Adaptive Rate Control Scheme for Streaming-based Content Sharing Service

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Abstract

This paper presents an adaptive rate control scheme for streaming-based content sharing service. This scheme delivers multimedia contents from a user device to another device or seamlessly redirects streaming service across heterogeneous user devices. In the proposed scheme, a streaming server adjusts video quality level according to the network and client status. Our scheme is different from other rate control schemes, because the video quality at the server is decided not only based on the available bandwidth, but also based on the device characteristics and bandwidth requirement at the access network. We also propose a bandwidth estimation method to achieve more equitable bandwidth allocations among streaming flows competing for the same narrow link with different Round Trip Times (RTTs). Through the simulation, we prove that our scheme improves the network stability and the quality of streaming service by appropriately adjusting the quality of the video stream. The simulation results also demonstrate the ability of the proposed scheme in ensuring RTT-fairness while remaining throughput efficient.

Keywords: Streaming Service, Rate Control, Quality of Service, Content Sharing Service, RTT-fairness

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1. Introduction

In the recent years, due to the prevalence of various user devices and wireless networks, content sharing services have emerged as one of the most popular services. Content sharing service allows multiple devices to view the same content at the same instant or at different time periods. Multimedia streaming has been a noticeable trend for content sharing services [1]. Streaming-based content sharing services transmit multimedia contents from a user device to another device or seamlessly redirect the streaming service across heterogeneous user devices as shown in **Fig. 1**.

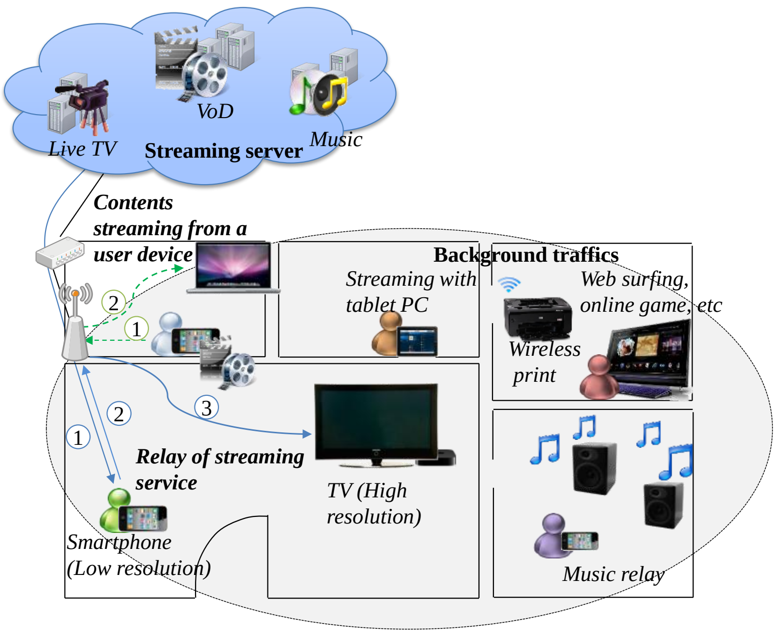

Fig. 1. Streaming scenario in a content sharing service

There are various devices that are equipped with wireless networking interfaces. Most of these devices have different resolution. In Fig. 1, a device, such as smartphone, can use a streaming service directly from the streaming server. However, the user can also use the streaming service on TV by relaying the streaming service from a smartphone to TV. This requires three times higher bandwidth at the access network. Also, the multimedia streaming service has stringent timing requirements for video playback because a server transfers the multimedia content to the client as a continuous stream and the client consumes the data as it arrives [2]. This requirement imposes bandwidth, delay and loss demands on the underlying network that is responsible for delivering the data. But, in a best effort network, network parameters such as delay and bandwidth vary unpredictably providing no guarantee for a timely delivery. This mismatch forms the fundamental challenge in streaming video over the Internet [3].

A widely accepted technique to counter this challenge is network-aware rate control scheme. Network aware applications sense the varying resource availability and accordingly adapt their transmission rate or video quality. However, as the network stability is only taken in to consideration, previous works do not consider the characteristics of the multimedia streaming application. In the streaming-based content sharing services, more bandwidth is required to relay streaming service and the client may suffer from interruption of video playback due to buffer underflow [4]. Furthermore, it is not easy to support cross-device handover, which dynamically and seamlessly switches between display devices on demand. For instance, a mobile device may not often have the capability to receive services with the same quality as a high-performance PC. For such cases, the quality of the multimedia content has to be adjusted based on the capabilities of the mobile device [5].

These problems can be solved by using Scalable Video Coding (SVC) which provides a bit stream with a layered structure. This includes a base layer and enhancement layers. The base layer presents the minimum quality of a bitstream. The more enhancement layers the client receives, the higher the video quality on the client side. SVC is a convenient solution that adjusts the data rate according to the characteristics of a diverse device and varying bandwidth in the Internet [6]. However, many streaming protocols like TCP-Friendly Rate Control (TFRC) estimate the transmission rate based on the session RTTs. This leads to RTT-unfairness, which severely affects the performance of the long-RTT flows [7,8]. When users in a home network are served with streaming from servers which have different end-to-end propagation delays, a long RTT flow uses less bandwidth than a short RTT flow. Therefore, the long RTT flow receives lower quality video than the short RTT flow. **Fig. 2** shows the transmission rate of two TFRC flows which have different end-to-end propagation delay.

