TCP Packet Control for Wireless Networks

Wan G. Zeng and Ljiljana Trajković

Abstract— **In this paper, we propose packet control algorithms to be deployed in intermediate network routers. They improve TCP performance in wireless networks with packet delay variations and long sudden packet delays. The ns-2 simulation results show that the proposed algorithms reduce the adverse effect of spurious fast retransmits and timeouts and greatly improve the goodput compared to the performance of TCP Reno. The TCP goodput was improved by ~30% in wireless networks with 1% packet loss. TCP performance was also improved in cases of long sudden delays. These improvements highly depend on the wireless link characteristics.**

Index Terms—**TCP, packet control, wireless networks, packet delay, packet delay variation.**

I. Introduction

The performance of Transmission Control Protocol (TCP) [1], [2] has greatly improved since 1988, when the congestion avoidance and control algorithms [3] were first introduced. TCP is currently the most widely used Internet transport protocol. In 2002, TCP traffic accounted for 95% of the IP network traffic [4]. This was due to a variety of popular Internet applications and protocols. Web (HTTP), file transfer (FTP), and e-mail (SMTP) rely on TCP as the underlying transport protocol. Internet applications that rely on TCP today are likely to do so in the future. With a growing deployment of wireless networks, it is important to support these applications in both wireline and wireless environments. Hence, wireless networks will also require good TCP performance.

Wireless networks have different characteristics compared to wireline networks. TCP, which was carefully designed and tuned to perform well in wireline networks, suffers performance degradation when deployed in wireless networks.

II. Transmission Control Protocol

TCP is a connection-oriented transport layer protocol. It provides reliable byte stream services for data applications. Its key features include reliability, flow control, connection management, and congestion control. Major TCP versions are Tahoe [2], Reno [5], and NewReno [6]. They differ mainly in their congestion control algorithms. Tahoe, the original version of TCP, employs three congestion control algorithms: slow start, congestion avoidance, and fast retransmit. TCP Reno extends Tahoe with a fast recovery mechanism. NewReno, the latest major version of TCP, modifies TCP Reno's fast recovery algorithm and addresses the issue of partial acknowledgements (ACKs) [6].

Differences between the characteristics of wireline and wireless networks have significant impact on TCP performance. TCP was designed and optimized to perform well in wireline networks. Wireless links, with considerable packet losses due to link errors, delay variations, and long sudden delays, violate TCP's essential design assumptions. Improving TCP performance in wireless networks has been an ongoing research activity since the mid 90's. Most improvements dealt with TCP's reaction to high bit error rate (BER) and TCP performance degradation due to delay and delay variation in wireless links. Performance of TCP's congestion control algorithms particularly deteriorates when TCP is deployed in mixed wireline/wireless networks. We describe here TCP's timer and window management, congestion control algorithms, and round-trip time (RTT) estimation.

A. TCP Windows

TCP maintains two windows to perform congestion control and avoidance: the receiver's advertised window (*rwnd*) and the congestion window (*cwnd*). They define the maximum number of bytes the receiver may receive and the sender may send, respectively. The number of bytes that may be sent to the network is the minimum of the two. With *rwnd* sufficiently large, the larger the *cwnd*, the more data TCP can send, resulting in larger TCP throughput.

The growth of the *cwnd* is ACK paced: with every segment that TCP sends, the receiver issues an ACK to acknowledge the receipt of the data. The receipt of the ACKs increases the *cwnd* and enables the sender to send more data.

B. TCP Congestion Control Algorithms

TCP packets may be lost due to link errors or network congestion. Since losses due to link errors in wireline networks are rare, TCP deals only with packet loss due to network congestion. Hence, packet loss always implies network congestion. TCP congestion avoidance and control [3] were first introduced when Internet experienced its first series of "congestion collapses."

TCP detects network congestion via duplicate ACKs and timeouts. Each byte of the transmitted data is assigned a unique sequence number (*seqno*). When a data packet loss occurs, TCP receiver issues a duplicate ACK for any out-of-sequence data packet received. Upon receiving a predefined number of consecutive duplicate ACKs, TCP assumes that a packet is lost.

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In most TCP implementations, the threshold is set to three (known as three duplicate ACKs). Note, however, that when *cwnd* < 4 or the network is temporarily disconnected, the number of duplicate ACKs is less than three, and thus insufficient to trigger three duplicate ACKs. TCP handles this situation by keeping a timer called Retransmission Timeout (RTO). When the timer expires, it assumes packet loss [1], which triggers congestion control. TCP congestion control mechanism includes [7]:

- increasing *cwnd* by one segment size per RTT and halving *cwnd* for every window experiencing a packet loss (Additive Increase Multiplicative Decrease, AIMD)
- Retransmission Timeout (RTO), including exponential back-off when timeout occurs
- slow start mechanism for initial probing of the available bandwidth
- ACK clocking (self-clocking) the arrival of ACKs at the sender, used to trigger transmission of new data.

TCP Reno congestion control algorithms are shown in Fig. 1 [5].

