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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

$$P_e \simeq N_e Q(\sqrt{3\Gamma}), \tag{4}$$

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

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

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)

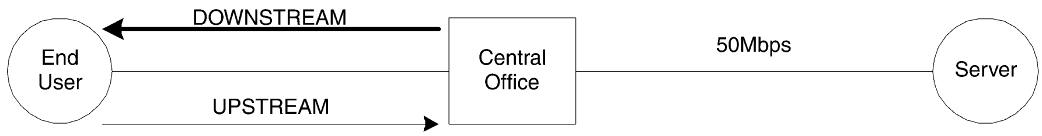

Fig. 1. Network Model.

loading algorithm. The RA loading algorithm maximizes the data rate with given energy constraint and the MA loading algorithm minimizes the energy with given data rate. The MA loading algorithm is desirable when the data rate cannot be changed. Since in this paper, the focal point is on the identification of TCP performance limit over ADSL links, we only consider the RA loading algorithm.

2.3. Rate-adaptive loading algorithm

The RA loading algorithm maximizes the number of bits per dimension subject to a fixed energy constraint:

$$\max_{E_i} \bar{b} = \sum_{i=1}^{N} \log_2\left(1 + \frac{E_i g_i}{\Gamma}\right),$$

subject to

$$\sum_{i=1}^{N} E_i \leq E_{\text{total}},$$

where $g_i = |H_i|^2/\sigma_i^2$, which represents the SNR with a unit energy of a subchannel.

It is proven in Ref. [7] that the RA design objective is achieved when

$$E_i + \frac{\Gamma}{g_i} = \text{constant}.$$

The term 'water-filling' is used to describe this solution. The curve of Γ/g_i is being a bowl into which the water (energy) is poured, filling the curve until all the energy is used.

In the application of the RA loading algorithm to ADSL systems, the subchannel data rates can be any multiple of 4 kbps with RA loading. Usually in these systems, the gap is intentionally increased by 6 dB. With nominally 100–250 tones being used and average numbers of bits per tone of 2–9 bits, data rates from 1 *Mbps* — 9 Mbps are possible.

2.4. Noises

In this paper, we consider White Gaussian Noise, NEXT and FEXT but not the Impulse Noise, which is assumed to be removed by error correction codes. In what follows, we briefly describe NEXT and FEXT.

2.4.1. NEXT noise

NEXT is the noise caused by other transmitters at the same end of the cable as the source. NEXT is usually considered for full duplex systems such as T1 originally deployed in the trunk network, the DSL used for ISDN Basic Access, and the HDSL used to support DS1 rate access. In general, the PSD of the NEXT can be expressed as

$$\mathrm{PSD}_{\mathrm{NEXT}} = \mathrm{PSD}_{\mathrm{Disturber}} x_n f^{3/2} \tag{6}$$

where $x_n = 8.818 \times 10^{-14}(n/49)^{0.6}$ and n is the number of disturbers. The $\mathrm{PSD}_{\mathrm{Disturber}}$ is the PSD of the disturber. The PSDs of other disturbers used in this paper can be found from Ref. [2]:

For example, the PSD of an HDSL disturber can be expressed as [2]:

$$\mathrm{PSD}_{\mathrm{HDSL}} = K_{\mathrm{HDSL}} \frac{2}{f_0} \frac{\left[\sin\left(\frac{\pi f}{f_0}\right)\right]^2}{\left(\frac{\pi f}{f_0}\right)^2} \frac{1}{1 + \left(\frac{f}{f_{3\,\mathrm{dB}}}\right)^8}, \tag{7}$$

where $f_{3\,\mathrm{dB}} = 196$ kHz, $f_0 = 392$ kHz, $K_{\mathrm{HDSL}} = (5/9) \times (V_p^2/R)$, $V_p = 2.70$ V, and $R = 135\ \Omega$.

