

Mabrur Roh Bintang Jaya¹
Faculty of Information Technology,
Universitas Teknologi Digital Indonesia
Yogyakarta,
Yogyakarta, Indonesia
email: bintang.dev@gmail.com

Widyastuti Andriyani
Faculty of Information Technology,
Universitas Teknologi Digital Indonesia
Yogyakarta,
Yogyakarta, Indonesia
email: widya@utdi.ac.id

Dommy Kristomo
Faculty of Information Technology
Universitas Teknologi Digital Indonesia,
Yogyakarta, Indonesia
email: domy@utdi.ac.id

Muhammad Agung Nugroho
Faculty of Information Technology
Universitas Teknologi Digital Indonesia,
Yogyakarta, Indonesia
email: m.agung.n@utdi.ac.id

Dynamic Bitrate Adjustment in Web-based Video Streaming Applications Using HTTP Live Streaming (HLS)

This research aims to implement Adaptive Bit Rate (ABR) in the web-based video streaming application JBTv using HTTP Live Streaming (HLS). ABR is a technique that enables automatic adjustment of video bitrate according to user network conditions, while HLS is a streaming protocol that supports adaptive streaming based on HTTP. The research methodology encompasses requirements analysis, system design, implementation, and evaluation. During the requirements analysis phase, the identification of JBTv application requirements and the features needed to implement ABR with HLS were conducted. System design involves the selection of suitable ABR algorithms and the architecture design of the JBTv application that supports HLS. Implementation is carried out by developing the JBTv application capable of generating variant streams with various bitrates and performing adaptive playback according to network conditions.

KeyWords: Adaptive Bit Rate, ABR, HTTP Live Streaming, HLS, JBTv, video streaming application

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1 Introduction

Education has undergone a significant paradigm shift in line with technological advancements. Technology, especially in the increasingly advanced digital era, has shown rapid development. One area that illustrates this progress is learning methods, where distance learning has become an essential need in the field of education [1,2]. The support of information and communication technology has opened new opportunities to form new ways of teaching and learning. Emerging platforms and systems facilitate online learning, providing various alternatives to meet learning needs, especially in the context of a global pandemic that limits physical interaction. One significant innovation supporting the development of distance learning is Jogja Belajar TV (JBTv). JBTv is an interactive television system initiated by the Balai Tekkomdik DIY. Through JBTv, students can access digital learning content via gadgets or laptops. The main advantage of JBTv is its ability to provide controlled live streaming, managed by the Jogja Belajar management team. This allows students to engage in learning in an interactive and enjoyable manner. However, like many technological implementations, JBTv also faces some challenges that need to be addressed. One of the main challenges is the limitation in real-time video streaming. In network environments with limited capacity, JBTv users may encounter issues such as prolonged buffering, connection drops, or low video quality. These

issues can disrupt the learning process and potentially damage the user experience. To address this challenge, a solution is needed to maintain the quality of live video streaming services on JBTv, especially when users operate on networks with limited bandwidth. One technological solution that can be applied is Adaptive Bit Rate (ABR) using HTTP Live Streaming (HLS) [1,3]. ABR allows applications to automatically adjust the quality of the video being played according to the user's network conditions. On the other hand, HLS is a streaming protocol that supports adaptive streaming through the HTTP protocol [4,6,8].

By implementing ABR with HLS on JBTv, it is expected that the quality of live video streaming services can be significantly improved. Thus, users will have a more satisfying experience with optimal video quality, even in situations with limited bandwidth. This step will positively impact the distance learning process, allowing students to access learning content smoothly and without interruptions [6,7,9]. Therefore, this research aims to implement ABR with HLS on the JBTv web-based video streaming application. By optimizing video playback according to user network conditions, this research is expected to enhance the quality of JBTv services in supporting distance learning. This research will involve requirements analysis, system design, implementation, and testing to integrate ABR with HLS on JBTv [9–12]. Thus, it is expected that this research will provide an effective solution for maintaining the quality of live video streaming services on the JBTv application, particularly in network situations with bandwidth limitations. The results of this research are expected to make JBTv a reliable and effective tool for delivering learning content to students, especially in the context of distance learning, which remains an urgent need in the current era. Consequently, education can continue to evolve and adapt to the ongoing technological transformation.

2 Theory

2.1 Adaptive Bitrate Streaming (ABR). Adaptive Bitrate Streaming (ABR) is a technique used in video streaming where the quality of the video is dynamically adjusted based on the current network conditions. The primary goal of ABR is to provide a continuous and smooth playback experience by minimizing buffering

¹Corresponding Author.

and interruptions [9–12]. ABR achieves this by offering multiple streams of the same video content encoded at different bitrates. The client device selects the appropriate stream to match the available network bandwidth and device capabilities. The concept of ABR was introduced to tackle the variability in internet speeds and network congestion, which are common challenges in delivering high-quality video over the internet. Traditional fixed bitrate streaming methods often lead to buffering or reduced video quality when the network conditions deteriorate. ABR provides a more flexible approach, allowing the video quality to degrade gracefully rather than abruptly stopping playback.

