Investigating Interactions between ARQ Mechanisms and TCP over Wireless Links

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Abstract: **In this paper we investigate interactions between TCP and wireless ARQ mechanisms. The aim is to understand what is the best configuration of the wireless link protocol in order to guarantee TCP performance, seemingly a controversial issue. Interactions between TCP and different link layer mechanisms are evaluated by means of an analytic model, that reproduces a generic selective repeat ARQ protocol (widely used in the current wireless environments) and the TCP behavior in a wired-cum-wireless network scenario. A numerical investigation is carried out in a specific case study (TCP over 3G radio access) by means of simulations collected with a very detailed UMTS-TDD simulator based on *ns*. Our main finding is that fully reliable ARQ protocols are the best choices from the TCP perspective; in fact, whereas a residual packet loss left over by not fully reliable ARQ protocols may not degrade appreciably TCP throughput performance as long as it is a fraction of the overall end-to-end TCP packet loss, no apparent performance advantages (e.g. energy savings) come from limiting the number of retransmission attempts at the wireless link layer.**

1 Introduction

In the last years, the issues regarding the behavior of TCP over wireless links have been extensively addressed in the networking research community, both because TCP represents the most widespread transport protocol in the current Internet (about 90 percent of the overall IP traffic is carried by TCP according to recent experimental measurements [1]) and because Internet access through wireless technologies is rapidly taking up. Recent and thorough reviews of general problems and possible solutions regarding the specific "TCP over wireless" topic can be found for example in [2], [3] and [4].

The specific aim of the present work is to investigate the impact of wireless link layer ARQ mechanisms on TCP performance through both analytic methods and simulations. In particular, we are interested in studying how the persistence degree of the radio link layer affects end-to-end TCP performance and wireless link utilization.

In the current literature the link layer design appears to be quite a controversial issue. In fact, with respect to the three commonly considered solutions - persistent and fully reliable ARQ (i.e., unlimited number of transmission attempts), semi-reliable ARQ (i.e., with a fixed limit on the number of transmissions), and ARQ with a dynamically adaptive limit - there is no agreement on which of them is the best in order to maximize TCP performance (see the recent RFC 3366 [5] for a discussion on the possible ARQ strategies in IP networks).

Most of the works on TCP over wireless implicitly assume that wireless links are not fully shielded from "random" errors caused by the non-ideality of the radio channel (i.e., they assume an unreliable or semi-reliable link layer); according to a number of works (e.g. [6] and [7]), the advantage of a not fully persistent ARQ is that it would reduce spurious TCP retransmission timeouts and packet reordering with a straightforward benefit for TCP. Conversely, another "school of thought" (e.g., [8], [9] and [10]) asserts that the wireless link layer should be fully reliable to preserve TCP performance. This choice guarantees that any loss detected by TCP is due to network congestion, so it fully respects TCP syntax.

As for the adaptive solutions, in [11] the authors propose a link layer algorithm that adapts the maximum number of link layer data units transmission attempts according to a target loss rate, used as parameter to describe a desired QoS for a TCP connection. They show that TCP performance can be improved by adapting the maximum number of transmissions to the desired QoS rather than leaving it constant.

In this work we evaluate the impact of different ARQ policies on TCP behavior by adopting an end-to-end point of view including a wireless link plus a non-ideal fixed network; we use a cross-layer approach that takes explicitly into account both TCP and wireless link layer dynamics.

Our aim is to thoroughly understand the best ARQ policy on the wireless link for TCP data transfer support and the reason why.

Our finding is that fully reliable ARQ protocols are the best choices from the TCP perspective; a residual packet loss left over by not fully reliable ARQ protocols does not degrade appreciably TCP delay and throughput performance as long as it is a fraction of the overall end-to-end packet loss. However it appears that the limitation of the number of retransmissions does not bring any performance advantages by itself (e.g. energy savings).

Our results are based on a quite general model of a selective repeat ARQ and of TCP connections. This model gives us an analytical tool that we validate in a specific and realistic case study (TCP over UMTS radio access) by means of a very detailed *ns* [12] based UMTS-TDD simulator.

