Optimizing TCP and RLC Interaction in the UMTS Radio Access Network

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Abstract

TCP, the dominant transport protocol for Internet applications, suffers severe performance degradation due to packet losses when a wireless link is present in the end-to-end path. For this reason, the 3G specification entity, 3GPP, has defined a reliable link layer protocol, RLC, to support packet switched services over UMTS. RLC provides error recovery in the radio access network by means of an ARQ algorithm. Early studies supported the benefit of using a reliable link protocol, while more recent studies outline new problems arising from RLC and TCP interaction and how to overcome them. This article describes the most relevant issues concerning TCP–RLC interaction and evaluates the most practical enhancement approaches, based on optimum parameter configuration at the transport and link layers. We devote special attention to RLC, whose specific configuration decisions are left to operators, and provide specific guidelines for setting its parameters. In addition, we propose two operational changes for enhancing the buffer management strategy, one of the main drawbacks of RLC.

Internet applications are gaining momentum in the third-generation (3G) mobile communication systems, and are considered a key factor for the success of the Universal Mobile Telecommunication System (UMTS). Many important applications, such as file transfer, the Web, and mail, rely on the dominant transport protocol in the Internet, Transmission Control Protocol (TCP). TCP is a reliable, connection oriented, end-to-end protocol that (flow) controls the rate at which a source injects packets into the network, aiming to avoid congestion in the network and in a remote host. TCP was primary designed for wired networks, where packet losses are mainly caused by packet discarding in congested network routers. Therefore, TCP congestion control mechanisms respond to packet loss with retransmission and reduction in the rate of the source. This reduction may be unnecessary if packets are lost in the network for reasons other than congestion.

Wireless channels are characterized by propagation phenomena, including fading and outage, which cause sporadic high bit error rates that lead to packet loss bursts in the wireless link. The performance of TCP over wireless networks has been extensively studied in recent years. From the outset, the results [1] have shown the significant throughput degradation TCP suffers when there is a wireless link in the end-to-end connection.

Many research results [2] have demonstrated that a reliable link layer protocol at the wireless network improves TCP performance. For the UMTS radio access network, the Third Generation Partnership Project (3GPP) specified the Radio Link Control (RLC) protocol [3], which offers reliable delivery when configured to work in acknowledge mode (AM). RLC AM controls the user plane connection between the user mobile equipment and the radio network controller (RNC), providing a transport service to upper layers known as radio access bearers, as seen later in Fig. 4. RLC AM consists of a sliding window link layer protocol that can recover from frame losses in the radio access network by means of a selective repeat automatic repeat request (SR-ARQ) algorithm.

RLC specification defines several operation options and configurable parameters (e.g., the transmission window size and polling frequency, explained later), the choice of which is left to operators. Parameter setting has been studied in previous work [4, 5]; however, there is still no general guideline for RLC configuration when serving TCP flows.

Several features of the radio bearer connection have been found to have a negative impact on TCP performance, such as delay jitter caused by link layer retransmissions or abrupt changes in the available bandwidth. Several studies have described the performance degradation a TCP connection running over a 3G link may suffer [2, 6–8], such as unnecessary retransmission timeouts, considered a severe congestion symptom in current TCP and therefore the cause of reduced end-to-end throughput.

Extensive research has been directed at the problem of optimizing transport protocol performance over wireless links. The most developed approaches are TCP configuration for 3G links (end-to-end approach) and the optimization of link layer parameters (link layer approach). The Internet Engineering Task Force (IETF) has proposed several recommendations following both end-to-end [6] and link layer [9, 10] trends. Comparison of our simulation results with other findings in the research literature has enabled us to outline some relevant conclusions about the practical benefit of each recommendation. In addition, following a link layer approach, we disclose how each particular RLC parameter should be configured to optimize end-to-end performance within the boundaries of the 3GPP standards and identify the main dangers: incorrect parameter configuration, buffer overflow, wireless access network delay, and error burstiness in the channel. Additionally, we propose slight modifications to the radio link layer protocol that, in contrast to other proposals in the literature, are scalable, do not violate the end-to-end semantics of TCP, and respect the principle of protocol layering, while improving TCP performance. Our proposals are:

- Active queue management for RLC buffers
- Acknowledgment delay control based on the status of the wireless channel

The rest of the article is organized as follows. We give an overview of TCP and describe the problems arising from the interaction between the transport layer and the link layer in 3G. We focus on the end-to-end approach, evaluating the IETF recommendations. We explain RLC operation and give a general guideline for its parameter configuration. We describe the proposed modifications for RLC. Finally, we outline open issues and offer some conclusions.

Problem Description

An Overview of TCP

TCP is the most commonly used protocol to provide reliable transport service in the Internet. TCP sends data segments (fixed amounts of application data that fill the TCP payload) according to a sliding transmission window, while the receiver generates cumulative acknowledgments (ACKs) for packets received in sequence. The arrival of an out-of-sequence packet causes retransmission of the last ACK. The size of the send window limits the amount of data the sender may have outstanding in the network at any given time. When a packet loss is detected, the congestion control mechanisms decrease the throughput of the source by reducing the size of the send window. Packet losses are detected by means of two mechanisms, each triggering a different congestion response:

- The arrival of three duplicate ACKs activates the Fast Retransmit — Fast Recovery algorithm, which reduces the window to 50 percent.
- The expiration of a retransmission timer, whose value depends on an estimation of the round-trip time (RTT) of the connection.

If no ACK is received in the meantime, the timer expires and the congestion window is reduced to one packet. In congestion-free periods, the sender is allowed to increase the window size by one segment for every ACK received (slow start phase) as long as the window is below a certain threshold. When this threshold is exceeded, the source enters the congestion avoidance phase, where every segment in the window has to be acknowledged to increase the window size by one segment.

TCP Problems over 3G Links

In this section we describe the characteristics of the radio bearers that may harm TCP performance, an issue that has been studied by several researchers [1, 7, 8] and the IETF Performance Implications of Link Characteristics (PILC) working group, whose conclusions are summarized in [6]. Taking this previous work into consideration, we survey the most relevant features focusing on 3G links.

