Analysis of the Effect of Frame Variations on Improving the Quality of Streaming Network Services on the Local Network

Atthariq, Mursydah, Azhar
Department of Information Computer Technology, Politeknik Negeri Lhokseumawe, Lhokseumawe, 24301, Indonesia
E-mail: atthariq.huzaifah@pnl.ac.id, mursyidah@pnl.ac.id, azhar.tik@pnl.ac.id

Abstract. The quality of network services on streaming services is influenced by several aspects, one of which is the quality of the video used, the variation of frames and the right video format in streaming communication is expected to improve the quality of streaming services. In this study using IEEE 802.11n Wireless LAN devices as transmission media and 2-layer switches. The switch will classify incoming data packets at the input port based on access-list here incoming packets will be defined and marked according to the access-list attachment that will give priority so that higher-class packages will be prioritized first. This research uses HTTP protocol in streaming communication with H264 / MPEG-2 video format and frames 320x576 pixels, 352x576 pixels, 426x284 pixel, 544x576pixel and 704x576pixel. In this study using WireShark as a data analysis tool and service quality on streaming networks. From the test results obtained an average delay of 86.14484 ms for MPEG-2 video codec with a resolution of 426x284 pixels 25 fps. The jitter of 13.96518221 ms, packet loss of 15.22%.

1. Introduction

Wireless multimedia communication via Wireless Local Area Networks (WLAN) has received attention in recent years, this is due to the agreement of technologies such as Bluetooth, IEEE 802.11, 3G, and WiMAX which specifically emerged as a technology applicable to broadband wireless access because it supports real-time Multimedia conversation applications such as Voice and video conferencing. Local Area Networks Wireless Networks, providing high-level data services at a low cost because they work on the 2.4GHz frequency which is license-exempt to be used in the fields of education, industry, scientific, and medical.

The use of (WLAN) as a means of streaming communication has become a trend for now, the IEEE 802.11n-2009 network commonly abbreviated as IEEE 802.11n is a standard of wireless networks (wireless networking standard) which uses many antennas to improve performance. With IEEE 802.11n, bandwidth can be increased from 54 Mbit / s to 600 Mbit / s. Channel width also increased from the previous 20 MHz to 40 MHz. In IEEE 802.11n added support for MIMO (Multiple Input Multiple Output). and frame aggregation. Thus the use of the IEEE 802.11n network is very supportive of streaming services.

In building a video streaming service network over a network, we need to take into account the bandwidth requirements for the success of a video conference. Generally, there are two bandwidth requirements that need to be fulfilled, the first is the bandwidth requirement for sending picture/video signals and the second is the bandwidth requirement for sending audio/video signals. Between the two requirements above, the need for video sending bandwidth is very important because it will take up most
of the existing communication bandwidth. It is not surprising that techniques for data compression have become very strategic which allows telecommunications bandwidth savings.

The problem faced in building streaming services is the process of capturing and live to decode on the server-side. In addition to problems with the server, the biggest problem faced by this technology is bandwidth limitations. Streaming media applications that require a high enough bit rate cause network load to increase, causing the services provided to not run properly. The urgency of this research is to find out how much influence the frame rate in the network, with the available bandwidth.

2. Method

Noriaki Kamiyama [1] has conducted research related to streaming HD video on the internet. When transmitting the bit-rate generated is quite large, so that traffic on the network is too high and takes data traffic. Yung-Sung Huang [2] conducted a study related to video streaming services using webcam media on hospitals that were applied to patients and all-important medical data was transmitted using a 3G-wireless network communication system to various client devices.

There are two standards bodies responsible for locating video coding standards, the International Organization of Standards (ISO) and the International Telecommunications Union (ITU), which have developed a set of standards that have shaped the development of the visual communication industry. ISO JPEG, MPEG-1, MPEG-2, and MPEG-4 standards have the biggest impact: JPEG has become one of the most widely used formats for image storage and MPEG-2 forms the basis of video system formats on digital television and DVDs [4].

