Making retransmission delays in wireless links friendlier to TCP

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***Abstract*— Heterogeneous communication networks with their variety of application demands, uncertain time-varying traffic load, and mixture of wired and wireless links pose several challenging problem in modeling and control. In this paper we focus on the packet delay, which is a particularly important variable for efficient end-to-end congestion control. In particular, we study the delay effects of radio links which use power control and link-layer retransmissions.**

Link-layer retransmissions induce delays which do not conform to the assumptions on which the transport protocol is based. This causes undesired TCP control actions which reduce throughput. A link layer solution based on adding carefully selected delays to certain packets is proposed to counteract this problem. All information needed for this is available locally at the link.

I. Introduction

Congestion control is one of the key components that has enabled the dramatic growth of the Internet. The original idea [1] was to adjust the transmission rate based on the loss probability. The first implementation of this mechanism, denoted TCP Tahoe, was later refined into TCP Reno. This algorithm (together with some of its siblings) is now the dominating transport protocol on the Internet. The throughput and delay experienced by individual users depends on several factors, including the TCP protocol, link capacity, and competition from other users. As illustrated in Figure 1, there are also lower layers that may affect the achieved delay and bandwidth, particularly if part of the end-to-end connection is a wireless link.

Poor TCP performance over wireless links is a well known problem. The traditional explanation for poor TCP performance is that the wireless link drops packets due to noise on the radio channel, and that TCP interprets all packet losses as indications of network congestion. This explanation is a little too simplistic when considering wireless links that employ link-layer retransmissions. The link-layer retransmission scheme transforms a lossy link, with fairly constant delay, into a link with few losses but random delays. The resulting delays, it turns out, are also problematic for TCP.

Several approaches to improve wireless TCP behavior have been suggested in the literature. Modifications of TCP have been proposed [2], [3]. Other methods try to more directly differentiate loss as being either due to congestion or due to lossy wireless transmissions [4], [5], [6]. Performance-enhancing proxies is an alternative in which either split connection schemes or interception schemes are used. The first approach introduces a virtual user at the link which acts as receiver to the source and source to the receiver. In the latter approach, acknowledgments are monitored and dropped if they indicate packet loss due to link-layer retransmissions. Finally, it is possible to counteract the influence from the wireless link by letting the receiver control the transmission via its advertised window. See [7], [8] for further details on these schemes.

We believe that, as far as possible, the link-layer should be engineered to be TCP-friendly, reducing the differences between wired and wireless links. There will naturally be some residual idiosyncrasies of wireless channels that cannot be dealt with in this way; our approach should be viewed as complementing both developments to make TCP more robust to "strange" links, and cross-layer developments that let the link and the end-node TCP:s exchange information about link and flow properties.

This article is organized as follows. Section II describes our models for the lower layers of the system. In sections III we use the models to derive IP-level properties, in particular the IP-packet delay distribution, and implications for TCP performance. In Section IV we argue that we should use the engineering freedom we have in the link layer to make the radio link more friendly to TCP, and show that adding carefully selected delays to certain packets can improve TCP performance.

II. System Model

When using TCP over a wireless link, there are several interacting control systems stacked on top of each other, illustrated in Figure 1. At the lowest layer, the transmission power is controlled in order to keep the signal to interference ratio (SIR) at a desired level. This is a fast inner loop intended to reject disturbances in the form of "fading", or varying radio conditions. On top of this, we have an outer power control loop that tries to keep the block error rate (BLER) constant, by adjusting the target SIR of the inner loop.

The target block error rate is a deployment trade-off between channel quality and the number of required base stations. For UMTS the reference block error rate is often chosen to be about 10%, see [10], which is what we will use.

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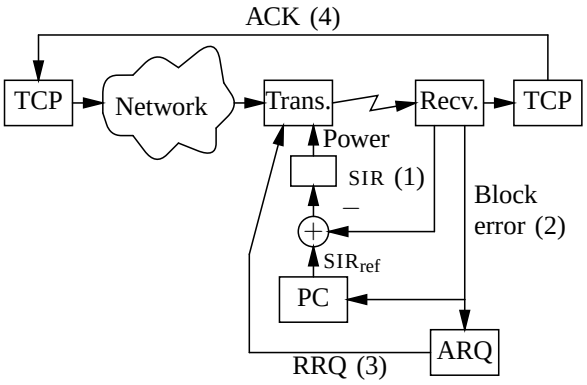

Fig. 1. End-to-end congestion control is affected by the delay and the bandwidth of the wired part of the network, but also by feedback mechanisms in lower layers of the wireless links.