The rest of the paper is structured as follows. Section 2 briefly reviews the recent related research works. In Section 3 we depict the network reference scenario. In Sections 4 we describe the ARQ link layer model and the TCP model. In Section 5 we discuss analytic and simulation results. Finally we give the conclusions and some hints to further work.

[1] This work was partially supported by the Italian FIRB Project PRIMO.

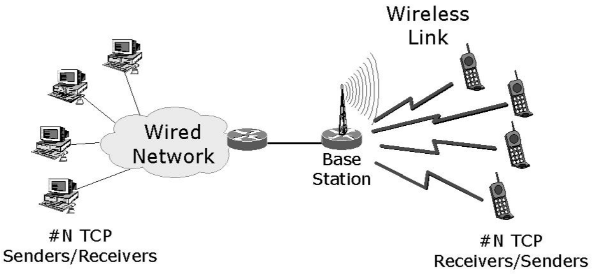

Figure 1: Network scenario.

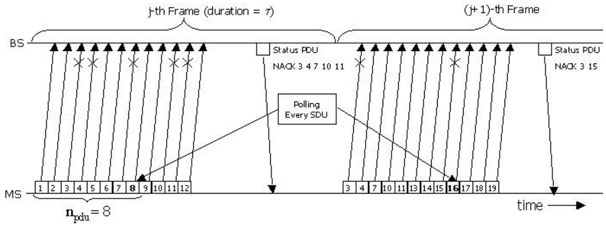

Figure 2: Selective Repeat ARQ protocol.

2 Related Works

Quite a number of recent works analyze interactions between TCP and wireless ARQ mechanisms, testifying the current interest for this topic. In this section we briefly review those researches in order to point out the differences with the present work.
In [13] a simulative study about wireless TCP with hybrid FEC/ARQ is presented and it is shown that a sufficiently (but not fully) persistent ARQ improves TCP performance.
In [14] and [15] analytical studies, based on the well-known TCP model presented in [16], are carried out on TCP with ARQ in a UMTS network environment. As for ARQ, only the fully persistent policy of a Go-back-N ARQ is analyzed; moreover, it is not studied the effect of different ARQ persistence degrees on TCP performance and no simulation validations are presented.
In [17] a simulative evaluation of TCP behavior in a WLAN environment implementing Stop&Wait ARQ is carried out. The main conclusions are similar to the ones presented in this paper; however, here a different network scenario is considered, an analytic study is carried out to support our statements and some factors (e.g., ARQ dynamics) are analyzed in a greater detail.
In [18] a mixed analytical/simulation study analyzes interactions between hybrid FEC/ARQ link layer mechanisms and TCP. As in the present work persistent TCP flows are analyzed and a selective repeat ARQ is taken in consideration. In that work, the analysis is still based on the model proposed in [16]. An assumption of independence on packet error process in the wireless link is done to model the ARQ dynamics.
To the best of our knowledge, the work presented in this paper is original under the respect that it accounts all the following factors both via analytic models and case study simulations: i) a correlated error generating radio channel; ii) a selective repeat ARQ in the wireless link.

3 Reference Network Scenario

The reference network scenario considered in this work is depicted in Figure 1. A number N of TCP senders (receivers) placed on mobile nodes are connected to N TCP receivers (senders) placed within the wired section of the Internet. We use the TCP Reno version as TCP implementation since today it is the most popular version of TCP. TCP Reno adopts an end-to-end closed-loop adaptive window congestion control, whose working details can be found in [19].
TCP sources are assumed to be "greedy", always ready to send new data (persistent TCP connection model). The rapid development of "peer-to-peer" applications and high-speed access links, that allow the exchange of a large amount of data among users, makes this hypothesis realistic. Besides, already now, according to [1], persistent streams carry a high proportion (50 or 60 percent) of the total amount of traffic on the Internet.
A number of earlier experimental works (see for example [20] and [21]) has shown that Internet is far from being ideal, given that congestion packet loss events are quite frequent (even 1 to 10 percent). In order to reproduce this non-ideality, TCP packets are dropped in the wired portion of our network scenario with probability p_f according to a random process assumed independent of TCP connection status[2]. The wired network also introduces a round trip delay T_f, assumed fixed.
As for the radio interface we assume a centralized radio access handled by a Base Station (BS). Mobile Stations (MSs) communicate with wired nodes through the BS, connected to the wired network with a dedicated link. Time is divided into frames of duration τ and every frame provides capacity for both uplink and downlink.
The Link Layer (LL) implements a selective repeat ARQ mechanism for error recovery. LL Service Data Units (SDUs) are accepted from the upper layers[3] and, once segmented into n_{pdu} Packet Data Units (PDUs), are placed in the LL transmission buffer, assumed infitite.

