Multimedia Services over Digital Subscriber Lines

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Multimedia communications is one of the most challenging research areas emerging recently, due to the advance of multimedia compression and coding technologies. With the recent successes in the establishment of some multimedia communications standards as well as the rapid proliferation of the uses and demands, we can predict the creation of a vast number of applications destined to offer users new, efficient, and sophisticated services in the near future. The word "multimedia," in general, is considered to mean simultaneous consideration of a mixture of signal types, such as speech/audio, image/video, graphics/animation, and text. In particular, video conferencing, multimedia mail, video-on-demand, HDTV broadcast, and many other multimedia applications through the Internet—either by wireline such as asymmetric digital subscriber line (ADSL), ISDN, or by wireless networks—post new problems with a distinctive nature, making multimedia communications one of the most interesting and challenging research fields.

In this tutorial, we study the problem of reliable and yet efficient multimedia communications over ADSL through joint consideration of compression/coding and channel transmission techniques. ADSL will be delivering multimedia services to millions of users. The transmission of digital multimedia data requires the existing systems to be augmented with functions that can handle not only ordinary (nonmultimedia) data. In addition, the high volume of multimedia data can be handled efficiently only if all available system resources are carefully optimized. Here we consider the special characteristics of ADSL channels to formulate optimization criteria. We present a system where the encoder consists of a layer coder that divides and compresses the source data into coded layers of multimedia data with different performance and quality-of-service requirements. The encoded bit streams are then transmitted over a noisy channel, where channel noise may distort the data. The decoder undoes all the coding and compression applied in the encoder to obtain as close as possible the original data. It is conceivable that for such a system the loss of information occurs not only during the source compression but also in the channel transmission. The source coding scheme and bit stream arrangement

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have a direct impact on the channel distortion. Therefore, the joint consideration of source/channel coding design will be ideal to reduce the overall information loss.

Introduction

Layer Coding Schemes and Recent Multimedia Standards

Source coding considers mainly the compression of data, in either a lossless or lossy way. Source coding for multimedia data benefits from recent dramatic developments in compression algorithms and their implementations. Compression is necessary and important to reduce the volume of multimedia data for storage and to limit the bit rate to provide efficient transmission over normally bandwidth-limited networks. Ordinary digital data cannot afford to lose any information, as such the compression schemes used are mainly lossless. The multimedia data such as image, video, speech, and audio are subject to human perception, which translates to the loss of some fidelity as long as not perceivable. Therefore, such compression may discard some information to achieve more compactness, corresponding to the well-known lossy schemes.

In the past, multimedia source coding schemes were developed mostly with the assumption of error-free channels. Therefore, the objective of compression is purely on reducing the information rate as much as possible. Only recently, researchers started to realize the importance of joint considerations of the distortion induced in the channel into the source coding design. As such, layered and scalable coding have now been widely recognized and used since they can provide more error resilience for noisy channel transmissions.

In layered or scalable coding [1], [2], source data is decomposed into hierarchical relevant layers, with each resulting in a distinct data stream. Different layers may have distinctly different tolerances to channel errors and delay and thus should be handled differently by the transmitter or network. Layers containing high spatial-frequency components are generally more tolerant to channel error effects than layers containing low spatial-frequency components and thus may receive less protection for channel induced error.

Subband-based coding schemes have been the major class of approach for accomplishing layered coding. Subband coding uses filter banks to divide the source data into subbands corresponding to different frequency ranges [3], [4]. After transformation, the most significant parts of the information, e.g., the coefficients containing most of energy, are packed and easily identified. The encoder is then designed to present those significant (important) coefficients more accurately. Certain coefficients may also be neglected to achieve more compression. It involves a quantizer design and a rate allocation problem, which has been extensively studied in the literature [5]-[8].

By exploiting space-frequency property, significant improvements have been achieved in the new generation of wavelet-based encoders including wavelet packets, embedded zerotree wavelet, set partitioning in hierarchical trees.

A variant of subband coding uses wavelet transform and filter banks to present the data in multiresolution contents. Similar to subband coding, the image and video is decomposed into a set of subbands of different resolutions. The low spatial and temporal subbands carried more information and thus are more important compared to the high spatial or temporal subbands. Extensive research has shown that the images obtained with wavelet-based methods yield very good visual quality [9]-[15]. Recently, by exploiting space-frequency property, significant improvements have been achieved in the new generation of wavelet-based encoders including wavelet packets [17], [18], embedded zerotree wavelet (EZW), and set partitioning in hierarchical trees (SPIHT) [19], [20].

