International Journal of Recent Technology and Engineering (IJRTE)
ISSN: 2277-3878 (Online), Volume-8 Issue-6, March 2020

MAC-aware Loss Discrimination for Real-time Streaming Services in Wireless Networks

Seung Sik Choi

Abstract**:** ***This paper addresses the rate control problem for real-time applications streamed over wireless networks. In wired networks, an equation-based rate control such as TCP-friendly rate control (TFRC) can be used to control the rates of a source under the assumption that the loss is primarily due to congestion. But in wireless networks, packet loss may be due either to congestion or to channel errors. Thus, it is necessary to differentiate between packet loss due to wireless channel errors and that due to congestion. The MAC-aware rate control scheme for real-time streaming applications discriminates packet losses due to channel errors using the event generation in the MAC layer. The simulation results show that the MAC-aware rate control scheme has higher throughput than the rate control scheme without loss classification.***

Keywords **:** ***Rate Control, Streaming Service, TFRC, DCCP, real-time application.***

I. INTRODUCTION

Recently, there has been considerable interest in streaming technology over the internet. However, current internet technology only supports best-effort service and does not provide quality of service (QoS) guarantees. In this environment, multimedia applications are provided by modifying the routing algorithm and the end-to-end transport protocol. Modification of the router is not easy problem to address due to the cost involved. Alternatively, a reasonably simple approach is to enable the transport and application layers to adapt to varying network conditions, including available bandwidth, delay and jitter. This requires control of the transport layer in response to the dynamic service characteristics. One mechanism that can efficiently support streaming services involves control of the transmission rates based on the network conditions.

There are three viable approaches for implementing this. The main approach involves rate control of the sender with acknowledgement 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 source rates in a step-wise fashion in the absence of packet loss and reduce the multiplicative decrease in packet loss and timeouts. Another approach uses 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 packet size, packet loss rate and round-trip time (RTT) [4-7]. TFRC can only take into account packet loss due to congestion when rate control is applied.

However, these schemes cannot be used in wireless networks in which the main cause of packet loss is due to wireless channel errors. There are three approaches to solve this inefficiency over wireless networks. The first approach is to split a TCP connection into two separate connections[8-9]. By adding 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 But this approach violates TCP end-to-end semantics and it is reasonable to solve this problem based on 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 inter-arrival time. In [11], this problem can be solved by multiple parallel TFRC connections when a single connection is inefficient. But it is difficult to discriminate accurately the cause of packet losses because the boundaries of loss discrimination are vague. The third approach is to solve this problem using the cross layer design (CLD) concept [13-20]. In [14], the packet loss is recognized as due to wireless channel errors if recent signal strength is low, and due to congestion otherwise. In [15], using loss discrimination of [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 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 wireless losses and congestion losses are simultaneously occurred.

In this paper, the author proposes an explicit discrimination method of packet losses due to wireless channel and those due to congestion.

Manuscript received on March 15, 2020.
Revised Manuscript received on March 24, 2020.
Manuscript published on March 30, 2020.
* Correspondence Author
Seung Sik Choi*, Computer Eng., Incheon National University, Incheon, Korea. Email: sschoi@inu.ac.kr
This work was supported by the Incheon National University Research Grant in 2015.

This is differentiated from the previous works which use implicit discrimination methods such as inter-arrival time[10] and signal strength[14]. In this method, when a receiver detects a packet loss, it checks a loss event in the link layer, which is generated by a packet loss over a wireless link.

If a loss event is detected, the packet loss is due to wireless channel errors. Otherwise, the packet loss is due to congestion. When a packet loss is caused by congestion, the rate of the sender is controlled by the feedback information of the receiver.

The first advantage of this algorithm is that it can be applied in such a way as to not change the network infrastructure and protocols, but rather to change the end-to-end application. The second advantage of this algorithm is that it removes the inaccurate discrimination of the previous works. The third advantage is that it can efficiently use the wireless bandwidth by not decreasing the source rates when there is a wireless channel error. Essentially it prevents a system from underutilizing its wireless bandwidth.

