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Adaptive Quality Control Scheme Based on VBR Characteristics to Improve QoE of UHD Streaming Service

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***Abstract*—Recently, with the development of networks and smart devices, the demand for Ultra High Definition (UHD) video has risen, and HTTP adaptive streaming has attracted attention. HTTP adaptive streaming can guarantee high Quality of Experience (QoE) because it adaptively selects the video quality according to the network state. However, the existing quality control schemes experience unnecessary quality changes and low average video quality due to the bandwidth measurement and the quality control that do not consider the Variable Bit Rate (VBR) content characteristics of the UHD video. In this paper, we propose an adaptive quality control scheme based on VBR content characteristics to improve QoE of UHD streaming service. The proposed scheme measures the bandwidth using the actual bit rate of the segment and the difference in the network adaptability between segments. Furthermore, the proposed scheme defines a quality control region by considering the buffer state of the client. The quality control region consists of four subregions based on buffer thresholds, and the client selects the quality differently according to each subregion. Experimental results have shown that the proposed scheme improves the QoE compared to the existing schemes by minimizing the unnecessary quality changes and maximizing the average video quality.**

***Index Terms*—adaptive control, data transfer, high definition video, quality of service, streaming media.**

I. Introduction

According to the Cisco Visual Networking Index, the proportion of video traffic is currently estimated to be more than 70% of the Internet traffic and will increase to be more than 82% in 2021 [1]. Furthermore, with the spread of various smart devices and a rise in network speeds, the demand for high-quality media, such as Ultra High Definition (UHD) video, is increasing. Therefore, HTTP adaptive streaming to provide seamless streaming service for users is attracting attention [2].

In streaming based on the Real-time Transport Protocol (RTP) using the User Datagram Protocol (UDP), the streaming service is limited by firewalls and Network Address Translation (NAT). In addition, the scalability of service is degraded because the users need a separate streaming server. To solve these problems, adaptive streaming based on the Hypertext Transfer Protocol (HTTP) using the Transmission Control Protocol (TCP) has emerged recently. HTTP adaptive streaming is not limited by firewalls or NAT due to the use of the HTTP protocol and the client is able to use existing web servers. Commercialized services include Microsoft's Smooth Streaming [3], Apple's HTTP Live Streaming [4], Adobe's HTTP Dynamic Streaming [5], and the international standard called Dynamic Adaptive Streaming over HTTP (DASH) [6].

Fig. 1 shows the structure of HTTP adaptive streaming. The server encodes a video at various bit rates and stores it in the form of segments. The data on the segment's bit rate, the request's Uniform Resource Locator (URL), and the playback length are stored in a Media Presentation Description (MPD). When streaming starts, the client requests the MPD from the server. After receiving the MPD, the client determines the bit rate of the segment that will be requested using the measured bandwidth and the information described in the MPD. The client measures the available bandwidth using the download time and the amount of data in the received segment. HTTP adaptive streaming can guarantee the Quality of Experience (QoE) by matching the available bandwidth and video quality to adapt to the current network state [7-8].

In the previous researches for HTTP adaptive streaming, many adaptive quality control schemes have been proposed to adjust the video quality according to time-varying network condition [9]. To select the appropriate quality utilizing available bandwidth efficiently, one solution is that the client uses the estimated throughput in determining the video quality. Another solution is that the client uses the buffer occupancy in determining the video quality. By quality control considering the buffer occupancy, the client can avoid the playback interruptions due to the buffer depletion.

However, the existing quality control schemes still experience unnecessary quality changes and playback interruptions because these schemes measure the bandwidth and adjust quality without considering the Variable Bit Rate (VBR) content characteristics [10]. In addition, since the UHD video requires more than four times the bandwidth used to stream the Full High Definition (FHD) video, we need to measure the bandwidth and control the quality while considering the characteristics of the UHD video [11-12].

The VBR content characteristics of the video appear when the amount of data in each frame is dynamically

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allocated for transmitting high-quality video efficiently. If the video is encoded using a VBR encoding method, then a different bit rate is assigned to each segment even at the same video quality [13]. The difference in bit rate between segments is due to the dynamic scene included in each frame. If a frame contains a significant amount of motion, then we need more data in the encoding process to represent the corresponding scene. A UHD video has a large volume of data in each frame compared to that of a low-quality video. Therefore, when we encode the UHD video into the form of segments, the variation in the bit rate of each segment caused by the VBR content characteristics will be high. However, the existing bandwidth measurement uses the average bit rate of the segment. If the client measures the bandwidth without considering the changes in the download time caused by the difference between the actual bit rate and the average bit rate of the segment, then the download time is frequently changed even if there is no intervening traffic on the network. The client experiences unnecessary quality changes that result in QoE degradation [14-15].

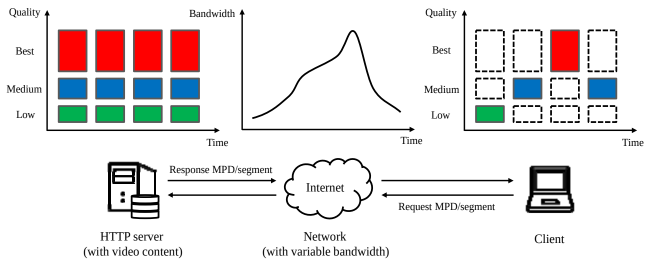

Figure 1. Structure of HTTP adaptive streaming

In this paper, we propose an adaptive quality control scheme based on VBR content characteristics to improve the QoE of a UHD streaming service. The proposed scheme measures the available bandwidth using the actual bit rate of the segment to minimize the error of the bandwidth measurement. Then, the client promptly responds to changes in the bandwidth by smoothing the measured bandwidth while considering the network adaptability of each segment. The proposed scheme defines the quality control region using the buffer state of the client. The quality control region consists of four subregions specified on the basis of buffer thresholds. These thresholds are calculated using the predicted changes in the buffer level. The proposed scheme determines the subregion that controls the video quality using the current buffer level. The client selects the quality differently according to the determined subregion. In addition, the proposed scheme resets the thresholds after the quality is changed in order to reflect the changes in the network state.

