ACK RATE CONTROL STRATEGY AT THE LINK LAYER TO ENHANCE TCP OVER 3G LINKS

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Abstract: When TCP is carried over 3G links, overbuffering and buffer overflow at the RLC layer degrades its performance. In order to prevent these undesired cross-layer interactions, we propose an algorithm to control the rate of the acknowledgements (ACK) traversing the RLC entity at the network side. Since the sending rate of a TCP source is determined by the arrival rate of the ACKs, the basic idea of this scheme is to adjust the inder-departure time of the ACKs in the uplink direction, according to the congestion level of the downlink buffer. By means of extensive simulation experiments, we disclose the influence of each parameter on TCP performance. We provide several configurations that improve end-to-end goodput while reducing the delay. The operation of the algorithm is deterministic; it does not require changes in 3G specifications nor in TCP itself, and preserves TCP end-to-end semantics.

1 INTRODUCTION

Third generation cellular networks (3G) are expected to be an important part of the Internet. Many Internet applications like e-mail, web surfing and file transfer, rely on TCP for the end-to-end transport. In 3G radio access networks, the link layer is managed by the Radio Link Control (RLC) protocol (3GPP, 2005). For packet switched services, RLC is usually configured to provide a reliable service, recovering from propagation errors. A reliable RLC layer reduces packet losses perceived at TCP layer, avoiding the triggering of unnecessary congestion control measures (Inamura, 2002).

Although, in general, RLC improves TCP transfer rate over a radio bearer (RB), several characteristics of 3G links like high and variable latency and buffer overflow of the RLC downlink buffers (Alcaraz, 2006), have undesired effects on TCP performance.

One of the strategies to overcome these effects is to improve the RLC layer. In contrast to other proposals, like split-connection proxies, this approach does not require changes in the TCP itself and does not break the end-to-end semantics of TCP. Several works propose the use of Active Queue Management (AQM) techniques in the RLC downlink buffer. It was shown in (Alcaraz, 2006) that these mechanisms can enhance TCP goodput while reducing end-to-end latency. A slightly different approach is the use of ACK Delay Control at RLC. This mechanism was presented in (Wu, 1999) as an algorithm to improve the performance of TCP over satellite links. The underlying idea is to delay acknowledgements travelling through a node where its forward connection is congested. Since the ACK arrival rate at a TCP source determines the sending rate of new packets, this is a simple mechanism to reduce packet drops due to buffer overflow. The congestion is detected when the buffer occupancy exceeds a fixed threshold (*minth*). When this happens a delay is applied to the ACK packets in the reverse flow to slow down its departure rate. This mechanism takes advantage of the TCP self-clocking principle, as the reduction in the ACK flow rate will reduce the sending rate of the source. In the original proposal the delay applied to ACKs is a constant value.

In this paper we propose an adaptation of this algorithm for 3G links, where ACK rate is adapted according to the downlink queue length. We consider a general delay computing function, in which the fixed delay and the linear relation are particular cases. This generalization gives us a wider point of view to find optimum configurations. We

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provide a feasible implementation of the algorithm, and by means of extensive simulation tests, we discuss the influence of each parameter and propose configurations that clearly improve end-to-end performance compared to the conventional RLC operation.

The rest of the paper is organized as follows. Section 2 describes the characteristics of the 3G radio bearer that degrade TCP performance. Section 3 explains our ACK rate control algorithm in detail. Section 4 provides a brief description of the simulation environment. Section 5 discusses the influence of each parameter in end-to-end performance based on extensive simulation results. The paper concludes in section 6.

2 MOTIVATION

Previous experimental (Chakaravorty, 2005) and simulation (Rossi, 2003) results provide a clear view of the characteristics of 3G wireless links. The behaviour of the link buffer occupancy has shown a great impact on TCP performance.

3G links employs per-user buffering. Flows going to a single mobile terminal share a single buffer. Therefore, radio bearers are expected to multiplex a number of simultaneous connections ranging form 1 to 4 TCP flows (Gurtov, 2004). At a reliable RLC layer, the upper layer packets will be stored in the downlink buffer until they are fully acknowledged by the receiver side. The consequence is that, as described in (Bestak, 2002) frame losses in the downlink channel result in higher RLC buffer occupancy at the network side. Considering that the current RLC specification propose a drop-tail scheme, the buffer may overflow causing consecutive packet losses. This situation is especially harmful in the first stages of a TCP connection (slow start) and has a higher impact in TCP Reno, which can only recover from consecutive losses with a Retransmission TimeOut (RTO). An RTO reduces TCP transmission window to one, causing the highest reduction of the source rate.