This paper is organized as follows: Section II is devoted to explain the system model of MAC-aware loss discrimination. In Section III, the proposed rate control scheme and its main characteristics are described. Numerical results for the proposed rate control scheme are provided in Section IV, and concluding remarks are contained in Section V.

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 net¬works. The targeted wireless technology is based on IEEE 802.11 wireless LAN. The wireless link is assumed to have an available bandwidth of Bw, and a packet loss ratio pw due to wireless channel errors. A streaming server transmits packets to clients over wired and wireless networks. For streaming services, TCP, TFRC or 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. But this protocol has proven to be inefficient for providing multimedia data[5]. The main problem with TCP is that it cannot support bursty traffic due to window-based rate control. Because of this reason, a rate-based transmission protocol, TFRC has been proposed by the Internet Engineering Task Force (IETF) for the transmission of multimedia traffic. TFRC calculates the sending rate based on the TCP throughput equation [7] and provides a smooth traffic rate. UDP is a reasonable choice for real-time streaming services because it does not allow retransmission of packets and control of the source rate. But, 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, Datagram Congestion Control Protocol (DCCP) has been proposed to control UDP traffic[21]. DCCP provides a way to gain access to congestion control mechanisms without having to implement them at the application layer. It allows for flow-based semantics like in TCP, but does not provide reliable in-order delivery.

Based on these protocols, the architecture for MAC-aware loss discrimination is shown in Fig. 2. 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 using the congestion algorithm.

The basic idea of the loss discrimination is to detect the loss event in the MAC layer. In the MAC layer, packet loss due to wireless channel error is detected by the sequence number and a loss event is forwarded to the transport layer. It consists of 3 sub-modules: the loss classification sub-module (LCS), the congestion command sub-module (CCS), and the channel status sub-module (CSS).

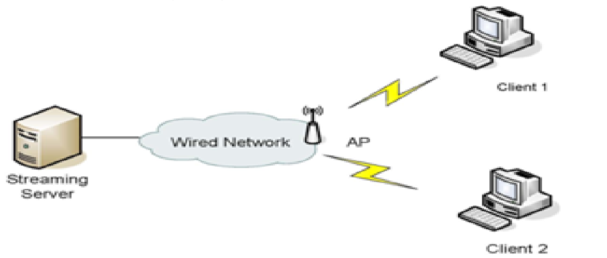

Figure 1. A streaming application transmits packets over wired and wireless networks

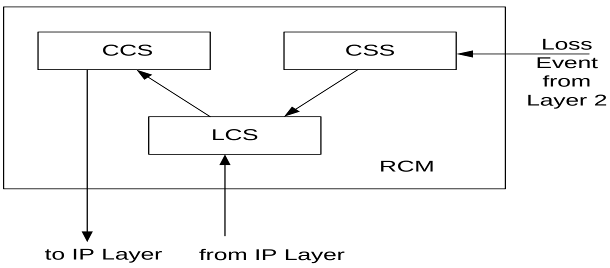

Figure 2. RCM architecture

In the LCS on the client side, 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, LCS notifies the congestion event to the CCS and the command is transmitted by the CCS to the server. 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. The CSS can be used to monitor the wireless channel status and to receive loss event from a link layer.

III. RATE CONTROL ALGORITHM FOR REAL-TIME STREAMING SERVICES

A. Loss Discrimination

In the transport layer, sequence number (SN) field in the header is defined to show the sequence of packets and packet losses are detected by this field. In the MAC layer, a IEEE802.11 frame also has SN field, thus it is re-defined as link sequence number (LSN) field to discriminate SN field in the transport layer. The access point writes LSN on the frame header of received packets from the server and sends them to the client.

The client accurately detects a packet loss due to wireless channels using LSN and generates a loss event. This event is forwarded to the transport layer. When a packet loss is detected by the transport layer, the transport layer first checks whether it was caused by congestion or channel errors using a loss event in the link layer. If the loss event is detected, the packet loss is due to wireless channel errors. Otherwise, the packet loss is due to congestion.