The remainder of this paper is organized as follows. Section 2 provides the background and reviews the related studies. In Section 3, the proposed VBR content characteristics-based quality control scheme is described. The results of the performance evaluation are provided in Section 4. Finally, the concluding remarks are given in Section 5.

II. Related Work

In this section, we illustrate what is the QoE first. Next, we describe the VBR content characteristics of the video, QoE degradation problems caused by these characteristics, and the existing quality control schemes to improve the QoE of video streaming.

A. Quality of experience

QoE means the degree of delight or annoyance of the user for an application or service [16]. In video streaming, many factors related to network, device, user expectations, and video content affect the QoE. For example, we are able to measure the QoE by considering startup delay, packet loss rates, average video quality, rebuffering events, and quality variations [17].

Table I shows the QoE metrics of HTTP adaptive streaming. According to researches for the QoE in video streaming service, perceived service quality is determined by video waiting times, video adaptation, and video quality [18-19]. When viewing a video, a user wants to view the video as clearly and smoothly as possible. Initial playback latency, playback interruption frequency, and playback interruption duration are metrics of the video waiting times. The initial playback latency due to video buffering degrades the QoE if the time is too long. The playback interruption frequency and duration are dominant in the QoE degradation. When the playback interruption frequently occurs and its duration is long, the user stops viewing the video due to the serious QoE degradation. Quality switching frequency, quality switching magnitude, and quality playback duration are metrics of the video adaptation. If the video quality frequently changes and degree of the changes is large, the user feels that the video has noise and artifacts. To improve the QoE, the high-quality video should be played for a long time. These metrics greatly affect the QoE, but the QoE degradation from frequent quality switching is not more severe than that from the playback interruption. Video quality is determined by a resolution, frame rate, and image quality. When the video is played with high resolution and high frame rate, the user is able to view the video clearly and smoothly. To achieve high perceived service quality in terms of the video quality, the image quality should also be considered. The distortion, artifacts, and noise should be minimized for the high image quality.

TABLE I. QOE METRICS OF HTTP ADAPTIVE STREAMING

Perceived Service Quality	Metrics
Video Waiting Times	Initial Playback Latency
	Playback Interruption Frequency
	Playback Interruption Duration
Video Adaptation	Quality Switching Frequency
	Quality Switching Magnitude
	Quality Playback Duration
Video Quality	Resolution
	Frame Rate
	Image Quality

B. VBR content characteristics of video

Fig. 2 shows the changes in the video quality and the buffer level during streaming of a video encoded with the Constant Bit Rate (CBR) and VBR methods. The CBR-encoded video in (a) assigns the same amount of data to each frame. In this case, the client does not experience the changes in the download time while the bandwidth is fixed

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because there is no difference in the bit rate between segments. On the other hand, the VBR-encoded video in (b) dynamically allocates the amount of data in each frame. In this case, there is a difference in the bit rate between segments even at the same video quality. Therefore, the download time is frequently changed according to the variation in the bit rate of each segment. However, the existing quality control schemes measure the bandwidth inaccurately because these schemes use the average bit rate of the segment. If the average bit rate of the segment is less than the actual bit rate of the segment, then the measured bandwidth is lower than the actual bandwidth. Therefore, the average video quality is degraded due to abrupt decreasing in the quality. If the average bit rate of the segment is greater than the actual bit rate of the segment, then the measured bandwidth is higher than the actual bandwidth. As a result, playback interruptions occur due to unnecessary increasing in the quality.

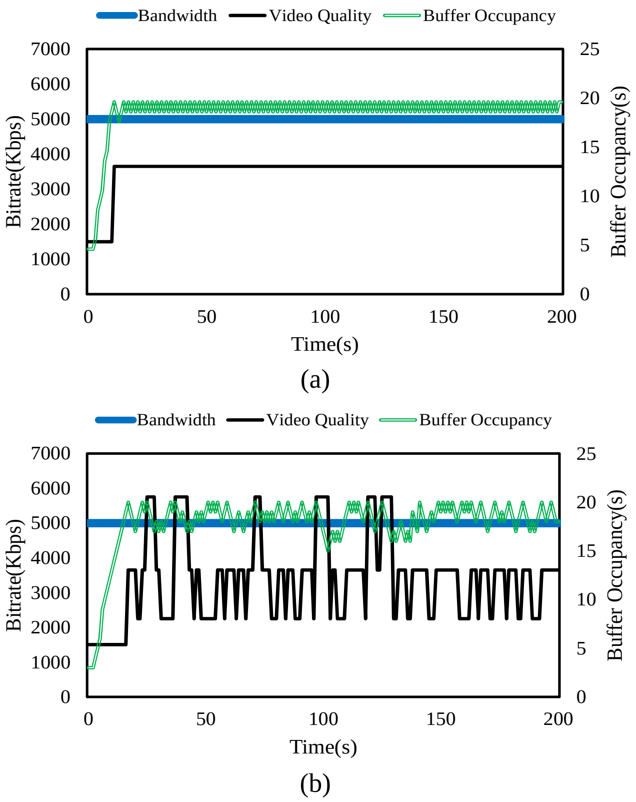

Figure 2. Changes in the video quality and the buffer level during streaming of (a) CBR video and (b) VBR video

C. Quality control schemes

Various quality control schemes have been proposed to improve the QoE of video streaming. These schemes can be classified as throughput-based, buffer-based and VBR content characteristics-based quality control schemes.

A throughput-based quality control scheme measures the available bandwidth of the network using the segment throughput. The segment throughput is calculated as the size of the last downloaded segment divided by the time taken to download it. The client selects the quality that has a lower bit rate than the measured bandwidth to minimize playback interruptions and utilize bandwidth efficiently.

Rate Adaptation for Adaptive HTTP Streaming (RAHS) defines a parameter that assesses the network state using the Media Segment Duration (MSD) and the Segment Fetch Time (SFT) [20]. To minimize unnecessary quality changes, the client selects the quality by comparing the parameter to quality control thresholds. These thresholds are calculated using the difference between the bit rates of each video quality. RAHS is able to use bandwidth efficiently because it responds to changes in the bandwidth promptly. However, RAHS does not consider the VBR content characteristics and the buffer state of the client. Therefore, unnecessary quality changes and playback interruptions occur.