The buffer should be large enough to avoid frequent overflow. However, excessive queuing causes some additional problems (Chakaravorty, 2005) like Round Trip Time (RTT) inflation, unfairness between competing flows and viscous web surfing.

Fig. 1 illustrates the end-to-end goodput and delay of TCP over an RB for different RLC buffer sizes and a number of flows ranging from 1 to 4. The goodput accounts for the successfully delivered packets at the receiver and the delay is the transfer time of a packet in the downlink direction, at the TCP layer. The buffer size is given in RLC Service Data Units (SDU) of 1500 bytes. Table 1 shows the parameter configuration for the RLC and TCP protocols. The RLC parameters were set according to the optimizing considerations described in (Alcaraz, 2006) and (Rossi, 2003). Further details on the simulator are provided in section 4. As expected, Fig. 1 reveals that a larger buffer benefits the goodput performance but the overbuffering increases the latency.

Fig. 2 shows the trace of a TCP connection over a 384 kbit/s radio bearer multiplexing one TCP flow. The curve at the top shows the RLC buffer occupancy (BO). The buffer size is limited to 50 SDUs. The TCP congestion window (*cwnd*) is depicted under this curve, and the curve at the bottom shows the sequence number of the packets when they are sent, received and dropped.

Table 1: Simulation Parameters

3G link parameters	Setting
PDU payload size	320 bits
TTI (Transm. Time Interval)	10 ms
Transmission window	1024 PDUs
maxDAT	10
In-order-delivery	true
Status Prohibit Timer	60 ms
Missing PDU detection	true
Poll Timer	60 ms
Wireless Round Trip Delay	50 ms
Normalized doppler frequency	0,01
Poll window	50 %
Last PDU in buffer Poll	yes
Last retransmitted PDU Poll	yes
Frame Error Ratio (FER)	10%
TCP parameters	**Setting**
Maximum TCP/IP packet size	1500 bytes
Maximum allowed window	64 kbytes
Initial window	1
Wired Network Round Trip Delay	200 ms

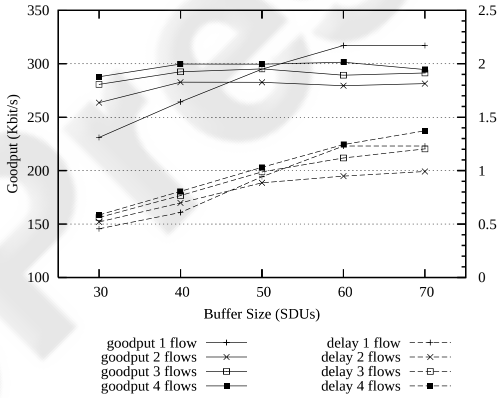

Figure 1: TCP performance over a 384 kbit/s RB.

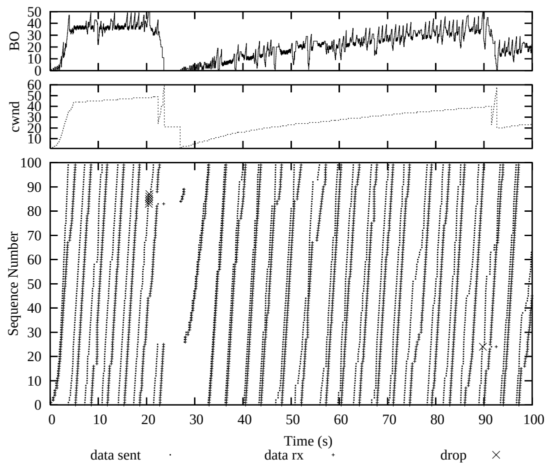

Figure 2: TCP trace over a 384 kbit/s RB.

At the first stages of the connection, multiple packets are dropped due to buffer overflow, causing an RTO. The buffer is drained because the connection reduces its rate, with the consequent underutilization of the resources. After that, the rate is recovered slowly and the overbuffering appears again, causing high delay and additional packet losses.