The ITU H.261 standard was originally developed for video conferencing via ISDN, but H.261 and H.263 are now widely used for real-time video communication over various networks including the Internet. H.264 from ITU-T is known as the International Standard video encoding, which is the latest standard in the H.261 video coding standard sequence [5]. Each international standard also has a common approach to meet these goals. Video coding standards explain the syntax for representing compressed video data [6].

2.1. MPEG-2

MPEG-2 is a determination for a group of coding and compression for audio and video, which is approved by MPEG and published as an international standard ISO / IEC 13818. MPEG-2 is usually used to encode audio and video for broadcast signals, including direct broadcast satellites and cable television. MPEG-2 with some modifications is also the coding format used in commercial DVD films. Using MPEG2 it is necessary to pay license fees to patent holders through the MPEG Licensing Association. The MPEG-2 standard is designed to provide the ability to compress, encode, and transmit multimedia signals on high-quality multichannel, satellite distribution, and broadband networks [7]. MPEG-2 consists of three main parts namely: video, audio (based on MPEG-1 audio coding).

2.2. H264

H.264 / AVC is the latest motion-compensation codec standard. This codec standard was developed by ITU-T Video Coding Expert Group (VCEG) together with ISO / IEC Moving Picture Expert Group (MPEG) and is a partnership effort known as the Joint Video Team (JVT). The ITU-T H.264 standard and the ISO / IEC MPEG-4 AVC standard (official ISO / IEC 14496-10 - MPEG-4 Part 10, Advanced Video Coding) are developed together so that they have identical technical content. H.264 is used in applications such as Blu-ray Discs, videos from YouTube and iTunes Store, DVB broadcasts, live broadcast satellite television services, cable television services, and real-time video conferencing.

The goal of the H.264 / AVC standard is to be able to provide good video quality at a price slightly lower than the previous standard (for example half or less the MPEG-2 H.263 bit rate, or MPEG-4 Part 2), without increasing the design complexity so much so it would be impractical or too expensive to implement.
Additional objectives are to provide flexibility to be applied to various applications on various networks and systems, including low and high bit rates, low and high-resolution video, broadcasting, DVD storage, RTP/IP packet networks, and multimedia ITU-T telephone systems.

The first H.264 / AVC standardization was completed in May 2003. The JVT then developed an extension to the original standard known as the Range Fidelity Extension (FRExt). This extension enables high-quality video by supporting precise coding with increased color sharpness and higher resolution, including the sampling structure known as Y’CbCr 04:02:02 (= YUV 04:02:02) and Y’CbCr 04:04:4. Some other features are also included in the Range Fidelity Extensions project, such as adaptive changing between 4×4 and 8×8 integers.

The name H.264 follows the ITU-T naming convention, where the standard is a member of the H.26x VCEG video coding standard; The MPEG-4 AVC name relates to the naming conventions in MPEG ISO / IEC, where the standard is part 10 of ISO / IEC 14496, which is a standard known as MPEG-4. [8]

The H.264 video format has a very wide range of applications which include all forms of digital video compression from low-bit streaming internet applications, to broadcast HDTV and Digital Cinema applications. By using H.264, save a bit rate of 50% or more. The quality of digital satellite TV can be achieved at 1.5 Mbit / s, compared to the current operating point of MPEG-2 video at around 3.5 Mbit / s. To ensure compatibility and adoption that is trouble-free, many standard bodies have changed or added H.264 / AVC to their video standards so users can use H.264 / AVC.

3. Experimental scenario

3.1. Network Testing Scenarios

In this study applying two different video codecs H.264 and MPEG-2. Video streaming files are transmitted using two different protocols namely HTTP and UDP. Transmission performance will be measured using the WireShark tool. In streaming network testing, several scenarios are performed which aim to analyze the performance of services using Wireshark software. This software will capture all data passing through and then analyzed by calculating the value of delay, jitter, packet loss, and throughput.

