ISSN : 0974 - 7435 Volume 10 Issue 13

2014

BioTechnology

An Indian Journal

FULL PAPER

BTAIJ, 10(13), 2014 [7051-7057]

A self-adapted QoS mechanism in wireless multimedia networks

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ABSTRACT

This paper applies TCP/IP behavior characteristics and wireless link BER change to establish a wireless heterogeneous model with binary-state Markov error model, to describe the practical congestion degree in network more accurately. Then, the price mechanism of REM algorithm is adopted to design a congestion control method for rapid response and queue cache rate. By comparing the receiving rate of access point, link bandwidth and the rate of queue on node buffer are used to control queue length, so the adaptability of queue changes is improved. Then, by means of analyzing lacking price production mechanism in standard random exponential marking algorithm, the maximum frame is introduced into REM price mechanism so that it can be more adapted to wireless environment. In addition, queue cache ratio and link virtual capacity are taken as congestion measurement index and a new price-based congestion control strategy is proposed. Simulation results show that our scheme shows better performance and advantage on adaptability of dynamic bandwidth as well as QoS assurance of multimedia transmission delay, which also effectively improves the utilization of link bandwidth.

KEYWORDS

QoS; Wireless multimedia network; Congestion; REM; Bandwidth.

INTRODUCTION

Due to practical convenience of wireless service in life, more and more current multimedia service will basically be provided through wireless transmission. However, it obviously needs some support of key technologies to implement these services such as wireless transmission technology of video stream media[1,2]. In particular, the quality of services (QoS) is one of the most important technologies to influence performance of wireless video service. Different from traditional data service (FTP, HTTP web page browsing, E-mail, etc), the multimedia transmission has high requirement on time delay. Meanwhile, when multimedia service data is transferring in wireless network, there also appears the problem that it can be ignored in wired network. That is, random bit mistake is caused by environment of transmission channel and it is mainly determined by transmission characteristics of wireless channel. Due to the mismatch of access point rate between wired network and wireless network, serious congestion will be caused.

The bandwidth resource of wireless link lacks dynamic change, so active queue management algorithm cannot be directly introduced into the research of wireless network. Reference[3] applies the multiplication of MAC layer packet service time and retransmission number of packet as networked congestion indication to improve drop probability function of original RED algorithm. This algorithm has perfect adaptability towards dynamic change of wireless bandwidth. Reference[4] is based on QoS requirement of multimedia average access time delay to dynamically change the target queue length, so as to guarantee QoS requirement of multimedia transmission time delay. However, the algorithm is not enough for dynamic change of wireless bandwidth. Reference[5] considers that too large dropping probability only increases more packets drop and it is not effective for congestion control towards wireless network of longer RRT transmission time delay. Therefore, an improved RED/ARED is proposed to increase one time dropping probability through restricting the largest dropping probability as 0.1 as well as when average queue length is more than the maximal threshold value. This algorithm does not have adaptability towards wireless bandwidth of dynamic change.

Our research focuses on the time variation of wireless that influences AQM algorithms. We adopt adaptive queue length and the maximum delay of multimedia transmission when design the AQM controller. Then by the analysis on the defects of standard REM, we introduce maximum frame delay to generate the price to enhance its adaptability in wireless environment. After the congestion degree is determined, the loss rate will be improved in a phased mode. The experiments show that, compared to similar algorithms our control scheme in wireless network has better adaptability of bandwidth and its has advantage in QoS ensurence of multimedia transmission delay.

PRICE-BASED ACTIVE QUEUE MANAGEMENT ALGORITHM

In the early research on wired network, there appear many AQM algorithms, such as RED, PI and REM[6]. The conception of price in REM proposes a novel idea for the traffic control. AS the congestion performance indicator of REM, the price is decided by two factors: one is the difference between instantaneous queue length and target queue length; the other is the difference between packet arriving rate and bandwidth. The weight of above difference contains the price. REM computes the price change by the accumulation of the whole link price, reflecting the congestion state of the network and adjusting the congestion windows of sending terminal.