3 PROPOSED ALGORITHM

TCP is basically a sliding window algorithm. The transmission window limits the total amount of data that the source can transmit without receiving any acknowledgement from the receiver side. The window moves forward and its length increases with the arrival of ACKs, thus the source can inject new segments in the network. This is usually called TCP self-clocking mechanism.

In this paper we propose a scheme for controlling the rate at which TCP ACKs cross the RLC layer. Since the throughput of a TCP source is basically determined by the arrival rate of ACKs, it is possible to avoid overbuffering and buffer overflow in the RLC layer if this rate is conveniently adapted to the link situation. The buffer occupancy (BO) is the variable used to perceive the link quality, and the rate control measures are applied only when certain BO threshold is reached (*minth*). In our algorithm, the inter-departure time, t_{id}, of ACKs traversing the RLC layer is computed with a non-linear function of BO, $t_{id} = f(BO)$, which is obviously a continuous and only-increasing function because the rate has to be reduced when the BO increases.

The motivation of the variable delay strategy is to gradually adapt the source's perception of the

```
for each event
   if (event == ack_arrival)
      insert ack in uplink buffer
   if (event == id_timer_expiration)
      send first ack in buffer
      LADT ←current_time
   if (uplink buffer nonempty) and(id_timer off)
      delay ← 0
      if (BO>=minth)
         t_id = f(BO)
         actual_t_id ←current_time - LADT
         if (actual_t_id < t_id)
            delay ← t_id – actual_t_id
      if (delay == 0)
         send ack
         LADT ←current_time
      else
         id_timer ← delay
         activate id_timer

Events:
ack_arrival: arrival of an upcoming ACK
id_timer_expiration: expiration of the id_timer

Saved Variables:
LADT: Last Ack Departure Time

Other:
current_time: system clock
delay: artificial delay applied to outgoing ACKs
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Figure 4: Proposed ACK rate control algorithm.

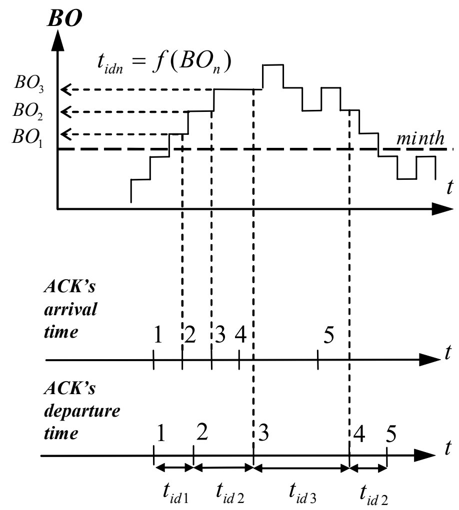


Figure 3: TCP performance over a 384 kbit/s RB.

available bandwidth and the round trip delay. Higher buffer occupancies require higher reductions in the sending rate of the source, while for a short error burst an excessive delay on the ACKs could be unnecessary. This should also help to avoid excessive delay spikes causing spurious TCP timeouts.

It may be argued that a drawback of the ACK delaying strategy is the Retransmission Time Out (RTO) inflation caused by the increment on the perceived RTT. The RTO inflation slows down the reaction of the source upon the eventual loss of consecutive packets. However, the higher delay of the reverse path is compensated by a lower buffer occupancy, which reduces the downlink packet latency. In addition, a main goal of this algorithm is the avoidance of buffer overflow, thus TCP timeouts are expected to happen rarely.

