A Novel Receiver-aided Scheme for Improving TCP Performance in Multihop Wireless Networks

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Abstract

In general, TCP sender will take all the packet losses as congestion caused and reduce the packet sending rate to avoid the worsening of network condition. In multihop wireless networks, the long propagation time, poor link quality and high bit error rate make this scheme slow-moving, overcompensated and inefficient.

This paper proposes a new optimized data delivery scheme, called TCP receiver-aided (TCP-RA) scheme. By continuously observing the packet's end-to-end delay condition, the receiver can estimate the congestion degree and sense the changes of situation. When packet losses happen, the receiver will estimate the reason of loss and assist the sender to control sending rate more efficiently in congestion control. Compared with several contemporary TCP schemes, experimental studies reveal that the TCP-RA scheme can accurately differentiate various packet losses and efficiently improve the throughput performance by control the sending rate smartly.

1. Introduction

With the explosive deployment of the IEEE 802.11 wireless endpoints, wireless data communication over multihop wireless networks is becoming wildly used. Meanwhile the TCP over the IEEE 802.11 based multihop wireless networks has been challenged over these years [1-3] because of its poor throughput performance. The well-known challenge in providing TCP congestion control is that current TCP implementations rely on packet loss as an indicator of network congestion. Multihop wireless networks bring some critical challenges to TCP because of the distributed topology environment, poor quality of wireless link, and high level of bit error rate. In IEEE 802.11 based multihop wireless networks, a high level of bit error rate could not be ignored.

In a high bit error rate network environment, when random transmission losses caused by sporadic wireless channel failure, how does the sender tell it from congestion losses and not mistreat it as a symptom of congestion by current TCP schemes and how does the sender avoid leading to an unnecessary window reduction?

In the previous works, the TCP sender takes the responsibility to estimate the network condition and make the congestion control [4, 6, 7]. Their schemes depend on the measurement of packet's Round Trip Time (RTT). It is obviously that the ACK packet's behavior will misadvise the real condition of the link and affect the performance of TCP schemes.

There are many research works focus on the receiver side for improving the TCP performance. [8][10] explores the end-to-end loss differentiation algorithms. and Cen et al. proposes the ZBS and evaluate the algorithm for the network where wireless link is the last hop or wireless link acts as backbone network topology.

[5] tracks loss events and determines the data delivery rate at the receiver. [12] add functionality to the receiver to control the bandwidth shares of incoming TCP flows, i.e., by adapting the receiver's advertised window and delay in transmitting ACK messages, the receiver is able to control the bandwidth share on the access link according to the client's needs.

The techniques above do not achieve high accuracy in classifying both congestion and wireless channel losses. Simply consider the packet inter-arrival time or delay time to make a classification about losses is not enough at all, because it is difficult to make a distinct boundary about packet inter-arrival time or delay time. The methods also do not consider the complex environment of multihop wireless network which can impact the TCP performance seriously.

We have recognized that the receiver has a better position to measure the bandwidth of network and performs well in determining the cause of packet loss. Without the interfering of ACK packet, the receiver can

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measure the packet's end-to-end propagation time more accurately. Further considering the history of end-to-end delay of packet, the receiver can provide more accuracy in knowing the packet loss, which is extremely useful for sender to take smart control when packet loss happens. At the same time, we introduce the "dubious loss" which helps the sender take a moderate sending rate control in some complex conditions.

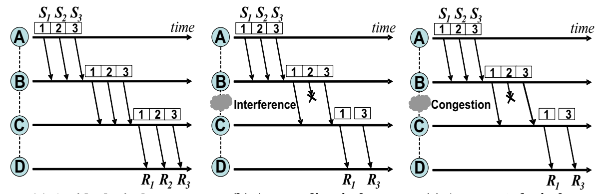
The results of simulation illustrate that TCP-RA works well in wireless multihop networks, and get a high differentiation accuracy of packet losses. The scheme significantly improves the network performance.

