Resource Provisioning in a Multi-Service Enabled ADSL System

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Abstract— This paper addresses the resource provisioning issue for the Discrete Multi-Tone (DMT) based asymmetric digital subscriber loop (ADSL) network. Currently, the provisioning scheme for ADSL network provisions the network purely based on the physical layer performance. However, provision the network at the physical layer cannot provide assurance to TCP/UDP quality. The simulation results presented in our previous work showed that there is a threshold for the bit error rate (BER) above which TCP (transport control protocol) throughput drops quickly. This threshold takes its value in a wide range depending on the TCP round-trip time as well as channel noises etc. A simple solution is to set sufficient noise margin so that the BER is low enough such that the threshold effect is avoided. However, setting a large noise margin itself reduces bit loading rate and thus TCP throughput performance. In this paper, we propose an new idea of provisioning the resource based on the TCP/UDP performance. Based on the joint optimization scheme of bit loading rate at the physical layer and TCP throughput performance at the transport layer which we have discussed in the previous paper. The architecture provides Quality of Service (QoS) guarantee for voice traffic and throughput assurance for TCP applications. Moreover, this architecture provides systematic procedures to allow an Internet service provider to establish Service Level Agreement (SLA) with its customers. A case study is presented to demonstrate how the provisioning scheme works.[1]

Keywords— **ADSL, DMT, VoDSL, TCP, Resource Provisioning, Performance Analysis, Bit Loading Algorithm**

I. Introduction

As one of the two most popular residential broadband technologies, asymmetric digital subscriber loop (ADSL) has attracted tremendous attention due to its capability to deliver broadband access over the traditional telephone lines. By using the spectrum above 4 KHz, an ADSL link is able to provide data and voice services simultaneously in addition to the traditional baseband telephone service. In particular, the capability of DSL in general to carry voice traffic (i.e., Voice over DSL or VoDSL) offers a competitive edge for competitive local exchange carriers (CLECs) to compete in the lucrative voice market, traditionally dominated by incumbent local exchange carriers' (ILECs) as well as long-distance phone service companies. DSL in general and ADSL in particular provide an economic means for residential and small business users to access the Internet as well as public switched telephone networks (PSTNs). As shown in Fig. 1, a single telephone line can connect multiple computers and telephone lines all the way to the Internet and PSTNs. While data traffic is directed to IP networks, Voice traffic can go to either IP networks (i.e., Voice over IP or VoIP) or PSTNs.

However, unlike a backbone network based on fiber optics with a bit error rate (BER) on the order of 10^{-11} to 10^{-13}, an ADSL local loop 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 Cross Talk (FEXT), Near End Cross Talk (NEXT), and impulse noise. Therefore, a challenging issue is how to provide quality of service (QoS) guarantees for voice and mission critical data services in an ADSL segment. Both the physical layer and higher layer mechanisms need to be combined to achieve desired QoS guarantees for these services.

Much research effort has been made in the analysis of the impact of channel noises on the ADSL performance, e.g. [8], [12], and in the design of loading and dynamic loading algorithms to optimize the channel performance [5], [6], [9], [10]. Three bit loading schemes have been proposed, i.e. Rate Adaptive (RA) loading algorithm [13], Margin Adaptive (MA) loading algorithm [13], and Disjoint Bandwidth Search (DBS) loading algorithm [9]. The RA algorithm maximizes the bit rate under the energy constraint. The MA algorithm minimizes the energy to achieve a minimum bit rate guarantee. The Disjoint Bandwidth Search (DBS) algorithm [9] maximizes the overall bit rate under both energy and certain bit rate constraint. These bit loading algorithms are intended to be used to provide Variable Bit Rate (VBR), Constant Bit Rate (CBR), and both VBR and CBR services, respectively.

Most of the data applications are built on top of 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, e.g. [14], [1], [11]. By assuming that the channel bit rates in both directions are given, these papers investigate the effects of buffering [14], asymmetry [3], and random loss or BER [15], [4] on TCP throughput performance.

However, none of the studies mentioned above considered both the physical layer (ADSL) and the upper layer performance simultaneously. Since the objective of performance optimization at any given layer is to deliver the best service to its upper layer, the TCP/UDP performance should be the ultimate performance measure for applications using it as their underlying protocol.