The proposed ACK rate control algorithm is detailed in Fig. 3. The algorithm relies in the use of a single timer (*id_timer*), which delays ACKs assuring that they are spaced by t_{id}. The pseudocode identifies two different events, the arrival of an ACK packet (*ack_arrival*) and the expiration of the *id_timer* (*id_timer_expiration*). Figure 4 illustrates the operation of the algorithm.

3.1 Calculating the Inter-departure Time

The original proposal for controlling the rate of the reverse ACK flow relies on a fixed delay (Wu, 1999). This strategy may be enhanced using a variable delay, e.g. setting a linear relation between the delay and the downlink queue length.

In this paper we consider a more general function (1) to estimate t_{id} with BO, where *maxd* and *mind* are the maximum and minimum values for t_{id}, and *maxth* is the buffer size.

$$t_{id} = (maxd - mind)\left(\frac{BO - minth}{maxth - minth}\right)^{\alpha} + mind$$

This function let us evaluate the effect of the variation rate of t_{id} respect BO. This variation rate, which is related to the “aggressiveness” of the algorithm, is determined by α (≥0). Lower values of α are tied to more aggressive delaying policies. Figure 5 shows the shape of (1) for different values of α. From now on, to reduce complexity, we consider $mind = 0$ s. The shape of *f(BO)* with $mind > 0$, can be approximated using *f(BO)* with $mind = 0$ and a lower α. In consequence, in out tests, the value of *mind* showed very little influence on performance.

In (1) it is easy to see that, if $\alpha \rightarrow 0$, *f(BO)* tends to the step function, i.e. t_{id} is constant when BO ≥ *minth*. On the other hand, if $\alpha = 1$, *f(BO)* is a linear function. Therefore, the step and the linear functions are particular cases of (1). Moreover, the absence of delay is also a particular case of this function, when $\alpha \rightarrow \infty$. These cases are discussed in Section 5.

3.2 Implementation Issues

An important advantage of the ACK delay control algorithm compared to random or RED-like mechanisms is that it does not need to generate random numbers to compute the discarding probability because of its deterministic operation. This reduces the computational cost of the algorithm, and makes it more feasible for its implementation at the RLC level where the buffering is done in a per-user basis.

The maintenance of a new timer, *id_timer*, does not add too much complexity to the RLC operation, which already handles several timers, e.g. *Poll Timer* and *Status Prohibit Timer* (3GPP, 2005). The RLC can synchronize *id_timer* to the Transmission Time Interval (TTI), which is equivalent to a clock signal with a granularity of 10 ms (similar to that of other RLC timers).