PDUs are transmitted when the MAC layer indicates that there are available transmission resources.
On reception of a PDU, after decoding, the physical layer performs error detection, and passes the result of the check to the LL, together with data. When all the n_{pdu} PDUs belonging to a SDU are received correctly, the SDU is reassembled and delivered to the upper layer preserving the original SDU sequence.
Every SDU, the transmitter entity polls the receiver for a status report. According to the Selective Repeat ARQ, the receiver sends a PDU containing the status report (Status PDU) indicating the PDUs received correctly and the ones to be retransmitted. When the Status PDU is received, PDUs buffered in the retransmission buffer of

[2] This hypothesis is reasonable when the traffic generated by the target TCP connections is a negligible fraction of the overall traffic crossing the TCP connection path in the considered wired network.

[3] A link layer SDU corresponds to an IP packet carrying a TCP segment.

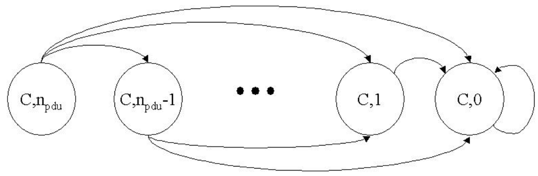

Figure 3: Markov chain describing ARQ protocol.

the sender entity are deleted or retransmitted according to the status report. Every PDU can be transmitted at most M_t times. When this number is reached, the transmitter entity discards that PDU and all PDUs belonging to the same SDU. Given the limited value of M_t, a SDU loss probability p_w is present in the wireless link.
An example of the ARQ mechanism when the number of PDUs in a SDU is 8 is depicted in Figure 2.

4 Analytic Models

4.1 Selective Repeat ARQ Model

In this section we describe the discrete Markov chain modeling the wireless link selective repeat ARQ of Section 3.
The model will be used in this work for two aims. On the one hand, its results will be integrated in the TCP model described in Section 4.2. On the other hand, in Section 5, we use it to define a cross-layer adaptive ARQ algorithm.
The radio channel is modeled as a Gilbert-Elliot channel with two states (GOOD and BAD) [22], widely used in literature to characterize wireless transmission medium (see for example [23]). The GOOD state is the one with very low PDU error probability p_G, whereas the BAD state is the one with high PDU error probability p_B.
The discrete Markov chain, with time unit τ corresponding to a radio frame, is characterized by the transition probabilities g and b, that represent the conditional probabilities that the channel state remains respectively in the state GOOD and BAD during the sampling time τ provided it was in the same state in the previous sampling interval. The b and g parameters can be estimated by direct field measurements on the basis of the mean time in the BAD state $\tau/(1-b)$ and of the mean time in the GOOD state $\tau/(1-g)$ [23].
The selective repeat ARQ protocol is modeled with a discrete time Markov chain which reproduces the transmission of a single and isolated SDU under the following assumptions: i) the SDU is segmented at the link layer in n_{pdu} PDUs of equal size; ii) all the PDUs belonging to the SDU are transmitted in a single time interval of duration τ; iii) the ack/nack is delivered without errors by the end of the time interval; iv) all the PDUs which require retransmission are sent again in the next time interval, and so on; v) the error events of different PDUs conditional on the state of the channel are independent of one another.
A chain state is defined by the vector $\overline{s} = (C, X)$, where:

- $C \in \{\mathrm{GOOD}, \mathrm{BAD}\}$ is the Gilbert-Elliot channel state; in the following the GOOD and the BAD state are referred often as G and B respectively.
- $X \in [0, n_{pdu}]$ is the number of PDUs, belonging to the considered SDU, not successfully transmitted yet.