In order to solve these problems, in our previous work[20], we performed a TCP performance analysis for the DMT based ADSL network and developed a dynamic bit loading scheme is developed. The results presented in [20] show that, with given total energy input, there exists a threshold for the bit loading rate, or equivalently, BER, above which the TCP throughput per-

[1] The work was completed when authors were with The Pennsylvania State University

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formance drops drastically. The threshold value is a function of both channel characteristics and RTT value. However, setting a large noise margin itself reduces bit loading rate and thus TCP throughput performance.

In general, Voice over IP over ADSL uses UDP (i.e., User Datagram Protocol) as its transport layer (ADSL Forum recommended protocol stack is RTP/UDP/IP/AAL5/ATM/ADSL [19]). Due to the connectionless nature of UDP, the lower layer behaviors generally do not cause UDP protocol specific problems. So the lower layer performance optimization can directly translate into optimized UDP performance. On the other hand, due to the TCP congestion control mechanism and its connection-oriented nature, TCP throughput performance is highly sensitive to bit errors generated at lower layers. Since most of the data applications use TCP as their underlying transport protocol, how to provide assured throughput performance for TCP-based applications over ADSL lines is a critical issue.

A. Motivation

In the current ADSL network, the resource, i.e. the energy distribution over the subchannels, is provisioned based on the raw bit rate at the physical layer. It means that the energy will be distributed in such a way that the bit rate will be maximized under certain noise margin. This resource provisioning approach does not attempt to optimize the transport layer TCP/UDP throughput performance. However, as an access technology for the network, the TCP/UDP throughput performance is the most important factor for the end users. With the threshold effect, it is obvious that the existing service provisioning at the physical level only is inadequate. "..., Actual speeds will vary and depend on numerous factors, including the condition of your phone line and the wiring inside your home or office, the distance between your location and Central Office, network and Internet congestion and other factors. Due to the sophisticated nature of DSL, we cannot guarantee uninterrupted or error-free service, or the speed of your service", this is a common expression from the DSL service provider's advertisement. As a result, we argue that it will be more important to provision the network based on the UDP/TCP performance since the TCP/UDP throughput and delay performance are the end-user's primarily concern.

In this paper, we propose a resource provisioning scheme which provisions the network resource based on the TCP/UDP performance. This scheme enables three classes of service (CoSs). The first CoS provides CBR type of service for voice. The second CoS provides throughput assurance for TCP data service. The third CoS is the best-effort data service. These three CoSs can be directly mapped into the Expedited Forwarding, Assured Forwarding, and the Default Forwarding Per hop behaviors of DiffServ model, respectively. Hence, combined with DiffServ in backbone networks, this architecture enables seamless, end-to-end QoS guarantees for voice and mission critical data services.

The rest of the paper is organized as follows. Section 2 presents a background introduction to DMT based ADSL systems. Section 3 describes a bit loading architecture to enable service differentiation for the three CoSs. Section 4 provides a case study based on the proposed architecture. Finally, Section 5 concludes the paper and proposes future research directions.

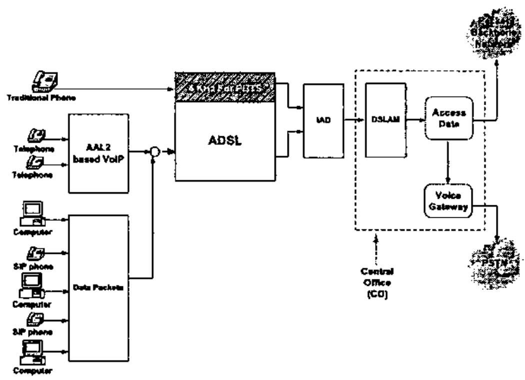

Fig. 1. VoDSL Architecture

II. Background Introduction

This section gives the fundamental background on the ADSL technology, DMT modulation, and TCP/UDP protocols.

