TELKOMNIKA, Vol.14, No.4, December 2016, pp. 1534~1544
ISSN: 1693-6930, accredited A by DIKTI, Decree No: 58/DIKTI/Kep/2013
DOI: 10.12928/TELKOMNIKA.v14i4.3295

A Methodology for Characterizing Real-Time Multimedia Quality of Service in Limited Bandwidth Network

Yoanes Bandung*[1], Idi Sumardi[2]
[1,2]School of Electrical Engineering and Informatics, Institut Teknologi Bandung,
Jalan Ganesa 10 Bandung, Indonesia, Ph.: +62-22-2500985
*Corresponding author, e-mail: bandung@stei.itb.ac.id[1], idisumardi@gmail.com[2]

Abstract

This paper presents how to characterize the quality of multimedia which consists of audio and video that are transmitted in real-time communication through the Internet with limited bandwidth. We developed a methodology of characterizing the multimedia Quality-of-Service (QoS) by measuring network parameters (i.e., bandwidth capacity, packet loss rate (PLR), and end-to-end delay) of testbed network and simulating the audio-video delivery according to the measured network parameters. The analysis of network parameters was aimed to describe the network characteristics. Multimedia QoS was characterized by conducting a simulation using data which was collected from the previous network characterization. A simulation network model was built using OMNet++ representing a delivery of audio-video in real-time while a background traffic was generated to represent a real condition of the network. Apllying the methodology in a network testbed in Indonesia's rural area, the simulation results showed that audio-video could be delivered with accepted level of user satisfaction.

Keywords: *quality of service, multimedia communication, audio and video, network characterization, limited bandwidth network*

1. Introduction

Multimedia communication over the Internet has been the subject of many research works over the last few years [1] -[3] . Multimedia communication technology has been available to the transmission of a variety types of media such as text, graphics, animation, audio, and video over the Internet that are carried out continuously into data stream. Basically, multimedia streaming applications are divided into 3 categories [4] : (i) streaming of stored audio/video, (ii) streaming of live audio/video, and (iii) conversational of voice/video-over-IP. These categories are also known as (i) video on demand, (ii) live streaming, and (iii) real time streaming. First, video on demand allows users to perform playback control such as pause, rewind, or fast forward. Second, live streaming allows users to receive live broadcasts such as broadcast television or radio. Third, real time streaming allows users to communicate with audio or video in real-time.

One application of multimedia communication technology for education is distance learning through virtual classroom technology. This technology allows a learning process facilitated by a teacher in a real classroom can be virtually shared into other classrooms at different locations [5] . With virtual classroom technology, learning process between two or more classrooms can be conducted through a real-time multimedia communication over the Internet. Learning process in each classroom is recorded into audio and video and transmitted between those classrooms. Moreover, data also can be communicated between classrooms for example in remote presentation or desktop sharing.

Quality of audio and video during the learning process through a virtual classroom technology becomes a significant factor for the successful delivery of learning materials. This is because students in virtual classroom participate virtually in a learning process with a teacher in the real classroom. Thus, the comfort of learning materials delivery depends on the quality of audio-video transmitted through the Internet. Low quality of audio-video can make the learning proccess being interrupted or even cannot be held. Our study revealed that the quality of

Received December 18, 2015; Revised September 4, 2016; Accepted September 19, 2016

multimedia services which produce a large amount of data [6] can be affected by several network parameters [7] such as bandwidth, end-to-end delay or latency, jitter, and packet loss rate (PLR). When the available bandwidth become narrow, the audio and video transmission may be disturbed [5] . Moreover, the quality of audio and video requires end-to-end delay and packet loss rate as low as possible.

The concept of quality of service (QoS) is defined in ITU-T Recommendation E.800 [8] as the collective effect of service performance, which determines the degree of users satisfaction [9] . QoS aims to provide a better quality of service for various needs of the existing network infrastructure so that users get satisfaction in using network-based applications. QoS can arrange the provision of different services quality for the diverse needs of service such as providing specific bandwidth, decreasing packet loss, decreasing delay time and jitter. QoS functions can be describe as follows [10] : (1) grading packets to provide different services for each class of packets; (2) congestion handling to handle the needs of different services; (3) controling of packet traffic to restrict and controlling delivery of data packets; (4) signaling for control the device functions that supports communication in the Internet.

There are several obstacles which can affect the quality of audio, video, or other data transmitted through the Internet. One of the obstacles is available bandwidth of the network which is typically very limited particularly in rural areas like in Indonesia [5] . Increasing access links of the network is very expensive and often takes several years for their deployment [11] so that the Internet access is varied between regions. Another obstacle is asymetric property of the network where the bandwidth capacity of the upload link is different with those of the download link. Typically, the upload link of the network is greater than the download link of the network. Technically, real time multimedia communication requires symmetrical bandwidth capacity in order to transmit the multimedia which consists of audio, video, or other data in full-duplex communication scenario.

