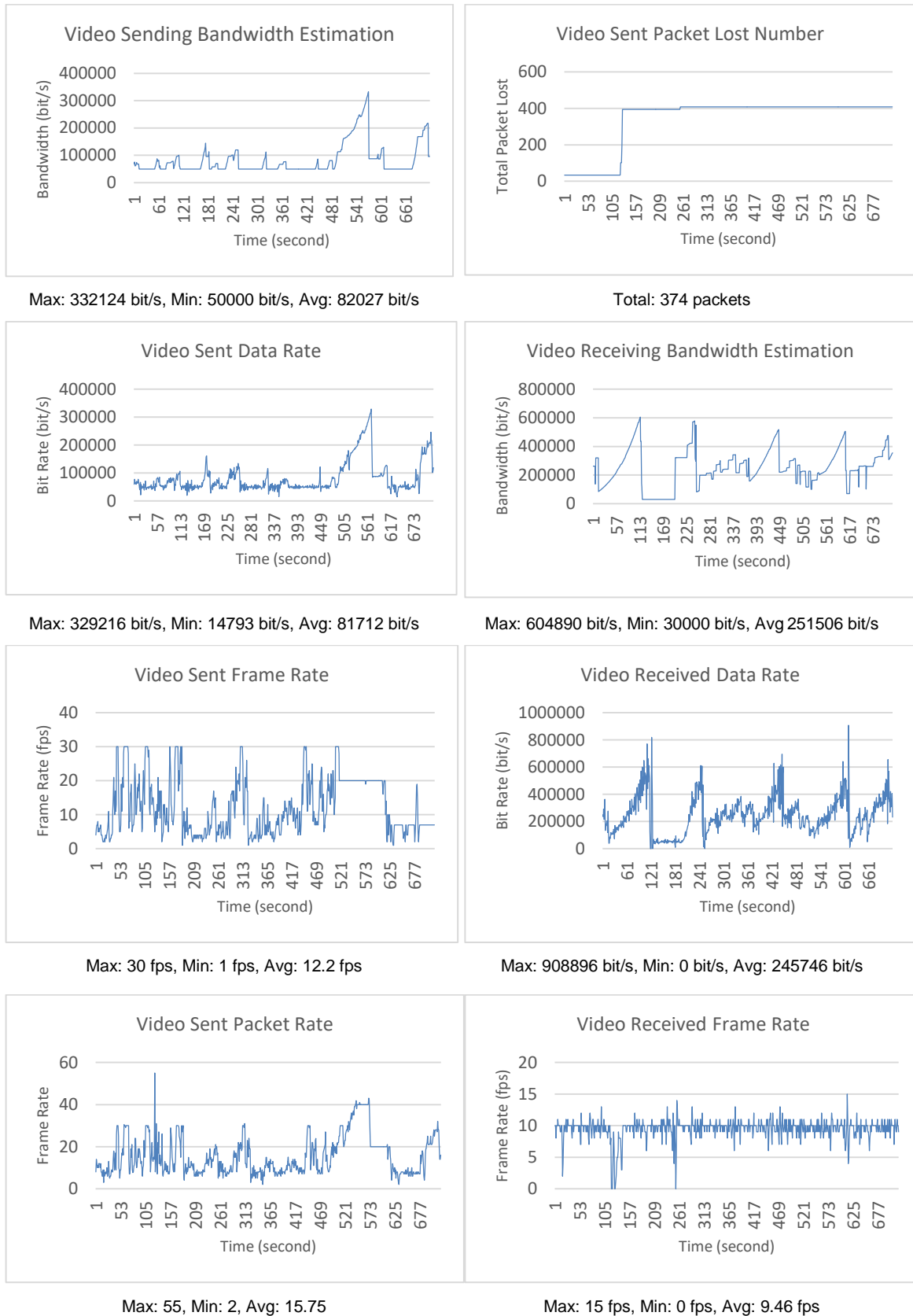


Figure 6. Graphs of QoS Parameters

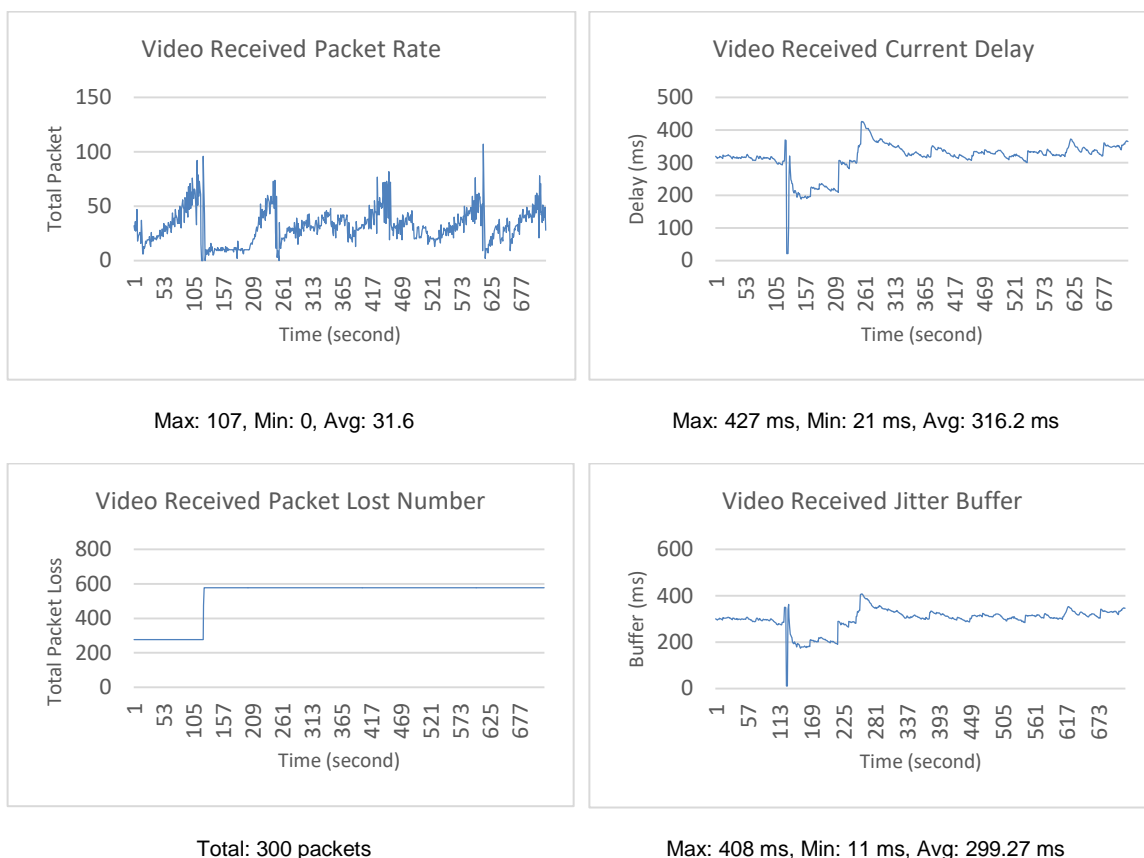


Figure 7. Graphs of QoS Parameters

5. Conclusion

We have described several issues and challenges in developing and implementing videoconference system for rural education. We have proposed a dedicated system of videoconference service as a solution. The solution was implemented in Cianjur, Jawa Barat, for supporting distance learning between two schools while several QoS parameters were measured during that activity. WebRTC technology was chosen as base platform to build the videoconference system. This technology can provide mechanism that ensure reliability of videoconference service. From the measurement, we obtained 82 kbit/s of video sent data rate, 245 kbit/s of video received data rate, 316 ms of average receiving delay time, and 1.32% of packet lost rate. The experiment results showed that the audio and video quality still can be accepted by users. From that result, we concluded that our proposed solution can provide videoconference services for rural education with acceptable QoS.

References

- [1] B Grozev, G Politis, E Ivov, T Noel, V Singh. Experimental Evaluation of Simulcast for WebRTC. *IEEE Commun. Stand. Mag.* 2017; 1(2): 52–59.
- [2] MP Sharabayko, OG Ponomarev. RI Chernyak. Intra Compression Efficiency in VP9 and HEVC. *Appl. Math. Sci.* 2013; 7(137): 6803–6824.
- [3] D Grois, D Marpe, A Mulyoff, B Itzhaky, O Hadar. *Performance comparison of H.265/MPEG-HEVC, VP9, and H.264/MPEG-AVC encoders.* in Picture Coding Symposium (PCS). 2013: 394–397.
- [4] M Rerabek, T Ebrahimi. Comparison of compression efficiency between HEVC/H.265 and VP9 based on subjective assessments. in *SPIE - The International Society for Optical Engineering.* 2014.
- [5] S Loreto, S Pietro Romano. Real-Time Communications in the Web: Issues, Achievements, and Ongoing Standardization Efforts. *EEE Internet Comput.* 2012; 16(5): 68–73.
- [6] S Loreto, S Pietro Romano. *Real-Time Communication with WebRTC.* O'Reilly Media, Inc. 2014.