Latency — The TCP source has to maintain a window size larger than or equal to the bandwidth delay product (BDP) of the connection in order to use the whole available bandwidth. In this way, the source can fill the link until the window slides forward upon the arrival of an ACK. In 3G networks the latency may range from 100 ms to 1 s, depending on RLC configuration and channel radio conditions. In the early stages of a TCP connection or after a timeout, the window size starts with a small value, generally one segment. A connection over a 3G link with a data rate of 384 kb/s and an RTT of 500 ms has a bandwidth delay product of 24 kbytes, which is large enough to cause link underuse for three to six round-trips, depending on the segment size and initial window size.

Link Data Rate — The data link rate offered by the radio bearer is dynamic. This rate variability is caused by changes in the traffic load of the radio cell, and alterations in the propagation conditions and user mobility. For example, a user moving from a cell offering a low bit rate to a higher-bit-rate cell might experience underutilization of the new available bandwidth if TCP is in congestion avoidance. Changing from a cell offering a high data rate to a lower one may cause overflow at the link layer buffer, packet losses, and a sudden increase in RTT (delay spikes).

Delay Spikes — A delay spike is a sudden increase in end-to-end delay. The most frequent reasons for a delay spike are:

- The ARQ algorithm recovering from an outage
- A handover
- Blocking due to the presence of higher-priority traffic in the channel
- Withdrawal of the channel when the network must provide access to higher-priority users

A delay spike may cause a retransmission timeout (RTO) in the TCP source. This event is known as spurious timeout if packets are just delayed at the link layer, but none is actually dropped. The retransmission triggered by a spurious timeout is redundant, and the reduction of throughput is generally excessive. In addition, several consecutive timeouts cause a decrease in the slow start threshold (*ssthresh*); therefore, the optimal window size is recovered more slowly.

ACK Compression — This phenomenon, also called burstiness of ACKs, is studied in [8], and affects the self-clocking nature of the TCP flow control. The queuing of ACK packets in the reverse path of a TCP flow may result in the almost instantaneous arrival of bursts of ACKs at the sender. This can break the TCP self-clocking operation and cause long packet bursts. This situation may happen when the downlink connection is delivering packets and the receiver, at the user side, is generating ACKs while the uplink connection is suffering a burst of frame losses. The uplink and downlink channels in the wideband code-division multiple access (WCDMA) radio interface are separated by about 200 MHz, so they may experience different and independent channel fading behavior. Therefore, one direction could be error-free while the other could be suffering a deep fading period.

Packet Losses — Depending on the link layer configuration, the radio bearer can support a low packet loss rate even in the presence of a transient high frame loss ratio in the channel. The ARQ mechanism may mask almost every packet loss in the wireless channel, but the recovery at the link layer appears to the higher layer as delay jitter. However, even if the link layer is configured to offer a highly reliable connection, packets may be lost because, as described in [4, 8], frame losses in the downlink wireless channel result in higher buffer occupancy at the network side RLC entity. If the situation persists, packets may be lost due to buffer overflow.

Enhancement Strategies

Many solutions have been proposed to improve data delivery at the transport layer over wireless links. They can be classified into four categories:

- *End-to-end*: This consists of optimizing the transport proto-

RLC parameter	Setting
PDU payload size	320 bits
PDUs per TTI	12
TTI (transmission time interval)	10 ms
RB nominal bit rate	384 kb/s
Transmission window	2047 PDUs
Allowed retransmisions (maxDAT)	10
In-order-delivery	True
Buffer size	80 SDUs
RLC one-shot polling triggers	
Poll window = 50%	
Last PDU in buffer	
Last retransmitted PDU	
RLC recurrent poll timers	**Setting**
Poll timer	110 ms
Receiver status generation	**Setting**
Status prohibit timer	300 ms
Missing PDU detection	True
Connection characteristics	**Setting**
RTT wireless access network	100 ms
RTT fixed network	200 ms
Wireless channel average frame loss ratio	10%
Normalized doppler frequency	0.01
TCP parameters	**Setting**
TCP flavor	Reno
Maximum TCP/IP packet size	1500 bytes
Maximum allowed window	64 kbytes
Initial window	1

Table 1. *Simulation parameters setting values. RLC parameters not present in this table are deactivated.*

col to improve its performance in the presence of wireless links. There are two subcategories within this approach:
—Modifications in the TCP protocol itself
—The selection of TCP configuration options within the framework of approved IETF recommendations
- *Link layer*: The objective of this approach is to hide wireless errors from the transport layer, while minimizing negative interactions. Proposals in this area can be further classified into two subcategories:
—The development of new algorithms to enhance interlayer interactions
—The search for the optimum parameter configuration of the current link layer protocol standard
- *Split connections*: An intermediate node, generally called a performance enhancing proxy, splits the TCP connection between the sender and receiver into two separate connections. Specific enhancements are applied to the connection containing the wireless link.
- *Cross-layer*: The main concept of this scheme is to break the client-server working principle of the traditional layer stacks, allowing information exchange between protocols of different layers to adapt the behavior of each protocol accordingly.

Optimizing TCP

The most practical end-to-end approach consists of configuring TCP according to the properties of 3G radio bearers. In [6] the IETF summarizes and recommends some currently approved configuration options that may enhance TCP over 3G links. In this section we explain these options and present our own evaluation based on simulation experiments. The simulated channel generates error bursts according to the model described in [11] where the Doppler frequency (f_d) of the user equipment determines the average burst length with lower f_d values causing longer bursts of errors. The Doppler frequency represents the deviation from the nominal channel frequency at the receiver side, and depends on the user speed. It is usual to employ the normalized Doppler frequency, which, in our environment, is equal to the product of f_d and the radio frame duration (10 ms).

To evaluate the real benefit of the proposed TCP options, the RLC parameters were set according to the optimizing considerations described later and detailed in Table 1 (see Fig. 4 for an explanation of the abbreviations). The radio bearer nominal rate is 384 kb/s in both directions. The error probability is the same in the uplink and downlink. Unless stated otherwise, the frame loss ratio is fixed at 10 percent, a typical UMTS design value.

