QoS IMPROVEMENTS RESULT FROM TCP/RLC AND MAC IN A MOBILE CHANNEL

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Abstract: Mobile telecommunication new services are based on data networks specially Internet. These services include http, telnet, ftp, Simple Mail Transfer Protocol (SMTP), etc. Besides we recognize a mobile network as a multi-user network. Transmission Control Protocol/Internet Protocol (TCP/IP) which is sensitive to link congestion in wireline data links is also used in wireless networks. In order to improve the system performance, the TCP layer uses flow control and congestion control. Besides, Radio Link Control (RLC) has been introduced to compensate the deficiency of TCP layer in wireless environment. MAC and RLC have important roles in quality of service improvement of UMTS. In this paper we verify TCP over Automatic Repeat reQuest (ARQ) error control mechanism and finally quality of service improvement results from it in the fading channels.

1 INTRODUCTION

Transmission Control Protocol (TCP) is an end-to-end transport protocol in the Internet Protocol (IP) suite which is widely used in popular applications like SMTP, ftp and http. TCP guarantees reliable and in-sequence delivery of packets. TCP performance gets severely affected when used on channels which are typically characterized by high error rates (e.g., wireless channels). Although Tahoe and Reno versions of TCP are commonly used implement-tations, several other enhancements to TCP, including New Reno, Selective Acknowledgement (SACK), and Vegas, have been proposed to improve TCP performance. However, it has been shown that TCP enhancements like New Reno and Vegas do not offer significant improvement compared to Tahoe on slowly fading, low bandwidth-delay product wireless channels. It was suggested by (Chockalingam and Zorzi, 1999) that a clever design of the lower layers where memory in the packet error process (naturally present on wireless links because of the fading behavior) is preserved could be more effective than the development/use of more complex TCP versions. Wireless radio channel is affected by shadowing and fading. These, in turn, are affected by environment and mobility of subscribers. On the other hand, developing data networks and their connections to the mobile network has introduced new attractive services. These services are in addition to customary voice services. Data services include data file, video and voice transfer within the mobile network or over the Internet connection. Internet traffic is a combination of the above traffic services.

We know data traffic is sensitive to packet loss and voice traffic is sensitive to delay (Chockalingam, 2000). TCP provides end-to-end recovery and flow control. Link layer recovery protocols in cellular systems operate below the TCP layer and have complex interactions with TCP.

TCP has been designed and tuned for networks in which segment losses and corruptions are mainly due to congestion. TCP utilizes a flow control mechanism by which the sender host doesn't overflow the receiver host by transmitting too much

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data and too fast. In order to avoid long delays when there is no response from the receiver in a TCP connection, a time-out mechanism is employed. Besides a congestion control mechanism is used in TCP to avoid packet drop due to lack of resources and buffer space.

In wireless systems most of errors are due to lossy media. The reason is that in the wireless channels the main cause for the packet loss may be the high BER in the channel not the network congestion. So the low efficiency of the TCP in a wireless channel is a direct result of the fact that the TCP misinterprets the packet loss resulting from high channel error rate or from the congestion. In order to enhance QoS seen by TCP layer on a wireless link, a radio link control (RLC) is generally introduced at link layer. Typically the RLC uses an ARQ error recovery mechanism to improve the QoS (3GPP TS 25 322, 2007).

RLC is a protocol above MAC and blew RRC. Every outgoing TCP packet is put into an interface buffer which is picked up by the RLC. RLC is responsible for error and flow control (by ARQ mechanism) of the frames and provides transparent mode (TM), unacknowledged mode (UM) and acknowledged mode (AM) services. The RLC breaks the TCP packet into 10-ms frames and sends them to MAC. MAC chooses a user queue according to the scheduling mechanism and after adding a MAC header sends them to the Physical layer (Chockalingam and Zorzi, 1999) and (Borgonovo, 2001). In this paper we review the ARQ protocol effects on the TCP throughput and show how it improves the throughput. In section 2 we define a system model and in section 3 we simulate the TCP and TCP/ARQ protocols in wired and wireless systems. Finally conclusion will be offered.

2 SYSTEM MODEL

TCP is in layer 4 and locates in the hosts in end nodes but isn't a part of UMTS network. Implementations of TCP contain four intertwined algorithms: slow start, congestion avoidance, fast retransmit and fast recovery (RFC, 2001). Although TCP has been designed, optimized and tuned in wired networks to react to the packet loss due to congestion, in wireless systems service degradation can be due to bit (packet) errors. In UMTS, TCP and ARQ protocols operate against loss and error in wired and wireless sections respectively.

