‘A Comparison of Mechanisms for Improving TCP performance over Wireless Links’

NAME: SHWETA SUBRAMANI UNITY ID: ssubra10

1. What is the main contribution of the paper?

There has been a significant amount of change in the type & scale of users of the Internet over the years since the Transmission Control Protocol was first proposed [2]. The fundamental problem lies in the fact that traditional TCP, originally designed for wired networks, which were less susceptible to random losses, is being used even today in the presence of wireless links and highly overprovisioned routers. Specifically, TCP assumes any kind of packet loss to be an indication of the network being congested, which may be true in most cases for wired networks. However for internetworks with wireless links, a lot of the packet loss could be attributed to random losses over the wireless links. In such situations, when TCP invokes its congestion control algorithm, the result is that, neither is it able to take advantage of the existence of the fast and ‘thick’ data pipe, nor is it able to cope with the errors rampant in wireless links resulting in degraded performance in terms of throughput and good put from an end-to-end perspective.

This paper[1] compares a few of the proposals to improve the performance of TCP over wireless links based on their performance obtained from the results of experiments conducted by the authors. The paper broadly characterizes the solutions under consideration into 2 categories based on their approach:

i. The first category of solutions is based on an approach that hides any non-congestion-related losses from the TCP sender and does not require any changes to be made to existing sender implementation.
ii. The next category makes the sender aware of the existence of wireless hops and realize that some packet losses are not due to congestion.

Based on these broad categories, they further group the techniques under scrutiny into 3 classes as mentioned in the table 1[1].

Of all the above mentioned schemes analyzed by the authors, the claim made is that LL-TCP-AWARE protocol performs the best. This further leads into the conclusion that it is not required to split the TCP-connection in order to improve the performance. Another inference drawn is that of all the above schemes the ones incorporating SMART[4] based selective acknowledgement mechanisms perform better in comparison with their counterparts in the same category. The paper also claims that end-to-end schemes are not as effective as local techniques but still seem to be promising since they do not require support from intermediate nodes in the network. [1]

2. **(i) What were the weaknesses or problems of the paper? (ii) Did the authors address these weakness or problems? (iii) Are these weaknesses/problems impossible to correct? (iv) If not, do you have some ideas of how to correct them?**

   a. The authors assume in all their experiments that they have **perfect knowledge about packets that were lost due to errors on wireless (lossy) links**. In reality determining this information is not easy [3]. The results of all experiments conducted by the authors of this paper must be verified when the various protocols under scrutiny incorporate such detection mechanisms to actually distinguish the cause for packet losses. From my understanding, the results may not be same since there is additional processing being done now to distinguish the cause for lost packets which might introduce a significant amount of delay in the response of the TCP sender to the missing packet which may only aggravate the existing problem rather than alleviate it.

      This problem or weakness of the paper can be rectified by following techniques mentioned in [3] which examines error patterns within a physical layer symbol to distinguish loss over the wireless link from one due to congestion.

   b. Furthermore, all the experiments have assumed **a Poisson-distributed bit-error model** for errors on the lossy link. In reality, errors in the wireless medium occur in bursts and are highly random. The authenticity of the results must hence be validated with different error models before concluding them ‘to be applicable under other patterns of wireless losses’.

      Such a problem can be rectified by running simulations where the error rate can be modeled by a different process. We could then infer a trend from all the samples that we collect by running these simulations before we come to a general conclusion about the validity of the results over any loss model.

   c. The paper claims that from the experimental results obtained the TCP-aware link layer protocol LL-TCP-AWARE performed the best. This protocol takes advantage of the knowledge of the higher-layer transport protocol (TCP). This protocol **violates the basic principle of layering** which is characteristic of the internet protocol stack. While we may obtain better results by crossing the two layers, it might not be a good idea to bring this protocol into practice since it may prove counter-productive in some cases. Crossing layers would no doubt boost the performance but this is only a short term achievement. From a futuristic point of view, if suppose a radical change takes place in the transport layer protocol then this will call for a subsequent change in the link layer protocol as well due to the interdependency between them which will eventually create loops. Protocols thus, cannot be designed independently. This invariably reduces the robustness of the protocol subsequently affecting the stability of such a system. Thus in my opinion, such a protocol which violates the fundamental principles should not be advocated.

d. The authors also conclude that selective acknowledgement schemes are very useful in the presence of lossy links. However, the SMART-based selective acknowledgement scheme employed by them [4] in all protocols requires **additional information to be incorporated in the packet header which means additional processing of the TPDU** [5] at the transport layer entity. This is also true in the “Explicit Loss Notification” scheme. Furthermore, these schemes require us to make changes in the existing sender implementation in terms of processing the TPDU.

   What could be done is to find out if the overhead of this additional information amortizes over the performance of the transport layer protocol. If the tradeoff between the throughput and additional overhead is reasonable then we can follow such a scheme.

e. Other schemes that seem to have yielded good results in the authors’ experiments are the ones falling into the category of split-connection. One main disadvantage of split connection is that the intermediary needs to maintain ‘per-connection’ state information. This has its own implications. Firstly, as identified by the authors themselves, it **does not preserve TCP’s end-to-end semantics**. Secondly, in wireless networks unlike the ones in their experiment, where handoffs are frequent, there will **be frequent context switching** as well. The performance of this protocol is questionable under such circumstances. If the intermediary maintaining the state information crashes, then the whole **connection is lost**. This might prove fatal in daily transactions. The authors address the former problem and state that since its performance is not any better than any of the TCP-aware link layer schemes, splitting the connection at a base station may not be required in order to improve TCP’s performance on the wireless hop.