2.4.2. FEXT noise

FEXT is the noise effect of the other transmitters at the far end of the cable of the source. The effect of ADSL self-FEXT cannot be ignored. At high frequency range ADSL self-FEXT noise power can exceed that of HDSL NEXT and white background noise combined. In general, the PSD of the FEXT can be expressed as

$$\mathrm{PSD}_{\mathrm{FEXT}} = \mathrm{PSD}_{\mathrm{Disturber}} |H_{\mathrm{channel}}(f)|^2 klf^2 \tag{8}$$

where $H_{\mathrm{channel}}(f)$ is the channel transfer function, k is defined as $8 \times 10^{-20}(n/49)^{0.6}$ which is called the coupling constant, and l is the coupling path length.

2.5. TCP congestion control protocol

In this paper, we study two versions of TCP congestion control protocols, i.e. TCP-Reno and TCP-Tahoe. TCP-Tahoe is the current standard version of TCP and TCP-Reno provides fast retransmission and reduces the number of the slow start occurrences.

2.5.1. TCP-Tahoe

As it is explained in Ref. [18], in TCP-Tahoe, a packet loss is detected by a timer based on the estimated round trip time (RTT). When a packet loss is detected, the window size

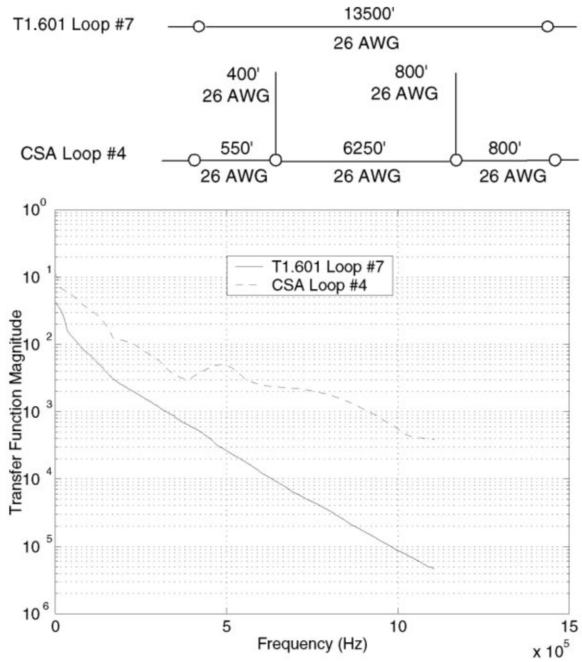

Fig. 2. Two standard test loops.

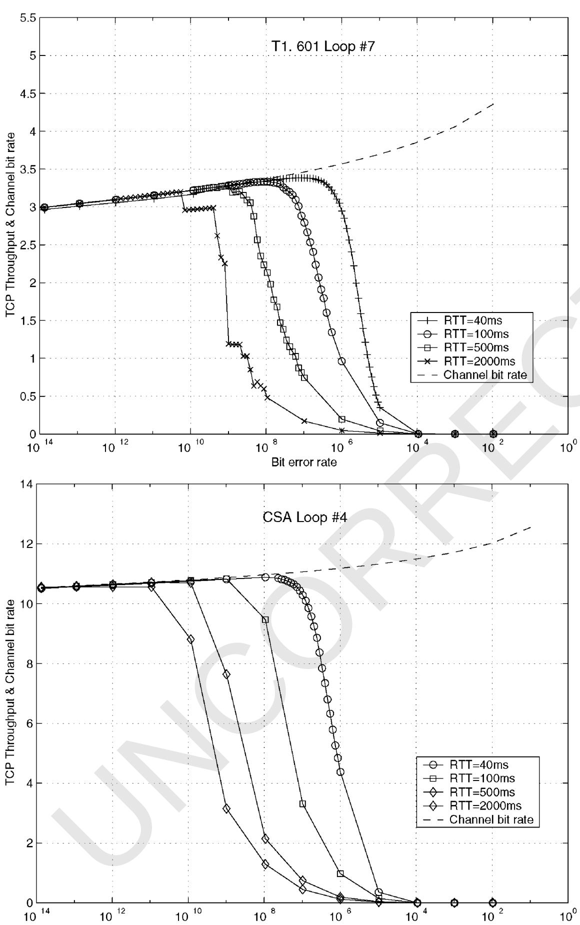

Fig. 3. Different RTT times.

