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TCP NCE: A unified solution for non-congestion events to improve the performance of TCP over wireless networks

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Abstract

In this article, we propose a unified solution called Transmission Control Protocol (TCP) for Non-Congestion Events (TCP NCE), to overcome the performance degradation of TCP due to non-congestion events over wireless networks. TCP NCE is capable to reduce the unnecessary reduction of congestion window size and retransmissions caused by non-congestion events such as random loss and packet reordering. TCP NCE consists of three schemes. Detection of non-congestion events (NCE-Detection), Differentiation of non-congestion events (NCE-Differentiation) and Reaction to non-congestion events (NCE-Reaction). For NCE-Detection, we compute the queue length of the bottleneck link using TCP timestamp and for NCE-Differentiation, we utilize the flightsize information of the network with a dynamic delay threshold value. We introduce a new retransmission algorithm called 'Retransmission Delay' for NCE-Reaction which guides the TCP sender to react to non-congestion events by properly triggering the congestion control mechanism. According to the extensive simulation results using qualnet network simulator, TCP NCE acheives more than 70% throughput gain over TCP CERL and more than 95% throughput improvement as compared to TCP NewReno, TCP PR, RR TCP, TCP Veno, and TCP DOOR when the network coexisted with congestion and non-congestion events. Also, we compared the accuracy and fairness of TCP NCE and the result shows significant improvement over existing algorithms in wireless networks.

Keywords: Wireless Networks, TCP, Congestion loss, Non-congestion events

Introduction

Transmission Control Protocol (TCP) [1] is the most popular transport layer protocol used in the current internet. The pervasiveness of the internet in combination with the increased use of wireless technologies makes TCP over wireless networks an important research topic. TCP provides connection-oriented, end-to-end in-order delivery of packets to various applications. In wireless networks, packets are transmitting with the presence of wireless links. When TCP operates in wireless networks, the end-to-end performance of TCP degrades significantly because of the unnecessary usage of TCP congestion control algorithms. The congestion control algorithms of TCP are designed for wired networks with the assumptions of order packet delivery and error-free transmission. As a result, when the receiver receives out-of-order packets, it will send back a duplicate acknowledgment to its corresponding sender. At the sender side, when the number of duplicate acknowledgments (dupacks) which is equal to three, the sender consider it as a loss due to network congestion and triggers the congestion control algorithm such as fast retransmission and will reduce the size of congestion window. However, in wireless networks, the packet loss can be due to either congestion or non-congestion losses such as random loss due to transmission errors. In fact, the latter case is more common than the former case.

In addition to that, recent internet measurement studies show that packet reordering plays an important role in the packet transmission and it is not a rare event in wireless networks [2,3]. As a result, three dupacks may cause due to non-congestion events such as random loss or packet reordering. In the former case, the TCP sender reduces the size of congestion window

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unneccessarily and hence wasting bandwidth and in the latter case, the TCP sender not only reduce the size of congestion window but also retransmit the packet needlessly. Several loss differentiation algorithms have been proposed for improving the performance of TCP. Among that TCP NewJersey [4], TCP Veno [5], and TCP CERL [6] have been proposed to differentiate congestion losses from random losses whereas RR TCP [7], TCP PR [8], and TCP DOOR [9] have been proposed to differentiate congestion losses from packet reordering. However, these algorithms have no unified solution to differentiate the non-congestion events when the sender receives three dupacks [10]. When random loss and packet reordering are co-existed, the number of unnecessary retransmission increases and will have adverse effects on TCP and its congestion control mechanisms, which deteriorate the poor performance of TCP over wireless networks. As a result, it is an important issue of TCP to guide the TCP sender for triggering the congestion control algorithms properly by providing a unified solution for non-congestion events in addition to network congestion to improve the performance of TCP over wireless networks.

To address this issue, we propose a unified solution called TCP NCE for improving the performance of TCP over wireless networks by reducing the unnecessary reduction of congestion window size and retransmissions due to non-congestion events. Our unified solution TCP NCE has three schemes.

1. NCE-Detection which is used for detecting the non-congestion events from network congestion by computing the queue length of the bottleneck link using TCP timestamp based RTT measurement.
2. NCE-Differentiation is used for differentiating the non-congestion events especially random loss from packet reordering by utilizing the flightsize information of the network with a dynamic delay threshold value.
3. NCE-Reaction guides the TCP sender to react to non-congestion events accordingly by introducing a new retransmission algorithm called 'Retransmission Delay' which delays the packet retransmission upto the expiration of the dynamic delay threshold value.

We evaluated TCP NCE with other TCP schemes such as TCP Veno, TCP CERL, RR-TCP, TCP PR, TCP NewReno, and TCP DOOR and compared the performance by using the metrics such as end-to-end throughput, accuracy, and fairness through extensive simulations using Qualnet 4.5 [11]. Simulation results show that TCP NCE has significant improvement over other popular TCP variants. The rest of this article is organized as follows. In 'TCP in wireless networks' section, we describe the behavior of TCP in wireless networks. We briefly summarizes the previous works in 'Related work' section. In 'TCP-NCE' section, we introduce TCP NCE and its three schemes. We describe the performance evaluation of TCP NCE with other TCP variants in 'Performance evaluation' section. Finally, 'Conclusion' section concludes this article and highlights future works.

TCP in wireless networks

TCP was designed to provide reliable connection-oriented services between any two end systems on the internet. The congestion control algorithms of TCP consists of Slow-Start, Congestion Avoidance, Fast Retransmission and Recovery as shown in Figure 1 in conjuction with several different timers.

In Slow-Start, the size of congestion window (cwnd) increases exponentially at the sender whereas in Congestion Avoidance algorithm, cwnd increases linearly. Fast Retransmission and Recovery algorithm triggers only when the sender receives three dupacks. As a result, when the sender receives three dupacks, traditional TCP assumes that the loss of packets are caused by network congestion. However, when TCP deployed in wireless networks, this assumption is no longer true. This is because in wireless networks non-congestion events are more common than network congestion. When TCP sender receives three dupacks, the sender has to consider non-congestion events as shown in Figure 1 in addition to network congestion. If the three dupacks is due to packet reordering then the sender need not retransmit the packets by reducing the size of cwnd. On the other hand, if the three dupacks is caused by random loss, the sender has to retransmit the packet without reducing the size of cwnd. Below, we discuss the main causes of non-congestion events in wireless networks.

Random Loss

In wireless networks, the loss of packets are due to transmission errors which is more common than congestion loss. The frequent causes of non-congestion losses in wireless networks are high bit error rate in the wireless medium, exposed and hidden terminal problems, multipath routing, MAC designs etc. [12]. Packet losses due to channel collision depend on the number of contention of nodes.

Moreover, in wireless networks, the interferences between neighboring nodes are much higher compared to local area networks. As a result, the bit error rates of wireless links are more variable in wireless medium. As shown in Figure 2, TCP sender transmits packet from P_1 to P_5. Among that packet P_1 was lost due to transmission error. As a result, the receiver sends three

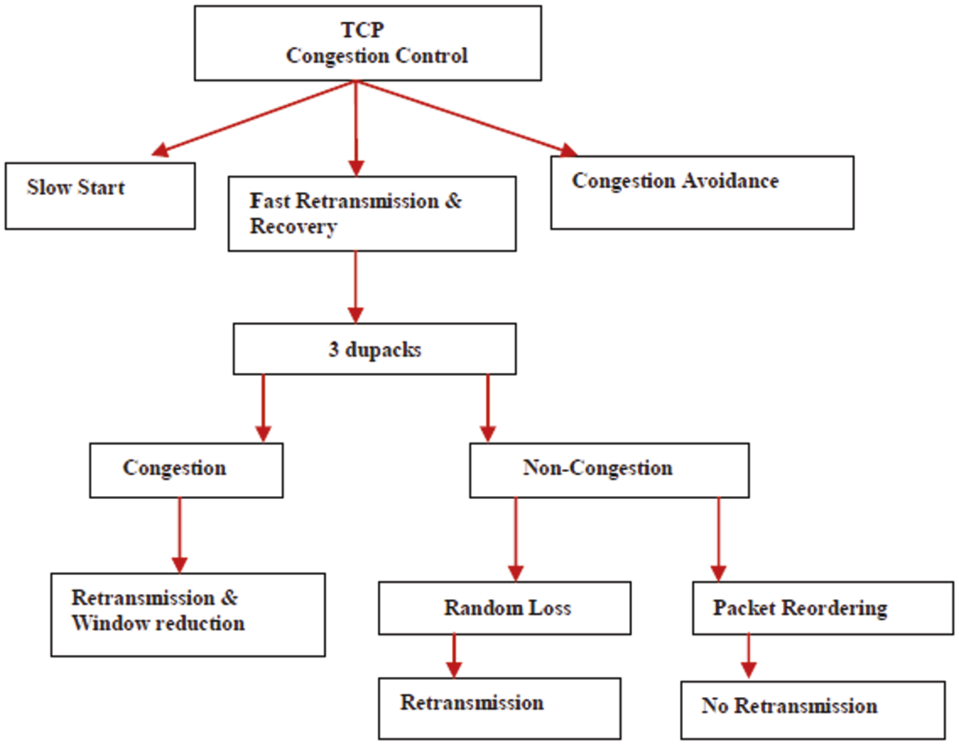


Figure 1 Congestion control algorithms of TCP.

dupacks by packets P_2 to P_4. Upon the arrival of three dupacks the sender trigers fast retransmission unnecessarily and retransmits the packet by reducing the size of cwnd needlessly and thereby degrade the performance of TCP.

Packet Reordering

Packet reordering [10] refers to the network behavior, where the relative order of packets is altered when these packets are transported in the network. As shown in Figure 3, the packets P_2, P_3, P_4, P_5, and P_1 are sent in the order of P_1, P_2, P_3, P_4, and P_5. However, the packet P_1 reaches the destination after the arrival of P_5. As a result, the receiver sends three dupacks of packet P_1 to the sender. Upon receiving the three dupacks of packet P_1, the sender trigers fast retransmission and retransmits the packet by reducing the size of cwnd needlessly. In wireless networks, packet reordering may cause due to route fluttering, inherent parallelism in routers, link-layer retransmissions, router forwarding lulls, multipath routing etc. TCP inability to distinguish packet reordering from packet loss causes unnecessary retransmissions, slow down the growth of cwnd and reduces the efficiency of the receiving TCP.