Joint Photographic Experts Group (JPEG) [21], Motion Picture Experts Group (MPEG-1, MPEG-2) [22], [23] and H.26x [25], [26] are all classified as block-based coding techniques, which divide the image or video frames into small blocks and apply a combination of discrete cosine transform (DCT), quantization, run-length, and variable-length coding (VLC) techniques as well as interframe motion prediction and compensation to remove both spatial and temporal redundancy. VLC helps to increase the compression efficiency. However, if channel error occurs and the decoder wrongly decodes a codeword of length different from the transmitted one, the decoder may lose the synchronization and the bit stream followed cannot be decoded correctly until the synchronization is regained. A way to overcome such a problem is the use of error resilience [16].

H.263 [26] is a low bit-rate video coder which is a hybrid of interpicture prediction, transform coding, and motion compensation. Half-pixel precision is employed in motion compensation to further reduce the temporal redundancy. H.263+ carries an option for layered video coding, or scalability, i.e., SNR and spatial/temporal scalability. Spatial scalability generates multiresolution bit streams. Increasing the picture resolution by a factor of two in the vertical, horizontal, and/or both directions can enhance the picture quality. Temporal scalability enhances the picture quality by increasing picture display rate. SNR scalability generates multirate bit streams where the enhancement layer carries the finely quantized difference of the base layer and original picture [28]. Recently, H.263++ is proposing a data partition option which groups all the macroblock headers, motion vectors, and DCT coefficients separately with a synchronization word in between. The headers and motion vectors are reversible VLC (RVLC) coded so that more information can be extracted from the corrupted bit stream [29].

MPEG-4 [24] is an object-based approach designed for both low bit-rate video communications and mixed multimedia data such as video, image, graphics, text, audio, and speech. By multiplexing and synchronizing the streams associated with different data defined as audiovisual objects (AVOs), they can be transported over networks with appropriate quality of service (QoS) matching the nature of the AVOs. Such a layered structure also permits error resilience to enable robust transmission of compressed data over noisy channels. For low bit-rate video, a data partition option is proposed for MPEG-4 where the important information such as header, motion vector, and DC component in DCT coefficient is separated from texture (other DCT coefficients) and grouped separately with synchronization words placed regularly in between. This allows for the isolation of errors and helps the decoder to regain synchronization. The groups are deemed as layers and treated differently.

The popularity of the Internet is fueling the emergence of multimedia applications. A mixture of data, speech, and video can provide entertainment, online services, interactive shopping, video conferencing, etc. Those services are associated with different data rates and different QoS requirements. A typical example is shown in Table 1, which involves video conferencing, speech data, and e-mail. Those services can be viewed as data sources of different layers with different bit rates and bit error rate (BER) requirements. As specified by MPEG-4, integrated services in terms of AVOs can also be achieved.

Joint Considerations of Source and Channel Schemes

Most of the channel coding schemes in the past considered primarily (nonmultimedia) data transmission. Such data communication has a strict low BER requirement, but can tolerate latency well. In addition, every information bit is of the same importance. Therefore, the performance measure is mainly on BER. Unlike ordinary digital data, multimedia data such as image, video, and audio have a high tolerance to error and distortion. A low percentage of bit errors can usually be tolerated. Some portion of the content can even sustain high BER without causing too many visible fidelity distortions. Therefore, the BER requirement of multimedia communications is much relaxed compared to data communications, typically ranging from 10^{-5} to 10^{-2}. The performance metric is usually audio/visual quality and measured as the mean squared error (MSE) between the original and received data. The performance evaluations about multimedia data can be found in [30] and [31]. On the other hand, multimedia data is usually delay sensitive. In data transmission, automatic retransmission (ARQ) is often employed to retransmit corrupted or lost data until successfully received. Nevertheless, in real-time multimedia communications, the number of ARQ operations is quite limited since it yields additional delay. Shannon's separation principle states that if the minimum achievable source coding rate is below a channel's capacity, then the source can be reliably transmitted through the channel [32]. Such theory allows separate design of source coding and channel coding, without any loss of the performance for the overall system. It also relies, however, on the availability of an appropriate long source signal as well as high computational cost and associated delays. In practice, due to limited resources, such conditions can not be met.

Table 1. Data Rate and BER for Various Services.

Service	Data Rate	BER
Video Conferencing	200K b/s	10^{-6}
Speech	10K b/s	10^{-3}
E-mail	64K b/s	10^{-5}

It has been proved that, in practice, combining source and channel coding design has led to a substantial gain in performance. Due to layered coding, multimedia data consist of different layers with different error sensitivities. One way to efficiently combat channel errors is to employ unequal error protection (UEP) to layers of different importance. For example, it has been shown that by separating the source material into a base layer and an enhancement layer and applying priority queuing or different loss probabilities, the results are superior to the single layer approach obtained from statistically multiplexed networks [33]. In addition, it is generally argued upon that, in practice, most multimedia applications involve stationary users or users moving at relatively slow speeds, which results in a fixed or slowly changing channel. Powerful channel equalization and estimation techniques allow transmitters to obtain the current channel condition correctly. Therefore, joint source and channel coding is applied to adapt the source coding, channel coding, modulation, and power control according to the current channel condition and the coded layer to be transmitted.