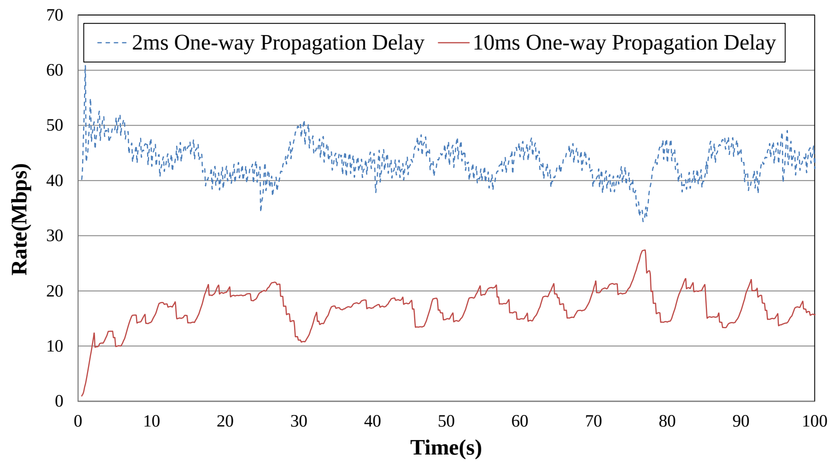

Fig. 2. Comparison of transmission rate between two TFRC ows with 2ms and 10ms end-to-end propagation delays

To prevent the throughput reduction caused by a long end-to-end propagation delay, there are a few works recently proposed for new RTT-fair TCPs like TCP Vegas, FAST TCP and CUBIC. However, these protocols are not designed for an efficient streaming service.

In this paper, we propose a Network and Device Adaptive Rate Control (NAD-RC) scheme to improve the quality of service (QoS) for streaming-based content sharing service. The NAD-RC estimates RTT-fair bandwidth by modifying Padhye's analytical model for the available bandwidth share of TCP connection [9]. It adjusts the quality level of SVC bitstream, based on the estimated available bandwidth, required bandwidth to relay the streaming service, device resolution, and client buffer status. Therefore, the proposed scheme improves the network stability and video quality by reducing the packet loss and increasing the throughput even a long end-to-end propagation delay. It also provides smoothed playback by preventing buffer underflow or overflow.

The rest of the paper is organized as follows. In the next section, we review some of the related works. In Section III, we present an adaptive rate control scheme for streaming-based content sharing service. Detailed description of our simulation results are presented in Section IV. Finally, Section V concludes the paper and discusses some of our future works.

2. Related Work

2.1 Adaptive Rate Control Schemes

In a streaming service, the end-user can start displaying the video data or multimedia data before the entire file has been transmitted. To support a seamless playback at the client, the streaming system requires an efficient mechanism for delivering continuous media. To achieve this, the bandwidth efficiency and flexibility between the streaming servers and end-user devices are challenging technical issues. Zhou et al. propose an adaptive rate scheduling scheme to improve the QoS of multi user video streaming over multi-channel multi-radio multi hop wireless networks [10]. However the adaptive rate scheduling scheme does not consider the characteristics of client devices. In response to such challenges, SVC has been proposed. In order to support QoS for multimedia streaming services, various rate control schemes using the SVC have been studied. The network-aware rate control system using SVC, which is proposed in [6], transmits the maximal bitrate according to an estimated available bandwidth. This system adjusts the quality level of the SVC bitstream in order to adapt the transmission rate to a varying bandwidth. However, this system has a buffer underflow problem which does not support the continuity of media playback, as it does not consider the status of the client buffer.

Unlike the network adaptive rate control scheme which is discussed above, a buffer-driven scheme is proposed in [4]. It scales the video quality and schedules data for transmission based on the client buffer occupancy and the server buffer occupancy. Therefore, the buffer-driven scheme provides smoothed playback by preventing buffer underflow or overflow. During underload periods, the video quality is increased, which results in higher information rate, whereas during overload periods, the video quality is decreased. However, the buffer-driven scheme has inherited instability and unfairness from the UDP-based streaming protocols as it has no concern of the network situation.

