Journal of Zhejiang University SCIENCE A
ISSN 1009-3095 (Print); ISSN 1862-1775 (Online)
www.zju.edu.cn/jzus; www.springerlink.com
E-mail: jzus@zju.edu.cn

Equation based rate control scheme for video streaming over wireless channels with link level ARQ*

ZHU Xiao-liang†, YANG Zong-kai, DU Xu, CHENG Wen-qing
(*Department of Electronics & Information Engineering, Huazhong University of Science & Technology, Wuhan 430074, China*)
†E-mail: zhuxlhust@yahoo.com.cn

Received Dec. 15, 2005; revision accepted Feb. 17, 2006

Abstract: Equation based TCP-friendly rate control (TFRC) protocol has been proposed to support video streaming applications. In order to improve TFRC performance in wireless channels, the link level automatic repeat request (ARQ) scheme is usually deployed. However, ARQ cannot ensure strict delay guarantees, especially over multihop links. This paper introduces a theoretical model to deduce an equation for packet size adjustment in transport layer to minimize retransmission delay by taking into consideration the causative reasons inducing retransmission in link layer. An enhanced TFRC (ETFRC) scheme is proposed integrating TFRC with variable packet size policy. Simulation results demonstrate that higher goodput, lower packet loss rate (PLR), lower frame transmission delay and jitter with good fairness can be achieved by our proposed mechanism.

Key words: TFRC, Video streaming, Wireless channels, Automatic repeat request (ARQ), Rate control
doi:10.1631/jzus.2006.A1702 **Document code:** A **CLC number:** TN919.8

INTRODUCTION

Video streaming applications generally require continuous high bandwidth guarantees as well as stringent bounds on delays and jitters (Shen *et al*., 2004; Hassan and Krunz, 2004). Providing video streaming with acceptable quality degradation over wireless channels is fraught with challenges due to the adverse influence of bandwidth limitation, time variation, high error rate, and so on (Etoh and Yoshimura, 2005; Shen *et al*., 2004; Chen and Zakhor, 2004; Hassan and Krunz, 2004). To cope with these challenges, rate control aimed to provide appropriate traffic regulations for streaming applications is one of the main counter measures (Hassan and Krunz, 2004). UDP is usually used to provide video streaming to guarantee real-time performance, but it may result in unfair competition with TCP without effective traffic regulation. Therefore, maintaining fairness in current TCP flows is also a concern (Shen *et al*., 2004; Chen and Zakhor, 2004).

A widely popular rate control scheme with perfect fairness and relative smoothness over wired networks is equation based rate control, also known as TFRC, which assumes that packet loss, delay and jitter are caused by network congestion. The equation is described below (Floyed *et al*., 2000):

$$R_{\text{est}}=\frac{1}{\sqrt{\dfrac{2PLR}{3}+RTO\times\left(3\sqrt{\dfrac{3PLR}{8}}\right)}}\times\frac{8S}{RTT\times PLR\times[1+32(PLR)^2]}. \tag{1}$$

According to Eq.(1), available bandwidth R_{est} can be estimated by *PLR*, *RTT* (Round Trip Time), *RTO* (Retransmission Timeout) and packet size *S*. However, in wireless networks, packet loss and jitter may also be caused by radio link errors. Therefore,

* Project supported by the National Natural Science Foundation of China (No. 60302004) and the Natural Science Foundation of Hubei Province (No. 2005ABA264), China

when applying TFRC in wireless networks, a sender cannot identify the network congestion condition correctly, and this leads to inappropriate rate control, and then results in underutilization of the wireless channel (Chen and Zakhor, 2004). To address the issue, link layer ARQ is anticipated to be deployed. However, how to achieve strict delay guarantee using ARQ scheme is still a major consideration (Shen *et al.*, 2004; Hassan and Krunz, 2004; Nadeem and Agrawala, 2004). Raptis *et al.*(2005) indicated that retransmission is the main factor increasing average packet delay. Shen *et al.*(2004) proposed an analytical model to select optimal system parameters including the maximum retransmission times of the truncated ARQ scheme and the interface buffer size. IEEE 802.11 proposes a fragmentation mechanism via partitioning large packets into smaller fragments to increase transmission reliability and decrease packet transmission delay, but it lacks adaptive mechanism to adjust fragment threshold (Nadeem and Agrawala, 2004). It is obvious that improving delay performance of TFRC over wireless links, especially over multihop links, was not well addressed in previous works.