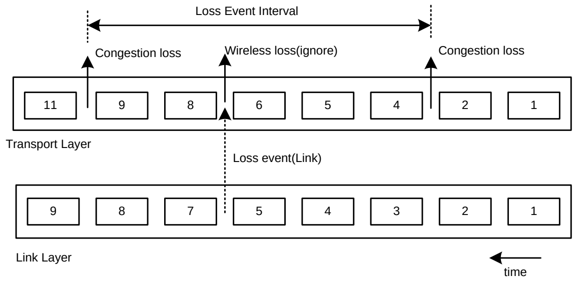

Figure 3. Loss Discrimination Example

Fig.4 shows a loss discrimination example. In this figure, SN and LSN are represented by numeric and alphabet symbols, respectively. In this example, there are three packet losses(3, 7, 10) in the transport layer, but in the link layer, there is only one loss event(f). This event shows that 7 is a packet loss due to wireless channels. Thus, 3 and 10 are packet losses due to congestion. The loss event interval τ is measured by packet losses due to congestion as shown in Fig. 4. The loss event rate which is used to get the rate of the sender can be obtained as 1/ τ [4].

In order to compute the transmission rate of the sender, the sender should get round trip time and packet loss due to congestion. The end-to-end round trip time RTT can be estimated from packet start time and feedback arrival time. And packet loss is measured using sequential number field.

In initial phase, the sender transmits DATA packet to the receiver and the receiver transmits feedback to the sender. Let RTT_i be round trip time in the i-th packet. And ts,i and ta,i indicate the start time of the i-th DATA packet and the arrival time of feedback to the i-th packet, respectively. N is the number of received packets in the t interval. Then, the initial RTT0 can be estimated by ta,0 −ts,0. In the next phase, the RTTi in the i-th packet can be estimated as

$$RTT_i = \sigma RTT_{i-1} + (1-\sigma)(t_{a,i} - t_{s,i}) \qquad (1)$$

where σ is adjustable parameter. For the compu¬tation of packet loss due to congestion, the sequential field of received packets and wireless loss events are used. Let SNi be the sequential number of the i−th received packet in the interval τ. And ηi is the number of packet loss due to congestion in τ when the i-th packet receives. Let ei(i-1) be the number of wireless loss event from the link layer between i-th and (i-1)-th packet. Then, the packet loss due to congestion is computed as

$$\eta_i = (SN_i - SN_{i-1} - 1) - e_{i(i-1)} \quad (2)$$

From (2), ηi>0 means there are packet losses due to congestion. Otherwise ηi=0 means there is no packet loss due to congestion (no packet loss or only wireless packet loss). When ηi<0, this means there are negative congestion loss due to the nonconsecutive arrived packet. This equation should be changed because the received packet at the receiver is not arrived at the same order. In other words, the sequence number of packet at the receiver may be changed due to different routing paths. This should be considered in (2). To reflect this problem, (2) is changed as

$$\eta_i = \begin{cases} (SN_i - SN_{i-1} - 1) - e_{i(i-1)} & if\ SN_i \geq SN_{i-1} \\ -1 - e_{i(i-1)} & if\ SN_i < SN_{i-1} \end{cases} \qquad (3)$$

where SNi-1 is set to be the largest sequence number until i-1.