- [7] J Depesova, I Turekova. *Implementation model of teaching practice with the application of a videoconference system*. 12th IEEE Int. Conf. Emerg. eLearning Technol. Appl. 2014: 91–96.
- [8] MA Bochicchio, A Longo, M Zappatore, D Tarantino. *A collaborative multi-videoconferencing platform for online optical microscopy*. in 12th International Conference on Remote Engineering and Virtual Instrumentation (REV). 2015: 143–151.
- [9] P Barkman, D Brown, P Brusilovsky, JR Burke, M Fore, J Hyde, I Kegel, B Melton, B. Nielsen, B Peoples. *Draft Standard for Learning Object Metadata*. 2002.
- [10] A Gordillo, E Barra, D Gallego, J Quemada. *A model for integrating learning object repository resources into web videoconference services*. in Frontiers in Education Conference. FIE. 2013: 383–392.
- [11] P Aji. *Summary of Indonesia's Poverty Analysis*. 2015.
- [12] Y Bandung, AM Gani, HC Tanuwidjaja, J Sembiring. *The Challenges of Delivering Multimedia-based Learning Services in Rural Areas*. in International Conference on Information Technology System and Innovation (ICITSI). 2015.
- [13] B García, F Gortázar, L López-fernández, M Gallego, M París. *WebRTC Testing: Challenges and Practical Solutions*. *IEEE Commun. Stand. Mag.* 2017; 1(2): 36–42.
- [14] Y Bandung, I Sumardi. *A Methodology for Characterizing Real-Time Multimedia Quality of Service (QoS) in Limited Bandwidth Network*. *TELKOMNIKA*. 2016; 14(4): 1534–1544.
- [15] HW Barz, GA Bassett. *Multimedia Networks: Protocols, Design, and Applications*. Wiley Publishing, Inc. 2016.
- [16] M Brinkmann. *Web Browser System Requirements Overview*. 2012. [Online]. Available: <http://www.ghacks.net/2012/05/21/web-browser-system-requirements-overview/>. [Accessed: 03-Jan-2017].
- [17] The WebM Project. *SoCs Supporting VP8/VP9*. 2017. [Online]. Available: <http://wiki.webmproject.org/hardware/socs>. [Accessed: 03-Mar-2017].
- [18] J Rosenberg, R Mahy, P Matthews, D Wing. *RFC 5389 - Session Traversal Utilities for NAT (STUN)*. 2008.
- [19] R Mahy, P Matthews, J Rosenberg. *RFC 5766 - Traversal Using Relays around NAT (TURN): Relay Extensions to Session Traversal Utilities for NAT (STUN)*. 2010.
- [20] Y Bandung, MT Hutabarat. *Network Characterization for Delivering Multimedia-based Learning in Rural Areas*. Int. Conf. Adv. Educ. Technol. 2014: 54–57.
- [21] IK Kim, S Lee, Y Piao, J Chen. *Coding efficiency comparison of new video coding standards: HEVC vs VP9 vs AVS2 video*. in IEEE International Conference on Multimedia and Expo Workshops (ICMEW), 2014, pp. 2–7.
- [22] WL, Y-FC, YL. *An Enhanced Safety Algorithm for Network QoS Multicast Routing Optimization*. *TELKOMNIKA*, 2014; 12(4).
- [23] T Sziget, C Hattingh. *End-to-end QoS Network Design*. Cisco Press. 2014.
- [24] L De Cicco, G Carlucci, S Mascolo. *Experimental Investigation of The Google Congestion Control for Real-time Flows*. in ACM SIGCOMM Workshop on Future Human-centric Multimedia Networking (FhMN). 2013: 21–26.
- [25] J Nainggolan, G Christian, K Adari, Y Bandung, K Mutijarsa, LB Subekti. *Design and Implementation of Virtual Class Box 5.0 for Distance Learning in Rural Areas*. in International Conference on Information Technology and Electrical Engineering (ICITEE). 2016.