This paper introduces an analytical model to describe video transmission over multihop wireless channels with link level ARQ. An equation for adjusting packet size in transport layer is deduced via a delay-constrained theory proposed by Zhang (2004). ETFRC scheme is proposed based on a practical packet adjustment method to decrease retransmission delay and jitter. In addition, smooth PLR and probability of packet retransmission (PPR) evaluation method is proposed to decrease fluctuation. A rate discount policy is also proposed to guarantee smooth throughput in wireless channel.

ANALYTICAL MODEL

Performance degradation over wireless channel can be due either to collisions or bit error. According to IEEE802.11 DCF (Distributed Coordination Function) scheme, for both cases, the sender cannot receive acknowledgement from the receiver, thus will backoff and retransmit, until the number of retries reaches the threshold or the packet is transmitted successfully (Raptis *et al.*, 2005). An analytical model is introduced to describe packet retransmission events with link level ARQ. In advance, because RTS, CTS and ACK packets are much shorter, we make an assumption that packet error occurs only with DATA packets. In addition, we assume that all the hops face the same condition. Let M represent the total hop number, A denote retransmission times threshold and P_e mean packet error rate via every hop link. The analytical model describing the transmission process of a packet evolving via M hops can be simplified to the multilayer finite Markov chain as depicted in Fig.1. State (i,j) means the packet is processed by ith node and has already transmitted j times via this hop, where $0 \le i \le M$, $1 \le j \le A$. The state probability is denoted as $P(i,j)$. If j=1, one packet may go through via this hop directly, or else, it is blocked and retransmitted.

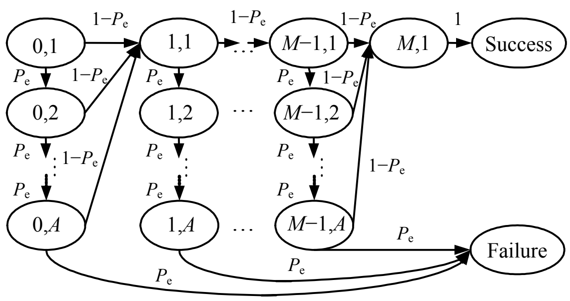

Fig.1 A multilayer finite Markov chain for packet transmission process via *M* hops

From this Markov chain, the probability of packet retransmission can be calculated by

$$PPR = 1 - \prod_{i=0}^{M-1} P(i,1) \times (1 - P_e). \tag{2}$$

Assuming that all retransmission events are due to generalized bit error (collisions or link error), we adopt GBER (generalized bit error rate) to denote the probability of generalized bit error events in every hop. We get $P_e=1-(1-GBER)^{S+H}$. The relationship between PPR and $GBER$ is:

$$PPR = 1 - (1 - GBER)^{M(S+H)}. \tag{3}$$

Let S represent packet size and H be the size of overhead including TFRC/UDP/IP/MAC/PHY layer header. Using Eqs.(2) and (3), we can calculate $GBER$ as

$$GBER = 1 - (1 - PPR)^{\frac{1}{M(S+H)}}. \tag{4}$$

DELAY-CONSTRAINED EQUATION FOR PACKET SIZE ADJUSTMENT

Theorem 1 Let bandwidth be BW bps and generalized bit error rate of one hop be $GBER$. There exists an optimal payload size S^* for reliably transmitting video streaming frame via theoretical Gaussian wireless channel with the least frame delay, which can be depicted by

$$S^* = \frac{-2}{M\ln(1-GBER)\times\left[1+\sqrt{1-\dfrac{4}{MH\ln(1-GBER)}}\right]}. \quad (5)$$