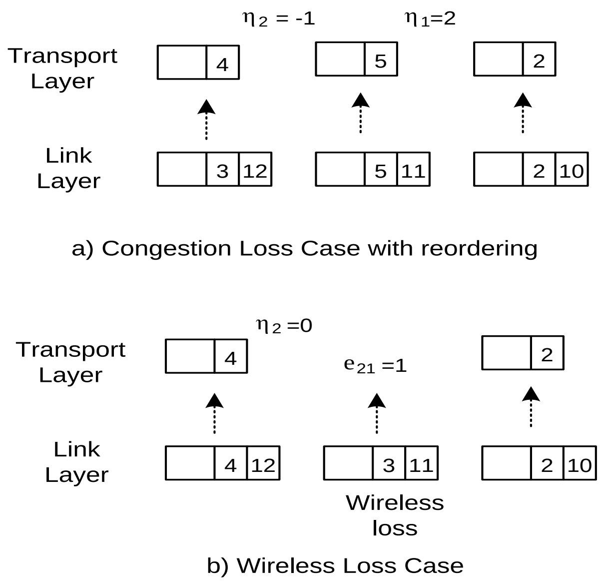

Figure 4. Loss Classification Example

Fig. 4 shows a loss classification example. In Fig.4a, the client receives packets with nonconsecutive sequence number and there was a congestion loss (SN=3). In this case, we can estimate that there was congestion loss from η_1+ η_2=1. In Fig.4b, the client receives packets with wireless loss. In this case, we can also estimate that there was wireless loss from η_2=0(It can be computed by η_2=4-2-1-1). As another example, sequence numbers arrive at the transport layer in order such as "1 4 2 3 5" without channel errors. In this case, η_1=4-1-1=+2, η_2 =-1, η_3 =-1 and η_4=5-4-1=0. Thus, the summation of η equals to 0. Therefore, there is no congestion loss. This method can be extended to the more complex case.

B. Rate Control Using Loss Discrimination

Let the packet loss ratio in a client be p_l. This can be caused either by congestion or by wireless channel errors. When there is congestion in wired networks, the loss ratio due to congestion is represented by p_c. When there are wireless channel errors in the wireless network, the packet loss ratio can be written as:

$$p_l = p_c + p_w (1 - p_c) \quad (4)$$

Equation (4) means that the packet loss ratio pl in a client is the summation of the packet loss ratio, pc, in wired links due to congestion and the packet loss ratio, $p_w(1-p_c)$, due to wireless channel errors without congestion.

From this method, pw can be computed as

$$\rho_w = \frac{\sum_{i=1}^{M} e_{(i+1)i}}{N + \sum_{i=1}^{M} e_{(i+1)i}} \quad (5)$$

Using (5), pc can be computed as

$$\rho_c = \frac{\sum_{i=1}^{M} \eta_i}{N + \sum_{i=1}^{M} e_{(i+1)i} + \sum_{i=1}^{M} \eta_i} \quad (6)$$

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 pmin be the minimum probability for rate control respectively. Then, R(t + τ), the transmission rate at (t + τ) seconds, can be written as

$$R(t+\tau) = \begin{cases} R(t) - \Delta & \rho_c > \rho_{\min} \\ R(t) + \Delta & otherwise \end{cases} \quad (7)$$

In (7), Δ can be written as

$$\Delta = \frac{L(1+\rho_c)}{RTT} f \quad (8)$$

where L is the packet length and f is the constant factor which determines scales of rate control.

IV. NUMERICAL RESULTS

A. Rate Control over UDP

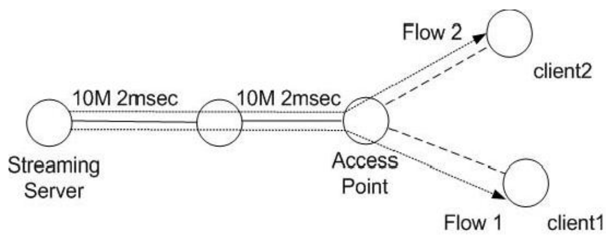

Figure 5. The simulation model from NS-3

To evaluate the performance of the proposed scheme, we consider UDP with congestion control mechanisms. For this purpose, we use DCCP modules implemented in NS-3 for simulation. The DCCP is a transport protocol that implements bidirectional, unicast connections of congestion-controlled, unreliable datagrams. The DCCP module with TFRC-like is selected as the congestion control mechanism for real-time streaming services. As shown in Fig.5, we consider a streaming server in a wired network and two clients (client 1 and client 2) in IEEE 11Mbps 802.11 LAN. The bandwidth of the wired link is 10Mbps and the propagation delay is 2 msec. First, flow 1 is established between the server and client 1. For the rate control algorithm, the transmission rate in the server is determined by the feedback information from client 1. The rate control is based on the DCCP module with TFRC-like congestion control. The MAC-aware loss discrimination algorithm is implemented in client 1 and ftp traffic is generated by the server. For the simulation under normal traffic, only flow 1 is considered. The throughput and congestion loss ratio of wireless DCCP with loss discrimination is compared with those for DCCP without loss discrimination.