![Figure 1: The results of capturing streaming video from the VideoLan application using the Wireshark tool that analyzes packets on the UDP protocol.](image-url)

For the first scenario video streaming from server to client will be done with the VideoLan Client software with the H.264 video codec using the HTTP protocol. Testing in this scenario uses different video resolutions with frame number 25. Then proceed with the second scenario testing using the UDP protocol, the same video file format as the first scenario, the third scenario testing using the same method with the first scenario only format the file is replaced by the MPEG-2 codec and the fourth scenario tests...
using the UDP protocol and video format the same as the third scenario. Then Wireshark that has been installed on the client will be used to capture packets that pass through the network, namely UDP packages. Figure 1 shows the results of the Wireshark capturing tool.

3.2. Measurement and Analysis
In this section analysis will be made of the performance of the video streaming network, this analysis will be carried out when streaming video from the server to the client. The analysis process is carried out on the QoS parameters, namely delay, military, packet loss and throughput obtained from each scenario. For discussion of the scenario has been explained in the previous section. Data retrieval is done by using two files with different formats, namely H.264 and MPEG-2 with different file sizes.

3.2.1. Delay Measurement and Analysis
Delay is the time taken by the packet from the sender to the receiver. Delay is a parameter needed to determine the performance of a streaming video network quality. Based on the ITU-T G.1010 standard for good video streaming quality, the delay must be <150ms.

This test will stream video from the server to the client with the VideoLan Client software. Packets that pass on the network will be captured from the client-side using Wireshark software. The results of the measurement of the delay parameters can be shown in Tables 3.1 and 3.2

| Video format and resolution | Protocols  | Protocols  |
|-----------------------------|------------|------------|
|                             | HTTP       | UDP        |
| H.264 320x576 pixel 25 fps  | 86.3545    | 29.4403    |
| H.264 352x576 pixel 25 fps  | 84.1125    | 29.4307    |
| H.264 426x284 pixel 25 fps  | 86.8876    | 29.5205    |
| H.264 544x576 pixel 25 fps  | 86.8741    | 30.9401    |
| H.264 704x576 pixel 25 fps  | 86.4955    | 30.7741    |

| Video format and resolution | Protocols  | Protocols  |
|-----------------------------|------------|------------|
|                             | HTTP       | UDP        |
| MPEG-2 320x576 pixel 25 fps | 86.4686    | 29.5762    |
| MPEG-2 352x576 pixel 25 fps | 90.1779    | 30.3425    |
| MPEG-2 426x284 pixel 25 fps | 98.9255    | 39.4704    |
| MPEG-2 544x576 pixel 25 fps | 95.1976    | 61.0027    |
| MPEG-2 704x576 pixel 25 fps | 85.4732    | 65.6233    |

From the measurement results show the difference in the value of delay for each of the data in the test delay generated from the results of the H.264 video code data testing with testing on the HTTP and UDP protocols can be seen the results of an average delay difference of 64.96% between these protocols. The increase in the delay value occurs in the HTTP protocol. The biggest increase in delay value is owned by the test data with MPEG-2 video codec at a resolution of 426x284 pixels 25 fps. The magnitude of the measured delay is due to the small number of packets that can reach the client within the timeframe of sending video streaming packets. When there is communication there is a disruption so there is a delay in the arrival of packets. An increase in the value of the delay can also occur due to the increasing number of packets that come. The increase in the value of this delay results in a decrease in the value of throughput in accordance with the analysis of the average throughput can be seen in table 3.8.
Based on Figure 2 it can be seen that each delay has changed, the largest delay value occurs in the testing of the HTTP protocol with MPEG-2 test data resolution of 426x284pixel 25 fps, which is equal to 98.9255ms file size that is streamed at 15.7 MB. The delay value from the implementation results is still within the tolerance limits according to the International Telecommunication Union (ITUT G.1010) recommendation [3] for video streaming services. Therefore, the delay value from the implementation is still acceptable for video streaming services.

