ADAPTIVE END-TO-END LOSS DIFFERENTIATION SCHEME FOR TCP OVER WIRED/WIRELESS NETWORKS

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Abstract: We improve a well-known TCP throughput enhancement scheme over heterogeneous networks of wired/wiress paths. The scheme (Westwood+) adjusts the congestion window using the estimate of currently available bandwidth instead of halving the window in the event of a packet loss, enhancing the TCP throughput. Our scheme infers the cause of packet loss(if it is due to congestion or regular bit errors over wireless paths) from the changes of the RTT values. We adapt the RTT threshold used for determining the cause of packet loss to the current congestion level. For example, we increase the RTT threshold for a packet loss to be regarded as due to congestion when the congestion level of the path is estimated to be low. This avoids unnecessary halving of the congestion window on packet loss due to regular bit errors over wireless path in a more precise way than previous schemes. Compared with the Westwood+, our scheme improves the TCP throughputs by 29% and 55% on 1 Mbps and 10 Mbps paths, respectively. The fairness degradation is negligible.

1 INTRODUCTION

TCP congestion control runs under the basic assumption that any packet loss is the indication of congestion. However, the assumption does not hold when the TCP flow path includes wireless part. In such a case the packet loss may not come from congestion but from regular bit errors over wireless path. TCP throughput may be unnecessarily degraded due to the packet loss from bit errors over wireless part even though there is little congestion.

Known schemes to address this problem can be divided into three categories(Sumitha, 2005): First, network-based schemes locate an agent at the access point/base station on the TCP path to locally recover the wireless loss at transport or link layer. M-TCP(MTCP, 1997) and I-TCP(ITCP, 1995) approaches split the TCP flow into the two parts at transport layer and deal with the wireless part in different way to improve the throughput. Network-based schemes at transport layer do not maintain the endto-end semantics of TCP and may require state to be maintained and packets to be buffered at the base station. Snoop(Snoop, 1995) scheme places an observer in the base station at link layer. However, it can be purely local and aware of the semantics of the TCP protocol.

In next, in the explicit loss notification approach the receivers/network routers mark the acknowledgements with appropriate notification of distinguishing the channel errors from congestion losses. Then the senders respond to the notification. In TCP-Casablanca(Biaz, 2005), the sender sends packets with different discard priorities. Routers drop marked packets to de-randomize congestion losses. The receiver discriminates the cause of packet loss and marks the acknowledgement with explicit loss notification. Then the sender responds to the notification. Like TCP-Casablanca, the explicit loss notification approaches require modifications to network infrastructure, the receiver, and the senders.

Finally, end-to-end schemes modify the TCP congestion control algorithm to distinguish the losses due to congestion in the network from other random losses over wireless paths. They can be deployed more easily because they are a modification to the TCP congestion control at sender or receiver side. Westwood+(West, 2004) is the best end-to-end scheme as far as we know in terms of TCP throughput

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Figure 1: The heterogeneous network model of wired/wireless paths.

enhancement. It estimates currently available bandwidth and adjusts accordingly the congestion window size instead of halving it in the event of a packet loss. When three duplicate ACKs (TDACKs) are received, both the congestion window (cwnd) and the slow start threshold (ssthresh) are set equal to the estimated bandwidth times the minimum measured RTT. On timeout, the ssthresh is set as before while the cwnd is set equal to one.

We propose a new end-to-end scheme which improves the Westwood+. Instead of estimating the available bandwidth, our scheme precisely infers the cause of individual packet loss whether it is due to congestion or regular bit errors of wireless nature. The congestion window is reduced only when the packet loss is determined to be due to congestion. A moving threshold for relative change of RTT against the minimum RTT is employed to differentiate the cause of each packet loss. The moving threshold is lowered to be more sensitive to congestion loss when the network is is congested while it is increased when the network is unloaded.

Compared with the Westwood+, our scheme improves the TCP throughputs by 29% and 55% on 1 Mbps and 10 Mbps paths, respectively based on ns-2 simulations(NS-2). The fairness degradation is negligible.

2 NETWORK MODEL

Figure 1 shows a heterogeneous network model of wired/wireless paths to be used in the following description of our scheme. p_c denotes the probability of packet loss due to congestion in the wired path while p_w denotes a uniformly random packet loss probability over the wireless path. Without loss of generality, we assume there is no congestion over the wireless path. C denotes the link bandwidth in Mbps and B denotes the number of packet(size S) currently occupying the buffer. B_{max} denotes the size of the buffer. d denotes the propagation delay between the base station(R2) and the receiver. Let T_p denote the propagation delay on the path between the sender and receiver. The queuing delay is represented by $\frac{B \cdot S}{C}$.