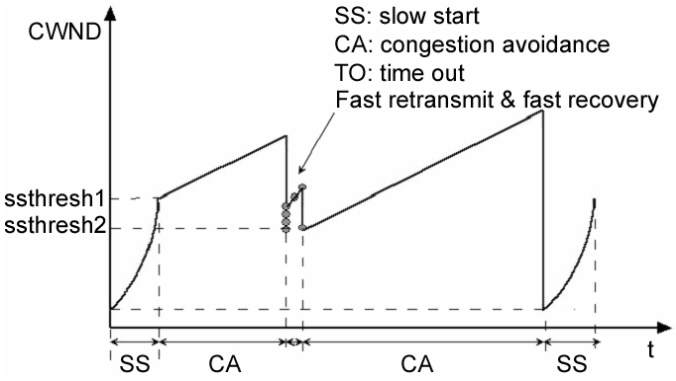

Fig. 1. TCP congestion control algorithms.

Slow Start: At the onset of a TCP connection, TCP employs the slow start mechanism to probe the network capacity. Slow start is also employed after a packet loss is detected by the RTO mechanism. When the transmission starts, the sender's *cwnd* is set to the initial window (IW) size. Congestion window *cwnd* is increased by at most SMSS (sender maximum segment size) bytes for each ACK received that acknowledges new data. The slow start threshold (*ssthresh*) may be arbitrarily high and could be reduced when congestion occurs. When congestion is detected by the RTO mechanism, *cwnd* is set to IW and *ssthresh* is set to 0.5×*cwnd*. In both situations, slow start is used as long as *cwnd* < *ssthresh*. Slow start ends when *cwnd* > *ssthresh* or when congestion is detected. When *cwnd* = *ssthresh*, the sender may use either slow start or congestion avoidance.

Congestion Avoidance: If *cwnd* > *ssthresh,* congestion avoidance is employed to probe the network capacity more slowly than during the slow start. Congestion window *cwnd* is incremented by one full-size segment per RTT. In most cases, TCP operates in the congestion avoidance phase. Congestion avoidance ends only when congestion is detected.

Fast Retransmit: When three duplicate ACKs are detected, TCP moves from congestion avoidance to fast retransmit. The incoming segments are considered out-of-order by the receiver when a packet loss occurs. For any out-of-order packet received, the receiver immediately sends a duplicate ACK acknowledging the next expected *seqno*. After receiving three duplicate ACKs, the sender retransmits what appears to be the lost packet without waiting for the retransmission timer to expire. It uses the sequence number contained in the duplicate ACKs. Along with the retransmission, TCP also sets *ssthresh* to

$$ssthresh = \max\left(\frac{FlightSize}{2}, 2\times SMSS\right),$$

where *FlightSize* is the size of the outstanding data in the network.

Fast Recovery: Fast recovery takes place immediately after the sender performs fast retransmit. Here, a new ACK is defined as the ACK acknowledging the sequence number beyond the lost segment. TCP first inflates *cwnd* to $ssthresh + 3\times SMSS$. This reflects the three segments that have left the network (three duplicate ACKs would require three packets to leave the network). For every additional duplicate ACK received, the sender increments *cwnd* by SMSS to reflect that an additional segment has left the network. This new *cwnd* may also allow the sender to transmit a new segment. When a new ACK is received, the sender sets *cwnd* to *ssthresh* to deflate the *cwnd*, and the congestion avoidance phase continues.

C. Karn's algorithm: RTT estimation and RTO

After a segment is transmitted, an ACK is expected by the sender. If the RTO timer expires before the ACK is received, the segment is retransmitted. This resynchronizes the transmission in case the segment is lost. Therefore, if the calculated RTO is too large, unnecessary time will be spent waiting for the timer to expire. Thus, it will cause TCP performance degradation [1]. If the calculated RTO is too small, the timer may expire prematurely and cause unnecessary retransmissions.

RTT is estimated using Karn's algorithm. RTO is calculated based on the estimated RTT and the RTT deviation. TCP measures the round-trip time of the ACKs for data segments and this interval is called sample RTT. The moving average of RTT, called a smoothed RTT (*srtt*), and the mean deviation (*rttvar*) are calculated as:

$$srtt = (1 - g)\times srtt + g\times sampleRTT$$
$$rttvar = (1 - h)\times rttvar - h\times|sampleRTT - srtt|,$$

with recommended parameter values:

$$g = 0.125 \text{ and } h = 0.25.$$

RTO is calculated as:

$$RTO = srtt + 4\times rttvar.$$

III. Characteristics of Wireless Networks

Mobile connectivity provided by wireless networks allows users to access information anytime and anywhere. The growth of cellular telephone systems is accompanied with a growing number of wireless-enabled laptops and personal digital assistants (PDAs). Cellular networks evolved from 1G analog systems to 2G systems (GSM and PDC), 2.5G systems (GPRS and PDC-P), and 3G systems (Wideband CDMA and cdma2000). During the past decade, the quality of wireless links has been improved in terms of BER and link bandwidth.

Wireless networks still exhibit the following characteristics:

A. High bit error rate (BER)

Wireless networks experience random losses. BER in wireless networks is significantly higher than in wireline networks. Packet error rates range from 1% in microcell wireless networks up to 10% in macrocell networks [4]. Even with optimized link layer retransmission algorithms in 3G networks, packet error rate remains ~1%.

