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TCP performance analysis and optimization over DMT based ADSL system

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Abstract

This paper studies the transmission control protocol (TCP) performance over a discrete multi-tone (DMT) based asymmetric digital subscriber loop (ADSL) network. The impact of DMT subchannel bit loading on the TCP throughput performance is studied. The simulation results show that there is a threshold for the signal-to-noise ratio (SNR) gap or bit error rate (BER) above which TCP throughput drops quickly. This threshold takes its value in a wide range depending on the TCP round-trip time as well as channel noises. This suggests that it would be insufficient to set a fixed target BER at, e.g. 10^{-7}, when calculating the number of bits to be loaded in each subchannels. Instead, the bit loading should take TCP performance into account. Finally a dynamic bit loading scheme is proposed, which jointly optimizes the channel bit rate and TCP throughput performance.

Keywords: Asymmetric digital subscriber loop; Discrete multi-tone; Water-filling; Transmission control protocol; Performance analysis

1. Introduction

As an emerging technology, asymmetric digital subscriber loop (ADSL) has attracted a lot of attention due to its ability to deliver broadband access over the traditional telephone lines. However, unlike a backbone network based on fiber optics technology with a bit error rate (BER) on the order of 10^{-11}–10^{-13}, an ADSL system has to live with potentially high and variable BER, ranging from 10^{-3} to 10^{-9}. Except the ADSL channel characteristics, four major noises contribute to the high and variable BER, including white Gaussian noise, far end crosstalk (FEXT), near end crosstalk (NEXT), and impulse noise.

Tremendous research efforts have been made in the analysis of the impact of channel noises on the ADSL performance, e.g.[10,15], and in the design of loading and dynamic loading algorithms to optimize the channel performance [7,9]. Of particular interest is the rate-adaptive (RA) loading algorithm [9] for ADSL systems based on discrete multi-tone (DMT) modulation. The RA loading algorithm maximizes the overall bit rate subject to a fixed energy constraint and signal-to-noise ratio (SNR) gap or BER.

Most of the data applications are built on top of the transmission control protocol (TCP). Therefore, in parallel to the above development, research effort has been made on the study of the performance of TCP over asymmetric and lossy channels [1,14,16]. By assuming that the channel bit rates in both directions are given, these papers investigate the effects of buffering [16], asymmetry [4], and random loss or BER [5,18] on TCP throughput performance.

However, none of the studies mentioned above considered both the physical layer (ADSL) and the upper layer (TCP) performance simultaneously. Existing TCP performance papers mentioned above assume the underlying channel conditions are given. For instance, by setting BER at 10^{-7} and the maximum bit rates at 8 Mbps downstream and 800 kbps upstream, respectively, the TCP performance can then be independently evaluated regardless of the actual underlying ADSL processes. In reality, however, the maximum bit rates are complicated functions of BER as well as subchannel SNRs, which may change from time to time. Hence, analyzing TCP performance over ADSL should take the physical channel processes into account. On the other hand, the research on the loading and dynamic loading algorithm design for DMT modulation did not take higher layer performance into account. Although it is mentioned in Ref. [9] that some higher-layer entity may arbitrate when the reloading should occur for dynamic loading, the question as to which higher layer entity and how a higher layer entity makes the reloading decision is not addressed. Since the objective of

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performance optimization at any given layer is to deliver the best service to its upper layer, TCP performance should be the ultimate performance measure for applications using TCP as their underlying protocol.

In this paper, the performance analysis of TCP over ADSL is performed by taking into account of the underlying ADSL processes. This approach enables us to find the true TCP *performance limit* for different standard ADSL test loops and under various noise conditions. The performance analysis further leads to the development of a bit loading scheme which *jointly* optimizes channel bit rate and TCP throughput performance. The idea is first to take BER as a variable, rather than a fixed target, and run the RA bit loading algorithm [9] to find the functional relationship between the maximum bit rate and BER. Then locate an operating point on the curve of the maximum bit rate versus BER, which maximizes the TCP throughput performance. Finally, load the subchannels with the energies or numbers of bits per symbol calculated at this operating point. Obviously, at this operating point, both physical layer and TCP throughput performance are jointly optimized.