Then, the RTT(T) can be written as Equation 1.

$$T = T_p + \frac{B \cdot S}{C} \tag{1}$$

3 PROPOSED SCHEME

3.1 The loss differentiator

We present a new scheme to precisely infer the cause of each packet loss encountered whether it is due to congestion or not. The RTT(T) measured immediately before the current packet loss is used through Equation 2 as an indicator. \overline{T} and T_{dev} are an exponentialy weighted moving average of RTT(T) and the deviation. They are updated by $\overline{T} = \frac{7}{8}\overline{T} + \frac{1}{8}T$ and $T_{dev} = \frac{3}{4}T_{dev} + \frac{1}{4}|T - \overline{T}|$ as in most TCP implementations. The current packet loss is determined to be a congestion loss if Equation 2 is satisfied.

$$T > \overline{T} + T_{dev} \cdot \left(2 \left(\frac{T_p}{T} \right)^k - 1 \right)$$
 (2)

The rationale behind the indicator represented by Equation 2 can be better explained using Figure 2. It shows the relationship among the buffer occupancy, RTT and p_c against $\sum_{i=1}^{n} W_i$, the aggregate sum of the congestion windows of the TCP flows established. W_i denote the congestion window size of *i*-th TCP flow out of *n* flows.



Figure 2: Three-state of network buffer and the change of the RTT with the aggregation of *cwnd* of each flow.



Figure 3: Accuracies of loss differentiator versus k for various network conditions.

When the buffer occupancy nears overflow, we have $T \gg T_p$ and the probability of congestion $loss(p_c)$ becomes increasingly high. In order not to miss any congestion loss, we decrease the threshold. On the contrary, when T approaches T_p , we increase the threshold not to mistakenly count the wireless loss as congestion loss. We estimated the accuracy of the indicator against the value of k through a simulation involving packet losses of known causes. A_c is the ratio of the number of congestion losses correctly inferred over the total number of congestion losses. A_w is similarly defined for wireless losses. As k increases, A_c increases as shown in Figure 3(a), while A_w is decreased as shown in Figure 3(b). Thus k can be chosen through trade-off analysis for best performance in terms of throughput and fairness.

The thresholds in Zigzag scheme(Song, 2003) are special cases of our indicator while they employ the one-way delay approximately equal to half of the RTT. The capability of our loss differentiation for wireless losses is better by 100% than Zigzag scheme while for congestion losses both schemes have the same accuracy when only one flow is established.

3.2 TCP modification

To apply the loss differentiator, we modify two blocks in TCP congestion control: they are RTT update and receipt of TDACKs. In RTT update, the TCP sender updates the minimum of RTT which indicates the propagation delay. On receipt of TDACKs, the sender evaluates Equation 2 using the sample RTT measured just before the TDACKs only if the TCP connection is in the congestion avoidance phase. The sender halves *cwnd* if Equation 2 is satisfied. Otherwise, the sender keeps current *cwnd*. We explain the procedure of the receipt of TDACKs in Figure 4.



Figure 4: Flow chart for the modified congestion control block

3.3 Throughput model

We derive the TCP throughput of the proposed scheme following arguments developed by Kelly(Kelly, 1999). The throughput model of Westwood+(Luigi, 2002) is also derived similar to (Kelly, 1999). The TCP throughput model can be defined as Equation 3. For the simplicity, we do not model the behavior after a timeout.

$$x_{TCP} = \frac{1}{T} \sqrt{\frac{a(1-p)}{(1-b)p}}$$
(3)

a and b denote the increase and the decrease parameter in TCP, respectively. p is the total packet loss rate. Let x_{prd} and b_{prd} denote the throughput and the decrease parameter of the proposed scheme, respectively. x_{ww+} and b_{ww+} denote those of the Westwood+, respectively. For both scheme, a is set to 1.

The decrease parameters (b_{prd} and b_{ww+}) derived as a function can characterize the congestion control of the proposed scheme and Westwood+, respectively. Figure 5(a) shows the change of *cwnd* in NewReno over wireless paths. In this case, all of packet loss are regarded as due to congestion. Figure 5(b) shows that the proposed scheme infers the cause of packet loss and keeps the current *cwnd* if the loss is regarded as the wireless loss. Figure 5(c) shows that changing *b* can be equivalent to the proposed scheme. We first derive x_{prd} similar to (Kelly, 1999) and (Luigi, 2002), and then compare the proposed scheme with the Westwood+ using b_{prd} and b_{ww+} .