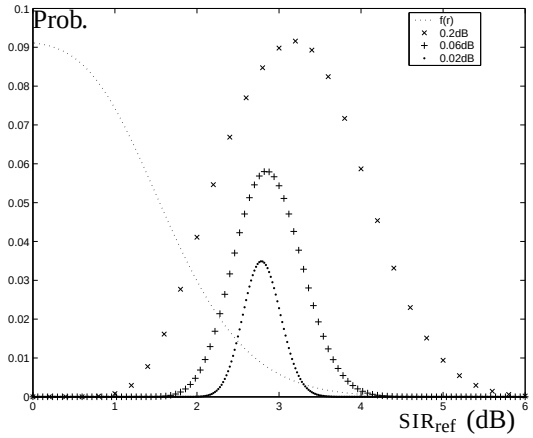

Fig. 2. Stationary distribution for the power control. Each mark represents one state of the power control, the corresponding value of SIR_{ref}, and its stationary probability. The dotted curve is the threshold-shaped function $f(r)$, scaled to fit in the figure, which represents the block error probability as a function of SIR_{ref}.

On top of power control, we have local, link-layer, retransmissions of damaged blocks. Finally, we have the end-to-end congestion control of TCP.

By modeling the lower layers, we can investigate the effects the link layer control have on TCP performance. We refer to our previous paper [9] for further details on the radio model.

A. Power control Markov chain

As there is no simple and universal relationship between the SIR and the block error rate, the outer power control loop uses feedback from the decoding process to adjust SIR_{ref}. The outer loop uses a fix step size Δ. It decreases SIR_{ref} by Δ for each successfully received block, and increases SIR_{ref} by 9Δ each time a block is damaged.

This process can be modeled as a discrete Markov chain, where state k corresponds to $\text{SIR}_{\text{ref}} = k\Delta$. Assuming that the inner loop power control manages to keep the actual SIR close to SIR_{ref}, and using an appropriate channel model, we get a threshold shaped function $f(r)$ which gives the probability of block damage for any $\text{SIR}_{\text{ref}} = r$. There are two transitions from state k of the Markov chain: To state $k + 9$, with probability $f(k\Delta)$, and to state $k - 1$ with probability $1-f(k\Delta)$. The operating point of the outer loop power control is close to the point where $f(r) = 10\%$, i.e., the desired block error rate.

From $f(r)$ and Δ, it is straight forward to compute the stationary distribution of the Markov chain. Figure 2 shows the stationary distribution for a BPSK channel (see [9] for the parameters) and three different values for Δ.

B. Link-layer retransmission

Since a packet loss probability on the order of 10% would be detrimental to TCP performance, the link detects block damage (this is the same feedback signal that is used for the outer loop power control), and damaged blocks are scheduled for retransmission. We will consider one simple retransmission scheme, the (1,1,1,1,1)-Negative Acknowledgment scheme [11], which means that we have five

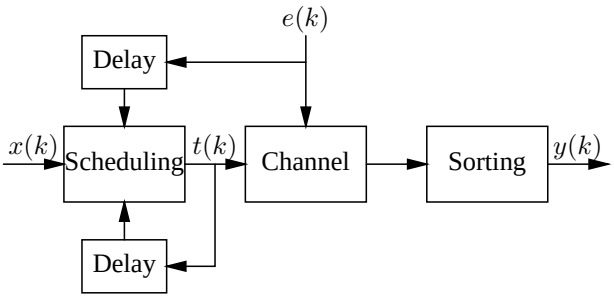

Fig. 3. Retransmission model

"rounds", and in each round we send a single retransmission request. When the receiver detects that the radio block in time slot k is damaged, it sends a retransmission request to the sender. The block is scheduled for retransmission in slot $k+3$ (where the delay 3 is called the RLP NAK guard time). If also the retransmission results in a damaged block, a new retransmission request is sent and the block is scheduled for retransmission in slot $k + 6$. This goes on for a maximum of five retransmissions.