B. Long and varying delay

Wireless links have a large latency. Typical RTTs in 2.5G and 3G networks vary from a few hundred milliseconds to one second. Furthermore, they are likely to experience sudden delay changes (delay spikes) greatly exceeding the typical RTT [8]. (Delay spike is defined as a sudden increase in the latency of a communication path [8].) Wireless WANs have a typical latency of up to 1 s. [9]. These delay changes may cause spurious TCP timeouts. Wireless links experience delay changes due to link recovery, temporary disconnections, traffic priority, and link/MAC layer protocols [9].

C. Bandwidth

Bandwidth of cellular networks increased as they evolved from 1G analog systems to 2G systems (10–20 kbps for uplink and downlink), to 2.5G (10–20 kbps uplink and 10–40 kbps downlink), and 3G systems (up to 64 kbps uplink and 384 kbps downlink) [8]. Data rates vary due to mobility and the interference from other users [8]. Mobile users share the bandwidth within a cell. As users move among cells, they affect the bandwidth available to other users. Furthermore, a user may move to another cell with higher or lower bandwidth. These factors cause variable wireless link data rates. TCP was designed to handle the changes in bandwidth with its self-clocking scheme. However, a sudden increase in RTT could still cause spurious timeouts.

D. Path asymmetry

Cellular 2.5G and 3G systems employ asymmetric uplink and downlink data rates.

IV. Improving TCP Performance

A number of solutions have been proposed to solve the problem of non-congestion related packet losses misinterpreted by TCP [10] – [12] and to reduce the impact of delays and delay variations on TCP performance in wireless networks [4], [9], [13] – [15].

A. Wireless Link Errors

The main characteristic of a wireless network is the high BER on its links. It violates the fundamental assumption of TCP that packet loss caused by link error is negligible (<< 1%) [3] and that packet loss is caused only by network congestion. High BER in wireless networks causes packet loss regardless of network congestion. The main cause for TCP's performance degradation in a mixed wireless/wireline environment is its inability to detect the origin of the packet loss.

When a packet loss is detected, TCP employs congestion control algorithms to reduce the transmission rate. A single packet loss on the link will cause duplicate ACKs and *cwnd* to be reduced by half according to the fast retransmit and fast recovery algorithms. TCP resolves the congestion in the network by lowering its transmission rate. However, lowering the transmission rate will degrade TCP performance if the packet loss is not caused by congestion.

One approach to improving TCP performance is to reduce the adverse effect of wireless link errors. Proposed solutions either hide the wireless link error from the TCP sender or make the sender aware of the causes of segment losses. The first approach resolves the error within the wireless domain without the TCP sender being aware of the error. These solutions often modify the base station and/or the mobile host. If the link error is well shielded from the sender, modifying the sender is not necessary. The examples are I-TCP [10], M-TCP [16], and Snoop [12], [17]. The second approach explicitly makes the sender aware of the wireless link error by handling differently segment losses caused by wireless link errors and losses due to network congestion. This approach requires the base station to send explicit congestion messages to the sender or a mechanism to detect the causes of loss at the sender. An example is TCP Westwood [18].

Based on the design principles [4], the solutions may also be categorized as: split connection (I-TCP [10]), link layer retransmission (Snoop [12]), and end-to-end (TCP Westwood [18], WTCP [19]).

B. Wireless Link Delays

Wireless networks have larger latency and delay variations than wireline networks. Long sudden delays during data transfers are common in GPRS wireless WANs [9], [13]. Furthermore, experimental [13] and analytical [15] data indicate that mobility increases packet delay and delay variation and degrades the throughput of TCP connections in wireless environments. Three major adverse effects are: spurious fast retransmit, spurious timeouts, and ACK compression.

Spurious fast retransmit: Its primary source is the link delay. It can also be caused by a spurious timeout. TCP generates a duplicate ACK whenever an out-of-order data segment is received. The number of out-of-order segments that had arrived consecutively prior to this segment is called the re-ordering length [13]. Thus, the re-ordering length represents the number of duplicate ACKs expected to arrive at the TCP sender. Fig. 2 shows no segment loss or network congestion. Nevertheless, fast retransmit is triggered because TCP misinterprets the duplicate ACKs as packet loss and a sign of network congestion. This event is called spurious fast retransmit.

TCP sender halves its congestion window to reduce the transmission rate in response to fast retransmits. As illustrated in Fig. 2, a packet was held in a queue by the *hiccup* (a delay generator [13]) at time 37.7 s (marked +) and then retransmitted after six segments at time 41.9 s (marked ⟶). Upon receiving the six segments prior to receiving the queued segment, the receiver generates six duplicate ACKs, triggering a fast

retransmit.

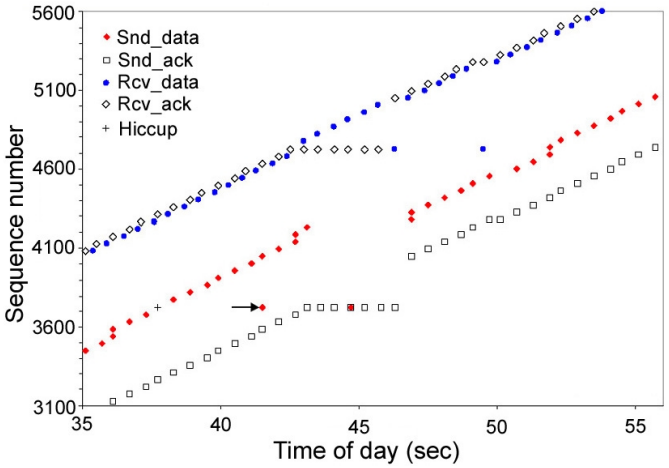

Fig. 2. The effect of packet re-ordering [13].