W is reduced to one. In the slow start phase, the window grows rapidly for every successfully acknowledged packet until it reaches half of the window size. The algorithm then switches to the congestion avoidance phase, probing for extra bandwidth by incrementing the window size by one for every window's worth of acknowledged packets. This growth continues until another packet loss is detected, at which point another cycle begins.

2.5.2. TCP-Reno

TCP-Reno retransmits the packet after the number of duplicate acknowledgements exceeds a threshold. However, instead of cutting the window size back to one, it reduces the window size by half. Further, in order to prevent a burst of packets from being transmitted when the retransmission is finally acknowledged, it temporarily permits new packets to be transmitted with each repeated acknowledgement until the 'next expected' number in the acknowledgement advances.

3. Simulation results and analyses

This section presents the performance analysis of TCP over ADSL based on simulation. We run simulations using ns-2 network simulator tool developed by VINT research group at University of California, Berkeley [13].

3.1. Experiment setup

3.1.1. Network model

The network topology is composed of three nodes: a server, a central office and an end user, as shown in Fig. 1. The link between the end user and the central office represents the ADSL local loop and the other error-free link is a simplified model of the links between the central office and the server across the backbone networks. The server and the end-user communicate using TCP. We use a file transfer protocol (FTP) connection to generate data packets from the server to the end-user, and assume that the server has always packets to send. Each data packet is assumed to be 8000 bit-long. The end-user does not send any data packets but just sends acknowledgements back to the server.

The connection between the server and the central office has fixed bandwidth of 50 Mbps which is at least around five times higher than the ADSL link supporting up to 10 Mbps data rate downstream. The data rate ratio between ADSL downstream and upstream is assumed to be 8:1. The length of the link between the central office and the server is changed to run the simulation at different RTT. A 20 ms Reed-Solomon coding/decoding delay is assumed to be added to the RTT. The central office has a finite size first in first out (FIFO) buffer. The size of the buffer is proportional to the bandwidth-delay product of the network [18]. Here, the bandwidth refers to the channel rate of the ADSL link and the delay in RTT.

Table 1
TCP throughput with different RTT (unit: Mbps)

RTT (ms)	Reno/Tahoe	
	Test Loop: T1.601	Test Loop: CSA #4
40	2.31/2.04	7.42/6.65
100	2.16/1.95	3.24/2.89
200	1.57/1.41	1.66/1.50
500	0.72/0.61	0.73/0.63
2000	0.16/0.15	0.16/0.15

3.1.2. ADSL link model

In this study, we use two standard ADSL test loops and their transfer functions [3] to characterize the ADSL links, as shown in Fig. 2.

The first test loop, T1.601 Loop #7, has a bridged tap with 13 500 feet in length and a American wire gauge (AWG) twisted pair cable is used. The second test loop, CSA Loop #4, has three bridged taps with 7600 feet in length and a 26 AWG twisted pair cable is used. The CSA Loop #4 is substantially shorter than T1.601 #7 and is generally used as a test loop for very high speed DSL. Here we use it as an extreme case for ADSL.

Assume that the system will drop those packets which have bits in error that cannot be corrected. We consider a