Increase of Bandwidth Utilization — The goal is to avoid idle initial periods of the connection and react faster to sudden bandwidth increases even if TCP is in congestion avoidance. Two configuration options were proposed to meet this target, *Increased Initial Window* and *IP MTU Larger than Default*. The first mechanism has been proven as a safe option and is especially effective when transmitting few data packets. The second scheme allows the source to use larger packet sizes; therefore, the transmission window increases faster. Figure 1 shows how the configuration values of these parameters affect the downloading time of a 5-kbyte file. The figure shows that using 1500-byte packets, which is also the maximum allowed in the UMTS radio access network, in combination with an initial window of 4 packets provides the best download times.

Robustness against Packet Losses — Packet losses should not be a cause of concern because, according to 3GPP quality of service (QoS) objectives, the packet loss probability of the radio bearer should range from 10^{-6} to 10^{-3}. Nevertheless, the IETF recommends two mechanisms, *limited transfer* and SACK, whose main objective is efficient recovery from packet losses. In a SACK connection, the receiver can explicitly inform which packets are received, even if they arrive out of sequence. This permits the sender to retransmit only the missing packets. Limited transfer extends Fast Retransmit — Fast Recovery mechanisms to connections with small congestion windows, and is therefore best suited to short-lived flows.

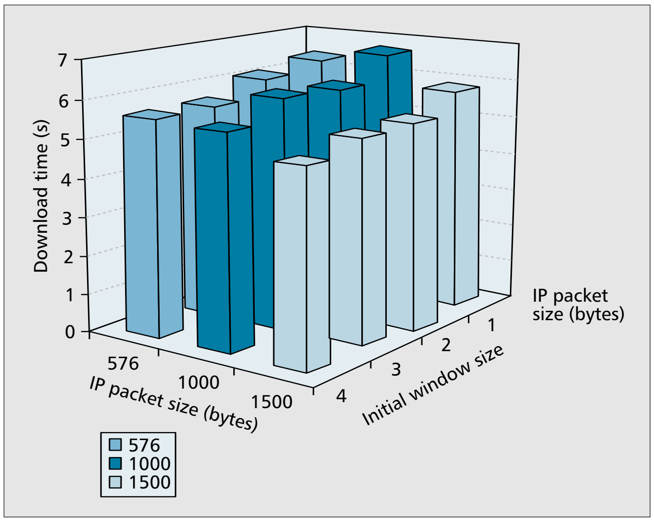


Figure 1. *Download time of a 5-kbyte file for several IP packet sizes and different initial window sizes.*

For connections with a large bandwidth delay product, the probability of multiple packet losses in a single window increases. In such cases TCP SACK has proven to be effective, providing higher robustness than other TCP implementations. Previous results [12], based on extensive measurements on 2.5G networks, showed that the SACK option improves the goodput of TCP flows by about 10 percent. The goodput estimates the amount of successfully delivered data per second, providing the user perceived quality of the connection, instead of the sender data rate (throughput), which does not discount packet losses and retransmissions. Our simulation results, shown in Fig. 2, compare TCP Reno and SACK goodput performance in a 3G network for several RLC buffer capacities. We can see how the SACK option improves TCP goodput, especially for lower buffer sizes, where overflow causes frequent packet losses.

Reduction of Delay Jitter — The conservative computation algorithm of the TCP retransmission timer can mitigate but not avoid the problems arising from the delay jitter and delay spikes in a radio bearer. The IETF proposal for solving this problem is the *TCP timestamp option*, which permits TCP to sample the RTT for each packet instead of sampling it once in each window. A higher sampling rate should allow the sender to react more quickly to sudden increases of the delay by updating sooner the RTO value to a more suitable one. However, this is not always true. Many previous studies [11] have shown that when the user equipment is stationary or moving slowly, frame error bursts in the wireless channel are highly correlated. This means an increase in the average length of frame loss bursts in the channel. The problem arises when relatively long periods of good channel conditions are followed by a long degraded period. In this case the *timestamp* option results in a higher probability of spurious timer expiration because, due to the higher sampling rate, it is probable that, during an error-free period, the estimation of the RTT could rapidly converge to an accurate but excessively small value, making the sender very vulnerable to delay spikes. The non-*timestamp*-based measure is coarser, but also more conservative in this scenario.

Figure 3 shows an example of the evolution of the RTT measure, with and without the *timestamp* option, in a situation where only the *timestamp* option suffers from spurious timeouts. The effect of the *timestamp* option can also be noticed in Fig. 2. Experimental measurements in [12] also showed this degradation in 2.5G networks, although no explanation was provided.

Optimizing RLC

Optimizing TCP performance by means of a suitable configuration of RLC parameters has the advantage of not requiring any modification at the end hosts. The following subsections describe the RLC operation and provide specific recommendations for setting the parameters, considering the permitted 3GPP values, to optimize data transport.

RLC AM Operation

The RLC AM is basically an SR-ARQ link layer protocol. As shown in Fig. 4, packets from higher layers (service data units, SDUs) are segmented and concatenated into link layer frames (protocol data units, PDUs) for transmission over the radio link. Figure 5 shows the basic operation of the error recovery algorithm. The receiver side detects missing PDUs based on the error control field or by detecting gaps in the frame sequence numbers. The RLC receiver side requests retransmission by sending back a bitmap report, called status PDU, informing which frame within the receiving window has been received (ACK) or lost (negative ACK, NACK). Upon reception of a status message, the sender can advance its transmission window if one or more in-sequence frames are acknowledged, so that new PDUs can be sent. If there are NACKs in the status message, the sender retransmits the missing PDUs giving them priority over new ones.

A status message is either issued when the receiver is polled by the sender or self-triggered. At the sender side, a polling request is made by marking the poll bit in the header of a PDU. The poll request can be triggered by seven configurable mechanisms, two of them considered recurrent:

- *Timer-based polling*, consisting of a periodic timer whose expiry triggers a poll.
- *Poll timer*, also driven by a periodic timer that is only started when a poll PDU is sent. The aim of this timer is to retransmit the poll if it is lost in the link, so the timer is stopped upon receiving a status message.

The rest of the mechanisms are considered one-shot polling methods, because they are triggered when the sender is in a particular state:

- *Last PDU in buffer* activates the poll bit in the last frame in the transmission buffer.
- *Last PDU in retransmission buffer*, same as above for frames in the retransmission buffer.
- *Poll every* N *PDU* triggers the poll for every *N* PDUs sent.