The *cwnd* is updated upon ACK reception. Each time an ACK is received back by the sender the *cwnd* is increased by 1/cwnd. On the receipt of TDACKs, the proposed scheme involves to infer the cause of the packet loss. The *cwnd* is reduced by half if the proposed scheme regards the packet loss as due to congestion. Otherwise, the *cwnd* will be kept. Let $\Pr(c|l)$ denote the probability of packet losses regarded as congestion losses. It results from the aggregate sum of the probability to correctly detect the congestion losses represented by $\frac{p_e}{p}A_c$ and the probability to wrongly infer the wireless loss into the congestion loss represented by $\frac{p_w}{p}(1 - A_w)$ as shown in Equation 4. Therefore, the change in *cwnd* is $\frac{1}{2} \cdot \Pr(c|l) \cdot cwnd$.

$$\Pr(c|l) = \frac{1}{p} \left[p_c \cdot A_c + p_w \cdot (1 - A_w) \right]$$
(4)

Since the total packet loss rate is p, it follows that the expected change of cwnd per update step is:

$$E[\Delta cwnd] = \frac{1-p}{cwnd} - \frac{1}{2} \cdot \Pr(c|l) \cdot cwnd \cdot p \quad (5)$$

Since the time between update steps is about $\frac{T}{cwnd}$, by recalling Equation 5, the expected change in the rate x per unit time is approximately:

$$\frac{\delta x(t)}{\delta t} = \frac{1-p}{T^2} - \frac{1}{2} \cdot \Pr(c|l) \cdot p \cdot x^2(t) \tag{6}$$

Equation 6 is a separable variable differential equation. By separating variables, Equation 6 can be written as:



Figure 5: The change of cwnd in (a) NewReno, (b) proposed scheme, and (c) equivalent model of the proposed scheme changing b

$$\frac{\delta x(t)}{x^2(t) - \frac{2(1-p)}{T^2 \cdot \Pr(c|l) \cdot p}} = -\frac{1}{2} \cdot \Pr(c|l) \cdot p \cdot \delta t \quad (7)$$

and by integrating each member the following solution can be obtained

$$x(t) = \frac{x_1 \cdot \left(1 + C_0 \cdot e^{-\frac{1}{2} \cdot \Pr(c|l) \cdot p \cdot t}\right)}{1 - C_0 \cdot e^{-\frac{1}{2} \cdot \Pr(c|l) \cdot p \cdot t}}$$

where $x_1 = \frac{1}{T} \sqrt{\frac{2(1-p)}{\Pr(c|l) \cdot p}}$ is the root of the equation $x^2(t) - \frac{2(1-p)}{T^2 \cdot \Pr(c|l) \cdot p} = 0$ and a constant C_0 depends on the initial conditions. The steady state throughput of the proposed scheme is then,

$$\therefore x_{prd} = \lim_{t \to \infty} x(t) = \frac{1}{T} \sqrt{\frac{2(1-p)}{\Pr(c|l) \cdot p}} \qquad (8)$$

and by recalling Equation 3 and 4, b_{prd} can be written as Equation 9.

$$x_{prd} = \frac{1}{T} \sqrt{\frac{a(1-p)}{(1-b_{prd})p}} = \frac{1}{T} \sqrt{\frac{1-p}{\frac{1}{2} \Pr(c|l) \cdot p}}$$

$$\therefore b_{prd} = 1 - \frac{1}{2} \cdot \Pr(c|l)$$

$$= 1 - \frac{1}{2p} \left[p_c \cdot A_c + p_w \cdot (1-A_w) \right] \quad (9)$$

From Equation 9, the proposed scheme can adapt b_{prd} into p_c since it increases A_w as p_c decreases(See Figure 3). For example, b_{prd} can be set to 1 when $p_c \approx 0$. However, b_{prd} can approach $\frac{1}{2}$ as p_c increases.