The protocol could be further simplified in terms of implementation with the use of pre-calculated t_{id} values. Therefore, the buffer could be considered as divided in gaps, each one having a corresponding t_{id} value. In our implementation, the number of t_{id} values equals the number of SDUs fitting in the buffer space above *minth*.

Our implementation does not deal with ACK identification, because the services considered to benefit from our algorithm are assumed to be strongly asymmetric, e.g. file downloading and web browsing. Therefore, in our simulations, every upcoming packet is an ACK. However, a simple way to implement IP packets monitoring and ACK filtering is described in (Wu 1999). It is a subject of further research if, given the traffic profile of a 3G user, an ACK filtering mechanism is justified.

4 SIMULATION ENVIRONMENT

The simulation environment for this research has been developed in OMNeT++ (Varga, 2001) and comprises a complete implementation of TCP and RLC protocols. Similar simulators were described in (Rossi, 2003) and (Bestak, 2002). The simulation topology, shown in Fig. 6 consists of one or several TCP sources connected to their respective receivers in the user’s equipment (UE). The end-to-end connection consists of two sections, the wired

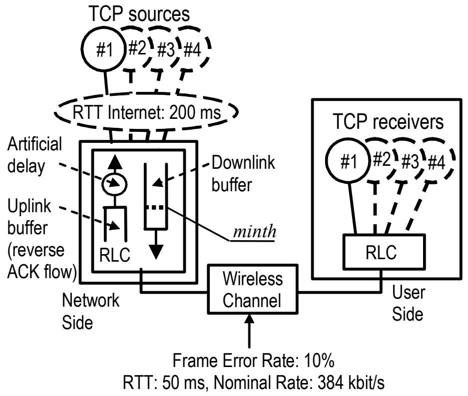


Figure 6: Simulator topology.

network and the radio bearer. The wired network comprises the Internet and the 3G core network. The radio bearer has a round trip time (RTTw) of 50 ms (Holma, 2004) and a nominal transmission rate equal to 384 kbit/s in both directions, representing the bottleneck link, which is the situation expected in most cases (Gurtov, 2004). The wired network is modeled with a 1 Mb/s link with a round trip delay (RTTf) of 200 ms.

The wireless channel generates error bursts according to the model described in (Chockalingam, 1999) where the Doppler frequency, f_d, of the UE determines the average burst length. Lower f_d causes longer bursts of errors. It is usual to employ the normalized Doppler frequency, equal to the product of f_d and the radio frame duration (10 ms).

In order to obtain more realistic results, the error probability is the same in the uplink and in the downlink direction. The frame loss ratio is 10%, a typical UMTS design value (Meyer, 2003).

The simulation results exposed in this paper are obtained averaging 20 runs per sample. Each run is a 60 second download session. The confidence intervals are obtained with a confidence degree of 90% according to a t-student distribution.

The TCP flavour employed is TCP Reno, one of the most extended in the Internet (Gurtov, 2004). RLC and TCP parameter setting is shown in Table 1.

5 PERFORMANCE EVALUATION