The arrangement of this paper is organized as follows. Section 2 introduces the proposed TCP receiver-aided scheme. Section 3 gives the simulation setup, scenario, and results. Finally, the concluding remarks are drawn in Section 4.

2. The TCP Receiver-aided Scheme

2.1. Monitoring Congestion State at the Receiver

In order to show how the receiver can monitor the contention state of a connection, we depicted the situations in Figure 1. All the links between node A, B, C and D are wireless. With the simplest chain topology, three back-to-back data packets are emitted to the channel from node A to node D. Under an optimal condition, three packets traverse the space without any hesitation as depicted in Fig. 1(a). The corresponding data packet arrival pattern is depicted in the simplified handshaking. In Fig. 1(b), considering a sporadic wireless channel error has occurred, the packet number 2 is corrupted because of error in some bits that causes a false signal decoding at the receiver. Since the channel is still in a good state, the packet number 3 still can pass through with a low congestion state. The time difference between the arrival of data packet number 1 and number 3, R_1 and R_2, respectively, is slightly higher compared with the situation shown in Fig. 1(a). In Fig. 1(c), considering a packet loss is caused by an elevated level of channel congestion, the packet number 2 is dropped because the channel contention failure (after seven retries of request-to-send control packet at the MAC). Because of the delay in contention, the packet number 3 is further delayed by another time gap. Consequently, the time difference between R_1 and R_2 becomes larger. According to the above observations, we develop the following heuristics scheme:



(a) An ideal wireless channel case (b) A sporadic wireless channel case (c) A congested wireless channel case

Figure 1. Data packet arrival patterns

Suppose S_i is the sending time while R_i is the receiving time of a packet with sequence number i. EED_i is the end-to-end delay of the packet, or a result of R_i–S_i. N is the number of hops of the connection. Then, under optimal condition without contention, EED_i can be described as following [9].

$$EED_i = \sum_{h=1}^{N} (T_q + T_{DIFS} + CW_h \times T_{slot} + T_{RTS} + T_{CTS} + T_{DATA} + T_{ack} + 3T_{SIFS}) \quad (1)$$

Where T_q refers to the queuing delay at each node; T_{RTS} refers to the transmission delay of a request-to-send (RTS) packet; T_{CTS} refers to the transmission delay of a clear-to-send (CTS) packet. T_{DATA} refers to the transmission delay of the data packets. CW_h is the contention window, which is dictated by IEEE 802.11 standard and h is the number of hops. If we take into account the contention at each node, then EED_i can be described as follows:

$$EED_i = \sum_{h=1}^{N} \left[(T_q + T_{DATA} + T_{ack} + 2T_{SIFS} + T_{CTS}) + K_h^i (T_{DIFS} + CW_h \times T_{slot} + T_{RTS} + T_{SIFS}) \right] \quad (2)$$

The above equation shows that a data packet experiences an contending time before accessing the channel. The level of contention is described by K, which is round number of contention that is experienced by the packet i at the h hops. At the receiver, we can derive a series of EED_i for each coming data packet.

For each arrival of the successive data packet, we define and obtain the following equation:

$$\begin{aligned} EED_{i+1} - EED_i &= \sum_{h=1}^{N} K_h^{i+1} (T_{DIFS} + CW_h \times T_{slot} + T_{RTS} + T_{SIFS}) \\ &- \sum_{h=1}^{N} K_h^{i} (T_{DIFS} + CW_h \times T_{slot} + T_{RTS} + T_{SIFS}) \\ &= (T_{DIFS} + CW_h \times T_{slot} + T_{RTS} + T_{SIFS}) \sum_{h=1}^{N} (K_h^{i+1} - K_h^{i}) \end{aligned} \quad (3)$$

This calculation tells us how many contentions that the packet i+1 experiences more than the packet i. If a packet experiences many rounds of contention at each node, we can conclude that the connection path is in a congested state. For the sake of brevity, we define the *Contention Indicator (CI)* and compute it by exponential average:

$$CI_{new} = \frac{7}{8} CI_{old} + \frac{1}{8}(EED_{i+1} - EED_i) \quad (4)$$

At the same time, we evaluate the network congestion grade. With the changing of network congestion situation, the static criterion is not suitable for describing the congestion grade. A dynamic and adaptive mechanism is effective in such constrained environments. After analyzing wireless multihop network congestion situation, we defined the network's congestion grades with slight, middle, and heavy congestion by following rules.

