TCP-ELN: Efficient Communication over Wireless Channels

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As wireless technologies becoming more wide-spread, the importance of efficient communications over radio channels increases. TCP versions applied currently do not take into account the special characteristics of radio channels, thus their performance can be significantly lower compared to the available resources. In this paper, we propose a new TCP variant (called TCP-ELN) which is capable to considerably improve the transfer rate. The performance of the new solution is evaluated using simulations.

1. Introduction

TCP congestion control has been designed for the application in wired networks that have negligible packet losses due to bit error on the channel. Consequently, standard TCP versions do not distinguish between losses due to congestion and losses due to bit-error on the channel and simply reduce the congestion window for both types of packet losses.

In the recent years, TCP/IP has been used in wireless networks to support Internet access through radio interfaces. In a wireless environment, where bit errors on the channel happen with high probability, reduction of the congestion window without distinguishing losses due to bit error and congestion at the TCP layer makes TCP congestion control ineffective. If a TCP flow is transmitted through a wired part without packet errors and a wireless link with bit errors, TCP reduces its congestion window in response to packet losses caused by bit errors on the wireless channel. Therefore, the throughput of TCP flows may be close to zero although the network is not congested at all.

Several options exist to optimize TCP performance over wireless networks. A major part of today's operational wireless networks employ an ARQ mechanism at the link layer for IP based data services (e.g. 802.11, UMTS acknowledged mode). Link layer retransmission has been shown to be efficient and scalable in many scenarios. The number of retransmissions attempts before a link layer frame is considered lost, however, is usually rather small in order to avoid high delay variation and buffering. Thus, even in the presence of link layer retransmission a residual packet loss probability exists from the perspective of the transport layer. Authors in [1] report an average residual packet loss probability of more than 6% for the case of WLAN in a standard industrial environment. Therefore, additional mechanisms are needed at the TCP layer to improve performance in the case of critical radio conditions even in the presence of link layer ARQ. In case of no

link layer ARQ and FEC (e.g. UMTS transparent mode), TCP is obviously unprotected against packet loss due to bit errors, making a clear case for protection at the transport layer.

Several proposals have been worked out for transport layer mechanisms to help TCP distinguish between losses due to bit error and losses due to congestion. The indirect TCP approach [2] splits a TCP connection between a fixed and a mobile host into two separate connections and hides TCP from lossy link by using a protocol optimized for lossy links. The Berkeley SNOOP protocol [3] caches packets at the base station and performs local retransmissions over the lossy link. Note that both approaches require storage of extensive per micro-flow states and even data packet retransmissions at the base station make them scale badly in terms of link bandwidth and number of flows. Bakshi et al. [4] investigated the performance of TCP over wireless links varying the MTU size and proposes to send ICMP packets to inform the TCP source about the wireless link experiencing bit-errors.

Balan et al. [8] proposed the TCP header checksum option to make TCP detect whether a bit-error happened in the body of the TCP segment or in the header. In the first case, the congestion window is not reduced.

TCP Westwood [5] has the potential to improve the performance over lossy channels as it adapts the congestion window based on throughput variations rather than packet losses. TCP Westwood is a sender-only modification to TCP thus it scales and is easy to deploy. The mechanism of throughput based window adaptation, however, needs some initial time to become effective and TCP Westwood cannot avoid unnecessary window reductions in the presence of packet losses due to bit-error. Thus Westwood will not show significant improvements of TCP flows over lossy channels.

Explicit Loss Notification (ELN) [6] is a mechanism by which the reason for the loss of a packet can be communicated to the TCP sender. The base station monitors all TCP segments that arrive over the wireless link but does not cache any TCP segments since it does not perform any retransmission. Rather, it keeps track of the segments that have been lost over the wireless link and sets the ELN bit in the corresponding ACKs. This original ELN proposal implies the drawbacks of requiring a list of the lost sequence numbers per microflow at the base station which is impossible in case of IPSEC encryption. Moreover, this proposal does not take ACK loss into account and provides poor signaling information for the TCP sender.