In Figure 3 the Markov chain for a single state channel C is shown. A transition from the state (C, k) to the state (C, k') represents the event that $k - k'$ PDUs are transmitted successfully and k' PDUs are transmitted with error; only transitions from k to k' with $k' \leq k$ are allowed. The transition probability $b_C(k, k')$ is the probability to fail k' transmissions out of k:

$$b_C(k, k') = \binom{k}{k'} p_C^{k'} \cdot (1-p_C)^{k-k'}$$

where p_C is the error probability in the state C. The transition matrix $\mathbf{T}_C$ is:

$$\mathbf{T}_C = \begin{bmatrix} 1 & 0 & \dots & 0 \\ b_C(1,0) & b_C(1,1) & \dots & 0 \\ \vdots & \vdots & \ddots & \vdots \\ b_C(n_{pdu},0) & b_C(n_{pdu},1) & \dots & b_C(n_{pdu},n_{pdu}) \end{bmatrix}$$

Let $\mathbf{T}_C^*$ be a matrix obtained from $\mathbf{T}_C$ by removing the first row and the first column, the matrix $\mathbf{Q}$ is defined as:

$$\mathbf{Q} = \begin{bmatrix} g \cdot \mathbf{T}_G^* & (1-g) \cdot \mathbf{T}_G^* \\ (1-b) \cdot \mathbf{T}_B^* & b \cdot \mathbf{T}_B^* \end{bmatrix} \quad (1)$$

$\mathbf{Q}$ is the one step transition probability matrix of a transient Markov chain, whose absorption time η gives the number of frames needed to complete the delivery of the PDUs making up an SDU. The residual SDU loss probability, provided the number of transmission attempts per PDU is n, is $F(n) = Pr\{\eta > n\} = \boldsymbol{\pi}\mathbf{Q}^n\mathbf{e}$ for $n \geq 0$ where $\mathbf{e} = [1 \dots 1]^T$ and $\boldsymbol{\pi}$ is the $2n_{pdu}$ components row vector of the initial probabilities; $\boldsymbol{\pi}$ is equal to $[0, 0, \dots \pi_G, 0, 0, \dots \pi_B]$, where π_G and π_B are assumed to be the steady-state probabilities of the Gilbert-Elliot Markov chain.
The expected value of η is the mean number of transmissions per SDU and can be evaluated as follows:

$$E[\eta] = \sum_{n=0}^{\infty} nPr\{\eta = n\} = \sum_{n=0}^{\infty} F(n) = \boldsymbol{\pi}(\mathbf{I}-\mathbf{Q})^{-1}\mathbf{e} \quad (2)$$

The mean number of transmissions per SDU, when the number of transmissions per PDU is limited to M_t, is:

$$E[\eta | n \leq M_t] = \sum_{n=0}^{M_t-1} F(n) = \boldsymbol{\pi}(\mathbf{I}-\mathbf{Q}^{M_t})(\mathbf{I}-\mathbf{Q})^{-1}\mathbf{e} \quad (3)$$

Using equations (2) and (3) and considering the transmission process of a SDU consisting of a single PDU (i.e. $n_{pdu} = 1$), we can calculate the mean number of transmissions per PDU, $\mathcal{N}_\infty$ and $\mathcal{N}_{M_t}$, respectively in the case of unlimited transmissions and in the case of a limited number of transmissions.
In [24] an approximate expression for the residual SDU loss probability $F(n)$ has been derived:

$$\hat{F}(n) = \left[1 - [\pi_G(1-p_G)^{n_{pdu}} + \pi_B(1-p_B)^{n_{pdu}}]\right]\lambda_{11}^{n-1} \quad (4)$$

for $n \geq 1$, where λ_{11} is the maximum eigenvalue of $\mathbf{Q}$. Equation (4) (evaluated for $n = M_t$) can be easily inverted in order to obtain an explicit expression for the maximum number of transmissions per PDU M_t given the residual SDU loss probability required. This will be used in Section 5 to define an adaptive ARQ algorithm. The validation of the model has been presented in [24].

4.2 Analytic Model for TCP Behavior in Wireless Environments

In this section we discuss the TCP analytic model able to reproduce the stationary behavior of TCP.