For delivering information successfully over wireless networks, the modification of TCP congestion control algorithms is necessary especially fast retransmission and recovery. For the higher performance of TCP over

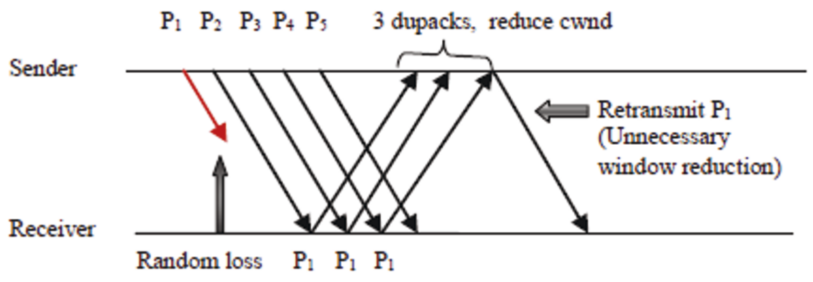


Figure 2 Fast retransmisssion due to random packet loss.

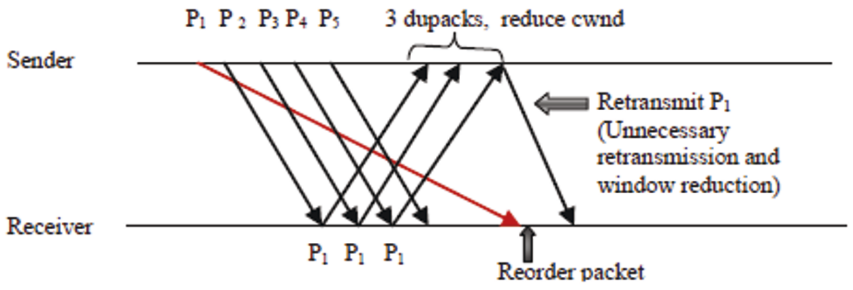


Figure 3 Fast retransmisssion due to packet reordering.

wireless networks, the sender not only needs to differentiate non-congestion losses from congestion losses but also need to differentiate the reordering of packets from random losses as it is not a rare event in wireless networks.

Related work

In this section, we describe a set of algorithms that have been proposed for improving the performance of TCP that TCP NCE is compared to in this article. 'Solutions for random loss' section gives an overview of three random loss solutions and 'Solutions for packet reordering' section gives an overview of three packet reordering solutions. In 'Other solution' section, we describe TCP NewReno as it is the most widely deployed protocol in current internet.

Solutions for random loss

TCP Veno differentiate the random losses from congestion losses by adopting the mechanism of TCP Vegas [13] to estimate the size of the backlogged packets (N) in the buffer of the bottleneck link. The calculation of N is given below.

$$N = \text{Diff} * \text{BaseRTT} \tag{1}$$

where Diff is the difference between expected and actual rates and BaseRTT is the minimum measured round-trip times. The Expected and Actual rates are measured as,

$$\text{Expected} = \text{cwnd}/\text{BaseRTT} \tag{2}$$

$$\text{Actual} = \text{cwnd}/\text{RTT} \tag{3}$$

where cwnd is the current size of congestion window and RTT is the measured smoothed round-trip time. TCP Veno used the measurement of N to differentiate the type of packet loss. Specifically, when a packet is lost, Veno compare the measured value of N with β (backlog threshold). If $N < \beta$, TCP Veno assumes the loss to be random rather than congestive, otherwise Veno assumes the loss to be congestive.

TCP CERL (Congestion Control Enhancement for Random Loss) distinguishes random losses from congestion losses based on a dynamic threshold value. TCP CERL is a sender side modification of TCP Reno. TCP CERL and TCP Veno are similar in concept. However, the mechanisms utilized by TCP CERL differ greatly from those used in TCP Veno. TCP CERL utilizes the RTT measurements made throughout the duration of the connection to estimate the queue length (l) of the link, and then estimates the congestion status. The calculation of l is as shown below,

$$l = (\text{RTT} - T)\,B \tag{4}$$

where RTT is the measured round-trip time, B the bandwidth of the bottleneck link, and T the smallest RTT observed by the TCP sender and l is updated with the most recent RTT measurement. Using the values of l and A (a constant which is equal to 0.55), TCP CERL used to set the dynamic threshold value (N),

$$N = A * l_{\max} \tag{5}$$

where $l_{\max}$ is the largest value of l observed by the sender. If $l < N$ when a packet loss is detected via three dupacks, TCP CERL will assume the loss to be random rather than congestive. Otherwise, TCP CERL will assume the loss is caused by congestion.

TCP NewJersey introduced as the extension of TCP Jersey [14] as a router assisted solution for differentiating random packet loss from congestion loss and react accordingly. TCP New Jersey has two key components in its scheme, timestamp based available bandwidth estimation (TABE) and congestion warning scheme. To estimate the available bandwidth, TCP Jersey follows the same idea of TCP Westwood's rate estimator to observe the rate of acknowledged packets by acknowledgments (ack), but with a different implementation. Upon receiving n acks, the available bandwidth B_n is estimated as shown below.

$$\mathrm{B_n} = \frac{\delta \mathrm{B_{n-1}} + \mathrm{L_n}}{(\mathrm{t_n} - \mathrm{t_{n-1}}) + \delta} \tag{6}$$

where δ is the TCP's estimation of the end-to-end RTT delay at time t_n, L_n the size of the data, t_{n-1} the arrival time of the previous ack, and t_n the arrival time of *n*th packet at the receiver. The sender interprets the estimated rate as the optimal congestion window (ownd) in unit of the size of segment (*S*) and is calculated as,

$$\mathrm{ownd} = (\delta * B_n)/S \tag{7}$$

When the sender receives three dupacks, TCP New-Jersey checks whether the received ack has congestion warning mark or not. If it has mark, TCP NewJersey assumes that the loss is caused by network congestion and proceeds as TCP NewReno [15] after estimating the available bandwidth for adjusting the size of cwnd, whereas, if the ack has no mark, TCP NewJersey assumes the loss is due to non-congestion and retransmits the dropped packet without reducing cwnd.

Solutions for packet reordering

RR TCP, the reordering-robust TCP proposed as an extension of the Blanton-Allman algorithms [16]. RR TCP is a sender side solution, which adjust the threshold (dupthresh) of dupacks dynamically to detect and recover from spurious fast retransmissions. However, this solution differs in three ways compared to Blanton-Allman algorithm. First, RR TCP uses a different mechanism to adjust dupthresh dynamically. The author utilizes a combined cost function for retransmission timeouts (RTO) and false fast retransmissions to adapt the false fast retransmit avoidance ratio (FA ratio). Second, the authors considered the extended version of the limited transmit algorithm [17] which permits a source to send up to one ack-clocked additional cwnd's worth of data. Third, for the estimations of RTT and RTO the authors proposed an idea to correct the sampling bias against long RTT samples. Compared to Blanton-Allman algorithm, RR TCP needs excessive computational and storage overhead.

TCP Persistent packet reordering (TCP PR) proposed to improve the poor performance of TCP under persistent packet reordering by delaying solely on timers. To detect a packet loss, TCP PR maintained timers to keep track of how long ago a packet was transmitted instead of relying dupacks. When TCP PR detects a packet drop, the sender reduces the size of cwnd into half and carried out congestion avoidance algorithm. In order to avoid the over-reaction to congestion, TCP PR will not reduce the size of cwnd for subsequent occasional segment drops in the same cwnd. When more than half of a cwnd's worth of packets is detected to be lost, cwnd is set to one and triggers the slow start algorithm. One of the major advantages of TCP PR is that it can able to maintain ack clocking in the presence of packet reordering. Another merit is the new RTT and RTO estimator are very effective in packet reordering. However, TCP PR has some limitations. First, TCP PR is computationally expensive and second, the new RTT estimator is overly sensitive to spikes in RTT.

TCP DOOR (Detection of out-of-order and response) is a state reconciliation method, to solve the performance problems caused by spurious retransmissions and to eliminate the retransmission ambiguity by disabling the congestion response for a period of time. In order to detect reorder packets, TCP DOOR insert the sequence numbers of data and acks on each data packets and acks, respectively. Upon the detection of out-of-order events, the sender can either disable the congestion response or trigger congestion avoidance algorithm. TCP DOOR detects out-of-order events only after a route has recovered from failures. As a result, TCP DOOR is less accurate and responsive than a feed-back based approach, which can determine whether congestion or route errors occur in a responsive manner.

Other solution

TCP NewReno changes the fast retransmit algorithm for eliminating Reno's waiting time for the retransmission timeout when multiple segments are lost within a single window. More than 76% of web servers deployed TCP NewReno as the standard protocol [18]. In fast retransmission, when the sender receives three dupacks the current implementation of TCP NewReno stores the highest sequence number transmitted in a variable 'Recover', retransmit the lost segment and set cwnd to ssthresh (slow start threshold) plus 3 * mss (maximum segment size). Then, TCP sender enters into fast recovery and increment cwnd by one mss for each additional dupacks and transmits new packets, if allowed by the new value of cwnd and the receivers advertised window. When the sender receives a new ack including Recover, the sender sets cwnd to ssthresh and goes to congestion avoidance state. On the other hand, if this new ack does not include Recover, then the sender consider it as a partial ack, retransmit the first unacknowledged segment and add back one mss to cwnd and send a new segment if permitted by the new value of cwnd. This way, TCP NewReno can recover multiple packet losses from a single window of data. However, TCP NewReno assumes all duplicate acks are due to the cause of network congestion.

Opposed to above approaches, TCP NCE is able to detect, differentiate and react to non-congestion events accurately while maintaining responsiveness against situations with purely congestive loss. TCP NCE can increase the performance of TCP over wireless networks by reducing the unnecessary reduction of cwnd size and spurious retransmissions due to non-congestion events.

TCP NCE

In this section, to tackle the end-to-end performance degradation problem of TCP over wireless networks, we introduce our unified solution named as TCP NCE, which is capable of reducing the unnecessary retransmissions and reduction of cwnd size by detecting, differentiating, and reacting to non-congestion events while maintaining responsivess against situations with purely congestive loss. In the following subsections, we describe the three schemes of TCP NCE such as NCE-Detection, NCE-Differentiation, and NCE-Reaction.

NCE-Detection

For detecting the non-congestion events from network congestion, we measure the queue length of the bottleneck link of a TCP connection. We use a similar method to that used in [6] for measuring the queue length. Compared to former method, the main difference lies in the measurement of RTT. When computing the queue length, the estimation of RTT is important because RTT includes the delays of forward and reverse paths. In our scheme, we calculate RTT using the timestamp option fields defined in RFC 1323 [19] as shown in Figure 4. The timestamp option contains two fields namely, timestamp (TS) value and timestamp echo reply. Each field has four bytes.

