A NEW MULTIMEDIA PACKET LOSS CLASSIFICATION ALGORITHM FOR CONGESTION CONTROL OVER WIRED/WIRELESS CHANNELS

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ABSTRACT

In a wireless network environment, common channel errors due to multipath fading, shadowing, and attenuation may cause bit errors and packet loss, which are quite different from the packet loss caused by network congestion. In congestion control, the packet loss information can serve as an index of network congestion for effective rate adjustment; therefore wireless packet loss can mistakenly lead to dramatic performance degradation. This paper proposes a packet loss classification algorithm based on trend detection of relative one-way trip time (ROTT) when it falls in the ambiguous zone where the packet loss classification is not straightforward. We show that the proposed algorithm greatly benefits rate-based congestion control algorithms for multimedia over IP networks.

1. INTRODUCTION

Congestion control for TCP/UDP flows is an active topic in multimedia networking. Most of the existing congestion control algorithms deal with end points to adjust the sending rate of a controlled flow based on observed network conditions, such as delay statistics and packet loss information. TCP is the most popular protocol reacting to this information by regulating the sending rate with additive-increase and multiplicative-decrease (AIMD) mechanism. Rate-based congestion control for multimedia streaming applications generally try to be friendly to TCP flows, by either equation-based congestion control [1][2] or AIMD rate adjustment [3][4] to approach inter-session fairness. Congestion control of PLM [5] introduces the concept of utilizing available bandwidth estimation tools (e.g., packet pair [6]) for layered video streaming. Other tools for estimating available bandwidth, such as Pathload [7], IGI [8], Spruce [9], and Pathchirp [10], can also be adopted to assist congestion control. In [17], a receiver-driven layered multicast protocol also suggests using an available bandwidth tool, which is similar to Pathload, for receivers to subscribe to the most suitable layers.

In addition to the implicit bandwidth inference, most congestion control algorithms also rely on packet loss information to indicate network congestion. However, for network topology containing wireless links, packet loss can be caused by either congestion loss or wireless channel errors, resulting from mulitipath fading, shadowing, or attenuation. Packet loss due to wireless channel errors will result in improper sending rate reduction and dramatically throttle the throughput.

There have been studies to improve TCP/UDP over mobile networks. The Snoop protocol [11] uses a retransmission mechanism for the lost packets at the base station to improve TCP throughput. Westwood TCP [12] exploits TCP acknowledged packets for the sender to estimate the bandwidth and adequately adjust slow-start threshold and congestion window. Lee et al. [20] explores the linear relationship between the probability of packet loss and packet size, under the assumption of uniformly distributed wireless errors. Elaarag [13] surveys various techniques for TCP to improve performance over wireless networks.

Another approach of this subject is to perform packet loss classification so that congestion control algorithms can more effectively adapt the sending rate based on congestion loss instead of from wireless loss. Biaz and Vaidya [14] suggest using the inter-arrival time at the receiver to discriminate congestion loss from wireless loss, so that a sender can respond appropriately. Spike-train in [15] provides two predefined thresholds on relative one-way trip time to classify packet loss. Cen [16] investigates packet loss classification algorithms introduced in [14][15] and proposes a *ZigZag* scheme which uses different threshold values based on the mean and deviation of ROTT for different number of lost packets. It is reported in [16] that neither ZigZag nor methods in [14][15] could perform well at different network topologies and suggests a switching algorithm based on [14][15] and ZigZag, depending on various values of ROTT.

The main drawback of using thresholding on either packet inter-arrival time or delay time is that it may cause misclassification of packet loss, because it is difficult to conclude that congestion loss and wireless loss will exhibit distinct boundary on either packet inter-arrival time or packet ROTT time.

In this paper, we propose a packet loss classification algorithm based on the trend of ROTT to assist packet loss classification in the ambiguous area of ROTT distribution. To demonstrate the effectiveness of our proposed packet loss classification algorithm, we perform network simulations based on a multilayer multicast congestion control algorithm, Bandwidth Inference Congestion control (BIC) [17]. Furthermore, this packet loss classification algorithm can also assist other congestion control protocols that might lead to unnecessary bandwidth reduction in the presence of wireless packet loss.

