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Research Paper

Application-Specific Rate Control for Realtime Streaming Services in Wireless Networks

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ABSTRACT: *In wireless networks, packet loss may be due either to congestion or to channel error. Thus, it is necessary to differentiate between packet loss due to wireless channel errors and that due to congestion. To apply this concept to real-time applications, we consider a rate control method without retransmission for UDP traffic. For implementation of rate control, we utilize a rate control module at the application layer and use NS-2 to determine the performance of adaptive rate control. The results show that the adaptive rate control scheme has a higher throughput than that of the rate control scheme without loss classification. Also, packet loss with adaptive rate control is approximately the same as with rate control without loss classification.*
KEYWORDS: Rate Control, Streaming Service, UDP, TFRC, real-time application

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I. INTRODUCTION

UDP has likely been used for real-time applications, such as video and audio. UDP supplies minimized transmission delay by omitting the connection setup process, flow control, and retransmission. Hence, the quality of service (QoS) of real-time applications using UDP is affected by TCP traffic and the rate control of UDP is required to privide reliable TCP services. There are some viable approaches for implementing this type of system. The main approach involves rate control of the sender with acknowledgment from the receiver [1-3]. The rate of the sender is controlled using information about packet loss and timeouts caused by network congestion. The well-known additive increase and multiplicative decrease (AIMD) rate control protocols increase the source rates in a step-wise fashion in the absence of packet loss and reduce the multiplicative decrease in the packet losses and timeouts. The other approach uses the TCP-friendly rate control (TFRC), which is an equation based rate control in which the throughput of a TCP sender is determined as a function of the packet size, packet loss rate and round-trip time (RTT) [4-7]. TFRC can account for only the packet loss due to congestion when the rate control is applied.

However, these schemes cannot be used in wireless networks in which the main cause of the packet loss is due to wireless channel errors. There are three approaches for solving this inefficiency over wireless networks. The first approach is to split a TCP connection into two separate connections[8-9]. By adding a gateway function to an access point, loss discrimination can be solved. In [9], loss discrimination can be solved by adding the additional sequence number into TFRC headers at the access point. However, this approach violates TCP end-to-end semantics, and it is reasonable to solve this problem based on a basic transport concept. The second approach is to discriminate packet losses using end-to-end transport protocols[10-12]. In [10], this problem can be solved by excluding wireless losses from congestion losses using packet interarrival times. In [11], this problem can be solved by multiple parallel TFRC connections when a single connection is inefficient. However, it is difficult to determine the cause of the packet losses accurately because the boundaries of the loss discrimination are vague. The third approach is to solve this problem by using the cross-layer design (CLD) concept [13-20]. In [14], the packet loss is recognized as arising from wireless channel errors if the recent signal strength is low, and it is due to congestion otherwise. In [15], using the loss discrimination from [10], the ratio of collision losses to wireless losses is estimated by a cross-layering approach, and it is used to control the rates of the sender. In [17], the channel efficiency in MAC protocol is used to control the transmission rate in TCP by comparing with the virtual channel efficiency. In [18] and [19], automatic repeat request(ARQ) and reservation backoff in MAC layer are used for video streaming. In [20], MAC layer contention is used to improve the rate control of the transport layer. But these schemes cannot accurately discriminate the cause of packet losses when

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wireless losses and congestion losses are simultaneously occurred. However, these schemes cannot accurately discriminate the cause of the packet losses when the wireless losses and congestion losses occur simultaneously.

II. SYSTEM MODEL

In this section, we show the system model to which the rate control can be applied. Fig. 1 shows the system model for streaming services in wired and wireless networks. A server S in a wired network transmits packets to client C in a wireless network. For streaming services, either transmission control protocol (TCP) or user datagram protocol (UDP) can be used as a transport layer. TCP permits congestion control and end-to-end flow control, and is a good approach for reliable and delay tolerant applications. UDP is a reasonable choice for real-time streaming services because it does not allow retransmission of packets and control of the source rate. When TCP and UDP applications exist in the same network, there is a starvation of TCP traffic because UDP traffic lacks congestion control.