In [7] we have redefined the ELN technique keeping only the fundamental idea of it, informing the TCP source explicitly on packet losses on the wireless channel. We proposed a Reno based TCP congestion control algorithm that can improve the performance of TCP significantly if the source is able to inform the sender about the number of the corrupted TCP segments. However, the technique to gain loss information that makes the protocol work is only briefly described in that paper and implemented in the simulator in a course grained manner.

In this paper we give the details of gaining loss information from corrupted packets, taking the different configuration cases of wireless environments into account. We also introduce a novel enhanced congestion algorithm that exploits all information from the loss notifications in order to maximize its performance.

The rest of the paper is organized as follows. The ELN based TCP congestion control algorithm is discussed in section 2. In section 3 we describe the details of our loss recovery technique. Section 4 specifies the simulation environment and shows the results. Finally section 5 concludes the paper.

2. The TCP-ELN proposal

In order to improve the performance of TCP by collecting loss information, both the server and the client side of the TCP agent must be modified. Note that the approach for ELN taken in this paper does not necessarily require modification of network elements. Therefore, it can be implemented in an end-to-end manner. The TCP receiver must be able to receive loss information from lower layers and notify the sender, who needs to act properly in reaction to the reception of loss and congestion information.

2.1. Receiver side modifications

By using a loss recovery technique the TCP receiver may receive not only intact TCP segments but also loss notifications. When the TCP data receiver receives intact TCP segments it operates in compliance with standard TCP behavior i.e. an acknowledgement is generated. If a loss notification is received then it is stored. This loss information is sent to the TCP source embedded into the option field of the acknowledgement. In case of bursty bit errors several TCP segments can get damaged possibly producing a high amount of loss notifications before an ACK is generated. Thus these notifications *must be stored* at the receiver to avoid dropping of loss information.

No negative ACKs are used thus loss notifications can be sent to the source only piggybacked on ACKs. As a consequence the use of *delayed acknowledgements is feasible though not recommended* since it decreases the signaling speed.

The TCP receiver must take also ACK loss into account. Since loosing an ACK causes the loss of loss notifications the notifications *must be sent redundantly*. Depending on the implementation it must be defined how many notifications can be stored in an ACK and how many times a notification is resent. In our implementation we have implemented a SACK like solution with sending loss notifications three times.

During the lifetime of a TCP flow timeouts can occur at the sender. After the timeout the sender starts sending packets regardless of the ongoing TCP ACKs. In this case all loss notifications must be cleared by the receiver since these losses occurred before the timeout and sending them to the source can cause inconsistent TCP behavior. Thus the sender must be aware of sending timeout notifications and the receiver must be aware of *handling these signals*. How timeout signals are generated by the source can be found in the next subsection.

2.2. Sender side modifications

The flow control algorithm of TCP-ELN is based upon TCP Newreno's standard. If no loss notification is received by the source TCP-ELN behaves exactly the same way as Newreno. If the source receives loss notifications in the ACKs from the receiver then it stores the notifications and resends the lost segments. Until receiving the 3rd duplicate ACK TCP-ELN operates according to Slow Start or Congestion Avoidance algorithms. The 3rd duplicate ACK can be a sign of congestion loss or loss due to bit-error. To decide which recovery state the TCP source should enter the stored loss notifications are examined. If a notification of the lost segment indicated by the duplicate ACKs was stored previously then TCP-ELN enters the wireless recovery state. If no loss notification is stored for the segment indicated by the triple duplicate ack event then TCP-ELN treats the 3 consecutive duplicate ACKs as a sign of congestion. Thus Fast Retransmit and Fast Recovery are invoked.

In the wireless recovery state the source waits for a recovery ACK that acknowledges all segments that were lost on the wireless channel. During this recovery phase reported segments that are lost due to wireless bit-error are resent. Every time a duplicate ACK arrives the source increases the size of its congestion window with one segment. As a consequence, the TCP data flow and self clocking is kept going, except if the effective window size is limited by the maximum value of the send window. If a partial ACK arrives at the source then the source decides whether it should stay in the wire-