Fig. 6 shows the throughput variation of MAC-aware rate control under the normal traffic condition. When the MAC-aware rate control algorithm is applied, the throughput is higher than that of rate control without loss discrimination. When packet loss is detected, the rate control algorithm without loss discrimination should decrease the rates of the sender. However, wireless DCCP maintains the rates of the sender because these packet losses are due to wireless channel errors.

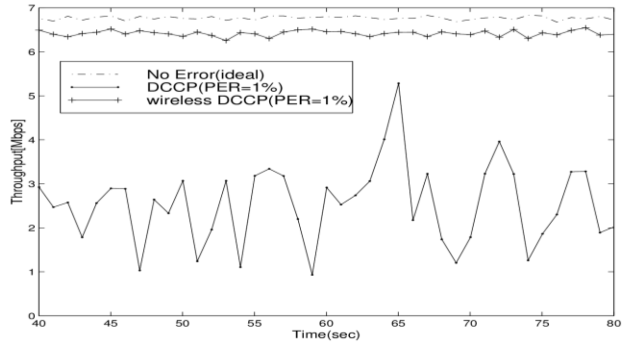

Figure 6. Throughput variation under the normal traffic

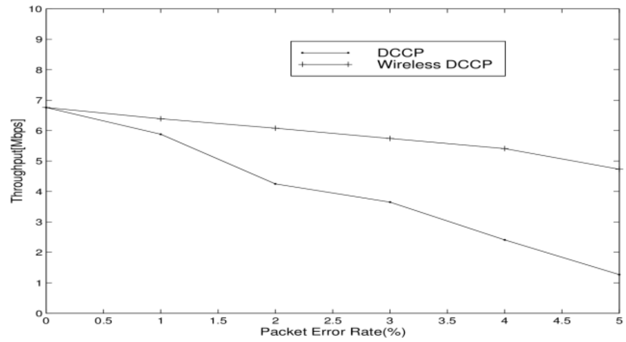

Figure 7. Throughput of DCCP and wireless DCCP under the normal traffic condition

Fig. 7 shows the throughput of DCCP and wireless DCCP as a function of different PERs under the normal traffic condition. When PERs due to wireless channel errors are increased, the throughput in wireless DCCP is higher than that in DCCP because the MAC layer in client 1 can discriminate packet losses due to congestion or channel errors

Fig. 8 shows the congestion loss ratio of DCCP and wireless DCCP as a function of different PERs under the normal traffic condition. When PERs due to channel errors are increased, the congestion loss ratio of DCCP is also increased because DCCP cannot discriminate the cause of packet losses. In DCCP, packet losses due to channel errors are considered as the congestion losses and it deteriorates the throughput. But in wireless DCCP, it discriminates the cause of packet losses and there is no change in congestion loss ratio.

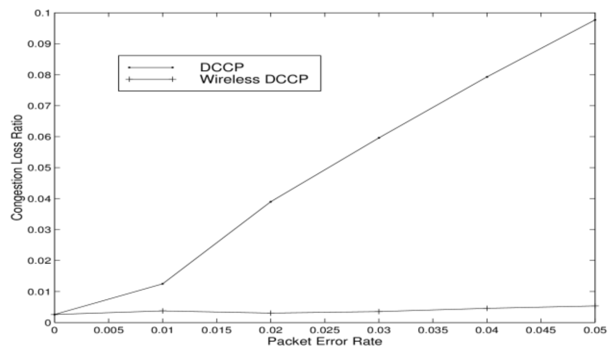

Figure 8. Congestion loss ratio of DCCP and wireless DCCP in client 1 vs. different PERs

International Journal of Recent Technology and Engineering (IJRTE)
ISSN: 2277-3878 (Online), Volume-8 Issue-6, March 2020