2.2 HTTP Live Streaming (HLS). HTTP Live Streaming (HLS) is a protocol developed by Apple for delivering media content over the internet. HLS is widely adopted due to its compatibility with various devices and platforms, including web browsers, mobile devices, and smart TVs. HLS works by breaking down the video content into smaller, discrete segments, typically a few seconds long. Each segment is available in multiple bitrate versions, allowing the client to switch between different quality levels in real time based on network conditions [9–12].

- a. Segmented Media Files: The video content is divided into small segments, each encoded at different bitrates.
- b. Master Playlist (Manifest): A playlist file that lists the available media files and their corresponding bitrates.
- c. Media Playlist: A playlist file that lists the media segments for a specific bitrate.

HLS supports live streaming as well as video on demand (VOD) and provides mechanisms for adaptive bitrate switching, allowing a smooth transition between different quality levels.

2.3 MPEG-DASH (Dynamic Adaptive Streaming over HTTP). MPEG-DASH is another popular standard for adaptive bitrate streaming. It functions similarly to HLS but is an open standard, meaning it is not tied to any specific company or ecosystem. MPEG-DASH provides more flexibility in terms of codec support and can deliver high-quality streaming across a wide range of devices and platforms. Like HLS, MPEG-DASH divides the video into segments, and each segment can have multiple quality levels. The client dynamically selects the appropriate segment based on real-time network performance and playback requirements [13–17].

2.4 Quality of Service (QoS) and Quality of Experience (QoE). Quality of Service (QoS) refers to network performance metrics like bandwidth and latency, while Quality of Experience (QoE) gauges user satisfaction with video quality and playback [16,18–20].

- a. Quality of Service (QoS) refers to the performance characteristics of a network, such as bandwidth, latency, jitter, and packet loss, which directly affect the delivery of video streams.
- b. Quality of Experience (QoE) focuses on the end-user's perception of the video quality. ABR streaming aims to enhance QoE by adapting the video quality to the network conditions, thereby ensuring a satisfactory viewing experience even under fluctuating network scenarios.

2.5 Bandwidth Estimation Techniques. Accurate bandwidth estimation is crucial for the effective implementation of ABR. Various techniques are used to estimate available network bandwidth.

- a. Throughput-based estimation: Measures the rate at which data is being successfully transferred.
- b. Buffer-based estimation: Monitors the state of the playback buffer to infer available bandwidth.
- c. Hybrid methods: Combine multiple approaches to achieve more accurate and reliable bandwidth estimation.

2.6 Scalability and Content Delivery Networks (CDNs). Scalability is a critical factor for video streaming services, especially for handling large audiences. Content Delivery Networks (CDNs) play a vital role in distributing video content across geographically distributed servers, reducing latency, and ensuring consistent video quality. CDNs facilitate the delivery of adaptive bitrate streams by caching multiple versions of video segments, allowing faster access and smoother playback for users [21].

2.7 Challenges and Considerations in ABR Implementation. Implementing ABR involves addressing various challenges, such as:

- a. Latency: Minimizing the delay between the actual event and its display on the user's device.
- b. Buffering: Managing the trade-off between buffering time and playback quality.
- c. Encoding Overhead: Balancing the computational cost of encoding multiple bitrate versions of the same video.
- d. Network Variability: Adapting to rapidly changing network conditions without compromising user experience.

2.8 Challenges Faced by JBTv.

- a. Bandwidth Limitations, One of the main challenges faced by JBTv is the limitation in real-time video streaming. JBTv users often experience prolonged buffering, disconnections, or low video quality especially in network conditions with limited capacity.
- b. Network Variability, Unstable and variable network conditions are a significant challenge because they can cause sudden and unexpected drops in video streaming quality.
- c. User Experience (QoE), Another challenge is maintaining a high quality of user experience (Quality of Experience, QoE). Users often feel dissatisfied if they experience frequent buffering or significant drops in video quality during a streaming session.
- d. Difficulty in Implementing New Technologies, Implementing new technologies such as Adaptive Bit Rate (ABR) with HTTP Live Streaming (HLS) requires extensive customization and testing to ensure that the system can operate smoothly in various network conditions.
- e. Efficient Data Traffic Management, Efficiently managing streaming data traffic is another important challenge to ensure that video can be delivered at optimal quality without causing excessive load on the network.

2.9 Specific Aspects of Video Streaming Quality to be Addressed.

- a. Dynamic Bitrate Adjustment, ABR is used to automatically adjust the video bitrate based on the user's network conditions. This aims to reduce buffering and ensure a smoother and uninterrupted viewing experience.
- b. Multi-Device Compatibility, Implementing HLS that supports adaptive streaming and is compatible with various devices such as web browsers, mobile devices, and smart TVs, thus ensuring that users can enjoy video content on various platforms.
- c. Buffering Reduction, By using ABR and HLS, JBTv strives to reduce buffering time and maintain smooth video playback even on networks with limited bandwidth.
- d. Latency Reduction, Reducing latency is one of the main focuses to ensure that the delay between real events and display on the user's device can be minimized.
- e. Real-Time Video Quality Management, With ABR, video quality can be managed in real-time to adapt to changing network conditions, ensuring that quality degradation occurs gradually and does not surprise users.