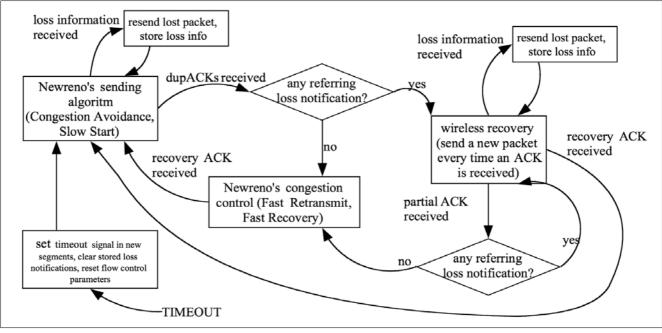


Figure 1. The operation of the TCP-ELN sender

less recovery state or enter the congestion recovery state. To make the decision the stored loss notifications are checked just like after receiving the 3rd duplicate ACK. If there is a referring loss notification then the source stays in the wireless recovery state otherwise it enters the congestion recovery state. If the recovery ACK is received then Slow Start or Congestion Avoidance, respectively, continues. On The operation of the TCP-ELN sender can be seen in *Figure 1*.

During the operation of the flow control algorithms timeouts can occur. In this case all flow control parameters are reset like in Newreno, the list of the loss notifications is cleared and a timeout signal is sent to the receiver. Since this timeout signal can be lost it must be sent in a redundant manner. The sender includes the number of timeouts since the start of the flow in every TCP segment's header option field. Thus even if packets get lost the timeout signal will reach the receiver. It will be received when the first intact TCP segment arrives.

3. Loss information recovery techniques

It is crucial for the operation of TCP-ELN to gain loss information from packets damaged on wireless channel. Three parameters the TCP sequence number, port number, and the source IP address of the lost segments must be collected by the wireless client in order to identify a single TCP flow.

There are two major methods to recover loss information; both of them influenced by the network topology and configuration. To demonstrate the recovery techniques the topology of *Figure 2* is used. This is a widely deployed scenario that contains a wireless link between the client and the access point.

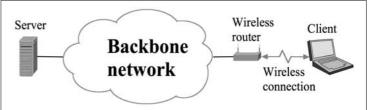
3.1. Cross layer communication

The first method is built upon the fact that MAC frames are not simply lost but received in a damaged manner by the destination node in the case of radio link error. Sensitive information can be found in the TCP/IP headers so extracting the intact headers from a damaged MAC frame solves the identification of a single TCP flow. To check whether the TCP header is damaged or not an additional header checksum is needed since TCP header has no dedicated protection [8].

This checksum can be inserted in the TCP header's option field. When the modified MAC layer of the client receives a corrupted frame instead of dropping it, it propagates the payload of the frame to the upper layers. If all the headers belonging to the different protocol levels are undamaged then the TCP segment reaches the TCP layer where loss information is extracted and sent to the source of the data flow. As the payload is much longer than the headers, bit errors are likely to occur in the payload leaving the headers intact. So this recovery method can be very efficient to collect loss information.

The advantage of the MAC modification method is that only the destination peer needs to be modified in order to collect loss information. All other network nodes remain untouched making this method absolutely

Figure 2. Typical wireless access scenario



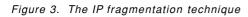
end-to-end compliant. Since the MAC layer is modified only at the wireless client only the last wireless link can be monitored and loss information can be extracted only from the download flows.

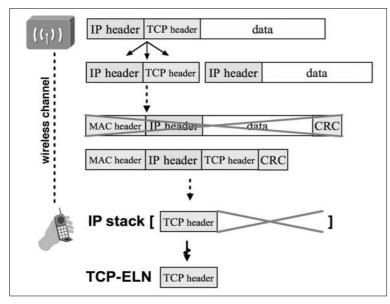
If modifying the MAC layer or recovering damaged packets at the MAC level is impossible, the ELN proposal may remain applicable by using the IP fragmentation method.

3.2. IP fragmentation

The fundamental idea of the IP fragmentation method is that the probability of loosing small packets on the wireless channel is smaller than the probability of loosing large packets. So if every TCP segment reaches the access point encapsulated in a single IP packet then fragmenting these IP packets at the access point into at least 2 packets where the first small part contains only the TCP header and the 2nd large part contains the payload can be used to collect loss information efficiently.