The proposed scheme in [11] includes more sophisticated features that considers both network and user requirements. From the network viewpoint, the proposed mechanism estimates the TCP-friendly rate that is suitable for the network status. Therefore, it is able to adjust the transmission rate of the video stream in a TCP-friendly manner. From the user viewpoint, this scheme provides smooth playback by preventing the receiver buffer underflow or overflow. Recent paper by Koo et al. [12] propose a rate control scheme called the Network and Client-Aware Rate Control (NCAR) scheme, to improve the user perceived QoS of multimedia streaming services. The NCAR scheme has two key functions for rate-control, congestion control and flow control. The congestion control function adjusts the server’s transmission bit-rate based on the estimated value of network-aware information such as the network congestion degree (i.e., packet loss ratio, delay) and bandwidth variation. To prevent client buffer overflow and underflow, the flow control function adjusts the quality level of the transmitting SVC bitstream. This is done based on the estimated value of the client-aware information, such as the buffer occupancy ratio. However, these schemes do not consider RTT-unfairness between the two competing flows which have different end-to-end propagation delay. A flow which has long end-to-end propagation delay shares less bandwidth than a flow that has short end-to-end propagation delay. This leads to the degradation of the SVC quality level.

2.2 Scalable Video Coding (SVC) Scheme

In January 2005, the ISO/IEC Moving Pictures Experts Group (MPEG) and the Video Coding Experts Group (VCEG) of ITU-T started jointly the Scalable Video Coding (SVC) project as an Amendment of H.264/AVC standard that was to be finalized in 2007. Furthermore, in November 2005, Audio/Video Transport (AVT) Working Group of the Internet Engineering Task Force (IETF) started to draft RTP (Real Time Protocol) payload format for the scalability extension of H.264/AVC and signaling for the layered coding structures. It is finalized in 2011 [13].

SVC is an extension of motion-compensated transcoding that achieves a high degree of spatial, temporal and quality scalability. An SVC bit-stream consists of a base layer and one or more enhancement layers. The omission of some or all of the enhancement layers still allows a reasonable quality of, albeit with some combination of decoded video reduced SNR (Signal-to-Noise Ratio), temporal and spatial resolution. The base layer is a bitstream conforming to H.264/AVC that ensures backward compatibility with legacy decoders. It provides low quality of video with low resolution, low frame rate (temporal resolution), or low picture fidelity (PSNR). The enhancement layers include the information of the frames with higher resolution, frame rate, or PSNR, but it cannot be decoded without a base layer. For example, in temporal scalability, each frame in the temporal base layer is entirely predicted from the previous frame in this layer as shown in **Fig. 3**.

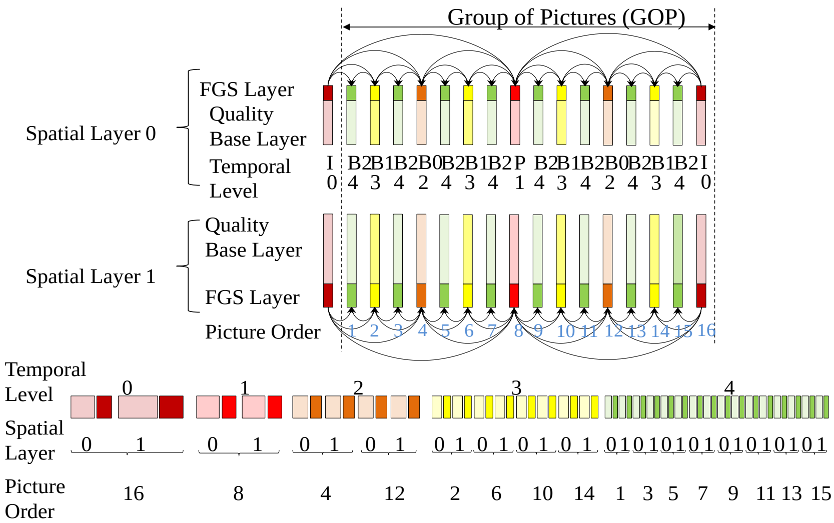

Fig. 3. Example of a video coding based on the scalabilityextension of SVC using hierarchical B-pictures

However, an enhancement-layer frame can be bi-directionally predicted, using two or four adjacent frames of lower temporal resolution as references. The proposed adaptive rate control scheme takes advantage of SVC.

2.3 Addressing RTT-unfairness

Most of the rate control schemes decide the transmission rate based on the network delay. However, senders with different end-to-end propagation delays will typically receive feedbacks at different rates. They will accordingly adapt their transmission rates. This phenomenon causes the RTT-bias, or RTT-unfairness problems.

To solve RTT-unfairness problem, a few works have been recently proposed new RTT-fair TCPs. Among the most relevant ones, TCP Hybla [14] implements a constant increase algorithm. It provides RTT-fairness under a certain stability bond. TCP Vegas [15] provides a good RTT-fairness but disregards friendliness. FAST TCP [16] increases TCP Vegas's stability bounds, but with a behavior that results either too timid or too aggressive when coexisting with the legacy TCP protocols. Finally, CUBIC [17] features a linear RTT-fairness that claims to improve BIC [18]. In particular, CUBIC tries to decouple the window growth from the returning ACK process. With CUBIC, the window size is a function of the time elapsed since the last packet loss. Thus, it allows higher efficiency in the case of long fat networks and it reduces, even though it does not completely eliminate, the throughput dependency from the RTT. Indeed, the throughput still corresponds to the ratio between the window size and the RTT, where two flows with similar packet loss trend and may have the same window but different RTTs. These researches tried to improve the RTT-fairness of TCP, but were not designed for streaming services.