3.2.2. **Jitter Measurement and Analysis**

Jitter is a variation of the time of arrival of each package. Jitter can be measured from the time between the packet received now and the package received previously. Jitter is caused by different trajectories between packages, variations in queue length, and data processing times. Telecommunication Union (ITUT G.1010) [3] recommending a good jitter is <30 ms. At the time of the trial, the measured jitter is the average jitter (jitter) of the jitter of all video packages captured by the Wireshark tool. In the measurement of jitter also performed the same scenario with the measurement of delay. The following is an example of jitter calculation and the results in table Table 3.3 and Table 3.4.

**Figure 2:** (a) Delay Measurement Results for H.264 codec with different video resolutions
(b) Delay Measurement Results for MPEG-2 codec with different video resolutions
Average jitter = \frac{\text{total delay variation}}{\text{total package received} - 1}
= \frac{176173.210155.82286}{12404 - 1}
= 13.96518221 ms

In Table 3.3 and Table 3.4 there is an increase in the value of jitter 14.8361 ms in the test data with MPEG-2 video codec resolution of 426x284 pixels 25 fps. The increase in the value of jitter for files measuring 15.7 MB. Large value of jitter is influenced by the delay that occurs in the packet while in the router can be seen in Table 3.2. In testing with this test data the result is smaller than the value of 30 ms which is a good jitter standard. The results of this experiment stated the value of jitter still met the standard for Quality of Service. This happens because of the buffer used by the real-time video streaming application. Therefore, jitter does not greatly affect the video that is run by real-time video streaming applications.

(a)
Based on Figure 3 it can be seen that the jitter value of each time changes. The size of jitter is in the range of 4,706 ms to 14,836 ms. The value of jitter is caused by buffers used by real-time video streaming applications and the influence of unstable wireless-based network conditions. Wireless speed and signal strength from access points can change at any time.

3.2.3. Measurement and Analysis of Packet Loss

Packet loss aims to determine the amount of packet loss when streaming video from source address to destination address. The greater packet loss causes the quality of the video received. The results of the measurement of packet loss parameters are shown in table 3.5 and table 3.6. Give an example of the calculation of test results using the UDP protocol with H.264 video format 320x576 pixel 25 fps resolution.

\[
Packet Loss = \frac{(Packets\ sent - packets\ received)}{packets\ received} \times 100\%
\]

\[
Packet Loss = \frac{(14556 - 14376)}{14556} \times 100\% = 1.236603462489695\%
\]

| Video format and resolution | Protocols |
|-----------------------------|-----------|
|                            | HTTP | UDP   |
| H.264 320x576 pixel 25 fps  | 0    | 1.23  |
| H.264 352x576 pixel 25 fps  | 0    | 1.35  |
| H.264 426x284 pixel 25 fps  | 0    | 1.36  |
| H.264 544x576 pixel 25 fps  | 0    | 1.46  |
| H.264 704x576 pixel 25 fps  | 0    | 1.41  |
Table 3.6: Jitter Measurement Results for MPEG-4 codecs

| Video format and resolution | Protocols | HTTP | UDP |
|-----------------------------|-----------|------|-----|
| MPEG-4 320x576 pixel 25 fps | 0         | 1.38 |
| MPEG-4 352x576 pixel 25 fps | 0         | 1.45 |
| MPEG-4 426x284 pixel 25 fps | 0         | 1.3  |
| MPEG-4 544x576 pixel 25 fps | 0         | 0.61 |
| MPEG-4 704x576 pixel 25 fps | 0         | 0.5  |

