An Analysis of Quality of Service (QoS) In Live Video Streaming Using Evolved HSPA Network Media

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Abstract—Evolved High Speed Packet Access (HSPA+) is a mobile telecommunication system technology and the evolution of HSPA technology. This technology has a packet data based service with downlink speeds up to 21.1 Mbps and uplink speed up to 11.5 Mbps on the bandwidth 5MHz. This technology is expected to fulfill and support the needs for information that involves all aspects of multimedia such as video and audio, especially live video streaming. By utilizing this technology it will facilitate communicating the information, for example to monitoring the situation of the house, the news coverage at some certain area, and other events in real time. This thesis aims to identify and test the Quality of Service (QoS) performance on the network that is used for live video streaming with the parameters of throughput, delay, jitter and packet loss. The software used for monitoring the data traffic of the live video streaming network is wireshark network analyzer. From the test results it is obtained that the average throughput of provider B is 5,295 Kbps bigger than the provider A, the average delay of provider B is 0.618 ms smaller than the provider A, the average jitter of provider B is 0.420 ms smaller than the provider A and the average packet loss of provider B is 0.451% smaller than the provider A..

Keywords— Evolved HSPA, Live Video Streaming, throughput, delay, jitter, packet loss.

1. Background

Nowadays the development of mobile telecommunications technology experience very rapid progress. Evolved High Speed Packet Access (HSPA +) represents the technology in mobile telecommunications system which constitutes an evolution of HSPA technology. This technology has a data packet-based services with downlink speeds of 21.1 Mbps and uplink speed of 11.5 Mbps in the 5MHz bandwidth. Indeed, the technology is expected to meet the basic necessities for the life of people who need the support for their daily activities, which is one of the example is the need for information involving all aspects of multimedia such as video, audio, and text. The effectiveness and efficiency of these technologies is much expected to help and to support this based video streaming technology. Utilization of this technology will facilitate the delivery of information, e.g. for monitoring the condition of the house, the news coverage from a certain place, and other events in real time or live. Apostolopoulos et.al say in the world of the Internet, live streaming is a technology that is capable to compress or shrink the size of audio and video files to be easily transferred through the Internet. The distribution of audio and video files is continuously processed. From the point of the process, streaming means the technology of file delivery from the server to the client via a packet-based network.

This final project research aims to identify and examine the performance of the Quality of Service (QoS) on a live video streaming service with Evolved HSPA network media. The parameters of QoS analyzed, among others are: Delay, Jitter, Packet Loss, and Throughput. The result of the analysis of the final project is expected to be one of the reference for the users in selecting a good network service.

2. Theoritical Framework

2.1 *Evolved HSPA*

Evolved HSPA or HSPA+ constitutes the name of a set of HSPA development beginning with realease 7. With this HSPA Evolved, the type of specific higher-order modulation can be supported to the uplink of 16QAM and downlink of 64QAM. The modulation of 16QAM enables the obtained uplink peak data speed up to 11.5 Mbit/s and 21.1 Mbit/s for downlink peak data speed with 64QAM modulation [1].

With the addition of 2x2 MIMO (Multiple Input Multiple Output) antenna then improves the support for Evolved HSPA downlink direction. The system uses two antennas to increase significantly the peak downlink speeds effectively in order to obtain peak data speed of 28 Mbit/s. If 2x2 MIMO is combined with 64 QAM modulation, the downlink peak data speed can reach of 42 Mbit/s. Latency is also reduced with this Evolved HSPA [3].

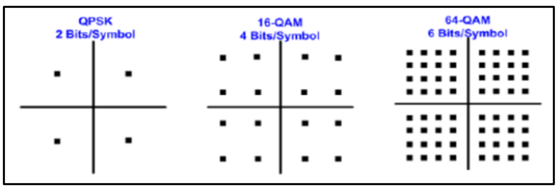

Fig.1 Higher Order Modulation (HOM) [3]

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HSPA supports the 16 QAM modulation of the downlink direction and QPSK in the uplink direction. As shown in Figure 2.1 data capacity (bits/symbol) increases as the transition from QPSK to 16QAM and 64 QAM. Evolved HSPA R7 is using 64QAM for downlink, which increases data speed of 50% in good signal conditions (high SNR). On the uplink, 16QAM increases the data speed. Wireless signals transmitted with high modulation techniques have more sensitivity against the interference and need higher SNR at the receiver for the success of the demodulation process. HOM significantly increases data speed for users with high SNR. Thus, the traffic to these users can be served more quickly, leaving the B Node with more time and resources to serve users in weaker signal areas (such as the edge of the cell). Overall, this will provide high data speed and improve the experience of users who are inside the cell [3].The network architecture of WCDMA release 7 can be seen in Fig.2 below.