In this paper, we aim to study the effects of network characters to the audio and video quality in a real-time multimedia delivery especially in limited bandwidth network. Moreover, we propose a methodology for characterizing multimedia QoS which is affected by network conditions. There have been many studies on the measurement of network and multimedia, but only a few studies on the development of a methodology for characterization of multimedia QoS in limited bandwidth network. In previous studies of [5] [11] [12] , network characterization was done by measuring the broadband networks. Measurement hosts were connected to high-speed academic network. Study in [12] proposed methodology for characterizing residential broadband networks but did not yet include the characterization of multimedia QoS. In this technique, network measurement was conducted by using a probe train sent from hosts located in four academic networks - three in North America and one in Europe. In the study of [11] , characterizing real-time video traffic in residential broadband networks was conducted by analyzing networks performance for transmitting real-time video. Measurements were performed between DSL end-hosts in apartment complexes and single houses which were connected to the university network. In [5] , network measurement and simulation was done from DSL host located in rural area connected to a high-speed academic network.

2. Methodology for Multimedia QoS Characterization

This section describes our proposed methodology to characterize multimedia QoS when various media such as audio, video, and data are transmitted over limited bandwidth network. Basically, the methodology presents two important procedures: i) network characterization to obtain network conditions based on several network parameters measurement in testbed network and ii) network simulation to draw the quality of multimedia contents if transmitted across the network. The proposed methodology can be described in more detail as follows.

2.1. Choosing Network Parameters for Measurement

Several studies reveal that network parameters such as bandwidth availability, end-to-end delay, and packet loss rate have great impact on the quality of audio, video, or data in the multimedia delivery services [5] [11] [12] . Based on those studies, we suggest to choose the three parameters to be measured on the multimedia QoS characterization. We present several reasons behind the selection of the parameters as follows.

1. In data transmission, bandwidth capacity is related to the width of the communication pipe and how quickly bits can be sent.
2. Multimedia QoS will decline with increasing delay time. Moreover, if the multimedia contents spend much time towards the destination host, it can cause failures in real-time multimedia communication.
3. Packet loss can be occurred when the packets are discarded during its transmittion to destination host. This is because of connection failure, sudden route change, traffic overload, or congestion in the network. In general, packets loss is the major cause of degradation on audio and video quality.

2.2. Measuring and Investigating Rural Network

Network measurement aims to capture variations of network properties. Measurement method used in this study can be described as follows. 1) Bandwidth capacity of network link is measured by using TCP (Transfer Control Protocol) traffic. In our experiments, several measurements were done repeatedly using certain time intervals and traffic load. According to the study of [5] [11] [12] , minimum measurements should be conducted during a day. Here, each iteration of measurements for up to 10 seconds was conducted with interval time of 30 minutes. Bandwidth capacity was measured by flooding the link with data packets sent from a source host to a destination host. This scenario was aimed to saturate bandwidth. 2) End-to-end delay was measured from host-to-host using ICMP (Internet Control Message Protocol) packets with different sizes. First, source node sent the same size ICMP echo request packets and destination node responded with the same size ICMP echo response packets. Second, measurement was conducted using the MTU (Maximum Transmission Unit) value on ADSL (Asymmetric Digital Subscriber Line) network. Measurements were performed repeatedly in a day by using a specific interval to characterize the end-to-end delay. 3) Packet loss rate parameters were measured using UDP (User Datagram Protocol) by sending UDP datagram packets from source to destination by using different traffic load. We used UDP datagram packets because it could represent multimedia delivery. UDP datagram packets with size 1470 bytes per datagram were sent repeatedly with different data rate.

Characterization of limited bandwidth network was done by investigating each network parameters measured in the previous section. 1) Bandwidth capacity was investigated and characterized by using the result of bandwidth measurement. The data was retrieved from source host and destination host during the measurement process. We investigated bandwidth properties, i.e. allocated bandwidth, upstream and downstream capacity, and ratio between upstream and downstream capacity. 2) End-to-end delay characterization was conducted by using RTT (Round Trip Time) variation of ICMP packets delivery. End-to-end delay between source host to destination host was obtained by calculating half of the RTT (equation (1)). Investigation of end-to-end delay refers to ITU-T recommendation G.114 [13] . The delay that occurred in each second during the measurement processes was also investigated. 3) Packet loss rate is the number of packets which are not received by destination host. In our case study, PLR was characterized by using measurement results from two different locations at the testbed network. PLR was calculated using equation (2). Furthermore, we investigated whenever packet loss has occured, whether it has occurred during the measurement process or has only occurred at certain times. We also investigated how much the packets loss at certain times and how many times the emergence of packets loss during the measurement process.

$$End\text{-}to\text{-}end\ Delay = \frac{RTT}{2} \quad (1)$$

$$PLR = \frac{P_{sent} - P_{Received}}{P_{sent}} \ x \ 100\% \quad (2)$$

2.3. Simulating and Characterizing Multimedia QoS

We characterized multimedia QoS through network simulation with a simulation model that approaches real network conditions. The simulation was set up according to the results of measurement parameters: bandwidth, end-to-end delay, and packet loss rate.