Proof If generalized bit error rate of M hops are $GBER_1$, $GBER_2$, …, $GBER_M$ respectively, the probability for successfully transmitting a K-bit data packet without retransmission at any hop P_s is given by

$$P_s = (1-GBER_1)^K(1-GBER_2)^K\cdots(1-GBER_M)^K. \quad (6)$$

Supposing $GBER_1=GBER_2=\ldots=GBER_M=GBER$, we get

$$P_s = (1-GBER)^{MK}. \quad (7)$$

Let I-bit streaming media frame be partitioned into F fragments, so that the size of each packet S is $S=I/F$ and that the overall size including packet size and overhead size K satisfies $K=(1/F)+H$. Then the probability for successfully transmitting I-bit streaming frame can be defined as

$$P_s = (1-GBER)^{M(I/F+H)}. \quad (8)$$

Supposing packets are retransmitted independently of other packets, the aggregate of equivalent end-to-end retransmission times $\{N_i\}$ is a sequence of independent and identically distributed random variables. The conditional probability that $N_i=k$ can be described as

$$P(N_i=k) = (1-P_s)^{k-1}P_s, \quad (9)$$

Hence, the expectation of N_i is computed by

$$\overline{N} = \sum_k k\cdot P(k) = \sum_k k(1-P_s)^{k-1}P_s = P_s^{-1}. \quad (10)$$

Eq.(10) represents that each packet must successfully transmit $\overline{N}$ times from end to end.

In wireless environment, end-to-end delay includes access delay, transmission delay and propagation delay, among which transmission delay can be decreased by packet fragmentation. Let end-to-end transmission delay be d as computed by

$$d = \frac{I/F+H}{BW}. \quad (11)$$

Then, the expectation of reliable transmission delay of I-bit frame can be described by

$$D = F\cdot d\cdot\overline{N} = F\cdot d\cdot P_s^{-1}. \quad (12)$$

To minimize delay, we can first compute the derivative of D with respect to F as:

$$\frac{\partial D}{\partial F} = \frac{1}{BW}\times\frac{H+(I+H\cdot F)\times[\ln(1-GBER)]\cdot M\cdot I\cdot F^{-2}}{(1-GBER)^{M(I/F+H)}} \quad (13)$$

Letting $\partial D/\partial F=0$ yields the following equation:

$$H = -(I+H\cdot F)\times[\ln(1-GBER)]M\cdot I\cdot F^{-2}$$
$$\Rightarrow H\cdot F^2+(M\cdot I^2+M\cdot I\cdot H\cdot F)\times\ln(1-GBER)=0. \quad (14)$$

The optimal packet number for least frame delay is solved from Eq.(14) and is expressed by

$$F^* = -\frac{1}{2}I[\ln(1-GBER)]\times\left[M+\sqrt{M^2-\frac{4M}{H\ln(1-GBER)}}\right] = -\frac{1}{2}I\cdot M[\ln(1-GBER)]\times\left[1+\sqrt{1-\frac{4}{H\cdot M\ln(1-GBER)}}\right]. \quad (15)$$

Finally, we have optimal packet S^* as:

$$S^* = \frac{I}{F^*} = \frac{-2}{M[\ln(1-GBER)] \times \left[1+\sqrt{1-\frac{4}{H \cdot M \ln(1-GBER)}}\right]}.$$

The least end-to-end frame transmission delay, frame delay for short D^* is estimated by

$$D^* = \frac{I}{(1-GBER)^{M(S^*+H)}} \times \frac{1+H/S^*}{BW}. \tag{16}$$

Thus, Theorem 1 is proved.

Eqs.(5), (15) and (16) express the least delay to guarantee reliable transmission for I-bit streaming frame and corresponding optimal packet size and optimal fragment number respectively. If hop number is 1 and the influence of content is ignored, $GBER$ equals BER and the above equations are similar to those deduced by Zhang (2004). Since the optimal packet size is only determined by $GBER$ when M and H are fixed, we can easily conclude from Theorem 1 that the packet size should adapt to the characteristic of generalized bit error rate. However, it is difficult to directly evaluate $GBER$ by Eq.(4). As PPR reflects generalized bit error events, we adopt PPR to estimate $GBER$ and get Lemma 1.

Lemma 1 Supposing the probability of packet retransmission via theoretical Gaussian wireless channel is PPR, the evaluation of optimal packet size to reliably guarantee least frame delay at ith interval is estimated by

$$S_i = -\frac{2K_{i-1}}{[\ln(1-PPR_{i-1})] \times \left[1+\sqrt{1-\frac{4K_{i-1}}{H\ln(1-PPR_{i-1})}}\right]}, \tag{17}$$

where K_{i-1} expresses the overall packet size of the $(i-1)$th interval.