3. Adaptive Rate Control Scheme to Improve QoS of Content Sharing Service

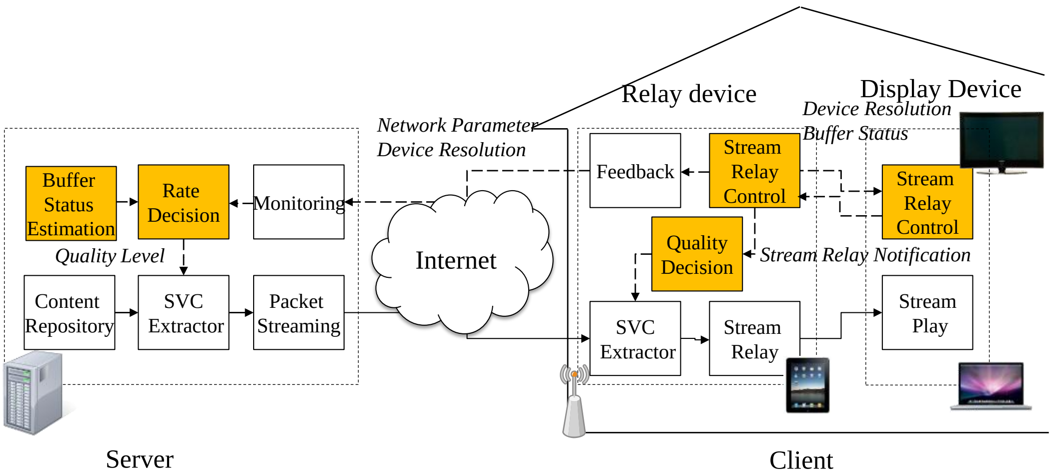

Fig. 4. Adaptive rate control system for streaming-based content sharing service

Fig. 4 shows the proposed streaming system for content sharing service. Our system consists of a server, a relay device and a display device. The server streams a multimedia content to the relay device. The relay device transmits the multimedia stream to the display device. The display device plays the arrived multimedia stream. The relay device act as a server when it streams stored contents to the display device. Also, the relay device can play the multimedia stream as a display device. In the proposed system, we use the RTP (Real-time Transport Protocol) and the RTCP (Real-time Transport Control Protocol) protocols for the delivery of video data and for the feedback on network quality. We also use the RTSP protocol for the establishment and control of media streams. There are four closely interacting modules in our system.

- Transmission Rate Decision Module (TRDM) determines the transmission rate based on Padhye's analytical model. We modify the RTT parameter of Padhye's model in order to improve the RTT-fairness.

- Quality Decision Module (QDM) determines the quality level of the SVC bitstream to be sent according to the available bandwidth, client buffer status and the resolution of the display device. Both the server and the relay device have QDM, because the relay device can be a streaming server when the relay device streams the stored contents in our system.

- Stream Relay Control Module (SRCM) controls messages to switch the display device seamlessly and to notify the status of display device. SRCM in relay device sends notification messages to current and next target display devices to switch the display device. It also collects the buffer status of the current display device.

- Buffer Status Estimation Module (BSEM) estimates the client buffer size and informs it to the QDM to prevent buffer underflow or overflow.

3.1 Estimating Available Bandwidth

In this paper, we present an adaptive rate control scheme. This is based on the estimated available bandwidth and the bandwidth required by the stream relay service. To adapt to the varying network status, the server should have fairly accurate information about the network status. TFRC is proposed for a smooth change in the transmission rate than that in the TCP, while maintaining TCP-compatibility. TFRC uses Padhye's analytical model for the available bandwidth share of TCP connection as shown in Eq. (1). This gives an upper boundary on the transmission rate T in bytes/sec, as a function of the packet size S, round-trip time RTT, steady-state loss event rate p and the TCP retransmit timeout value RTO [9].

$$T = \frac{S}{t_{RTT}\sqrt{\frac{2p}{3}} + t_{RTO}(3\sqrt{\frac{3p}{8}})p(1+32p^2)} \tag{1}$$

To estimate the transmission rate, one of the key issues faced in the TFRC protocol is concerning the response function's specification of the allowed transmission rate. This is inversely proportional to the measured RTT. When sessions share the same bottleneck, they are expected to receive the same share of bandwidth. Thus it achieves the same transmission rate. Unfortunately, this is not the case when the RTTs are very different among the sessions. Total RTT of a network can be presented as follows:

$$RTT_{total} = RTT_q + RTT_{ps} + RTT_t + RTT_{pp} \tag{2}$$

Where, RTT_{total} is the total RTT of a network, RTT_q is a RTT from a queuing delay, RTT_{ps} is a RTT from processing delay, RTT_t is a RTT from transmission delay and RTT_{pp} is a RTT from a propagation delay. Due to the improvement in device's capability, RTT_{ps} and RTT_{ps} are negligible, so the total RTT can be presented as the sum of the RTT_q and RTT_{pp} as follows:

$$RTT_{total} = RTT_q + RTT_{pp} \tag{3}$$

When several sessions share the same bottleneck link, they experience a similar degree of congestion. Therefore, the RTT unfairness problem is mainly caused by RTT_{pp}. To mitigate the effect of RTT_{pp}, we define RTT_{min} to be the RTT of a segment when the connection is not congested. It is generally the RTT of the first segment that is sent by the connection, before the router queues increase due to the traffic generated by these connections. Therefore, RTT_{min} is given by:

$$RTT_{pp} \cong RTT_{min} \tag{4}$$

RTT_{min} is updatedas follows:

$$\begin{gathered} if\,(RTT_{cur} - RTT_{min}) < 0: \\ RTT_{min} = RTT_{cur} \end{gathered} \tag{5}$$