The authors in Adaptive Streaming of Audiovisual Content (ASAC) reflects the measured bandwidth adaptively in the bandwidth smoothing [21]. If the fluctuations of the bandwidth are low, then the client increases the reflecting ratio of the previously measured bandwidth. In other cases, the client increases the reflecting ratio of the currently measured bandwidth. However, ASAC does not consider the buffer state of the client and the VBR content characteristics. Therefore, unnecessary quality changes and playback interruptions occur.

The conventional scheme measures the bandwidth using the Exponential Weighted Moving Average (EWMA) of the previously received segments [22]. The EWMA is able to improve the accuracy of bandwidth measurement because it reflects both instant fluctuations of bandwidth and the previously measured bandwidth. However, the smoothing parameter used in the EWMA is fixed depending on the type or the state of the network. Therefore, the bandwidth cannot be measured adaptively in various networks.

Buffer-based quality control schemes assess the buffer state of the client using the buffer level after receiving the segment. These schemes minimize playback interruptions by selecting the quality considering the current buffer state.

The Buffer-Based Adaptation for Adaptive HTTP Streaming (BAHS) determines the number of selectable quality levels according to the buffer level of the client [23]. The method adjusts the number of selectable quality levels by considering whether the buffer level is high. When the buffer level is high, the BAHS improves the average video quality by extending the range of selectable quality levels. In other cases, the BAHS minimizes playback interruptions by imposing restrictions on selectable quality levels. However, BAHS has a problem that the quality is frequently changed according to the changes of the buffer level and the number of video quality.

The Buffer-Based Approach to Rate Adaptation (BBA) defines a rate map that associates the buffer level with the video quality using the rate at which data of the received segment is stored and the rate at which the stored data is consumed [24]. The rate map determines buffer thresholds according to the number of quality levels. The BBA selects the quality using the current buffer level of the client. The method is able to minimize unnecessary quality changes thanks to the insensitive response to changes of bandwidth. However, since the BBA uses fixed thresholds, the quality is frequently changed when the buffer level fluctuates near the thresholds.

The Buffer Occupancy-based Lyapunov Algorithm (BOLA) defines a rate map that associates the video quality with the buffer state through optimization [25]. The buffer thresholds that exist in the rate map is changed according to

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the network state of the client. By determining the quality using the buffer level and the rate map, the BOLA improves the average video quality and minimizes unnecessary quality changes. Nevertheless, the BOLA changes the quality unnecessarily when the bandwidth is abruptly decreased due to a slow response to changes of bandwidth.

VBR content characteristics-based quality control schemes minimize unnecessary quality changes and playback interruptions by measuring the bandwidth using the actual bit rate of the segment and selecting the quality considering the buffer state of the client.

Segment Aware Rate Adaptation (SARA) measures the bandwidth with a weighted harmonic mean using the actual size of the segment described in MPD [26]. The method classifies the buffer state into four states based on fixed buffer thresholds. These states consist of a fast start, additive increase, aggressive switching, and delayed download. By adapting the quality on the basis of the buffer state, SARA minimizes unnecessary quality changes and improves the average video quality. However, the quality is changed frequently when the buffer level fluctuates near the thresholds. Furthermore, the client experiences a degradation of the average video quality because the client selects the lowest quality when the bandwidth abruptly decreases.

Future Buffer-based Adaptation for VBR Video (FBAV) predicts the changes in the buffer level for several segments assuming that the client receives the corresponding segments [27]. The client identifies whether or not playback interruptions occur when it maintains the previous quality level. If the playback interruptions do not occur, then the client selects the highest quality that has a lower bit rate than the measured bandwidth. Otherwise, the client decreases the quality by one level to guarantee seamless playback. However, FBAV changes the quality unnecessarily due to an inaccurate prediction of changes in the buffer level when the bandwidth is abruptly changed.

Smooth Quality Adaptation for VBR Video (SQAV) defines a quality control region using the changes in the buffer level [28]. The quality control region consists of up-switching, no-switching, and down-switching regions. SQAV minimizes playback interruptions and unnecessary quality changes by selecting the quality using the quality control region and the buffer state of the client. On the other hand, SQAV frequently changes each buffer threshold used in composing the quality control region when fluctuations of bandwidth are high. Therefore, unnecessary quality changes occur due to frequent changes in the control region.

As a result, existing quality control schemes suffer the QoE from the inaccurate bandwidth estimation and wrong buffer control. These schemes do not consider the abrupt changes in segment size, resulting in low adaptability to the bandwidth fluctuations. Thus, the client is not able to estimate the bandwidth by reflecting current changes in the bandwidth accurately. Wrong buffer control means the inappropriate use to the buffer thresholds for determining the video quality. According to the bandwidth fluctuations, quality control scheme adapts the video quality matching the available bandwidth. The buffer level also fluctuates by the determined video quality. These fluctuations cause the unnecessary quality changes, degrading the QoE.

III. VBR Content Characteristics-Based Quality Control Scheme

In this section, we propose a quality control scheme based on VBR content characteristics. The proposed scheme consists of bandwidth measurement using the information of the segment, and determination of quality according to the quality control region.

A. Overview

Fig. 3 shows the structure of the proposed quality control scheme. The *bandwidth estimation* measures the bandwidth using the throughput of the received segment and smooths the bandwidth while considering the difference in network adaptability between segments. The *utilization estimation* predicts bandwidth utilization of the segment using the changes in the buffer level after receiving the corresponding segment. The *control region management* defines the quality control region that consists of four subregions using the estimated bandwidth utilization. Furthermore, it resets the buffer thresholds that divide each subregion to reflect the changes in the network state after the client changes the quality. The *quality selection* determines the quality according to the subregion that corresponds to the current buffer level. The *segment request* demands the segment corresponding to the selected quality from the server.