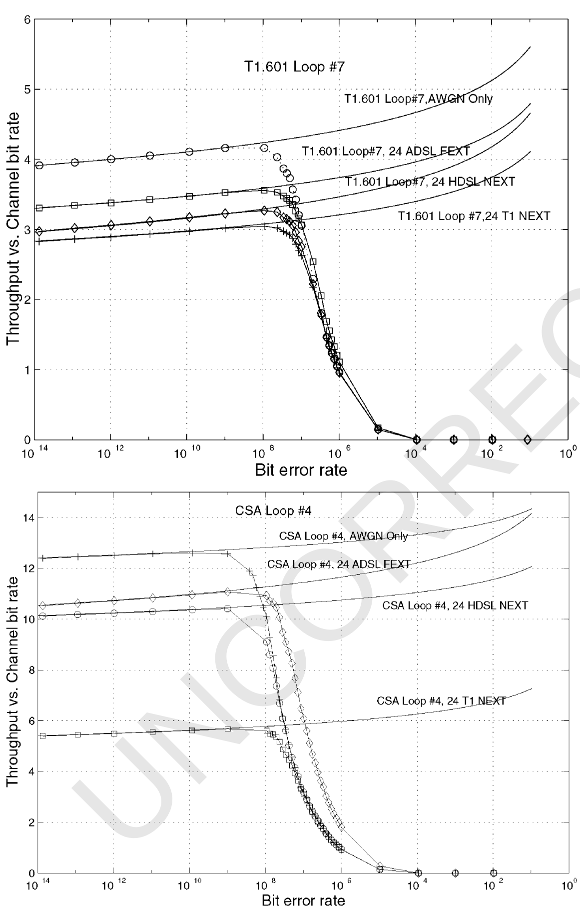

Fig. 4. TCP performance with different types of crosstalk.

scenario where packets incur random loss, i.e. the bit error is generated independently for each bit based on the given BER and the error pattern is assumed to be the same for both the receiver and sender sides. As a result, the simulation results in our paper represents the worst-case TCP performance. In our simulation, we focus on studying the impact of the TCP RTT time since the RTT can be changed significantly between different TCP sessions. Other parameters like TCP SCK and MSS are usually pre-configured so that have little impact on the different TCP sessions.

3.2. Simulation results and analysis

Throughout the simulation studies, we assume that the input power is fixed at 20 mW, the background additive white Gaussian noise (AWGN) is fixed at -140 dBm/Hz, and a powerful trellis code is used which achieves a 5 dB coding gain.

3.2.1. Impact of RTT

Assume there is no crosstalk. The RA loading algorithm mentioned in the Section 2 results in optimal bit loading or maximum data rate at any given Γ. By changing Γ from 3 dB to 13 dB (or equivalently, the BER from 10^{-2} to 10^{-14}), the maximum channel bit rate can be obtained from the RA bit loading algorithm. The TCP throughput under different maximum channel rates and BERs can then be calculated for different RTTs.

Fig. 3 shows the simulation results for both T1.601 Loop #7 and CSA Loop #4 when TCP-Reno is used. Interesting enough, for both cases, there is a sharp threshold BER beyond which TCP throughput starts to deviate from the maximum allowable bit rate of the channel and it drops quickly to zero as the BER further increases. This phenomenon is found in all the cases of different RTT values. Moreover, for RTT in between 40 ms and 2 s, the range in which the threshold occurs is pretty large, ranging from 10^{-10} to 10^{-6} and 10^{-11} to 10^{-7} for T1.601 Loop #7 and CSA Loop #4, respectively. This observation clearly demonstrates that care must be taken in bit loading to ensure that the selected operating point is below the threshold.

One possible solution is to select a safe Γ or sufficiently small BER for bit loading, say, BER $= 10^{-11}$. This conservative approach offers a simple solution at the expense of up to 15% TCP throughput loss. Another approach is to take advantage of the dynamic loading capability of DMT to achieve higher throughput performance. Since subchannel characteristics and SNRs may change with time, dynamic loading is desirable. In Section 4, we shall propose a scheme to enable dynamic loading.

To better observe the impact of RTT, Table 1 lists the TCP throughput for both TCP-Reno and TCP-Tahoe with fixed BER at 10^{-7} and the corresponding maximum channel bit rates at 2.45 and 10.4 Mbps for T1.601 Loop #7 and CSA Loop #4, respectively. One observes that the TCP throughput drops rapidly as the RTT increases.

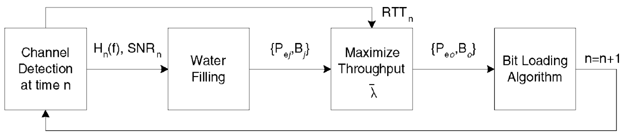

Fig. 5. Optimization process.

