Adaptive Rate Control Scheme to Improve QoS of Multimedia Streaming Application

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Abstract—This paper presents an adaptive rate control scheme for streaming-based content sharing service which delivers multimedia contents from a user device to another device or seamlessly redirect streaming service across heterogeneous user devices. In 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 on the device characteristics and bandwidth requirement at the access network. Through the simulation, we prove that our scheme improves the network stability and quality of streaming service by appropriately adjusting the quality of video stream.

Keywords-Quality of Service; Streaming Service; Content Sharing Service

I. INTRODUCTION

In 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 means that the same content is viewed at the same time or in different time by multiple devices. The multimedia streaming has been a noticeable trend for the content sharing service [1]. Streaming-based content sharing services transmit multimedia contents from a user device to another device or seamlessly redirect streaming service across heterogeneous user devices as shown in Fig. 1. There are various devices that are equipped with wireless networking interfaces and most devices have different resolution. In Fig. 1, a user device relays streaming service to another device and it requires having much higher bandwidth at the access network. Also, in the multimedia streaming service, a server transfers the multimedia content to the client as a continuous stream and the client consumes the data as it arrives. Such multimedia streaming has stringent timing requirements for data consumption. 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 on timely delivery. This mismatch forms the fundamental challenge in streaming video over the Internet [2].

However, by considering only the network stability, previous works does not consider the characteristics of multimedia streaming application. The client may suffer from interrupt of video playback because of the buffer underflow [3]. Also, in streaming-based content sharing service, the more bandwidth is required to relay streaming service. Furthermore, it is not easy to support cross-device handover, which dynamically and seamlessly switches between display devices on demand. A mobile device may often not have the capability to receive services with the same quality as a high-performance PC. In such cases, quality of the multimedia content has to be adjusted based on the capabilities of the mobile device [4]. These problems can be solved by using scalable video coding which provides a bit stream with a layered structure, including 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 is. Scalable video coding (SVC) is a convenient solution to adjust the data rate to characteristics of diverse device and varying bandwidth in the Internet [5].

In this paper, we propose an adaptive rate control scheme to improve the quality of service (QoS) for streaming-based content sharing service. The proposed rate control scheme adjusts the quality level of SVC bitstream based on the available bandwidth, required bandwidth to relay streaming service, device resolution and client buffer status. Therefore, the proposed scheme improves the network stability by reducing the packet loss. It also provides the 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 and 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 work.

II. RELATED WORKS

Multimedia streaming addresses the problem of transferring multimedia data as a continuous stream. With streaming, the end-user can start displaying the video data or multimedia data before the entire file has been transmitted. To achieve this, the bandwidth efficiency and flexibility between streaming servers and device of end-users are very important and challenging problems. In response to such challenges, SVC has been proposed.

The JVT (Joint Video Team) of the ITU-T VCEG (Video Coding Experts Group) and the ISO/IEC MPEG (Moving Picture Experts Group) have standardized a SVC extension of the H.264/AVC standard [6]. SVC enables the transmission and decoding of partial bit-streams to provide video services with lower temporal or spatial resolutions or reduced fidelity while still retaining a reconstruction quality that is high relative to the rate of the partial bitstreams. However, this is insufficient because the Internet provides the best-effort service and has a time variant characteristic. Moreover, it is not explicitly optimized considering the specific characteristics of multimedia streaming services.

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 [5], transmits the maximal bitrate according to an estimated available bandwidth. This system adjusts the quality level of the SVC bitstream to adapt the transmission rate to a varying bandwidth. This system, however, has an underflow problem, that does not support the continuity off media playback because it does not consider the status of the client buffer.

Unlike the network adaptive rate control scheme discussed above, a buffer-driven scheme, which is proposed in [3], scales video quality and schedules data of transmission based on client buffer occupancy and server buffer occupancy. Therefore, the buffer-driven scheme provides smoothed playback by preventing buffer underflow or overflow. More in details, 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 the unstability and unfairness from the UDP-based streaming protocols because it has no concern about network situation.

The proposed scheme in [7] includes more sophisticated features that consider both network and user requirements. From the network viewpoint, the proposed mechanism estimates the TCP-friendly rate suitable for network status. Therefore, it is able to adjust the sending rate of video stream in a TCP-friendly manner and improves the network stability by reducing the packet losses. From the user viewpoint, this scheme provides the smooth playback by preventing the receiver buffer underflow or overflow.