Where, RTT_{cur} is a currently estimated RTT. We calculate RTT form to estimate transmission rate as follows:

$$RTT = RTT_{total} - RTT_{min} + \sigma \tag{6}$$

σ is a constant to decide the responsiveness. Large σ reduces the effect of RTT which is caused by the network congestion and small σ can result in unnecessary oscillations. In our simulation, 2ms shows a good performance. From equation (6), we can mitigate the effect of end-to-end propagation delay.

3.2 Network Adaptive Rate Control Scheme

In the proposed system, the relay device collects the network parameters to estimate the available bandwidth and sends acknowledgements to the server for network information. With the estimated available bandwidth information, the quality decision module on the server will decide up to which layer of the scalable bitstream data will be sent to the relay device. To extract these layers from the bit stream, a bitstream extractor is used.

We define S and Q as the maximum spatial level and SNR level provided by the used bitstream; s and q are the minimum spatial level and the SNR level. We do not adjust the temporal resolution to avoid jerky movements in the output video. R_{ABW} is the estimated available bandwidth and $R_{n,m}$ is the data rate of the spatial-SNR resolution. New spatial–SNR resolution with data rate to adapt to the bandwidth variation will be chosen according to the algorithm described in Fig. 5. The server estimates available bandwidth between server and relay device based on the Equation (1). However, the streaming-based content sharing service requires much higher bandwidth at the access network than the estimated available bandwidth at the server, because the relay device uses bandwidth to relay a streaming service to display device as shown in Fig. 1. Therefore, if the estimated available bandwidth is three times higher than current enhancement layer's data rate, the quality level is increased while the estimated available bandwidth is lower than three times of the next-higher enhancement layer's data rate. On the other hand, if the estimated available bandwidth is equal or lower than three times of current enhancement layers' data rate, the quality level decreased until the data rate of the SVC bitstream is three times lower than the estimated available bandwidth. The server adjusts SNR level first, and then adjust the spatial level.

```
Calculate R_ABW
If ( R_ABW > 3*R_n,m )
    Until (R_ABW < 3*R_n+1,m or R_ABW < 3*R_n,m+1 )
        If (m < Q)
            Increase a SNR layer
        Else
            Increase a spatial layer

Else
    Until (R_ABW ≥ 3*R_n,m)
        If (m > q)
            Decrease a SNR layer
        Else
            Decrease a spatial layer
```

Fig. 5. Network adaptive rate control algorithm

3.3 Device Adaptive Rate Control Scheme

To switch the streaming service that is being provided from one device to another seamlessly, an adaptive rate control scheme is required by considering not only the network status but also the device capabilities. While switching between the display devices, the new spatial-SNR resolution (S', T') with a data rate will be chosen according to **Fig. 6**.

The server estimates available bandwidth between server and relay device based on the Equation (1). Then, the server chooses the maximum spatial level for the corresponding display device. However, if the estimated available bandwidth is lower than the bandwidth required, the server decreases quality level. On the other hand, if the estimated available ban-

Calculate R_{ABW}
$R_{n,m} = R_{S,q}$

If $(R_{ABW} > 3*R_{n,m})$
    Until $(R_{ABW} < 3*R_{n,m+1})$
        Increase a SNR layer
Else
    Until $(R_{ABW} \geq 3*R_{n,m})$
        If $(m > q)$
            Decrease a SNR layer
        Else
            Decrease a spatial layer

Fig. 6. Device adaptive rate control algorithm

dwidth is higher than the bandwidth required, the server increases quality level. The server adjusts SNR level first, and then adjust the spatial level. To provide smoothed playback by preventing buffer underflow or overflow of display device, the proposed scheme controls the quality of the SVC bitstream by estimating the client's buffer state. Fig. 7 describes the client buffer status estimation at the server.