Typical joint source and channel coding designs include the following.

▲ *Source Oriented Channel Coding Design*

A popular form of joint source and channel design is based on choosing appropriate channel coding techniques for different source components. UEP is achieved by allocating different source coding rate and channel coding rate to different layers. For example, the joint design of rate compatible-punctured-convolutional

(RCPC) codes and bit-rate assignment in source coders [34]-[40] is a typical example.

▲ *Source Oriented Power Control*

For power constrained channels, such as wireless channels, power control schemes can be employed. Layers assigned with higher transmitted power experience lower channel BER than those with lower transmitted power. The objective of the power optimization is to efficiently distribute the transmitted power to the layers to minimize the overall channel effect [42].

▲ *Source Oriented Modulation Design*

Another critical part of joint source and channel design is the mapping between quantized source code word and their representative channel modulation symbols [43]. The mapping from source codeword space to two-dimensional signal space is designed in such a manner that the close code vectors in the modulation space are assigned to the close points in the input space. Therefore, most channel errors lead to the output code-vectors within the nearest neighbors of the input vectors [43]-[45], [47]. The design of quantization code book can take into account both source distortion and channel noise affect to improve robustness [48]-[51]. A detailed survey of joint source and channel coding can be found in [53].

The Goal of this Tutorial

ADSL services are available in many places in the world. Soon ADSL will deliver multimedia services to millions of users. Such a demand motivates us to understand how to best deliver multimedia data over ADSL channels. Most of the recent research on ADSL focuses on reliable nonmultimedia data delivering at very low BER such as 10^{-7} with equal error protection. We will re-examine why it is not such a good idea to do so and then discuss what we think will be a better approach.

Our discussion centers around the development of a multimedia communications framework for ADSL channels. We emphasize the channel resource allocation problems needed to achieve required performance and quality using minimum amount of channel resources. Providing UEP, rather than equal error protection, in a resource-efficient way is the key aspect of most algorithms in discussion. The basic principle is to match the characteristics of telephone line channels to the error performance requirements of source layers.

ADSL Fundamentals

ADSL [54], which evolved during the early 1990s, uses the existing twisted-pair copper loops to offer an effective alternative to the next generation of broadband access networks and to support high speed communications at very affordable prices. It is specifically designed to support asymmetric data traffics to exploit the one-way nature of most multimedia communications where large amounts of information flow toward the subscribers (downstream) and only a small amount of interactive

One way to efficiently combat channel errors is to employ unequal error protection to layers of different importance.

control information is returned (upstream). ADSL transmits more than 6 Mb/s (optionally up to 9 Mb/s) downstream to a subscriber, and as much as 640 kb/s (optionally up to 1 Mb/s) upstream.

The ADSL system can place the upstream and downstream frequency band separate from each other to avoid the self-crosstalk or overlap them to reduce transmission bandwidth. It operates at the frequencies above those of standard telephony, allowing simultaneous data and voice services on the same pair. Typical applications of ADSL are fast access to Internet facilities (e.g., web services) and fast delivery of multimedia services [e.g., video on demand (VoD), digital TV, streaming video/audio].

Channel Characteristics

We first examine the channel characteristics of the twisted-pair copper loop, namely the ADSL channel. Major impairments in the twisted pair phone lines are the attenuation and crosstalk. The attenuation comes from the loop transfer function, e.g., insertion loss. It forms a large variation of signal levels over frequency and makes digital transmission very difficult. Crosstalks are generally caused by electromagnetic radiation of the phone lines. Typical crosstalks include the so-called near-end crosstalk (NEXT) and far-end crosstalk (FEXT). NEXT is the transmitted signal that leaks into the receiver via capacitive and induction coupling paths. It increases with frequency. Recently, many efforts have been taken to suppress or cancel NEXT [56]. In contrast, FEXT occurs when signals from the transmitters on other pairs within the same cable leak into the transceiver at the other end. FEXT is proportional to $|H(f)|^2 \cdot d \cdot f^2$, where $H(f)$ is the loop transfer function, d is the line length, and f represents frequency. Therefore, symmetric transmission would yield a significant amount of FEXT interference while the asymmetry of ADSL resolves the limitation of the downstream data rate and increases the delivering distance. In addition to crosstalks, a variety of sources producing short electrical transients could generate the impulse noise that is random in frequency, duration, and amplitude. Together, the impairments tend to increase as a function of frequency and distance except for the impulse noise. The resulting ADSL channels are spectrally shaped. A typical ADSL channel is shown in Fig. 1. Interested readers can find more detailed information about ADSL in [55] and [57]-[60].