Proof Given packet size of last adjustment interval S_{i-1} and overhead size H, according to Eq.(3), PPR_{i-1} is computed by

$$PPR_{i-1} = 1-(1-GBER_{i-1})^{M(S_{i-1}+H)}. \tag{18}$$

From the above equation, we have

$$M\ln(1-GBER_{i-1}) = \frac{\ln(1-PPR_{i-1})}{S_{i-1}+H}. \tag{19}$$

By introducing Eq.(19) into Eq.(5), we can get

$$S_i = -\frac{2(S_{i-1}+H)}{[\ln(1-PPR_{i-1})] \times \left[1+\sqrt{1-\frac{4(S_{i-1}+H)}{H\ln(1-PPR_{i-1})}}\right]}. \tag{20}$$

Let $K_{i-1}=S_{i-1}+H$ and then Eq.(17) is proved.

Theorem 2 If $GBER$ is bounded and satisfies $0<GBER_{\min}\le GBER\le GBER_{\max}<1$, Eq.(17) is convergent.

Proof From Eq.(17), we can get $\ln(1-GBER_{\max})\le \ln(1-GBER)\le\ln(1-GBER_{\min})$, which can be expressed as:

$$C_0 \le \frac{\ln(1-PPR)}{S+H} \le C_1 < 0, \tag{21}$$

where $C_0=M\ln(1-GBER_{\max})$, $C_1=M\ln(1-GBER_{\min})$. As $C_0\le C_1<0$, optimal packet size S satisfies the following inequation:

$$S_{\min} = -\frac{2}{C_0 \times \left[1+\sqrt{1-\frac{4}{H\cdot C_0}}\right]} \le S \le -\frac{2}{C_1 \times \left[1+\sqrt{1-\frac{4}{H\cdot C_1}}\right]} = S_{\max}. \tag{22}$$

Therefore S is bounded. According to the above analysis, we can also get $S_1<S_2$ when $GBER_1>GBER_2$. It namely indicates that Eq.(17) has monotonicity. Thus, Eq.(17) is convergent.

ETFRC SCHEME AND IMPLEMENTATION

Using Eqs.(1) and (17), we construct ETFRC scheme, which includes sender functionality on server and receiver functionality on client. The two functionalities can cooperate with each other to implement reliable rate control. The overall operation consists of two sequential periods, i.e., probe period and steady period, which is basically similar to TFRC. We pay attention to the second period. Our work includes 4 parts:

(1) The first part is modifying data structure of ACK packet by adding a PPR report field.

(2) The second part is adopting smooth PLR and PPR evaluation method to decrease fluctuations. The method introduces a weighted factor $\lambda=2^{-5}$, and defines exponential average value $\overline{P}$ to replace the statistical value P_i of PLR and PPR as follows:

$$\overline{P}_i = (1-\lambda)\overline{P}_{i-1} + \lambda P_i. \tag{23}$$

(3) The third part is to guarantee smooth throughput in wireless channel. We define a discount factor $\rho_i=1-PRR_{i-1}$. We have sending rate as

$$R_i' = R_i \rho_i. \tag{24}$$

(4) The fourth part is packet fragment mechanism. Let initial packet size S_0=1000 bytes. At the same time, in order to compare with TFRC, 1000 bytes is set to be the packet size threshold. If sender gets PRR_i from ACK, it computes packet size S_i according to Eq.(17). Subsequently, a practical packet fragment method is adopted as follows:

(1) Set packet size $S\in\Gamma$ and aggregate $\Gamma=\{S_1, S_2, \ldots, S_K\}$. Let K be an integer from 1 to 11 and S_1=100, S_{11}=1000. The step for adjusting S is 100 bytes;

(2) If $S_i\geq S_0$, then $S_i=S_0$; otherwise, if $S_i\in[S_K, S_{K+1}]$, S_K, $S_{K+1}\in\Gamma$, then $S_i=S_K$.

The ETFRC procedure is as follows:

(1) Receiver evaluates *PLR*, *PPR* and feeds them back to sender.

(2) Sender gets *ACK* and estimates *RTT* and available rate using Eq.(1);

(3) Sender discounts sending rate according to ρ and gets the practical sending rate;

(4) Sender determines packet size and sending interval via the above packet fragment mechanism.

