A Control-Theoretic Method for Rate-Based Flow Control of Multimedia Communication

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Abstract. In this paper, based on the feedback of buffer occupancy (BO), a new control-theoretic flow control mechanism is proposed. Our mechanism applies feedback control to keep BO running to a given level to increase the time available for packet recovery without violating real-time requirements. It tightly couples gap-based loss detection and packet retransmission to provide effective and efficient error control. Therefore, reliable communications and optimized playback QoS can be achieved at a minimal cost. Moreover, our implicit prediction and prevention for buffer underflow and overflow can help clients with a limited buffer (such as set-top box, PDA and cellular phone) to achieve acceptable playback QoS. We have deployed the proposed mechanism into a true VOD service in an ADSL trial. Experiments show that the proposed mechanism outperforms the previous mechanisms. The proposed mechanism contributes an innovative way to resist the network uncertainty with minimal overhead.

Keywords: *feedback control, buffer-occupancy, control theory, error control, video on demand.*

1. Introduction

The bandwidth of a public network backbone is shared by all kinds of applications to achieve statistical multiplexing gain. It introduces considerable uncertainty in workload and resource requirements. While delivering packets through such a network, the unpredictable delay jitter may introduce underflow or overflow in a limited client buffer even if the network is error-free at a time period (as illustrated in Figure 1). However, different from conventional text/image network applications, multimedia applications require end-to-end quality-of-service (QoS) with jitter-free playback of audio and video. Thus a good end-to-end flow control mechanism is needed to maintain high throughput and keeping average delay per packet at a reason level for such time-critical applications like multimedia communications.

Rate-based and window-based mechanisms are the two of the best-known candidates for flow control. Though window-based flow control can help to avoid buffer overflow, it does not regulate end-to-end packet delays well and does not guarantee a minimum data rate for guaranteed-QoS. On the other hand, rate-based flow control can provide end-to-end deterministic and statistical performance guarantees over modern packet-switching networks [1]. Its short propagation delay has brought wide deployment of the rate-based flow control in real-time multimedia applications.

Usually, the rate adjustment is performed by the intensive feedback controls from client to achieve guaranteed-QoS. Such a feedback control idea has been used successfully in computer systems to control the unpredictable workload. In this paper, based on the control theory, a new rate-based flow control mechanism is proposed with the feedback of buffer occupancy (BO). The proposed mechanism applies feedback control to keep BO running to a given level away from buffer overflow and underflow. Because of the stability of proposed control function and the caution of two guarded thresholds, the control overhead is low and would not scarify the playback QoS. Our mechanism can increase the availability of time for packet recovery without violating the real-time constraints of playback. By coupling the gap-based loss detection and packet retransmission, error control can be achieved effectively at low overhead of retransmission acknowledgements.

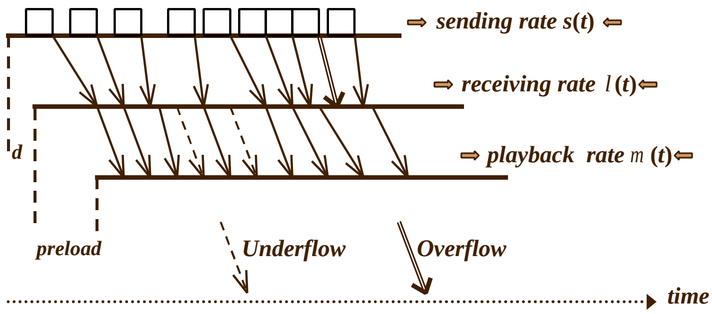

Fig. 1. Buffer underflow and overflow from unpredictable jitter.

Recent literatures [6-7] have shown that the control theory can be successfully applied to rate control in video coding and high-speed network infrastructure respectively. They demonstrated the effectiveness of feedback control mechanisms in achieving predictable system performance without precise knowledge of worst-case load patterns. In this paper, we extend this idea on the unpredictable network performance of real-time multimedia communications. It takes much more challenge than the conventional problem that deploys local buffer occupancy in the smoothing algorithm of the MPEG stream encoder [6]. The four objectives that the proposed mechanism targets to meet is shown in the following:

- *minimize fluctuations of buffer occupancy* (*i.e. minimize the fluctuation between observed BO* $b(t)$ *and target BO* b_m).
- *maximize the possibility to recover the lost packet by retransmission in error control.*
- *avoid sending unnecessary requests of feedback control and retransmission due to the communication overhead.*
- *considering the friendliness behavior of rate adjustment to the network traffic.*

The remainder of this paper is organized as follows. The related work of rate-based feedback control is presented in Section 2. In Section 3, we introduce the feedback control mechanism and the linear proportional and derivative (PD) control function. Experimental results of our proposed method compared with other method are shown in section 3. In section 4, we analyze the proposed scheme by mathematical proof. Concluding remarks and future works are given in the last section.