The rest of this paper is organized as follows. Section 2 presents the proposed packet loss classification algorithm. We describe an error model for wireless channel in Section 3 and present simulation results in Section 4 and 5, followed by the concluding remarks in Section 6.

2. THE PROPOSED PACKET LOSS CLASSIFICATION ALGORITHM

We consider two packet loss classes, *congestion loss* and *wireless loss*. Generally, the classification algorithms of packet loss depend on analysis of statistical behavior of some observed values, such as packet timestamp and packet serial number in the packet header. Spike-train in [15] and ZigZag in [16] investigate the ROTT difference between these two classes of packet loss. Unfortunately, these methods may produce unreliable classification performance when ROTT is around the threshold.

In the proposed classification method, we also exploit the ROTT of received packets to assist packet loss classification. In this session we first examine the chosen packet-loss classification index, ROTT, and then describe our packet classification algorithm.

2.1. Network Congestion, Packet Loss, and ROTT

ROTT is defined as the relative one-way trip time measured by the receiver as the time difference between the receiving time and the packet sending timestamp recorded in the field of multimedia UDP packet header plus a fixed bias.

The end-to-end packet delay can be modeled as the summation of following quantities: propagation delay for the electromagnetic waves to traverse all the link media along end-to-end path, queuing delay which is the main cause of congestion that leads to packet loss, and router processing delay which is required for the router to multiplex, reassemble, and forward packets. Propagation delay and router processing delay are usually constant for a given end-to-end path and invariant packet length. We can conclude that packet delay information infers both network congestion and packet loss resulting from network congestion. The model of packet delay can be summarized by the following equation:

$$\begin{aligned} &T_d = \sum_i T_{q,i} + \sum_i \frac{P_s}{C_i} + \sum_i T_{p,i}. \\ &T_{p,i} \propto P_s. \end{aligned} \tag{1}$$

Where T_d is the packet delay, $T_{q,i}$ is the queuing delay of link i,, $T_{p,i}$ is the router processing delay, and P_s is the packet size.

In Pathload [7], the basic idea is that the one-way delays of periodic probing packets show increasing trend of delay when the probing rate is larger than the available bandwidth. BIC [17], as a modified Pathload congestion control, agrees this proposition and further suggests that the variance of packet size can be taken into account of trend decision so that the proposition still holds for time-variant packet size. BIC proposes a delay model similar to Equation (1) except that in BIC the router processing delay is assumed to be packet-size independent. Nevertheless, this difference will not affect delay trend decision in BIC.

Since network congestion is directly related to congestion packet loss, we suggests the following proposition based on similar reasoning as in Pathload,

Proposition: When a packet loss is observed at time t, it should be considered as a congestion loss if T_d is in an ascending phase; otherwise it is categorized as wireless loss.

Based on this proposition, we propose a packet loss classification algorithm that uses trend detection for packet loss in the gray zone of ROTT**,** which is defined to be the interval between TG^{up} and TG^{low}, where TG^{up} denotes the upper bond of the gray zone and TG^{low} is the lower bound of the gray zone. When the receiver observes packet loss from the information of packet serial number, and the ROTT of the received packet is greater than TG^{up}, the packet loss will be classified as congestion loss. If the ROTT is smaller than TG^{low}, the packet loss will be classified as wireless loss. For ROTT falls in the gray zone, a trend detection process is performed to classify packet loss.

2.2. Proposed Packet Loss Classification

We propose a trend detection algorithm based on a moving average of trend index S_f:

$$S_f = (1-\gamma)\cdot S_f + \gamma \cdot I(D_i > D_{i-1}). \tag{2}$$

Where $I(X)$ is defined as 1 if X is valid, and 0 otherwise; D_i is the ROTT of the i^{th} packet and γ is the smoothing factor of S_f. Our simulations reveal that γ =1/30 achieves the best results. S_f can take value from 0 to 1. S_f will be around 0.5 if D_i is randomly distributed without increasing trend. If there is a strong increasing trend, S_f will approach unity. A threshold $S_{f,th}$ is used so that if $S_{f,th}<S_f$, the packet loss will be classified as congestion loss. For the sake of packet loss classification used in congestion control, we choose a relatively conservative value $S_{f,th}=0.4$ such that we would assume a packet loss as congestion loss at the ambiguous moment due to the concern of congestion control stability.