The following hypotheses are assumed: i) TCP has always data to send; ii) TCP Slow Start is neglected; iii) TCP packet losses are assumed to be independent; iv) the TCP receiver does not limit the value of the TCP sender congestion window, according to [26]; v) TCP only reveals packet loss by using Fast retransmit/Fast recovery mechanisms; vi) the ACK packets are always delivered to TCP sender.

The model is based on a discrete Markov chain. Although the Markov chain approach has been already used in earlier works (e.g., [27] [28]), with respect to them, we also include the multiple Fast Retransmit events due to multiple loss within a single TCP transmission window[4] and the effects of queueing delay within the LL buffer.

The discrete Markov chain describing the stationary dynamics of TCP is depicted in Figure 4. Every state is characterized by the current TCP congestion window $W = 1, 2, ...$ and its time unit is the time t_W spent for transmitting a window W of TCP packets. The transition probability q_{ij} from the state i to the state j is:

$$q_{ij} = \begin{cases} (1-\nu)^i & \text{if } j = i+1 \\ \binom{i}{c}\nu^c(1-\nu)^{i-c} & \text{if } j = \left\lfloor \frac{i}{2^c} \right\rfloor \text{ for } \log_2(i) > c \geq 1 \\ 1 - \sum_{k \neq 1} q_{ik} & \text{if } j = 1 \\ 0 & \text{otherwise} \end{cases} \tag{5}$$

where $\nu = 1 - (1 - p_f)(1 - p_w)$ is the TCP packet loss probability due to wireless impairments and to wired network congestion supposing them mutually independent. The packet loss probability in the wireless link p_w is $F(M_t)$ (as derived in the earlier section) and depends on the channel transitions and error probabilities, the SDU size and the ARQ persistence degree.

The first item of (5) represents the probability of having no packet losses when the window size is i: in this case TCP increments the congestion window by one packet. The second item represents the probability of having multiple Fast Retransmit: if there are k packet losses within a time unit, we assume TCP performs k times the Fast Retransmit algorithm so to halve the window k times; we make the rough simplification that the multiple Fast Retransmit is performed in a single time unit[5]. This simplification is supported by the more than satisfactory agreement with simulations as shown in Section 5.1. The third item in (5) is derived from the condition that $\sum_j q_{ij} = 1, \forall i$.

[4] From simulations we observed this is a quite significant occurrence.

[5] Actually multiple packet losses can produce either multiple Fast Retransmit or Time Out events. Moreover, the events caused by a multiple loss last in general more than a single time unit.

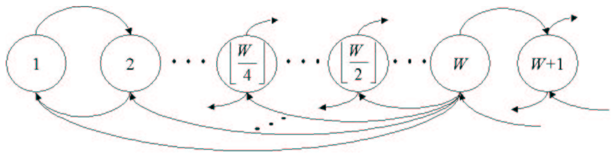

Figure 4: Markov chain describing TCP behavior.

The time t_W spent in the state W is variable and state dependent; according to a fluid approximation, used in a number of TCP related works, we have:

$$t_W = \begin{cases} \frac{W}{C_d} & \text{if } W \geq C_d \cdot E[T] \\ E[T] & \text{otherwise} \end{cases} \tag{6}$$

where C_d is the capacity available for the TCP connection at link layer and $E[T]$ is the average end-to-end round trip delay computed excluding queueing delay experienced within the LL buffer.

In (6) the first item represents the time necessary to transmit W packets when the capacity available is saturated; the second item represents the case in which the packets within a window of size W are transmitted without suffering queuing delays since there is available bandwidth. Assuming the bottleneck in the wireless link, the available capacity C_d is the radio interface capacity C assigned to the TCP connection divided by $\mathcal{N}_{M_t}$, i.e. the mean number of transmission attempts per PDU given the maximum number of transmissions at the LL is M_t. $E[T]$ is the sum of the average round trip time in the wired network $E[T_f]$ and the TCP packet delay spent to cross the radio interface $E[T_w]$:

$$E[T] = E[T_f] + E[T_w] = T_f + \tau \cdot E[\eta | n \leq M_t] + \tau \tag{7}$$

where T_f is the round trip time of the wired network, τ is a frame interval and $E[\eta | n \leq M_t]$ is the average number of frames necessary to send correctly a TCP packet. The last term τ takes into account the transmission delay of the TCP ACK on the wireless link, assuming that it is sent without errors.