The rest of the paper is organized as follows. Section 2 presents a background introduction to DMT based ADSL systems. Section 3 presents the performance evaluation of TCP performance on two test loops of a ADSL system. Section 4 describes a joint optimization scheme to maximize the TCP performance. Finally, Section 5 concludes the paper and presents a future research direction.

2. Background introduction

This section gives the necessary background on the ADSL technology, DMT modulation, and TCP protocol.

2.1. Asymmetric digital subscriber line

ADSL is a standardized transmission technology facilitating simultaneous use of normal telephone services and data transmission. ADSL can be seen as a frequency division multiplexing (FDM) system in which the available bandwidth of a single copper-loop is divided into three subbands. The baseband of 4 kHz is used for analog voice telephony. The band between 25 and 138 kHz is for upstream data transmission. The band between 200 and 1100 kHz is for downstream data transmission. The lower cutoff frequency for downstream data can be extended down to the lower frequency of the upstream data if echo cancellation is used. According to ANSI standard, ADSL should run at a minimum of 6.144 Mbps downstream and 640 kbps upstream over the existing copper telephone lines [2,6].

Basically, two types of modulation schemes can be used for ADSL modems: carrierless amplitude-phase (CAP) and DMT. Since the DMT modulation technique is chosen by ANSI as the standard modulation scheme for ADSL, in this paper, we consider only DMT modulation. Interested readers can refer to Refs. [7,8] for more information about CAP.

2.2. DMT

The basic idea of DMT is to divide the available bandwidth into a fixed number of N parallel, independent subchannels. Quadrature amplitude modulation (QAM) is used for each subchannel. Different numbers of bits can be assigned to different subchannels. Subchannels with larger SNR carry more data and those with smaller SNR carry less data. The algorithm which achieves the overall maximized bit rate is the RA bit loading algorithm, which will be introduced shortly. A detailed explanation of DMT can be found in Ref. [7].

Each subchannel's SNR is given by

$$\mathrm{SNR}_i = \frac{E_i|H_i|^2}{\sigma_i^2}, \tag{1}$$

where E_i is signal energy of the ith subchannel, $|H_i|^2$ is the power spectral density of the ith subchannel, and σ_i^2 is the noise variance for the ith subchannel. Therefore, the number of bits per dimension (QAM has two dimensions) carried in the ith subchannel is given by

$$b_i = \tfrac{1}{2}\log_2\left(1 + \frac{\mathrm{SNR}_i}{\Gamma}\right), \tag{2}$$

and the total number of bits, $\bar{b}$, that can be sent over the channel is the sum of the number of bits on the used subchannels,

$$\bar{b} = \sum_{i=1}^{N}\log_2\left(1 + \frac{\mathrm{SNR}_i}{\Gamma}\right), \tag{3}$$

where Γ is the SNR gap which measures the SNR loss from the theoretical maximum channel capacity. A 0 dB gap ($\Gamma = 1$) means that the channel capacity is achieved. With the symbol error probability P_e fixed, Γ is approximately a constant, independent of the number of bits per symbol with fixed minimum QAM distance. For QAM, we have

$$P_\mathrm{e} \simeq N_\mathrm{e}Q(\sqrt{3\Gamma}), \tag{4}$$

where N_e is the number of nearest neighbors of an input signal constellation for the ith subchannel, and

$$Q(x) = \frac{1}{\sqrt{2\pi}}\int_x^\infty \mathrm{e}^{-u^2/2\,\mathrm{d}u}.$$

For uncoded QAM transmission, Γ is found to be 9.8 dB, which means that the SNR is reduced by that amount to achieve the probability of error of 10^{-7}. Γ can be reduced by the coding gain [17].

Therefore, the achievable bit rate B can be calculated by dividing the total number of bits with the symbol period T:

$$B = \frac{1}{T}\bar{b} = \frac{1}{T}\sum_{i=1}^{N}\log_2\left(1 + \frac{\mathrm{SNR}_i}{\Gamma}\right). \tag{5}$$

Bit loading algorithms calculate the bit or energy distribution for subchannels. There are two types of loading algorithms: RA loading algorithm and margin-adaptive (MA)