This is a serious problem with current internet technology. Thus, the system model we use assumes that rate control of UDP traffic is applied via the real-time transport (RTP) / real-time control protocol (RTCP). In Fig. 1, the protocol architecture of real-time streaming service over a UDP layer is shown. The transmission rate of the server is determined by the client's loss classification module (RCM). The session information for the streaming service is exchanged by a real-time streaming protocol (RTSP). Then, RTP over UDP is used for transmitting stream data.If a packet loss was caused by congestion, rate control of the sender is initiated by a command of the receiver. When the sender receives the command, it decreases its transmission rate. The basic idea behind rate control for UDP is to use statistical data obtained over a discrete period of time. Using such statistical data as packet loss and delay, RCM on the client side determines a command for the server and transmits it over the RTCP. The packet losses are measured over a discrete period and then computed wireless channel losses and congestion losses using the proposed loss discrimination method in section 3. If congestion losses are out of the minimum requirement, the command is transmitted by the client to the server using RTCP. In the case of congestion, it is reasonable to control the rate of the sender. But for channel errors, it is useless to control the sending rate. Packet loss caused by channel error requires addition of some redundant bits such as FEC. An application in the client monitors the wireless channel status and can receive loss event from a link layer

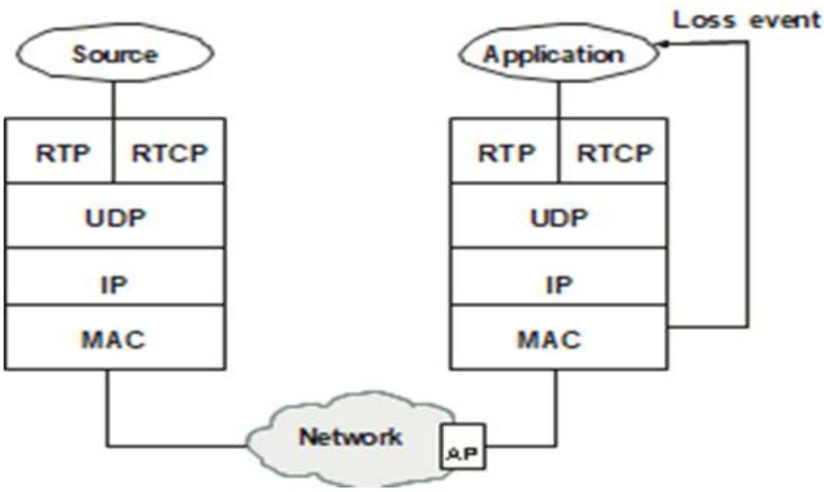

Fig. 1 Protocol architecture

III. RATE CONTROL FOR REAL-TIME STREAMING SERVICES

When packet loss is detected by a client, the client first checks whether it was caused by congestion or channel errors. If it was caused by congestion, rate control of the sender is implemented by sending a command using RTCP. When the sender receives the command, it decreases its transmission rate. For rate control of the sender, the following rate control algorithm can be used.

Let $R(t)$ be the sending rate at t seconds, RTT the end-to-end round trip time, Δ the minimum incremental rate of a sender and D the statistical average delay of packets. Let τ be the rate control interval of the sender, which is greater than RTT. A client receives packets and calculates the average packet delay within the τ interval. $R(t + \tau)$, the sending rate at $(t + \tau)$ seconds, can be written as

$$R(t+\tau) = \begin{cases} \dfrac{kL}{RTT\sqrt{p_c}} & \text{if } D \geq \dfrac{RTT_{max}}{2} \\ R(t) + \Delta & \text{if } D < \dfrac{RTT_{max}}{2} \end{cases} \quad (1)$$

where RTT_{max} is the maximum bound of RTT, k is a constant factor that determines scales of the rate control and L is the packet length.