According to the Little's Law, the TCP throughput is the ratio between the average window and the average value of t_W:

$$Th_B = \frac{E[W]}{E[t_W]} = \frac{\sum_i i \cdot \pi_i}{\sum_i t_i \cdot \pi_i} \tag{8}$$

where π_i are the steady state probabilities of the Markov chain described above.

Summing up, the TCP analytic model depends on the following input parameters: packet loss probability and average round trip time in the wired network, the physical radio interface capacity, the maximum number of transmissions, TCP packet size and the set of parameters that characterize the radio channel.

5 Results and Discussion

In this section we discuss the results provided by either the analytic models and the simulations. In Section 5.1,

we investigate, by using the TCP model, how the presence of wireless impairment affects TCP performance. In Section 5.2, we examine interactions between different persistence degree ARQ policies and TCP.
Numerical investigation is carried out with reference to UMTS-TDD radio access interface.
Since a detailed presentation of the UMTS-TDD system is beyond the purpose of this paper, here we briefly describe the UMTS-TDD radio access and the acknowledged mode (AM) of the UMTS link layer (named Radio Link Control (RLC) in the 3GPP specifications [25]) with the options used in our simulations.
The UMTS-TDD radio interface implements a combined time division and code division multiple access (TD-CDMA). Time is divided into frames of duration $\tau = 10$ ms and every frame is slotted in 15 time slots. Within each time slot a set of variable spreading factor orthogonal codes are defined to spread QPSK modulated data symbols.
The AM LL uses a selective repeat ARQ mechanism for error recovery, similar to that described in Section 3. With respect to that, the AM LL implements a number of timers to avoid deadlock states and a sliding window mechanism that regulates the transmission of PDUs.
Many parameters of the selective repeat ARQ protocol can be configured; among them the size of the transmission window, the size of Data PDU and Status PDU, the duration of the timers and the maximum number M_t of transmissions per each PDU. The LL has been configured for in-sequence delivery.
All the simulations have been carried out with a UMTS-TDD module for *ns* [6] that simulates the TD-CDMA radio interface. In the simulations, TCP sources are placed on the mobile nodes and the following parameters are left unchanged: TCP packet size is 1000 bytes, the LL payload size is 40 bytes (so that the number n_{pdu} is 25), the error probabilities in the GOOD and BAD states are respectively $p_G = 0.01$ and $p_B = 0.9$, the bottleneck link is consistently assumed to be the wireless one.

5.1 What is the packet loss degree tolerable by TCP in the wireless link?

In a related work [29] it has been verified by means of simulations that, when the packet loss probability experienced by the TCP connection in the wireless link is one order of magnitude lower than the one experienced in the wired network, TCP performance is essentially not affected by the wireless impairment. Here, we substantiate that result by applying the TCP model defined in Section 4.2 and carrying out further detailed simulations.
We define the coefficient α as the ratio between the packet loss probability in the wireless link p_w and the packet loss probability in the wired network p_f:

$$\alpha = \frac{p_w}{p_f} \tag{9}$$

In Figure 5 the normalized TCP throughput, defined as the ratio between TCP throughput and the maximum reachable one (C_d), is shown as a function of α for $p_f = 0.01$ and $p_f = 0.1$.

[6] available via http://net.infocom.uniroma1.it

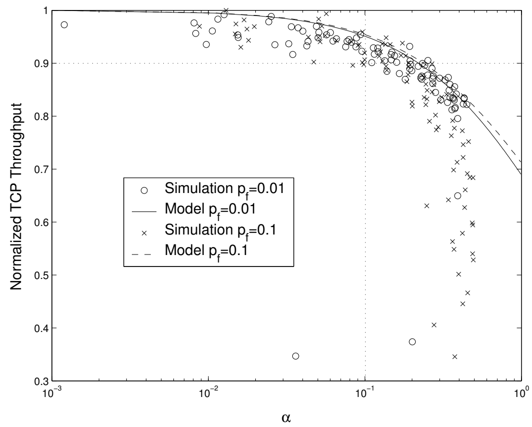

Figure 5: Normalized TCP Throughput *vs* packet loss degree, α, in the wireless link (RTT=1.1s).