A. Asymmetric Digital Subscriber Line (ADSL)

ADSL is a standardized transmission technology facilitating simultaneous use of normal telephone services and data/voice 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 sub-bands. The baseband of 4 KHz is used for analog voice telephony. The band between 25 KHz and 138 KHz is for upstream data transmission. The band between 200 KHz to 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 ITU standard, ADSL should run at a minimum of 6.144 $Mbps$ downstream and 640 $kbps$ upstream over the existing copper telephone lines [2], [7]. Since the Discrete MultiTone (DMT) modulation technique has been chosen by ITU as the standard modulation scheme for ADSL, in this paper, we focus on the DMT based ADSL network.

B. 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. Suchannels with larger SNR carry more data and those with smaller SNR carry less data. DMT is able to allocate data so that the overall bit rate is maximized. A detailed explanation of DMT can be found in [6].

Each subchannel's SNR is given by

$$SNR_i = \frac{E_i|H_i|^2}{\sigma_i^2}, \tag{1}$$

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

$$b_i = \frac{1}{2}\log_2(1 + \frac{SNR_i}{\Gamma}), \quad (2)$$

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,

$$\bar{b} = \sum_{i=1}^{N}\log_2(1 + \frac{SNR_i}{\Gamma}), \quad (3)$$

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,

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

where N_e is the number of nearest neighbors of an input signal constellation for the i^{th} subchannel, and

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

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,

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

Bit loading algorithms calculate the bit or energy distribution for subchannels. As mentioned in the Introduction Section, there are three proposed bit loading algorithms: RA, MA, and DBS. The following three subsections briefly describe the three bit loading algorithms, respectively.

C. Rate-Adaptive (RA) Bit Loading Algorithm

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

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

$$\textit{subject to:} \sum_{i=1}^{N} E_i \leq E_{total},$$

where $g_i = |H_i|^2/\sigma_i^2$, which represents the SNR with a unit energy of a subchannel. E_{total} is the total input energy.

It is proven in [6] 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.

D. Margin-Adaptive (MA) Bit Loading Algorithm

The MA bit loading algorithm minimizes the total energy subject to a fixed bit rate constraint:

$$\min_{E_i} \sum_{i}^{N} E_i$$

$$\textit{subject to:} \sum_{i=1}^{N} log_2(1 + \frac{E_i g_i}{\Gamma}) \geq B,$$

$$E_i \geq 0$$

Many applications require a fixed data rate. Under this constraint, one wants to maximize the performance margin given a fixed data rate of B bits/symbol. It is proven in [6] that the MA design objective is achieved when

$$\epsilon_n = K_{ma} - \frac{\Gamma}{g_n}.$$

The traditional formulation of this optimization problem is not convex in the unknowns E_i and Γ, and requires an iterative method to find the solutions [5].

E. Disjoint Bandwidth Search (DBS) Bit Loading Algorithm

The problem formulation of the DBS bit loading algorithm is as follows.[9], [10]:

$$\max_{S,E_i} \sum_{i\in\overline{S}} log_2(1 + \frac{E_i g_i}{\Gamma_2}),$$

$$\textit{subject to:} \sum_{i\in S} log_2(1 + \frac{E_i g_i}{\Gamma_1}) \geq B,$$

$$\sum_{i} E_i \leq E_{tot}.$$

where S, a subset of the N tones, gives the tone assignment for CBR service and $\overline{S}$, the complement of S, gives the tone assignment for VBR service.

In [9], [10], the authors showed that an exhaustive search for the optimal tone assignment S^* is NP-hard. In their paper, a suboptimal algorithm, i.e. Disjoint Bandwidth (DBW), has been developed. The DBW algorithm meets the requirement of CBR service and provide maximum bit rate for the VBR service at the same time. This algorithm is particularly useful for allocating resources between Voice and Data services, by assigning CBR channels to voice calls and VBR channels to data traffic.

F. 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.

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 is used for ISDN Basic Access, and the HDSL for DS1 rate access. In general, the PSD of the NEXT can be expressed as:

$$PSD_{NEXT} = PSD_{Disturber} \times x_n \times f^{3/2}, \qquad (6)$$

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

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

$$PSD_{HDSL} = K_{HDSL} \times \frac{2}{f_0} \times \frac{[sin(\frac{\pi f}{f_0})]^2}{(\frac{\pi f}{f_0})^2} \times \frac{1}{1+(\frac{f}{f_{3dB}})^8}, \qquad (7)$$

where $f_{3dB} = 196\ kHz, f_0 = 392\ kHz,\ K_{HDSL} = \frac{5}{9} \times \frac{V_p^2}{R}, V_p = 2.70\ V, andR = 135\ Ohms.$