When a segment leaves the sender, the field TSval stores the current time of sending packet. If that segment reaches the receiver, it stores the TSval. When the receiver sends ack, it attaches the time of previously received segment in the TSecr field. When the source receives this ack, it takes the TSecr value and use for calculating the RTT as shown in (8).

$$\mathrm{RTT} = \text{current time} - \mathrm{TSecr} \tag{8}$$

This way of RTT measurement works correctly in the face of non-congestion events especially in the case of packet reordering rather than using an algorithm that samples one RTT per window of data. The reason is, in the presence of spurious fast retransmits, TCP is likely to have to discard most of its potential samples. As a result, the RTT estimator will not sample the RTT very frequently and may not keep a good estimate of the RTT [20]. By using the measured RTT, we calculated the queue length (Ql) of the bottleneck link as shown in (9),

$$\mathrm{Ql} = B(\mathrm{RTT}_{\mathrm{now}} - \mathrm{RTT}_{\mathrm{min}}) \tag{9}$$

where $\mathrm{RTT}_{\mathrm{now}}$ is the current round-trip time when the sender receives an ack, $\mathrm{RTT}_{\mathrm{min}}$ is the minimum RTT observed by the TCP sender, and B is the bandwidth of the bottleneck link. As shown in Figure 5, for detecting the non-congestion events at the time of receiving the three dupacks, the sender checks the current queue length which is greater than a threshold value. If it is greater than a threshold value (Th-Val), the TCP sender confirms that the dupacks is due to network congestion and proceeds as TCP NewReno otherwise the sender assumes that the dupacks is due to non-congestion events and delays the retransmission upto the expiration of dynamic delay threshold value.

Determination of threshold value

For determining the threshold value in order to detect non-congestion events from network congestion, we assume that the router uses drop-tail queueing policy as it is the most widely deployed router queue management scheme [21]. Figure 6 shows the network environment that we considered for determining the threshold value. There are '*n*' TCP flows from source (S to Sn) connected to the router R1 and the router R2 connected to the destinations (D to Dn). The congested uplink from R1 and R2 is with capacity *C*. Based on drop-tail algorithm, when the queue length becomes equal to the buffer size (BS), then all the newly arrived packets are being dropped. As a result, for determining the threshold value we use the percentage of usage buffer size. However, how much percentage of buffer size we need to use for determining the threshold value for detecting non-congestion event from congestion? For that, we divide the router buffer space into three different loads

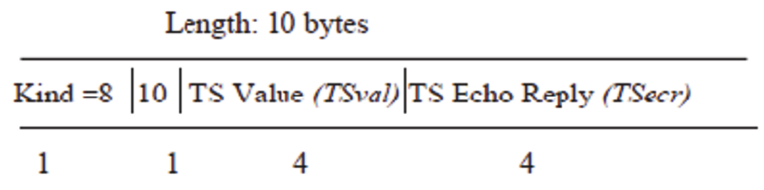


Figure 4 TCP Timestamp options.

```
Algorithm 1: Whenever the sender receives three dupacks

(1)  If (three dupacks) {
(2)      If (Ql > Th-Val){
(3)          Report congestion
(4)      } else {
(5)          Report non-congestion
(6)  }
(7)  }
```

Figure 5 Psuedocode of TCP NCE-Detection of non-congestion events.

as shown in Figure 7. It consists of light load, medium load and heavy load.

When the router buffer space is less than 30%

We consider it as a light load and the router is not congested at this time. As a result, when the sender receives three dupacks, we can predict that the three dupacks is due to non-congestion events.

When the router buffer space is less than 90% and greater than 30%

We consider it as a medium load and the router is not congested at this time, but it is easy to become congested at the next period of time. In this case also, we can assume that the arrival of three dupacks is due to non-congestion events.

When the router buffer space is greater than 90%

It means that the router is in the heavy load and it is under congestion at this time and the buffer will easily overflow, which results the packet loss due to network congestion.

Furthermore, for fixing the threshold value we did experiments by using different buffer loads in terms of accuracy as it is the most important performance metric of both events. Because when accuracy of non-congestion event increases, obviously the TCP performance also increases [22-24] compared to traditional TCPs. The topology we used for our experiments as shown in Figure 6. We use TCP connection with 3% random packet loss and 1% packet reordering with bottleneck capacity 6 Mbps and propagation delay 10 ms. We measured the accuracy of non-congestion events ($NCE_{accuracy}$) using equation (10),

$$NCE_{accuracy} = NCP_{exact}/NCP_{total} \tag{10}$$

where NCP_{exact} is the number of non-congestion packets exactly identified as non-congestion events and NCP_{total} is the total number of non-congestion packets caused by transmission errors and packet reordering. Figure 8 shows the result of accuracy for varying buffer loads. It is evident that when buffer load increases upto 90%, the accuracy of non-congestion event becomes higher. On the other hand, when the buffer load is greater than 90%, the accuracy of non-congestion event decreases. Because when the buffer becomes full, all the incoming packets may drop. As a result, if more than one TCP connection, all the sender receives three dupacks and the sender assumes that the packet loss are due to network congestion even in non-congestion events and thereby decrease the accuracy. As a result, in order to use the buffer resources fully, we set the

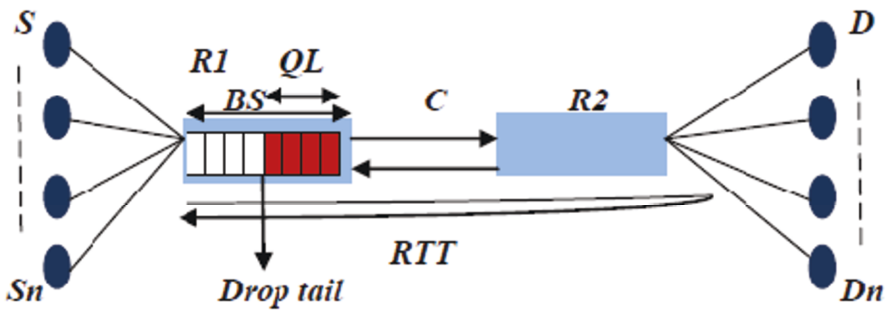


Figure 6 Network environment.

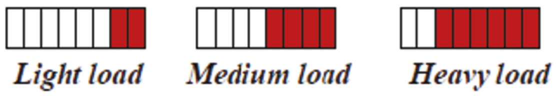


Figure 7 Different buffer loads.

threshold value which is equal to 90% of the buffer size. Moreover, this value has another advantage. That is, when the queue length becomes greater than 90% at the time of receiving three dupacks, we reduce the size of cwnd and can avoid the loss of multiple packet drops from different TCP sources due to network congestion.

NCE-Differentiation

When the sender detects three dupacks is the cause of non-congestion event, the sender of TCP NCE computes a dynamic delay threshold (delay-thresh) for differentiating whether the received dupacks are due to random packet loss or packet reordering and delays the retransmission upto the expiration of delay-thresh. For computing the delay-thresh, we need to consider three things.

(1) If the value of delay-thresh is high, then retransmission timeout happens and the packet gets retransmitted by reducing the size of cwnd to one.

(2) If the value of delay-thresh is too small, then the TCP will continue to retransmit packets unnecessarily.

(3) If the value of delay-thresh is too high, retransmission may not triggered leading to retransmission timeout.

As a result, by considering these things TCP NCE computes the best value of delay-thresh by utilizing the flightsize information of the network. Let ‘$\text{Pack}_{\text{LSent}}$’ be the last sent packet from the source and ‘$\text{Pack}_{\text{LAck}}$’ be the last acknowledged packet from the receiver. Then the total number of outstanding packets ‘$\text{Pack}_{\text{TotNum}}$’ in the network at the time of receiving dupacks is calculated as shown below,

$$\text{Pack}_{\text{TotNum}} = \text{Pack}_{\text{LSent}} - -\text{Pack}_{\text{LAck}} \tag{11}$$

From the total number of outstanding packets in the network, the sender receives three dupacks. That means three more packets that has left the network, then the remaining packets in the network ‘$\text{Pack}_{\text{Remain}}$’ is,

$$\text{Pack}_{\text{Remain}} = \text{Pack}_{\text{TotNum}} - \text{ndupacks} \tag{12}$$

After receiving ndupacks which is equal to three, the sender can expect this much of additional dupacks (add-dupacks) from the receiver. As a result, we can set the value of delay-thresh as,

$$\text{delay - thresh} = \text{Pack}_{\text{Remain}} \tag{13}$$

When the sender receives add-dupacks in addition to first three, which is greater than or equal to the value of $\text{Pack}_{\text{Remain}}$, then that add-dupacks are the sign of newly sent packets. As a result, the TCP sender can confirms that the corresponding packet is lost from the old window of data due to transmission errors. Otherwise, the sender confirms that the add-dupacks were due to reordered packets because if the packet is reordered from

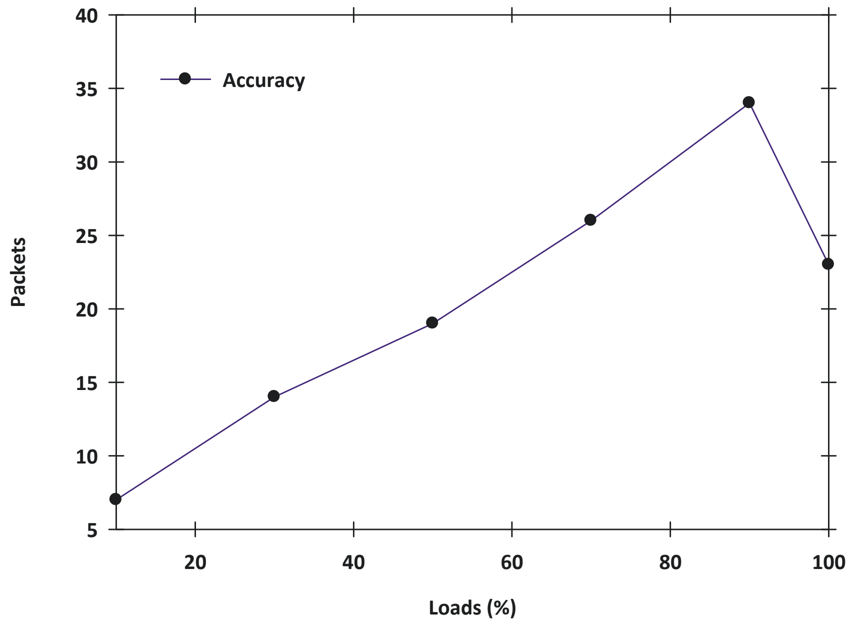


Figure 8 Accuracy of detecting non-congestion events with different buffer loads.

Algorithm 2: Whenever the sender detects non-congestion event

```
1. If (non-congestion event)
2. {
3.   send new packet
4.   compute 'delay-thresh'
5.   cwnd increment to one mss
6.   ssthresh= now
7.   Retransmission Delay()
8. }
```

Figure 9 **Psuedocode of TCP NCE Reaction of detecting non-congestion events**.

one window of data, the reordered packet may reach the destination before the packets from new window of data reaches the destination [25]. Not only that, the time taken to reach the newly sent packet to the destination is much higher than the arrival of the reordered packet at the destination [26]. As a result, our delay threshold value helps the TCP to avoid unnecessary retransmissions and reduction of cwnd.