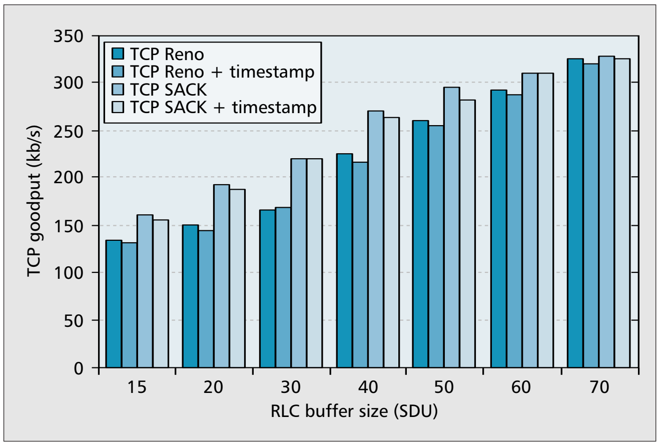
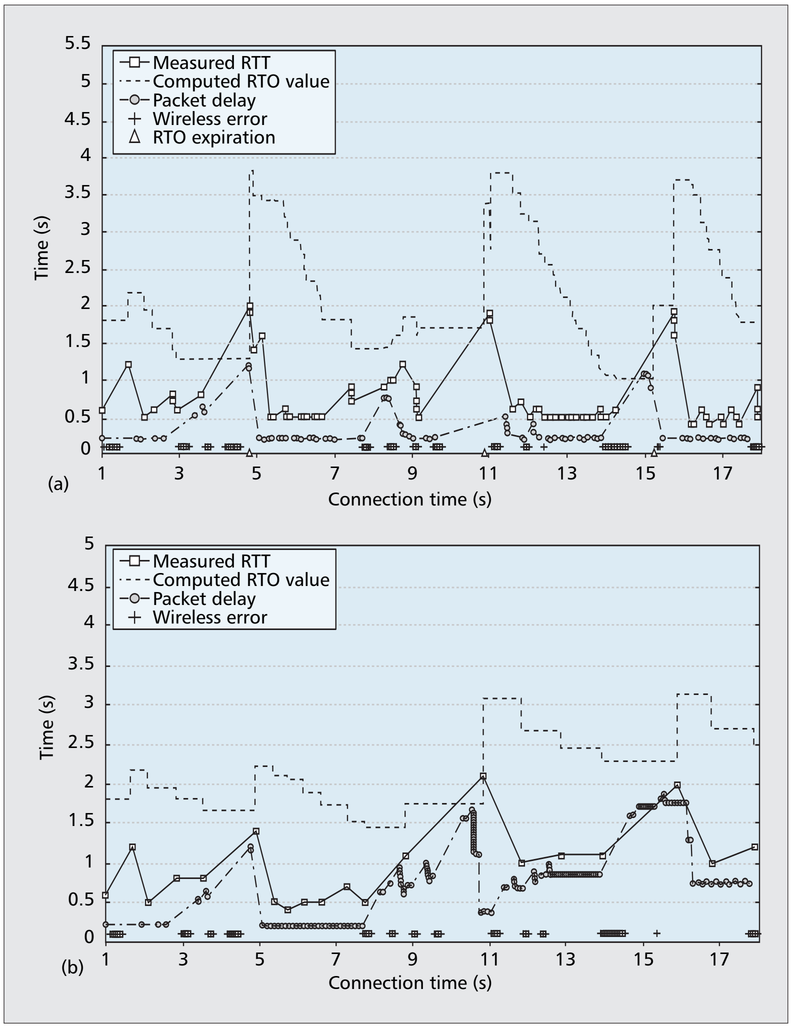


Figure 2. *End-to-end goodput for different TCP configuration options and different RLC buffer sizes.*



■ Figure 3. *Variables of a TCP Reno connection with and without the timestamp option, running over an RLC radio bearer with the same channel evolution. The wireless link suffers an average frame error ratio of 20 percent in both directions. Errors in the channel occur in bursts. For the sake of clarity, only frame errors in the downlink direction are shown. a) TCP with timestamp option. The value of the computed RTO rapidly converges to the measured RTT value. Three consecutive delay spikes result in three spurious RTO timeouts. b) TCP without timestamp option maintains a more conservative RTO value under good wireless conditions. There are no timer expirations.*

- *Poll every* N *SDU*, for SDUs.
- *Window-based polling* consists of marking the poll bit when a certain percentage of the transmission window is sent.

The receiver side transmits a spontaneous status message when any of the following status triggers are configured: *detection of missing PDUs*, which causes the sending of a status message when a missing PDU is detected, or *periodic status report*, which is driven by a periodic timer.

An excessive generation of status messages may waste energy in the mobile device and trigger redundant retransmissions at the link layer. The maximum signaling rate is controlled by means of two timers. The *poll prohibit timer*, in the sender side, and the *status prohibit timer*, in the receiver side, set the minimum allowed time lapse between successive transmissions of signaling frames at each side.

General Framework for RLC Parameter Configuration

Based on previous research and our own findings, this section provides a general guideline for configuring the radio network link layer to enhance upper layer performance. These configuration options have been checked by means of extensive simulation experiments. The TCP version used was TCP Reno, without the *timestamp* option. Unless stated otherwise, the parameter settings are those shown in Table 1.

Polling Strategy — The RLC AM polling function should be sufficiently frequent to make full use of the available link bandwidth and reduce packet queuing time. However, an excessively aggressive polling strategy has two main drawbacks:

- The possible generation of unnecessary control messages, wasting radio resources and increasing mobile energy consumption
- The possible triggering of redundant retransmissions

It is advisable to use a combination of polling mechanisms in both the sender and the receiver [5]. For example, in the proposed configuration (Table 1), the sender would poll the receiver when it has transmitted a significant amount of data, and the receiver would generate a status report as soon as a frame loss is detected. The idea is to generate status signals when they are really needed. Protocol stalling can be avoided by means of a periodic trigger, generally the *poll timer*. Finally, to control signaling overload, the *status prohibit timer* should be configured to a value above the RTT of the radio access network. This timer will be activated after the sending of a status signal, preventing the sender from generating any other control message until the timer expires (Fig. 5).