We present the quality control scheme based on the VBR content characteristics in this paper. To mitigate the problem caused when not considering the inherent variability of the video, the proposed scheme estimates the bandwidth using the actual segment size first. Next, the proposed scheme selects the adaptation strategy based on the calculated buffer thresholds. The client calculates the value of these thresholds using the bandwidth utilization of the received segment. The bandwidth utilization of each segment varies dramatically according to the variations in the segment size. Thus, the buffer level is changed abruptly by the segment download time. Through considering the changes in the buffer level, the proposed scheme applies the different strategy to select the video quality. Thus, the client can avoid the playback interruptions efficiently. Furthermore, the proposed scheme updates the buffer thresholds to set the quality control region matching the current network state. With the quality control region, the proposed scheme can determine appropriate video quality that does not occur frequent quality switching and low average video quality.

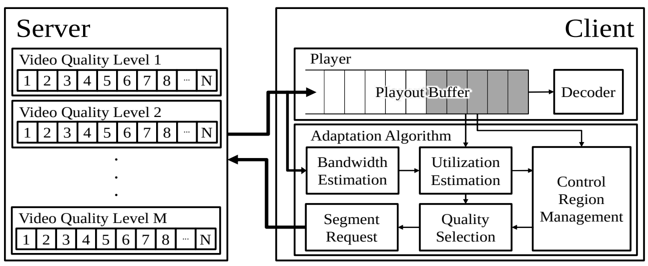

Figure 3. Structure of the proposed quality control scheme

B. Bandwidth measurement

To reflect the changes in the download time when streaming a VBR video, as given in (1), the proposed scheme calculates the throughput of the segment using the

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actual size of the segment. T_n is the throughput of the n th received segment, S_n is the actual size of the n th segment, and D_n is the download time after receiving the n th segment.

$$T_n = \frac{S_n}{D_n} \tag{1}$$

The client smooths the measured bandwidth using the EWMA, as given in (2). T_n^s is the smoothed bandwidth after receiving the n th segment, and α_n is the weight parameter.

$$T_n^s = \alpha_n \cdot T_{n-1}^s + (1-\alpha_n) \cdot T_n \tag{2}$$

If we use a fixed weight parameter, then the client responds slowly to changes in the bandwidth because the difference in network adaptability of each segment is not considered. Network adaptability means the responsiveness that how quickly the client is able to adapt to the network state changing in real time. The proposed scheme adjusts the weight parameter using (3). d_n^Q and d_n^R are the quality difference parameter and the bit rate difference parameter respectively.

$$\alpha_n = \frac{1-d_n^R}{1+e^{-d_n^Q}} \tag{3}$$

The quality difference parameter is calculated, as given in (4), using the changes of the requested quality. Variable l_n is the quality level of the n th segment, ranging from 1 to M . If the client demands a higher quality than the previous quality, then the network adaptability of the segment is degraded due to the increase of download time. In this case, the proposed scheme increases the reflecting ratio of the currently measured bandwidth by setting the value of the quality difference parameter to be less than 0. Therefore, the proposed scheme is able to react promptly to the changes in the bandwidth.

$$d_n^Q = 1 - \frac{l_{n+1}}{l_n} \tag{4}$$

The bit rate difference parameter is calculated according to (5) using the relative error between the actual bit rate and the average bit rate of the segment. $r_{avg}^{l_n}$ and $r_n^{l_n}$ are the average bit rate and the actual bit rate of the segment corresponding to quality level l_n , respectively. If the bit rate of the segment is high, then the network adaptability of the segment is degraded because the client needs to consume more time to receive the corresponding segment. Therefore, the proposed scheme increases the reflecting ratio of the currently measured bandwidth by decreasing the value of the bit rate difference parameter.

$$d_n^R = \frac{\left| r_{avg}^{l_n} - r_n^{l_n} \right|}{r_{avg}^{l_n}} \tag{5}$$

C. Quality control region

After the bandwidth measurement, the proposed scheme defines the quality control region that assesses the buffer state of the client using the current buffer level. The quality control region consists of four subregions, and each region is divided by the buffer thresholds. These thresholds are calculated using the predicted changes in the buffer level after receiving the segment. To predict the changes in the buffer level, the proposed scheme calculates the bandwidth utilization of segment u_n using the download time and the playback length of the segment as shown in (6).

$$u_n = \frac{D_n}{\tau} \tag{6}$$

Fig. 4 shows the bandwidth utilization of the segment at various quality levels. In (a), the client has received a segment that has a lower bit rate than the available bandwidth. The client is able to utilize more bandwidth than the actual required bandwidth in this case. Therefore, the buffer level increases as the difference between the download time and the playback length of the segment as a result because the download time becomes shorter than the playback length of the segment. On the other hand, in (b), the client needs more bandwidth to receive the segment within the time shorter than the playback length of the segment. Therefore, the buffer level decreases as the difference between the download time and the playback length of the segment. The bandwidth utilization of the segment denotes the ratio of the available bandwidth and that used to receive the segment. As a result, we are able to predict the changes in the buffer level of the client based on the playback length of the segment.

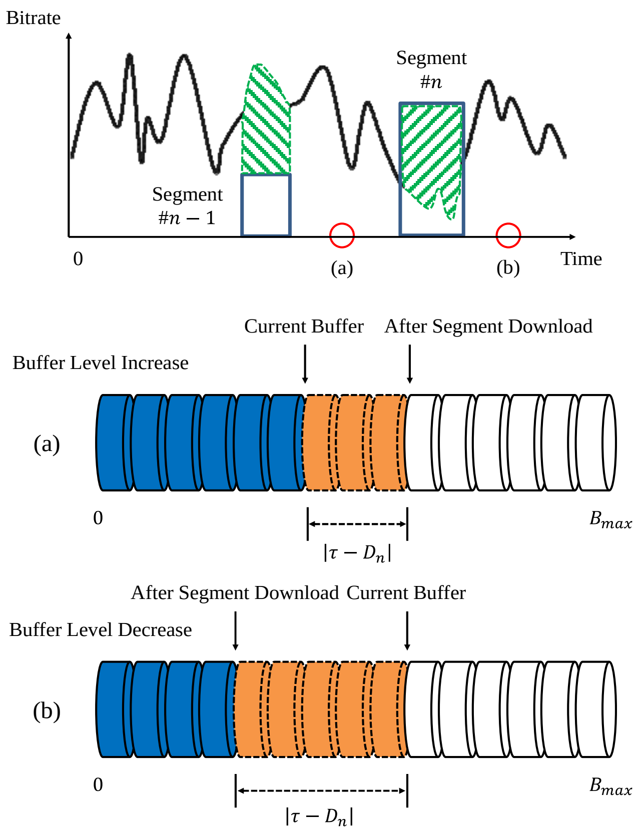

Figure 4. Bandwidth utilization of the segment for (a) the low-quality and (b) the high-quality