To compare with Westwood, the throughput approximation is shown as following (Luigi, 2002):

$$x_{ww+} = \frac{1}{\sqrt{T(T-T_p)}} \sqrt{\frac{1-p}{p}}$$
(10)

By recalling Equation 3, b_{ww+} can be derived as Equation 11.

$$x_{ww+} = \frac{1}{T}\sqrt{\frac{a(1-p)}{(1-b_{ww+})p}} = \frac{1}{T}\sqrt{\frac{1-p}{\left(1-\frac{T_p}{T}\right)\cdot p}}$$

$$\therefore b_{ww+} = \frac{T_p}{T}$$
(11)

From Equation 11, we can find that the Westwood+ tries to empty the buffer to suppress the increase of RTT since the b_{ww+} is sensitive to the buffer occupancy. Consequently, compared with Westwood+, the proposed scheme can be expected to improve the TCP throughput.

4 PERFORMANCE EVALUATION

4.1 Experimental setup

We evaluate the performance of proposed scheme via ns-2 (Ver 2.26)(NS-2) simulation. The network topology is shown in Figure 1. The bottleneck (C in the Figure 1) is set to 1 Mbps or 10Mbps. The size of the buffer (B_{max}) is set to the bandwidth-delay product which is 12 packets or 120 packets, respectively. We set the packet size equal to 1500 bytes. The value of kis set to 2. We run simulations 50 times. In the each of the 50 runs, we estimate the average throughput, link utilization and fairness index. We use the Jain's fairness index(Jain, 1984) as in Equation 12 where x_i and n denote the throughput of *i*-th flow and the number of flows, respectively. The Westwood+ module for ns-2 is obtained from an official site(nsWestwood).





Figure 6: Link utilization and fairness index on 1 Mbps of bottleneck for $p_w = 1\%$ and 5%

$$F(x) = \frac{\left(\sum_{i}^{n} x_{i}\right)^{2}}{n\sum_{i}^{n} x_{i}^{2}}$$
(12)

4.2 Results

Figure 7 shows the throughput with increasing wireless packet loss rates. Compared with Westwood+, the proposed scheme improves the throughput by up to 29% and 55% when the bottleneck bandwidth is 1 Mbps and 10Mbps, respectively. Figure 6(a) shows the link utilization with varying the number of flows for 1% and 5% of wireless packet loss rates on 1 Mbps link. The increase of wireless packet loss rate may prevent the flows from fully utilizing the bottleneck link capacity. Compared with Westwood+, the proposed scheme improves the utilization by 0.1%-59% depending on the number of flows. Figure 6(b) shows that the fairness index of the proposed scheme stays



Figure 7: Throughput comparisons with Westood+ and NewReno



Figure 8: Throughput comparisons with Westood+ and NewReno versus change of delay and buffer length

over 0.995 which is similar to those of Westwood+ and NewReno.

Figure 8(a) shows the TCP throughput decreases as wireless propagation delay(d) increases (C=1Mbps and p_w =1%). The NewReno suffers most as high as 47.5% when the delay increases from 10ms to 100ms. While the throughput of the Westwood+ and the proposed scheme decrease 17.6% and 19%, respectively, the proposed scheme improves the throughput by 11.5% - 14.5% compared with the Westwood+. Figure 8(b) shows the TCP throughput as B_{max} increases. When $B_{max} \ge 6$ packets, the proposed scheme improves the throughput as shown above.

4.3 Implementation and results

We implement the algorithm based on Linux Kernel Ver. 2.4.20 as shown in Figure 9(a). The sender runs

a FTP client and the receiver runs a FTP server. The data flow is sent through the Nist Net(NIST). We set 50 packets to the maximum buffer size and 70ms to forward path delay, and we vary the wireless packet loss rate from 0% to 9%. We record the goodput defined as Equation 13 for 3MB and 100MB of file size. We uploads a file of size 3MB 10 times. As shown in Figure 9(b) and 9(c), the proposed scheme have the best goodput with increasing the wireless packet loss rate.

$$Goodput(\%) = \frac{file_size}{duration \times C} \times 100(\%) \quad (13)$$





Figure 9: (a) Emulation setup using NistNet and goodput for (a) 3MB and (b) 100MB file

5 CONCLUSION

We propose a simple TCP modification to improve the Westwood+ through precise inferring of the cause of packet losses. The proposed scheme is an end-to-end scheme which is easy to implement. Modifications is made only to the TCP sender. From the simulation results, an adaptative loss differentiator can improve the throughput compared with Westwood+, and the degradation of fairness is negligible. Westwood+ can estimate the available bandwidth less than the real one due to frequent wireless losses. Thus, it may fail to retain the current cwnd. However, the proposed scheme can retain the current cwnd through more precise inferring of the cause of packet losses. Our scheme is more effective in improving the throughput especially when the bottleneck bandwidth is high(Compare Figure 7(a) and 7(b)).

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