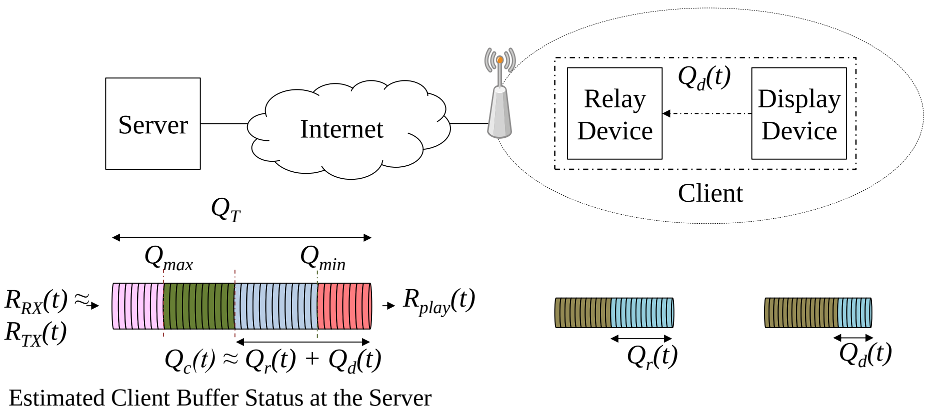

Fig. 7. Estimation of the client buffer status at the server.

As shown in **Fig. 7**, the server regards the summation of the relay device buffer and the display device buffer as a client buffer. In **Fig. 7**, Q_T is the total buffer size, $Q_c(t)$ is the current buffer occupancy, $Q_{max}(t)$ is the maximum threshold and $Q_{min}(t)$ is the minimum threshold, $R_{RX}(t)$ is the input data rate in a client, $R_{TX}(t)$ is equal to the outgoing rate in a server. If we ignore the network transmission delay, $R_{RX}(t)$ is equal to the $R_{TX}(t)$. $R_{Play}(t)$ is the data consuming rate in a client and it is same as encoding rate of video streams. The server can estimate the client buffer state as follows:

$$Q_c(t+1) = Q_c(t) + \{R_{RX}(t) - R_{Play}(t)\} \times RTT \tag{7}$$

$$Q_c(t) \approx Q_r(t) + Q_d(t) \tag{8}$$

After updating the next transmission rate and the video quality, $Q_c(t)$ is recalculated on the basis of the new transmission rate and video quality. In Eq. (7), the server predicts the client buffer state one-RTT-early. The client buffer size can be presented as the summation of the buffer size of relay and display as shown in Eq. (8).This scheme efficiently prevents buffer underflow or overflow of the client buffer with no additional overhead in a high-speed network with longer delay. Depending on the client buffer occupancy, QDM in the server decides the quality level of the SVC bitstream as shown in **Fig. 8**.

```
Estimate Q_c(t+1)

If ( Q_max < Q_c(t+1))
    Until (R_TX < R_n,m)
        If (m < Q)
            Increase a SNR layer
        Else
            Increase a spatial layer

Else If (Q_min > Q_c(t+1))
    Until (R_TX > R_n,m)
        If (m > q)
            Decrease a SNR layer
        Else
            Decrease a spatial layer
```

Fig. 8. Client buffer adaptive rate control algorithm.

If $Q_c(t+1)$ is estimated between 0 and Q_{min}, QDM in the server decreases the quality. To prevent the client buffer overflow in the proposed rate control scheme, the Q_{max}, equation is defined as follow:

$$Q_{max} = Q_T - \int_0^{RTT} (R_{RX}(t) - R_{Play}(t))dt \tag{9}$$

$$Q_{min} = \int_0^{RTT} R_{Play}(t)dt \tag{10}$$

However, the quality level at the display device is not changed immediately because of the buffered data in routers and relay device. To react fast to the buffer status of display device, QDM in relay device controls the video quality according to the status of the display device. If the $Q_d(t)$ of display device is larger than the maximum threshold, the relay device reduces the transmission rate to the bitrate of a lower quality level than the current quality level without decreasing the video quality to prevent a buffer overflow. If the $Q_d(t)$ is smaller than the minimum threshold, relay device reduces the quality level according to the algorithm described in **Fig. 6**.

4. Simulation and Evaluation

This section presents the simulation results for the proposed rate control scheme. In order to evaluate the performance of the proposed scheme, we perform experiments on the basis of the ns-2 (Network Simulator) of LBNL (Lawrence Berkeley National Laboratory) [19]. JSVM (Joint Scalable Video Model) is used to encode the test video clip [20].We constructed a dumbbell topology as shown in Fig. 9. In this simulation, the video servers transmit video streams over the networks. We use two video servers which have different RTTs to show the RTT-fairness of the proposed scheme. The relay device and display device are connected using wireless links. There are two background traffic flows in the simulation. One is FTP traffic transmitted by TCP. The second one is traffic transmitted by other rate control schemes such as TFRC or CUBIC.

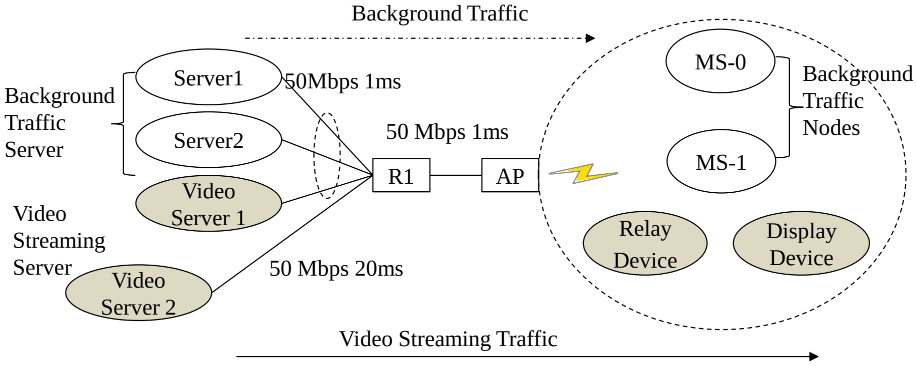

Fig. 9. Network configuration to evaluate the performance of the proposed rate control scheme.