Slight congestion:

$EED_i \le 0.8 * EED_{mean}$

Middle congestion:

$0.8 * EED_{mean} \le EED_i \le \min\{1.2 * EED_{mean}, 0.8 * EED_{lc}\}$

Heavy congestion:

$EED_i \ge \min\{1.2 * EED_{mean}, 0.8 * EED_{lc}\}$

Where EED_{mean} is the mean of EED and calculated by the equation $EED_{mean} = 0.95 * EED_{mean} + 0.5 * EED_i$. The factor is selected by simulation. EED_{lc} is the EED of the packet which is successfully transmitted before the last congestion (LC) loss happened. In most cases, the neighbor congestion losses are similar and directed.

In the simulation, we find that 1% congestion losses and 44% wireless losses happened when the network was in slight congestion grade. In middle grade, 23% congestion losses and 47% wireless losses happened. In heavy grade, 76% congestion losses and 9% wireless losses happened. We draw the ratio of packet loss distributions in table 1.

Table 1. Ratio of loss distribution

	wireless loss	congestion loss
slight	47%	1%
middle	44%	23%
heavy	9%	76%

Table 2. Network congestion grade and trend

Congestion grade \ Trend	UP	Stabilized	Down
Slight	Wireless loss	Wireless loss	Wireless loss
Middle	Congestion loss	Dubious loss	Wireless loss
Heavy	Congestion loss	Congestion loss	Dubious loss

In middle and heavy grade, it is difficult to classify the type of losses. With the help of *CI*, we can get a precise decision of packet loss classification. Since some losses are very difficult to be differentiated, it is necessary to make a compromised response to these losses, and we name it as "dubious loss".

When the network runs stably, the end-to-end delay of packets is similar and their difference fluctuates in a reasonable range. In such condition, the value of *CI* is limited and the absolution of *CI* has an upper limit, α. When the network becomes congested gradually, the EED_{i+1} is larger than EED_i and the *CI* is positive and greater than α. When the network recovers gradually, the EED_{i+1} is smaller than EED_i and the *CI* is negative.

$|CI| \le \alpha$, the tendency of network is stability;

$CI > +\alpha$, the tendency of network is up;

$CI < -\alpha$, the tendency of network is down;

Simulation shows that $\alpha = 0.05 * EED_{mean}$ provides the best results. Based on our simulation, we build the empirical table 2 with the support of the congestion grade and the tendency of network as follows. The receiver can estimate the packet loss type and congestion state of network while feeding back to the TCP sender.

When packet loss occurs, we consult the Table 2 to find whether the packet loss is from the sporadic wireless channel error or the elevated level of channel contention and assist the sender to make the proper action for getting better performance. We have defined three kinds of losses and adopted different reactions to them, wireless channel loss, dubious loss and congestion loss. Wireless channel error causes the packet loss but it is occasional and will not impact the following data transport. So the best reaction for wireless loss is to neglect them and do not need congestion control. When packet loss is caused by the over-congested network environment, it is named congestion loss and the sender should slow down the sending rate to ensure the health of network. In some conditions, it is hard to make a precise decision about loss type and the network is on the way into congestion. So we will take a great risk to simply classify the loss as congestion loss or wireless loss. We have defined the dubious loss and its corresponding actions. The TCP sender slows down the sending rate and modifies the CWND by the information of loss type to avoid the continual depravation of network.