f. An assumption made by the authors while conducting their experiments is that since intermediate host is close to the final destination, there is little reordering of packets. They also identify that ‘SMART retransmission schemes trades off some resilience to reordering and lost acknowledgements in exchange for a reduction in overhead to generate & transmit acknowledgements’. Despite this, solely based on the assumption made by them, they claim that the scheme is ‘well suited to situations where there is little reordering of packets which is true for one-hop wireless systems such as theirs’. This is a highly idealistic scenario! Thus the results obtained, based on which the authors infer that the mechanisms adopting SMART-based retransmissions are better than their counterparts in the category, once again, are not fool-proof. In reality packet reordering is not a trivial issue. The **transport layer protocol is responsible for**

fragmenting and reassembling frames, which takes some processing time and hence reordering, could severely affect the performance of TCP.

g. The results of the experiments conducted by the authors are based on data corresponding to **one-hop wireless network** scenario i.e. only the last hop to the receiver is a wireless link. Based on these statistics they go on further to state that they ‘believe it will be applicable in a wider context’. This assumption needs to be validated for a multi-hop wireless network or at least for a path with 2 wireless hops probably one from the sender to an intermediate router and one from the last wired router to the receiver. In case of multi hop wireless networks, there’s a possibility of an increase in the probability of the acknowledgements getting lost. In such a case, the sender’s retransmission timer will time-out, eventually reducing the congestion window size thus defeating the purpose of devising a new protocol to prevent this. This would probably require more careful estimates for the Retransmission Time-Out (RTO) value. Furthermore, in case of multiple wireless hops, there will be more out-of-order delivery of packets. This increases the transport layer’s workload to process and reassemble the out-of-order packets.

In conclusion, the paper makes many **idealistic assumption**s and fails to account for scenarios in which these assumptions may prove detrimental to the protocol’s performance. To justify these assumptions, the suggested protocols should be validated against these scenarios, so that the results of these experiments can deemed to be fool-proof.

3. **(i) Do you have any better solution to the problem than suggested by the authors? (ii) If so, please describe clearly and provide all details. (iii) Then explain why it is a better solution.**

We know that:

a. Air is a shared medium

b. There are many users contending for this medium

c. Losses in this medium are highly inevitable since it is also used by other devices such as mobile phones, Bluetooth devices and hence our internet packets experience interference due to them.

d. The annual estimated growth in IP traffic is already of the order of Exabytes and is predicted to go up to the order of Zettabytes by the end of 2016, the same time during which over half the world's internet traffic is expected to come from Wi-Fi connections.[7]

e. As the number of internet users are growing at a larger than an exponential rate there will always **be huge amount of data to be transmitted from the system's point of view**. System here will refer to the set of hosts competing for access to the shared(wireless) channel on the first hop. Thus, in such a system there will always be ~100% utilization of the shared resource(which is the medium in this case).

Suggested Solution

Under normal operation of the TCP, if there are 4 users, namely A, B, C, & D that have corresponding connection with senders A', B', C' & D'. Now suppose A or A' faces packet loss due to corruption over the wireless link. This means, one of them which was sending the data will reduce their congestion window size. For users B, C & D this is an advantage as this means there will be lesser traffic from A to A'. The wireless router will see a reduced incoming traffic. So this difference in capacity can be used to quickly process packets belonging to the connections between B & B', C & C' and D & D'. In this way, these 3 connections will see an increase in their individual throughputs. The router could probably use a **weighted fair queuing** scheme as discussed in class, but the weights should be adapted in an intelligent manner to favor the processing of packets belonging to B-B', C-C' OR D-D'. When all these flows are aggregated, from the network operator's point of view, his line is always being used and therefore is close to 100% efficiency.

Another possible improvement I see is to use **wireless repeaters** to boost the signal strength of packets being transmitted. This will at least ensure from packets being lost due to multipath fading. Since the crux of the problem lies in the fact that packets could get lost on the wireless link, we must follow the least possible measures to check the loss of packets on the wireless link.

Also following from point e above, we had earlier learnt in class that a static channel allocation technique like **Time Division Multiplexing** does not perform well under low load conditions. However, if there could be a way in which when we find a user to be idle during one particular time slot, and then allot this slot to another connection then we can improve the performance of this scheme. In any case, in the future, there is always going to be a lot of data that is transmitted over the 'long fat pipes' and 'low load' would never be a hindrance. Probably the only obstacle that could be faced here is synchronizing the users of a connection.

Thus, at a macroscopic level, we shall never see drastic degradation in the efficiency of our transmission control protocol (TCP-Reno/NewReno/SACK/etc.) from the system's perspective. Since the channels are 100% utilized, the network operator should not have any complaints.

Although this is true, a fact to be considered is that while in the methods stated in the paper, reduction in congestion window size was being avoided for as long as possible, thereby preventing an individual connection's throughput from decreasing, the method stated by me above does not follow such a policy. It allows TCP to work naturally, but ensures maximum channel utilization by giving advantage to other users in the network. This is another example of a situation where we see a **tradeoff between efficiency and fairness** (as studied in class). However, the solution requires no change to current TCP (hence it can be applicable over any version of TCP being used), it does not violate the end-to-end semantics of the transport layer protocol or the fundamental layering principle of the internet protocol stack(which makes it stable and robust). Hence, in my opinion this is a better solution when compared to the ones stated by this paper.

4. REFERENCES

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