At first, we measure the fairness in sharing the bottleneck bandwidth between the competing two proposed Network and Device Adaptive Rate Control (NAD-RC) flows that have different RTTs. For this experiment, we fix the propagation delay of one flow to 4ms and the other flow to 24ms. **Fig. 10** presents the bandwidth share between two NAD-RC flows.

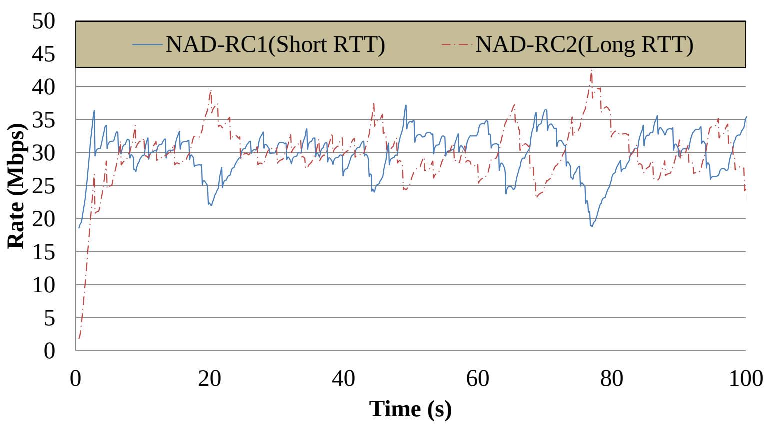

Fig. 10. Estimated sending rate of two NAD-RC with short(4ms) and long(24ms) propagation delay

To compare RTT-fairness with the other rate control schemes, we define the fairness index as a transmission rate ratio between two flows, which use the same protocol, as follows:

$$f = \frac{sending\ rate\ of\ \ flow\ 1}{sending\ rate\ of\ \ flow\ 2} \tag{11}$$

This metric represents a degree of bandwidth shares between two flows of the same protocol. We compared f of NAD-RC with TCP, TFRC and TCP-CUBIC. **Fig. 11** shows the fairness index of the protocols, which have different rate control schemes.

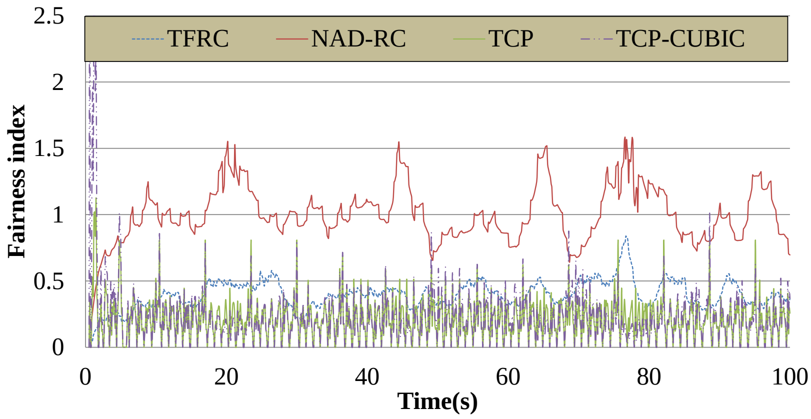

Fig. 11. Comparison of the fairness index between different rate control schemes

From **Fig. 10** and **Fig. 11**, we can see that the proposed NAD-RC share a fair bandwidth than the other rate control schemes.

Another simulation was performed for the Scalable Rate control (SRC) scheme, which is the representative network-aware rate control scheme using SVC, and the proposed rate control scheme in order to compare their performance in terms of packet loss rate, PSNR and the buffer underflow frequency. To evaluate the performance of proposed scheme, the simulation was performed for 100 secs. From 0th sec, the video server sent a video stream. Beginning at the 20th sec, the first TCP flow started. It began competing with the existing video stream for a bandwidth share. This connection finished at the 70th sec. The second TCP flow started at the 30th sec. As shown in **Fig. 12**, the resulting packet loss rate of the SRC is higher than the proposed scheme, as the SRC does not consider the increased bitrate caused by the stream relay and the available bandwidth to adjust the quality level of SVC bitstream.

To demonstrate the impact of the NAD-RC on the smoothness of playback video quality, we compare the number of buffer underflows between NAD-RC and SRC using the buffer driven scheme. From **Fig. 13**, we can see that NAD-RC can achieve high smooth playback quality as NAD-RC reacts rapidly to the buffer status of the display device.