2. Related Work

Recent literatures [2-5] proposed the rate adjustment schemes to consider TCP friendliness for the congestion condition. Their adaptive rate adjustments are basically based on the feedback of packet loss and delay indicated in the periodic RTCP packet of the RTP protocol suite [13]. Their AIMD (Additive Increase/Multiplicative Decrease) rate adjustment discipline (i.e. rate control function) consider packet loss ratio, utilization and fairness to perform the actual rate change dynamically to achieve maximal communication throughput. However, achieving maximal communication throughput does not explicitly indicate the playback quality will be optimal. The buffer underflow and overflow introduced by these schemes may jeopardize the playback QoS for clients that do not preserve much system resource (such as set-top box, PDA and cellular phone).

To keep BO running away from underflow and overflow, Sun et al. [9] proposed an adaptive flow control based upon client's queue length. The packet sending rate is derived from a formula based on timely measured packet delay, previous sending rate, average decoding rate, packet loss ratio and target queue length. The rate control period is a round trip time based on the average one-way delay and standard deviation measured in the previous period. Note that, this formula is quite complicated to derive. Although this scheme has considered the dynamic of networks, it assumes the playback rate is a constant and ignores the realistic impact of variance in playback rate to the running BO. Therefore, it may introduce inevitable rate and BO oscillations. While the longer the control period sustains, the higher possibility that the BO may run far away from the target queue length and may result in buffer overflow or underflow.

In [10], a condition-ARQ rate control algorithm is proposed over wireless channel under the buffer constraints to achieve maximal end-to-end channel utilization with smooth playback QoS. Though they claim the control scheme is simply the function of BO only, the algorithm necessitates embedding coding to control the playback rate at different video quality perceived instead of affecting the sending rate. The sender infers instantly the channel condition and current BO of the receiver from previous ARQs to decide either to transmit another packet of the current video frame for better playback quality or to transmit a packet of next video frame for speeding up the playback rate at receiver. To compensate the buffer underflow and overflow, the perceived playback QoS will be degraded smoothly because of the embedded coding. Thus, they can not guarantee the original playback QoS. Moreover, it’s not cost-effect to deploy such a codec-dependent algorithm concerning the overhead of acknowledgements of stop-and-wait ARQ while coding data will traverse series of routers to client.

Therefore, without constraint of coding scheme, we explore further for an aggressive and fine-grain control function of only BO feedback to alleviate the dynamics along the transmission path to the user at client (e.g. network delay, variable playback rate, etc.). The control message is triggered to acknowledge only while BO running away from the interval of two guarded thresholds. Moreover, based on the rate adaptation function for running the BO at target threshold, our mechanism can maximize the possibility that the lost packets can be recovered before the retransmitted packets miss the playback deadline. Thus, an economical and effective error control mechanism is furnished to provide reliable real-time multimedia communication service to keep the original playback QoS at its best.

Attention to achieve the optimized error control based on the fine-grain position control at playback buffer is a salient feature of our study. This issue for optimized error control upon BO using control theory has not been addressed in other related work.

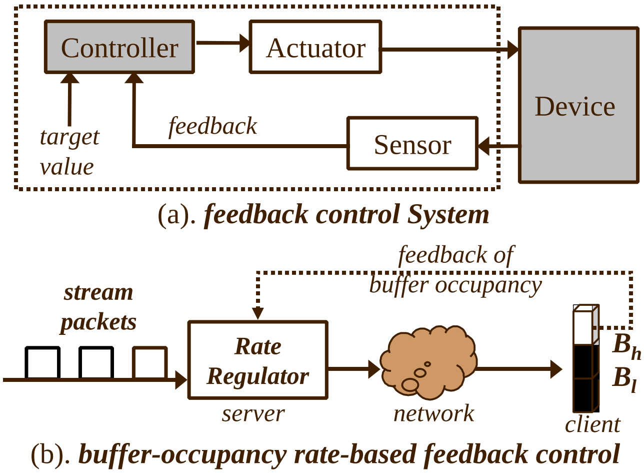

Fig. 2. A control theoretic feedback control system.

3. Control Theoretic Rate-based Flow Control with Buffer-Occupancy Feedback

In this section, firstly the applied control theory will be simply introduced and we describe the proposed buffer-occupancy rate-based control mechanism in detail. Secondly, we will present the applied PD control function. Finally, we continue to the retransmission mechanism based on our rate adaptation scheme.