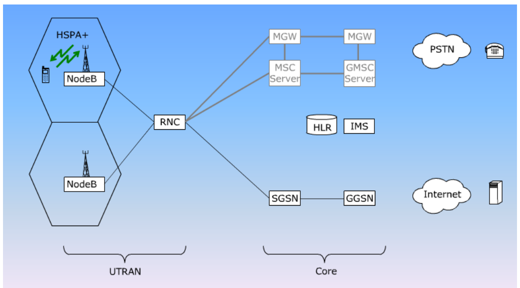

Fig.2 Architecture of WCDMA Release 7 (Evolved HSPA)

2.2 *Streaming Technology*

Streaming is a technology to play or run a video or an audio file directly or real-time or by pre-recorded on a server machine. First, the data file must be encoded using a particular data rate matching to transmit over a network suitable with the bandwidth capacity of the client. Thus it is necesssarry to carry out the encoding of video or audio files at varying data speeds rate, then the client can adjust the speed of the network and the speed of data access system.

At the time the video files are streamed they will form a buffer on the client's computer and the audio or video data will start downloading to the buffer that has been formed on the client's computer. Once the buffer is filled within in seconds, then the video or audio file will automatically run by the system. The system will read the information from the buffer while it is still doing the process of downloading files so that the stream process remains going to the client computer as shown in Fig.3 below [4].

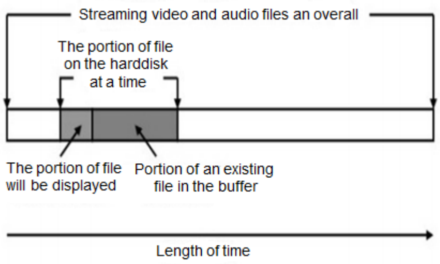

Fig.3 The working principle of streaming [4]

2.3 *Quality of Service*

Quality of Service is defined as a measure of how well the network and constitutes an attempt to define the characteristics and nature of the service. QoS refers to the ability of a network to provide better service at a specific network traffic through various technologies. QoS becomes a major challenge in IP-based networks and for the Internet as a whole. The goal of QoS is to meet the needs of various services, using the similar infrastructure. QoS offers the ability to define the attributes of the services provided, both qualitatively and quantitatively.

Computer network performance can vary due to several problems, such as problems of bandwidth, latency and jitter, which can make fairly significant effect for many applications. For example, voice communications (such as VoIP or IP Telephony) as well as video streaming can make the user frustrated when the application data packets streamed over the network with insufficient bandwidth, unpredictable latency, or excessive jitter. These features of Quality of Service (QoS) can make bandwidth, latency, and jitter predictable and is matched with the needs of applications used in the existing network.

Parametric Quality of Service (QoS)include:

1) Delay

Delay is the total time a packet takes to cover the distance from the origin (sender) to the destination (receiver). The delay from sender to receiver is comprised of hardware latency, delay of access, and delay of transmission. Delay most often experienced by the transmitted traffic is the transmission delay [2]. TIPHON recommends that the delay time is not longer than 150 ms for a variety of applications, with a limit of 450 ms for voice communication which is still receiveable, as shown in Table 1 below.

Table1 Delay Standardization based on TIPHON [6]

Delay (ms)	Quality
< 150	Very Good
< 250	Good
< 350	Enough
< 450	Poor

2) Jitter

Jitter is defined as the variety of the delay occuring due to the time difference or interval between the arrival of a packet at the receiver. Jitter creates problems causing the recipient must receive or encode or display the frame with a constant rate, and each of late coming frame will complicate the reconstruction of the video received. To resolve the jitter then the data packet coming is first collected in the jitter buffer during a predetermined time until the packet is received at the receiver side in the correct order. Jitter can cause data loss especially at high transmission speeds [5]. In Table 2it is explained the jitter grouping based on standard of TIPHON.