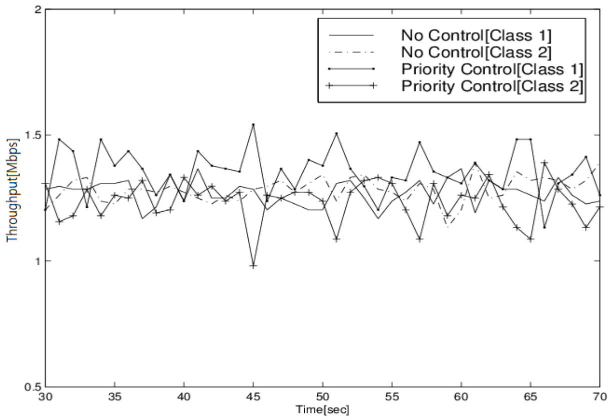

Figure 9. Throughput of MAC-aware rate control with priority under the heavy traffic condition(class1 has high priority)

Table I. Average throughput and standard deviation of MAC-aware rate control with priority under the heavy traffic condition (class1 has high priority)

	No Control		MAC-aware Rate Control with Priority	
	Class 1	Class 2	Class 1	Class 2
Avg. Thr.	1.147	1.158	1.348	1.027
Stand. Dev.	0.166	0.169	0.098	0.076

(unit : Mbps)

For the simulation under heavy traffic, flow 1 is established between the server and client 1 and flow 2 is also established between the server and client 2. The server simultaneously generates ftp traffic over DCCP in flow 2 and CBR traffic over UDP in flow 2. The packet length and inter arrival time of CBR traffic are assume to be 1000 bytes and .004, respectively.

Fig. 9 shows the throughput of DCCP and wireless DCCP under the heavy traffic condition. Here, wireless DCCP with priority is applied. The throughput with high priority is higher than that of the rate control without priority. Table I shows average throughput and standard deviation of wireless DCCP with priority under the heavy traffic condition. The standard deviation of wireless DCCP with priority has stable characteristics because packet loss due to wireless channel loss is discriminated. These results show that wireless DCCP with priority is an effective method when networks are congested. In this case, wireless DCCP with priority is more effective than DCCP without priority.

B. Rate Control over TFRC

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. The wireless TFRC (r=4) and wireless TFRC (r=2) represent methods for limiting retransmission in the link layer by 4 and 2 times, respectively

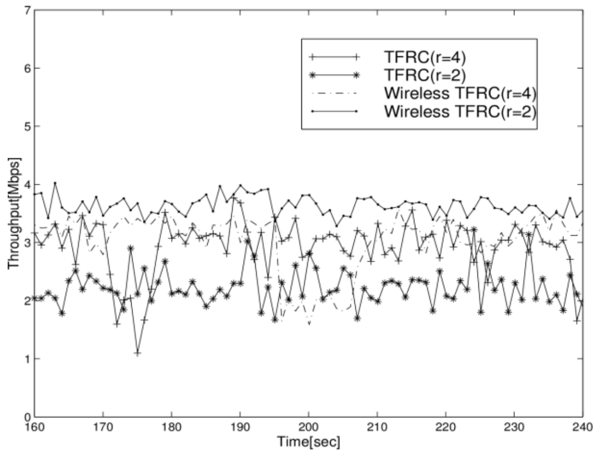

Figure 10 Throughput of TFRC, wireless TFRC(r=4) and wireless TFRC(r=2) when PER is 6%