PERFORMANCE EVALUATION

The simulation scenario is set as shown in Fig.2, which includes 15 wireless nodes, server node *S* and routing node *R* with two TCP sources and sink nodes. The wireless nodes access the wired network via base station node *BS*. There are 3 lines of nodes. Each line contains 5 mobile nodes. The horizontal distance of each adjacent node in a line is 150 m. The vertical distance of top line to middle line and middle line to bottom line is 200 m and 150 m, respectively. In wireless links, we adopt 802.11 as typical MAC type and adopt DSDV routing protocol. We also add wireless transmission interference simulating program to ns2.26. Default packet size is 1000 bytes. IEEE802.11 parameters adopt defaults in ns2.26. The size of overhead is 134 bytes including 8-byte UDP header, 20-byte IP head, 58-byte MAC and PHY header, and ETFRC header. The size of frame I is 3000 bytes.

Theoretical results

Let bandwidth be 1 Mbps and *GBER* change from 10^{-7} to 10^{-4}. Theoretical results of optimal packet size and relative improvement ratio (RIR) of frame delay relative to TFRC with fixed packet size are shown in Figs.3a and 3b, respectively. *X*-coordinate represents setting *GBER* using scientific notation; *Y*-coordinate expresses theoretical results.

Fig.3a shows optimal packet size decreases with increasing hop number and *GBER*. As shown in Fig.3b, frame delay can effectively dispense with using our packet fragment method, especially when *GBER* is larger than 5×10^{-6}. In the following simulation, we mainly test ETFRC performance at high bit error rate (10^{-5}~10^{-4}).

Simulation results

A good transport protocol for streaming delivery must have high goodput, low frame delay and frame jitter with good fairness. Two types of fairness including homogeneous and heterogeneous fairness

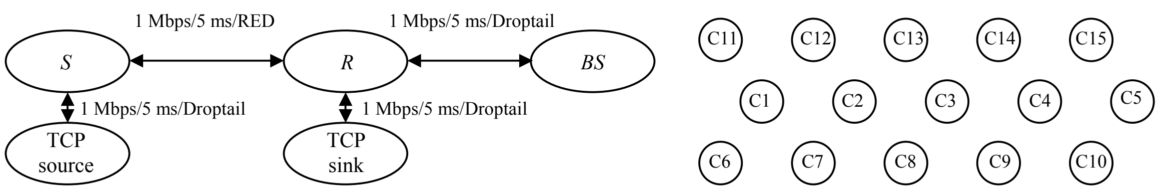

Fig.2 Simulation scenario

must be examined. To evaluate fairness of two flows at a given time scale, we adopt equivalence function as defined by Floyed *et al*.(2000). The value is named as fairness ratio in this paper, and is between 0 and 1. The closer it is to 1, the more "equivalent" the two flows are.

Firstly, let *BER* be 10^{-5}, 5×10^{-5} and 10^{-4}, respectively. Judging from the definition of GBER, it can be concluded that the value of *GBER* is greater than *BER*. Comparisons of ETFRC and TFRC during 400 s simulations are shown in Figs.4~6. From Fig.4a, *PLR* is dropped obviously when using ETFRC, especially when *BER* is 10^{-4}. As shown in Fig.4b, when *BER* is 10^{-5} or 5×10^{-5}, the two schemes get approximate goodput. When *BER* is 10^{-4}, improved ratios are 211.94%, 1267%, 2700%, 1625% and 3100% respectively. Fig.5 shows comparison of relative improvement ratio of average frame delay and average frame jitter using ETFRC relative to TFRC with different BER settings. From Fig.5, we can conclude that average frame delay and frame jitters are basically dropped more and more with increasing *BER*.

Secondly, this paper compares homogeneous fairness of ETFRC with that of TFRC as shown in Fig.6a when sender delivers the same type of traffic to two receivers with the same hop number. Basically, with different hop numbers and BER settings, homogeneous fairness ratio of ETFRC is better than TFRC.

Finally, Fig.6b shows comparison of heterogeneous fairness of ETFRC and TFRC with TCP flow. It can be concluded that values of fairness ratio decreased with increasing hop number and *BER*. When hop number is larger than 2, the fairness ratio of ETFRC is basically better than that of TFRC, especially when *BER*=10^{-4}.