3.1 Buffer-occupancy rate-based control (BORC)

In control theory, the feedback control system as shown in Figure 2(a) can be described by a device to be controlled, an actuator and a sensor. The system will timely monitor and compare error between the current value and the target value of the sensor. If necessary, the controller will perform a feedback function to the system based on this error. An adaptive algorithm is introduced to support applications in dynamic and uncertain environments. In our corresponding control system, the block of the device in Figure 2(a) is the buffer-occupancy we try to control. The sensor is a mechanism that observes the BO and will send feedback to the controller. Then the controller will perform the proposed control function and ask the actuator to adapt the actual sending rate.

The feedback information is needed to provide the network dynamics and then to predict the workloads and to support a good rate regulator as shown in Figure 2(b). The running BO of the client results from not only the elastic traffic introduced by the rate adaptation but also the playback rate while the client decoding the received packets. Therefore, the proposed rate-based feedback control mechanism must be adapted to the variations of server sending rate, client receiving rate and client playback rate (as shown in Figure.1) to keep the playback QoS intact at its best.

Note that in multimedia communications, each client has only a limited buffer. If the buffer is flooded, then the incoming packets are discarded. On the other hand, if the buffer is underflow, the display is then temporally in freeze. They will jeopardize the playback QoS. In this paper, we regard the rate regulator as a controller, and use buffer's occupancy to predict its possible overflow/underflow. In this setting, the target values of a sensor are thresholds of BO. In this paper, we use the low threshold b_l and the high threshold b_h of BO to guard against buffer overflow and underflow. Let the target value of BO be b_m where b_m is simply defined as the middle point of b_l and b_h. Based on the feedback control theory, at any time t, our algorithm controls the BO $b(t)$ to be targeted at b_m. Therefore, the controller can prevent buffer overflow/underflow to achieve satisfactory playback quality. To save the communication overhead of timely feedback control, client can send the feedback control to server only when the BO does not run within the $[b_l, b_h]$ interval. Statistically, two buffer intervals $(b_h, b_{Max}]$ and $[b_0, b_l)$ will play an working area that can sustain the integrated error from the proposed PD rate control function where the b_{Max} represents the buffer size. That is the reason why PID (Proportional, Integral, Derivative) control is not considered here because of the proposed PD rate controller with two additional guarded thresholds is good enough to predict and prevent the buffer underflow and overflow at minimal cost.

Two terms are raised here as the criteria to judge the performance to achieve the first one of four objectives aforementioned in section 1 to minimize the fluctuations of BO. The first term is defined as “threshold miss” (*T.M.*) while the receiving packet did not arrive in the position within [b_l, b_h]. The second one is defined as “serious miss” (*S.M.*) while the client buffer was in the condition of overflow or underflow. We will utilize these terms in the experiments discussed in the section 4 to measure the performance.

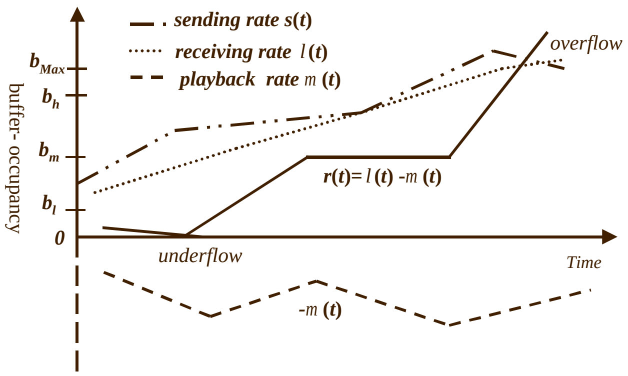

Fig. 3. Illustrated curves of rate functions will affect the BO at client.

3.2 Stabilized control function

For the feedback controller, many possible designs are available. We focus on a linear proportional and derivative (PD) control [10] for its theoretic stability and practical values. Because the only proportional control of rate adaptation may result in rate oscillations theoretically, it won't reach stable state to maintain target level of BO and the playback quality.