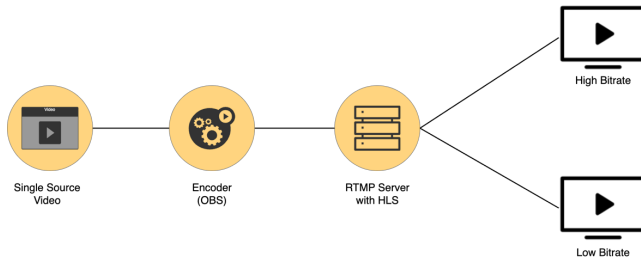


Fig. 1 Modeling of RTMP Server with HLS

- f. Bandwidth Usage Optimization, Using accurate bandwidth estimation techniques to ensure that the video displayed matches the user's network capacity, thus optimizing bandwidth usage and preventing excessive data usage.

By addressing these challenges, JBTV is expected to improve the quality of video streaming services, providing a more effective and enjoyable distance learning experience for its users.

3 Method

3.1 Analysis. JBTV uses the RTMP (Real-Time Messaging Protocol) protocol as the main streaming medium, as seen on Figure 1. However, to improve the quality and user experience, JBTV also implements HLS (HTTP Live Streaming) as an adaptive streaming method. When a user watches a video stream on JBTV, the streaming server will receive the request and use HLS to serve the video according to the user's network and device conditions.

3.2 Modeling RTMP Server with HLS. In the system built, the video source comes from a video test with mp4 format. OBS is used as an encoder, which then sends to a streaming video server that uses RTMP and HLS to manage and deploy video content in real-time and adaptively to connected users according to each user's bandwidth capabilities. Ubuntu Server 18.04 acts as the main platform to run all these components efficiently.

3.3 Experimental Setup, Sample Size, and Statistical Methods.

3.3.1 Experimental Setup. The experiment was conducted with the aim of observing how the implementation of HTTP Live Streaming (HLS) affects the quality and user experience under various network and bandwidth conditions. There are two test scenarios designed to illustrate various possible situations: the first scenario is conducted without HLS and without bandwidth throttling, while the second scenario is conducted with bandwidth throttling at 900 kbps. In both scenarios, the bitrates used are 600 kbps (for 720p video) and 450 kbps (for 240p video). The live stream URL address used for testing is <https://live.devjbtv.jogjabelajar.org/full/test1.m3u8>. This test was conducted using an online tool, Bitmovin, which measures video streaming performance.

3.3.2 Sample Size. The sample size in this test is not explicitly stated in the uploaded document. However, this test involves several video streaming sessions to observe performance under various bandwidth conditions. Samples in this context refer to video streaming sessions tested under varying network conditions.

3.3.3 Statistical Methods. The statistical methods used in this experiment are not specifically described in the documentation. However, observations were made to compare various metrics such

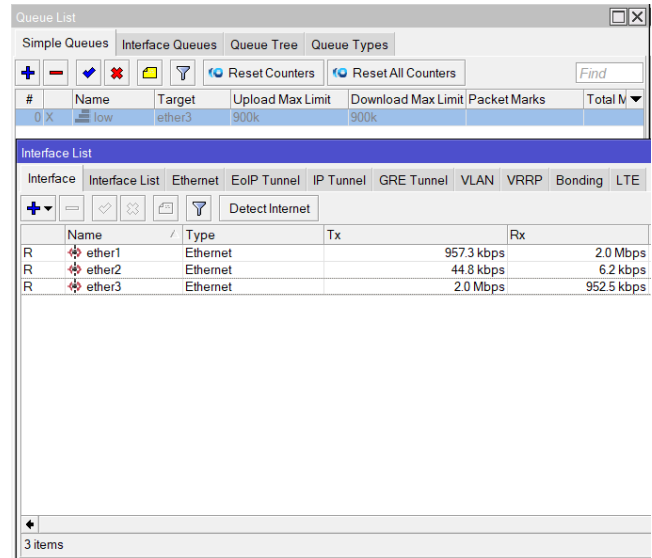


Fig. 2 Before limitation on proxy

as bitrate, buffering level, and video quality before and after bandwidth throttling was applied and removed. These measurements provide a basis for analyzing changes in streaming performance and their impact on user experience, allowing for interpretations that can be used to assess the effectiveness of HLS and ABR under varying network conditions.

The experimental setup was designed to evaluate the effectiveness of HLS in optimizing video streaming quality according to the existing network conditions, with a focus on the adaptability of HLS in the face of bandwidth throttling and network variability.