Spurious timeout: It may occur on links with long sudden delays. With its RTO timer, TCP is designed to handle even large gradual changes in delays. Nevertheless, TCP cannot handle well long sudden delays because it is unable to adjust its RTO fast enough. When the RTO timer expires, TCP assumes that the outstanding packets are lost and triggers the congestion control.

Spurious timeout is illustrated in Fig. 3. The three arrows show three critical events. The sudden long delay on the link occurs at 5 s. The first arrow indicates the moment when the TCP sender's RTO timer expires. TCP sender assumes that the previously sent packets are lost. The *cwnd* is reduced to the initial window (two segments). TCP then retransmits the first two unacknowledged segments. At 11 s, the link delay terminates (marked by the second arrow). The sender receives the first new ACK and starts recovering from timeouts by entering the slow start phase. All the unacknowledged segments are to be retransmitted. Since some ACKs on the wireless link have also been delayed, they accumulate and arrive together at the sender when the link recovers. This causes a burst of data segments to be sent. This is known as ACK compression. The retransmission unnecessarily utilizes the scarce wireless bandwidth and may potentially increase the recovery time.

The unnecessary retransmission of segments may introduce an additional spurious fast retransmit. At 11.97 s, the retransmitted segments arrive at the receiver. Since previously transmitted segments have been received after the link recovered, TCP receiver generates a duplicate ACK for every out-of-order segment. These duplicate ACKs (ACK 136) are shown between 11.97 s and 12.54 s. When the number of duplicate ACKs exceeds the duplicate ACK threshold, another spurious fast retransmit is triggered. This further worsens the situation. A gap appears after 12.54 s (graph labeled *seqno*) immediately after ACK 137 is received. During the fast retransmit, for every duplicate ACK received, the sender artificially inflates the *cwnd* by one segment and, if the *cwnd* permits, transmits the next segment. (The change in *cwnd* is not shown. It can be seen from the *seqno* showing new segments that are sent with ACKs received.) When the new ACK 137 is received (marked by the third arrow), the fast retransmit is terminated and *cwnd* is deflated back to the size that it had at 11.97 s. No segments are transmitted during the period between 12.50 s and 13.10 s (graph labeled *seqno*) due to this decrease of *cwnd*.

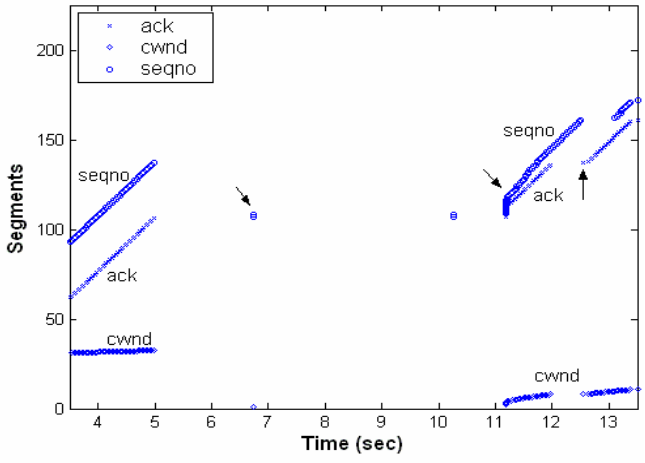

Fig. 3. Spurious timeout.

Eifel algorithm [13] was proposed to enhance TCP's adaptation to link delays in wireless networks. Both spurious timeout and spurious fast retransmit are caused by TCP's retransmission ambiguity, which occurs when an ACK arrives for a segment that has been retransmitted. Hence, there is no indication which transmission is being acknowledged [20]. Eifel algorithm is an end-to-end solution, which requires modifying only the TCP sender. It first eliminates the retransmission ambiguity by using additional information in the ACKs. Then, it restores the payload and resumes transmission with the next unsent segment [13]. Timestamp option is used to provide the additional information to identify the segment that triggered the duplicate ACK. Timestamp clock is stored in the header of every outgoing segment and echoed back with its corresponding ACK. The sender also keeps track of the timestamp of the first retransmission. The received ACK can be identified by comparing the timestamp stored in the sender with the timestamp in the received ACK. If the ACK was triggered by the original segment, spurious retransmission has occurred. The sender then restores the *cwnd* and possibly RTO. Instead of retransmitting the unacknowledged segments, the next unsent segment is transmitted.

Although Eifel algorithm effectively reduces the impact of spurious timeouts and spurious fast retransmits by eliminating the retransmission ambiguity, it has two major drawbacks: it requires modification of all TCP clients in the wireline domain and requires that both the sender and the receiver have the 12-byte TCP timestamp option enabled in every segment and the corresponding ACKs. Furthermore, its performance in the cases of high link errors is questionable [14].

V. Proposed TCP With Packet Control

We propose a set of packet control algorithms designed to avoid the adverse effect of long delays and delay variations on

TCP performance in wireless networks. We describe the algorithms, their implementation, and evaluate their performance using the ns-2 simulator [21].