The price of REM is updated by equation 1 when each packet arrives at the router:

$$p_t(t+1) = [\, p_t(t) + \gamma(\alpha(q_t(t) - q^*) + x(t) - c_i)]^+ \qquad \textbf{(1)}$$

α and γ are weighted parameters. $q_t(t)$ and q^* denote the instantaneous queue length and target queue length. $x(t)$ is packet arriving rate and c_i is link bandwidth. When the price updating equation is decided, the loss rate of REM adopts the index-mode of link price value, as equation 2 shows:

$$P_{drop} = 1 - \varphi^{-p_t(t)} \qquad \textbf{(2)}$$

In the analysis of reference[7], we define $cm(t)=\gamma(\alpha(q_t(t)-q^*)+x(t)-c_i)$ as instantaneous congestion indicator, then

$$cm(t)=\beta e(t)+\gamma\Delta e(t) \quad \textbf{(3)}$$

$e(t)=q_t(t)-q^*$ is the queue length error, we rewrite the updating equation as:

$$p_t(t+1)=\sum_{l=0}^{t}cm(l)=\sum_{l=0}^{t}(\beta e(l)+\gamma\Delta e(l))=\beta\sum_{l=0}^{t}(e(l)+\gamma e(l)) \quad \textbf{(4)}$$

From equation 4 we find that, similar to PI, REM cannot effectively recognize the congestion state of the link. It just pays attention to the performance of steady state. So when the price of link is close to ∞ the loss rate can reach 1, which will create many defects to control the queue. Considering the features like high time-delay and big packets loss rate in wireless network, it is hard for REM to be suit for such kind of network.

RESEARCH ON SELF-ADAPTED REM ALGORITHM

Principle idea

Based on current research of wireless stream media, the key problem to influence flow is how to judge dropping packet phenomenon. However, the key problem is to judge whether the packet loss rate and the method of judging dropping packet rate is in effective time complexity scope. In existing methods, discontinuous data packet and transmission timeout of data packet are often adopted. However, the packet loss method of these two kinds of data packets is not suited to the situation accuracy of error control. Because is it is very easy to cause transmission delay. At the same time, unstable factors in data transmission link such as packet jitter, packet order turnover, etc, will also cause large interference on this research.

In wireless multimedia network, due to dynamic change of link bandwidth, the RTT in loop time generates large fluctuation which causes serious delay jitter. This seriously affects the service quality of multimedia. So the design idea of this paper is satisfying QoS requirement of multimedia transmission delay, and dynamically adapting the wireless bandwidth change. So the RTT in loop time can satisfy the requirement of QoS and reduce the delay jitter as much as possible. As known from reference[8], under the condition that propagation delay is hardly adjustable, we take changing target queue length of active queue as the realization target. To control target queue length of active queue more accurately, the overall anadiplosis time RTT can be remained in a certain scope. Then a price-adapted REM algorithm (PAREM) is proposed to solve the problems mentioned above.

Determination of congestion degree

We first provide the definition of the maximum frame delay $D_{\max}$ as: the maximum end-to-end delay of multimedia data packets, from sending end to target end, and meet QoS demand of transmission delay. No error probability of a packet with N-bit in ideal wireless link is $P_i=(1-BER_p)^N$. Assuming the size of multimedia frame is I bits, then its error probability is $P_i=(1-BER_p)^{(I/F+H)}$. F and H denote the number of packets and the size of packet head. Q denotes the instantaneous queue length. Then the maximum frame delay is:

$$D=\frac{F}{P_i}RTT=\frac{(I+FH)/c_i+FQ/c_i}{(1-BER_p)^{(I/F+H)})} \quad \textbf{(5)}$$

In heterogeneous network, since the channel interference of wireless link, packet random dropping will often occur, we use a binary-state Markov error model[9] to express packet random loss of wireless link under wireless environment and we propose that all bit error rate (BER) of wireless link corresponds to this model.