4 Results and Discussion

4.1 Testing. The test aims to observe how the implementation of HTTP Live Streaming (HLS) affects the quality and user experience under various network and bandwidth conditions. In order to test the effect of HLS on video streaming applications, we designed two different test scenarios to illustrate various situations that might occur, the bitrate setup in HLS is 600 kbps (720p) and 450 kbps (240p), hls will detect the client's internet speed to determine the video quality to be provided. if the client's internet speed can reach 600kbps, then 720p video will be provided. but if the internet speed is less than 600kbps, then 240p video quality will be provided. This test is carried out using online tools, namely Bitmovin then what is tested is the live stream url, in this case it is <https://live.dev-jbtv.jogjabelajar.org/full/test1.m3u8>.

4.1.1 Without HLS. In this scenario, testing is done without applying HTTP Live Streaming (HLS) to the video streaming application. The purpose of this test is to observe how the video streaming application will behave in the absence of HLS.

Sufficient Bandwidth / Before bandwidth limitation on proxy. Testing is done without applying HLS and without bandwidth limitations on Mikrotik. This means that video streaming will be done at the available bitrate speed without restrictions.

It can be seen in Figure 2 that the user's bandwidth reaches 2 Mbps and the recorded bitrate runs at 0.6 Mbps, where the bandwidth is sufficient or more than 600 Kbps, a video with 720p quality will be run.

In the buffer levels in Figure 3, there are no buffers that reach 0 seconds in a row so that the user experience does not experience buffering on the video.

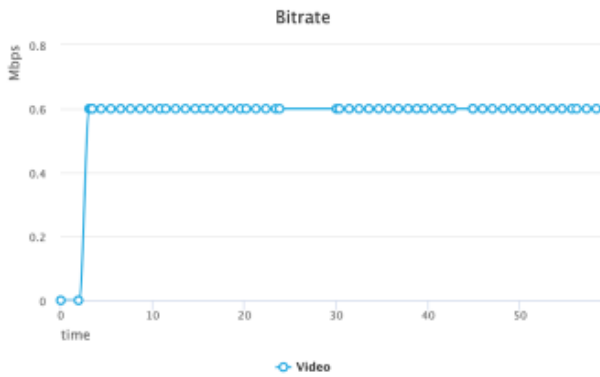


Fig. 3 Bitrate before limitation on proxy

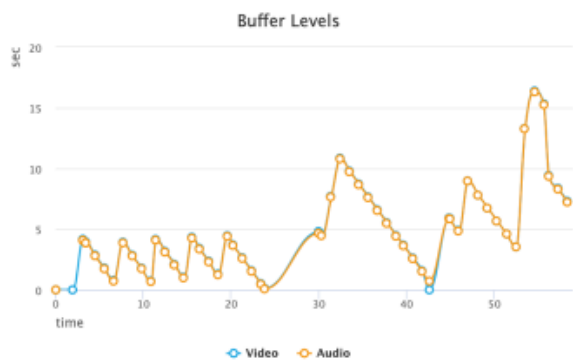


Fig. 4 Buffer Levels Before limiting on the proxy



Fig. 5 Video Streaming Before limitation on proxy

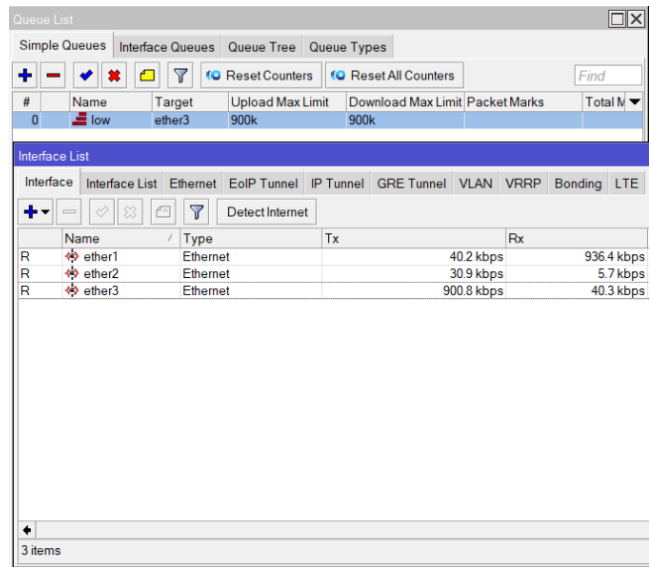


Fig. 6 Performed limitation on proxy

Then in conditions without HLS with no Bandwidth restrictions, the video can run smoothly without any buffering and interference as shown in Figure 4.

Low Bandwidth / Bandwidth limitation on proxy. Tests were carried out without using HTTP Live Streaming (HLS) and with bandwidth limitation applied to 900 kbps. The purpose of this test is to observe how the video streaming will behave in the absence of HLS under low bandwidth conditions.

When a bandwidth limitation of 900 Kbps is performed on the proxy, the bitrate is recorded to remain at 0.6 MBps.