C. Retransmission as feedback

To be able to analyze the impact of the scheduling mechanism on link properties such as the delay distribution, it is of interest to model the retransmission scheme. Feedback is an intrinsic property of the retransmission mechanism. Below we propose a model for the relationship between the input blocks, the block error process, and the in-order output blocks, where this feedback is explicitly shown. We believe that this model will be useful for further studies of retransmission scheduling.

Let k denote time in units of time slots, and consider the

following input and output signals, also shown in Figure 3.

$$x(k) = \#\text{ of input blocks up to time } k \quad (1)$$

$$e(k) = \#\text{ of errors up to time } k \quad (2)$$

$$y(k) = \#\text{ of in-order output blocks up to time } k \quad (3)$$

These are increasing functions, like the accumulated rate functions used in network calculus. Also let

$$t(k) = \text{Index of block transmitted at time } k \quad (4)$$

which is *not* an increasing function. Consider a simple one-parameter family of retransmission schemes, where each damaged block is retransmitted g slots later, and there is no limit on the number of times a block may be resent. The parameter g corresponds to the RLP NAK guard time.

To describe the process mathematically, we start with the queue at the input to the scheduler. Let $s(k)$ be the number of time slots up to time k that are not used for retransmissions, and let $f(k)$ be the number of blocks that have been transmitted (but not necessarily received successfully) up to time k. Then

$$s(k) = k - e(k-g) \quad (5)$$

$$f(k) = \min_{\ell \le k} \left(x(\ell) + s(k) - s(\ell) \right) \quad (6)$$

where the minimum in the latter equation is obtained when ℓ is the start of the current busy period.

The scheduling can be described as

$$t(k) = \begin{cases} f(k) & \text{if } e(k-g) = e(k-g-1) \\ t(k-g) & \text{if } e(k-g) > e(k-g-1) \end{cases} \quad (7)$$

Finally, $y(k)$ is defined by $y(k) = n$ if all blocks up to n have been received properly at time k, but block $n+1$ has not. In symbols,

$$y(k) = \max\{n : \forall \ell \le n, \exists j \le k, t(j) = \ell, e(j) = e(j-1)\} \quad (8)$$

So where in this model is the feedback? It is included explicitly, in Equation 7. We believe that this model, together with a model for the stochastic process $e(k)$, lets us optimize the parametrized retransmission scheme. The delay at time k is defined by

$$d(k) = \min\{\tau \ge 0 : y(k+\tau) \le x(k)\} \quad (9)$$

If x and e are stationary processes, with average rates that sum to less than 1, then also $d(k)$ is a stationary process, and its properties can, at least in principle, be calculated from x, e and the retransmission model. If $Q(d)$ is a quality measure, we can formulate the optimization problem

$$g^*(e, x) = \arg\max_g Q(d) \quad (10)$$

which gives the optimal value for the retransmission delay.

Intuitively, we expect that g^* will depend on the autocorrelation of the e process; it seems reasonable to use a retransmission delay such that loss of the original transmission, and loss of the retransmission, are negatively correlated.

This one parameter retransmission model is quite limited. Other schemes can be modeled analogously, as long the relation between e and s is simple, and the scheme does not need an additional queue for retransmitted packets. The challenge is to find a powerful but simple parameterization of an interesting class of schemes.

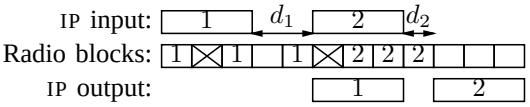

Fig. 4. IP packets divided into radio blocks

III. TCP/IP Implications

Consider the system at a randomly chosen start time, with the state of the power control distributed according to the stationary distribution. For any finite loss/success sequence (for example, the first block damaged, the next six received successfully, the eighth damaged), we can calculate the probability by conditioning on the initial power control state and following the corresponding transitions of the Markov chain. We can then use these probabilities to investigate the experience of IP packets traversing the link.