In order to disclose the effect of each parameter in end-to-end performance, multiple parameter combinations were tested: The values of α ranged from 0 to 40; for *minth*, the following values were evaluated: 0, 10, 20, 30 and 40 SDUs; *maxd* ranged from 50 ms to 2000 ms. The value of *maxth* equals the size of the RLC buffer, 50 SDUs, enough to prevent buffer overflow in most configurations achieving low packet latency. Fig. 7 shows the average goodput and delay of some selected results (α = 0.5, α = 1, α= 5), considering a single TCP flow over the RLC layer.

Fig. 8 shows the same measures in a scenario with four TCP flows. In these figures, the goodput and delay performance for the conventional RLC drop-tail (DT) operation is also shown as a reference value.

It should be stated that our objective is to improve the goodput while reducing the delay, which, as can be seen on first inspection of figures 7 and 8, is achieved for the single-flow scenario in many configurations. In the multiple-flow scenario, the goodput improvement is negligible but the delay reduction is up to 50%.

The aggressiveness of the algorithm is determined by the combined values of *minth*, *maxd* and α. Given a certain BO value, the algorithm is more aggressive if the ACK rate reduction is higher (higher t_{id}). For an optimum performance, the relation between BO and t_{id} should balance two opposed objectives: avoiding overbuffering and achieving full use of the radio bearer bandwidth. Figures 7 and 8 show that the aggressiveness increases for lower values of *minth* and α, and for higher values of *maxth*. As expected, if the configuration is too aggressive, the goodput decays.

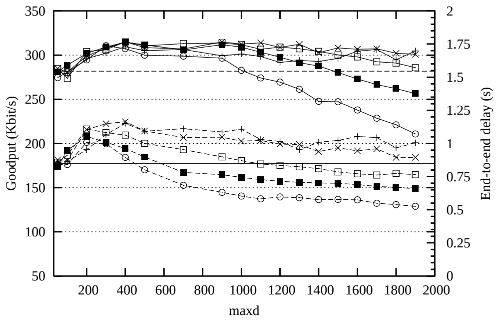


Figure 7 (a): α = 0.5

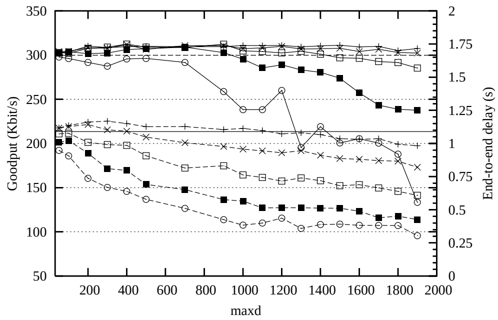


Figure 8 (a): α = 0.5

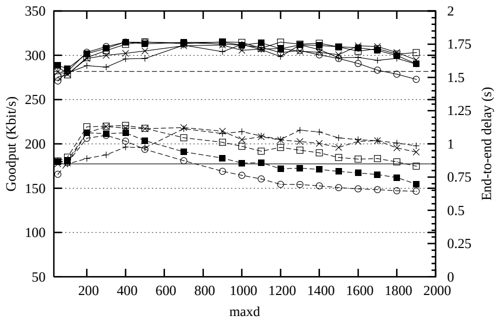


Figure 7 (b): α = 1

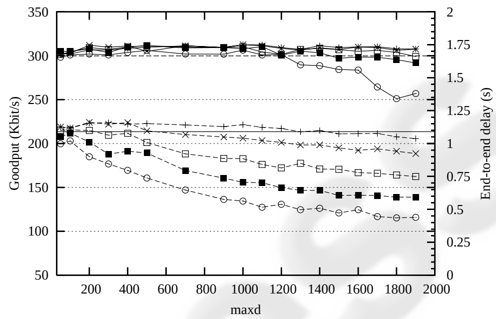


Figure 8 (b): α = 1

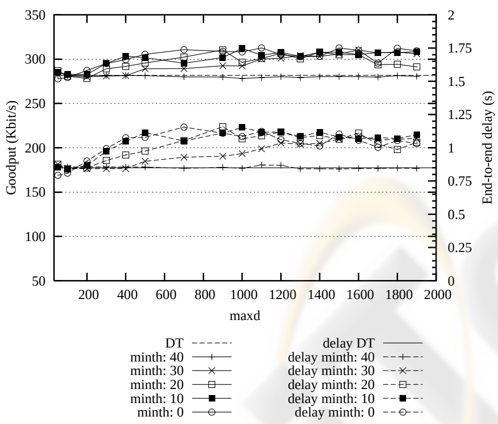


Figure 7 (c). α = 5

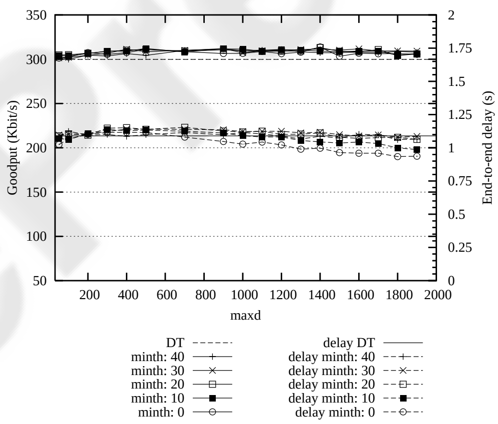


Figure 8 (c). α = 5

Figure 7: Performance of the single-flow scenario.

Figure 8: Performance of the 4-flows scenario

In most TCP implementations, one ACK packet is sent when two TCP segments are received in sequence. In our environment it implies that, in normal operation, two consecutive ACKs are spaced approximately 140 ms. Therefore, if t_{id} is around this value, the effect of the algorithm is imperceptible, and if t_{id} reaches e.g. 240 ms, the ACK rate (and thus the source's rate) is halved. Our simulations show that optimum configurations follow roughly a similar pattern: t_{id} is lower than 240 ms for BO values below 10 SDUs, and for BO values around 20 SDUs, the rate is divided by 3.

Table 2 shows the performance figures of four possible optimum configurations, contrasted with that of RLC without ACK rate control algorithm. Fig. 9 shows the shape of the function $t_{id} = f(BO)$ for these configurations. It can be seen that the rate reduction is similar for certain BO levels, as stated before.