A. Network Architecture

Network architecture, shown in Fig. 4, represents a cellular network or a wireless LAN (WLAN). A mobile host (MH) initiates a TCP connection with a fixed host (FH) through a base station (BS), which is an edge node in the wireless network. TCP packets are sent from the FH to the MH through the BS and MH acknowledges every data packet received [15]. TCP data may be either a long lived FTP connection with a large volume of data traffic or a short lived HTTP connection with a typically smaller volume of data traffic. We assume that the condition of the wireless link may change with time (leading to variable wireless link delay), that the mobile device roams between cells, and that mobile applications have limited data bandwidth.

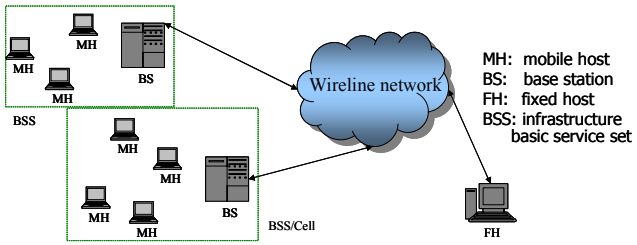

Fig. 4. Network architecture.

B. TCP with Packet Control

TCP with packet control consists of ACK and data packet filters. The two filters improve TCP performance in mixed wireline/wireless networks and maintain TCP's end-to-end semantics. They deal with wireless links with long sudden delays and delay variations, handle handoffs, and maintain regular TCP functions. They do not depend on end-user TCP flavors.

The filters are to be deployed at the wireless network edge (typically the BS). This is a TCP-aware link layer solution. The algorithms keep track of TCP data and ACK packets received from the FH and the MH, respectively. Packet control filters forward packets to both client ends based on the information gathered in the BS.

1. ACK Filter: Packet control reacts to ACKs received from the MH using the ACK filter. It drops the old ACKs and duplicate ACKs classified according to the duplicate ACK threshold defined by the user. It remembers the last new ACK received from the wireless receiver, called the last received ACK. When an ACK arrives, its ACK number is checked against the last received ACK. We consider three cases:

Old ACK: The ACK is considered old if the ACK number has already been received and/or is smaller than the last received ACK. It is immediately dropped.

Duplicate ACK: If the newly received ACK number is identical to the largest ACK currently received, it is considered to be a duplicate. Packet control keeps track of the current number of duplicate ACKs received at the BS. Based on the number of duplicate ACKs received and the user-defined duplicate ACK threshold, duplicate ACKs are evenly dropped and are not sent to the sender. The number of ACKs to be dropped is equal to the difference between the user-defined duplicate ACK thresholds at the BS and at the FH. For example, if the user-defined duplicate ACK threshold is 6 and TCP has defined the three duplicate ACK threshold, every second duplicate ACK is dropped.

New ACK: If the ACK number has not been previously received, the ACK is considered new. The last wireless ACK is updated, the counter for the current number of duplicate ACKs is reset, and the ACK is forwarded to the sender.

The design of the ACK filter is based on the observation that a wireless link has a high number of re-ordered segments, which is the primary cause of spurious fast retransmit. By filtering some duplicate ACKs at the BS, the spurious fast retransmit may be reduced. If there is no packet loss in the network, filtering duplicate ACKs results in better TCP performance.

2. Data Filter: When the packet control receives a data segment from the FH, it passes it to the MH. The data filter at the BS is designed to prevent the spurious fast retransmit caused by spurious timeout.

In the case of spurious timeout, retransmissions of the unacknowledged segments unnecessarily consume the scarce wireless link bandwidth and also trigger additional spurious fast retransmits. Therefore, their prevention is essential in solving spurious timeout. The data filter checks whether data segments have been acknowledged. The sequence number is checked against the last ACK received from the receiver. We consider two cases:

New data segment or unacknowledged segment: If the segment has not been acknowledged, it is forwarded to the receiver. The segment is either a new data segment or an unacknowledged segment. In the latter case, the system cannot distinguish whether the last transmission of the same segment has been received by the receiver or its ACK was lost. In both cases, even if the received segment is a retransmission, it should be forwarded.

Acknowledged segment: This segment is a retransmission due to spurious timeout. This occurs because the ACK from the BS is lost or has not arrived at the FH. In both cases, the segment should be dropped. We consider that a loss of ACKs could occur even though the BER and the possibility of congestion for ACKs are small in wireline networks. For every two identical retransmitted segments received, an ACK is sent from the BS to the sender. Hence, unnecessary retransmissions are eliminated and the problem of lost ACKs is resolved.

C. Design Considerations and Tradeoffs

Packet control filters designed to deal with the wireless link delays have to be simple to implement.

TCP option: Packet control has been designed as an option for TCP rather than a modification of TCP. Hence, it is less difficult to deploy in an existing network.

A link layer solution in BS: Packet control requires modification in the BS only. No modifications are required at the end users. Furthermore, it can be deployed incrementally

because it does not require changes in the protocol stack.

Scalable: With proper implementation, packet control filters only require retaining few constant state variables, and, hence, require minimal additional memory in the BS.