Table2 Jitter Standardization based on TIPHON [6]

Jitter (ms)	Quality
0	Very Good
< 75	Good
< 125	Enough
< 225	Poor

3) Packet Loss

Packet loss becomes a major cause of the weakening of the audio and video of multimedia streaming. This loss can cause damage of the reconstructed (decoded) of the video quality. One of the causes the packet loss is the queue exceeding the capacity of buffers at each node. Some causes of packet loss, are:

a. Congestion which is caused by the occurrence of excessive queues on the network.
b. Nodes that work beyond the capacity of the buffer.
c. Memory whis is limited to the node.
d. Policing or control of the network to ensure that the amount of traffic flowing is in accordance with the bandwidth capacity. If the volume of the flowing traffic in the network exceeds the capacity of existing bandwidth, then the policing control will remove the excess traffic.

Table3 Packet Loss Standardization based on TIPHON [6]

Packet Loss (%)	Quality
0	Very Good
3	Good
15	Enough
25	Poor

4) Throughput

Throughput, is the speed or rate of the effective data transfer, measured in bps [2]. Sometimes Throughput is always associated with bandwidth, because indeed it can also be considered as the actual bandwidth conditions. Throughput is the total number of observed successful packet arrival at destination during a specified time interval divided by the duration of the time interval (equal to the number of successful delivery of IP packet per service-second). When sending data is faster than the available bandwidth, there will be congestion on the network affecting the video quality that will be received. Here is the calculation of the formula to find the value of throughput.

3. Research Methods

3.1 System Architecture

At this stage, it is discussed the design system of live video streaming media with Evolved HSPA mobile network. The system design of this final project is using the system architecture as shown in Fig.4. Clients are connected to the Media Server can access the live video streaming multimedia resources derived from the webcam that has been compressed by the encoder.

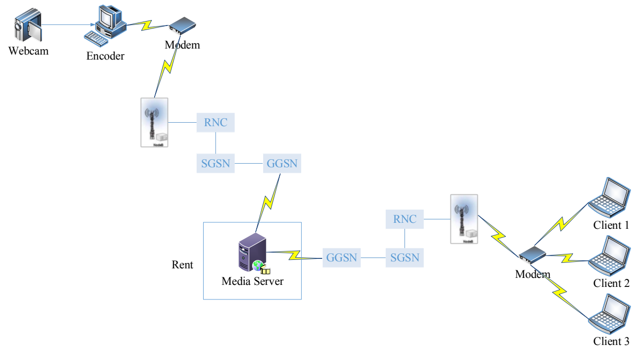

Fig.4 The System of the Live Video Streaming

This system architecture is using a webcam as a source of information, three Notebooks as the clients, one Notebook as an encoder and a modem as a data transmission medium with Evolved HSPA network connections in which the performance of the live video streaming on the network will be analyzed.

3.2 Flowchart Packet of Capturing the Client

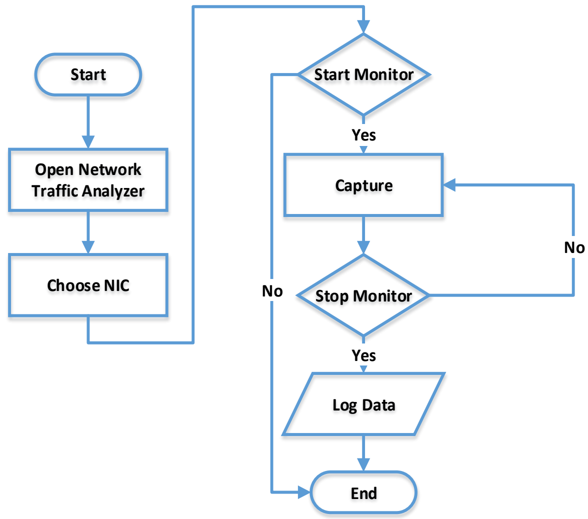

Fig.5 Flowchart of packet capturing process on the client

In Fig.5, when the software is run, then the list of NIC (Network Interface Card) will be displayed. Select one Network Interface that will be captured. Press the Start Monitor button to start the capture of the data traffic. To stop the capture of data traffic press the Stop Monitor button, it will display the log data containing the information of the captured traffic data.

4. Results and Discussion

This chapter discusses the network speed measurements made immediately before network (QoS) performance test according to the scenario using webtools (Speedtest.net). The measurement parameters of throughput, delay, jitter and packet loss are carried out according to the scenario, the network performance test with a variety of clients running live video streaming simultaneously using Evolved HSPA network. In this scenario there are two providers used, namely A and B. In this test the client to access the live video streaming webcams with a specified variety of time.

4.1 Test Based on Network Speed

Test of the speed of data retrieval of Evolved HSPA network is using the webtools (speedtest.net). Data retrieval is carried out at the time before the data collection of the live video streaming network quality (QoS). Fig.6An example of the results of Evolved HSPA networks Speedtest.