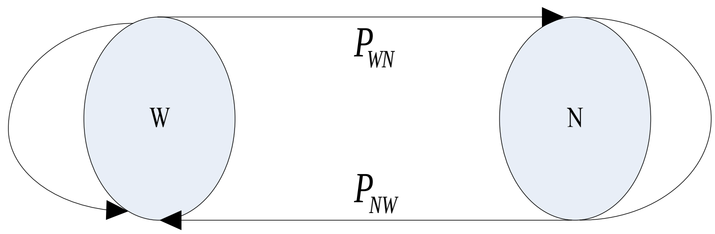

Figure 1 : Two-state Markov error model

The bit error rate (BER) of wireless link is unstable by the factors like channel disturbance. In the two-state Markov error model, we believe there are two states in the wireless link: normal and worse, represented as N and W. In state N, BER of the link is low; while in state B, BER is high. Since the wireless network often changes between these two states, we use P_N and P to denote the loss rate of two states. Similarly, P_{WN} and P_{NW} denote the transition probabilities from W to N, and from N to W. So the probability of wireless network in state N or W is $P_{WN}/(P_{WN}+P_{NW})$ and $P_{NW}/(P_{WN}+P_{NW})$. Further we deduce the equation of average error rate as:

$$BER_P = P_N P_{NW}/(P_{WN}+P_{NW}) + P_W P_{WN}/(P_{WN}+P_{NW}) \quad \textbf{(6)}$$

To ensure the QoS of multimedia transmission, the value of D should be controller to a smaller one. In ideal case, the queue length of intermediate node is 0, and the sending rate of end system equals to bandwidth. The burst feature of traffic and time variation of wireless link cause that the ideal case cannot be acquired. So when $D \le D_{\max}$, the congestion degree is not too high, we can further determine the status of link by setting some thresholds, to differentiate the congestion degree more accurately. In the following sections, this paper introduces a new threshold D' as the separation to better depicts the reason of queue overflow. That is, when $D \le D'$, it means that is probrably caused by the disturbance of wireless link. To ensure QoS of multimedia packets, we will wait to see the following status of queue buffer, until $D' \le D \le D_{\max}$, which means a real congestion begins. So in the following section, we will discuss the loss rate computation under different congestion degree.

Computation of loss rate

In the computation of packet loss rate, to acquire more sensitive result for queue changing, we add squared term to the price of REM. So the response to queue in router is in a timely manner. When sudden traffic occurs in the network, the squared error get rapid increases which leads to higher price. Then the active packet loss can decrease the length of instantaneous queue to be close to target value. Simultaneously, to avoid overly sensitive, we only adopt squared term instead of high orders to stabilize the queue length which may cause intensive jitter to influence the algorithm. Combined with the above analysis, the price updating equations are:

$$\begin{cases} Droptail\ , & D \le D' \\ p_t(t+1) = [p_t(t) + \gamma(\alpha(q_t(t) - q^*) + x(t) - c_i)]^+ \ , & D' \le D \le D_{\max} \\ p_t(t+1) = [p_t(t) + \gamma(\alpha(q_t(t) - q^*) + x(t) - c_i)^2]^+ , & D \le D_{\max} \end{cases} \quad \textbf{(7)}$$

The control framework of improved algorithm is shown in figure 2. $f(e)$ and $f(pr)$ are transformation functions. The input of $f(e)$ is queue error e and the output is $e^2 \operatorname{sgn}(e)$. The input of $f(pr)$ is price $\mathrm{P_r}$ and the output is loss rate P_{drop}.