Figure 6a shows the TCP goodput for different values of the status prohibit timer. Figure 6b shows the signaling efficiency, expressed as the percentage of status messages per delivered radio frame. For a wireless round-trip delay of 100 ms a status prohibit timer of 300 ms provides a good balance in terms of goodput and signaling efficiency.

Window Size — Previous research results show that there is an optimum window size at the link layer, for which optimum delay and throughput performance is achieved. Configuring a window size beyond this optimum does not bring any benefit [13], but requires more memory space in the terminal and on the network side. The optimum link layer window size in terms of throughput and delay performance is obviously related to the radio access network delay and the polling frequency, as shown in Fig. 7. A higher error rate also imposes larger optimal windows, although, for a fixed error rate, the average duration of the error period, also known as degree of burstiness, does not change the optimum window value. In 3G networks, the power control technique holds the error rate around a preset value, typically around 10 percent.

Transmission Buffer Size — In case of full buffer occupancy, the RLC specification [3] defines a simple droptail policy. The buffer size is not a protocol parameter but should be carefully selected when dimensioning the hardware and configuring the operating system of the equipment. Buffer overflow in the RLC buffer has been identified as one of the main threats to transport layer performance. In [4] it was shown that, under some assumptions (e.g., error-free uplink channel and uncorrelated downlink frame losses), buffer occupancy is limited to

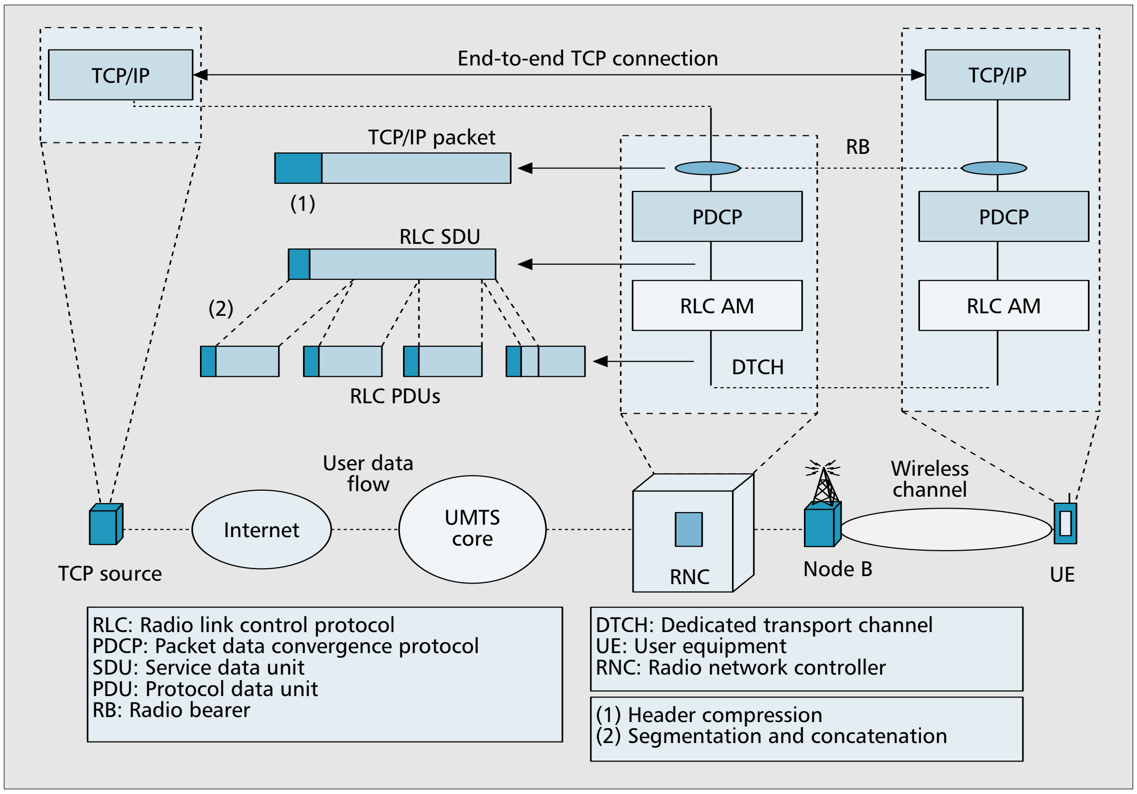


Figure 4. *Elements and protocols involved in the end-to-end connection. The wireless channel connects the user equipment with the UMTS radio station (Node B). The Node B transparently connects the user data flow with the radio network controller. The lower layer offers an unreliable transport service to the RLC protocol through the dedicated transport channel.*

the maximum allowed value of the TCP transmission window (*awnd*), usually 64 kbytes. However, more realistic network situations, with burst errors in the wireless channel, in both the uplink and downlink, show that the awnd value may be exceeded when the channel is temporally closed and a spurious timeout occurs at the transport layer. The performance degradation caused by a small buffer size can be seen in Figs. 3 and 10, where the RLC queue size is expressed in maximum sized SDUs (1500 bytes).

Discarding Mode — RLC can be configured in two different SDU discarding modes:
- When its buffering time exceeds a certain amount of time
- When one of the frames containing a segment of the PDU is retransmitted a specific number of times (*maxDAT*)

Previous results [4] show that the second strategy benefits TCP, especially when there is a high limit in the number of retransmissions. Our simulation experiments show that the optimum value of *maxDAT* depends on the round trip time of the radio access network, ranging from 6 to 8 when the round trip delay varies from 60 ms to 300 ms. On the other hand, the burstiness degree of frame losses in the channel has no notable impact on these optimum values. According to the common delays of the radio network [14], a value of 10 for *maxDAT* could be considered safe for a wide range of network situations.