Video streaming experiences problems in loading and maintaining optimal video quality. Because the applied bandwidth limitation is not sufficient to load video with 720p quality, the use of this bitrate causes the video to buffer repeatedly with the buffer level reaching point 0 seconds in a row (time > 100) as seen in Figure 8.

So in conditions without HLS, the video streaming experience becomes less smooth, visible buffering when the video is running as shown in Figure 9.

4.1.2 With Hls Adaptive Bitrate. In this scenario, testing is done by applying HTTP Live Streaming (HLS) to the video streaming application. The purpose of this test is to observe how the video streaming application will behave in the presence of HLS.

Before limitation on proxy. Testing using HLS with sufficient bandwidth conditions without any limitation on the proxy. This aims to see the performance of HLS in optimal conditions.

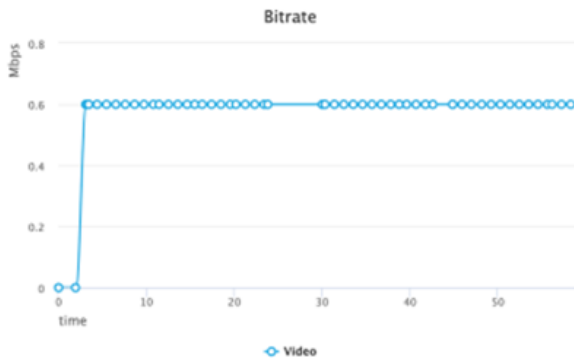


Fig. 7 Bitrate Performed Bandwidth limitation on proxy



Fig. 8 Buffer Level Performed Bandwidth limitation on proxy



Fig. 9 Video Streaming Performed Bandwidth limitation on proxy

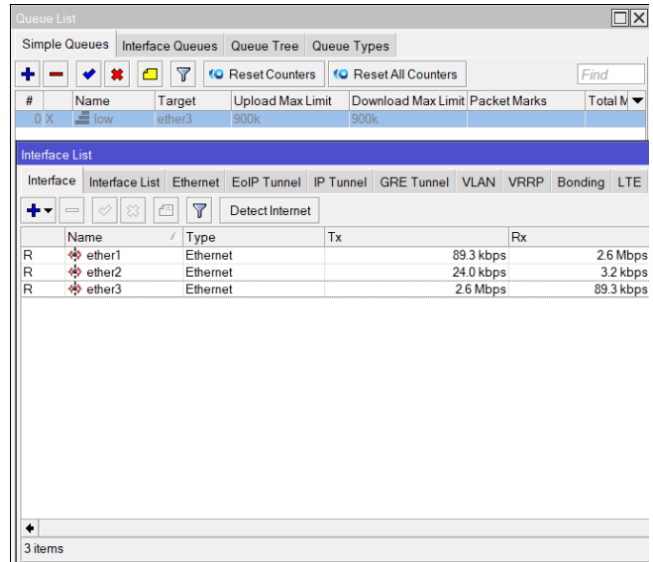


Fig. 10 Before limitation on the proxy

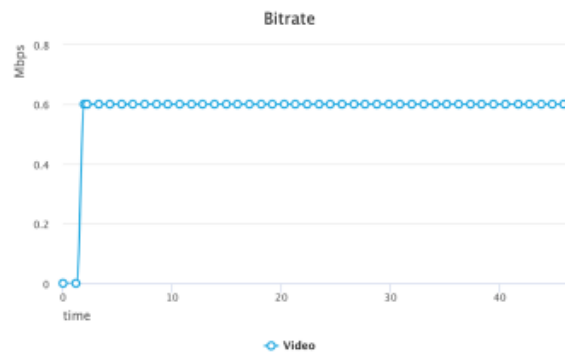


Fig. 11 Bitrate before limitation on proxy

Users have an internet connection without any bandwidth limitation on the proxy, a bandwidth of 2.6 Mbps is recorded as shown in Figure 10 then the bitrate runs at 0.6 Mbps

Video streaming starts with bitrate quality according to good network conditions and sufficient bandwidth. buffer level does not reach 0 seconds in a row as shown in Figure 11. With sufficient bandwidth availability without any limitation on the proxy, HLS can respond by providing optimal video streaming quality according to network conditions and user devices.

Testing by implementing HLS with bandwidth limitation on the proxy. Bandwidth limitations will affect the server's ability to transmit video with optimal quality. This test aims to observe how HLS responds when there is limitation on the HLS side. Bandwidth limitation is applied at 900 Kbps as shown in Figure 14, the streaming video starts with a bitrate quality of about 0.6 Mbps (720p), after a while the bitrate drops to 0.4 Mbps.

Since the bandwidth limitation is only 900 kbps, after a while the video will buffer because the internet speed is not enough to load the video with 720p quality. HLS detects that the available bandwidth does not meet the bitrate requirement of 0.6 Mbps (720p). In response, HLS will downgrade the video quality to 240p to accommodate the bandwidth limitation. The video bitrate will be reduced from 0.6 Mbps to 0.45 Mbps, and the video will continue to play at 240p to avoid excessive buffering.