These data indicate that the TCP throughput degradation is essentially negligible provided $\alpha \ll 1$ (e.g. $\alpha < 0.1$), i.e. some residual loss in the radio link is tolerable if it is a fraction of the fixed network segment loss. This behavior is captured by both model and simulations.
As for the TCP model accuracy against simulation results, it is quite poor only for very high values of α (close to 20% as appears in Figure 5). For such loss rates, TCP time-out events are no more negligible as assumed in our model.

5.2 Effects of ARQ persistence on TCP performance

In the previous section we have shown how a certain degree of packet losses in the wireless link does not reduce TCP performance significantly. This degree depends on both wired network packet loss (i.e. congestion) and wireless channel transmission quality and can be represented by the coefficient α.
Now, we describe an adaptive selective repeat ARQ mechanism similar to the one proposed in [11] and alternative to the traditional ones (that have either limited or unlimited but fixed retransmission attempts), able to adjust dynamically the ARQ persistence degree on the basis of the current network condition, so to impose a desired residual packet loss rate on the wireless link; such target packet loss rate must be chosen as not to limit TCP performance.
By following a cross-layer approach, the adaptive mechanism is based on the exchange of information between the wireless link layer and the TCP entity (either sender or receiver) placed on the mobile node.
The algorithm adapts the maximum number of ARQ transmissions M_t by estimating the channel status and the TCP end-to-end packet loss probability.
End-to-end packet loss probability p and wireless packet loss probability p_w are estimated by means of classical moving average runtime estimates respectively by the TCP entity (either sender or receiver) and by the LL entity both placed on the mobile node. Then the algorithm adapts the maximum number of ARQ transmissions so as to obtain a *target* link layer packet loss probability $\hat{p}_w$

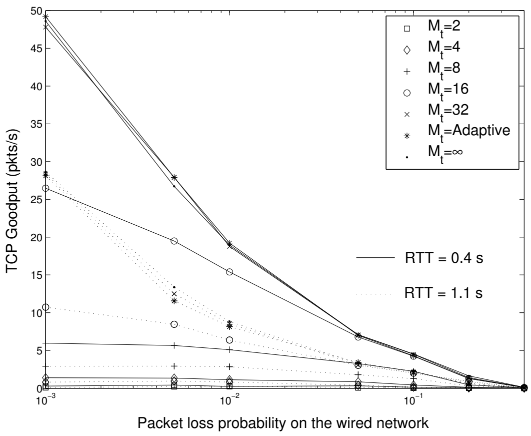

Figure 6: TCP Goodput varying p_f for 1 TCP Source.

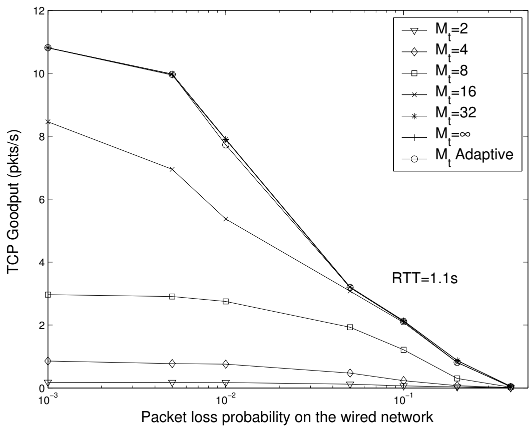

Figure 7: TCP Goodput varying p_f for 8 TCP Sources.

equal to $\alpha \cdot \left[1 - \frac{1-\bar{p}}{1-\bar{p}_w}\right]$, where $\bar{p}$ and $\bar{p}_w$ are the estimates of p and p_w respectively. Given the *target* error probability $\hat{p}_w$, we are able to compute the maximum number of transmission attempts by using the expression (4).

In the following we report the results of the simulations carried out to evaluate the three possible ARQ solutions (limited number of transmissions, unlimited number of transmissions, adaptive number of transmissions).