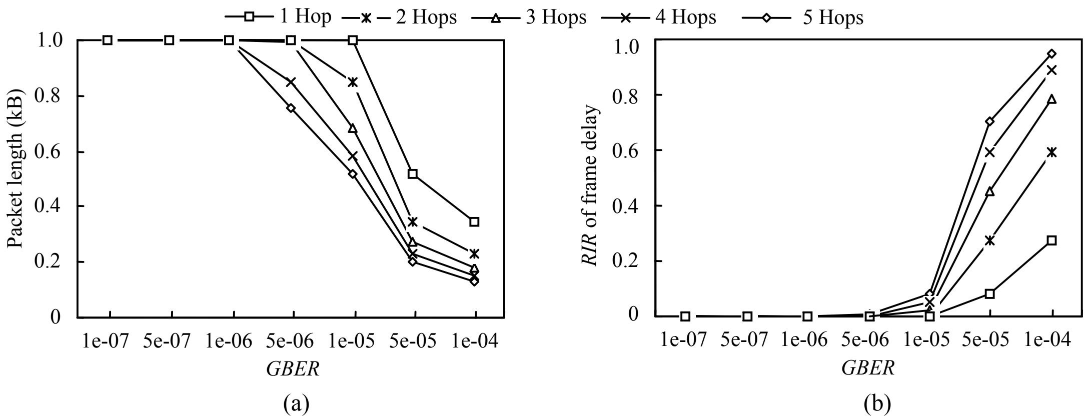

Fig.3 Theoretical result of (a) Packet size adjustment; (b) Relative improvement ratio of frame delay

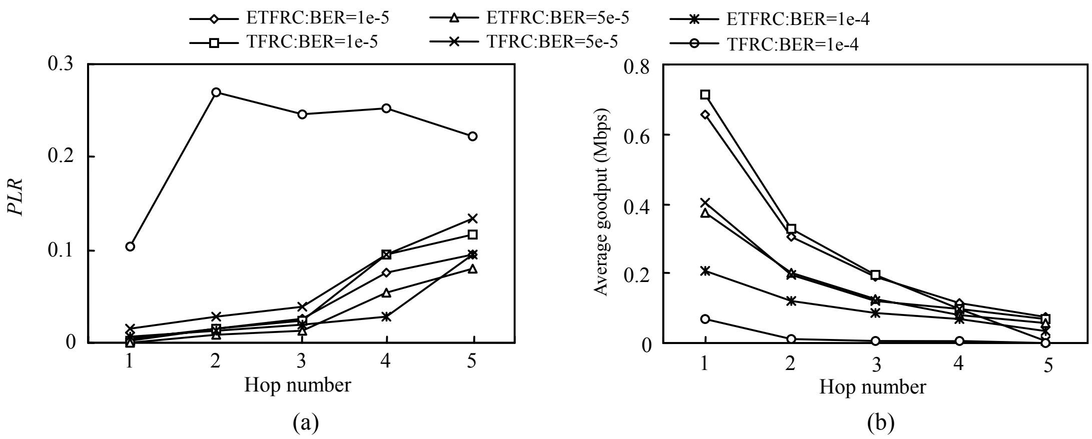

Fig.4 Comparison of (a) PLR with different BER settings; (b) Goodput with different BER settings

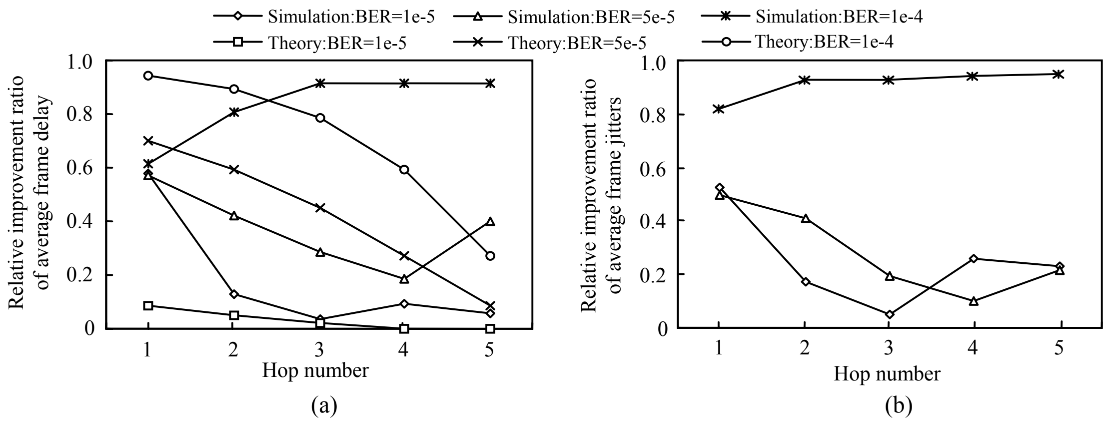

Fig.5 Comparison of relative improvement ratio of (a) Average frame delay with different BER settings; (b) Average frame jitters with different BER settings