2.3.1. Simulating Multimedia Delivery

In the simulation, we were sending audio and video packet along with background traffic. Parameters configured for the delivery of audio were data rate and size of the transmitted audio. In this research, we used data rate which representing Vorbis technology. Same as the delivery of audio, data rate for video delivery can use the size required for the delivery of compressed video such as VP8 [14] . Based on this data rate, we determine the length of each packet and the interval between packets. The size of audio and video packets used in the simulation depends on the data rate and duration of simulation and was calculated using the equation (3).

$$Packet\ Size = Packet\ Length * \frac{1}{interval} * duration \tag{3}$$

For mechanism of audio and video delivery, we installed a VoIP application in the first host as a sender and VoIP application in the second host as a receiver. Similarly, the video delivery used UDP applications installed together with VoIP applications which the first host as server and the second host as a client. Audio and video packets were transmitted simultaneously from the first host to the second host. We included background traffic so that approached real conditions in the field. Background traffic was set up using UDP traffic from source to destination. In addition, the delivery of the packets were limited to maximum delivery time of 100 seconds.

Simulation of multimedia delivery was repeated using several different data rates. Multimedia packets which were consisted of audio and video were sent over the network with the lowest bandwidth allocation up to the highest during the measurements. Parameters of end-to-end delay and packet loss were also configured according to network characteristics in the field.

2.3.2. Characterizing Multimedia QoS

In characterizing multimedia QoS, we measured and analyzed the network parameters using objective testing. After measuring and analyzing the multimedia packet delivery, we investigated the characteristics of multimedia QoS using PESQ [15] and PSNR. PESQ was used to estimate the audio quality and PSNR was used to estimate the video quality. Then we applied user's perception expressed in Mean Opinion Score (MOS) to characterize multimedia QoS. The correlation between PSNR and MOS can be seen in Table 1 while PESQ using MOS scale. MOS value was calculated using the E-model formulation [16] . The first step in E-model formulation is calculating the rating factor R.

$$R = 93.2 - I_d - I_e \tag{4}$$

$$I_d = 0.024d + 0.11(d - 177.3)H(d - 177.3) \tag{5}$$

$$I_e = 30\ln(1 + 15e) \tag{6}$$

The equation for R aims to describe disturbance factors on the network that affect multimedia QoS. Rating factor R is calculated using equation (4). The variable of I_d is a factor of quality degradation caused by delay. The variable of I_e is a factor of quality degradation caused by compression technique and packet loss. The constants in equation (4), (5) and (6) are the recommended numbers by [16] which are not changed because of the network conditions. To calculate MOS (ITU-T P.800) based on the estimated R-value, there are provisions presented in equation (7). ITU-T recommendation G.107 [16] has established a classification of user satisfaction based on rating factor R of the E-model formula and the estimated MOS as shown in Table 2.

$$MOS = \begin{cases} 1 & if\ R < 0 \\ 1 + 0{,}035\,\mathrm{R} + \mathrm{R}(\mathrm{R} - 60)(100 - \mathrm{R}) \times 7 \times 10^{-6} & if\ 0 < R > 100 \\ 4{,}5 & if\ R > 100 \end{cases} \tag{7}$$

Table 1. Correlation between PSNR and MOS

PSNR	MOS (upper limit)
> 37	4,5
32 − 37	3,5
26 – 31	2,5
20 – 25	1,5
< 20	0,5

Table 2. Relation between MOS and User Satisfaction

R-Value (lower limit)	MOS (lower limit)	User satisfaction
90	4,34	Very satisfied
80	4,03	Satisfied
70	3,60	Some users dissatisfied
60	3,10	Many users dissatisfied
50	2,58	Nearly all users dissatisfied

2.4. Validating Important Issues of the Methodology

In this section we consider to discuss the validation of the methodology proposed in this research work.

1. Does our measurement already reflect the characteristics of rural networks?

We perform measurement through VPN (Virtual Private Network) channel to ensure that sent probe packet is answered by corresponding destination host, not by any router in middle of the network link. Measurement host located in the testbed network is configured as a VPN client and a VPN server is located in a high-speed academic network. In this measurement, we use the same IP address segments between measurement hosts as if without any intermediate router. Each measurement is monitored carefully so that our measurement can be justified. To conduct measurement in real network condition, we designed measurement scenario as detail as possible.

The measurements are conducted repeatedly to ensure that the measurement results will have enough information for characterizing the network. In the measurement scenario that we have designed, bandwidth measurements are performed 200 times, end-to-end delay measurement are performed 2000 times, and packet loss measurements are performed 200 times. However, we limit each measurement only ± 10 seconds to keep the interests of other users. We set the interval time between measurements as an average of 30 minutes. In the measurements of bandwidth capacity, we blocked other users from using the same network so that the maximum bandwidth capacity can be measured.