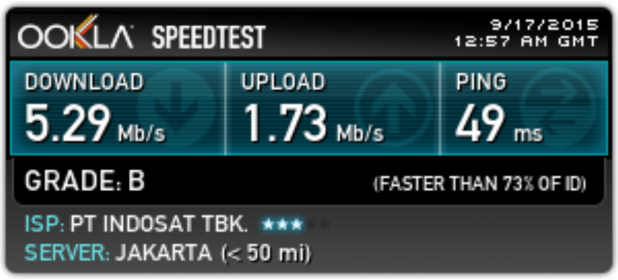

Fig.6 Speedtest results of evolved HSPA network

Teble4 SpeedTest Results of Evolved HSPA Network of Provider A and B

No.	Time	Provider A	Provider B
		Download	Download
1	24.00	5,532 Mbps	7,620 Mbps
2	04.00	9,208 Mbps	9,332 Mbps
3	08.00	3,244 Mbps	4,958 Mbps
4	12.00	3,848 Mbps	5.104 Mbps
5	16.00	5,002 Mbps	6,284 Mbps
6	20.00	4,340 Mbps	3,116 Mbps

From Table 4of the test Results of Evolved HSPA Network Speed of provider A it is obtained the highest download speeds of 9.208 Mbps during testing at 04.00. In Table 4.2 the Network Speed Test Results of Evolved HSPA of Provider B it is obtained that the highest download speeds is 9.332 Mbps during testing at 04.00.

4.2 Testing throughput based on the variety of clients and provider

Based on the results of test data throughput with a variety of clients and providers which have been averaged is presented in Table 5 and Table 6 as follows.

Table 5 Average throughput from Encoder to Media Server provider A and B

No.	Number of Client	Provider A	Provider B
		Average Throughput (Kbps)	Average Throughput (Kbps)
1	**1 Klien**	**468.729**	**475.667**
2	**2 Klien**	**464.625**	**474.479**
3	**3 Klien**	**466.979**	**476.063**

Table 6 Average throughput from Media Server to Client provider A and B

No.	Number of Client	Provider A	Provider B
		Average Throughput (Kbps)	Average Throughput (Kbps)
1	**1 Klien**	**446.354**	**444.417**
2	**2 Klien**	**424.979**	**441.052**
3	**3 Klien**	**405.319**	**407.069**

It is obviously the access time will determine the quality of throughput on the network used. This is due to the density of traffic on the network used will vary depending on the network usage time. From the data it can be said that the greater value of the throughput on the network is-the lower of delay will be. This shows that there is a relationship between the throughput with the delay which is inversely proportional to each other. The throughput can be influenced by the level of network traffic dense so the availability of bandwidth for each user will be divided evenly.

4.3 Testing delay based on the variety of clients and provider

The scenarios of delay test based on varietys of the client and the provider is carried out simultaneously with the sampling of data throughput, jitter and packet loss using wireshark network analyzer software. The test data delay variety based client and provider that has been averaged is shown in Table 7 and Table 8 as follows.

Table 7 Average delay based on variety of clients provider A

No.	Number of Client	Average Delay End to End (ms)
1	**1 Klien**	**31,153**
2	**2 Klien**	**32,949**
3	**3 Klien**	**32,913**

Table 8 Average delay based on variety of clients Provider B

No.	Number of Client	Average Delay End to End (ms)
1	**1 Klien**	**28,317**
2	**2 Klien**	**28,895**
3	**3 Klien**	**32,913**

What can cause delay is packetization delay which is the time for the establishment of an IP packet which must go through the process of encapsulation through 4 TCP/IP layers, serialization delay which is the time required for the transmission of IP packets from the originating (the sender), queuing delay which is the processing time required in the router to handle the transmission of packets on the network and the propagation delay which is the time needed to transfer the information during transmission. There is also a delay on the live video streaming or interval between the actual event with live streaming which is caused by a coding process of multimedia information (video and audio) to form a signal, i.e. the time required by the server to send the stream to the software of the client, and the client takes time to buffer the data received so that it can be displayed on the media player on the client later. From the observed data, it is known that the more number of clients is, the longer of the delay will be. The bigger of the throughput value is, the smaller delay value is generated-- and vice versa. The smaller of the delay shows the better of the performance of a network, because of the time it needs to transfer a packet from the source (server) to the destination (client) the faster. Based on TIPHON (Telecommunications and Internet Protocol Harmonization Over Networks), the average value of delay in the test is categorized excellent, because the maximum delay value obtained during the test on both the provider is less than 150 ms.