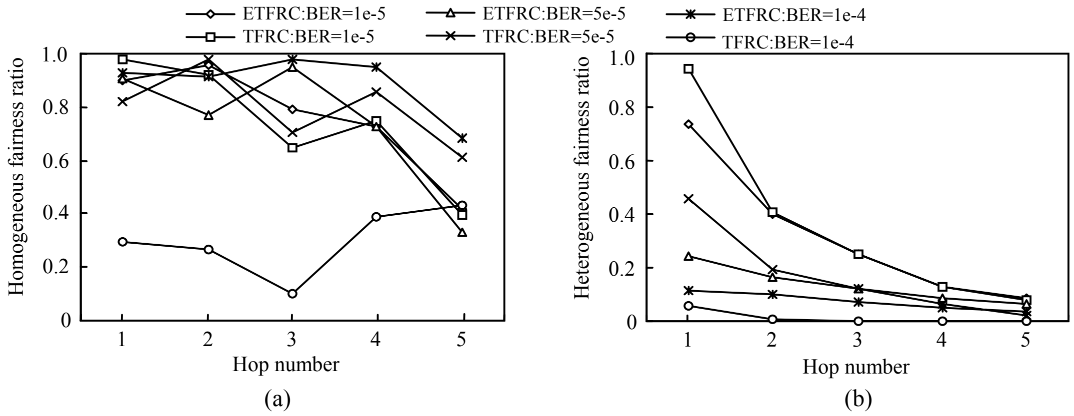

Fig.6 Comparison of (a) Homogeneous fairness with different BER settings; (b) Heterogeneous fairness with different BER settings

CONCLUSION

In order to reduce retransmission delay and modify shortage of packet fragment in the link layer, this paper proposes an enhanced TFRC with variable packet size. TFRC equation is first used to detect available bandwidth. Then PPR is used as the base of packet size adjustment and rate discount. Finally, sending packet rate is determined by sending rate and packet size. Simulation results indicate that ETFRC scheme can provide efficient rate control in error-prone situations, while maintaining fairness.

References

Chen, M.H., Zakhor, A., 2004. A rate control for streaming video over wireless. *Proceedings of the IEEE INFOCOM*, **2**:1181-1190.

Etoh, M., Yoshimura, T., 2005. Advances in wireless video delivery. *Proceedings of the IEEE*, **93**(1):111-122. [doi:10.1109/JPROC.2004.839605]

Floyed, S., Handley, M., Padhye, J., Widmer, J., 2000. Equation-based Congestion Control for Unicast Aplications. Proceedings of SIGCOMM, p.43-56.

Hassan, M., Krunz, M., 2004. A rate control scheme for video streaming over wireless channels. *Proceedings of the IEEE GLOBECOM*, **6**:4071-4075.

Nadeem, T., Agrawala, A., 2004. IEEE 802.11 Fragmentation-aware Energy-efficient Ad-hoc Routing Protocols. Proceedings of the IEEE International Conference on Mobile Ad-hoc and Sensor Systems, p.90-103.

Raptis, P., Vitsas, V., Paparrizos, K., Chatzimisios, P., Boucouvalas, A.C., 2005. Packet Delay Distribution of the IEEE 802.11 Distributed Coordination Function. Proceedings of the IEEE International Symposium on a World of Wireless, Mobile and Multimedia Networks, p.299-304.

Shen, H., Cai, L., Shen, X.M., 2004. Performance analysis of equation based TFRC over wireless links with link level ARQ. *Proceedings of the IEEE GLOBECOM*, **3**:1681-1685.

Zhang, Z.J., 2004. Guaranteeing end-to-end adaptive QoS in wireless multimedia networks. *Chinese Journal of Computers*, **27**(8):1064-1073 (in Chinese).