To point out the effect of limiting the maximum number of transmission attempts, the TCP goodput as function of the wired packet loss probability p_f is shown in Figure 6 for 1 TCP source and in Figure 7 for 8 concurrent TCP sources for different values of M_t, for M_t unlimited and for M_t adaptive according to the described algorithm. In these simulations the round trip delay on the wired section of the network is 0.4 s and 1.1 s, $\alpha = 0.1$, $b = 0.9$, $g = 0.9$.

As shown in the figures, the maximum TCP goodput is always obtained either with unlimited M_t or by adapting M_t according to the described algorithm. We observe a degradation in TCP performance when M_t is limited independently of the loss experienced by TCP in the wired network. By using the adaptive algorithm we introduce residual loss probability on the wireless link ($\alpha = 0.1$), nevertheless TCP performance is not affected.

As pointed out in the figures, a well-designed adaptive

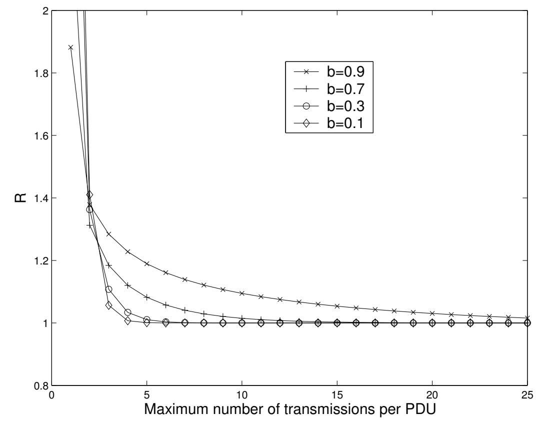

Figure 8: $\mathcal{R}$ varying the maximum number of transmission per PDU.

ARQ policy does not reduce TCP performance with respect to a fully persistent ARQ strategy, although it adds packet losses on the wireless link.

In the following we verify that (and explain why), although an adaptive ARQ protocol does not degrade TCP throughput, it does not bring advantages in terms of wireless link layer energy and bandwidth consumption, which is one of the main issues in a mobile environment. To study the energy consumption when using an ARQ with limited transmissions per PDU and when using unlimited transmissions we define

$$\mathcal{R} = \frac{\mathcal{N}_{M_t}/(1-p_w) - 1}{\mathcal{N}_\infty - 1}$$

that represents the ratio between the energy surplus consumed by retransmissions when M_t is limited and the energy surplus due to retransmissions with M_t unlimited considering that the packet discarded in the radio interface with probability p_w are retransmitted by TCP; this ratio is depicted in Figure 8 varying M_t for different value of b and $g = 0.9$. It is possible to note that, because of the TCP retransmission techniques, the energy consumption in the case of limited transmissions is always higher or equal than with unlimited transmissions. Summing up, on the one hand, for TCP performance it is not strictly necessary to provide unlimited transmissions in the wireless link (in fact, an adaptive algorithm achieves equal performance by limiting the transmission attempts). On the other hand, there is a loss in wireless efficiency (bandwidth and energy consumption) if we limit the number of transmissions. Hence, the best choice for TCP and the simplest from a design point of view is to adopt a fully reliable link layer protocol.

6 Conclusions

In this paper we investigated the interactions between ARQ mechanisms and TCP behavior. The main contribute of the work is to give some precise indications on the design of the link layer protocol for mobile computing applications. We have shown that there are no practically useful alternatives to the adoption of a fully persistent ARQ protocol on the wireless link to guarantee TCP performance, even though essentially equivalent TCP performance can be obtained also with non fully reliable but adaptive ARQ schemes. In fact, to limit retransmission attempts at the data link layer brings some disadvantages in terms of wireless energy and bandwidth savings.

To achieve such conclusions we developed an accurate analytic model for a generic selective repeat ARQ protocol and a TCP model in a wired-cum-wireless network scenario. Moreover we validated our results by means of simulation environment.

Further work is required to verify if our conclusions are still valid when: i) resource sharing in the radio interface (MAC layer) is adaptive to the wireless link transmission quality; ii) TCP connections are non-persistent (short-lived TCP flows).

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