Note that, as shown in Figure 3, the running position of BO depends on not only its current value $b(t)$ but also the rate-difference $r(t)$ between packet receiving rate $l(t)$ and packet playback rate $m(t)$. However, the server sending rate $s(t)$ will also effect the $l(t)$ because of the network dynamics. The network dynamics impact to the difference between $s(t)$ and $l(t)$ will be discussed in detail later in the section 5. Thus we can define the rate-difference $r(t)$ which affects the running BO $b(t)$ as follow in formula (1) where Δt is the control period, $\Delta r(t)$ in formula (2) indicate the rate adjustment upon the rate-difference $r(t)$:

$$r(t) = \frac{db(t)}{dt} = \frac{b(t) - b(t - \Delta t)}{\Delta t} = l(t) - m(t) \tag{1}$$

$$\Delta r(t) = \frac{r(t) - r(t - \Delta t)}{\Delta t} \tag{2}$$

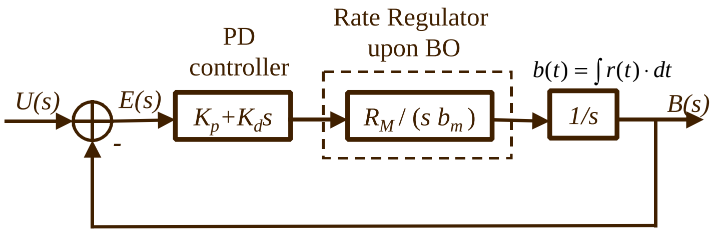

Fig. 4. Block diagram of the proposed BO feedback control system, $U(s)$ is the *Laplace* transform function of input function (e.g. the target BO), and $E(s)$ is the difference between $U(s)$ and the output transform function $B(s)$ (e.g. the observed BO).

Before we figure out the formula of the PD control function of the diagram illustrated in Figure 4 for the proposed control mechanism, we should consider some criteria in which the PD control function should meet in practice as follow:

- *prevent the jitter and congestion by smoothing the rate change with friendly behavior.* (*i.e. The given constant R_M represents the maximum rate adjustment we expect*).
- *the proportional part of the PD control function should indicate that the rate change must be linearly dependent on the difference between the target BO b_m and the current BO.* (*i.e. R_M/b_m is the proportional coefficient for the control function. Therefore, the K_p in PD controller is simply set to 1*).

- *the derivative part of the PD control function will not only stop the oscillations introduced by the proportional part to help the stability but also contribute to the rate difference.*(*i.e. the* K_d *coefficient for derivative part. We defer the discussion of optimal* K_d *to section 5*).

The first and second criteria indicate our intuitive rate regulator upon BO as shown in Figure 4. The BO feedback control system will achieve stable state because of the compensation of PD controller as indicated by the third criterion. After investigating the above criteria and the diagram shown in Figure 4, we can simply derive the discrete rate-adjustment control function with PD compensator as follows:

$$\Delta r(t) = \frac{R_M}{b_m}\left(bm - b(t) - K_d\left(\frac{b(t) - b(t-\Delta t)}{\Delta t}\right)\right) \qquad \mathbf{(3)}$$

The idea curves for rate difference $r(t)$ and BO $b(t)$ controlled by the proposed function are all illustrated in Figure 5 while the initial conditions of $b(0)$ and $r(0)$ are all zero. The mathematical results conducted from the continuous case of the PD control function.

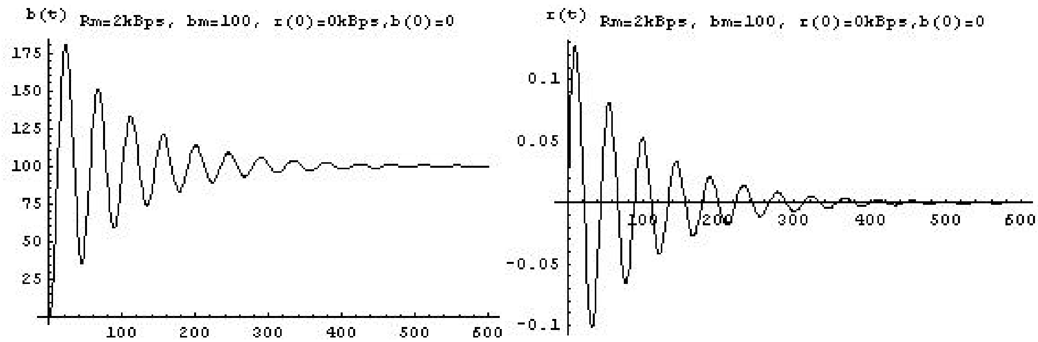

Fig. 5. Ideal curves of *b*(*t*) and *r*(*t*) for the BO feedback control function.

3.3 Efficient error control

Considering the efficiency of the error control of packet loss for reliable multimedia communications, the proposed retransmission mechanism with gap-based loss detection is simply illustrated in Figure 6. Each arrival packet will carry with sequence number to indicate the playback sequence. Therefore, while there is a gap between current arrival sequence number and the number the client currently expects, this detection of packet loss will trigger the retransmission request immediately. Besides, the decoder at client will continue to consume the packets in buffer in a FIFO manner as long as the buffer is not empty. After sending out the retransmission request for loss packet, the arrival time of packet in retransmission may miss the playback deadline due to the round trip delay (RTD) in communication. However, thanks to the stable control for BO in the proposed mechanism and observed RTD, the client can decide to issue the request or not for eliminating the communication overhead. If the real-time constraint for multimedia playback is allowed, the higher the BO running the more valid packets of retransmission will be received to playback at client. Thus, the multiple retransmissions of lost packet are applicable in our proposed scheme to achieve optimized playback QoS.