It is clear that optimum configurations require $\alpha < 1$. The effect of α is shown in Fig. 10 (single flow) where *minth* = 10. The performance is worse for $\alpha > 1$ because the rate reduction is concentrated on higher BO values (see Fig. 5). The algorithm tends to react only when the congestion is too high,

Table 2: Performance Figures.

TCP flows	α	*minth (SDUs)*	*maxd (ms)*	Goodput (kbit/s)	Delay (s)
1	Conventional RLC			281.9 ± 8.9	0.85±0.05
1	0	20	500	310.4±5.9	0.84±0.02
1	0.2	10	500	308.2 ± 6.2	0.78±0.03
1	0.5	10	700	305.9±5.6	0.78±0.03
1	1	0	1000	311.2±2.5	0.76±0.01
4	Conventional RLC			299.9 ± 3.9	1.09±0.02
4	0	20	500	302.7±7.9	0.72±0.02
4	0.2	10	500	301.4. ± 7.9	0.59±0.03
4	0.5	10	700	309±5.6	0.65±0.02
4	1	0	1000	306.3±6.9	0.56±0.03

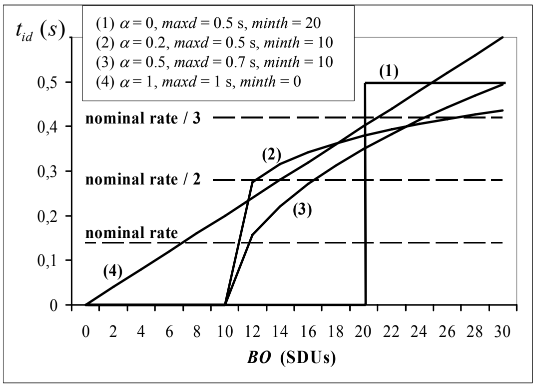


Figure 9: *f(BO)* of the selected configurations.

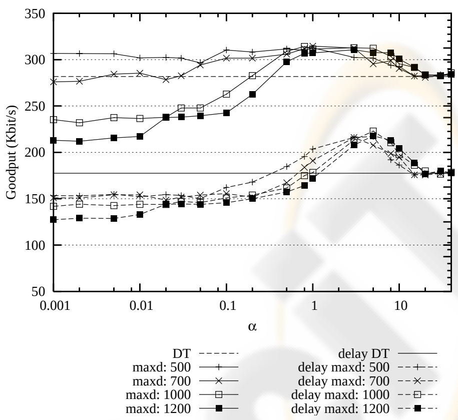
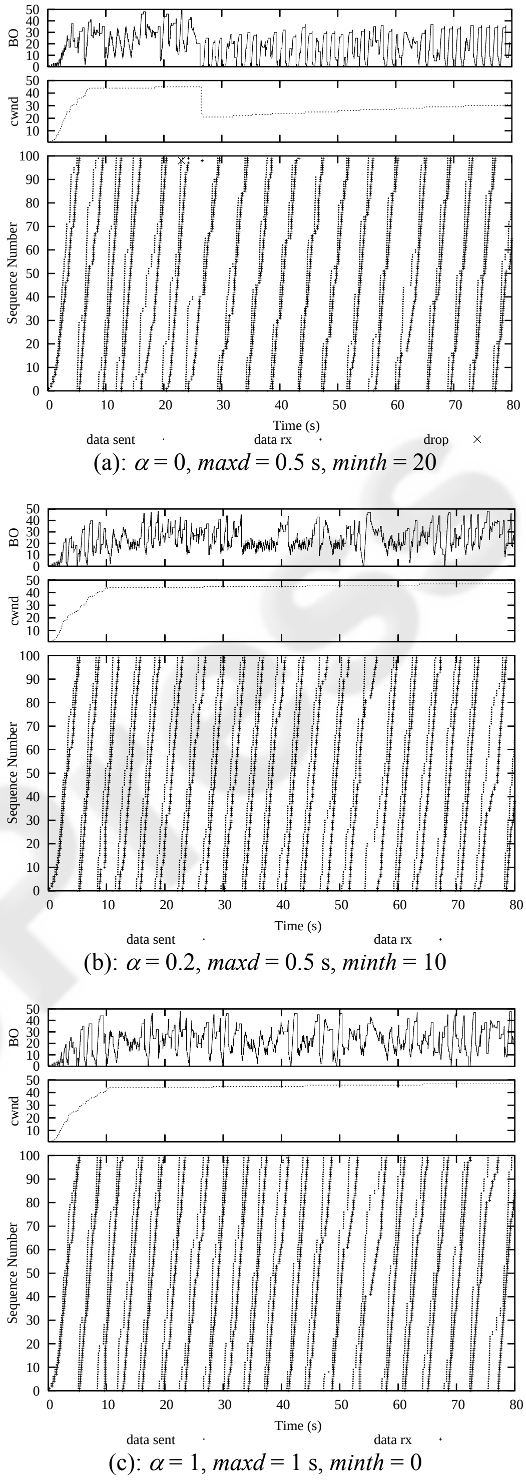


Figure 10: Effect of α. Single-flow scenario.

which is less effective preventing buffer overflow. In fact, for $\alpha >> 1$, the algorithm effect vanishes ($t_{id} = 0$).

Finally, Fig. 11 shows traces of three of the selected configurations ($\alpha = 0$, $\alpha = 0.2$ and $\alpha = 1$). The traces provide further insight on the algorithm performance. In general, buffer overflow is avoided, and the overall behaviour is better than that shown in Fig. 2. For $\alpha = 0.2$, we can see a single packet loss, because *minth* is higher in this scenario. The $\alpha = 0$ and $\alpha = 1$ cases show a noticeable oscillation in the buffer occupancy process. For $\alpha = 0$ it is caused by the abrupt increment on t_{id}. For $\alpha = 1$, the increment



(a): $\alpha = 0$, *maxd* = 0.5 s, *minth* = 20

(b): $\alpha = 0.2$, *maxd* = 0.5 s, *minth* = 10

(c): $\alpha = 1$, *maxd* = 1 s, *minth* = 0

Figure 11: TCP traces with ACK rate control.

is gradual but relatively slow given the transmission delay of the control loop. The source reacts "a bit late" and "a bit too stark" causing an apparent instability. The stability of the algorithm from a control system point of view is an issue for further research. For $0 < \alpha < 1$, the algorithm shows lower BO oscillations, which justifies the generalized delay computation approach proposed in this paper.

6 CONCLUSIONS

Our proposed ACK rate control algorithm improves end-to-end goodput and delay performance when RLC supports a single or multiple TCP flows. The algorithm performs equally well in the slow start and in the congestion avoidance phase of TCP connections. In contrast to similar proposals done in the field of satellite links, we consider a more general way of calculating the inter-departure time of ACKs. Moreover, we provide a detailed algorithm that clarifies the operation and the implementation.