The parameter *maxDAT* has a strong interaction with the polling frequency. If *maxDAT* is set to a low value (less than 10) and the polling strategy is aggressive, the link layer may be forced to discard long bursts

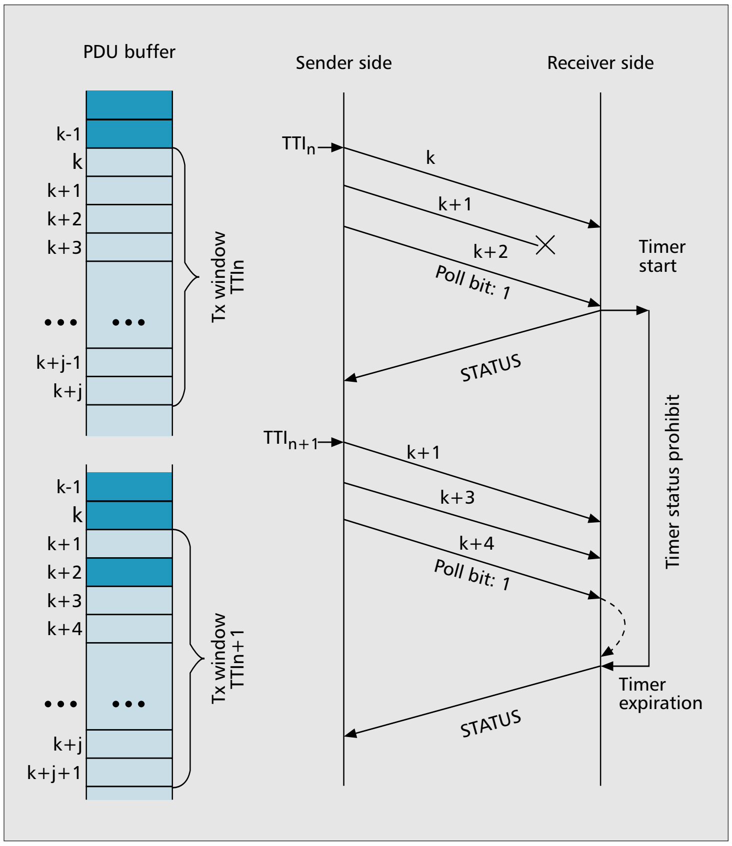
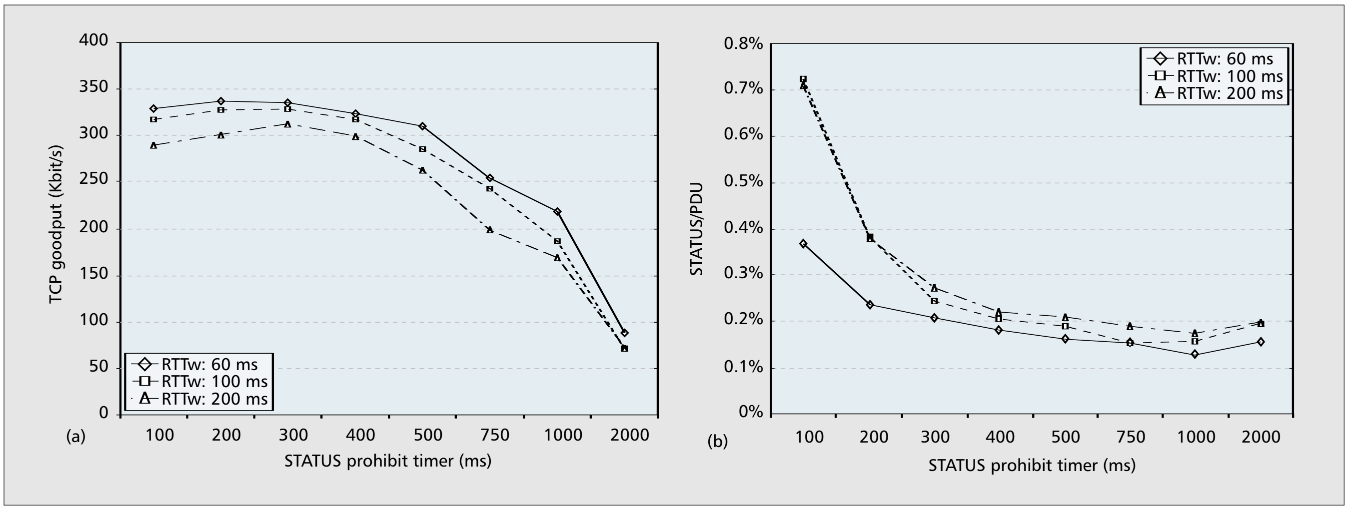
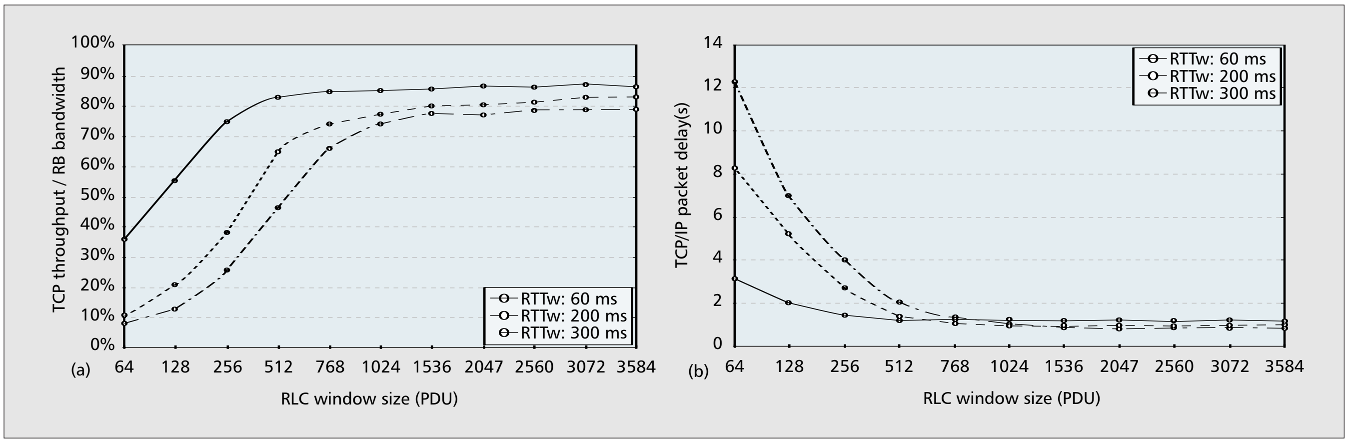