Thus, users can still enjoy video content even with lower quality, but still avoid buffering interruptions that can reduce the user experience.

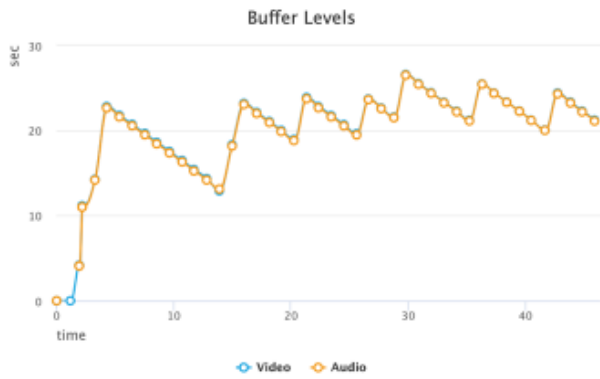


Fig. 12 Buffer Levels Before limitation on the proxy

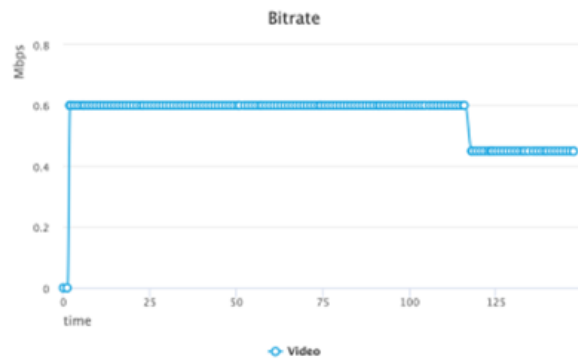


Fig. 15 Bitrate when limiting on proxy



Fig. 13 Video Streaming Before limitation on proxy

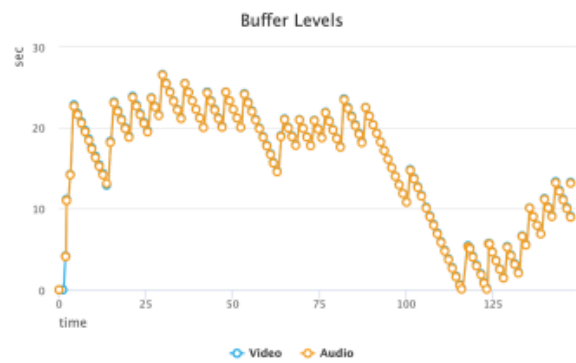


Fig. 16 Buffer Levels when limiting on proxy

Queue List						
Simple Queues						
#	Name	Target	Upload Max Limit	Download Max Limit	Packet Marks	Total M
0	low	ether3	900k	900k		

Interface List									
Interface	Interface List	Ethernet	EoIP Tunnel	IP Tunnel	GRE Tunnel	VLAN	VRRP	Bonding	LTE
Name	Type	Tx	Rx						
R ether1	Ethernet	41.8 kbps	846.4 kbps						
R ether2	Ethernet	41.3 kbps	8.8 kbps						
R ether3	Ethernet	880.6 kbps	41.8 kbps						

Fig. 14 Performed limitation on proxy



Fig. 17 Video streaming Performed limitation on proxy

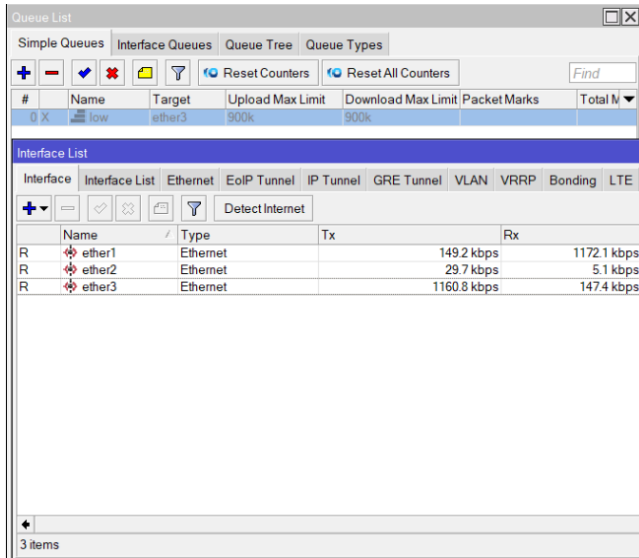


Fig. 18 After the limitation on the proxy is disabled

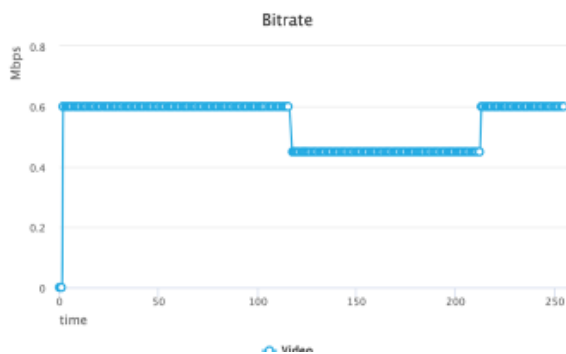


Fig. 19 Bitrate after the limitation on proxy is disabled

This test disables bandwidth limitation on the proxy when there is a restriction on the proxy with a limit of 900 Kbps then this limitation is disabled so that the normal bandwidth returns with a speed of 2 Mbps. The purpose of this test is to see how HLS operates in conditions without bandwidth restrictions.