NCE-Reaction

When the sender receives three dupacks and the current queue length is less than the threshold value, then the sender assumes that these dupacks are the sign of non-congestion events. In this situation, as shown in Figure 9, instead of triggering fast retransmission, the sender of TCP NCE sends a new packet by increment the value of cwnd to one mss without reducing the size of ssthresh and computes the retransmission delay-thresh value. This can maintain the ack clocking. Then the sender enters into the retransmission delay algorithm and receives add-dupacks. We introduce a new 'Retransmission Delay' algorithm for delaying the retransmission upto the expiration of dynamic delay-thresh value. As shown in Figure 10, in retransmission delay, the sender receives add-dupacks and for each add-dupacks the sender increments the size of cwnd to one mss and send new packets allowed by the value of cwnd. When add-dupacks is greater than or equal to the value of delay-thresh, the sender confirms that the packet is lost due to transmission errors and retransmit the packet immediately without reducing the size of cwnd and ssthresh. Otherwise, the sender ignores the add-dupacks and send packets continuously until the size of cwnd greater than the value of ssthresh.

Figure 11 presents an example of how TCP NCE differentiates random loss from reordering of packets. Consider that seven packets (5 to 11) are sent from a TCP sender to a TCP receiver in the order shown in Figure 11. Among that, the packet 5 is lost and the sender gets three dupacks of packet 5 by packets 6, 7, and 8. Consider the three dupacks are due to non-congestion event. As a result, when the sender receives three dupacks, it sends a new packet (12) to the receiver and computes the delay-thresh value by using the outstanding packets in the network. In this example, the total number of outstanding packets in the network is 7 using Equation 11. From that, the sender receives three dupacks and sets the delay-thresh value to 4 using Equation 12. For each add-dupacks, the sender sends new packets (13 to 15) allowed by the size of cwnd. When the newly sent packet (12) reaches the destination, the receiver sent one more add-dupacks to the sender which is greater than or equal to the value of delay-thresh. As a result, the sender confirms that the packet is lost due to transmission error and retransmits the packet immediately without reducing the size of cwnd. Otherwise the sender can confirm the packet is reordered and continue sending new packets for every dupacks until the value of cwnd greater than ssthresh. This helps the sender to increase the throughput of TCP by reducing unnecessary retransmissions and window reductions.

Behavior of TCP NCE

In this subsection, we describe the congestion control algorithms of TCP NCE and how the TCP sender behaves upon the arrival of three dupacks. We adopt the Slow Start (SS) and Congestion Avoidance (CA) algorithms of original TCP NewReno. Also, we adopts the fast retransmission and recovery algorithms of TCP NewReno when the sender of TCP NCE detects the packet losses due to network congestion. At the

Algorithm 3: Whenever the sender receives add-dupacks

```
1. Retransmission Delay() {
2. Receives add-dupaks
3.  Increment cwnd
4. Send new packets
5.   If (add-dupacks ≥ 'delay-thresh'){
6.      Retransmit the packet
7.      cwnd= now
8. }else {
9.      Continue sending packets until cwnd > ssthresh
10. }
11. }
```

Figure 10 Psuedocode of Retransmission Delay procedure.

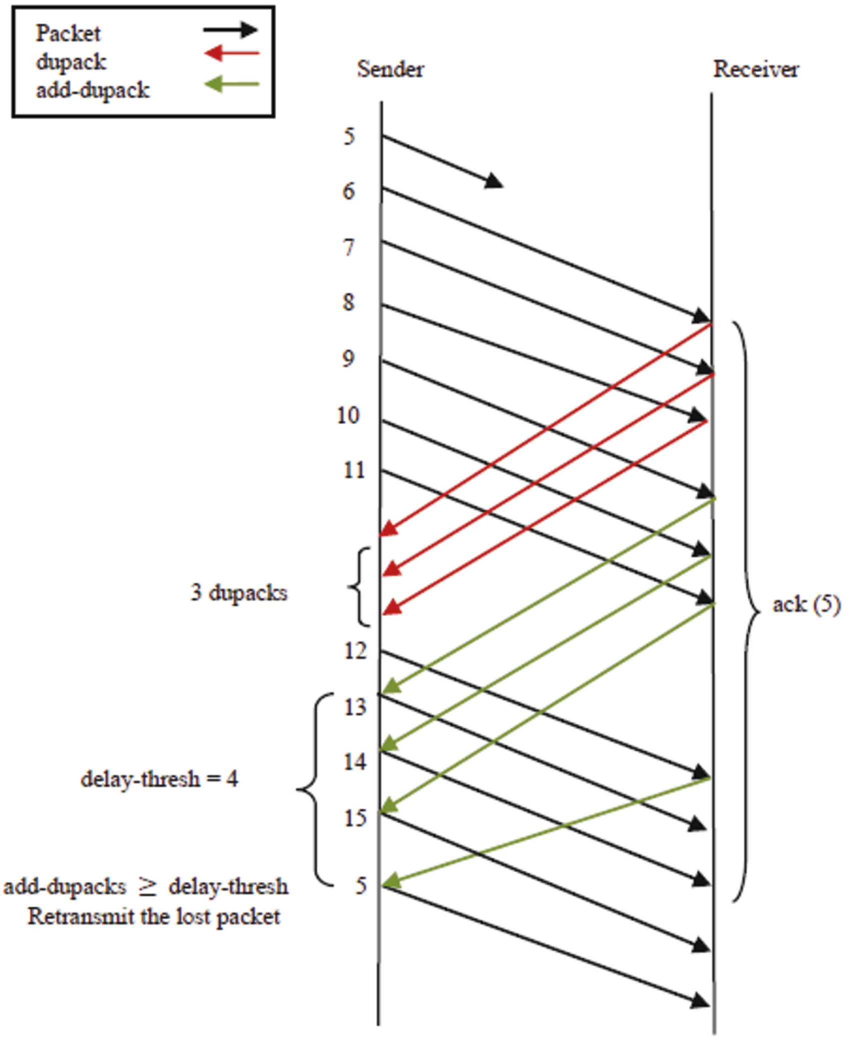


Figure 11 Example of TCP NCE detection of non-congestion event.

beginning of the TCP connection, the sender enters the SS phase, in which the cwnd increases by one mss for every receiving acks and grows the cwnd exponentially.

When the value of cwnd reaches the maximum of ssthresh which is equal to 65535 bytes, the sender enters the CA phase. During this phase, the sender increases its cwnd size linearly for every RTT. This linear growth of transmission rate helps the sender to slowly probe the available network bandwidth. When timeout occurs the sender retransmits the packet by reducing the size of cwnd to one mss and goes to SS phase as shown in Figure 12. When the sender receives three dupacks, it checks the current queue length is greater than threshold value. If yes, the senders triggers the fast retransmission algorithm of TCP New-Reno and retransmit the corresponding packet by storing the highest sequence number in the variable 'Recover' and then enters the fast recovery algorithm. During fast recovery the sender receives add-dupacks. For each add-dupacks the sender increments the size of cwnd by one mss and send new packets allowed by the value of cwnd. When the sender receives new ack including the value stored in the varaible Recover, the sender sets the size of cwnd to the value of ssthresh and then goes to CA phase. On the other hand, if the current queue length is less than the threshold value at the time of receiving three dupacks, the sender sends a new packet instead of retransmission by incrementing the size of cwnd to one mss without reducing the size of ssthresh and triggers the retransmission delay algorithm after computing the delay-thresh for detecting the non-congestion event whether the three dupacks is due to random loss or packet reordering. The box with green lines represents the procedure of retransmission delay algorithm. In retransmission delay, the sender receives add-dupacks and for each add- dupacks the sender sends new packet allowed by the value of cwnd. When the add-dupacks greater than or equal to the value of delay-thresh, the sender retransmit the packet by keeping the current size of cwnd otherwise the sender continues sending new packets until the value of cwnd greater than ssthresh.

Performance evaluation

In this section, we present the performance evaluation of TCP NCE by showing the metrics such as throughput, accuracy, and fairness. The below subsections shows the experimental set up and results of TCP NCE compare with other TCP variants.

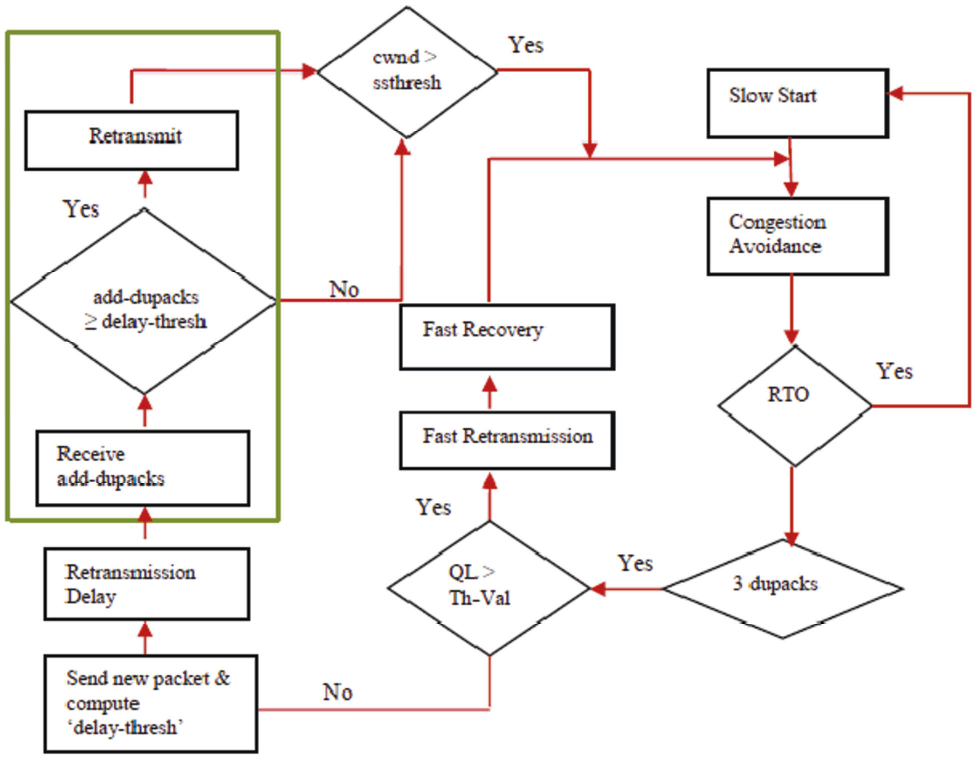


Figure 12 Behaviour of TCP NCE.