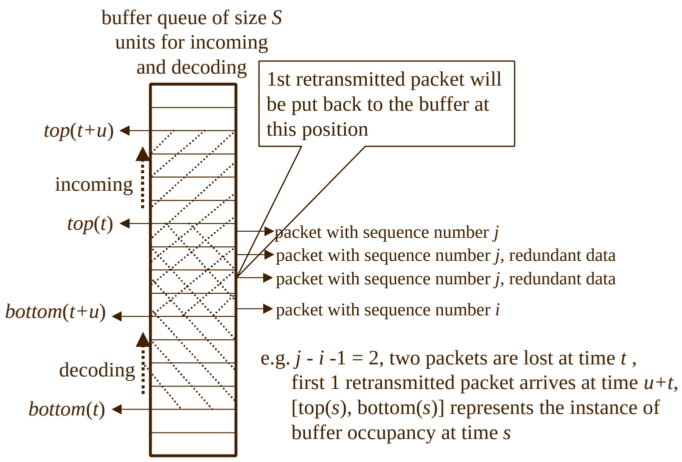

Fig. 6. Gap-base loss detection and packet retransmission.

Besides, more than one consecutive packets are lost in network can be considered as an indication of congestion. At the mean time, the multiplicative decrease (MD) part of the AIMD rate adaptation can be applied to help our scheme to achieve friendliness behavior [4][5].

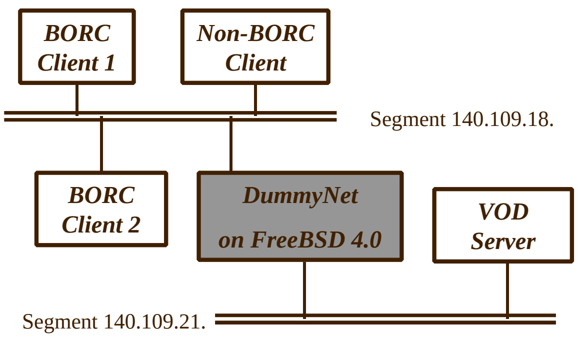

Fig. 7. Test bed.

4. Experiments and Performance Results

In this paper, the testbed is based on a previous work [8] of true VOD (Video-On-Demand) services in our ADSL trial. As shown in Figure 7, our clients are running on a different network segment with the server. A router in which a Dummynet simulation tool [12] is running then connects these two network segments. By applying this tool, we can approximate the real network model to simulate different effects of finite routing queue, bandwidth limitation, communication delay and packet drop rate. In the following, detailed experiments are shown to evaluate the performance of the proposed control scheme for BO.

4.1 Experimental background

In the test bed, there are three set-top boxes of clients are running. Two of them are the BORC (Buffer-Occupancy Rate Control) clients using our proposed mechanism. The last one is a non-BORC client. The server performs the discrete control function in formula (3) while receiving the BO feedback from BORC clients. While receiving the feedback of loss and delay from non-BORC client, the server performs the control function of TCP-friendly rate adaptation mechanism proposed in [5]. In the TCP-friendly rate adaptation mechanism, the server will pick up the minimal one among the values derived from the rate-change-smoothing function, the exponential-rate-limit function and the TCP-bandwidth-share function [5] to increase the rate while no congestion.

Much like the period of RTCP [13], the non-BORC client sends out the feedback control while it receives every 500 packets or more. It usually takes 2 to 3 seconds to send out the control message. BORC clients check for their buffer-occupancies after receiving every 100 packets. Considering the overhead of the feedback control mechanism including communication and computation overheads at the client, one of the BORC clients would send out feedback control to the server only when the buffer runs either above the high threshold b_h or below the low watermark b_l. The other BORC client would send out the feedback control while receiving every 100 packets.

Note that the system dynamic may be changed at any time. For fair comparisons, we need to make sure the system dynamics applied in different tests are the same. In our experiments, we first assume that the same content of media streams are requested to serve at the same time for the clients. The media content applied is MPEG-1 file of the motion picture "Star War, Episode I". We take 20 minutes length of the movie with size 206577K bytes in the experiments. The packet size segmented for delivery is 1K byte. The arrival packet will be accommodated in a buffer unit at client. In the testing period, we recorded the variations of the sending rate and the running BO according the different deployed rate control functions. The maximum sending rate of the VOD server is 183024 bytes/s (it is measured from the 1.5 Mbps bandwidth of the ADSL downstream that simulated by the Dummynet).