This method assumes that IP packets are not fragmented in the wired routers. The wireless router sends the IP packets embedded into MAC frames through the wireless channel. Since the MAC layer of the receiver is untouched, the damaged frames get dropped as usual. The IP layer tries to reassemble the received IP packets, but if any of them is lost then the original TCP segment cannot be reassembled. This means that at least one IP fragment was lost due to wireless bit error. But if the first IP packet with the TCP header included has arrived then the receiver's IP layer can partially reassemble the damaged TCP segment neglecting the missing parts. These missing IP packets contain the payload that is not used for loss information recovery. The damaged TCP segment is propagated then to the TCP layer where loss information can be extracted after verifying the TCP header checksum. The operation of this technique can be seen in Figure 3.





The drawback of the IP fragmentation method is that it violates the end-to-end semantics since it needs the wireless router to be slightly modified. However using this technique can be very efficient since it is *scalable* in contrast to other non-end-to-end proposals like Snoop, the needed modifications at IP layer are in accordance with the *normal IP level operation* (fragmenting) and it is *independent* of the MAC layer. This is a trade-off between the possibility of MAC level modifications and the end-to-end schematics.

4. Case study

4.1. Simulation environment

To measure the performance of TCP-ELN a wide variety of simulations were run.

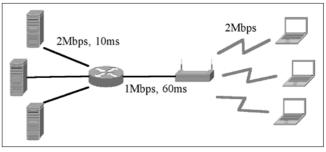


Figure 4. Simulation topology

The simulation topology illustrated in *Figure 4* consists of 3 servers, 3 wireless clients, a router and an access point. The bandwidth of the links between the servers and the router are 2 Mbps with 10 ms of delay. The router is connected to the access point with a 1 Mbps bandwidth link having a latency of 60 ms. The access point offers 2 Mbps for each client. All links are full-duplex. No specific MAC layer is applied on the wireless links, since the proposed techniques are general and do not depend on a specific wireless technology as strongly as on the topology. Due to the bandwidth set-

ting congestion can occur only at the router in the wired part of the network.

Two different types of traffic: FTP and web traffic were used to examine the behavior of TCP-ELN. While FTP traffic is used to investigate the steady state behavior of the protocol web traffic is used to examine the dynamic behavior. In the first case one FTP session is started between each server-client pair. All servers used the same TCP version – TCP-ELN or TCP-Newreno. The throughput of the two TCP versions were measured and compared.

In the web traffic case each client simulates a web browsing user that connects to one of the servers. The parameters of the web traffic are in accordance with the SURGE model [9] that is based upon real network traces and widely used to generate realistic workloads. The download speed (page size/download time) of the web pages with the different TCP versions are measured and compared.

To simulate wireless channel errors two error models were used. Uniform and Markov error models are two different approaches of modeling the lossy link behavior. Both of them are used to simulate packet loss at the TCP level. The mean value of the uniform distribution is varied between 0 and 0.2.

The Markov model presented in [10] is applied in the simulation study, which integrates channel fading and radio link parameters to take into account the characteristics of wireless channels.

The Markov error model simulates how the radio links with different drop probabilities are experienced by users with different speed (*Table 1*).

Model number	User speed	Average error rate	Average error burst length
1	Pedestrian	0.001	1.4913
2		0.01	4.0701
3		0.1	13.6708
4	Middle	0.001	1.0083
5		0.01	1.0838
6		0.1	1.8629
7	Vehicular	0.001	1.0024
8		0.01	1.012

Table 1. The parameters of the Markov error model

Since the efficiency of the loss recovery algorithms depends on how TCP segments are damaged, it is assumed that the probability of recovering a damaged segment is equal to the ratio of the payload size and the total packet size. In real environments this ratio is about 95% so in the simulations 5% of the damaged TCP segments cannot be recovered.

4.2. Results

Inspecting the throughput over FTP traffic TCP-ELN shows significant improvements in most cases, reaching as high as 400% depending on the particular error rate. As compared to Newreno, TCP-ELN is much better in the presence of error on the wireless link. In the error free case, i.e. without any packet loss on the wireless link, the performance of TCP-ELN is insignificantly lower (0.5%) than the performance of Newreno. This is due to the overhead, the larger header size of TCP-ELN segments. However this phenomenon also appears when using TCP-SACK.