Then, Δ can be written as

$$\Delta = \frac{L(1+p_c)}{RTT}k \quad (2)$$

In case of the practical situation, the sender has discrete tx levels, which depend on the modulation scheme in a physical layer. Thus it is required to approximate sending rates to tx levels. Let r_i be the rates of the sender with scale factors $i(0 \leq i \leq N)$. The scale factor shows that the sender can transmit N tx levels. Then, the scale factors i can be mapped to the sending rates r_i. $R(t + \tau)$ can be written as

$$R(t+\tau) = \begin{cases} r_{i-1} & \text{if } D \geq \dfrac{RTT_{max}}{2} \\ r_{i+1} & \text{if } D < \dfrac{RTT_{max}}{2} \end{cases} \quad (3)$$

When networks are congested, this approach is not effective because the advantage of rate control is greater than that of loss classification. It is reasonable to control the rate of multimedia applications based on service characteristics. From the assumption that a multimedia application has priority, the priority rate control can be used. When networks are congested, an application with low priority is controlled by the command of a client. But an application with high priority can sustain the transmission rate. If this scheme is not used when networks are congested, the throughput of multimedia applications decrease because packet losses are increased due to network congestion. Thus, it is useful to control the rate of senders with priority. Let the priority of a application be P_r. $R_p(t + \tau)$, the sending rate with priority at $(t + \tau)$ seconds, can be written as

$$R_p(t+\tau) = \begin{cases} \frac{kL}{RTT\sqrt{p_c}} & \text{if } P_r = low \text{ and} \\ & D \geq RTT_{max}/2 \\ R_p(t) + \Delta & \text{if } P_r = high \text{ and} \\ & D \geq RTT_{max}/2 \\ R_p(t) + \Delta & \text{if } D < RTT_{max}/2 \end{cases} \quad (4)$$

If the packet loss is due to network congestion, the rate control of the sender is required. Let $R(t)$ be the transmission rates of a sender at t and p_{min} is the minimum probability for rate control. Then, $R(t + \tau)$, the transmission rate at $(t + \tau)$ seconds, can be written as

$$R(t+\tau) = \begin{cases} R(t) - \Delta & \text{if } p_c \geq p_{min} \\ R(t) + \Delta & \text{if } p_c < p_{min} \end{cases} \quad (5)$$

From the assumption that wired links have sufficient bandwidth and wireless link is the bottleneck of the system, congestion loss can be computed. In this situation, packet losses due to congestion only occur in AP, which has K limited buffer. Then, the packet losses due to congestion can be modeled as M/M/1/K system. The blocking probability is the same as the packet loss probability. Let λ and μ be the arrival and service rates of AP. Then packet loss probability Π due to congestion, can be obtained as

$$\Pi = \frac{1-\lambda/\mu}{1-(\lambda/\mu)^{K+1}}\left(\frac{\lambda}{\mu}\right)^K \quad (6)$$

IV. NUMERICAL RESULTS

To evaluate the performance of the proposed scheme, we use UDP and TFRC modules implemented in NS-3 for simulation. We consider a streaming server (S1) in a wired network and two clients (C1 and C2) in IEEE 802.11 LAN. The two clients are located 40m from an access point. For the rate control algorithm, S1 has 5 different scale levels (500Kbps, 1Mbps, 1.2Mbps, 1.5Mbps, 2Mbps) and the transmission rate is determined by the feedback information from C1. The packet length is assumed to be 1460 bytes. The feedback information includes a scale factor which shows the transmission rate of S1. The scale factor can be determined from the statistical information related to the received packets. For the wireless channel, two-state error model with p=0.97 and q=0.43 is assumed.

To effectively simulate the performance of the rate control schemes, two kinds of traffic, are considered. The first case is a normal traffic situation in which there are wireless channel errors but no congestion. This case shows how a flow F1, which consists of S1 and C1, transmits using UDP. The other case simulates a heavy traffic situation in which there is packet loss due to congestion and channel errors. To create this heavy traffic condition, flow F2, which consists of S1 and C2, transmits UDP traffic. The adaptive rate control algorithm is implemented for both F1 and F2 flows, and the performance of the proposed scheme is compared with that for a rate control scheme without loss classification.

Fig. 2 shows the throughput of adaptive rate control under the normal traffic condition. When the adaptive rate control algorithm is applied, the throughput is higher than that for rate control without loss classification. UDP traffic is transmitted at 2Mbps through an 11Mbps wireless link (effectively about 4Mbps). When packet loss is detected, the rate control algorithm without loss classification should decrease the rates of the sender. However, the adaptive rate control maintains the rates of the sender because these packet losses are due to wireless channel errors.