All the BORC client and non-BORC clients are set-top box machines with the same limited buffer size of b and b equals to a very small size of 200 buffer units (i.e. 200K bytes). At first, we assume the chance of overflow and underflow will be the same after BO reaching steady state at b_m and requiring no further rate control, so b_m is set to the middle of buffer. Then, we keep the distance from overflow to b_h same as the distance from underflow to b_l to equalize the possibilities to predict and prevent the overflow flow and underflow respectively. Therefore, b_m is set to the middle point of b_l and b_h. At last, considering the trade-off between the effectiveness of prediction and the control cost of prevention, the low watermark b_l and high watermark b_h are simply set to $b/4$ and $3b/4$ respectively in our experiments.

4.2 Performance results and evaluation

To measure our mechanism for the performance, we have already defined two terms *S.M* and *T.M.* in section 3.1. The experimental result as illustrated in Figure 8 shows that the *T.M.* and *S.M.* ratios of a BORC and the non-BORC clients experiencing the delays of 0ms, 100ms, 500ms and 1 second. The results indicate that all the miss ratios of *T.M.* in BORC client are less than 2.3% and especially much less than the ratios in non-BORC client. It also indicates that our proposed rate control scheme with BO feedback outperforms [4] in controlling BO to within the designated size constraints.

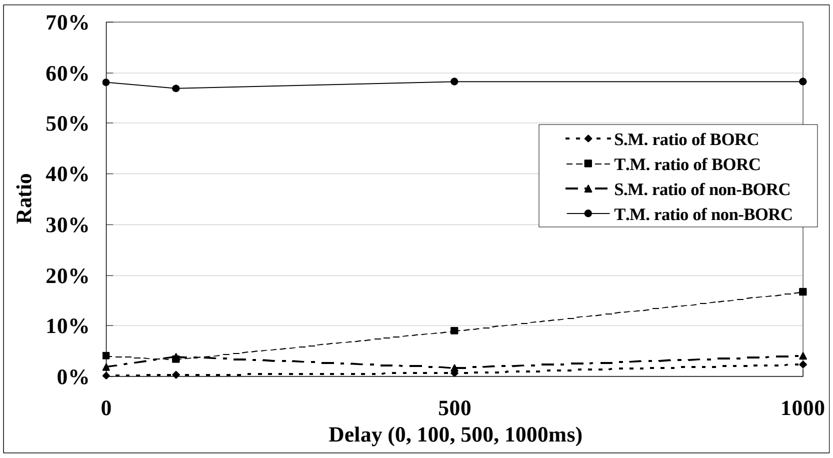

Fig. 8. S.M. & T.M. ratios for BORC and non-BORC clients under different network delay.

Besides, the playback quality index *S.M.* of BORC client approaches to the ratio of non-BORC client while the delay is longer. After studying Figure 9 and Table 1, buffering underflow actually dominates in our performance study. We observed that most the buffer underflow occurs at the beginning of streaming. Unfortunately, the feedback control mechanism is not yet stable for the preloading phase. Non-BORC client is also unstable at that time. However, non-BORC client suffers from much more overflow than does BORC clients do as shown in the experiments. Therefore, the BORC client outperforms the non-BORC client under all the measured criterions proposed in the experiments.

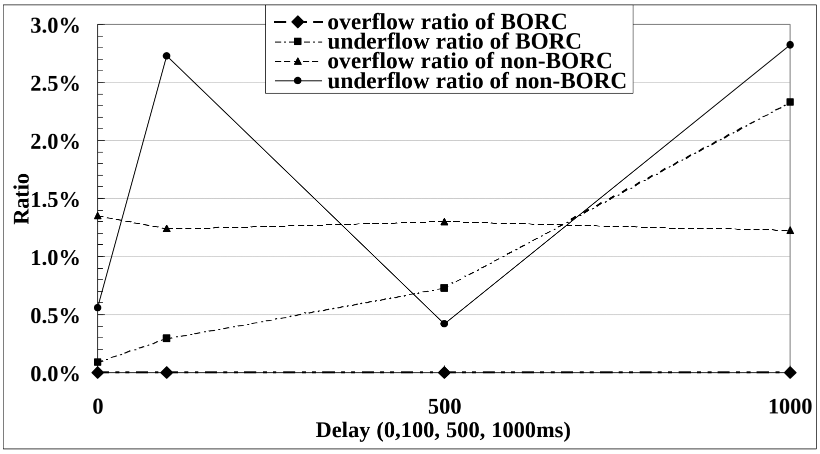

Fig.9. Overflow & underflow ratios for BORC and non-BORC clients under different network delay.