3.2.2. Impact of noises

In this section, we study the relatively impact of different noise types on the TCP throughput performance. The performance is evaluated with NEXT from HDSL and T1 lines as well as FEXT from other ADSL lines and end-to-end RTT is fixed at 100 ms. In all case studies, 24 disturbers are present. As the references, the maximum channel bit rates are also plotted. The results for both T1.601 Loop #7 and CSA Loop #4 are presented in Fig. 4.

One observes that these noises reduce the achievable channel bit rate and thus the TCP throughput. The dynamic range of the threshold due to different noise is much smaller than that due to the RTT difference. One also notes that T1 NEXT is the worst disturber for the throughput performance. This is due to the fact that the magnitude of the T1 NEXT PSD is significant over the entire downstream ADSL bandwidth.

4. Joint optimization of channel bit rate and TCP throughput

As we have seen from Section 3, due to the RTT and noise effects, the RA loading at a fixed Γ value or BER can result in poor TCP throughput performance. The problem can be compounded when the subchannel characteristics and SNRs change with time. Cioffi [9] proposed a dynamic RA loading algorithm assuming that some higher layer entity arbitrates when the adaptation should take place. However, the author did not address the issue as to which higher layer entity and how higher layer entity arbitrates. In this section, we propose a control scheme which jointly optimizes the channel bit rate and TCP throughput performance. The higher layer entity is TCP protocol at the transport layer.

The optimization problem can be formalized as:

$$\max_{P_e} \lambda(P_e, B(P_e)), \tag{9}$$

where λ is TCP throughput and $B = B(P_e)$ is obtained from the water-filling algorithm. Obviously, the optimal solution $\{B^*, P_e^*\}$ jointly maximizes the TCP throughput performance as well as the channel bit rate B at $P_e = P_e^*$.

The optimization process can be described as follows:

1. Measure the subchannel characteristic H_i and SNR_i, and estimate RTT of the TCP connection in use.
2. By changing Γ or P_e and using the water-filling algorithm, find $B = B(P_e)$.
3. Find $B^* = B(P_e^*)$ which maximizes the TCP throughput λ under the measured RTT.
4. Load the bits or distribute energy to subchannels according to the solution of the water filling algorithm at $\{B^*, P_e^*\}$.

The control scheme can be also seen in Fig. 5.

The key issue is how to find the maximum TCP throughput. A searching of TCP throughput in the parameter space $B = B(P_e)$ obtained from the water filling algorithm based on simulation is time consuming. Here, we propose to use an analytical result on the TCP throughput by Lakshman and Madhow [18]. Their result establishes the relationship between B, P_e and RTT of a TCP session. Our numerical test of their approach against the simulation results showed that their result is sufficiently accurate in capturing the threshold of TCP throughput.

To locate the optimal point, the maximum TCP throughput is searched. Since there are no local maxima, any fast standard search algorithms can be used. Each iteration involves an iterative search of B at a given P_e using water-filling algorithm and an one-dimensional integral to find λ. Our numerical analysis showed that the running time is on the order of seconds on the SPARC-10 SUN workstation.

5. Conclusions and future work

In this paper, the performance of TCP over DMT based ADSL is studied. In contrast to the previous papers, this study did not assume that the underlying ADSL channel conditions are fixed. Instead, it explicitly considered the impact of ADSL channel conditions on the TCP throughput performance. This study enabled us to identify the limit of TCP throughput performance under different standard ADSL test loops and various noise conditions. This study further lead to the design of a dynamic bit loading scheme which jointly optimizes the channel bit rate and TCP throughput performance.

However, the study only considered the TCP throughput as a performance measure. A possible extension of this work is to consider multiple classes of service. Results on providing multiple classes of service on the ADSL transmission line are available [11,12]. Our future work is to address the issue as to how the physical layer service differentiation is to be related to the higher layer service differentiation.

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