We verify the accuracy of the network measurements as follows. First, we compare bandwidth capacity from measurement with bandwitdh capacity advertised by Internet Service Provider (ISP). The results showed that the bandwidth capacity from the measurement was similar with the bandwidth capacity advertised by the ISP. Second, we compare the measurement results on the first location of testbed network with the second location. It shown similarity in the measurement results, therefore it can be concluded that the measurements have been able to capture the network characteristics.

2. Why do we characterize multimedia QoS through a simulation?

There are two reasons why we characterize multimedia QoS through simulation. First, network capacity is limited, particularly in rural area like in Indonesia. Testing with audio-video streaming directly on the network requires the majority of network resource allocation and it usually takes a long time. While our scenario requires streaming which is repeated many times, multimedia QoS characterization performed directly on the network will overload the network itself. On the other hand, the user has high intensity to use the network resources.

Secondly, our methodology requires two hosts for network measurement at the same time. However, we are not able to perform and control the experiments in two different places simultaneously. With simulation, efforts for characterizating multimedia QoS become more flexible. We can perform measurements repeatedly without burdening the network in the field.

3. Case Study of Network Measurements

This study uses testbed network in Cianjur Regency, West Java Province of Indonesia. The network consists of real-time multimedia delivery system that lies on a network of two educational entities based on Asymmetric Digital Subscriber Line (ADSL) access network. In its architecture, we define two Local Area Networks (LANs) in Cianjur and one Campus Network in

Bandung: i) LAN A is a local network of educational entity in location A where some end-systems consist of data end-system, audio end-system, and video end systems are deployed, ii) LAN B is a local network of educational entity in location B with the same end-systems as in LAN A, and iii) Campus network where a multimedia proxy server is installed to facilitate multimedia delivery between two LANs. The two educational entities may be located in several km until some tens km far away from each other. On the other hand, the distance between campus network in Bandung and the educational entities in Cianjur regency is around 60 km.

In many cases, broadband Internet connections in educational entities in many rural areas in Indonesia are predominated by ADSL access network as their last mile connection. As shown in Figure 1, two educational entities are connected to a telephone company through fixed line telephone cable. The data sent from a Digital Subscriber Line Access Multiplexer (DSLAM) is splitted by a splitter and sent to DSL modem using a dedicated point-to-point connection. The data connection to the global network is established using dial-up connection with Point-to-Point Protocol (PPP). In several educational entities in Cianjur, bandwidth capacities of the last mile connections are typically ranging from 512 kbps until 2 Mbps.

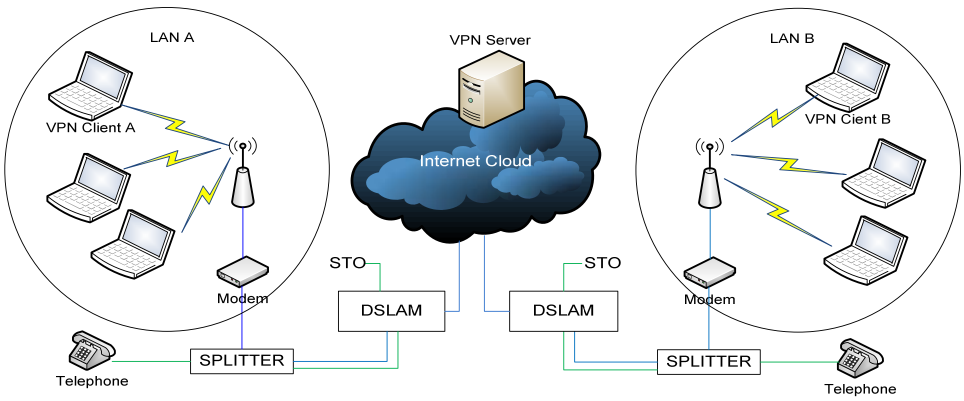

Figure 1. A topology of testbed network

3.1. Measurement Scenario

Measurement scenario is designed to determine characteristics of the testbed network. In the scenario, several network parameters including bandwidth, end-to-end delay, and packet loss are measured as described below:

1. Bandwidth

   In general, Internet bandwidth consists of downstream bandwidth and upstream bandwidth. In our experiments, the downstream bandwidth was measured by flooding the link bandwidth using Iperf application. Each measurement was run for up to 10 seconds with interval between measurements was approximately 30 minutes. On the other hand, the measurement of the upstream bandwidth was done by sending files of a certain size from client to server.
2. End-to-end Delay

   End-to-end delay was measured using ICMP echo request and ICMP echo response. Source host (client) sent ICMP echo request (1,000 packets) and destination host (server) responded with ICMP echo response packets of the same size. The packets have different sizes ranging from 32 bytes up to 1472 bytes. The amount of 32 byte refers to the default value of PING command, while the size of 1472 bytes refers to MTU value of ADSL networks. We repeated the experiment 10 times with 30 minutes time intervals.
3. Packet Loss Rate (PLR)

   PLR parameter was measured using various sizes of UDP datagrams with 1470 bytes per datagram on Iperf application. We delivered UDP datagrams from 564 Kbps until 10 Mbps. The size of 564 Kbps was calculated from the data rate of audio-video packet when applying Vorbis and VP8 compression schemes.