2.2. The Proposed TCP Receiver-aided Scheme

In this section we present our proposed scheme, called TCP receiver-aided (TCP-RA) scheme. After a connection is established, the sender enters a slow start stage as normal. Once packet loss occurs, the receiver will consult the contention indicator in the equation (4) and the congestion grade described in part A in order to judge whether the packet loss is caused by the wireless channel error or due to the traffic congestion. If the wireless channel error is identified, no further action needs to be performed and the sender does not need to modify the CWND size. If the congestion or dubious losses are recognized, the receiver will send out ACK with an option notifying the sender about the loss type. Upon receiving an ACK, the sender first checks whether it is a congestion state. If it is a good ACK without congestion information, the sender inflates the CWND according to the amount of acknowledged data packet and continues its data transmission. If the congestion occurs, it will deflate the CWND and slow down the data sending rate as follow. For the three types of losses, CWND will be modified with,

Dubious loss:

$$CWND = \beta * (CWND + Packet\_received)$$

Congestion loss:

$$CWND = 0.5 * (CWND + Packet\_received)$$

Wireless loss:

$$CWND = CWND + Packet\_received$$

For dubious loss, the sender is in a dilemma. Although the network congestion is still tolerable, a compromised modification is necessary to prevent the network from the further congestion for safe. In our simulation, $\beta = 0.8$ provides the best throughput performance.

3. Numerical Simulation and Discussion

In this section, the proposed TCP-RA is evaluated in its differentiation accuracy and the performance of goodput compared to TCP Veno [4], TCP NewReno [6], and TCPW [7]. We use the ns2 simulator in our evaluation. In the simulation, the network warms up 30s firstly and then records the data. With every combination of the parameters, we execute the algorithms for 30 times and each value in the graphs is the average over all 30 results.

3.1. Topology and Wireless Error model

In order to examine the performance of TCP-RA, we simulate the algorithm in a grid topology in Fig. 2. The wireless link between neighbor nodes is bidirectional and the neighbor nodes can exchange messages with each other directly. We run 8, 12, 16 TCP flows respectively in the simulation. In each case, half of the TCP flows go horizontally and the other half go vertically and they are arranged evenly as much as possible. Some nodes in the crossway which need to tackle horizontal flows and vertical flows simultaneously are bottlenecks in some conditions.

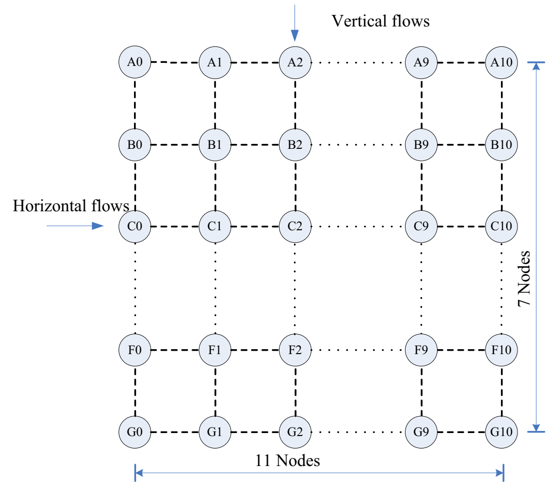


Figure 2. 11*7 grid topology

Table 3. Simulation parameters

Parameter	Value	Description
RxThresh	3.652×10^{-10}W	Transmission range is 250 m
CSThresh	1.559×10^{-11}W	Carrier sensing range is 550 m
Data packet size	800 bytes	
Physical data rate	11 Mbps	IEEE 802.11b
Alpha and Beta	1 and 3	For TCP Veno setting
Routing protocol	AODV	Ad hoc on-demand vector

In topology setting, the well-known hidden and exposed node problems are existent. Other simulation parameters are shown in Table 3.

For modeling the error process of Rayleigh fading channel behavior, a first-order Markov chain such as the two-state Markov model provides an adequate conformability. In the simulation, we use the two-state discrete Markov chain, Gilbert's Model [11], for the wireless link of the end-to-end connection path.