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 can not be ignored. At high frequency range ADSL self FEXT noise power can exceed that of HDSL NEXT and white background noise combined. In general, thhe PSD of the FEXT can be expressed as

$$PSD_{FEXT} = PSD_{Disturber} \times |H_{channel}(f)|^2 \times k \times l \times f^2, \qquad (8)$$

where $H_{channel}(f)$ is the channel transfer function, k is defined as $8\times10^{-20}\times(n/49)^{0.6}$ which is called is the coupling constant, and l is the coupling path length.

G. TCP Congestion Control Protocol

In this paper, we consider the TCP-Reno version of TCP congestion control protocols. TCP-Reno provides fast retransmission and reduces the number of the slow start occurrences.

TCP Reno retransmits the packet after a number of duplicate acknowledgments 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 acknowledgment until the 'next expected' number in the acknowledgment advances.

H. User Datagram Protocol

Different from the TCP protocol, UDP protocol is an unreliable, connectionless protocol. It is used in applications when prompt delivery is more important than accurate delivery. Most of such applications are real-time multimedia applications transmitting video or speech. Such applications can tolerate packet loss but can not tolerate significant delay and delay jitter.

III. A Resource Provisioning Architecture for ADSL

The simulation results presented in [20] show that, with given total energy input, there exists a threshold for the bit loading rate, or equivalently, BER, above which the TCP throughput performance drops drastically. The threshold value is a function of both channel characteristics and RTT value. As a first step towards a comprehensive solution, in this section, we propose a *resource provisioning* architecture which enables three CoSs, i.e., CBR or expedited forwarding type of service for voice, assured forwarding type of service for TCP applications (we call it TCP assured service), and the best-effort service for the rest of data traffic. This architecture is in line with the DiffServ architecture proposed for IP networks and thus offers a solution to provide seamless, end-to-end QoS guarantees for voice and mission critical data services.

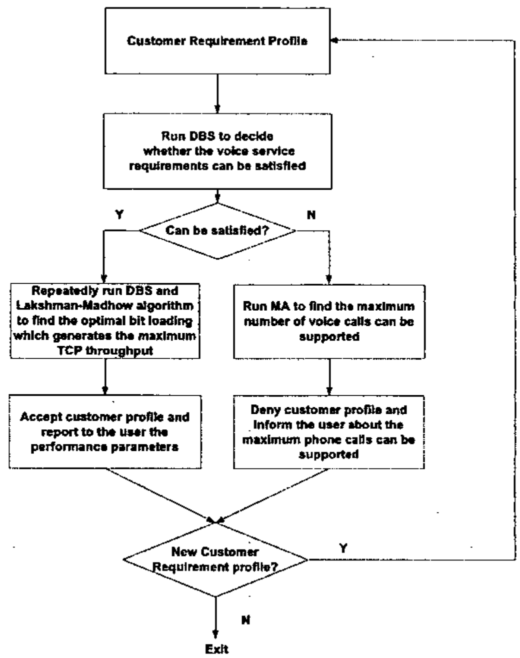

Fig. 2. Summary of resource provisioning process

In our architecture, to provide efficient use of bandwidth, a small percentage of the total energy and subchannels are allocated for the best-effort service under normal operation conditions. These resources may be re-allocated to the other two services under abnormal conditions such as temporary high noise situations. The bit loading algorithm for the best-effort service is based on a joint optimization of the RA bit loading and TCP throughput performance (assuming that the majority of the best-effort traffic is composed of TCP applications).