The buffer thresholds are calculated using the bandwidth utilization of the received segment. However, in the early stage of streaming, playback interruptions are able to occur according to abrupt changes in the network state due to an insufficient amount of data in the buffer. Therefore, the

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proposed scheme performs the quality control after the initial buffering. In the case of HTTP adaptive streaming, the client selects the lowest quality to fill the buffer after streaming has started. The initial buffering is performed until the buffer level exceeds the maximum buffer, and the client requests the next segment from the server immediately after receiving the segment. Nevertheless, the average video quality is degraded because the client selects the low-quality video even if the available bandwidth allows high-quality video. Therefore, the proposed scheme selects the quality during the initial buffering by considering the buffer charging rate γ_n .The buffer charging rate denotes how much buffer is consumed after receiving the segment. The proposed scheme calculates the buffer charging rate using the download time and the buffer consumption caused by video playback, as given in (7).

$$\gamma_n = \frac{\tau}{D_n} - 1 \tag{7}$$

The initial buffering ends when the buffer charging rate is less than 0, and when the client selects the quality that is lower by one level than the highest quality, or the buffer level exceeds the maximum buffer. The client immediately requests the next segment from the server after receiving the segment and increases the quality by one level from the lowest quality during the initial buffering.

The proposed scheme calculates the initial thresholds of the quality control region using the bandwidth utilization u_{init} and the buffer level B_{init} shortly after the initial buffering, as given in (8). B_{init}^{min} and B_{init}^{max} are the minimum region threshold and the maximum region threshold calculated after the initial buffering, respectively.

$$\begin{aligned} B_{init}^{min} &= B_{init} - \tau \cdot u_{init} \\ B_{init}^{max} &= B_{init} + \tau \cdot u_{init} \end{aligned} \tag{8}$$

Fig. 5 shows the quality control region. The quality control region consists of four subregions referred to as down-switching, no-switching, conservative up-switching, and aggressive up-switching regions. If the buffer level is lower than the minimum region threshold, then the subregion corresponds to down-switching. The down-switching region aims to obtain a sufficient buffer because to minimize the risk of playback interruptions. If the buffer level is between the minimum and the maximum region threshold, then the subregion corresponds to no-switching. The no-switching region aims to minimize unnecessary quality changes. If the buffer level is higher than the maximum region threshold and lower than the value of the difference between the maximum buffer and the minimum region threshold, then the subregion corresponds to conservative up-switching. The conservative up-switching region aims to minimize unnecessary quality changes and maximize the average video quality. If the buffer level is higher than the difference between the maximum buffer and the minimum region threshold and lower than the maximum buffer, then the subregion corresponds to aggressive up-switching. In the aggressive up-switching region, the risk of playback interruptions is low even if the client increases the quality. Therefore, this region aims to maximize the average video quality.

The proposed scheme resets the minimum region threshold and the maximum region threshold each time that the client changes the quality using the quality control region. If these thresholds are fixed, then unnecessary quality changes occur because the subregion to be determined is changed frequently according to changes in the buffer level. Furthermore, after the client changes the quality, the buffer state, the measured bandwidth, and the bandwidth utilization vary even at the same bandwidth as before. Therefore, we need to adjust each threshold to reflect the changes in the network state.

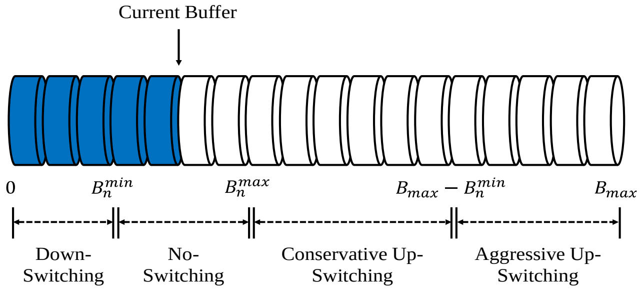

Figure 5. Quality control region

After the client changes the quality in the down-switching region, the proposed scheme calculates the minimum region threshold and the maximum region threshold using (9). B_n^{min} is the minimum region threshold after the client receives the n th segment. The proposed scheme adjusts the lengths of the down-switching and no-switching regions based on the minimum region threshold. Therefore, the proposed scheme improves the average video quality by increasing the quality only when the buffer level is sufficient.

$$\begin{aligned} B_{n+1}^{min} &= B_n^{min} - \tau \cdot u_n \\ B_{n+1}^{max} &= B_n^{min} + \tau \cdot u_n \end{aligned} \tag{9}$$

The proposed scheme calculates the minimum region threshold and the maximum region threshold as given in (10) after the client increases the quality in the conservative up-switching region. B_n^{max} is the maximum region threshold after the client receives the n th segment. The client moves the location of the down-switching region in the direction that corresponds to a higher buffer level. Therefore, the proposed scheme minimizes playback interruptions by decreasing the quality when there are the abrupt changes in the buffer level due to high bandwidth utilization of the segment.

$$\begin{aligned} B_{n+1}^{min} &= B_n^{max} - \tau \cdot u_n \\ B_{n+1}^{max} &= B_n^{max} + \tau \cdot u_n \end{aligned} \tag{10}$$

If the client increases the quality in the aggressive up-switching region or maintains the previous quality in the no-switching region, then the proposed scheme maintains the minimum region threshold and the maximum region threshold, as given in (11). Therefore, the client does not experience playback interruptions and unnecessary quality changes caused by fluctuations in the length of each subregion.

