Chapter 21

Effects of Packet-Loss and Long Delay Cycles on the Performance of the TCP Protocol in Wireless Networks

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ABSTRACT

Many analytical models have been developed to evaluate the performance of the transport control protocol (TCP) in wireless networks. This chapter presents a description, derivation, implementation, and comparison of two well-known analytical models, namely, the PFTK and PLLDC models. The first one is a relatively simple model for predicting the performance of the TCP protocol, while the second model is a comprehensive and realistic analytical model. The two models are based on the TCP Reno flavor, as it is one of the more popular implementations on the Internet. These two models were implemented in a user-friendly TCP performance evaluation package (TCP-PEP). The TCP-PEP was used to investigate the effect of packet-loss and long delay cycles on the TCP performance measured in terms of sending rate, throughput, and utilization factor. The results obtained from the PFTK and PLLDC models were compared with those obtained from equivalent simulations carried-out on the widely used NS-2 network simulator. The PLLDC model provides more accurate results (closer to the NS-2 results) than the PFTK model.

DOI: 10.4018/978-1-60960-887-3.ch021
INTRODUCTION

The TCP is the dominant transport layer protocol in the Internet Protocol (IP) suite. It carries a significant amount of the Internet traffics, such as Web browsing, files transfer, e-mail, and remote access. It is a reliable connection-oriented protocol that allows a byte stream originating on one machine to be delivered without error to any other machine on the Internet. (Forouzan 2007). An Internet work differs from a single network because different parts may have different topologies, delays, bandwidths, packet sizes, and other parameters. The TCP was designed to be dynamically adaptable to outfit Internet work, robust in the face of many kinds of failures, handle flow control to make sure a fast sender cannot swamp a slow receiver with more messages than it can handle, and support full duplex and point-to-point connections (Tekala & Szabo 2008).

The TCP does not support multicasting or broadcasting, and it only support point-to-point connection in which the sending and receiving TCP entities exchange data in the form of TCP segments. The size of the segments are decided by the TCP software, which decides how big the segments should be, and can accumulate data from several writes into one segment or can split data from one write over multiple segments. Two parameters restrict the segment size, these are: the IP payload (each segment must fit in the 65515 Byte IP payload), and the network maximum transfer unit (MTU) (each segment must fit in one MTU).

In practice, the MTU is generally 1500 bytes (the Ethernet payload size) and thus defines the upper bound on segment size. Segments can arrive out of order, so that some segment arrives but cannot be acknowledged because earlier segment has not turned up yet. Segments can also be delayed so long in transit that the sender times-out and retransmits them. The retransmissions may include different segment size than the original transmission. TCP must be prepared to deal with these problems and solve them in an efficient way. The definition of the components of the TCP segment header and more details on TCP protocol can be found in many computer networks textbooks and literatures (Tanenbaum 2003).

The TCP was initially designed for wired networks for which a number of notable mechanisms have been proposed in the literature to improve the performance of TCP in such networks. For most of these mechanisms, analytical models have developed to predict and investigate their performance in wired networks in terms of the sending rate \(S\) and throughput \(T\), and utilization factor \(U\). Unfortunately, these mechanisms demonstrated a poor performance in wireless networks due to presence of packet-loss (PL) and long delay cycles (LDC) in such environment (Adarbah 2008).

Presence of LDC leads to spurious retransmissions (STs) or spurious fast retransmissions (SFRs), which produce serious end-to-end TCP performance degradation. However, since the emergent of wireless networks, new mechanisms have developed to enhance the performance of TCP in presence of STs and SFRs (Abouzeid & Roy 2003, Chen et al. 2008). Consequently, new and adequate analytical models need to be developed to accommodate these new TCP mechanisms. This is because none of the existed models for wired networks considers the effect of STs and SFRs on the steady-state performance of TCP. This due to the fact that STs and SFRs do not occur frequently in wired networks, and also STs and SFRs are considered to be a transient state in a wired network, and thus cannot produce much impact on the steady-state performance of TCP (Yang 2003). Practically, in wireless networks, STs and SFRs are more frequent and must be explicitly modeled to accurately estimate \(S\), \(T\), and \(U\) of the TCP.

There are a number of mathematical models that have been developed throughout the years for evaluating the performance of a TCP connection.