TCP in a wireless network experiences several challenges. One of the issues is how to deal with the spurious timeout caused by the abruptly increased delay, which triggers unnecessary retransmission and congestion control. It is known that the link-layer error recovery scheme, the channel scheduling algorithm, and handover often make the link latency very high. Bandwidth of the wireless link often fluctuates because the wireless channel scheduler assigns a channel for a limited time to a user. Thus, the variance of inter-packet arrival time becomes high, which may result in spurious timeout. The Eifel algorithm has been proposed to detect the spurious timeout and to recover by restoring the connection state saved before the timeout (Wennstrom, 2004) and (Gurtov, 2003).

Although the packet loss rate of the wireless link has been reduced due to link-layer retransmission and Forward Error Correction (FEC), losses still exist because of the poor radio conditions and mobility. Therefore, non-congestion errors could sharply decrease the TCP sending rate. Packet reordering at the TCP layer may be caused by link-layer retransmission, which also results in unnecessary retransmission and congestion. In the wireless networks, in general, bandwidth and latency at uplink and at downlink directions are different. Hence, the throughput over downlink may be decreased because of ACK congestion at the uplink (Lee, 2006).

Now we consider a TCP connection between two hosts such that the first link on the end-to-end path from the sender to the receiver is a wireless radio link (Lee, 2006) and (Canton 2001). Such a scenario is common in mobile communication and is illustrated in figure 1(a). The protocol stack on the way from mobile host to fixed host is illustrated in the figure 1(b).

We assume there is no packet loss due to congestion on the wireless link but some packets may be corrupted under adverse radio link conditions. In our study, we assume that the bit error patterns on the radio link are independent. On the wired network, packets may only get lost when congestion occurs.

As described in (Lee, 2006) and (Chahad, 2003) we assume that TCP sends one cumulative ACK_{TCP} for b consecutive TCP segments and is always in congestion avoidance. Besides, Packet loss is detected in one of the two ways, either upon reception of a triple-duplicate ACK_{TCP} (denoted by TD), or upon expiration of a Time-Out (denoted by T0). In case of a TD, window size is decreased by half, while upon expiration of a T0, it is decreased to 1. Moreover, we assume that the loss behavior is bursty, i.e., packet losses are correlated within a back-to-back transmission. Hence, when a packet is lost, all remaining packets in the same round are lost as well (Padhey, 1998). Furthermore, under the assumption that rounds are separated by each TCP round trip time, RTT_{TCP}, loss in one round is independent of loss in other rounds.

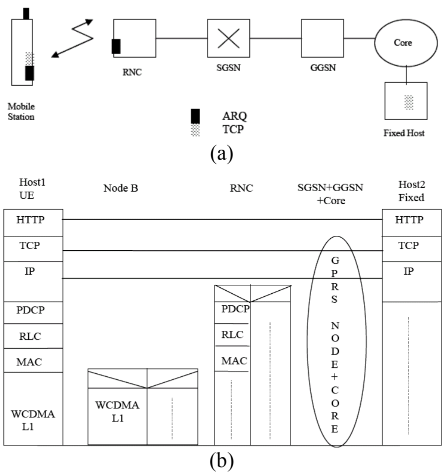

Figure 1: (a) An end-to-end system (b) The protocol stack on the way.

We consider TCP Reno version. Let T0 denote the TCP time-out and p denote the loss rate due to congestion in the wired portion of the network. For the steady-state of TCP throughput, in a wired context only we have (Canton, 2001) and (Chahad, 2003)

$$Th(p) = \frac{1}{RTT_{TCP}\sqrt{\frac{2bp}{3}} + T_0 \min(1, 3\sqrt{\frac{3bp}{8}})p(1+32p^2)} \quad (1)$$

and for an end to end protocol consisted of both wireline and wireless channel we have

$$[Th(p,PER)]^{-1} = RTT_{TCP}\sqrt{\frac{2b(p+PER-p*PER)}{3}} + T_0 \min \left(1, 3\sqrt{\frac{3b(p+PER-p*PER)}{8}}(p+PER-p*PER)\times (1+32(p+PER-p*PER)^2)\right) \quad (2)$$

which is the same as (1) in which p is substituted by

$$Global\ Average\ Packet\ Loss\ Rate = 1-(1-PER)(1-p) = p+PER-p*PER \quad (3)$$

In addition we have packet error rate as in equation $PER = 1-(1-FER)^n$ in which FER is frame error rate in wireless section and *n* is the number of frames in a packet. Equation (2) is for a complete (wireline and wireless) system without ARQ mechanism.