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$$\begin{aligned} B_{n+1}^{min} &= B_n^{min} \\ B_{n+1}^{max} &= B_n^{max} \end{aligned} \tag{11}$$

D. Quality control according to each subregion

When the subregion corresponds to the down-switching region, the proposed scheme decreases the quality by one level to minimize the playback interruptions, as given in (12).

$$l_{n+1} = l_n - 1 \tag{12}$$

However, when the minimum region threshold moves in the direction that corresponds to a higher buffer level, the length of the down-switching region increases compared to its earlier value. Therefore, the average video quality degrades because the client decreases the quality even if the buffer level is sufficient. The proposed scheme defines the optimal switching region, as shown in Fig. 6. In the optimal switching region, the client decreases the quality considering the change rates of the buffer level. By predicting the buffer charging rate and bandwidth utilization of the segment, the proposed scheme determines the range of the buffer level where playback interruptions do not occur even if the client maintains the previous quality. The expected buffer charging rate γ_{n+1}^{exp} is calculated, as given in (13), and the expected bandwidth utilization u_{n+1}^{exp} is calculated, as given in (14).

$$\gamma_{n+1}^{exp} = \gamma_n \cdot \left(1 - \frac{S_{n+1}}{S_n}\right) \tag{13}$$

$$u_{n+1}^{exp} = \frac{1}{1 + \gamma_{n+1}^{exp}} \tag{14}$$

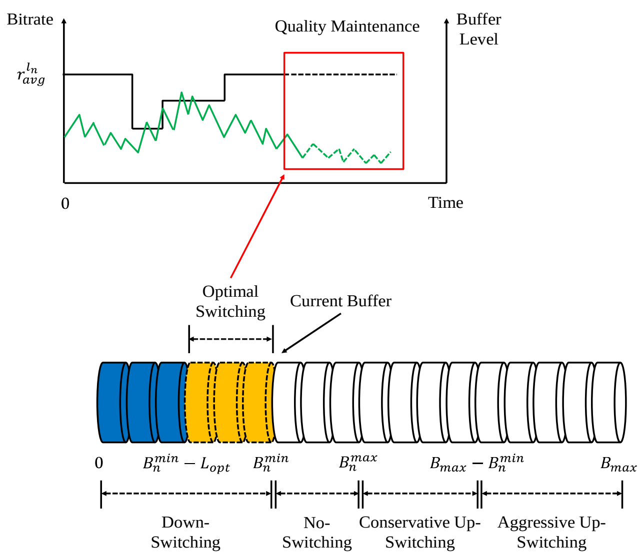

Figure 6. Optimal switching region

The proposed scheme calculates the length of the optimal switching region L_{opt} by comparing the expected buffer charging rate to the expected bandwidth utilization, as given in (15). The buffer level after receiving the segment decreases abruptly if bandwidth utilization is high and the buffer charging rate is low. Therefore, the proposed scheme defines the optimal switching region by considering the difference between the expected buffer charging rate and the expected bandwidth utilization. The proposed scheme judges that playback interruptions do not occur even while maintaining the quality when the value of the difference between the expected buffer charging rate and the expected bandwidth utilization is lower than the playback length of the segment.

$$L_{opt} = \begin{cases} \tau \cdot (u_{n+1}^{exp} - \gamma_{n+1}^{exp}), & if \;\; (u_{n+1}^{exp} - \gamma_{n+1}^{exp}) < \tau \\ 0, & else \end{cases} \tag{15}$$

In the optimal switching region, the proposed scheme calculates the change rates of the buffer level ΔB_{n+1}^r using (16). B_{n+1}^r is the relative error of the buffer level when the client maintains the previous quality.

$$\Delta B_{n+1}^r = \frac{B_n - B_{n+1}^r}{B_n} \tag{16}$$

Then, as given in (17), the proposed scheme calculates the quality maintenance probability p_r using the change rates of the buffer level. The client decreases the quality when the value of the generated random number is larger than the quality maintenance probability and maintains the previous quality otherwise. Therefore, the proposed scheme maximizes the average video quality by decreasing the quality only when the risk of playback interruptions is high.

$$p_r = 1 - \left|\tanh(\Delta B_{n+1}^r)\right| \tag{17}$$

When the subregion corresponds to the no-switching region, as given in (18), the proposed scheme maintains the previous quality to minimize unnecessary quality changes.

$$l_{n+1} = l_n \tag{18}$$

When the subregion corresponds to the conservative up-switching region, as given in (19), the proposed scheme increases the quality by one level more than the quality $\hat{l}_n$ which is determined using the measured bandwidth. Therefore, the client is able to increase the quality in the range where the changes of the buffer level are not high.

$$\begin{aligned} \hat{l}_n &= \max_{l_n}\{r_{avg}^{l_n} < T_n^s\} \\ l_{n+1} &= \hat{l}_n + 1 \end{aligned} \tag{19}$$

When the subregion corresponds to the aggressive up-switching region, as given in (20), the proposed scheme maximizes the average video quality by increasing the quality by one level more than the previous quality.

$$l_{n+1} = l_n + 1 \tag{20}$$

IV. Performance Evaluation

This section presents and discusses the results of the simulation. To evaluate the performance of the proposed scheme, we implemented a simulation environment and an HTTP adaptive streaming player using the NS-3 network simulator [29].

A. Simulation setup

Fig. 7 shows the simulation environment. The server supports video quality ranging from Standard Definition (SD) to UHD, and the actual bit rate of the segment corresponding to each quality ranges from 65% to 150% of the average bit rate of the segment as shown in Fig. 8. The playback length of the segment is set to 2 seconds, and the

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experiments are performed for 200 seconds. We have evaluated the performance of the proposed scheme and the existing schemes under the fixed and variable bandwidth scenarios. In the case of the fixed bandwidth scenario, the available bandwidth is not changed until the streaming ends. On the other hand, in the variable bandwidth scenario, the available bandwidth is changed by the CBR traffic. The change order is {25 Mbps -> 20 Mbps -> 16.5 Mbps -> 21.5 Mbps -> 14.5 Mbps -> 19 Mbps -> 21 Mbps -> 25 Mbps}.