3.2. Results of Network Measurement

This section presents the measurement results in accordance with the scenario presented in the preceding section. Furthermore, we investigate each parameter to obtain the characteristics of the rural network.

1. Bandwidth

   It was observed that bandwidth allocation of the testbed network during the measurement process has fluctuated. From ten times of measurement, it was known that the bandwidth ranged from 0.13 Mbps up to 12.5 Mbps with average 8.20 Mbps in the first location (LAN A). In the second location (LAN B), the bandwidth ranged from 0.546 Mbps to 12.1 Mbps with average 7.83 Mbps. We found that the lowest bandwidth on LAN A was 0.13 Mbps or 1.3% from subscription bandwidth up to 10 Mbps. In some measurements, the maximum bandwidth allocated exceeded the capacity subscribed bandwidth of up to 12.5 Mbps or approximately 25% from subscribed bandwidth.

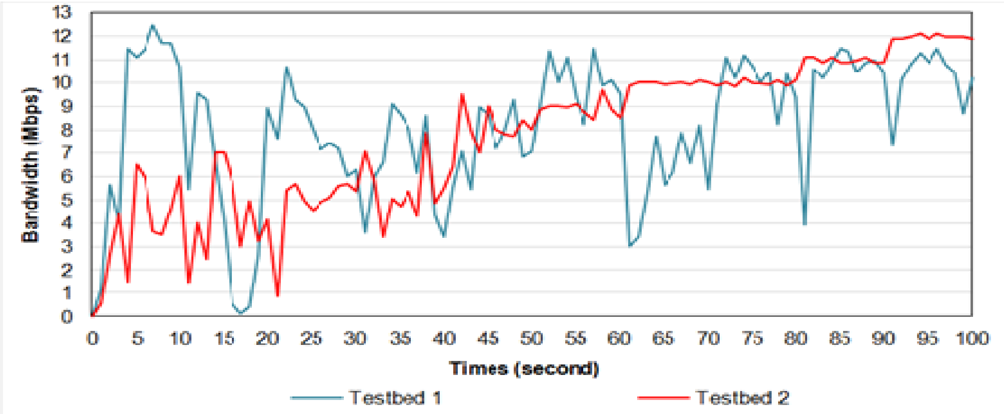

Figure 2. Bandwidth allocation of testbed network

   Upstream bandwidth has fluctuated in line with the bandwidth allocation although the ratio was not always directly proportional. The ratio between upstream and bandwidth allocation in the two LANs was ranging from 4 to 10. In order to characterize downstream, we reduce the bandwidth allocation by the upstream bandwidth. We also compared the downstream and upstream. Figure 3 shows that downstream was always larger than upstream with ratio of 3 to 9.

   We found that decrease of bandwidth do not directly decrease upstream bandwidth with the same ratio. When the bandwidth allocation is decreased, we found the ratio of upstream to downstream sometimes increased. The results of measurements on the first location showed when the bandwidth allocation was decreased to 4.71 Mbps, upstream to downstream ratio was increased to 1:3 (1.18 Mbps). On the second location, when bandwidth allocation was only 3.95 Mbps, the ratio of upstream to downstream was 1 : 3 (0,97 Mbps). To assure this result, we tried on other network and obtained similar results with testbed network.

2. End-to-end delay

   We calculated end-to-end delay using equation (1) and ignored no response packets. End-to-end delay on the first and second location was ranging from 7 ms to 556 ms and from 8 ms to 1918 ms, respectively. RTT was high only in a few moments, and this would cause interference on multimedia streaming. The average of end-to-end delay on the first and second location was 25.68 ms and 88.79 ms.

3. Packet Loss Rate

   PLR was calculated using equation (2). From the measurements we obtained that the PLR in LAN A and LAN B were 5.8% and 8.6%, respectively.