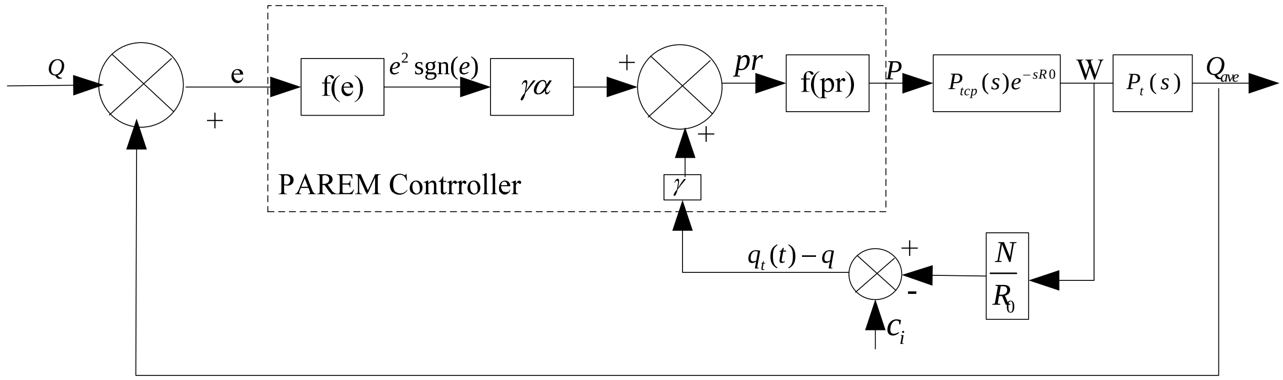

Figure 2: Control framework of PAREM

SIMULATION ANALYSIS

The simulation environment is set as a complicated one which includes changing data traffic, changing wireless BER and changing link bandwidth, to verify our method with other similar algorithms. The simulation holds 50 seconds. At the previous 10 seconds, the number of business flow based on UDP is added from 0 to 100. The duration time is 40s and they are terminated at 50s. The BER conforms to two-state Markov model and we set the initial BER of link as 0.01.

Figure 3 depicts the link utilization of different AQM algorithms. It is shown that the link utilization of PI is the worst, cased by poor performance in dynamic network. The link utilization of PAREM is higher than Proxy-RED, REM and Droptail. Because it both adopts the factors of queue length and the maximum frame delay as control indictor. It can effectively respond to the dynamic environment of wireless link.

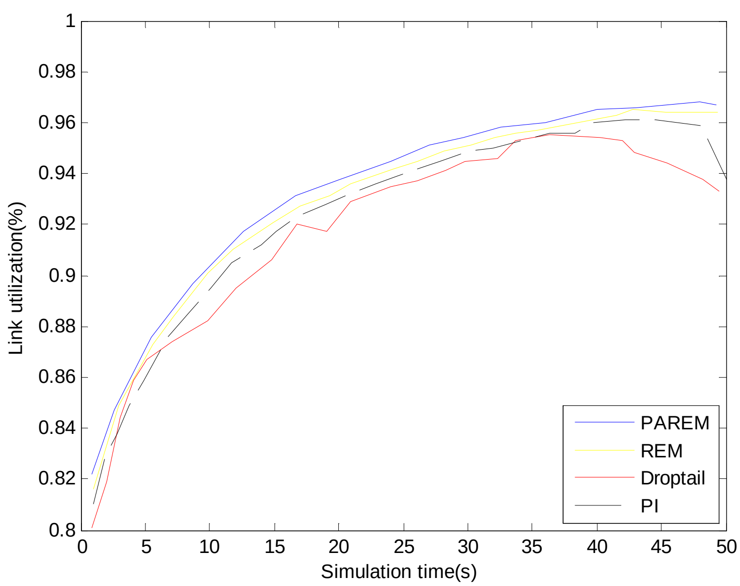

Figure 3 : Link utilization of different AQM algorithms

From figure 4 we see that given 3 algorithms are similar in their throughput. REM and PAREM have better performance. In the fact of loss rate, since we require better QoS of multimedia transmission based on bandwidth changing, the larger loss rate of PAREM is believed to be reasonable. When the bandwidth of wireless link gets smaller, the loss rate should be increased to adapt to the change of bandwidth, avoiding more congestion occur latter.