Fig. 3 shows the packet losses of UDP traffic under a random error condition. Here there is no difference between two control schemes because packet loss is due to wireless channel error.

Fig. 4 shows the throughput of adaptive rate control under the heavy traffic condition. Here, the adaptive rate control algorithm is applied and the throughput is lower than that of the rate control without loss classification because the rate control scheme is applied when packet loss is detected and the round-trip time is out of the predefined range. This means some packets are lost due to congestion. This result can vary with the characteristics of the congestion situation, but shows that adaptive rate control is not a effective method when networks are congested. In this case, rate control without classification is more effective than adaptive rate control.

Fig.5 shows the packet losses under the heavy traffic condition when packet losses are due to channel error and congestion. The rate control scheme without loss classification decreases the rate of the sender whenever there are packet losses. But the adaptive rate control scheme decreases the rate of the sender when packet loss is detected and round trip-time is out of the predefined range. Thus, the packet loss of the adaptive rate control scheme is higher than that of the rate control without loss classification. This result shows that the adaptive rate control is not effective under heavy traffic conditions. Table. 1 shows the average rx rate and standard deviation of adaptive rate control with priority under the heavy traffic condition. Thus, the packet loss of adaptive rate control with priority is lower than that of the rate control scheme without priority. These results shows that adaptive rate control with priority is a effective method when networks are congested. In this case, adaptive rate control with priority is more effective than that without priority. For real-time streaming services, the streaming server determines the rates of TFRC using feedback information from the client. The feedback information includes the packet loss ratio of the client. In the client, the packet loss ratio is computed by using the loss events in the link layer. The performance of the wireless TFRC is compared with that of TFRC.

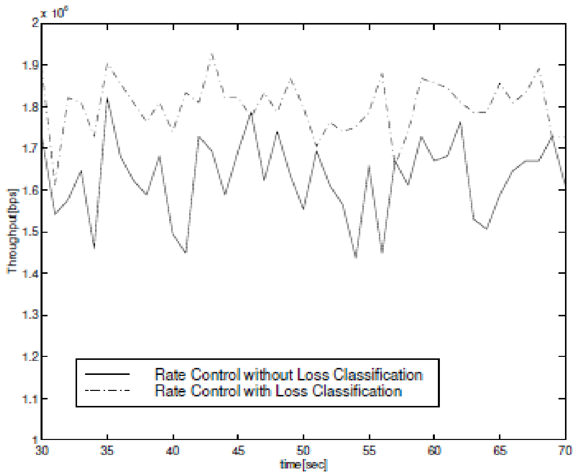

Fig. 2 Throughput of adaptive rate control under the normal traffic condition

Table 1 Average Rx rate and standard deviation of adaptive rate control with priority under the heavy traffic condition(class1 has high priority)

	No control		Priority control	
	class 1	class 2	class 1	class 2
avg. rx rate	1.147	1.158	1.348	1.027
std. dev.	0.166	0.169	0.098	0.076

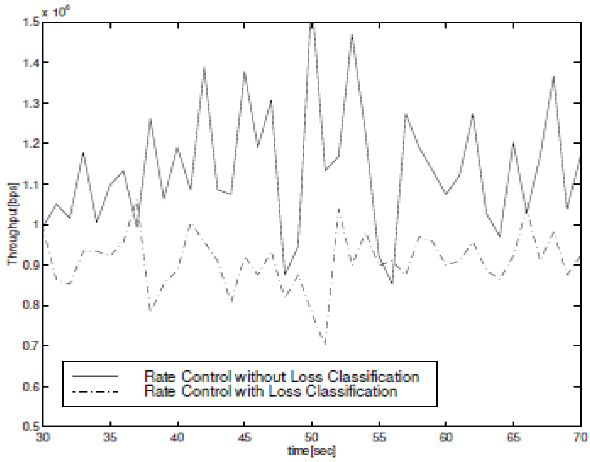

Fig. 3 packet loss of adaptive rate control under the normal traffic condition

The wireless TFRC (r=4) and wireless TFRC (r=2) represent methods for limiting the retransmission in the link layer by 4 and 2 times, respectively. The packet length is assumed to be 1460 bytes. For the wireless channel, a 0%-6% packet error ratio (PER) is assumed.