Recent paper by Koo et al. [8] proposes 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, that is 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. The flow control prevents the server from overwhelming the client such that in over buffered and under buffered conditions. In the NCAR, the flow control adjusts the quality level of the transmitting SVC bitstream based on the estimated value of the client-aware information, such as the buffer occupancy ratio. However, these schemes do not consider characteristics of device. In recent multimedia streaming applications such as content sharing service, it is important to support various types of devices with a wide range of capabilities, features, and characteristics.

III. ADAPTIVE RATE CONTROL SCHEME TO IMPROVE QoS OF MULTIMEDIA STREAMING APPLICATION

Fig. 2 shows the proposed streaming system for content sharing service. Our system consists of server, relay device and
display device. The server streams a multimedia content to relay device and 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 multimedia stream as a display device. In proposed system we use the RTP and RTCP protocols for delivery of video data and feedback on network quality. We also use the RTSP protocol for establishment and control of media streams. There are three closely interacting modules in our system.

- 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 display device. Both the server and the relay device have QDM, because relay device can be a streaming server when relay device streams stored contents in our system.

- Stream Relay Control Module (SRCM): controls messages to switch 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 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.

A. Network Adaptive Rate Control Scheme

In this paper, we present an adaptive rate control scheme which 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. In proposed system, the relay device collects the network parameters to estimate the available bandwidth and sends acknowledgements to server for network information. The server uses Padhye’s analytical model for the available bandwidth share of TCP connection as shown in Eq. (1) which gives an upper boundary on the transmission rate $T$ in bytes/sec, as a function of the packet size $S$, round-trip time $t_{RTT}$, steady-state loss event rate $p$, and the TCP retransmit timeout value $t_{RTO}$ [9].

$$T = \frac{S}{t_{RTT} + t_{RTO}(3.0 + \frac{3}{8}p(1 + 32p^{2}))}$$

With the estimated available bandwidth information, the quality decision module on the server will decide up to which layer of the scalable bit stream 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 bit stream. $s$ and $q$ are the minimum spatial level and the SNR level. We do not adjust temporal resolution to avoid jerky movements in the output video. The estimated available bandwidth and $R_{n,m}$ is the data rate of 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. 3. The quality level is increased when the available bandwidth is three times higher than the next-higher enhancement layer’s data rate. On the other hand the quality level is decreased until the data rate of SVC bitstream is three times lower than the available bandwidth.

B. Device Adaptive Rate Control Scheme

To switch streaming service being provided from one device to another seamlessly, an adaptive rate control scheme is required by considering not only network status but also device capabilities. When switching between the display devices, the new spatial-SNR resolution $(S', T')$ with data rate will be chosen according to Fig. 4. The proposed algorithm chooses the maximum spatial layer for corresponding display device. However if the estimated network bandwidth is lower than the bandwidth required, quality level is decreased from SNR layer.

Also, to provide the smoothed playback by preventing buffer underflow or overflow of display device, the proposed scheme controls the quality of SVC bitstream by estimating the client’s buffer state. Fig. 5 describes the client buffer status.

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Calculate $R_{ABW}$

if ($R_{ABW} < 3*R_{n,m}$) until ($R_{ABW} > 3*R_{n,m}$)
    if ($m > q$)
        Decrease a SNR layer
    else
        Increase a SNR layer

if ($R_{ABW} < 3*R_{n,m}$) until ($R_{ABW} > 3*R_{n,m}$)
    if ($m > q$)
        Decrease a spatial layer
    else
        Increase a spatial layer