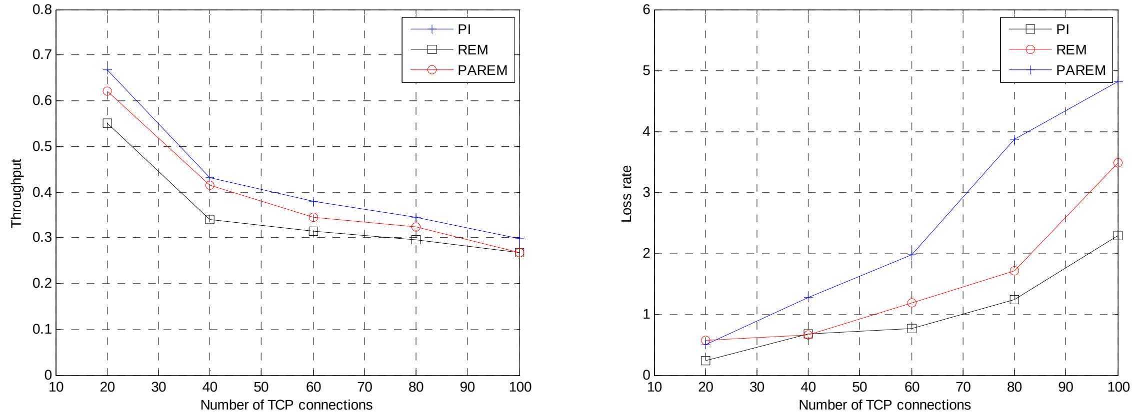

Figure 4 : Comparison of throughput and loss rate

In order to compare the performance among PAREM, REM, and PI algorithm under different BER of wireless links, we individually provide the simulation under the condition that average bit error rate of wireless link is 0.01%,0.1%, 0.5%, 1%,2%, 5%, respectively with 100 seconds simulation time. Figure 4 describes the throughput comparison in different bit error rates of wireless links with 3 algorithms. From this figure, it is easy to see that three kinds of algorithms throughput is decreasing with increasing wireless link bit error rates. It is because that the TCP congestion control algorithm will cause unnecessary retransmission with increasing bit error rate to decrease congestion window size and cause throughput decrease. When BER of wireless link is smaller than 0.1%, the throughput difference among three algorithms is not large. When BER of wireless link is larger than 0.1%, the throughput of REM and PI is far more than that in PAREM. Since REM and PI algorithms adjust the congestion window size just by RTT. However, the throughput of PAREM algorithm is a bit higher than REM algorithm because it adopts the maximum frame delay to improve the response speed in retransmission mechanism. Through this experiment, it is proved that the wireless link random error of PAREM has better adaptability in wireless environment.

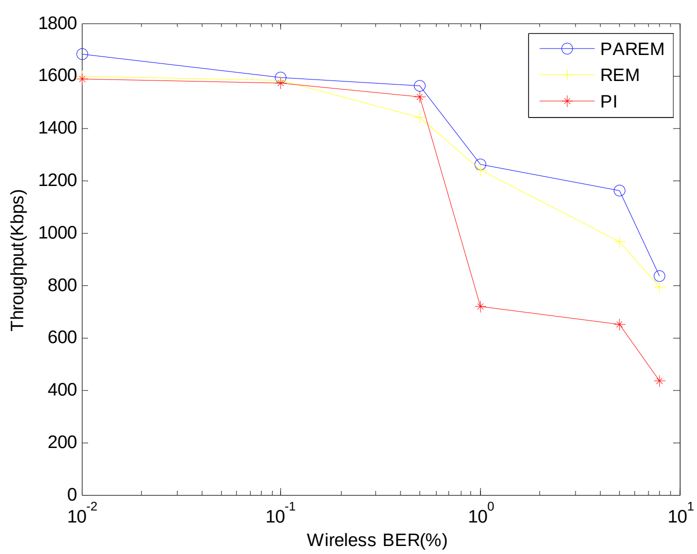

Figure 4 : Throughput under different BER

CONCLUSION

In wireless accessing network, due to high delay and uncertainty of environment, the packet loss is more serious than that of wired network and the packet loss rate of queue control-based AQM algorithm is even higher. In terms of this situation, we propose an adaptive method to design and realize congestion control algorithm satisfying QoS requirement of multimedia transmission time delay. By means of the simulations in NS2, under different TC'P connections, our algorithm has better performance in bandwidth utilization than REM, PI and Proxy-RED. Its average queue length does not have obvious relationship with TCP connection which also improves the stability of wireless access network.

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