Handoff: Packet control does not require additional operations during handoffs, such as additional memory requirements or message exchanges, and will not adversely affect handoffs.

VI. Implementation of TCP with Packet Control

We implemented TCP packet control in the ns-2.26 simulator [21] on RedHat Linux 9.

Fig. 5 illustrates the logic flow of the ACK filter. The variable *numOfLastDupAck* indicates the number of duplicate ACKs that have been received for the last received wireless ACK. It is updated when a new ACK is received. It is then used, along with the user-defined duplicate ACK threshold (*redefine3DupAck*), to determine whether an ACK should be sent or dropped. The next duplicate ACK to be sent (*nextDupAckToSend*) is calculated as:

$$numOfDupAckSent = \frac{numOfLastDupAck - 1}{\left(\frac{redefine3DupAck}{3}\right)}$$

$$nextDupAckToSend = (numOfDupAckSent + 1) \times \frac{redefine3DupAck}{3},$$

where *numOfDupAckSent* is the number of duplicate ACKs that should be sent, triggered by the previous duplicate ACKs received. This ensures that duplicate ACKs will be evenly sent to the sender according to the user-defined duplicate ACK threshold in the BS.

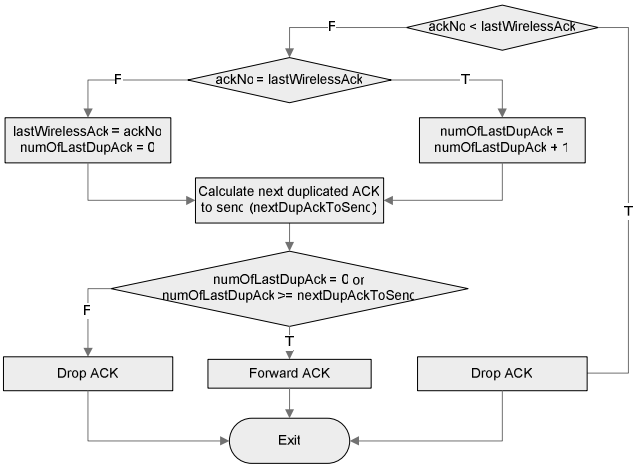

Fig. 5. Packet control: ACK filter.

Fig. 6 shows the logic flow of data filter. The variable *lstRetransWiredDataPkt* stores the segment numbers of retransmitted segments from the FH. A retransmitted segment is defined as a segment with the sequence number smaller or equal to the largest ACK number that has already been sent to the FH. These retransmitted segments are dropped. The number of retransmissions for each retransmitted segment *(m_iNumOfRtm)* is kept for each segment in the list. An ACK for a segment is generated and sent to the FH for every second retransmission of the same segment. This handles the rare situations when an ACK is lost on the path from the BS to the FH.

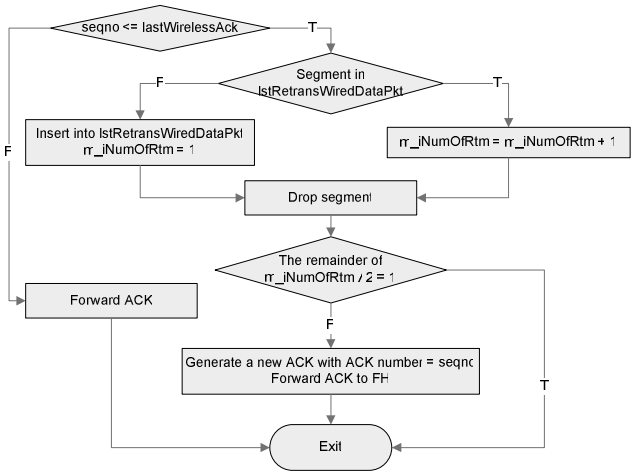

Fig. 6. Packet control: Data filter.

VII. Performance of TCP with Packet Control

The simulated network is shown in Fig. 7. A wired link connects the FH to the BS, while a wireless link connects the BS and the MH.

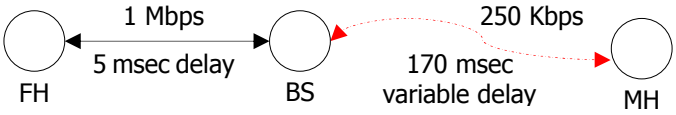

Fig. 7. Simulated network setup.

A. Scenario I: Link Delay Variation with Small Delay

This scenario is used to investigate TCP's reaction to link delay variations. For 20 seconds, FTP data are being sent from the FH to the MH in TCP packets of 1,040 bytes (default in ns-2). Link delay variation is introduced at 0.5 s. Links employ DropTail queues. The simulation results show improvement in TCP performance. The number of *cwnd* reductions vs. time is shown in Fig. 8. Due to the spurious fast retransmit caused by link delay variation, TCP without packet control has the largest number of *cwnd* reductions, which also implies the largest number of fast retransmits. TCP with packet control and the user-defined duplicate ACK threshold set to 12 has the smallest number of *cwnd* reductions. The larger the duplicate ACK threshold, the more duplicate ACKs will be dropped and fewer fast retransmits will be performed by TCP. These fast retransmits are spurious and reducing them results in higher TCP performance. Graph with no packet control and graph with user-defined duplicate ACK threshold of three overlap, validating the implementation of the filters. Since TCP sender also has the threshold of three, no duplicate ACKs are dropped and the two cases coincide.