International Telecommunication Union (ITU) has standardized several codec algorithms for voice, including G.711 with PCM (Pulse Code Modulation) at 64Kbps, G.726 with ADPCM (Adaptive Differentiated PCM) at 40, 32, 24 and 16Kbps [18], [16] with or without silence suppression. In this paper, we assume that VoDSL phones use encoding schemes without silence suppression. Based on this assumption, the voice service can be

Parameters: number of phones N, bit rate per voice call R, active call percentage at the peak hour k,
bit rate reserved for control messages $C[i]$, round-trip time $rtt[i]$,
optimal bit loading rate for the assured service traffic $r_{opt}[i]$, optimal SNR gap for the assured service $\Gamma_{opt}[i]$,
optimal BER P_{opt}[i] corresponding to $\Gamma_{opt}[i]$,
overall maximum TCP throughput for the assured service traffic $Th_{opt}[i]$,
required bit rate for voice and control traffic B_{voice}, total input energy $E_{total}[i]$, total subchannels $S_{total}[i]$,
total energy for non-best-effort traffic $E'_{total}[i]$, total subchannels for non-best-effort traffic $S'_{total}[i]$,
total energy reserved for the best-effort service $E''_{total}[i]$, total subchannels for the best-effort traffic $S''_{total}[i]$
percentage of total subchannels for the best-effort traffic $per_{N_b}[i]$,
percentage of total energy for the best-effort traffic $per_{E_b}[i]$

$E'_{total}[i] = (1 - per_{E_b}[i]) \times E_{total}[\mathrm{i}]$ and $E''_{total}[i] = per_{E_b}[i] \times E_{total}[i]$
$S'_{total}[i] = (1 - per_{N_b}[i]) \times S_{total}[\mathrm{i}]$
$S''_{total}[i] = S_{total}[i] - S'_{total}[i]$.
i=1 (upstream), 2 (downstream)

Input: $N, R, k, C, E'_{total}[i], S'_{total}[i], E''_{total}[i], S''_{total}[i], \Gamma_1, \mathrm{BER}_{voice}$
Output: **accept** or **reject** user requirements
When **accept**: for different $rtt[i]$, output $r_{opt}[\mathrm{i}]$, $\Gamma_{opt}[\mathrm{i}]$, $P_{opt}[\mathrm{i}]$, and $Th_{opt}[i]$ for i=1,2
When **reject**: report the maximum number of phones can be supported and may suggest new requirement parameters

Algorithm

1. **for** i=1 to 2
2. choose the subchannels
3. sort the subchannels according to the subchannel gains from highest to lowest
4. reserve per_{N_b}[i] precent of the total subchannels and per_{E_b}[i] percent of the total energy for the best effort service.
   default values : $per_{N_b}[i] = 10\%$, $per_{E_b}[\mathrm{i}] = 10\%$
5. for voice service: $B = B_{voice} = k \times N \times R + C[i]$
   choose the value of Γ_1 for the voice service, default Γ_1=7.7 dB correponding to BER=10^{-7};

   for TCP assured service: set a rtt scan range ($rtt \in [t_1, t_2]$). default values: $t_1 = 40\ ms, t_2 = 1\ sec$.
   choose a scan range for Γ_2, i.e. $\Gamma_2 \in [g_1, g_2]$ with g_{step} as step size, default values are: $g_1 = 1\ dB, g_2 = 15\ dB$, and $g_{step} = 0.5\ dB$

   repeat step 6 to step 10 twice with rtt = t_1 and rtt = t_2
6. set $r_{opt}[i] = 0$, $P_{opt}[i] = 0$, $Th_{opt}[i] = 0$.
7. **for** g = g_1 to g_2 **step** g_{step}
   run DBS algorithm with E'_{total}[i] , S'_{total}[i] , Γ_2= g for TCP assured service and Γ_1= 7.7dB for voice service as input to find the overall maximized bit rate. The rate achieved for the TCP assured service is denoted as $r[i]$;
   if (DBS doesn't find a solution when $g = g_1$) **then**
   run MA bit loading algorithm to find $\mathrm{N_{max}}$ [i] go to step 1
   end if
   calculate current BER, i.e. P_e , according to g
   run Lakshman-Madhow algorithm with $r[i]$, P_e[i] , and rtt as input to find the TCP throughput $Th[i]$
   if ($Th[i] > Th_{opt}$[i]) **then**
   $r_{opt}[i] = r[i]$
   $P_{opt}[i] = P_e[i]$
   $Th_{opt}[i] = Th[i]$
   end if
   end for
8. **run** RA bit loading algorithm with S''_{total}[i] and E''_{total}[i] as input to find the bit rate for the best-effort service
9. **end for**
10. **if** (both upstream and downstream processes find solutions) **then**
    accept the user requirement for voice and report to user $r_{opt}[i]$, P_{opt}[i], Γ_{opt}[i] and Th_{opt}[i] obtained for $rtt = t_1$ and t_2
    else
    reject the customer requirements. Report min($\mathrm{N_{max}}$[1], $\mathrm{N_{max}}$[2]) and suggest possible new requirement parameters
    end if

