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RESEARCH ARTICLE

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Rate control for streaming media transmission over WLAN

Abstract In order to solve the problems of link layer retransmission and packet fragment strategies for IEEE 802.11, this paper proposes a variable packet TCP-friendly rate control (VPTFRC) scheme for streaming media transmission over wireless local area network (WLAN) by researching on the policy of packet size adjustment in transport layer based on the minimum retransmit-delay constraint. Unlike other proposals, this process considers the impact of wireless packet error ratio (PER) on the packet size adjustment and the performance of rate control. Simulation results demonstrate that our proposed process can simultaneously achieve higher throughput, better fairness, shorter transmission delay and less jitter than TFRC. Among them, the ratio of delay, jitter and packet loss rate can reach a maximum improvement ratio of 58%, 42% and 85% respectively.

Keywords IEEE 802.11, wireless local area network (WLAN), TCP-friendly, streaming media, link layer retransmission

1 Introduction

Remarkable progress in constructing wireless local area network (WLAN) based on a series of protocols of IEEE 802.11 has paved the way for wireless streaming transmission [1]. Streaming applications generally require continuous high bandwidth guarantee as well as stringent bounds on delays and jitters [1]. The user data protocol (UDP) is usually used to convey streaming media to guarantee real-time performance, but it may result in unfair competition with the TCP traffic that dominates in the Internet without effective traffic regulation. Therefore, the scheme of wireless transmission is required to keep capability to fairly compete for the wired bandwidth with current TCP flows, which is referred to as "TCP friendliness" [2]. In addition, it is noteworthy that wireless channels are fraught with challenges because of high-error and time-variable features. Therefore, rate control for wireless transmission aims to make the transmitting rate adaptive to wireless link status with both TCP fairness and a better quality of service (QoS).

The most representative rate control scheme with perfect TCP fairness is TCP-friendly rate control (TFRC), which is deduced from the throughput function of TCP Reno protocol and adopted by Internet Engineering Task Force (IETF). TFRC equation is described as [3]:

$$R_{\text{est}} = \frac{f}{t_{\text{RTT}}\sqrt{\frac{2p}{3}} + t_{\text{RTO}}\left(3\sqrt{\frac{3p}{8}}\right)p\left(1+32p^2\right)}. \tag{1}$$

According to Eq. (1), the available bandwidth R_{est} (the unit is bit per second) can be estimated by packet loss ratio (PLR, represented by p), the round trip time (RTT, represented by t_{RTT}) and retransmission time-out (RTO, represented by t_{RTO}). In addition, f represents the packet size (the unit is bit). The unit of t_{RTT} and t_{RTO} is second. Although TFRC is proved with better capability of TCP friendly congestion regulation in the wired network, its application in the wireless channel still meets challenges such as decreases in the throughput due to bit errors.

To address the issue, in the IEEE 802.11 protocol, link level automatic repeat request (ARQ) scheme is deployed to improve TFRC performance [2]. However, it also increases delay [4]. IEEE 802.11 proposes a fragmentation mechanism that partitions large packets into smaller fragments to increase transmission reliability and reduce packet delay, but it lacks adaptive mechanism to adjust to the fragment threshold. Therefore, Zhang presents a method of packet size adjustment considering bit error ratio (BER) [4]. However, operating the method will be affected by how to evaluate BER and also does not bear TCP fairness as well.

To improve the fragmentation mechanism in IEEE 802.11 and construct a rate control scheme adapted to

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the streaming transmission via WLAN, packet adjustment in the transport layer will be proposed.

First, the rate is adjusted by available bandwidth based on the TFRC equation. Second, packet adjustment is chosen by BER. Finally, the packet send rate will be determined by both the send rate and the packet size. The rest of this paper is organized as follows. In Sect. 2, a delay-constrained equation for adjusting the packet size is deduced. In Sect. 3, an available rate control scheme is depicted and in Sect. 4 extensive simulation results and performance comparisons on NS2.26 are presented.