Fig. 4. Device adaptive rate control algorithm.
As shown in Fig. 5, the server regards the summation of the relay device buffer and the display device buffer as a client buffer. In Fig. 5, $Q_T$ is the total buffer size, $Q_c(t)$ is the current buffer occupancy, $Q_{\text{max}}$ is the maximum threshold and $Q_{\text{min}}$ is the minimum threshold, $R_{\text{RX}}(t)$ is the input data rate in a client, $R_{\text{TX}}(t)$ is equal to the outgoing rate in a server. If we ignore the network transmission delay, $R_{\text{RX}}(t)$ is equal to the $R_{\text{TX}}(t)$. $R_{\text{play}}(t)$ is the data consuming rate in a client, as same as encoding rate of video streams. The server can estimate the client buffer state as follow:

$$Q(t+1) = Q(t) + (R_{\text{RX}}(t) - R_{\text{play}}(t)) \times \text{RTT}$$ \hspace{1cm} (2)$$

$$Q_c(t) = Q_{\text{max}} + Q_{\text{min}}$$ \hspace{1cm} (3)

After updating the next transmission rate and video quality, $Q_c(t)$ is re-calculated on the basis of the new transmission rate and video quality. In Eq. (2), the server predicts one-RTT-early the client buffer state. The client buffer size can be presented as the summation of the buffer size of relay and display as shown in Eq. (3).

This scheme efficiently prevents buffer underflow or overflow of client 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 SVC bitstream as shown in Fig. 6. If $Q_{c}(t+1)$ is estimated between 0 and $Q_{\text{min}}$, QDM in server decrease the quality level of SVC bitstream without decrease of transmission rate to prevent buffer underflow. If $Q_{c}(t+1)$ is larger than $Q_{\text{max}}$, the quality level is increased without increase of transmission rate to prevent the buffer overflow. We adjust the SNR level first to support maximal resolution of the display device.

To prevent client buffer overflow in proposed rate control scheme, the $Q_{\text{max}}$ equation is defined as follow:

$$Q_{\text{max}} = Q_T - \int_{0}^{\text{RTT}} (R_{\text{RX}}(t) - R_{\text{play}}(t))dt$$ \hspace{1cm} (4)

Also, to prevent client buffer underflow, $Q_{\text{min}}$ is defined as follow:

$$Q_{\text{min}} = \int_{0}^{\text{RTT}} R_{\text{play}}(t)dt$$ \hspace{1cm} (5)

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 display device. If the $Q_d(t)$ of display device is larger than maximum threshold, relay device reduces transmission rate to the bitrate of a lower quality level than current quality level without decrease of video quality to prevent buffer overflow. If the $Q_d(t)$ is smaller than the minimum threshold, relay device reduces quality level according to Fig. 6.

### IV. SIMULATION AND EVALUATION

This section presents simulation results for the proposed rate control scheme. In order to evaluate the performance of proposed scheme, we performed experiments on the basis of the ns-2 (Network Simulator) of LBNL (Lawrence Berkeley National Laboratory) [10]. JSVM (Joint Scalable Video Model) is used to encode test video clip [11].

The NS simulation was performed for the existing rate control (RC) scheme, which is the network-aware rate control scheme using SVC, and the proposed scheme to compare their performance in terms of packet loss rate, PSNR and buffer underflow frequency. The simulation topology is shown in Fig. 7.

To evaluate the performance of proposed scheme, the simulation was performed for 100 sec. from 0th sec, the video server sent a video stream. Beginning at the 20th sec, the first TCP flow started and began competing with the existing video stream for bandwidth share. This connection finished at the 70th sec. The second TCP flow started at the 30th sec. As shown in Fig. 8, the resulting packet loss rate of the existing RC is higher than the proposed scheme, because the existing RC does not consider the increased bitrate caused by the stream.
relay and available bandwidth to adjust the quality level of SVC bitstream.

Fig. 9 demonstrates the impact of the proposed scheme on the smoothness of playback video quality, we compare the number of buffer underflow between proposed RC and existing RC using buffer driven scheme. From Fig. 9, we can see that our scheme can achieve high smooth playback quality because proposed RC reacts fast to the buffer status of display device.

In Fig. 10, we present the PSNR of frames received at the display device. It is shown that proposed rate control scheme achieve a higher PSNR by reducing packet loss and buffer underflow of display device.

V. CONCLUSIONS

To improve the quality of streaming service in 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 required bitrate for 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 previous rate control schemes, our scheme shows good performance by considering both network and device characteristics. Simulations show that the proposed scheme can efficiently adapt to changes in the network and device. Future work involves research on improving the performance of the adaptive rate control scheme under satisfying the QoE (Quality of Experience) to users.

REFERENCES