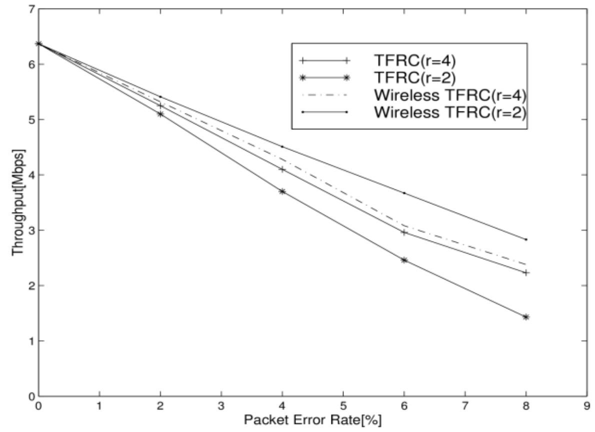

Figure 11. Throughput of TFRC, wireless TFRC(r=4) and wireless TFRC(r=2) vs. PER

Fig. 10 shows the throughput of wireless TFRC depending on the time when a 6% packet loss occurs in the wireless channel. When there is an error over the wireless channel, it increases the RTT due to retransmission in the link layer and also increases the packet loss ratio. The receiver accumulates packet losses during RTT and feedback to the sender. The sender reduces the transmission rates using the feedback information of the receiver. However, the wireless TFRC also reduces the throughput, but the reduction is less than that of TFRC because the packet loss due to wireless channel errors is excluded in the packet loss ratio. When the wireless TFRC algorithm is applied, the throughput of wireless TFRC(r=4) is higher than that of TFRC(r=4) because the congestion loss computed in the receiver is decreased. The wireless TFRC (r=4) has less throughput than that of the wireless TFRC (r=2) because the retransmission degrades the performance.

Fig. 11 shows the throughput of the TFRC and wireless TFRC for different PER. When the PER is increased, the throughput of each method is decreased because the RTT and packet loss ratio are increased. Specifically, the retransmission in the link layer due to channel errors causes a reduced rate for the sender. The amount of the reduced rate can be decreased by computing the packet loss ratio, excluding the wireless channel loss.

The advantage of wireless TFRC (r=2) is superior to TFRC(r=2), TFRC(r=4) and wireless TFRC (r=4) when PER is increased.

Fig. 12 shows the RTT of TFRC and wireless TFRC for different PER. 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. The gradual increase in the RTT has more stable characteristics than TFRC and wireless TFRC (r=4)

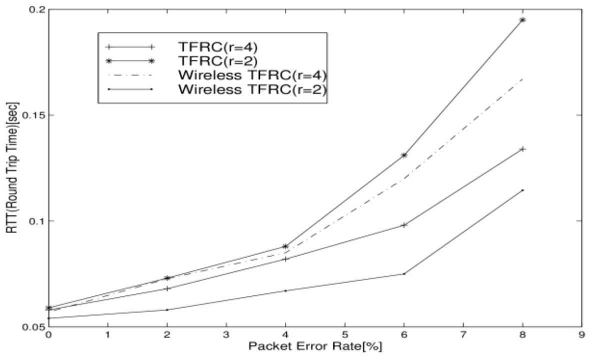

Figure 12. RTT of TFRC, wireless TFRC(r=4) and wireless TFRC(r=2) vs. PER

V. CONCLUSIONS

This paper proposed a form of MAC-aware rate control at the transport layer that can be applied to real-time streaming services over wireless networks. The proposed scheme controls the rate of a sender by differentiating between packet loss due to channel error and that due to congestion. Performance evaluations showed that the MAC-aware rate control scheme has a higher throughput than that of the rate control without loss discrimination when the wireless networks are not congested. In congestion networks, it has more stable characteristics than the rate control without classification.