If we consider a Go Back-N mechanism in layer 2 and also the case of independent and identically distributed bit errors (i.i.d.), we have the throughput as (Wennstrom, 2004)

$$[Th(p,FER)]^{-1} = \begin{pmatrix} RTT_{wire} + nbD_{ARQ} + RTT_{wless} + \\ ND_{ARQ}\frac{FER(nb-1)}{1-FER}\sqrt{\frac{2bp}{3}} \end{pmatrix} + T_0 \min (1, 3\sqrt{\frac{2bp}{8}})p(1+32p^2) \quad (4)$$

in which D_{ARQ} is the constant delay component as a result of ARQ frame processing and RTT_{wless} and RTT_{wire} are round trip times for ARQ (air channel from UE to RNC and vice versa) and wired sections respectively.

Now we consider a mobile channel where a subscriber moves in it. It is surrounded by some obstacles which incident rays strike them. We compute BER as described in (Jeruchim, 2000) which has been used in MATLAB as a reference for fading channel simulation. There, it is modeled as a linear FIR filter with tap weights given by

$$g_n = \sum_k h_k \operatorname{sinc}(\tau_k / T - n) \qquad for \quad -N_1 \le n \le N_2 \quad (5)$$

where

- the summation has one term for each major path.
- { τ_k } is the set of path delay.
- T is the input sample period.
- N_1 and N_2 are chosen so that $|g_n|$ is small when n is less than - N_1 or greater than N_2.
- { h_k } is the set of complex path gain which are not correlated with each other.

3 SIMULATION

We used RLC in acknowledged mode (AM) with a Go-Back-N mechanism. Besides we used a TCP packet with 120 bytes length then segmented it into 10 frames with 12 bytes length each. We began finding the ordinary TCP throughput in the wired section. The TCP parameters are as $RTT_{wire} = 0.2 \quad b = 10 \quad T0 = 0.4$. Then by MATLAB we made a slow fading channel and calculated the

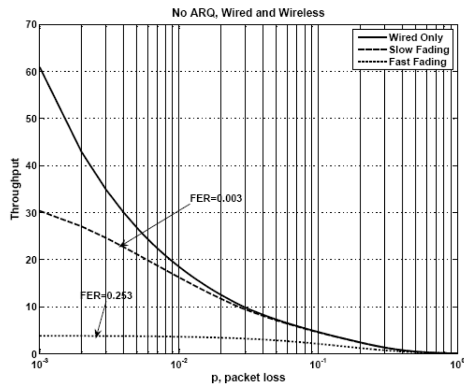

Figure: 2 Effect of air channel on TCP throughput (kbps).

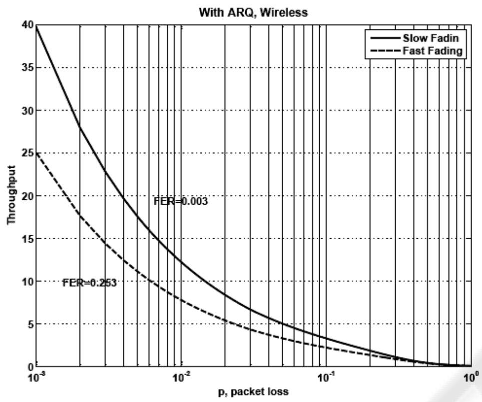

Figure: 3 Effect of air channel on TCP/ARQ throughput (kbps).

TCP throughput in it. Here we considered a pedestrian user with the speed of 3m/s (slow fading, $f_d = 20Hz$), then we considered a mobile user with the speed of 81km/h (fast fading, $f_d = 100Hz$) and found the throughput as shown in Figure 2. It consists of three kinds of curves. These are related to wired, slow and fast fading channels with No ARQ protocol. We see fading and FER effects on the throughput. The least throughput is related to a fast fading multipath channel where a user moves with the speed of 81km/h.

Then in another scenario like mentioned above we found the TCP/ARQ throughput with Go-Back-N mechanism as shown in Figure 3. We used Go-Back-N mechanism with the following parameters.

$RTT_{ARQ} = 0.005 \quad D_{ARQ} = 0.001 \quad n = 10 \quad N = 10$

ARQ protocol effects on throughput in slow and fast fading channels are illustrated in figure 3. The throughput improvement related to ARQ protocol is obviously observed on it.

4 CONCLUSIONS

We could compensate the TCP deficiency in wireless channels by ARQ protocol in the layer two. Besides we observed that although a fast fading channel degrades the system throughput more than a slow fading channel but ARQ protocol improves the throughput more.

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