2 Delay-constrained equation for packet size adjustment

Theorem 1 Let the size of the packet header be H (the unit is bit) and BER be e. There is an optimal payload size f^* for reliably transmitting the streaming media frame via a theoretic Gaussian wireless channel with the least frame delay, which can be depicted by [4]:

$$f^* = -\frac{2}{\ln(1-e)\left[1+\sqrt{1-\dfrac{4}{H\ln(1-e)}}\right]}. \tag{2}$$

We can easily draw a conclusion from Theorem 1 that to guarantee the least transmission delay, the packet size should adapt to the characteristic of the bit error. However, it is difficult to directly evaluate e, therefore, Eq. (2) should be simplified.

Lemma 1 Suppose the probability of the packet error in the theoretic Gaussian wireless channel is P_{e}, the evaluation of the optimal packet size to guarantee the least reliable frame delay at ith interval is estimated by:

$$f_i = -\frac{2K_{i-1}}{\ln\left(1-P_{\mathrm{e}}^{i-1}\right)\left[1+\sqrt{1-\dfrac{4K_{i-1}}{H\ \ln\left[1-P_{\mathrm{e}}^{i-1}\right]}}\right]}, \tag{3}$$

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

Proof P_{e} at $(i-1)$th interval (P_{e}^{i-1}) is computed by:

$$\begin{aligned} &P_{\mathrm{e}}^{i-1} = 1-(1-e_{i-1})^{f_{i-1}+H} \\ &\Rightarrow \ln(1-e_{i-1}) = \frac{1}{f_{i-1}+H}\ln\left(1-P_{\mathrm{e}}^{i-1}\right). \end{aligned} \tag{4}$$

Introduce Eq. (4) to Eq. (2). Let $K_{i-1} = f_{i-1}+H$, and then Eq. (3) is proved.

It can be easily proven that Eq. (3) is convergent. Both Eqs. (2) and (3) are defined as optimal delay-constrained equations for packet size adjustment to satisfy the status of a wireless network. Suppose the original packet size is 1000 B and the media frame size is 3000 B. H is 134 B. The maximum transmission unit (MTU) is 1500 B. If the packet size is larger than MTU, it equals MTU–H. If we set e within $10^{-8}-10^{-4}$, and list the results of the packet size adjustment and the effect of delay improvement, we can find that when $e \geqslant 10^{-5}$ the improvement is more evident than that in other setting. Among the results, the optimal packet sizes are 1229 B, 516 B and 348 B respectively when e equal 10^{-5}, 5×10^{-5} and 10^{-4}. The media frame transmission delay improves by 0.3%, 8.4% and 27.4% respectively compared with those of transmission by adopting the fixed packet size method.

3 VPTFRC scheme and implement

Based on Eqs. (1) and (3), we construct a variable packet TCP-friendly rate control (VPTFRC) scheme, which includes sender functionality on the server and receiver functionality on the client. The two parts are operated above the real-time transport protocol/real-time transport control protocol (RTP/RTCP), UDP and IP protocol. Acknowledge (ACK) packet is encapsulated by RTCP protocol. The two functionalities can cooperate with each other to implement a reliable rate control.

The overall operation consists of two sequential periods, i.e., probe period and steady period, which is basically similar to TFRC. Our work focuses on the second period, which includes:

1) In this paper a smoothing method to evaluate p and P_{e} is adopted on the side of the receiver functionality, which can make the throughput of VPTFRC more smoothing. Let the weighted factor $\lambda = 2^{-5}$, exponential average $\overline{P}$ can substitute the statistic of P_i (both represent p and P_{e}) via Eq. (5)