REFERENCES

1. R. Rejaie, M. Handley, and D. Estein, “Quality adaptation for congestion controlled video playback over the Internet,” in Proc. SIGCOMM’99., Aug. 1999, pp.189-200.
2. S. Jacobs and A. Eleftheriadis,“Streaming video using TCP flow control and dynamic rate shaping,” J. Vis. Commun. Image Represent., vol.9, no.3, pp.211-222, Sep. 1998.
3. S. Cen, C. Pu, and J.W Alpole,“Flow and congestion con¬trol for Internet streaming applications,” Proc. Multimedia Computing and Networking, pp.250-264, Jan. 1998.
4. J. Padhye,J. Kurose, and D. Towsley et al.,“A model based TCP-friendly rate control protocol,” Proc. 9th Int. Workshop NOSSDAV’98, June 1998.
5. W. Tan and A. Zakhor,“Real-time Internet video using er¬ror resilient scalable compression and TCP-friendly transport protocol,” IEEE Trans. Multimedia, vol.1, pp. 172-186, June 1999.
6. J. Padhye, V. Firoiu, D. Towsley, and J. Kurose, “Modeling TCP throughput : A simple model and its empirical valida¬tion,” in Proc. SIGCOMM’98., Aug. 1998, pp.303-314.
7. S.Floyd et al., “Equation-based congestion control for unicast applications,” in Proc. SIGCOMM’2000., Aug. 2000, pp.43¬
8. A.Bakre and B.R.Badrinath, “I-TCP:Indirect TCP for Mo¬bile Hosts,” in Proc. 15th IDCS, Vancouver, Canada, May 1995.
9. H. Lee and C.H. Choi “A Loss Discrimination Scheme for TFRC in Last Hop Wireless Networks,” in Proc. WCNC’2007, pp.3084-3088.
10. M. Chen and A. Zakhor ,“Rate control for streaming video over wireless,” IEEE Trans. Wireless, vol. 1, pp. 32-41, Aug. 2005.
11. S. Cen, P. Cosman, and G. Voelker ,“End-to-End Differenti¬ation of congestion and wireless losses,” IEEE/ACM Trans. Net., vol. 11, no. 5, 2003.
12. Panagiotis Papadimitriou and Vassilis Tsaoussidis, “Selective rate control for media-streaming applications in wireless internet environments”, in proc. of IEEE PIMRC’07, 2007.
13. S. Shakkottai, T. Rappaprot, and P. Karlsson, “Cross-Layer Design for wireless networks,” IEEE Commun. Mag., vol.41, no.10, Oct. 2003, pp.74-80.
14. F. Yang et al., “End-to-End TCP-Friendly Streaming Protocol and Bit Allocation for Scalable Video over Mobile Wire¬less Internet,” Proc. INFOCOM’2004, Mar. 2004.
15. S. Pack, X. Shen, J. Mark and L.Cai, “A Two-Phase Loss Differentiation Algorithm for Improving TFRC Performance in IEEE 802.11 WLANs,” IEEE Wireless Commun., vol. 6, no. 11, Nov. 2007, pp.4164-4175.
16. Zhu, et al., “Cross-Layer Design of Source Rate Control and Qos-Aware Congestion Control for Wireless Video Streaming” Proc. Advances in Multimedia, 2007.
17. X. Zhang, J. Lv, X. Han and D. K. Sung, “ Channel Efficiency-Based Transmission Rate Control for Congestion Avoidance in Wireless Ad Hoc Networks” IEEE Communications Letters, vol.13, no.9, Sep. 2009, pp.706-708.
18. D. Wu, S. Ci and H. Wang, “Cross-Layer Optimization for Video Summary Transmission over Wireless Networks," IEEE J. Sel. Areas Commun., vol.25, no.4, May 2007, pp.841-849.
19. Y. He, J. Sun, R. Yuan and W. Gong, “A reservation based backoff method for video streaming in 802.11 home networks," IEEE J. Sel. Areas Commun., vol.28, no.3, Apr. 2010, pp.332-343.
20. Le Minh Duong, L. Zitoune and V. Veque, “MAC-aware Rate Control for Transport Protocol in Multihop Wireless Networks", Proc. IEEE PIMRC'2012, 2012.
21. S. Floyd, E. Kohler and J. Padhye, “Profile for Datagram Congestion Control Protocol(DCCP), Congestion Control ID 3: TFRC", IETF, RFC 4342, L.A., USA, 2006.

AUTHORS PROFILE

Seung Sik Choi received the B.S. degree in Electronics engineering from Yonsei University in 1988, and the M.S. and Ph.D degrees in Electrical and Electronics Engineering from the Korea Advanced Institute of Science and Technology(KAIST) in 1990 and 2002, respectively. From 1990 to 2003, he was at Communication Network Lab. in Korea Telecom, where he worked on B-ISDN/ATM switch and network management system. From 2004, he is a professor in the department of computer engineering, Incheon National University. His interests include communication networks, sensor networks, resource management in CDMA systems, protocol design and medium access control.