Table 1. Counts for overflow and underflow under different network delay

	0ms		***100ms***		***500ms***		***1000ms***	
	BORC	***Non-BORC***	***BORC***	***Non-BORC***	***BORC***	***Non-BORC***	***BORC***	***Non-BORC***
Overflow	3	2800	4	2572	5	2688	6	2539
underflow	185	1147	614	5635	1500	866	4811	5836
Total	188	3947	618	8207	1505	3554	4817	8375
S.M. ratio	*0.001*	*0.019*	*0.003*	*0.04*	*0.007*	*0.017*	*0.023*	*0.041*

Moreover, as we studied the traces of BO and rate changes illustrated in the Figure 10-11, we may conclude that BO is controllable even under the long network delay with less communication overhead of feedback controls than the other without BO feedback control. The experimental results indicate that our predictive rate control mechanism of BO feedback control maintains low *T.M.* ratio to prevent the degradation of playback QoS for resultant serious miss ratio (i.e. *S.M*).

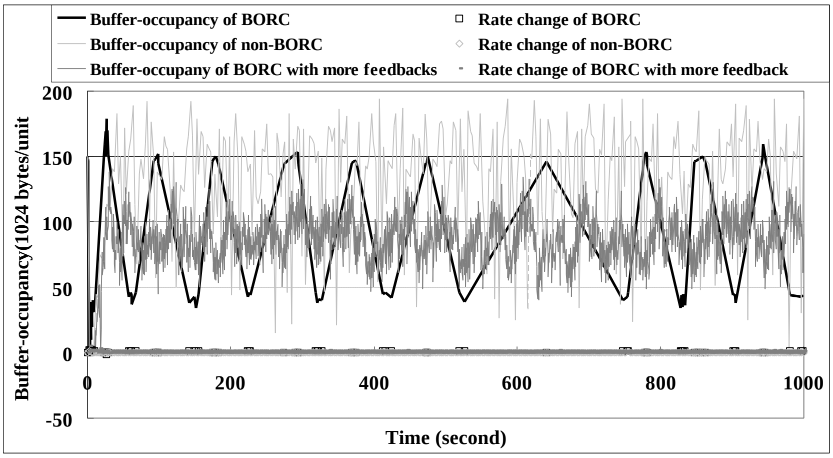

Fig. 10. Trace of BO for network delay from 0 to 10 ms. The darkest line indicates optimized BO control under minimal feedback control frequency. The lightest one shows the inevitable oscillations without BO control. The darker line shows the BO trace with more frequent BO feedback controls.

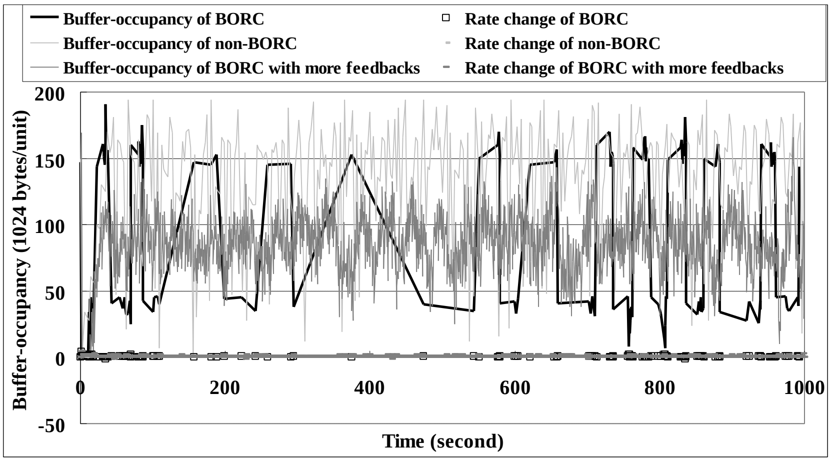

Fig. 11. Trace of BO for longer network delay from 500 ms to 1 second. More BO feedback control frequency is needed for longer network delay. But the darkest line still outperforms the lightest line for the perceived QoS at minimal controls.