$$PLR_A = \frac{3176}{54415} \times 100\% = 5.8\%$$

$$PLR_B = \frac{5128}{59763} \times 100\% = 8.6\%$$

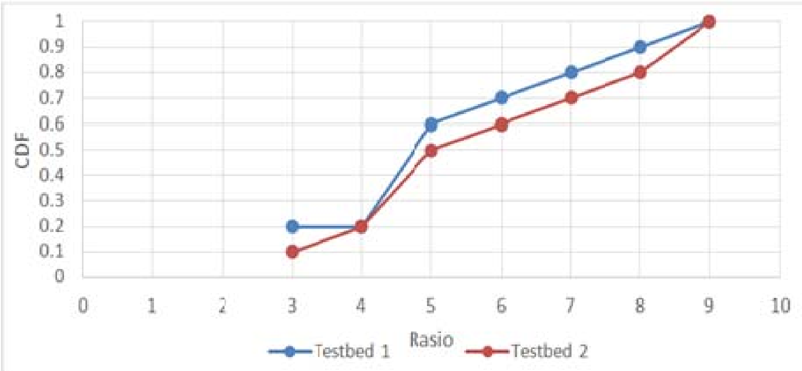

Figure 3. Downstream to Upstream Ratio

Figure 4. Packet Loss Rate of the Testbed Network

In this study we also investigate the occurrence of packets loss during network measurements. The experiments reveal that packet loss only occurs at certain times. Figure 4 shows the occurence of packet loss rate for the LAN A and LAN B during the measurements. It was shown that the packet loss rate varied from 8% to 90%.

4. Simulation Study of Characterizing Multimedia QoS
4.1. Validating Important Issues of the Methodology

We build a network simulation model that represents multimedia delivery in the testbed network which applies WebRTC technology. From the network topology on Figure 1, we draw the simulation model based on OMNeT++ Network Simulator with INET Framework. The simulatioin model on Figure 5 consists of several components, i.e. internet cloud, routers, access points, and hosts. In general, we arrange the components into three groups: Location 1 or LAN A to represent educational entity 1, Location 2 or LAN B to represent educational entity 2, and internet cloud to represent the Internet. In LAN A dan LAN B, each host is connected to an access point that works on frequency of 2.4 GHz with a maximum data rate channel 54 Mbps. The access point is connected to a router which acts as a gateway from each LAN to the Internet cloud via ethernet link with data rate of 100 Mbps. The data rate of connection between both routers to the internet cloud is set to the values that approach real condition in the field.

Data rate between router and internet cloud is configured based on the upstream bandwidth capacity in each location. Simulation study is built using symmetric bandwidth, while testbed network has asymmetric bandwidth which is in general, the upstream bandwidth is always smaller than the downstream bandwidth. We determine the data rate used in the simulation was twice of upstream bandwidth. The internet cloud is assumed as high speed network and is configured to 100 Mbps.

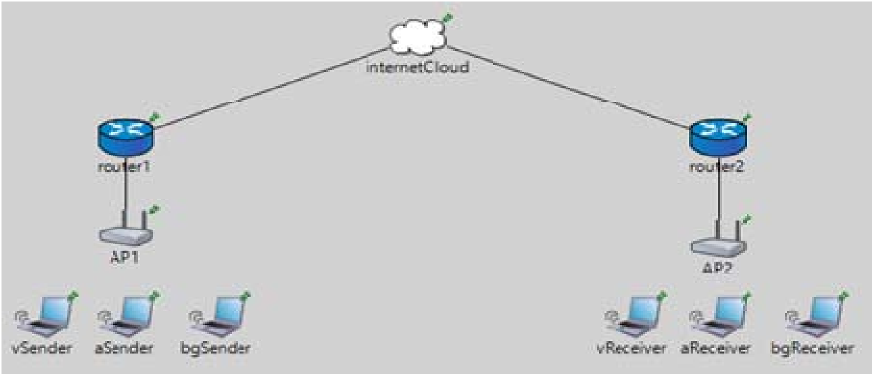

Figure 5. Network Simulation Model

We use measurement results on each network testbed to configure end-to-end delay parameter on the simulation. From the measurement results, end-to-end delay parameter is set to 25.68 ms at LAN A and 88.79 ms at LAN B. Delay parameter in internet cloud is assumed and configured to 5 ms. In addition, for configuring Bit Error Rate (BER) parameter on the link between router and internet cloud, it is assumed that the value for the parameter is 10^{-5}.

4.2. Multimedia Content Delivery Scenario

In this research, simulation of multimedia delivery that consists of audio, video, and background traffic is configured using one-way communication. The use of background traffic is aimed to approximate the real network condition. The simulation model is run on a laptop for 100 seconds. The delivery of audio, video, and background traffic was configured as follows.

1. Audio

To simulate audio packets delivery, we used *simpleVoIPSender* application that was installed in sender host. In the receiver host, it was installed *simpleVoIPReceiver*. In this simulation, audio packets was transmitted using *talkPacketSize*=160 bytes and *packetizationInterval*=0.02 seconds. This configuration was taken to approach audio delivery using Vorbis with data rate of 64 Kbps.

2. Video

We used *UDPVideoStreamSvr* that was installed in sender host. In the receiver host, we installed *UDPVideoStreamCli*. In this simulation, video packets was transmitted using *packetLen*=1024 bytes and *sendInterval*=0.016 seconds. This configuration was used in order to approach video delivery using VP8 with data rate of 500 Kbps.

3. Background Traffic

We used *UDPBasicApp* that was installed in sender host. While in the receiver host *UDPSink* was installed. In this simulation, video packets was transmitted using messageLength=512 bytes and *sendInterval*=0.032 seconds. Background traffic was transmitted during audio and video delivery process.