$$\overline{P_i} = (1-\lambda)\overline{P_{i-1}} + \lambda P_i. \tag{5}$$

2) A discount method to adjust the sending rate is proposed. Suppose the features of the data via wireless link keep steady, the processing delay is keep smoothing. If P_{e} at the ith interval (P_{e}^i) and the sending rate is R_i, the average transmission times is $1/(1-P_{\mathrm{e}}^i)$. We can also get available bandwidth of wireless link that is $(R_i/f_i)\left[1/\left(1-P_{\mathrm{e}}^i\right)\right]f_i$, which equals $R_i/\left(1-P_{\mathrm{e}}^i\right)$. This result is bigger than R_i, which makes the queue delay increase. Therefore, we define the discount ratio $\rho_i = 1-P_{\mathrm{e}}^{i-1}$, and the obtained discount sending rate is described by

$$R_i' = R_i\rho_i. \tag{6}$$

According to Eq. (6), the bandwidth of the wireless channel can be deduced by $R_i'/(1-P_{\mathrm{e}}^{i-1}) = R_i$, which equals to the adjustment bandwidth.

3) Packet adjustment scheme is proposed. Let the initial packet size $f_0 = 1000$ B. At the same time, 1000 B is set to be the packet size threshold to compare with TFRC.

If the sender gets P_e^{i-1} from ACK, it computes the packet size f_i according to Eq. (3). Subsequently, a practical integral packet fragment method is adopted as:

a) Set the packet size $f \in \Gamma$ and the aggregate $\Gamma = \{f_1, f_2, \ldots, f_K\}$. Let K be the integer among 1 to 10 and $f_1 = 100$, $f_{10} = 1000$. The step for adjusting f is 100 B.

b) If $f_i \geqslant f_0$, then $f_i = f_0$. Otherwise, if $f_i \in [f_k, f_{k+1}]$, $f_k \in \Gamma, f_{k+1} \in \Gamma$ then $f_i = f_k$.

The procedure of VPTFRC is designed as follows:

1) The receiver evaluates p, P_e and feedbacks them to the sender.

2) The sender gets ACK and estimates t_{RTT} and t_{RTO}, and available rate according to Eq. (1).

3) The sender discounts the sending rate according to ρ and obtains the practical sending rate.

4) The sender determines the packet size and sends the packet rate via the upper method of the packet size adjustment.

4 Performance evaluation

Rate control for streaming transmission requires smooth and higher throughput, lower delay and jitter, and TCP friendliness. Among them, effective parts of throughputs and their relative standard deviation (RSD) are adopted to analyze the throughput. Streaming frame delay and jitter is proposed to analyze real-time performance. Fairness function is adopted to check fairness performance, which is defined in Ref. [3]. When comparing two traffics, the value of the fairness function approximates to 1 and more fairness will be obtained.

4.1 Simulation setting

The network simulator NS2.26 and the environment shown in Fig. 1 are used to test the proposed scheme. TCP-Reno protocol is adopted for TCP traffics. N represents the number of sources and receivers. In wireless links, we adopt 802.11 as a typical MAC type and add wireless transmission interference simulating program to NS2.26. IEEE 802.11 parameters adopt default setting in NS2.26. The destination sequence distance vector (DSDV) routing protocol is adopted. The default packet size is 1000 B and used for TFRC. Backbone bandwidth is 1 Mbps. The size of the overhead is 134 B including 8 B UDP header, 20 B IP header, 58 B MAC and PHY header, whereas, the remainder is the TFRC header. Let the size of the streaming frame I be 3000 B.