By means of extensive simulation tests we provide insight about the influence of each parameter, which let us find multiple possible configurations improving overall performance. The key issue in selecting an accurate parameter setting is to tune the rate reduction of the algorithm according to the congestion level of the link, reflected in the buffer occupancy. Configurations showing similar performance figures (goodput and delay) presented a similar ACK rate reduction pattern for low and high BO levels. However, it was found that the value of α has a noticeable impact on the oscillatory behavior of the BO process.

An important advantage of our algorithm compared to random or RED-like mechanisms is its deterministic operation, which reduces the computational cost of the algorithm, and makes it more feasible for its implementation at the RLC level where the buffering is done in a per-user basis.

Some additional advantages of our algorithm are that it does not require changes in 3G specifications nor in TCP itself, and preserves TCP end-to-end semantics.

REFERENCES

3GPP, 2005. "Radio Link Control (RLC) protocol specification", v. 6.4.0., TS 25.322.

H. Inamura et al., 2003. "TCP over Second (2.5G) and Third (3G) Generation Wireless Networks" IETF RFC 3481.

J. J. Alcaraz, F. Cerdan and J. García-Haro, 2006. "Optimizing TCP and RLC Interaction in the UMTS Radio Access Network", *IEEE Network,* vol 20, no. 2, pp. 56 - 64.

Jing Wu et al., 1999. "ACK Delay Control for Improving TCP Throughput over Satellite Links", Proc. IEEE ICON'99, pp. 303- 312.

M. Rossi, L. Scaranari and M. Zorzi, 2003, "On the UMTS RLC Parameters Setting and their Impact on Higher Layers Performance", in *Proc. IEEE 57th VTC*, vol. 3, pp. 1827- 32.

R. Chakravorty, A. Clark and I. Pratt, 2005. "Optimizing Web Delivery over Wireless Links: Design, Implementation and Experiencies", *IEEE J. Select. Areas Commun.*, vol. 23, no. 2, pp. 402- 416.

A. Gurtov, S. Floyd, 2004. "Modeling Wireless Links for Transport Protocols", ACM SIGCOMM Computer Communication Review, vol. 34, no. 2, pp. 85-96.

R. Bestak, P. Godlewski and P. Martins, 2002. "RLC Buffer Occupancy when Using a TCP Connection over UMTS", in Proc. IEEE PIMRC'02, vol. 3. pp. 1161-65.

A. Varga, 2001. "The OMNeT++ Discrete Event Simulation System", in *Proc. European Simulation Multiconference*.

H. Holma, A. Toskala, 2004. *WCDMA for UMTS: Radio Access for Third Generation Mobile Communications, Third Edition*, Wiley.

A. Chockalingam and M. Zorzi, 1999. "Wireless TCP Performance with Link Layer FEC/ARQ", in *Proc. IEEE ICC'99*, pp. 1212-16.

M. Meyer, J. Sachs and M. Holzke, 2003. "Performance Evaluation of a TCP Proxy in WCDMA Networks", *IEEE Wireless Communication*, pp.70-79.