4.3. Characterization of Multimedia QoS

In this section we describe characterization of multimedia QoS both audio and video delivery. We investigate end-to-end delay and packet loss to obtain these characteristics. End-to-end delay of audio was varied approximately 0.14 seconds to 1.2 seconds. Whereas its average was 0.16 seconds. The highest delay was occurred in the first second, which was 1.2 seconds and next second was varied between 0.14 seconds to 0.19 seconds. This condition was similar as the network characteristics where in first second, bandwidth allocation tended to be small. According to ITU-T G.114, end-to-end delay of audio transmission has bad until good quality. However, overall end-to-end delay can be accepted with good quality.

Audio packets that was delivered for 100 seconds experienced 42 packets loss with varied rate between 0% to 25%. If the packet loss was observed every second, the highest packet loss (25%) occurred when simulation running until 2 seconds, whereas the next packet loss rate was varied between 0% to 5%. PLR was configured at 1.3%. According to ITU-T G.1010 [17] , PLR for audio conversations should be less than 3%. Overall, audio quality in this simulation can be well accepted because the value of PLR is still less than 3%, which is 1.3%.

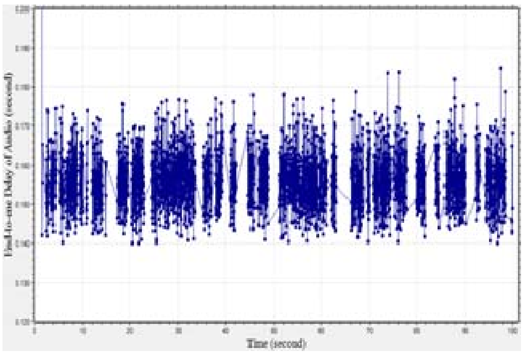

Figure 6. End-to-end delay of audio

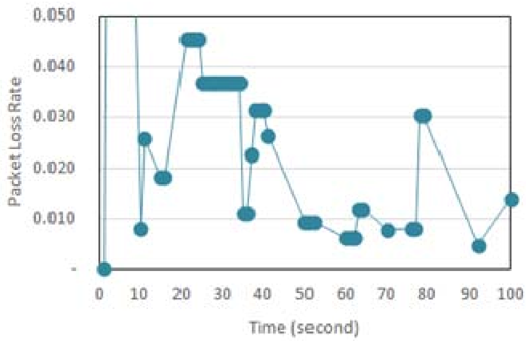

Figure 7. Packet loss rate of audio

End-to-end delay of audio was varied approximately 0.121 seconds to 0.172 seconds with average value of 0.135 seconds. Based on ITU-T G. 114 recommendation, end-to-end delay video delivery that can be well accepted is less than 0.4 seconds. In the simulation, end-to-end delay of video delivery was 0.135 seconds. This showed that the video can be accepted with good quality.

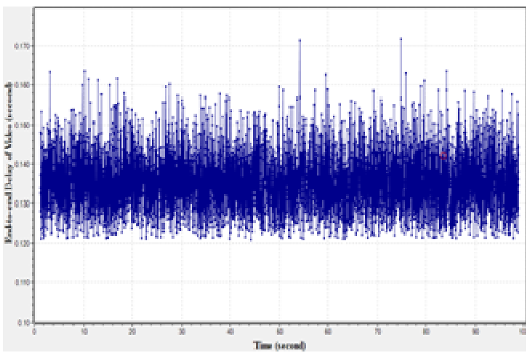

Figure 8 End-to-end delay of video

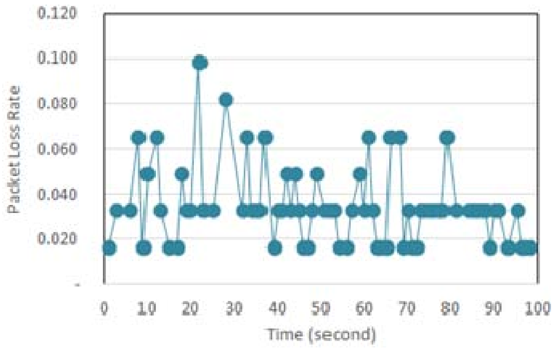

Figure 9. Packet loss rate of video

Within 100 seconds amount of 6,250,000 bytes video packets were delivered from the source and 6,088,704 bytes data were received at the destination. From this experiment, we calculated the PLR at approximately 2.6% that amount of 161,296 bytes packets were lost. Based on our observation, it was shown that a number of packet loss rate were not the same at every time, but it only occurred for certain second. According to ITU-T G. 1010, PLR for video delivery should be less than 1%. When the packet loss was observed at every second, video delivery at certain times was suffered loss that can not be tolerated.