3.2. Differentiation Accuracy of Packet Losses

At the same time, we evaluate the differentiation accuracy of congestion loss and wireless channel error loss of our algorithm. To measure the accuracy of our scheme, two metrics are used: the congestion accuracy A_c and the wireless accuracy A_w. A_c is the ratio of the number of congestion losses correctly diagnosed over

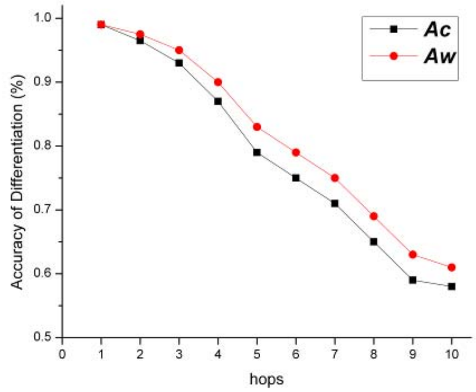


Figure 3. The accuracy of differentiation

Table 4. Accuracy of loss classification

	Estimation	
	Congestion loss	Wireless loss
Congestion loss	80%(correct)	6%(error)
Dubious loss	15%	18%
Wireless loss	5%(error)	76%(correct)

the total number of congestion losses. A_w is similarly defined for wireless losses. For some losses are classified as dubious losses, we take them as error classification in the simulation results. We just take the exactly judgment as right in the statistical figure.

To achieve high congestion accuracy is very important. If congestion losses are wrongly classified as wireless losses, TCP senders will not decrease their sending rate when congestion happens, and the network performance will be significantly degraded. On the other hand, a wireless loss mistaken for congestion loss will only degrade the performance of this TCP connection by over-reactively denting the sending rate, and it will be tolerable. With the help of dubious loss, we have a tradeoff between network congestion and high performance.

In the simulation, the average accuracy of congestion loss and wireless loss are 80% and 76% respectively. The increasing of hop count and TCP flows will contribute to the long packets trip time, and introduce the fluctuation of available bandwidth. It upgrades the difficulty of differentiation for the receiver. The complex network condition degrades the differentiation accuracy of the receiver.

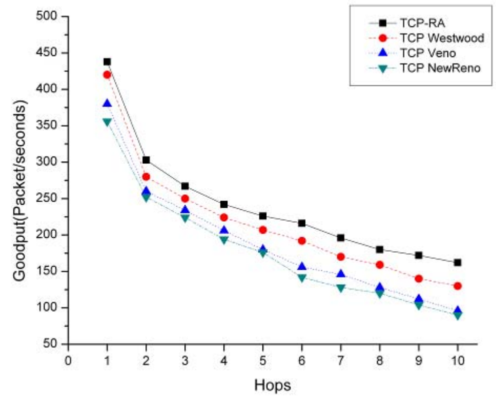


Figure 4. Goodput performance with different TCP variations

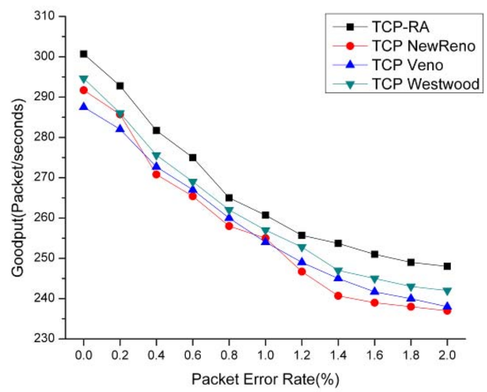


Figure 5. Goodput performance with different packet error rate

The simulation results show that our algorithm works well and the receiver can make an accurate judgment about the causes of packet losses.

3.3. Goodput Performance

Goodput is defined as the number of successful delivered packets from the sender to the receiver. Comparing TCP Veno, TCPW, and TCP NewReno with our scheme, we evaluate our proposed TCP-RA by varying the number of hops from 1 to 10 in order to investigate how they respond to various scenarios. Fig. 3 reveals that TCP-RA achieves better goodput performance regardless of the hop count than the other TCP variations do. An average of 40% goodput gain of our scheme over the TCP NewReno is revealed in Fig. 4. When the hop count is small, all protocols are similarly effective. As the hop count increase, the feedback information used to estimate the available bandwidth arrives too late to be of significant help to the other TCP variations. But the TCP-RA can get a better performance in this situation with the help of receiver.