A. IP *packet delay*

As a link employing link-layer retransmission yields a very small packet loss probability, the most important characteristic of the link is the packet delay distribution. If the distribution is sufficiently "friendly" to TCP, then the layering of the system works nicely, which means that upper layers like TCP need not be aware of any particular properties of individual links in the network.

We first compute the packet delay distribution explicitly from the models described above. Later, we will also assume that the calculated delay probabilities apply independently to all packets, which should be fairly close to reality as long as the power control is working.

When transmitting variable size IP packets over the link, each packet is first divided into fixed size radio blocks, see Figure 4. Let n denote the number of radio blocks needed for the packet size of interest. Typically, $1 \le n \le 10$.

The delay experienced by an IP packet depends on which, if any, of the corresponding radio blocks are damaged, and on the number and scheduling of the block retransmissions. When all the blocks are finally received correctly, the IP packet can be reassembled and passed on.

From the probabilities for all possible success/loss sequences at the radio block level, we can extract explicit probabilities for the possible IP packet delays. The resulting delay distribution for our example channel (see Figure 5), with a power control step size $\Delta = 0.06$dB, and $n = 2$, is shown in Table I.

This table includes only the delays due to radio block retransmissions, there is also a fix delay of 40 ms for the original transmission of two radio blocks.

d	0	40	60	100	120	160	180
p	80.6	8.8	9.3	0.6	0.6	0.03	0.03

TABLE I

B. TCP *performance degradation*

In observations and performance evaluations of TCP over wireless links [11], the properties of a wireless link can shine through to the TCP layer in three different ways:

- Genuine packet loss. With bad enough radio conditions, packet drops are inevitable. We will not consider genuine packet loss here, as we assume that the radio channel is good enough that the power control and link-layer transmissions can get packets through.
- Packet reorder. For a link with highly variable delay, packets can get reordered. Severe reordering can trigger a spurious TCP fast retransmit.
- Spurious timeout. A packet that is not lost, only severely delayed, can trigger a spurious TCP timeout.

If the bandwidth delay product is small compared to the maximum TCP window size, spurious timeouts and spurious fast retransmit need not lead to any performance degradation, as a modest buffer before the radio link will be enough to keep the link busy even when the sender temporarily decreases its sending rate. On the other hand, if the bandwidth delay product is larger than the maximum TCP window size, throughput is decreased.

The difference between these two cases can be seen for example in the performance evaluation [11]: In the scenarios that have a large maximum window size compared to the bandwidth-delay product, we get a throughput that is the nominal radio link bandwidth times $1 - p$ (where p is the average block loss probability), and there is no significant difference between different link retransmission schemes. Only when bandwidth or delay is increased, or the maximum window size is decreased, do we see a drastic changes in throughput when the BLER or retransmission-scheme varies.

We therefore concentrate on the case of a large bandwidth-delay product. Then both spurious timeout and spurious fast retransmit leads to a degradation of TCP performance, and we will consider them in turn.

C. Spurious fast retransmit

Spurious fast retransmit has been observed as an important factor in poor TCP performance over wireless links. The probability of spurious fast retransmit can be estimated from the loss/success-sequence probabilities. It turns out that unless the link is configured to do "in-order delivery", the probability is significant for $n = 1$ (0.25%–0.8%). It decreases rapidly with increasing n.

Fortunately, this problem is easy to solve: Let the radio link receiver sort packets so that they are always passed on in order. Typically, this is an option in the configuration of radio link equipment, and it should be enabled on links

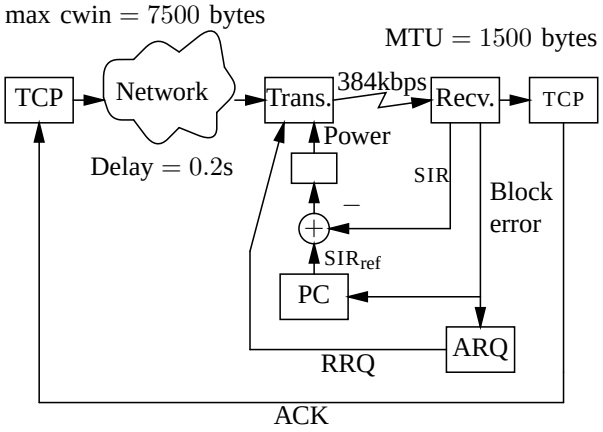

Fig. 5. Numerical example

where TCP performance is important. (One may want to use one channel with in-order delivery TCP-packets, and but not for real time streams like VoIP).