5. Analysis of Impact of System Dynamics to Buffer-occupancy

While analyzing the impact of system dynamics to the BO, the stability of proposed control function and the derivative K_d value should be considered at first in this section. As shown in Figure 4, the rate regulator with PD controller indicates a second order system in formula (3). And then we denote the Laplace transform function *G(s)* as shown in formula (4),

$$G(s) = \frac{B(s)}{U(s)} = \frac{(K_p + K_d s)(R_M / s^2 b_m)}{1 + (K_p + K_d s)(R_M / s^2 b_m)} \tag{4}$$

$\because K_p = 1$, (as explained in section 3.2)

$$G(s) = \frac{(1 + K_d s)(R_M / s^2 b_m)}{1 + (1 + K_d s)(R_M / s^2 b_m)} = \frac{\frac{R_M K_d}{b_m} s + \frac{R_M}{b_m}}{s^2 + \frac{R_M K_d}{b_m} s + \frac{R_M}{b_m}}$$

For underdamping system, poles in $G(s)$ are complex conjugate with negative real part,

i.e. $\left(\frac{R_M K_d}{b_m}\right)^2 < 4\frac{R_M}{b_m}, \qquad 0 < K_d < 2\sqrt{\frac{b_m}{R_M}}$

Usually $\frac{b_m}{R_M}$ is much larger than 1, for the example in our experiment, b_m is 100 buffer units,

And R_M is the rate of 2 buffer units per second, the value of K_d should be in the range of $(0, 10\sqrt{2})$.

Considering the smooth contribution to the rate change $\Delta r(t)$ while selecting the K_d to meet the first criteria mentioned in section 3.2, in our experiment we simply set K_d to 1 within the interval listed above although it's not chosen as an optimal value in theory. In our mechanism the two guarded thresholds b_l and b_h will also help to reduce the resultant overshoot from the chosen value of K_d coefficient. Therefore, the idea curves for BO and rate adjustment was obtained as shown in Figure 5 for their stability while we have the step input b_m/s of U(s), K_d equals to 1 and zero initial conditions.

In a network system, the sending rate $s(t)$ can be defined as $s(t) = l(t) + q(t)$ where $q(t)$ represents the jitters or the packet loss. By applying the same idea, we can calculate the adaptation of sending rate

$s(t+\Delta t) = s(t) + \Delta r(t) + \Delta q(t)$ where $\Delta r(t)$ is the picked up control function. For example, $\Delta q(t) = -(s(t) - l(t))*\varepsilon$ where $0 < \varepsilon < 1$ is a small constant.

Buffering the data of media stream may absorb the variation of playback rate, network bandwidth and end-to-end delay jitter. The sending rate $s(t+\Delta t)$ is ideally calculated by the summation of $s(t)$ and two delta functions obtained (i.e. $\Delta r(t)$ and $\Delta q(t)$). But in the experiments our proposed rate control mechanism, the $\Delta q(t)$ is considered to be zero for the assumption of network noise $q(t)$ remains invariant most of the time in formula $s(t) = l(t) + q(t)$. In the following, we will show the reason why the $\Delta q(t)$ can be discarded in the proposed feedback rate control algorithm in the experiments.

$$
\begin{aligned}
&\because r(t) = l(t-\Delta t) - m(t-\Delta t) = s(t-\Delta t) - q(t-\Delta t) - m(t-\Delta t) \\
&\because r(t+\Delta t) = s(t) - q(t) - m(t) \\
&\therefore \Delta r = r(t+\Delta t) - r(t) \\
&= s(t) - s(t-\Delta t) - (q(t) - q(t-\Delta t)) - (m(t) - m(t-\Delta t)) \\
&= \Delta s - \Delta q - \Delta m \\
&\because \Delta b = \Delta r \Delta t \\
&\therefore \Delta b = (\Delta s - \Delta q - \Delta m)\Delta t
\end{aligned}
\qquad \textbf{(5)}
$$

Therefore, the control function for $\Delta q(t)$ can be approximately associated into a single $\Delta r(t)$ control function, because the coming instance of BO $b(t+ \Delta t)$ will partially reflect the accumulated network noise $\Sigma(-\Delta q(t))$ by the observed difference $\Delta b(t)$.

6. Conclusions and Future Works

Our experimental results show that the proposed PD control function efficiently control the occupancy of client buffer of small size. It is shown to perform well even for networks with large network delay. Using the scheme of predictive prevention from overflow and underflow of client buffer, a significant improvement of playback QoS can be achieved. Such a fine-grain control-theoretic rate adaptation scheme will help not only to perform the error control but also the system with larger buffer to save more space to sustain the possible larger network jitter. Therefore, it indicates that BO should be taken into account in friendly rate control functions if the client buffer is not large enough. For example, this BO information can be placed in the optional field of RTCP packets. Therefore, we can study further on the implementation of this extension to transport video over wireless network using RTP/CRTP (Compressed RTP [14], a revised version of RTP/RTCP for multimedia transportation over low-speed serial links like wireless network).

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