Fig. 3. Resource provisioning process

Initial Customer Profile and Output

Input :

Voice source : *ITU G.711 PCM 64kbps*
Channel: CSA Loop #4
Customers required BER for Voice: 10^{-6}
Number of phones required by the customer : 70
Control Bit Rate: $R = 64Kbps$
Input Power : $20mW$
Noise Power : $140dBm/Hz$
k = 40%

Output :
Maximum number of phones supported are : 64

Adjusted Customer Profile and Output

Input :

Voice source: *ITU G.711 PCM 64kbps*
Channel: *CSA Loop #4*
Customer required BER for Voice: 10^{-6}
Number of phones required by the customers: 15
Control Bit Rate: $R = 64Kbps$
Input Power: $20mW$
Noise Power: $140dBm/Hz$
k=40%

Output :

Upstream

Voice Service :
Number of phones supported are: 15
BER : 10^{-6}
Γ=8.8 dB

TCP assured service :
$rtt = 40ms$
Sending Speed: 1.26 $Mbps$
Maximum TCP throughput: 1.26 $Mbps$
BER= 4.49×10^{-7}
Γ=9 dB
rtt = 1 sec
Sending Speed: 0.98$Mbps$
Maximum TCP throughput: 0.98 $Mbps$
BER= 3.78×10^{-9}
Γ=12 dB

Downstream

Voice Service :
number of phones supported are: 15
BER : 10^{-6}
Γ=8.8 dB

TCP assured service :
$rtt = 40ms$
Sending Speed: 6.49$Mbps$
Maximum TCP throughput: 6.49$Mbps$
BER= 4.49×10^{-7}
Γ=9 dB
rtt = 1 sec
Sending Speed: 5.26$Mbps$
Maximum TCP throughput: 5.26 Mbps
BER= 3.78×10^{-9}
Γ=12 dB

Fig. 4. A case study

modelled as a CBR or expedited forwarding type of service, i.e., voice streams in both downstream and upstream directions run at fixed bit rates.

The resource provisioning for the voice and TCP assured service are optimized by a joint optimization of the DBS bit loading and TCP throughput performance. This is achieved by searching for a BER (or Γ_2 in DBS) which optimizes DBS bit loading and maximizes the TCP throughput simultaneously at given subchannel SNRs and RTT. TCP throughput can be found based the existing analytical results, e.g., Lakshman-Madhow algorithm [15], or simulation. The study in [20] shows that the Lakshman-Madhow algorithm is accurate enough to serve for this purpose. When only voice service is required or the required voice capacity exceeds the total capacity, the MA bit loading algorithm is used to estimate the maximum number of supportable active voice calls.

The architecture facilitates the enforcement of the service level agreement (SLA) between an ADSL service provider (SP) and its customers. The resource provisioning takes place at the time when a customer subscribes to the ADSL services. The customer provides the information about the number of telephones to be connected to the ADSL link and the voice qualities such as call blocking probability, an average BER and a noise safe margin. The customer may also specify a minimum data rate and a noise safe margin for the TCP assured service. Based on this customer requirement profile, the SP:

1. decides whether the customer requirements can be met;
2. suggest new SLA parameters if the customer requirements cannot be met;
3. provides bit loading algorithms to enforce the SLA.

The SLA can be renegotiated and the resources re-provisioned due to customer requirement changes or considerable physical channel condition changes. One can expect that the changes will be infrequent, e.g., on the order of several weeks' or several months' interval. Hence, in our algorithm design which involves only off-line computation, the computation complexity should not be a major concern. The QoS is guaranteed under short-term noise condition variations by providing sufficient noise safe margins for value-added services.

As a rule of thumb, there will be 20 to 40 percent of voice calls simultaneous active during peak hours. Hence we use a configurable parameter in this range to estimate the voice call bandwidth requirement given the total number of telephones to be connected to the ADSL line. For simplicity, we assume that all the telephones connected to the same ADSL line use the same codec. In other words, we assume that voice sources are homogenous.