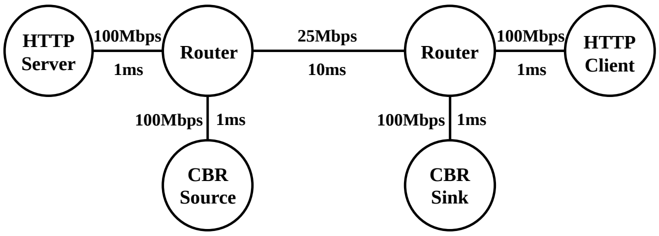

Figure 7. Simulation environment

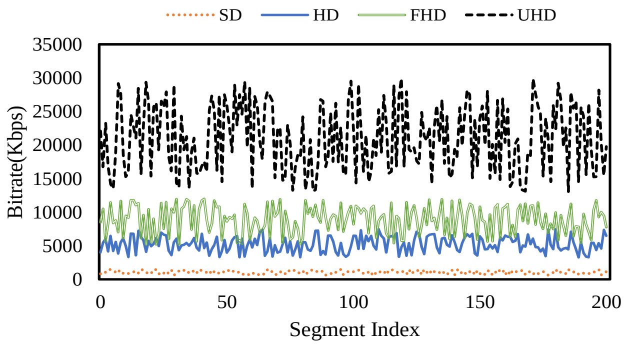

Figure 8. Changes in the bit rate of the segment by the quality

B. *Fixed bandwidth scenario*

Fig. 9 shows the changes in the video quality and the buffer level in the fixed bandwidth scenario. SQAV changes the quality unnecessarily for a while after streaming has started. This is because the buffer level fluctuates between the subregions in the quality control region so that the client changes the control strategy frequently. Furthermore, from 15 to 100 seconds, SQAV selects the low-quality even if the buffer level is sufficient because the client responds slowly to the changes in the bandwidth. In the BBA, the quality fluctuates from low to high after streaming has started. The frequent changes of quality are caused by changes in the buffer level near the thresholds. For example, from 140 to 160 seconds, the buffer level fluctuates around the threshold that determines whether the client selects the FHD or UHD quality. SARA selects the high-quality when the buffer level is sufficient. However, when the buffer level decreases to a lower level, SARA selects the lowest video quality degrading the average video quality. Furthermore, playback interruptions are able to occur if the network state is changed, because the client changes from the high-quality to the low-quality abruptly. On the other hand, the proposed scheme increases the quality promptly and maintains the highest quality when the client judges that the current buffer level is sufficient. For the duration of approximately 150 seconds after streaming has started, the proposed scheme does not change the quality unnecessarily. Nevertheless, at 152 seconds, the proposed scheme decreases the quality to avoid playback interruptions because the subregion at this time corresponds to the down-switching region. Therefore, the proposed scheme is able to minimize unnecessary quality changes and maximize the average video quality.

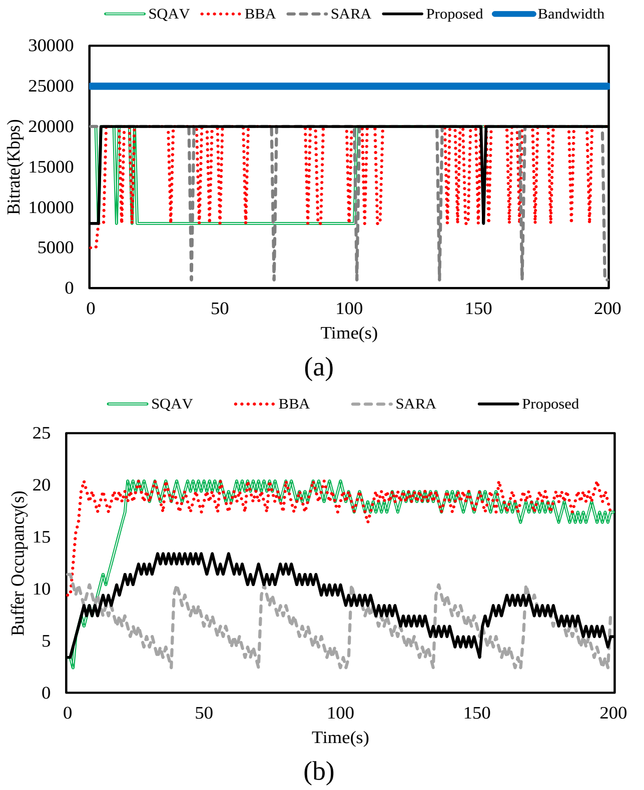

Figure 9. Changes in (a) the video quality level and (b) the buffer level in the fixed bandwidth scenario

C. *Variable bandwidth scenario*

Fig. 10 shows the changes in the video quality and the buffer level in the variable bandwidth scenario. Shortly after the streaming starts, SQAV changes the quality frequently according to the changes in the buffer level due to the fixed thresholds. In the initial buffering, the buffer level increases abruptly due to the segment request for the low video quality. As a result, the fluctuations in the buffer level occur resulting in unnecessary quality changes. After 20 seconds, the client selects the low video quality even if the risk of the playback interruptions is low. Although the bandwidth increases enough to stream the high video quality, SQAV does not increase the video quality. This is because SQAV responds slowly to the changes in the bandwidth. The buffer level in the BBA changes frequently near the thresholds. This is due to the difference in the download time of the received segment. When the available bandwidth increases again after decreasing, BBA changes the video quality. These changes cause the fluctuations in the buffer level. Therefore, QoE is degraded due to unnecessary quality changes. SARA is not able to manage the buffer stably. This is because the client changes the video quality abruptly from high-quality to low-quality or the opposite case. Thus, the client experiences unnecessary quality changes, and the risk of the playback interruptions in high. On the other hand, the proposed scheme selects the highest video quality quickly after video streaming has been started. By maintaining the high video quality and the sufficient buffer level during a long time, the proposed scheme is able to maximize the