D. Spurious timeout

A TCP timeout event occurs when a packet, or its acknowledgment, is delayed too long. Let RTT_k denote the round-trip time experienced by packet k and its corresponding acknowledgment. The TCP algorithm estimates the mean and deviation of the round-trip time. Let $\widehat{\text{RTT}}_k$ and $\hat{\sigma}_k$ denote the estimated round-trip time and deviation, based on measurements up to RTT_k. TCP then computes the timeout value for the next packet as $\text{RTO} = \widehat{\text{RTT}}_k + 4\hat{\sigma}_k$, which means that the probability that packet k causes a spurious timeout is given by

$$\text{P}(\text{RTT}_k > \widehat{\text{RTT}}_{k-1} + 4\widehat{\sigma}_{k-1}) \tag{11}$$

An idealized model of TCP is to assume that the the estimation is perfect, and that the timeout value is set to $\text{RTO} = \text{E}(\text{RTT}) + 4\sigma(\text{RTT})$. From the delay distribution of Table I, we get $\text{RTO} \approx 103$ ms and the probability that the delay is larger is $\approx 0.68\%$. This is the probability of spurious timeout events. When varying the parameters n and Δ, we typically get a probability of spurious timeout on the order of 0.5%–1% [9].

E. Performance implications

As explained above, we will concentrate on the case of a large bandwidth-delay product. We will also assume that in-order delivery is enabled, so there is no spurious fast retransmit, only spurious timeout. For a concrete example, we will consider the scenario in Figure 5: Radio link bandwidth 384 kbit/s, packet size $m = 1500$ bytes, maximum TCP window size $w = 7500$ bytes (i.e. five packets), and a constant round-trip delay time, excluding the radio link itself, of 0.2 s.

The available radio bandwidth (excluding losses) is 42.2 Kbyte/s. Due to the limited window size, TCP can not utilize the link fully. The ideal TCP throughput is one

maximum size window per RTT. For the untweaked link, the mean total RTT is $200 + 40 + 10.6 = 250.6$ ms, implying an ideal throughput of 29.2 Kbyte/s.

For each spurious timeout, the sending TCP enters slow start. The window size is reset to 1 packet, and the slowstart threshold is set to 2 packets. For the next four round-trip times, we will send 1, 2, 3, and 4 packets, 10 packets less than if we had kept sending a maximum window of 5 packets every RTT. This leads to

$$\text{Throughput} = \frac{w}{\mathrm{E}(\text{RTT})(1 + 10\mathrm{P}_{\text{TO}})} \tag{12}$$

(a more general formula is derived in [9]). Hence, over the example radio link, we get a throughput of 27.4 Kbyte/s. See also the summary in Table II.

IV. Improving the Link-Layer?

It is not trivial to define precisely what properties a link should have in order to be friendly to TCP. It seems clear that for example links with normal or uniformly distributed and independent delays are friendly enough. One crude measure is to examine the tail of the distribution. More precisely, if X is a stochastic variable representing the identically and independently distributed packet delays, define

$$\mathrm{P}_{\text{TO}}(X) = \mathrm{P}(X > \mathrm{E}(X) + 4\sigma(X)) \tag{13}$$

The motivation for this measure is the calculation of the timeout value in TCP. Timeout is intended to be the last resort recovery mechanism, and for TCP to work properly, spurious timeout must be a rare event.

Also note that $\mathrm{P}_{\text{TO}}(X)$ is invariant under the addition of constant delays.