Fig. 5 shows that our algorithm is effective for handing the high packet error rate environment than the others. In the simulation, the hop number is set to 3. The simulation proves that the classifying of loss type and taking corresponding reactions are essential and beneficial.

4. Conclusion and Future Work

In this paper, we have presented a novel TCP-RA scheme to enhance the TCP algorithm in the multihop wireless environment. We identified the culprits behind the conventional TCP performance degradation over wireless links. To mitigate these problems, we used a contention indicator, which assists in measuring the congestion state of a connection. We found out that the receiver always has better position in identification whether the packet losses are due to the sporadic wireless channel error or the elevated level of congestion state. With the aid of receiver, the TCP sender can take the efficient congestion control in time. In this way, our proposed mechanism behaves excellently in the multihop wireless network.

Our simulation results are encouraging and we are working to improve our scheme in various layers of dimension, which include the TCP-RA scheme incorporates with other versions of TCP, the fairness and friendliness of our scheme. The mobility and robustness issues are the next steps of our proposed TCP scheme.

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References

[1] Y Su, P Steenkiste, and T Gross, "Performance of TCP in Multi-Hop Access Networks," IWQoS2008, pp.181–190, 2008.

[2] K. Nahm, A. Helmy, and C. Kuo, "TCP over Multihop 802.11 Networks: Issues and Performance Enhancement," in Proc. MOBIHOC, pp.277–287, 2005

[3] D. Koutsonikolas, J. Dyaberi, P. Garimella, S. Fahmy, and Y. C. Hu, "On TCP Throughput and Window Size in a Multihop Wireless Network Testbed," in Proc. ACM WiNTECH, pp.51–58, 2007.

[4] C.P. Fu and S.C. Liew, "TCP Veno: TCP enhancement for transmission over wireless access networks," IEEE J. Sel. Areas Commun., vol.21, pp.216–228, 2003.

[5] V. Tsaoussidis and C. Zhang, "TCP-Real: Receiver-oriented congestion control, " The Journal of Computer Networks 40 (4) (2002) pp. 477–497

[6] R. Dunaytsev, Y. Koucheryavy, and J. Harju, "TCP NewReno throughput in the presence of correlated losses: The slow-but-steady variant," Proc. IEEE INFOCOM 2006, Apr. 2006.

[7] R. Wang, K. Yamada, M. Yahya Sanadidi, and M. Gerla, "TCP with sender-side intelligence to handle dynamic, large, leaky pipes", IEEE Journal on Selected Areas in Communications, 23(2):235-248, 2005.

[8] S. Biaz, N. H. Vaidya, "Discriminating congestion losses from wireless losses using inter-arrival times at the receiver," in Proc. IEEE Symp. Application-Specific Systems and Software Engineering and Technology (ASSET'99), pp. 10–17, Mar. 1999.

[9] IEEE Std. 802.11, "Wireless LAN Media Access Control (MAC) and Physical Layer (PHY) Specification, "1999.

[10] S. Cen, P. C. Cosman, and G. M. Voelker, "End-to-end differentiation of congestion and wireless losses," IEEE/ACM Trans. Networking, vol. 11, no. 5, pp. 703–717, Oct. 2003.

[11] Özgür B. Akan, and Ian F. Akyildiz, "ARC: The Analytical Rate Control Scheme for Real-Time Traffic in Wireless Networks," IEEE/ACM Trans. Networking, vol. 12, pp.634-644, Aug. 2004.

[12] P. Mehra, C. De Vleeschouwer, and A. Zakhor. "Receiver-driven bandwidth sharing for TCP," In Proceedings of IEEE INFOCOM, vol. 2, pp. 1145-1155, Apr. 2003.