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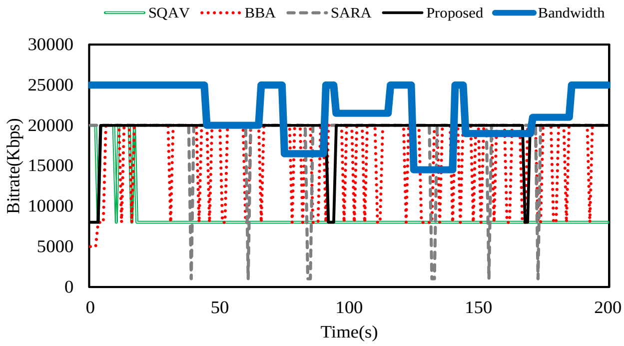
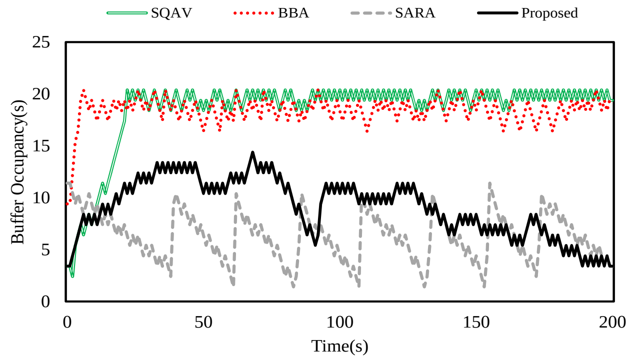
average video quality while minimizing the risk of the playback interruptions.



(a)



(b)

Figure 10. Changes in (a) the video quality level and (b) the buffer level in the variable bandwidth scenario

D. Evaluation of QoE

Both in the fixed bandwidth and the variable bandwidth scenarios, playback interruptions did not occur. We considered only the average video quality and the frequency of quality changes to evaluate the QoE. These metrics are important to the quality of user experience followed by the playback interruption frequency and duration. We used (21) to calculate the average video quality, R_{avg} , to the total received segments.

$$R_{avg} = \frac{1}{n}\sum_{k=1}^{n} r_{avg}^{l_k} \tag{21}$$

The frequency of the quality changes is calculated as given in (22). The value of $f(r_{avg}^{l_k})$ is set to 1 if the previous quality is different from the current quality and set to 0 in the opposite case. Thus, R_{freq} means how many the video quality has been changed during the whole streaming duration.

$$R_{freq} = \sum_{k=1}^{n} f(r_{avg}^{l_k}), \qquad f(r_{avg}^{l_k}) = \begin{cases} 1, & if \;\; r_{avg}^{l_{k-1}} \neq r_{avg}^{l_k} \\ 0, & otherwise \end{cases} \tag{22}$$

Fig. 11 shows the QoE evaluation of various quality control schemes. In the fixed bandwidth scenario, the frequency of the quality changes in the proposed scheme is 93.75%, 72.8%, and 62.5% lower than that of BBA, SARA, and SQAV, respectively. The average video quality of the proposed scheme is 9.2%, 1.8%, and 33.6% higher than that of BBA, SARA, and SQAV, respectively. In the variable bandwidth scenario, the frequency of the quality changes in the proposed scheme is 92%, 58.4%, and 28.6% lower than that of BBA, SARA, and SQAV, respectively. The average video quality of the proposed scheme is 14.2%, 1.1%, and 118.8% higher than that of BBA, SARA, and SQAV, respectively. From the results of the QoE evaluation, we confirmed that the proposed scheme improves the QoE of video streaming. The proposed scheme maximizes the average video quality and minimizes the unnecessary quality changes simultaneously.

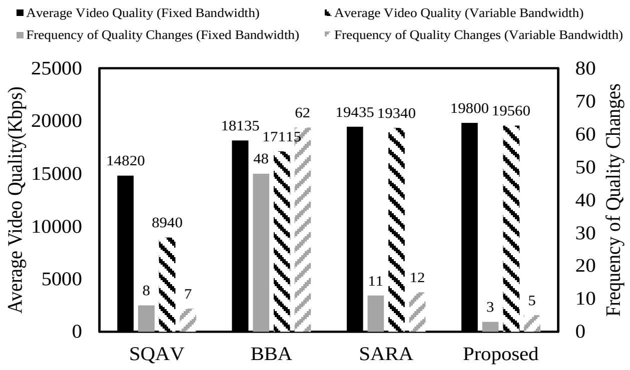


Figure 11. QoE evaluation of the quality control schemes

V. Conclusion

As the user demand for video streaming increases, HTTP adaptive streaming has become a promising technique for video streaming. Thanks to the adaptive transmission strategy matching the current network state, HTTP adaptive streaming can guarantee QoE of video streaming even for the UHD video. Various quality control schemes such as throughput-based and buffer-based quality control scheme have been proposed for HTTP adaptive streaming.

However, the existing quality control schemes still degrade QoE because these schemes do not consider the VBR content characteristics of the UHD video. In this paper, we have proposed the quality control scheme based on VBR content characteristics to improve QoE of a UHD streaming service. To minimize unnecessary quality changes due to inaccurate bandwidth measurement, the proposed scheme measures the bandwidth by using the actual size of the segment and considering the network adaptability of each segment. The proposed scheme defines a quality control region that consists of four subregions. The quality control region is divided into each subregion by using the buffer thresholds. These thresholds are calculated using the predicted changes in the buffer level based on the bandwidth utilization of the segment. The proposed scheme selects the quality differently according to each subregion and resets the buffer thresholds after the client changes the video quality. The changes in the video quality mean that the network state fluctuates differently compared to the previous state. Experimental results have shown that the proposed scheme improves QoE compared to the existing schemes by minimizing unnecessary quality changes and maximizing the average video quality.

In future research, we plan to implement the proposed scheme in a real-world video player and evaluate the

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performance of the proposed scheme in comparison with other quality control schemes under various network scenarios.

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