Figure 5 shows that if the error rate is increased, the proposed flow control algorithm makes TCP-ELN throughput decline more gently than the rapid decrease observed for Newreno. For instance, while the throughput of TCP-ELN drops to 90% at the error rate of 14% Newreno's throughput drops down to 90% at the error rate of 4%. The results with the Markov error model show that in the high loss scenarios (0.1 mean PER) model

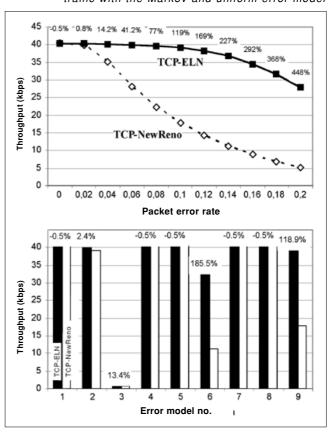
3, 6 and 9, the improvement depending on the burstiness can reach 185%. In the low loss scenarios (0.001, 0.01 mean PER) model 1, 4, 5, 7 and 8, an insignificant (0.5%) decrease can be observed, in model 2 the improvement is 2%.

Comparing the relative improvement measured over the uniform error with the ones measured for the Markov error model an interesting correlation can be recognized. As the speed parameter of the Markov error model increases the average length of error bursts decreases. Thus for the considered scenarios at higher speeds – as packet loss becomes sporadic – the Markov error model converges to the uniform error model.

In case of web traffic, less dramatic, but still significant improvements can be observed as shown in *Figure 6*. The improvement can reach 33% with the uniform error model at high error rates. When no packet loss occurs on the wireless channel then the performance decreases 0.1% due to the larger header size. Up to an error rate of 0.02 the performance of Newreno is almost identical to ELN performance. In case of the Markov model the improvement remains moderate 4% to 16% for the high error models (3, 6 and 9) and lower values for the low error models.

Comparing the web and FTP traffic results it can be seen that in the case of FTP traffic the improvement is significantly higher. The reason for this is that in the case of the used HTTP 1.1-like model for web traffic the majority of flows only has a few packets to send. When

Figure 5. The relative improvement and the throughput over FTP traffic with the Markov and uniform error model



the TCP flows are very short Newreno's congestion window remains small. Thus ELN's flow control algorithm cannot improve the performance with avoiding window halving as much as in the case of longer flows and larger congestion windows.

5. Conclusions

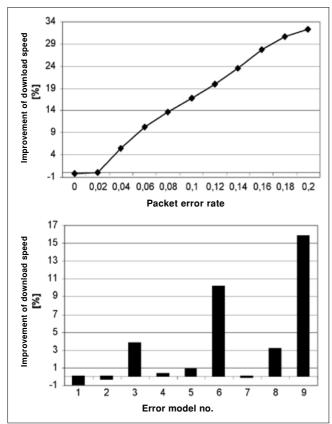
In this paper we proposed a new mechanism to improve the performance of TCP over wireless channels. Our solution is based on the idea of Explicit Loss Notification (ELN) which allows TCP to differentiate between packet losses due to congestion and packet losses due to radio channel errors. Thus unnecessary window reduction can be reduced substantially in the presence of radio channel errors. The resulting performance of the modified TCP protocol is greatly enhanced compared to TCP-Newreno.

We developed two alternatives to gain loss information in order to use ELN. The first solution is based on modifying the MAC layer; the second solution uses a special IP fragmentation technique. These methods can be used in a wide range of network environments depending on the topology and the setup of cross layer communication.

We developed a new TCP variant, TCP-ELN that integrates the loss recovery methods to improve its performance. The proposed receiver and sender side algo-

Figure 6.

The relative improvement of the download speed over web traffic with the uniform and the Markov error model



rithms and processes were discussed in detail. Security issues concerning TCP-ELN were also considered.

Our simulation experiments have shown that TCP-ELN can improve the performance of TCP over the radio channel substantially in a wide range of environments. The protocol was tested over random and bursty erroneous radio links with two different types of traffic, static FTP bulk load and dynamic web traffic. Results proved that TCP-ELN produces better performance than TCP-Newreno in all the cases. The performance improvement can reach up to 400% when using FTP traffic and 30% when using an HTTP 1.1-like model for web traffic with high error probabilities.

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