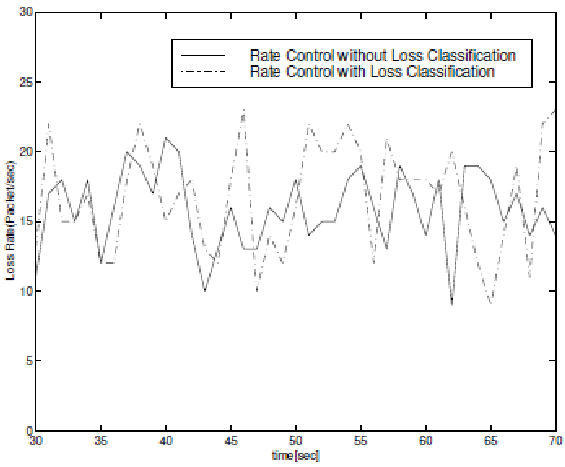

Fig. 4 Throughput of adaptive rate control under the heavy traffic

Fig. 6 shows the RTT of TFRC and wireless TFRC for different PERs. When PER is increased, the RTT of each method is increased because of the retransmission of packets in the link layer. Specifically, when PER increases above 4%, the RTT of the TFRC and wireless TFRC (r=4) increase drastically. This increase indicates that RTT is the major factor for determining the rate of the sender. Thus, for wireless TFRC (r=2), the reduction in the retransmission limit has a strong impact on the RTT.

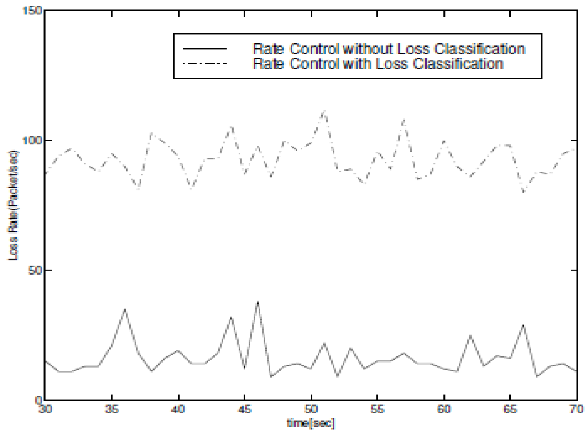

Fig. 5 Packet loss of adaptive rate control under the heavy traffic condition

The gradual increase in the RTT has more stable characteristics than in the TFRC and wireless TFRC (r=

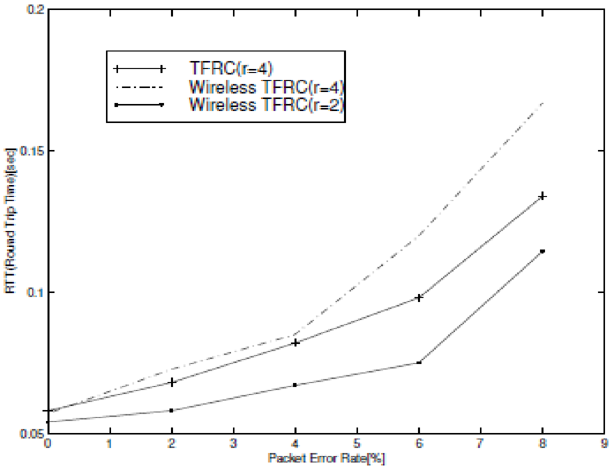

Fig. 6 RTT of TFRC, wireless

V. CONCLUSIONS

This paper proposed the rate control scheme that can be applied to real-time streaming services over wireless networks. The proposed scheme controls the rate of a sender with a reduced retransmission by differentiating between the packet loss that can be attributed to channel error and the packet loss that can be attributed to congestion. Performance evaluations showed that the adaptive rate control scheme has a higher throughput than that of the rate control without classification when the wireless networks are not congested. In congestion networks, it has more stable characteristics than the rate control condition

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