Furthermore we calculated R-values based on equation (4). R-value of audio and video delivery obtained are 84.11 and 80.08. MOS value was calculated using equation (7). MOS of audio delivery was 4.17 and MOS of video delivery was 4.03. Based on Table 2, the quality of multimedia delivery over testbed network achieved user satisfaction level which was satisfied. It can be concluded that the testbed network was feasible for real-time multimedia delivery such as distance learning activities through virtual classroom technology.

5. Conclusion

We have developed a methodology for characterizing multimedia QoS particularly in limited bandwidth network, with a case study in in rural area in Indonesia.The methodology is divided into two important phases i.e. network measurements in the field and simulation of multimedia delivery. The first phase consists of several activities, namely (i) selecting testbed network; (ii) choosing the testbed network parameters; (iii) measuring and characterizing the testbed network, with a case study in rural area in Indonesia. The second phase consists of the following activities: (i) building network simulation model that approaches the actual state of the testbed network; (ii) simulating multimedia delivery; and (iii) investigating the multimedia QoS.

From experiments in the testbed network, we obtained that bandwidth allocations were fluctuated from 3.95 Mbps up to 12.01 Mbps. Comparison between upstream and downstream bandwidth was approximately 1:6. The averages of end-to-end delay in the first and second location were 25.68 ms and 88.79 ms. Packet loss rate in the firtst and the second location were 5.8% and 8.6%. The simulation results showed that audio-video characteristics were accepted and user satisfaction level was satisfied.

References

[1] Silvestre-Blanes J, Almeida L, Marau R, Pedreiras P. Online QoS Management for Multimedia Real-Time Transmission in Industrial Networks. *IEEE Transactions on Industrial Electronics.* 2011; 58(3): 1061-1071.

[2] Veraraghavan W, Weber S. Fundamental Tradeoffs in Distributed Algorithms for Rate Adaptive Multimedia Streams. *Computer Networks.* 2008; 52(6): 1238-1251.

[3] Kesavan S, Jayakumar J. Conventional to Cloud: Detailed Survey and Comparative Study of Multimedia Streaming Rate Adaption. *International Journal of Cloud Computing and Services Science.* 2014; 3(2): 97-112.

[4] Kurose JF, Ross KW. Computer Networking: A Top-Down Approach Featuring the Internet. Sixth Edition. San Fransisco: Addison Wesley. 2013.

[5] Bandung Y, Erwin, Hutabarat MT. *Network Characterization for Delivering Multimedia-based Learning in Rural Areas.* Proceedings of the Intl. Conference on Advances in Education Technology. Bandung. 2014.

[6] Sun E, Shen X, Chen H, Wang C. Distributed Image Compression and Transmission Scheme in Wireless Multimedia Sensor Networks. *TELKOMNIKA Indonesian Journal of Electrical Engineering.* 2014; 12(1): 661-668.

[7] Wei L, Yong-Feng C, Ya L. An Enhanced Safety Algorithm for Network QoS Multicast Routing Optimization. *TELKOMNIKA Indonesian Journal of Electrical Engineering.* 2014; 12(4): 2797-2807.

[8] International Telecommunication Union. Recommendation ITU-T E.800. Terms and definitions related to the quality of telecommunication services. 2008.

[9] Iversen VB, Teletraffic Engineering and Network Planning. Denmark: Technical University of Denmark. 2010.

[10] Revathi P, Balasubramanian R. Efficiency Analysis on QoS Multicast Routing Protocols Under Cross-Layer Approach With Bandwidth Estimated Admission Control. *International Journal of Algorithms, Computing and Mathematics.* 2009; 2(3): 37-45.

[11] Raghavendra R, Belding E. *Characterizing high-bandwidth real-time video traffic in residential broadband networks.* Proceedings of the 8th Intl. Symposium on Modeling and Optimization in Mobile, Ad Hoc and Wireless Networks (WiOpt). Avignon. 2010; 597-602.

[12] Dischinger M, Haeberlen A, Gummadi KP, Saroiu S. *Characterizing Residential Broadband Networks.* ACM SIGCOMM Internet Measurement Conference. San Diego. 2007.

[13] International Telecommunication Union, Recommendation ITU-T G.114. One-Way Transmission Time. 2000.

[14] Feller C, Wuenschmann J, Roll T, Rothermel A. *The VP8 Video Codec - Overview and Comparison to H.264/AVC*. Proceedings of the IEEE International Conference on Consumer Electronics (ICCE). Berlin. 2011: 57-61.

[15] Rix AW, Beerends JG, Hollier MP, Hekstra AP. *Perceptual evaluation of speech quality (PESQ), An Objective Method for End-to-end Speech Quality Assessment of Narrowband Telephone Networks and Speech Codecs*. Proceedings of the IEEE Intl. Conference on Acoustics, Speech, and Signal Processing (ICASSP). Salt Lake City. 2001; 2: 749-752.

[16] International Telecommunication Union, Recommendation ITU-T G.107. The E-model: a computational model for use in transmission planning. 1999.

[17] International Telecommunication Union, Recommendation ITU-T G.1010. Quality of Service And Performance. 2000.