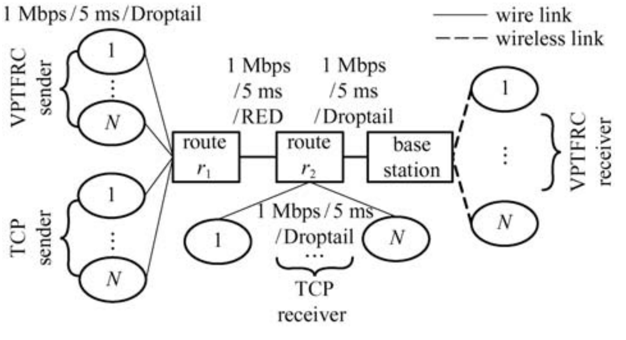

Fig. 1 Simulation scenario

4.2 Simulation results and comparisons

1) Let e be 5×10^{-5}. The results including the packet size (f B) and packet transmission delay (t_d) of different methods are shown in Figs. 2 and 3. Horizontal coordinate axis represents simulation duration (unit: s). The vertical coordinate axis shows the simulation results. The packet size and packet transmission delay of un-smoothed evaluation method, un-discount method and non-integrated method are respectively represented by f_{us}, $t_{\mathrm{d}}^{\mathrm{us}}$, f_{ud}, $t_{\mathrm{d}}^{\mathrm{ud}}$, f_{ui}, $t_{\mathrm{d}}^{\mathrm{ui}}$, compared with f_{v}, $t_{\mathrm{d}}^{\mathrm{v}}$, of VPTFRC method. Simulation results testify that the packet adjustment turns to be more smoothing and the packet transmission delay can be decreased by adopting VPTFRC method.

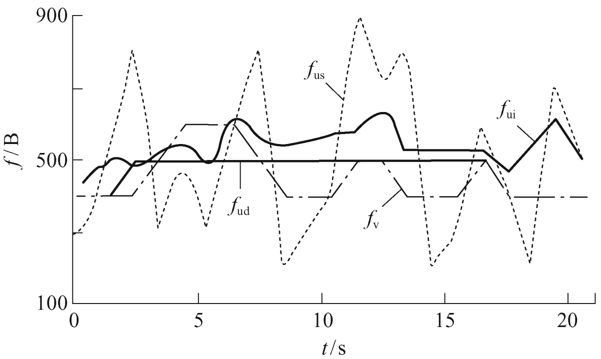

Fig. 2 Comparison of packet size adjustment

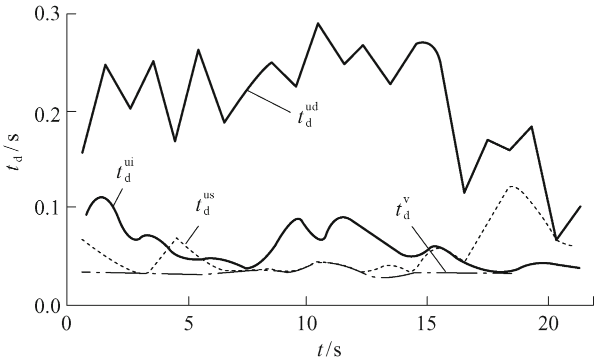

Fig. 3 Comparison of packet transmission delay

2) Comparison of the improvement ratio of different performance indexes during 400 s simulation with different e setting durations is shown in Table 1. Among them, L_{T}, L_{TRSD}, L_{DI}, L_{JI}, L_{P} and L_{F} represent the improve-

Table 1 Comparison of improvement ratio of different performance indexes/%

items	e		
	10^{-5}	5×10^{-5}	10^{-4}
L_{T}	35.80	27.20	153.10
L_{TRSD}	130.10	65.60	27.20
L_{DI}	11.50	58.00	−5.40
L_{JI}	−47.10	22.80	42.10
L_{P}	85.00	58.33	51.54
L_{F}	27.27	45.00	75.00

ment ratio of average throughput, RSD of average throughput and frame delay, PLR and fairness ratio, respectively. As shown in Table 1, the average throughput gets better when e is increasing. When e equals 5×10^{-5}, the improvement ratio of the average frame delay and its RSD is better than that of the other setting. When e equals 10^{-4}, the RSD of the average frame delay is improved. Whereas, when e equals 10^{-5}, this index is affected by augmenting the delay process because the throughput increases. In addition, the improvement ratio of PLR and fairness ratio are satisfactory at different settings. Hence, it is concluded that the overall performance of VPTFRC is better than that of TFRC. Simulation results demonstrate that VPTFRC is robust and effective against error feature and network congestion events. PLR, throughput, frame delay and jitter can be effectively improved when adopting VPTFRC. Therefore, the QoS of streaming media transmission accessed by WLAN is effectively guaranteed.

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