For distributions which we know are friendly to TCP, P_{TO} is small. For a general distribution, assuming only finite first and second moments, P_{TO} is bounded by Chebyshev's inequality. Comparing these values,

$$X \text{ uniform} \implies \mathrm{P}_{\text{TO}}(X) = 0 \tag{14}$$

$$X \text{ normal} \implies \mathrm{P}_{\text{TO}}(X) \approx 6.3 \cdot 10^{-4} \tag{15}$$

$$X \text{ wireless} \implies \mathrm{P}_{\text{TO}}(X) \sim 100 \cdot 10^{-4} \tag{16}$$

$$X \text{ arbitrary} \implies \mathrm{P}_{\text{TO}}(X) = 625 \cdot 10^{-4} \tag{17}$$

we see that the two friendly distributions yield a P_{TO} at least two orders of magnitude below the worst case given by Chebyshev. The wireless delay yields a significantly higher P_{TO}, although still with some margin to the worst case.

If we want to improve the system, where should we put the effort? The power control design have many constraints of its own, relating to radio efficiency and cost of deployment. It seems difficult to design and motivate changes to the power control for improving the delay distribution properties. Improvements to the TCP algorithm in the end-nodes are important, but also difficult both for technical and practical reasons, such as limited information about what goes on in the link (note that the link and the TCP implementations are not only in separate layers, they are also *geographically* separate), and the complex standardization and deployment process.

However, we do have some engineering freedom in the link itself. Even if we do not want to modify the power control, there are other link-local mechanisms we can add or optimize.

- Optimize the retransmission scheduling, taking advantage of the block loss correlation that we get after power control.
- Use error correction coding.
- Tweak the delay distribution by adding additional delays to selected packets.

In the remainder of this section, we investigate the simplest of these options, namely the third one.

A. Introducing additional delays

Assume that we have a discrete delay distribution for X, $\mathrm{P}(X = d_i) = p_i$, where $d_i < d_{i+1}$. It is typical, but not required, that also $p_i \geq p_{i+1}$. Let μ and σ^2 denote the mean and variance of X.

We consider the following class of tweaks to X. For each packet that experiences a delay $X = d_i$, buffer the packet so that it gets an additional delay x_i. This defines a new distribution $\tilde{X}$, $\mathrm{P}(\tilde{X} = d_i + x_i) = p_i$ (or if it happens that $d_i + x_i = d_j + x_j$ for some $i \neq j$, the corresponding probabilities are added up). For an example of what X and $\tilde{X}$ can look like, see Figures 6 and 7.

The parameters x_i are constrained only by $x_i \geq 0$.

What is the best choice for x_i? We select a maximum allowed value, ϵ, for $\mathrm{P}_{\text{TO}}(\tilde{X})$, and minimize $\mathrm{E}(\tilde{X})$ under the constraint that $\mathrm{P}_{\text{TO}}(\tilde{X}) \leq \epsilon$. This means that we want to push down our measure of "TCP-unfriendliness", while at the same time not adding more delay than necessary.

To simplify the problem a little, we require that $\mathrm{P}_{\text{TO}}(\tilde{X})$ corresponds to a tail of the original distribution X. Let k be the smallest value such that $\sum_{i \geq k+2} p_i \leq \epsilon$. Let $c = d_{k+1} + \delta < d_{k+2}$, where $\delta \geq 0$ is a robustness margin. We impose the additional constraints $\mathrm{P}_{\text{TO}}(\tilde{X}) = c$, $x_i + d_i \leq d_{k+1}$ for $i \leq k$, and $x_i = 0$ for $i > k$. Then, for any x_i satisfying these new constraints, we will have $\mathrm{P}_{\text{TO}}(\tilde{X}) = \sum_{i \geq k+2} p_i \leq \epsilon$. We get the optimization problem

$$\min \ \mathrm{E}(\tilde{X}) \tag{18}$$

$$\mathrm{P}_{\text{TO}}(\tilde{X}) = c \tag{19}$$

$$x \geq 0 \tag{20}$$

$$x_i \leq d_{k+1} - d_i, \text{ for } i \leq k \tag{21}$$

This is a quadratic optimization problem. To write it in matrix form, let x denote the vector $(x_1, \ldots, x_k)^T$, and similarly for p and d. Let $S = 16 \operatorname{diag} p - 17pp^T, b_i = 2p_i(16d_i + c - 17\mu), m_i = d_{k+1} - d_i$ and $\alpha = 16\sigma^2 - (c - \mu)^2$, and we can rewrite the problem as