Variations of *cwnd* are shown in Fig. 9. Since larger duplicate ACK threshold in packet control results in fewer spurious fast retransmits, *cwnd* remains large. *Cwnd* is directly related to TCP's throughput. TCP's performance may also be examined

by observing the goodput shown in Fig. 10. With an appropriate user-defined duplicate ACK threshold, TCP with packet control successfully reduces the number of spurious fast retransmits. It may improve TCP goodput by ~100%.

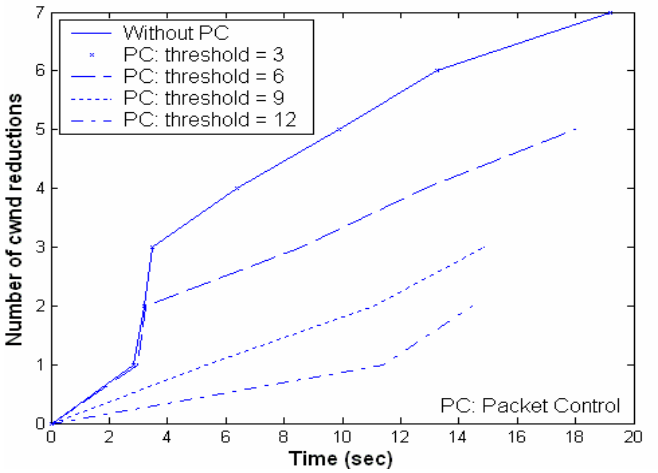

Fig. 8. Link delay variation: number of *cwnd* reductions.

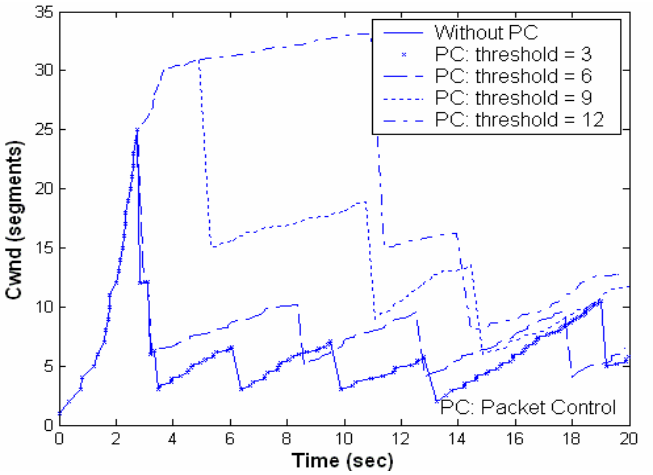

Fig. 9. Link delay variation: *cwnd*.

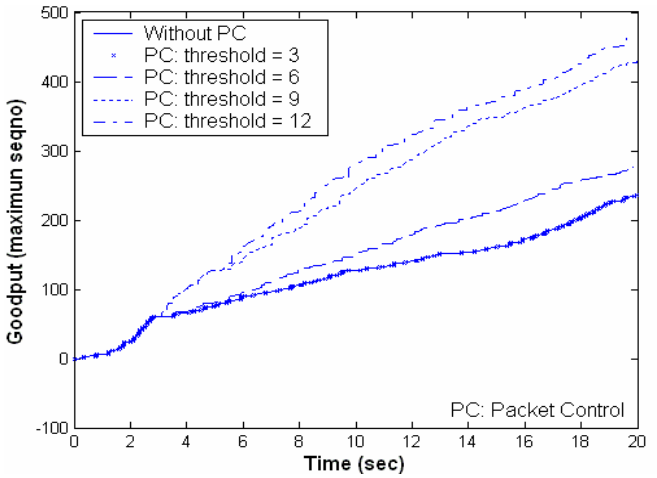

Fig. 10. Link delay variation: goodput.

B. Scenario II: Link Delay Variation with Small Delay and Link Errors

This scenario investigates the case with 1% bidirectional packet loss in the wireless link. Link layer retransmission protocols in 3G networks, such as RLP in 3G1X and RLC in UMTS, ensure that packet loss probability is less than 1% on the wireless link. TCP goodput is shown in Fig. 11. With user-defined duplicate ACK threshold of 9, TCP with packet control achieves ~30% improvement.

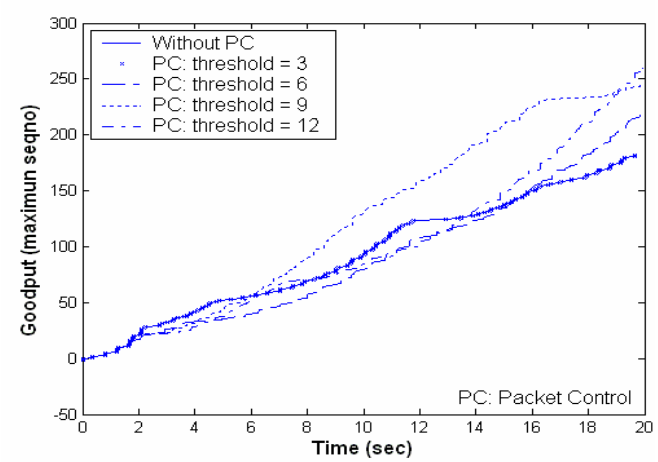

Fig. 11. Link delay variation (1% segment loss): goodput.