Experimental setup for end-to-end throughput performance

We have used three different network topologies for evaluating the performance of TCP NCE throughput in order to show that our solution is indeed efficient in different network conditions. Simulation topologies are illustrated in Figure 13. As shown in Figure 13a, in infrastructure based wireless network, a TCP connection between sender (S) and receiver (D) are connected to wired and wireless network and is routed through a base station BS. The wired link between the S and BS has a bandwidth of 100 Mbps with propagation delay 10 ms. The wireless link between BS and D has a bandwidth of 6 Mbps with propagation delay of 50 ms otherwise noted. On the other hand, in multi-hop wireless network as shown in Figure 13ba TCP connection traverses over five routers R1 to R5 with six hops to the receiver from sender. Each wireless link has a bandwidth of 6 Mbps with propagation delay of 10 ms unless otherwise stated. In Figure 13c, we simulate a dumbell shaped wireless network with 6 TCP sender (S1 to S*n*) and receivers (D1 to D*n*) having one bottleneck link *L*. R1 and R2 are two routers with drop-tail queueing policy. In all simulations, the length of the queue at routers is set to 50 kbytes and the maximum segment size of TCP is 512 bytes. The traffic source we implemented using FTP. The maximum window limit is set to 32 packets. the size of an ack packet is same as the size of data packet. We enabled the delay ack alogrithm. DSR is the main routing protocol in our simulation with a maximum message buffer size is set to 50 packets. The duration of our simulations was set to 300 s. During simulations, data packets are continuously transmitted upto the end of simulation and the source of all TCP flows originated from S1 to S*n*. The simulations have been conducted using Qualnet version 4.5, a software that provides scalable simulations of wireless networks. We compared the throughput with the main TCP versions and loss differentiation algorithms. The throughput '*t*' is calculated as specified in [27], $t = s/\text{stime}$, where '*s*' is the maximum sequence number transmitted and acknowledged and 'stime' is the simulation time.

In order to achieve our aims in the experiment, we used different scenarios of non-congestion events under three different network conditions. First condition is designed to check the throughput of random loss detection according to the rate of packet loss, bandwidth, delay, number of hops, and variation of cwnd size. Thus, in this condition, all packet losses are caused by transmission errors. The second condition aims to cause random loss and packet reordering according to the rate of delay, bandwidth, packet reordering rate, packet loss rate, and percentage of unneccssary retransmissions. Finally, in third condition, we planned to observe the throughput in terms of congestion loss, random loss, and packet reordering according to the rate of queue size, bandwidth, packet loss, delay, and number of hops in order to confirm that TCP NCE is efficient in the coexistence of congestion and non-congestion events.

Throughput evaluation of TCP NCE under first condition

In this section, we demonstrate the results of throughput performance in presence of random loss according

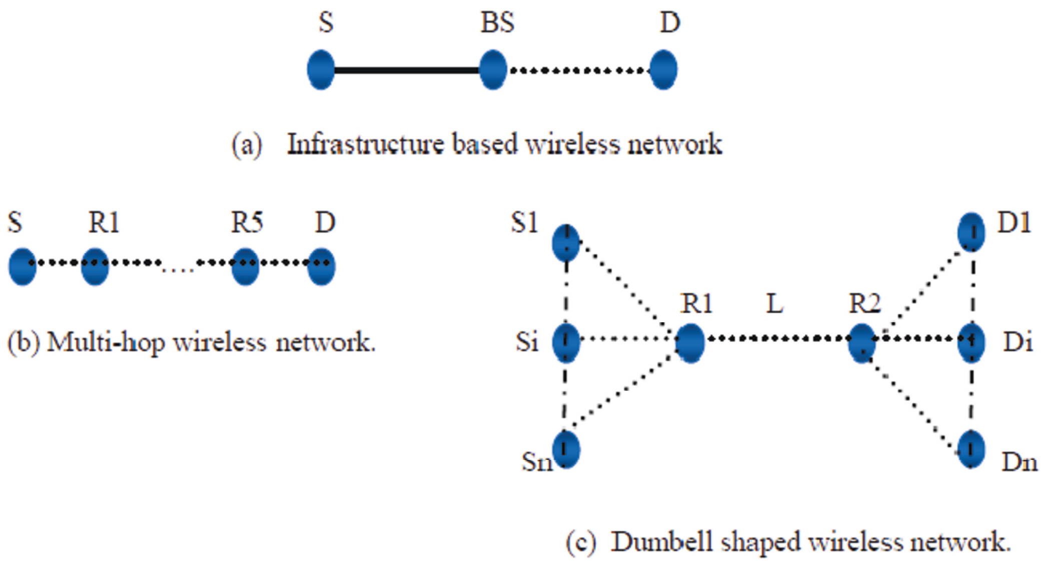

S
BS
D
(a) Infrastructure based wireless network
S
R1
R5
D
(b) Multi-hop wireless network.
S1
Si
Sn
R1
L
R2
D1
Di
Dn
(c) Dumbell shaped wireless network.

Figure 13 Network topologies for simulation.

to the rate of loss, bandwidth, delay, and cwnd size using infrastructure based wireless networks. For imposing random loss, we used the exponential error model available in qualnet. The link-layer retransmission is enabled and is set to zero. When the retransmission limit is set to zero there are no reordered packets due to link-layer retransmissions. Figure 14 shows the result of throughput in terms of varying loss rates and link propagation delays. In Figure 14a, the loss rate ranges from 0 to 10%. We run seven different simulations. One with TCP NewReno, one with TCP Veno, one with TCP CERL, one with RR-TCP, one with TCP-PR, one with TCP DOOR, and one with TCP NCE. When the percentage of packet loss rate increases, the throughput of all TCP's decreases. Although the throughput decreases, upto 3% all TCP's have almost similar throughput. But when the loss rate is greater than 3%, TCP CERL and TCP NCE begin to increase its throughput compared to other TCP's. This is because both of the TCP's can detect and differentiate the random packet losses via duplicate acknowledgments effectively and thus it can improve the cwnd evolution and thereby gain higher throughput. However, when the loss rate becomes 7%, TCP NCE acheives 85% greater throughput than TCP CERL and at 10% loss rate TCP NCE has 50% more throughput gain than TCP CERL and 85% more throughput than TCP NewReno. RR-TCP and TCP-PR acheives similar connection throughput and TCP DOOR does not yield any performance gain with respect to the measurement of congestion control algorithms because of no out-of-order event is detected. Figure 14b depicts the result of TCP throughput under varying link propagation delays from 50 to 150 ms in infrastructure wireless network with loss rate 2%. As delay increases, a larger size of cwnd is needed to utilize the full bandwidth of the link. Therefore, the random loss has much higher impact on the throughput of each TCP as the propagation delay of the link increases [6]. Among other TCP's, TCP NCE acheives higher throughput according to the increase in link propagation delays. This is because TCP NCE detect the loss by computing the current queue length using timestamp based RTT measurement. As a result, even high delay TCP NCE can accurately detect the type of loss and can trigger the congestion control meachanism accordingly. In the case of random loss, instead of retransmitting the packet when the sender recieves three dupacks, TCP NCE sends new packet without reducing the size of cwnd and increase the cwnd by one mss for each add-dupacks. This mechanism helps the sender to utilize the bandwidth efficiently and increase the throughput of TCP.

In Figure 14b at 150 ms delay, TCP NCE acheives 90% higher throughput than TCP NewReno and 81% higher than TCP CERL throughput. Figure 15 presents the result of typical variation of TCP throughput under different bandwidths. In this experiment, we set the packet loss rate 3% with delay 50 ms. From the graph, we observe that when bandwidth increases, the throughput also increases. Compared to other TCP's, TCP NCE achives the superiority. When bandwidth greater than 12 Mbps, TCP NCE begins to increase the throughput. This means that TCP NCE can utilize the bandwidth efficiently. However, in the case TCP Veno, due to frequent timeouts TCP Veno always reduce the size of cwnd which leads to the degradation of TCP Veno's throughput. In the case of TCP CERL, it missclassifies some random losses as congestion loss and reduce the size of cwnd unnecessarily. Moreover, other TCP's has no mechanism to detect random losses. As a result, TCP PR, RR TCP, and TCP DOOR frequently reduces the size of cwnd and thereby decrease the throughput performance. Figure 16 illustrates the typical cwnd size of TCP NCE and TCP NewReno according to the

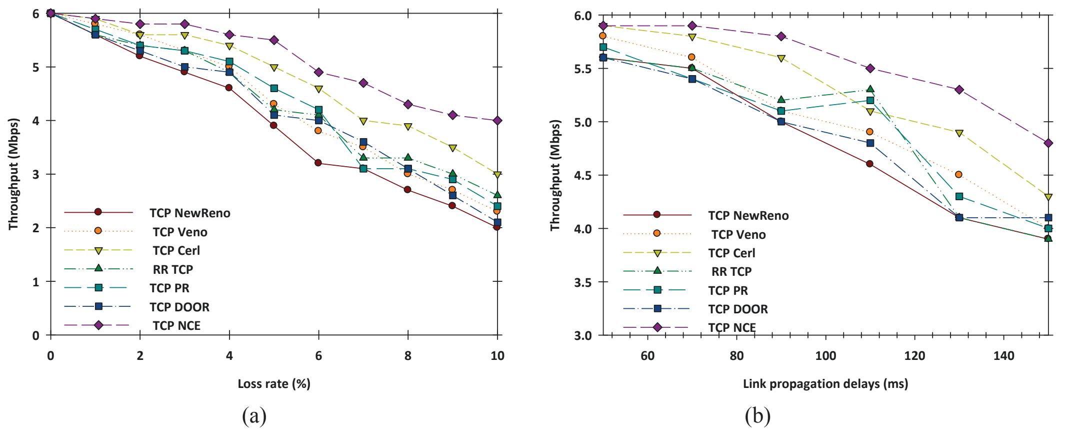


Figure 14 Typical variation in TCP throughput with packet loss rates and link propagation delay.

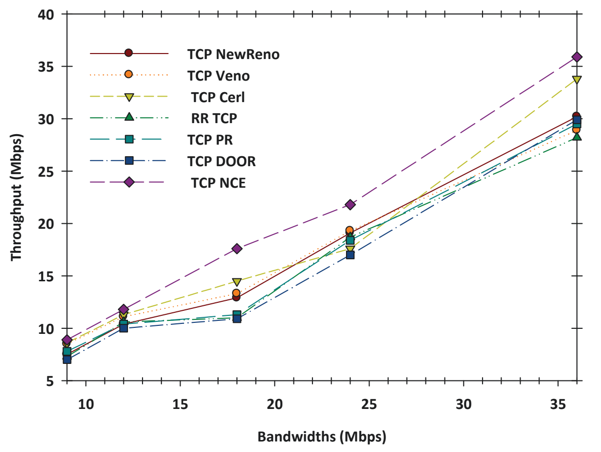


Figure 15 Typical variation in TCP throughput with different bandwidths.

simulation time. We imposed 3% random loss for this experiment. In this graph, it is evident that TCP NewReno is always trying to slow down the cwnd even in random packet loss. On the other hand, TCP NCE handled the random loss by increasing the size of congestion window. TCP NCE suffers only less time by reducing the cwnd size compared to TCP NewReno. Between 10 and 80s, TCP NewReno reduce cwnd more than 10 times. Opposed to TCP NewReno, TCP NCE reduced the size of cwnd only three times. The reason is TCP NCE can able to detect the random loss and is more effective in utilizing the bandwidth fully.