The resource provisioning procedure is presented in Fig. 3. A flow diagram (see Fig. 2) is also presented, which highlights the major steps of the resource provisioning procedure. The procedure is intentionally made symmetric with respect to upstream and downstream channels so that it applies not only to ADSL based system but also to other DMT based xDSL systems in general.

IV. A CASE STUDY

In this section, we present an example to show how the resource provisioning scheme works. Throughout the simulation studies, we assume that the input power is fixed at 20 mW, the background AWGN (additive white Gaussian noise) is fixed at $-140\ dBm/Hz$, and a trellis code is used which achieves a 5 dB coding gain. Furthermore, we assume there are 24 ADSL disturbers with 20 mW input power. The crosstalk noise PSD is calculated based on equation(6) and equation(8).

As shown in Fig. 4, initially, the user requires that 70 telephones be connected with BER equal 10^{-6}. The customer does not specify the requirements for the assured data service and would like to wait and see what the SP can offer. Based on the profile provided by the customer, the total input power, and the channel characteristics, the SP then runs the algorithm as shown in Fig. 3. At step 7, the DBS algorithm fails to find a feasible solution which meets the customer profile in the upstream direction. After calculating the maximum number of simultaneously supportable calls using the MA bit loading algorithm (which is 26) and a second run of the algorithm for the downstream which found a feasible solution, the maximum number of voice calls $N^1_{max} = 64$ ($N^1_{max}k = 64 \times 0.4 = 25.6 < 26$) is reported to the customer. Note that for ADSL, downstream bandwidth is much larger than the upstream bandwidth. Hence, in this case study, downstream can handle 70 phones required by the customer.

Realizing that the requested number of phones is too large to be acceptable, the customer decides to reduce the number of phones to 15. By comparing this number with $N^1_{max} = 64$, the customer knows that the SP will for sure be able to provide the required voice service and may also be able to offer sufficient bandwidth for his data service needs. To get an accurate estimate of the achievable throughput for his TCP applications, the customer sends an adjusted profile to the SP as shown in Fig. 4. The SP then re-runs the algorithm based on this new profile. The output of the algorithm shown in Fig. 4 is then reported to the customer. After the customer acknowledges the acceptance of these performance values, a SLA is achieved between the two parties. The SP will then record this SLA in its database. To comply with the SLA, the SP will then configure its ADSL control devices to operate at the bit loading rates obtained by the algorithm.

To validate whether the agreed SLA can be complied with, a simulation study is performed. In the simulation setting, the 6 voice channels are active and an ftp connection with RTT equal 50 ms is always on and is sending traffic at its full rate. The results show that 1.77 $Mbps$ the ftp throughput can be achieved which is above 1.65 $Mbps$ at $t1 = 40\ ms$ and 1.49 $Mbps$ at $t2 = 1\ sec$, complying with the SLA.

V. CONCLUSIONS AND FUTURE WORK

In this paper, based on the results presented in [20], we propose a resource provisioning architecture, which provisions the network resource according to the TCP/UDP performance. An algorithm is proposed to enable these services, which jointly optimizes bit loading rate at the physical layer and TCP throughput performance at the transport layer. Finally, a case study is given which demonstrates how the architecture can be used to enforce SLA between a customer and a SP.

In our architecture, the QoS is guaranteed under short-term noise condition changes by providing sufficient noise safe margins for value-added services. As mentioned in the previous section, the resources for the best-effort service can be re-allocated to the other two services under abnormal conditions such as temporary high noise situations. In other words, by dynamically turning off and on of some of the subchannels for the best-effort service and redistributing its energy to other subchannels used by other services, one can achieve another level of protection for value-added services. Hence, as part of our future work, we plan to study how to combine DMT's fast retraining capability with higher layer protocol mechanisms to allow dynamic resource re-allocation to achieve high service availability.

In our architecture, we didn't address the issue as to how to allow dynamic resource sharing. For instance, the best-effort and the assured service traffic should be able to take the subchannel resources allocated to voice service when it is available. This requires the design of fast algorithms, which can quickly respond to the load changes. We shall explore this issue as part of our future work.

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