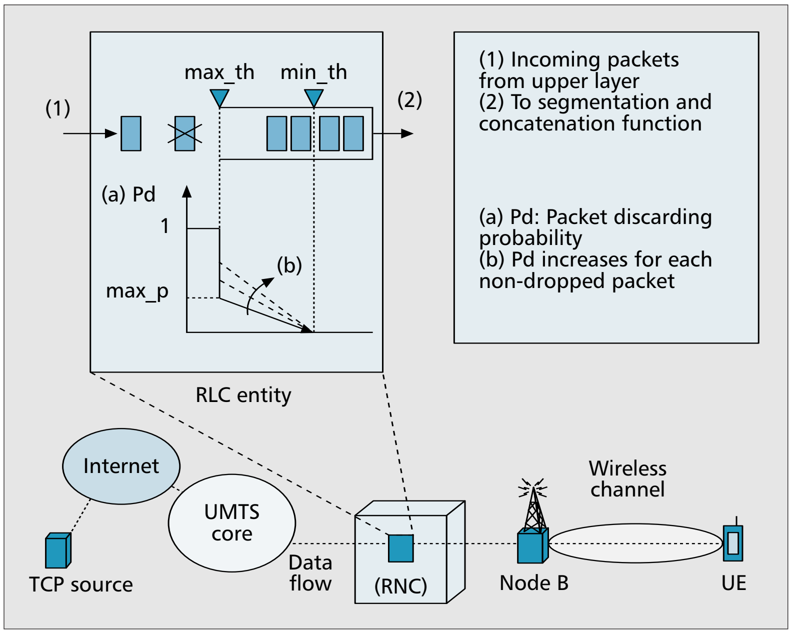
Figure 5. *Basic RLC AM error recovery operation with status prohibit timer.*



■ Figure 6. *a) TCP goodput for different values of the status prohibit timer; and b). signaling efficiency for different values of the status prohibit timer.*



■ Figure 7. *a) Influence of the RLC window size on TCP throughput for different radio access network delays; b) influence of the RLC window size on TCP end-to-end delay for different radio access network delays.*



■ Figure 8. *Implementation of the RED buffer management algorithm at the RLC layer.*

of packets when a transient period of high error rate occurs in the channel. The reason is that, according to the RLC protocol operation, every time a polling action is triggered, the sender retransmits the highest unacknowledged frame in order to obtain the maximum information from the generated status message. If the error period is sufficiently long, this polling frame could reach the maximum number of retransmissions before the connection is re-established, especially if the polling frequency is high. When *maxDAT* is reached, the frame is discarded, as well as all the unacknowledged ones, causing a burst of packet losses.

In-Order Delivery — The RLC layer may be configured to deliver packets in sequence to upper layers. In a TCP connection, out-of-sequence packets have a negative effect on TCP throughput, which was studied in depth in [15]. It is generally accepted that in-sequence delivery at RLC should be the selected option when serving TCP flows, as long as more robust TCP versions are not developed and deployed. In the simulation environment depicted in Table 1, the goodput performance of TCP Reno fell by almost 40 per-

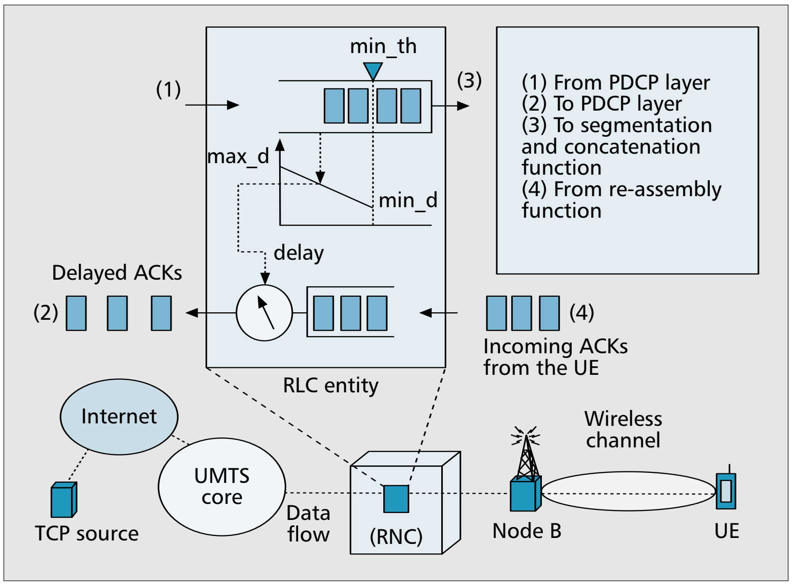


Figure 9. *Implementation of the ACK Delay Control algorithm at the RLC layer.*

cent when the RLC did not deliver packets in sequence.

Proposed Improvements to RLC

Our proposals for optimizing radio bearer behavior are based on the idea that when the link layer has been configured to be reliable, only buffer overflow may cause packet losses in the radio access network. The RLC entity will experience increasing buffer occupancy during periods with errors in the channel, such as a router in a congestion situation. The drop-tail packet dropping strategy of the current RLC standard may cause the loss of consecutive packets, which have a negative effect on TCP throughput. Hence, it seems appropriate to apply known congestion management techniques from Internet routers to handle this congestion-like situation within a radio bearer flow.

We propose an adaptation of the following two mechanisms:
- Random early detection
- ACK delay control

These approaches present two common advantages over other schemes like split connections. First, they do not require snooping into packet headers, so they are compatible with security schemes. Second, they keep the end-to-end semantics of TCP, because no transport layer packet is generated at intermediate nodes.

Random Early Detection

Random early detection (RED) [16] consists of detecting incipient congestion by computing the average queue size. The congestion is notified to the TCP source by dropping a packet arriving at the node. This lets the source adapt its sending rate before the buffer capacity is exceeded and consecutive packets are discarded. When the average queue size exceeds a preset threshold (*min_th*), this is taken to be a sign of congestion and the node could drop a packet with a certain probability. This probability is a function of the queue size and the number of packets accepted since the last discarded packet within the congested period. Figure 8 shows how this mechanism could be adapted to handle the congestion situation in the downlink buffer of the network side RLC entity. Given the small number of flows multiplexed in the RLC buffer, using the instantaneous queue size instead of the average queue size allows a faster reaction against signals of channel degradation and increases the algorithm efficiency.