Throughput evaluation of TCP NCE under second condition

As we mentioned earlier, the second condition aims to cause the random loss and packet reordering according to the rate of delay, bandwidth, packet reordering, packet loss, and percentage of unneccssary retransmissions. For these experiments, we used multi-hop wireless networks contains six hops with five routers as shown in Figure 13b. Figure 17 shows the variation in throughput gain according to packet loss rate and propagation delays at six hops connection. The packet loss rate ranges from 1 to 5% with 1% packet reordering. For causing packet reordering we used the link-layer retransmission limit which is equal to three as specified in [28]. As a result, when high error rate link-layer retransmission cannot guarantee successful packet retransmission and thereby reorder the packets in the same flow. We run simulation with bandwidth 12 Mbps and link propagation delay 50 ms. As is evident from Figure 17a, TCP NCE and TCP CERL perform significantly better than other TCPs. The throughput of RR TCP and TCP PR is fluctuating according to the increase in the loss rate. When packet loss rate reaches at 5%, the throughput of TCP NCE has 92% greater than TCP PR and 85 % greater than TCP CERL.

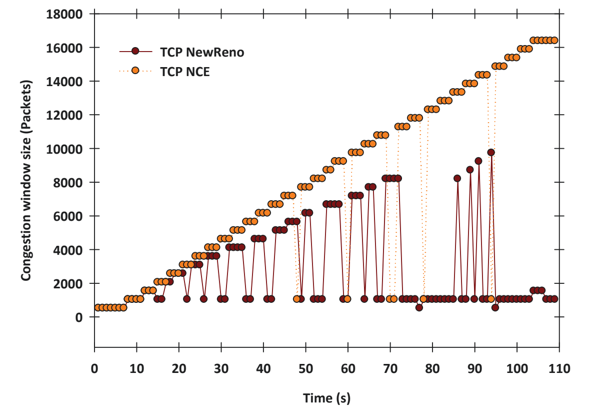


Figure 16 Typical congestion window size of TCP NewReno and TCP NCE.

Simulation results in Figure 17b shows that the performance of TCP NCE decreases with increase in propagation delays from 50 to 170 ms. For this experiment, we used 9 Mbps bandwidth, 2% random loss, and 4% packet reorder rate. When the delay increases, a large size of cwnd is needed to utilize the full bandwidth of the link. If we compared this results with Figure 17a, we can see that TCP PR and RR TCP outperfoms TCP CERL. Because when packet reorder occurs with less random loss, the solution for packet reordering such as RR TCP and TCP PR acheives higher throughput. However, even in the coexistence of random loss and packet reordering, TCP NCE achieves significant improvement in throughput compare to other TCPs by reducing the frequent reduction of the size of cwnd unnecessarily. As a result TCP NCE can send more packets and increase the throughput. Figure 18a depicts the throughput gain of TCP NCE under varying bandwidths ranges from 9 to 36 Mbps. The loss rate and reorder rate set to 5 and 3%, respectively. As shown in Figure 18a, when bandwidth increases, except TCP NCE the throughput of all other TCP's fluctuates lightly. When bandwidth reaches 36 Mbps, TCP CERL, TCP PR, TCP NewReno, TCP DOOR, and TCP Veno acheives only less than 30 Mbps throughput. Simulation results in Figure 18b shows the throughput gain of TCP NCE in presence varying reorder rate ranges from 1 to 5%. When reorder rate increases, the performance of RR TCP, TCP PR, and TCP DOOR becomes better than TCP Veno, TCP CERL, and TCP NewReno. However, still TCP NCE has higher throughput. The random loss differentiation algorithms such as TCP CERL and TCP Veno cannot perform well according to the different reorder rates because these solutions has no mechanism to detect the reorder packets and results in the degradation of

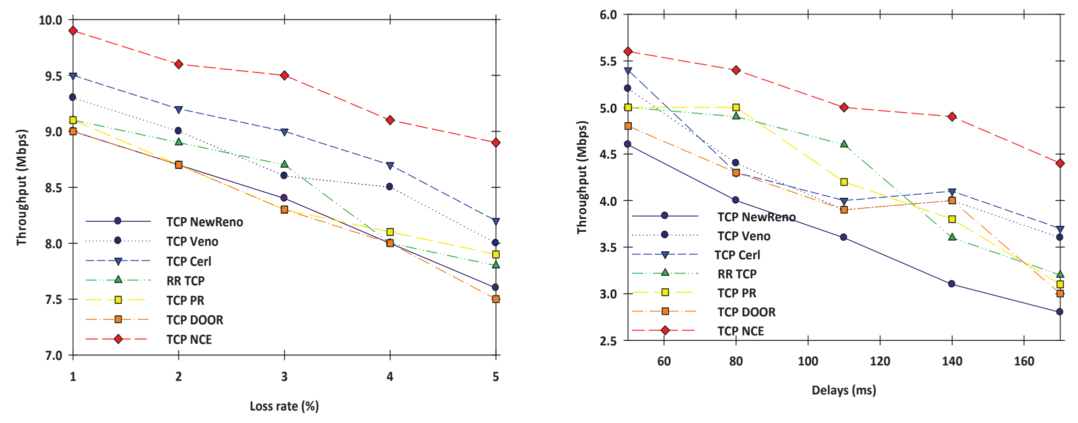


Figure 17 Typical TCP throughput according to loss rates and propagation delays.

throughput. On the other hand, TCP NCE can detect both the events compared to other TCPs.

In Figure 19, we analyzed the percentage of unnecessary retransmissions of various algorithms under varying reorder rate ranges from 1 to 5%. The unnecessary retransmissions rate, which is defined as the ratio of unnecessary retransmissions to the total number of packets transmitted. When the rate of reorder increases, the unncessary retransmissions of all TCP's increases. However, TCP NCE has less number of retransmissions compared to other TCP's due to the ability of detecting and differentiating the reorder packets. Compare to TCP Veno and TCP CERL, TCP PR, RR TCP, and TCP DOOR has less number retransmissions because of their capability to find the reorder packets. TCP NewReno has the worst performance. It has two times greater retransmissions than TCP NCE.

Throughput evaluation of TCP NCE under third condition

In third condition, we evaluated the throughput performance of TCP NCE by imposing congestion loss, random loss, and packet reordering in order to confirm that TCP NCE performs well in congestion and non-congestion events by using dumbell shaped wireless network as shown in Figure 13c. The experiments are based on different number of TCP connections, queue size, packet loss rate, reorder rate, unnecessary reduction of cwnd, and fast retransmissions. Figure 20a presents the typical throughput performance of TCP's under different number of TCP connections. In this experiment, all the senders send packets to destinations using more than one TCP connection ranges from 1 to 20 connections. Apart from that, we use 3% random loss and reorder rate with bandwidth 24 Mbps and

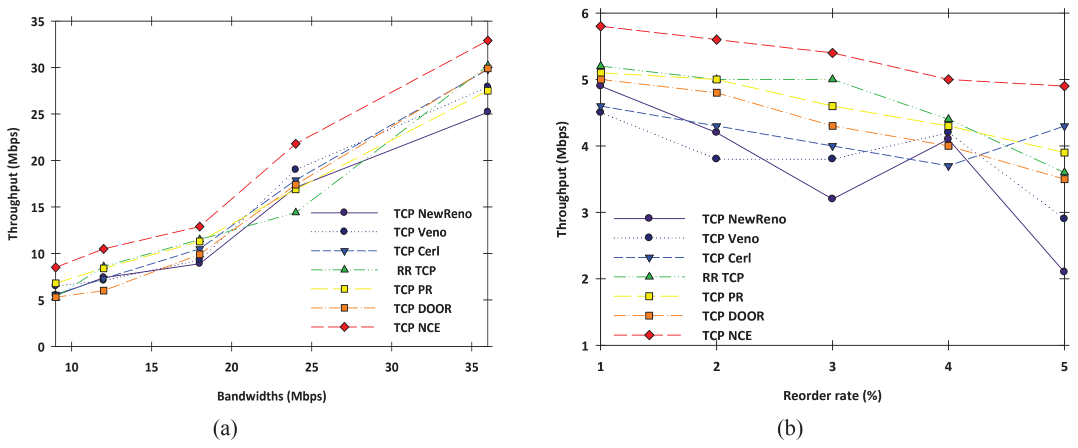


Figure 18 Typical TCP throughput according to various bandwidths and reorder rate.

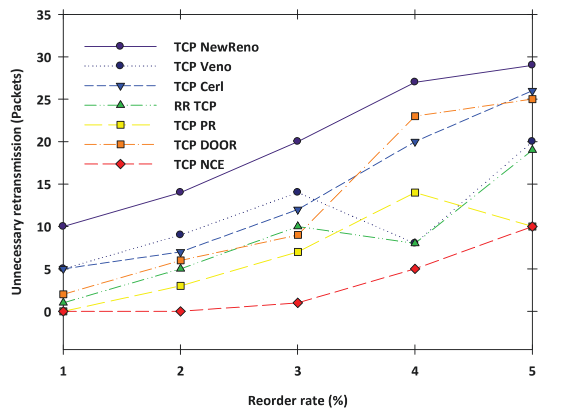


Figure 19 Comparison of unnecessary retransmissions vs reorder rate.

delay 50 ms. From the results of the graph, we can confirm that TCP NCE is indeed efficient in all types of network conditions such as the packet loss and packet reordering situations. When the number of TCP connections, the throughput of all TCP variants decreases. Even the throughput decreases, TCP NCE outperforms more than 70% from TCP CERL and more than 100% higher throughput than TCP NewReno. Figure 20b shows the result of throughput gain according to various queue size from 40 to 80K in bytes. In this graph, it is evident that the queue utilization of TCP NCE is much higher than that of other TCP variants and thus TCP NCE can achieve better throughput.

Figure 21a shows the comparison of TCPs under different bandwidth ranges from 12, 24, and 36 Mbps. In this experiment, we use five TCP connections from all senders to different destinations with 2% packet loss rate and 1% reorder rate with a delay of 80 ms. The throughput of all TCPs rise steadily according to the increase in bandwidth. However, TCP NCE has little more performance improvement compared to other TCPs.