4.4 Testing Jitter based on the variety of clients and provider

The scenario of Jitter test based on varietys of the client and the provider is carried out simultaneously with the sampling data of throughput, the delay and the packet loss using wireshark network analyzer software. Data of test results of jitter based on varietys of the client and the provider which has been averaged is shown in Table 9 and Table 10 as follows.

Table 9 Average jitter from Encoder to Media Server Provider A dan B

No.	Number of Client	Provider A	Provider B
		Average Jitter(ms)	Average Jitter(ms)
1	1 Klien	22.112	17.782
2	2 Klien	22.458	17.762
3	3 Klien	21.345	17.639

Table 10 Average jitter from Media Server to Client Provider A dan B

No.	Number of Client	Provider A	Provider B
		Average Jitter(ms)	Average Jitter(ms)
1	1 Klien	29.725	30.935
2	2 Klien	32.879	31.338
3	3 Klien	33.954	33.024

The greater the traffic load in the network can cause narrowing of the bandwidth and raises the density of the data queue that can cause the chances of collisions amongst the packets or start to create different delay among the packets or can be called a delay variety, so that the jitter value will be even bigger. From this observation it can be said that the bigger of the value of the delay is, the bigger of the value of the jitter is produced. Based on TIPHON (Telecommunications and Internet Protocol Harmonization Over Networks), the average value of the jitter in this test is categorized good,because the jitter value is less than 75 ms.

4.5 Testing packet loss based on variety of clients and provider

The test data packet loss based on the variety of the client and the provider whih is carried out simultaneously with the sampling of data throughput, delay and jitter that has been averaged is shown in Table 11 and Table 12 as follows.

Table 11 Average Packet Loss from Encoder to Media Server Provider A dan B

No	Number of Client	Provider A	Provider B
		Average Packet Loss	Average Packet Loss
1	1 Klien	0.282%	0.241%
2	2 Klien	0.257%	0.184%
3	3 Klien	0.230%	0.263%

Table 12 Average Packet Loss from Media Server to Klien Provider A dan B

No	Number of Client	Provider A	Provider B
		Average Packet Loss	Average Packet Loss
1	1 Klien	1.080%	0.062%
2	2 Klien	1.019%	0.720%
3	3 Klien	0.578%	0.541%

Packet loss can occur when the buffer capacity is full as a result of queuing and delay caused by the density of data traffic on the network, so that new data sent will not be received. Based on TIPHON (Telecommunications and Internet Protocol Harmonization Over Networks), average packet loss in this test is categorized good, due to packet loss value is less than 3%.

5. Conclusion

1. Time of test significantly affect the quality of network services used for live video streaming.
2. Throughput from the Media Server to Client based on the number of clients, it is concluded that the greater number of clients value is, the smaller the throughput is obtained with the highest throughput value of 467.750 Kbps when accessed by one client using the provider A. Throughput from Encoder to Media Server based on the number of clients, from the graphic looks unstable and highest throughput value is obtained at 483 000 Kbps when accessed by 3 clients using the provider B.
3. The Delay End to End based on the number of clients, the more number of clients accessing the live video streaming, the greater of the delay is. The lowest value of End to End Delay is 27.389 ms when accessed by one client using the provider B while the highest value obtained of Delay End to End is 36.180 2 ms when accessed by 2 clients using the provider A.
4. The Jitter from Media Server to the Client based on the number of clients, it is found that the more number of the clients, the bigger value of the jitter obtained with the lowest jitter value of 27.339 ms when accessed by one client using the provider A. The Jitter from Encoder to Media Server based on the number of clients, it is found that the more number of clients, the jitter value obtained at provider A look unstable and there is a decline at provider B with the lowest jitter value of 16.659 ms when accessed by 3 clients using the provider B.
5. The percentage of packet loss from Media Server to the Client based on the number of clients, it is found that the greater number of the clients, the percentage of packet loss acquired by provider A decreases while provider B look unsatble with the lowest percentage of packet loss is 0.007% when it is accessed by two clients using the provider A . The percentage of packet loss from the Encoder to Media Server based on the number of the clients, it is found that the more number of clients, the percentage of packet loss acquired by the provider A declines while provider B looks unstable with the lowest percentage of packet is 0.000% when accessed by clients using the provider B.
6. Based on the grouping of the delay, the jitter and the packet loss from Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) standard, the Evolved HSPA network on both of the providers used for live video streaming is categorized good.

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