Figure 4 can explain that the average packet loss when the system is streaming video ranges from 0.0% to 1.46%, where the packet loss is still within the tolerance limit. Packet loss below 10% is still permitted. Packet loss that occurs is caused at the time of data retrieval, as well as existing traffic conditions. From Figure 4 it can be seen that the large packet loss is owned by the H.264 video codec test data with a resolution of 544x576 pixels 25 fps with UDP protocol at table 3.5. The number of packets that are lost on network streaming results in a smaller measurable throughput of data seen in table 3.6. A large packet loss rate can reduce the value of throughput. A large packet loss rate in real-time video streaming applications results in certain parts of the video being interrupted. If there are a lot of broken parts, then the information will also be reduced.
3.2.4. **Measurement and Analysis of Throughput**

Throughput is measured based on packet transmission speed. The experiment was conducted several times to find the average (average) throughput. In the trial of measuring throughput for video transmission from server to client on this network, the average value of throughput is obtained. The throughput values are shown in Table 3.7 and Table 3.8. Data throughput in this study is taken from the packet capture with the Wireshark sniffing tool, which can be seen in Figure 5.

| Video format and resolution | Protocols | Protocols |
|-----------------------------|-----------|-----------|
| H.264 320x576 pixel 25 fps  | 0.858     | 0.834     |
| H.264 352x576 pixel 25 fps  | 0.844     | 0.791     |
| H.264 426x284 pixel 25 fps  | 0.855     | 0.826     |
| H.264 544x576 pixel 25 fps  | 0.858     | 0.771     |
| H.264 704x576 pixel 25 fps  | 0.845     | 0.806     |

**Table 3.8:** Throughput Measurement Results for the MPEG-2 codec

| Video format and resolution | Protocols | Protocols |
|-----------------------------|-----------|-----------|
| MPEG-2 320x576 pixel 25 fps | 1.159     | 0.82      |
| MPEG-2 352x576 pixel 25 fps | 1.281     | 0.756     |
| MPEG-2 426x284 pixel 25 fps | 0.817     | 0.834     |
| MPEG-2 544x576 pixel 25 fps | 1.763     | 1.798     |
| MPEG-2 704x576 pixel 25 fps | 2.251     | 2.271     |

In tables 3.7 and table 3.8 it can be seen that the comparison of throughput on the results of all tests. Based on Figure 6table from the measurement results of two file formats namely H.264 and MPEG-2 that the smallest throughput occurs in the test file with MPEG-2 format resolution of 352x576 pixels 25fps in table 2.8, which is 0.756Mbps, while the largest throughput occurs in the MPEG-2 test file with a resolution of 704x576 pixel 25 fps which is 2,271 Mbps. The size of the data packet received is due to different file sizes. This is in accordance with the understanding of throughput, namely the greater the data sent, the greater the data received.
Figure 5: (a) Results of Measurement of Throughput for codec H.264 (b) Results of Measurement of Throughput for codec MPEG-2

| Traffic | Captured | Displayed | Marked |
|---------|----------|-----------|--------|
| Packets | 19636    | 19636     | 0      |
| Between first and last packet | 173.210 sec |
| Avg. packets/sec | 113.365 |
| Avg. packet size | 946.324 bytes |
| Bytes | 18582026 |
| Avg. bytes/sec | 107280.045 |
| Avg. MBit/sec | 0.858 |

Figure 6: Wireshark summary to see the average packet size streamed, average streaming speed and bandwidth used by H.264 544x576 pixel 25 fps data with HTTP protocol

4 Conclusion

From the test results obtained an average delay of 86.14484 ms for the MPEG-2 video codec with a resolution of 426x284 pixels 25 fps. The jitter of 13.96518221 ms, packet loss of 15.22%. The results of the experiment show that MPEG-2 jitter time is less than H.264. So the MPEG-2 protocol is better than H.264 through the UDP protocol. In contrast, H.264 jitter time is less than MPEG-2 via the HTTP protocol. So H.264 is better than MPEG-2 via the HTTP protocol. But in terms of appearance H.264 is better than MPEG-2. H.264 image quality is also better and smoother playback than with MPEG-2 basic compression. But the most interesting feature is the lower bit-rate needed for network transmission.

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