The bandwidth limitation on the proxy is disabled after previously there was a limitation of 900 kbps, it was noted that the bandwidth began to return at a speed of 1160.8 Kbps. then the streaming video starts with a bitrate quality of around 0.6 Mbps conditions without bandwidth restrictions, then drops to 0.4 Mbps during bandwidth restrictions and then rises back to 0.6 Mbps when the restrictions on Mikrotik are disabled.

HLS will detect a higher bandwidth availability. In response, the video quality will be increased to 720p (0.6 Mbps) This allows users to enjoy higher resolution video content without buffering, it was noted in the buffer level that no buffer reached 0 seconds in a row when the bandwidth limitation increased or time > 200 as shown in Figure 20.

Thus, users can still enjoy video content with 720 p quality to provide a better and higher quality streaming experience.

5 Result

In this study, the following results can be drawn as conclusions:

The implementation of Adaptive Bit Rate (ABR) using HTTP Live Streaming (HLS) on the JBTV web-based video streaming application was successfully carried out using the Docker configuration on the Ubuntu 18.04 server. The configuration steps

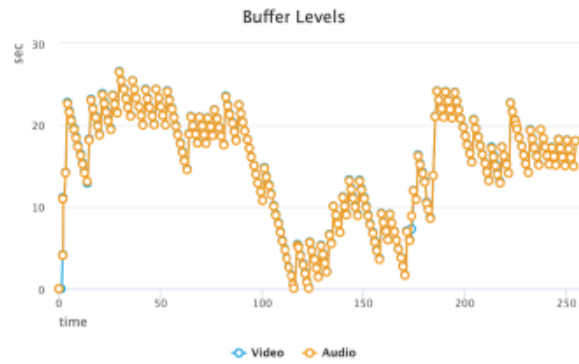


Fig. 20 Buffer Levels After the limitation on proxy is disabled



Fig. 21 Video Streaming After the limitation on proxy is disabled

involved installing the required software such as Nginx, libnginx-mod-tmp, and FFMpeg. Furthermore, the process also involved the creation of a Dockerfile and docker-compose.yml containing the rtmp-hls:1.14-rc1 image to facilitate the image creation and sharing process. Nginx configuration was also implemented using nginx.conf and default files. After all configuration and file creation steps are complete, the image build process is performed using the docker-compose command, followed by running the RTMP container on the server. The end result of these steps is a container named rtmp-hls that operates over ports 80 and 1935.

In the developed system, the source video comes from a test video with mp4 format. OBS is used as an encoder, which further sends the video to the streaming video server via RTMP and HLS protocols. This allows adaptive and real-time management and distribution of video content to connected users, according to the bandwidth capacity of each user.

Through the tests conducted, it is evident that the implementation of HLS with ABR on JBTV's RTMP server can improve the quality of live video streaming services. Under bandwidth limitation conditions, the video quality automatically adjusts by lowering the resolution to 240p. However, when bandwidth limitation is disabled, the video quality can increase up to 720p. This confirms the ability of the HLS implementation to perform well in the ABR scenario.

6 Discussion : Alignment and Contrast with Previous Research

6.1 Alignment with Previous Research. Implementation of Adaptive Bit Rate (ABR) and HTTP Live Streaming (HLS): This study is in line with many previous studies that show that implementing ABR with HLS can significantly improve video streaming quality under varying network conditions. For example, a study by Waheed et al. (2021) showed that ABR can effectively adjust video

bitrate based on user network conditions to reduce buffering and maintain optimal video quality. This study supports these findings with results showing that implementing ABR on JBTv allows automatic adjustment of video quality from 720p to 240p depending on bandwidth availability, which is in line with the main goal of ABR to provide a smooth viewing experience without buffering interruptions.

Improved Quality of Experience (QoE): This study is also consistent with the findings of Fiedler, Hossfeld, and Tran-Gia (2010) who emphasized the importance of QoE in video streaming services. The results of this study show that users can enjoy higher quality video content without annoying buffering, even when there is bandwidth limitation. This shows that the use of ABR and HLS in JBTv successfully improves QoE by dynamically adjusting video quality according to user network conditions.

Bandwidth Estimation Technique: The method used for bandwidth estimation in this study is also in line with the throughput-based and buffer-based estimation approaches widely discussed in the literature, as stated by George and George (2021). This technique allows the system to accurately assess bandwidth availability and adjust the video bitrate accordingly, which is also adopted in this study to ensure optimal streaming performance under various network conditions.