Several advantages may be obtained from the RED strategy:
- Long bursts of packet losses are avoided.
- The average buffer occupancy is held around a certain value, and so packet delay is also controlled.
- RED is a resource effective algorithm and, with a relatively small buffer we can achieve almost the same performance as a droptail scheme with an overdimensioned buffer size (no drops possible).
- When several TCP flows are multiplexed within the same radio bearer, this technique ensures a fair share of the bandwidth between the flows.
- It is not necessary to separate each transport flow, so its scalability is higher than other proposals [17] aimed at preventing buffer overflow at the link layer.

The main drawback of RED may be the additional computational cost of estimating a separate packet dropping probability for each user. In [16] simple algorithms were proposed

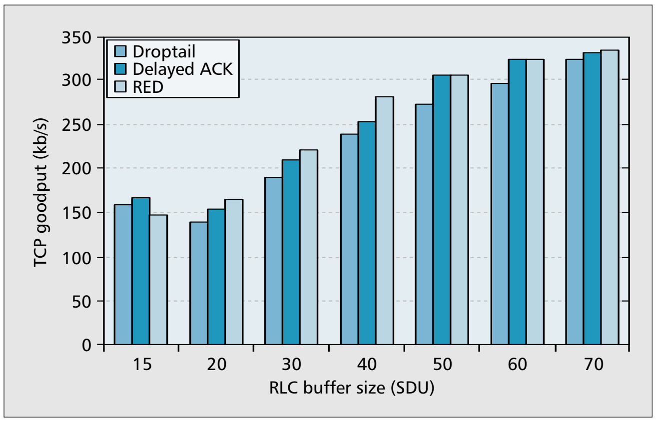


Figure 10. *Goodput performance of the three schemes for different RLC buffer sizes.*

for the computation of this probability, which were suitable for routers at the beginning of the last decade. The computational complexity of RED is still lower than that of schemes that require per-transport-flow management.

ACK Delay Control

ACK Delay Control was presented in [18] as an algorithm for routers aimed at improving the performance of TCP over satellite links. The basic idea of this algorithm is to delay acknowledgments traversing a node whose forward connection is congested. The congestion is detected when the buffer occupancy exceeds a fixed threshold (*min_th*). When this happens a delay is applied to the ACK packets in the reverse flow to slow down their departure rate. This mechanism takes advantage of the TCP self-clocking principle, as the reduction in the ACK flow will reduce the sending rate of the source.

In contrast to the basic algorithm, we propose that the delay applied to outgoing ACKs be determined by the buffer occupancy. This allows the source to gradually adapt its perception of the available bandwidth and the round-trip delay to conservative values. If the error burst in the wireless channel is long, the buffer occupancy will continuously increase. Higher buffer occupancies require higher reductions in the sending

Parameter settings for RLC RED buffer management							
Buffer size	15	20	30	40	50	60	70
min_th	1	5	20	30	30	40	40
max_p	0.01	0.02					
Parameter settings for RLC Delayed ACK							
Buffer size	15	20	30	40	50	60	70
min_th	0	0	0	20	20	20	40
min_d (ms)	30	10	30	20	20	50	10
max_d (ms)	200						

Table 2. *Parameter configuration of buffer management schemes.*

rate of the source, while for short error bursts excessive delay in the ACKs would be unnecessary. Another objective is to prevent spurious timeouts as long as no packet is dropped. A progressive increase of the ACK interarrival time helps the source increase its RTT measurement, preventing spurious timeouts. Besides, the magnitude of the delay spike caused by temporary channel unavailability is reduced, because the source receives ACKs within the degradation period.

The fixed delay scheme was compared to the variable delay strategy, simulating a wide range of configuration options. In most cases the variable delay scheme achieved better goodput performance, with gains ranging from 3 to 13 percent over the fixed delay strategy.

The diagram in Fig. 9 shows the main features of this algorithm. When the downlink buffer occupancy exceeds the minimum threshold (*min_th*), the next incoming ACK packet in the reverse direction is delayed by *min_d*. If the queue occupancy keeps growing, the value of the induced delay is augmented linearly until the maximum delay value, *max_d*, is reached.

The advantages of this algorithm are:
- For RLC entities with small buffer, the probability of packet dropping is reduced.
- The size of the buffer required to achieve a target throughput is lowered with respect of the simple droptail scheme.
- Reducing the reverse ACK pace helps the source to increase its round trip time estimation, reducing the probability of spurious timeouts.

This also benefits RLC entities with large buffers.

Figure 10 shows a comparison of the average goodput performance of these mechanisms. For almost every RLC buffer size, the new mechanisms improved the goodput. The parameter settings for the Delayed ACK and RED algorithms are shown in Table 2. Active queue management algorithms may also affect buffer dimensioning decisions. For the 384 kb/s bidirectional bearer of this example, when an active queue management algorithm is used, a buffer capacity of 60 packets provides almost the best performance, while a droptail scheme requires a buffering capacity of 70 packets to achieve the same performance. A smaller buffer size saves resources in network equipment and reduces the end-to-end delay.

Conclusions

One of the most used transport protocols for Internet applications, TCP Reno, requires protection against the frequent error bursts in wireless environments. This is the main purpose of the 3G link layer protocol when it is configured in acknowledge mode. We show that interaction between the RLC protocol and TCP may bring new problems. We have described the most practical approaches, based on suitable parameter settings in both the transport and link layers. IETF recommendations for configuring TCP over 3G links have been evaluated. Optimizing RLC does not require any changes at the end host, and is open to operator decisions. For this reason, we have offered a configuration guideline for improving performance in this layer. The protocol features are classified into five main areas: polling function, window sizing, buffer dimensioning, persistence degree, and in-sequence delivery. One of the main weaknesses of the RLC protocol is its buffer management scheme, which may cause consecutive packet losses in the downlink direction in certain channel fading situations. Packet loss bursts are especially harmful to TCP Reno flows, so we propose incorporating a buffer management scheme in the RLC operation. Two existing algorithms have been adapted, RED and ACK Delay Control, taking into consideration the characteristics of radio bearers. We have shown that with proper configuration, both schemes may improve transport layer performance.

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