$$\min \ p^T x \tag{22}$$

$$x^T S x + b^T x + \alpha = 0 \tag{23}$$

$$0 \leq x \leq m \tag{24}$$

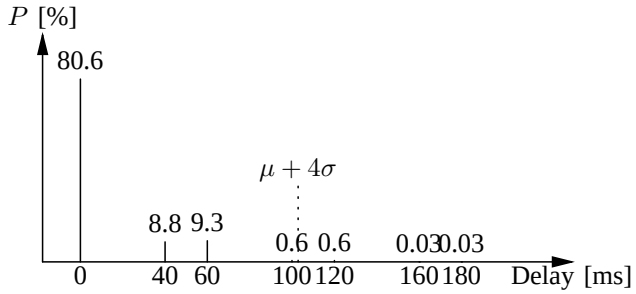

Fig. 6. Original delay distribution

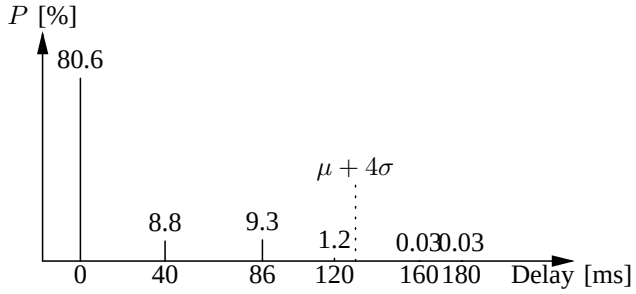

Fig. 7. Optimized delay distribution

Since the symmetric matrix S is typically indefinite, the problem is not convex. But it can be solved in exponential time $O(k^3 3^k)$, which has not been a problem thanks to the very limited size of k.

The typical solution is of the form $x = (0, \ldots, 0, x_j, m_{j+1}, \ldots, m_k)^T$. When the optimum has this form, it means that the cheapest way to to increase $\mathrm{P_{TO}}$, in terms of mean delay, is to increase the x_i corresponding to the smallest p_i. Necessary and sufficient conditions for the optimum to be of this form has not yet been determined.

Now consider the delays d_i and probabilities p_i in Table I, and assume that packets are independently delayed according to the given probabilities. This distribution is also shown in Figure 6. Before tweaking the delays, we have $\mathrm{E}(X) \approx 10.6$ ms, and $\mathrm{P_{TO}}(X) \approx 0.68$ %.

With $\epsilon = 0.1\%$ and $\delta = 10$ ms, the above optimization procedure yields $k = 4$ and the optimal additional delay $x \approx (0, 0, 26, 20)^T$ ms. This modified distribution is shown in Figure 7. The mean additional delay is only 2.54 ms, which seems to be a small cost, if we compare it to the transmission delay for the packet, which is 40 ms, or the end-to-end delay which necessarily is even larger. We also achieve $\mathrm{P_{TO}} < \epsilon$, if fact, we actually get $\mathrm{P_{TO}} \approx 0.06\%$.

For the tweaked link, we have a slightly larger RTT (which in itself would decrease the throughput slightly), and a significantly smaller $\mathrm{P_{TO}}$. The resulting throughput is 28.8 Kbyte/s, an improvement by 5% compared to the unmodified link, and only 1.4% below the ideal TCP throughput. These figures are summarized in Table II.

	Kbyte/s
Available radio bandwidth	42.2
Ideal TCP throughput	29.2
With wireless link	27.4
Modified wireless link	28.8

TABLE II

The important point is that a simple but carefully selected modification to the link-layer yields a modest but significant performance improvement.

V. Conclusions

In this contribution we have studied the effect retransmission in radio links has on packet properties. In particular, we have delineated the implications that it has on TCP. An input/output model has been suggested where the rôle of the scheduling mechanism is made explicit. The main contribution has been to show that a slight artificial increase of the delays of certain retransmitted packets may reduce the risk of spurious timeout in TCP and hence increase the throughput; in an example the increase was 5%. The artificial delay distribution is optimized off-line and applied on-line. The additional delay that is applied to a packet depends only on the retransmission delay experienced by that same packet, and this information is available locally at the link.

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