The proposed packet loss classification scheme can be summarized as in Fig. 1. We define the upper and lower bound of the gray zone of ROTT as in Equation (3).

$$\begin{aligned} TG^{up} &= ROTT_{\min} + \alpha(ROTT_{\max} - ROTT_{\min}), \\ TG^{low} &= ROTT_{\min} + \beta(ROTT_{\max} - ROTT_{\min}). \end{aligned} \tag{3}$$

Where α and β control the range of the gray zone and are empirically chosen to be 0.8 and 0.3 respectively. $ROTT_{min}$ and $ROTT_{max}$ are the minimum and maximum of ROTT, respectively. From our experiments, it shows that α and β are not very sensitive to the chosen values.

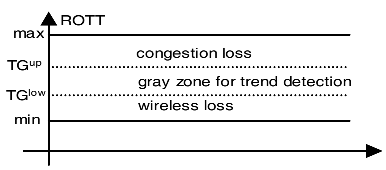

Fig. 1. Proposed packet loss classification range.

3. WIRELESS CHANNEL MODEL

Gilbert/Elliot's two-state Markov chain model [18] is a popular model to simulate the bursty nature of multipath wireless fading. The model assumes two states, good and bad channels, with the following transition matrix:

$$P = \begin{pmatrix} P_{gg} & P_{gb} \\ P_{bg} & P_{bb} \end{pmatrix}. \tag{4}$$

The transition probability P_{xy} is the probability of transition to state y given that the current state is x. The average bit-error rate (*ber*) P_b can be expressed as:

$$P_b = \frac{P_{gb}}{P_{gb} + P_{bg}}. \tag{5}$$

It has been reported that Gilbert/Elliot model is suitable for short-term instead of long-term error correlation [19]. For long-term bursty errors, block interleaving is usually adopted to remove the long-term bursty phenomenon. Besides the interleaving technique, channel coding, such as block coding and convolutional coding, is also an important component to reduce errors induced by impaired channels. In our simulations, (7, 4) Hamming code is applied to correct possible single error for each code word. The packet-error rate of this model will increase if packet size increases.

For P_{bb}=0.2 and P_{gb}=0.00005, packet size =500 bytes, and the number of packets is 5000, the average packet-error rate is about 4.36%. Without the (7, 4) Hamming code, the packet-error rate is 21.42%. The bit-error rate is 7.18e-5 while the theoretical *ber* from Equation (5) is about 6.25e-5.

4. PACKET LOSS CLASSIFICATION FOR MULTILAYER MULTICAST PROTOCOL

The knowledge of packet loss classification (PLC) can improve the performance of congestion control algorithms as long as they use packet loss information as a mean of congestion avoidance. To demonstrate the effectiveness of the proposed PLC algorithm, we use BIC [17] for multilayer video multicast as our test-bed.

The BIC for multilayer multicast performs periodic probing by the sender for receivers to analyze the trend of congestion. A receiver joins an additional layer if the following conditions hold. (1) There is no packet loss during the probing. (2) There is no trend of congestion. (3) Receiving rate during probing is at least 90% of the target rate. On the other hand, a receiver has to unsubscribe the highest layer if the packet loss rate exceeds 5%. Since a receiver needs to be idle for about 8~12 sec between two possible consecutive joins in order to avoid overloading of probing traffic, BIC has a startup mechanism so that a new receiver can subscribe more layers faster. Since this startup is aggressive, the startup will quit for any packet loss.

Since every step of the layered video multicast protocol described above is quite sensitive to the packet loss, experiments show that it will not work well for networks suffering from wireless packet loss.

We integrate the proposed PLC algorithm described in Section 2 with this BIC layered multicast protocol. Whenever a receiver observes a packet loss, PLC module checks whether the category belongs to wireless loss. If so, this loss information will not be reported to BIC control module.