The unnecessary reduction of the size of cwnd can be seen in Figure 21b. For this simulation, we use three TCP connections with 1% of packet reorder rate and the packet loss rate varies from 5, 7, and 9% due to network congestion and transmission errors. When the rate of packet loss increases, the unncessary reduction of cwnd also increases. This leads to decrease the throughput of TCPs. Among other TCPs, TCP NCE has less number of window reduction. Thus it can send more data and can increase the throughput. Figure 22 presents the unnecessary fast retransmissions of all TCP's according to the increase in loss rates which ranges from 5 to 10%. For doing this simulation, we use the same parameter settings of former comparison. The unncessary fast retransmission rate, which is defined as the ratio of unnecessary fast retransmission to the total number of packets transmitted. TCP NCE is superior to others by less number of fast retransmissions. TCP NCE can limit the retransmission by using the information from the network.

Accuracy of TCP NCE

Accuracy is another performance metric we used to evaluate the performance of TCP NCE. Because accuracy is one of the most important metric used in loss differentiation algorithms for evaluting the performance in addition to end-to-end throughput. We measured the accuracy of congestion loss (ACL), random loss (ARL), and packet reordering (APR) where,

$$\mathrm{ACL} = \left(\mathrm{NCL}/\mathrm{NCL}_{\mathrm{Total}}\right) * 100\,(\mathrm{ACL} \geq \mathrm{NCL} \geq 0, 100\% \geq \mathrm{ACL} \geq 0\%)$$

where NCL is the number of congestion packet loss exactly identified as congestion by TCP NCE compare

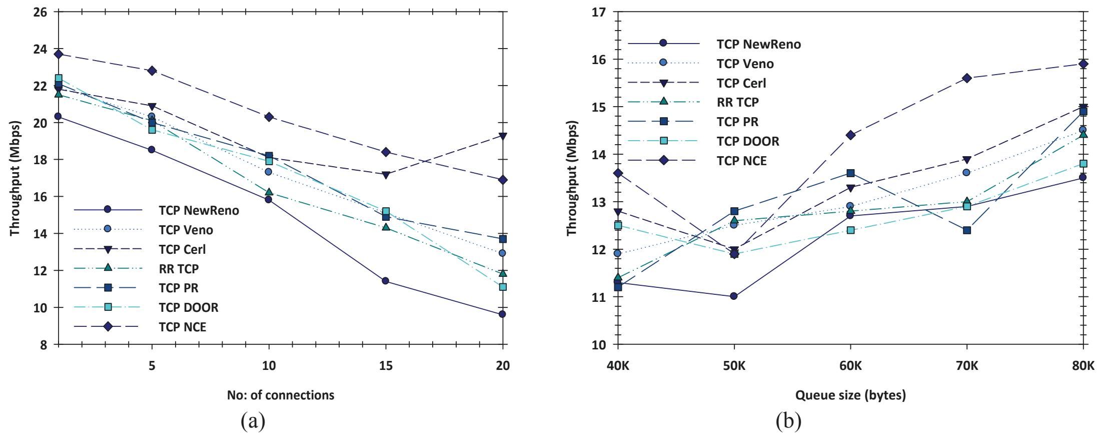
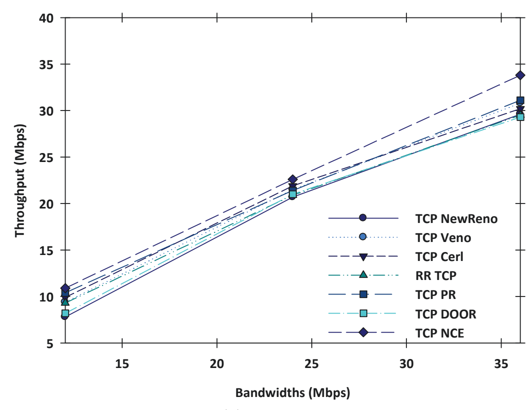


Figure 20 Typical TCP throughput according to different TCP connections and queue size.



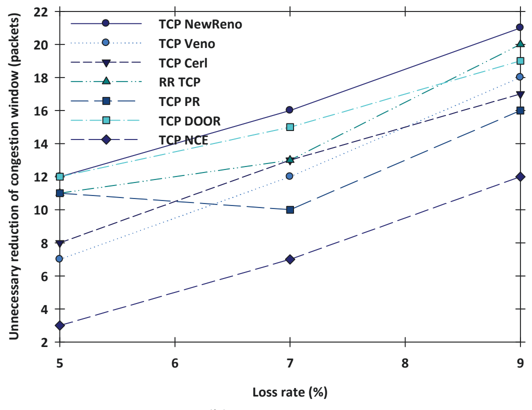


Figure 21 **Typical TCP throughput according to various bandwidths and comparison of unnecessary reduction of congestion window size vs loss rate**.

to other algorithms, and NCL_{Total} is the number of packet loss caused by network congestion.

$$ARL = \left(NRL/NRL_{Total}\right) * 100$$

$$(ARL \geq NRL \geq 0, 100\% \geq ARL \geq 0\%)$$

where NRL is the number of random packet loss exactly identified as random loss by TCP NCE compare to other algorithms, and NRL_{Total} is the number of packet loss caused by transmission errors.

$$APR = \left(NRP/NRP_{Total}\right) * 100$$

$$(APR \geq NRP \geq 0, 100\% \geq APR \geq 0\%)$$

where NRP is the number of reordered packet exactly identified as packet reordering by TCP NCE compare to other algorithms, and NRP_{Total} is the total number of reordered packet. Figure 23 presents the simulation results for checking the accuracy of TCP NCE in terms of congestion based packet loss with different TCP connections using dumbell shaped wireless network topology. We set 1% random loss and packet reordering in order to check the accuracy for the detection of congestion loss. From the graph, it is clear that TCP NCE gains more than 90% accuracy compared to other TCP's. The reason is, TCP NCE can utilize the maximum buffer space and this leads to reduce the missclassification of congestion and non-congestion events.

Figure 24shows the accuracy of random loss by varying packet loss rates which ranges from 1 to 5%. TCP NCE, TCP CERL, and TCP Veno has the highest accuracy compared to RR TCP, TCP PR, and TCP DOOR. The reason is these algorithms has no mechanism to detect random loss. The accuracy of TCP NCE caused by packet reordering is depicted in Figure 25. In this figure, TCP CERL and TCP Veno has worst performance due to the lack of mechanism for detecting the

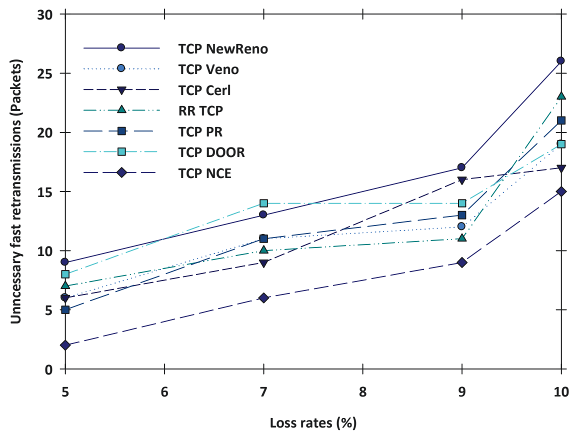


Figure 22 **Comparison of unnecessary retransmission vs loss rate.**

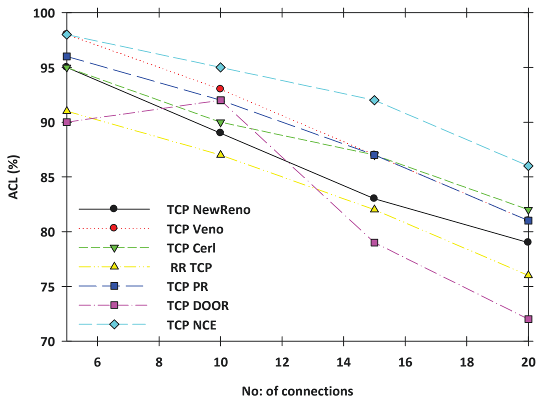


Figure 23 **Accuracy of packet loss due to congestion**.

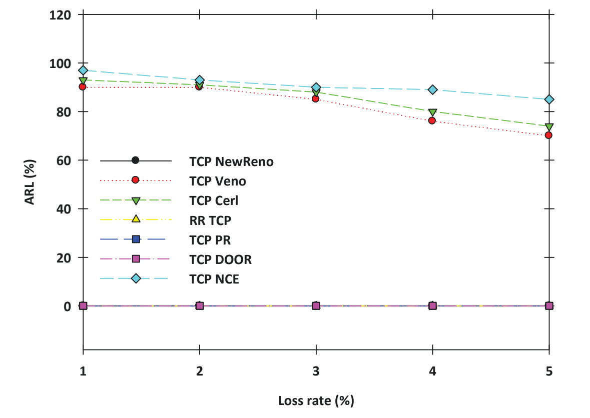


Figure 24 Accuracy of packet loss due to transmission errors.

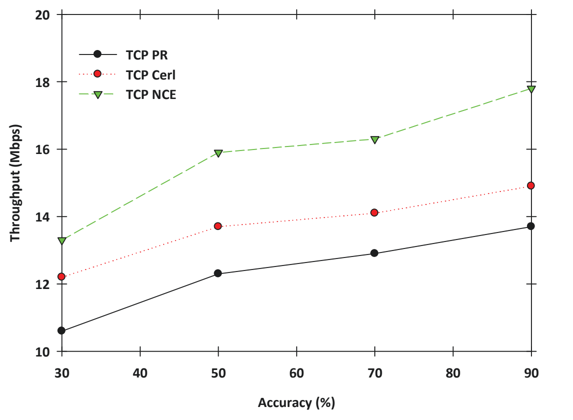


Figure 26 Acuracy vs. throughput.

reordering events. However, TCP PR, RR TCP, and TCP DOOR achieves higher performance when compare to TCP Veno and TCP CERL. TCP NCE is the superior one among all. Figure 26presents the result of throughput measurement under varying percentage of accuracies. From the figure, we observed that when accuracy increases, the throughput performance of TCP also increases. Compared to TCP CERL and TCP PR, TCP NCE acheives higher throughput when the accuracy increases.

Fairness of TCP NCE

Fairness is a measure of the relative throughput performance of a set of TCP flows of the same type. To investigate the fairness performance, 10 simultaneous TCP flows of the same type are run using the dumbell shaped wireless network topology consisting of 10 TCP sender and receivers with two bottleneck links L1 and L2 connected with three routers R1, R2, and R3 as shown in Figure 27.

We set 9 Mbps bandwidth with 80 ms link porpagation delay. The throughput of each flow is measured and calculated the fairness index using the well known Jain fairness index [29]. Fairness index $f(x)$ is a function of the variability of the throughput across the TCP flows and can be defined as,

$$F(x_1, \ldots, x_N) = \left(\sum_{i=1}^{N} x_i\right)^2 / N \times \sum_{i=1}^{N} (x_i)^2$$

where x_i is equal to the observed throughput of the ith flow ($0 < i \leq N$) normalized to the total achievable throughput in the link and N is equal to total number of flows sharing the link. Figure 28 shows the fairness of TCP NCE compared to other TCPs under varying packet loss rate ranges from 1 to 5%.