C. Scenario III: Spurious Timeout

We also investigate TCP's reaction to sudden large delay increase. A delay of 6 s is introduced at 5 s in the wireless link. The reaction of TCP without packet control to the long sudden delay was shown in Fig. 3. Identical simulation scenario, with packet control enabled, is used to generate the results shown in Fig. 12. They illustrate that TCP recovers faster (indicated by the third arrow) than in the case shown in Fig. 3.

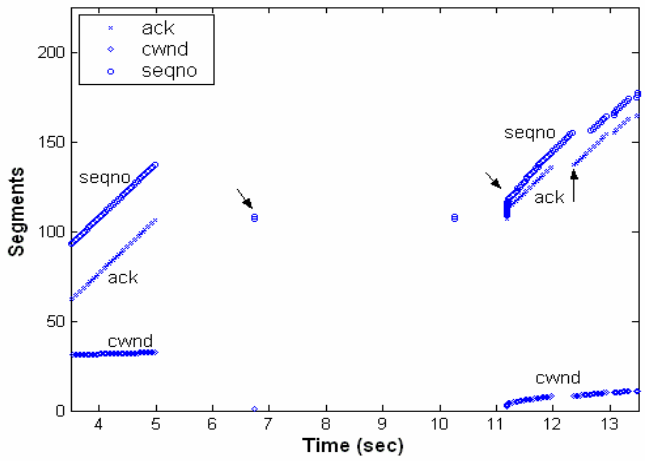

Fig. 12. TCP with packet control: spurious timeout.

A comparison of TCP goodput is shown in Fig. 13. The improvement is highly dependent on the path characteristics.

D. Delay Generator

Delays in ns-2 simulations are generated by a delay generator. We implemented two types of delays: short delay variations and relatively long sudden delays.

Short delay variations are generated based on the measurements of packet delays in wireless data networks [15].

The configuration for evaluating TCP performance was based on CDMA 1xRTT network architecture. A mobile host in a wireless network was connected to a host PC in a wired local area network (LAN) through a single BS. The "ping" application was generated by the Internet control message protocol (ICMP) ECHO. Ping packets were sent from the PC in the LAN to the mobile terminal moving at pedestrian speed. The delays vary from 179 ms to 1 s in a 3G1X system [15] and are in agreement with the "ping" latencies [22].

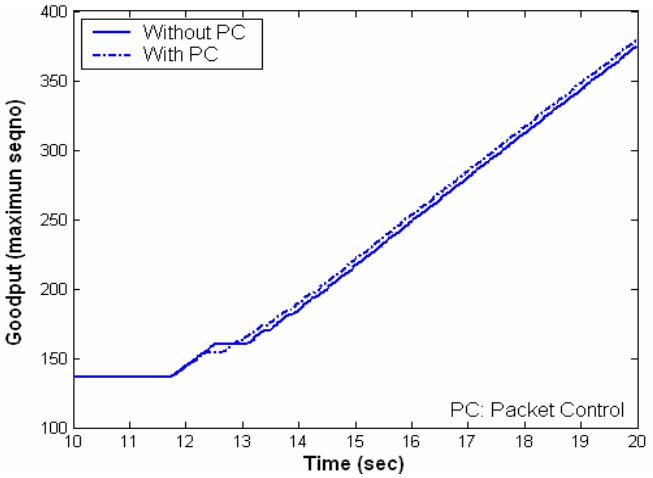

Fig. 13. TCP with packet control: goodput.

Delay values used in simulations are shown in Table I [15]. Each case is simulated with a uniform distribution generated by the ns-2 random generator. The wireline link delay was kept constant at 5 ms.

A long delay was generated by a timer. The one-way wireless link delay was kept constant at 170 ms. This value is close to the 300 ms average RTT [15]. The sudden increase of delay was simulated for 6 s, which was sufficiently long to cause a regular TCP timeout with at least one exponential back-off.

TABLE I
WIRELESS DELAYS FOR MOBILE TERMINALS

RTT (%)	Total RTT (ms)	Wireline RTT (ms)	Wireless RTT (ms)	Wireless link delay (ms)
80	316 – 400	10	306 – 390	153 – 195
10	400 – 460	10	390 – 450	195 – 225
8	460 - 605	10	450 – 595	225 – 297
2	605 - 1252	10	595 - 1242	297 – 621

VIII. CONCLUSION

In this paper, we proposed packet control filters to improve TCP performance in wireless networks with delay variations and long sudden delays. TCP connections were simulated in a mixed wireline and wireless network using the ns-2 simulator. The simulation results show that the proposed algorithms reduce spurious fast retransmit and spurious timeouts in TCP. They improve TCP's throughput, goodput, and bandwidth consumption. Goodput of TCP Reno is improved by ~100% in networks with delay variations and by ~30% in networks with 1% packet losses in the wireless link. In cases of long sudden delays, TCP performance is also improved, depending on the path characteristics. Packet control filters can be conveniently deployed at the intermediate routers to control the transmission of TCP segments and ACKs. Future improvements may include more accurate delay generators and multi-connection simulation scenarios while using genuine wireless traffic traces for performance evaluations.

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