6.2 Contrast with Previous Research. Testing and Implementation Approach, This study focuses on testing in scenarios with and without bandwidth limitation, which provides a practical perspective on the performance of HLS and ABR in real environments. Several previous studies, such as those conducted by Seufert et al. (2015), tend to focus more on simulations or testing in controlled laboratory environments, which may not fully reflect the dynamic and unpredictable network conditions in the field. The JBTv approach of directly testing in a real-world environment provides practical insights that may be more relevant for large-scale implementation.

Focus on Distance Education, While many previous studies have focused on the technical aspects of improving video streaming quality in general, this study makes a unique contribution with its specific application to supporting distance education through JBTv. This focus adds an additional dimension to the discussion on the importance of good video quality to support effective learning processes, which may not be explicitly addressed in other more technically oriented studies.

Recommendation for the Use of Content Delivery Network (CDN), This study also proposes the use of Content Delivery Network (CDN) as the next step to improve streaming service quality. While many previous studies have recognized the benefits of CDN for video streaming, this recommendation is emphasized more in the context of JBTv to compare the quality of service between RTMP and HLS implementations with and without the use of CDN. This demonstrates a more holistic approach to addressing video streaming quality issues by considering various supporting technologies. By analyzing the results of this study, it can be concluded that the implementation of ABR and HLS on JBTv has proven its effectiveness in improving the quality of video streaming and user experience, in line with the findings in previous literature.

7 Suggestions

7.1 Suggestions for Improving JBTv Streaming Quality.

(1) Use of Content Delivery Network (CDN)

To improve the quality and reliability of streaming services, it is recommended to use a CDN. A CDN can distribute video content to geographically dispersed servers, reducing latency, and ensuring that users get faster and more stable access to video content. The use of a CDN also allows JBTv to handle a larger number of viewers without any degradation in service quality.

(2) Implementation of More Accurate Bandwidth Estimation Techniques

The use of more advanced bandwidth estimation techniques such as hybrid methods can help in determining more accurate bandwidth availability. This will allow the system to better adjust the video bitrate according to network conditions, thereby reducing buffering and improving overall video quality.

(3) Optimization of ABR and HLS Usage

Although the implementation of ABR and HLS has yielded good results, there is still room for further optimization. For example, JBTv can explore the use of more advanced ABR algorithms that can be more responsive to changes in network conditions in real-time. Additionally, adjusting HLS parameters such as segment size and bitrate interval can also help in providing a smoother and more consistent streaming experience.

(4) Latency and Buffering Reduction

Reducing the latency between a live event and its display on the user's device can improve user satisfaction, especially for educational content that requires real-time interaction. Buffering reduction can also be achieved by improving buffer management and using more efficient codecs.

(5) Testing under Diverse Network Conditions

Testing under more realistic network conditions can provide a more accurate picture of how the system performs in the field. This testing can include variations in network speed, stability, and different usage scenarios such as access over cellular networks or public Wi-Fi.

(6) Improving Quality of Service and User Experience

Surveying users to get feedback on the quality of the streaming service can provide insight into areas for improvement. The data from these surveys can be used to make more precise adjustments to the streaming system, thereby improving the overall user experience.

7.2 Potential Areas for Further Investigation.

(1) Impact of 5G Networks on Streaming Quality

Investigating how faster and more stable 5G networks could impact the quality of video streaming on JBTv. This could include trials to see how 5G can reduce latency and increase video resolution without buffering, even in highly congested network conditions.

(2) Research on Dynamic Adaptation of Video Content

Conducting further research on how video content can be dynamically adjusted not only in terms of bitrate, but also in terms of format and resolution based on the device being used and the user's network conditions.

(3) Development of More Efficient ABR Algorithms

Investigating and developing ABR algorithms that are more efficient and responsive to rapidly changing network conditions. This research could include testing new algorithms and comparing them with existing algorithms to see which ones are most effective in improving streaming quality.

(4) Study on the Use of AI for Streaming Optimization

Investigating the use of artificial intelligence (AI) to automatically optimize streaming parameters, such as bitrate adjustment, CDN server selection, and buffer settings. AI can help make faster and more efficient decisions to improve streaming quality.

(5) Impact of Cloud-Based Streaming

Investigate the benefits and challenges of moving JBTv's streaming infrastructure to a cloud platform. This research could explore how cloud computing can improve scalability and flexibility in streaming service delivery, as well as how this impacts operational costs and service quality.

(6) Comparison of RTMP and HLS Effectiveness with CDN

Conduct a comparative study on the effectiveness of RTMP and HLS in the context of CDN deployment. This research

could involve measuring metrics such as buffering speed, video quality, and response time to determine which method is superior in providing an optimal streaming experience.

These suggestions and areas of inquiry are expected to assist JBTV in improving the quality of its streaming services and providing a better distance learning experience for users.

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