As expected, TCP NCE is more fairer than other algorithms such as TCP NewReno, TCP Veno, TCP CERL, RR TCP, TCP PR, and TCP DOOR due to its capability for detecting and differentiating the non-congestion events along with congestion and can utilize the bandwidth fully.

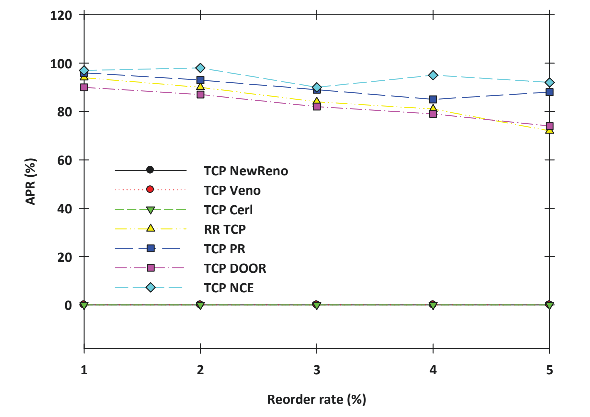


Figure 25 Accuracy of packet reordering.

Conclusion

In this article, for improving the performance of TCP over wireless networks, we proposed a new unified solution called TCP NCE, which is capable to reduce the unnecessary retransmissions and cwnd reductions by detecting, differentiating and reacting to non-congestion events such as random losses and packet reordering in addition to network congestion losses. For detecting the congestion from non-congestion events we used the queue length of the bottleneck link by measuring RTT

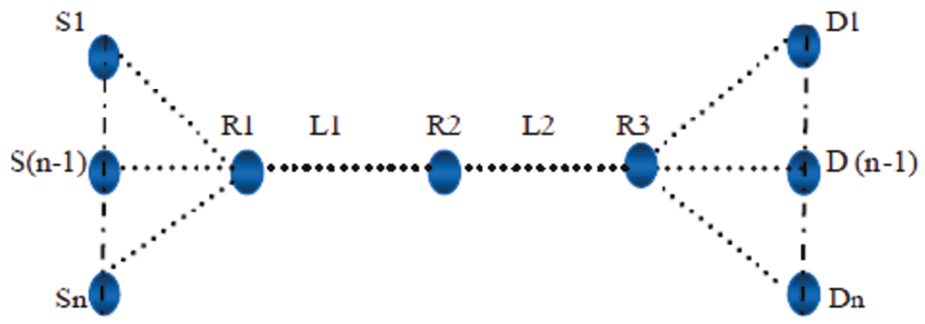


Figure 27 Dumbell shaped fully wireless network with two bottleneck links.

using TCP timestamp and compared with a threshold value. For differentiating the non-congestion events, we used the outstanding packets in the network when the sender receives three dupacks. Furthermore, we introduced a new TCP retransmission algorithm called 'Retransmission Delay' which guides the TCP sender at the time of detecting the non-congestion event via three duplicate acknowledgments by delaying the retransmission upto the expiration of dynamic delay threshold value. We evaluated the performance of TCP NCE in terms of throughput, accuracy and fairness over four different network topologies using qualnet 4.5. The simulation results have confirmed that TCP NCE has a significant improvement over existing variants such as TCP NewReno, TCP Veno, TCP CERL, RR TCP, TCP PR, and TCP DOOR. Three salient features of TCP NCE contribute to this improvement. First, detection of congestion and non-congestion events under different network conditions. Second, differentiation of these events and finally, the reaction of TCP sender when they detect congestion and non-congestion events.

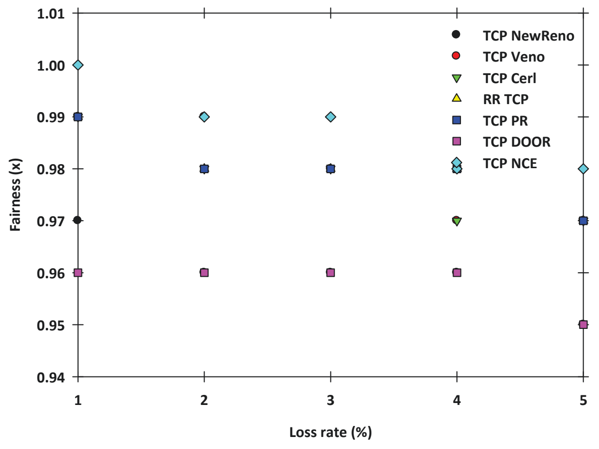


Figure 28 Fairness of TCP NCE towards other TCP algorithms.

These three features of TCP NCE helps the sender to reduce the size of cwnd unnecessarily and avoid spurious retransmissions and thereby increase the performance of TCP over wireless networks.

List of Abbreviations

ACL: accuracy of congestion loss; ARL: accuracy of random loss; APR: accuracy of packet reordering; BS: buffer size; CA: Congestion Avoidance; CERL: Congestion Control Enhancement for Random Loss; cwnd: congestion window; dupacks: duplicate acknowledgments; NCE-Detection: Detection of non-congestion events; NCE-Differentiation: Differentiation of non-congestion events; NCE-Reaction: Reaction to non-congestion events; SS: Slow Start; TABE: timestamp based available bandwidth estimation; TCP: Transmission Control Protocol; TCP NCE: TCP for Non-Congestion Events; TCP PR: TCP Persistent packet reordering; TCP DOOR: Detection of out-of-order and response.

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Competing interests

The authors declare that they have no competing interests.

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References

1. KC Leung, VOK Li, Transmission control protocol (TCP) in wireless networks: issues, approaches, and challenges. Commun. Surveys Tutorials IEEE. **8**(4), 64–79 (2006)
2. M Przybylski, B Belter, A Binczewski, Shall we worry about packet reordering? *in Proceedings of TERENA Networking Conference*, Poznan, Poland, 28–36, June 2005
3. A Sathiaseelan, T Radzik, Reorder Notifying TCP (RN-TCP) with explicit packet drop notification. Int J Commun Syst. **19**(6), 659–678 (2006)
4. K Xu, Y Tian, N Ansari, Improving TCP performance in integrated wireless communications networks. Comput Netw. **47**, 219–237 (2005)
5. CP Fu, SC Liew, TCP Veno, TCP enhancement for transmission over wireless access networks. IEEE J Select Areas Commun. **21**(2), 216–228 (2003)
6. H El-Ocla, TCP CERL: congestion control enhancement over wireless networks. J Wireless Netw. **16**(1), 183–198 (2010)
7. M Zhang, B Karp, S Floyd, L Peterson, RR-TCP: a reordering-robust TCP with DSACK, in *Proceedings of IEEE International Conference on Network Protocols (ICNP '03)*, 95–106, November 2003
8. S Bohacek, JP Hespanha, J Lee, C Lim, K Obraczka, A new TCP for persistent packet reordering. IEEE/ACM Trans Netw. **14**(2), 369–382 (2006)

9. F Wang, Y Zhang, Improving TCP performance over mobile ad-hoc networks with out-of-order detection and response. in *Proceedings of ACM MOBIHOC 2002*, Lausanne, Switzerland, **9-11**, 217–225, June 2002
10. KC Leung, V Li, D Yang, An overview of packet reordering in transmission control protocol (TCP): problems, solutions, and challenges. Parallel Distrib Syst IEEE Trans. **18**(4), 522–535 (2007)
11. Scalable Networks, http://www.scalable-networks.com/index.php
12. LP Tung, WK Shih, TCP throughput enhancement over wireless mesh network. IEEE Commun Mag. (2007)
13. LS Brakmo, S O'Malley, LL Peterson, TCP Vegas: New techniques for congestion detection and avoidance. Comput Commun Rev. **24**(4), 24–35 (1994)
14. K Xu, Y Tian, N Ansari, TCP-Jersey for wireless IP communications. IEEE J Select Areas Commun. **22**, 747–756 (2004)
15. S Floyd, T Henderson, The new Reno modification to TCP'S fast recovery algorithm. RFC 2582 (1999)
16. E Blanton, M Allman, On making TCP more robust to packet reordering. ACM SIGCOMM Comput Commun Rev. **32**(1), 20–30 (2002)
17. M Allman, H Balakrishnan, S Floyd, Enhancing TCP's loss recovery using limited transmit. IETF RFC, Network Working Group, January 2001
18. A Medina, M Allman, S Floyd, Measuring the Evolution of transport protocols in the internet. ACM SIGCOMM Comput Commun. **35**, 37–52 (2005)
19. V Jacobson, R Braden, D Borman, TCP extensions for high performance. RFC 1323 (1992)
20. J Feng, Z Quyang, L Xu, B Ramamurthy, Packet Reordering in high-speed networks and its impact on high-speed TCP variants. Comput Commun. **32**, 62–68 (2009)
21. T Reddy, A Ahammed, R Banu, Performance comparison of active queue management techniques. IJCSNS Int J Comput Sci Netw Security. **9**(2), 405–408 (2009)
22. MY Park, SH Chung, S Prasanthi, End-to-end loss differentiation algorithm based on estimation of queue usage in multi-hop wireless networks. IEICE Trans Inf Syst. **E92-D**(10), 2082–2093 (2009)
23. CH Lim, JW Jang, Robust end-to-end loss differentiation scheme for transport control protocol over wired/wireless networks. IET Commun. **2**, 284–291 (2008)
24. S Cen, PC Cosman, GM Voelker, End-to-end differentiation of congestion and wireless losses. IEEE/ACM Trans Netw. **11**(5), 703–717 (2003)
25. J Feng, Z Quyang, L Xu, B Ramamurthy, Packet reordering in high-speed networks and its impact on high-speed TCP variants. Comput Commun. **32**, 62–68 (2009)
26. Y Wang, G Lu, X Li, A study of internet packet reordering, in *Proceedings of Information Networking Technologies for Broadband and Mobile Networks International Conference, ICOIN 2004*, Busan, Korea, 18–20, February 2004
27. R De Oliveira, T Braun, A smart TCP acknowledgment approach for multihop wireless networks. Mobile Comput IEEE Trans. **6**(2), 192–205 (2007)
28. D Yang, KC Leung, VOK Li, Simulation-based comparisons of solutions for TCP packet reordering in wireless networks, in *Wireless Communications and Networking Conference, 2007. WCNC 2007. IEEE*, 11-15, 3238–3243, March 2007
29. R Jain, D Chiu, W Hawe, A quantitative measure of fairness and discrimination for resource allocation in shared computer systems. Research Report TR-301. (1984)

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