5. SIMULATIONS

We use *ns2* to evaluate the performance of the packet loss classification algorithm on top of BIC layered multicast. Network topology used is shown in Fig. 2. Some topology settings, such as link capacity and link delay, are indicated in the figure. A BIC sender is at node 0 with receivers located at node 1 to node 3 whose wireless links are modeled as Gilbert/Elliot Markov chain with P_{bb}=0.2 and P_{gb}=0.00005. Background traffic will travel from node 8 to node 4 to compete for the bandwidth of the bottleneck between node 5 and node 6.

Figure 3 and Fig. 4 show the BIC layered multicast protocol performance with and without wireless fading errors, respectively. Clearly BIC protocol fails with the presence of wireless errors.

For the receiving throughput with the presence of wireless errors, Fig. 5, Fig. 6, and Fig. 7 show the improved performance with packet-loss differentiation mechanisms, such as Spike-train, ZigZag, and our proposed PLC method, respectively. It shows that the throughput is much improved with the proposed PLC algorithm. Under various background traffics, such as sessions of BIC, UDP, and FTP-TCP, the simulation results are consistent and we can conclude that the proposed PLC algorithm works well for distinguishing packet loss.

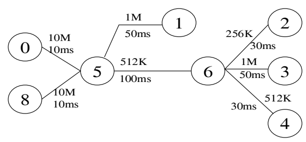

Fig. 2. Network topology for simulations.

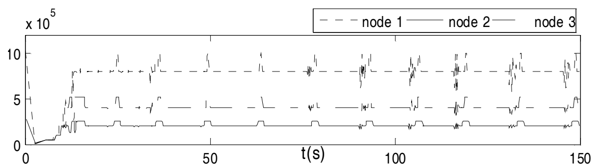

Fig. 3. The separately attained throughput (bps) at nodes 1,2,3, in a BIC session without wireless packet loss.

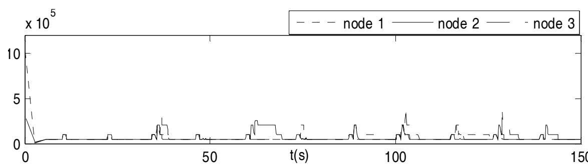

Fig. 4. BIC with wireless packet loss.

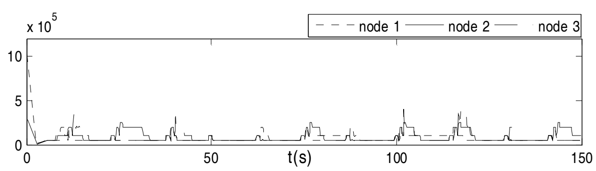

Fig. 5. BIC/ZigZag with wireless packet loss.

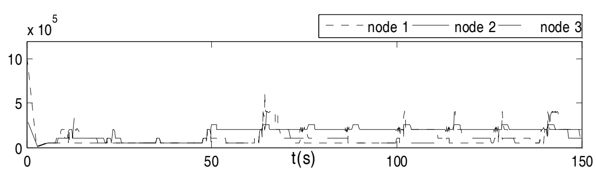

Fig. 6. BIC/Spike-train with wireless packet loss.

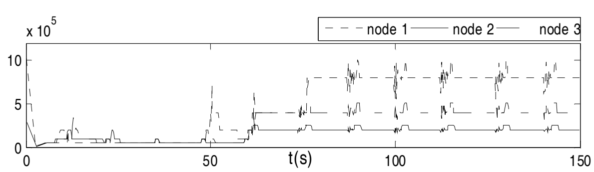

Fig. 7. BIC/Proposed PLC with wireless packet loss.

6. CONCLUSION

We present a new packet loss classification algorithm which utilizes trend detection to assist packet loss differentiation. Simulations show that the classification algorithm works quite well. With the knowledge of the classes of lost packets, congestion control algorithms can work much more effectively. Another benefit of this algorithm is that it can notify the sender that enhanced channel coding for error correction is required in the case that packet losses are mostly due to wireless errors.

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