In **Fig. 14**, we present the PSNR of frames received at the display device. It is shown that the proposed rate control scheme achieves a higher PSNR by reducing the packet loss and the buffer underflow of a display device.

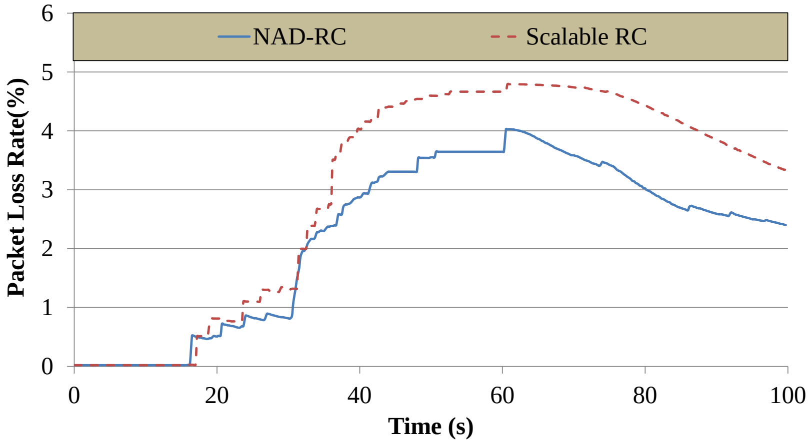

Fig. 12. Comparison of packet loss rate between the NAD-RC and the Scalable RC

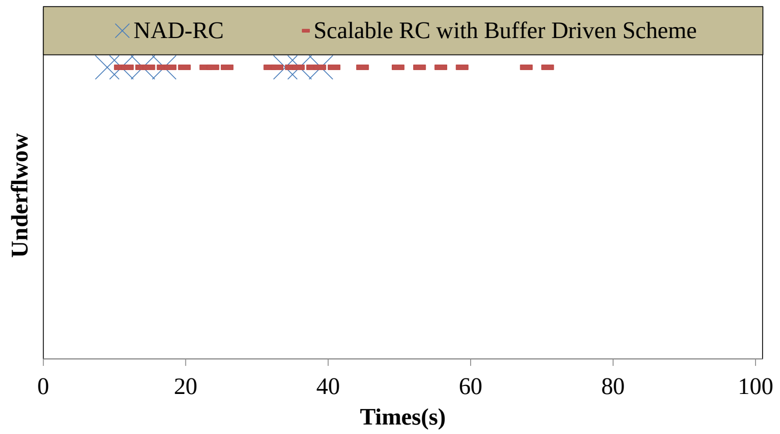

Fig. 13. Comparison of a number of client buffer underflows between the NAD-RC and the Scalable RC using a buffer driven scheme

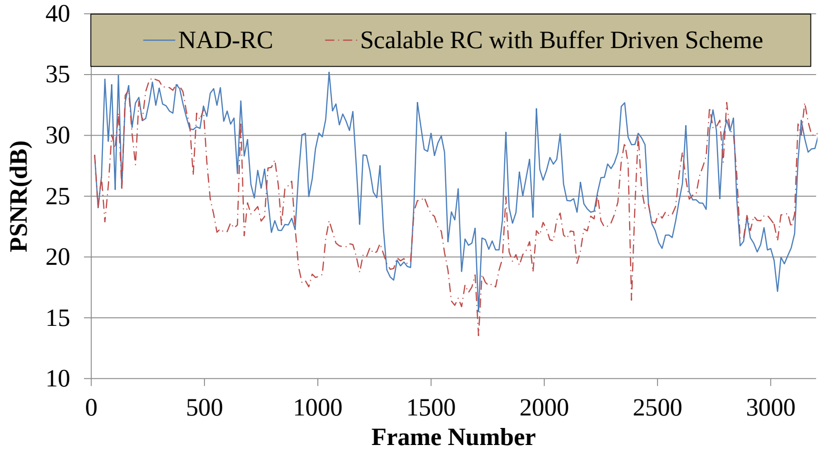

Fig. 14. Comparison of the PSNR between the NAD-RC and Scalable RC

5. Conclusion

To improve the quality of the streaming service in a content sharing service, we proposed a network and device adaptive rate control scheme. The proposed scheme adjusts the quality level of the SVC bitstream to adapt to the time-varying network bandwidth and the required bitrate for a stream relay. Also our scheme adapts quality level of SVC bitstream to the device characteristics such as device resolution and buffer occupancy. In contrast to the previous rate control schemes, such as TCP, our scheme shows good performance with low computational burden by estimating the client buffer state and network status only once per a RTT. The proposed scheme also prevents the quality degradation caused by end-to-end propagation delay.

Simulations show that the proposed scheme can efficiently adapt to the changes in the network and device status. Future work involves research on improving the performance of the adaptive rate control scheme for satisfying the QoE (Quality of Experience) to users.

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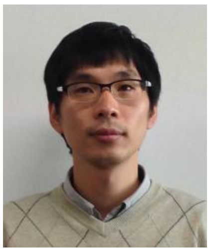

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