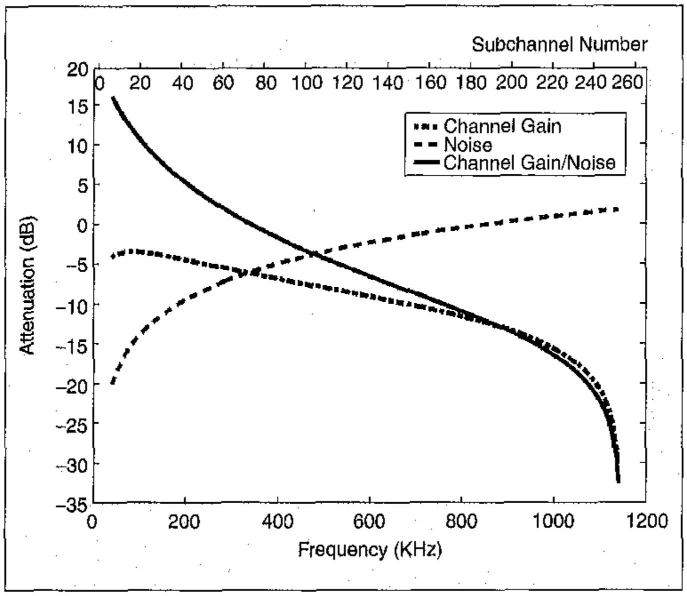
Channel Modulation Scheme

The wide variations in the ADSL channel lead to increased difficulty and complexity for channel equalization, which is necessary for any single-carrier system. Multicarrier modulation (MCM) [61] is currently considered as a standard channel coding scheme for ADSL. The basic function of an MCM system is to partition the channel into a set of independent subchannels, each with a smaller bandwidth. At the transmitter, a block of input bits is modulated as a set of quadrature amplitude modulated (QAM) symbols and passed to an inverse fast Fourier transform (IFFT), which combines the complex symbols into a set of real-valued time domain samples. The receiver applies FFT to convert the received (possibly corrupted) time domain samples into the QAM symbols and performs demodulation. In practice, the so-called cyclic prefix is added to remove the intersymbol interference (ISI) among the subchannels [65]. Other subchannel partitioning techniques are summarized in [55].

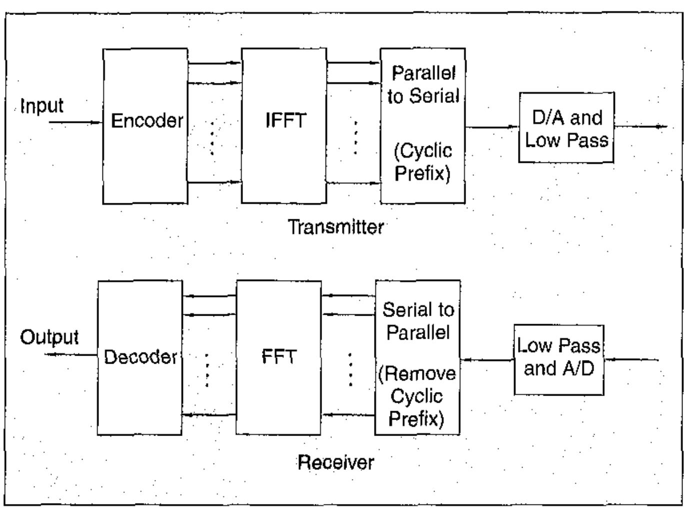
When the number of the subchannels is large, each subchannel is sufficiently narrow and the Gaussian noise is shown to be independent among the subchannels. Therefore, each subchannel can be approximated as an additive white Gaussian noise (AWGN) channel which requires little or no equalization. The subchannel transfer function can be approximated as a set of rectangles using the value at the center frequency in the subchannel. As such, any subchannel can be completely described as

$$T_k = H_k S_k + N_k \tag{1}$$

where T_k and S_k represent the received and transmitted signal at the kth subchannel, and H_k and N_k represent the subchannel gain and noise at the same subchannel, respectively. A remarkable feature of the MCM system is that it allows data rate, transmitted power, as well as channel encoder/decoder at each subchannel to be changed flexibly without affecting the others. This not only achieves the optimal transmission, but also allows the fast adaptation to the time-varying phone line channels.



▲ 1. *Typical spectrally shaped ADSL channels. The channel gain is defined as the square of the amplitude of the loop transfer function, while the noise represents the total effect of the crosstalks. The ADSL band resides in 40 KHz to 1397 KHz, which is further partitioned into 256 subchannels of 4.3125 KHz each.*



▲ 2. *Multicarrier basic model.*

Subchannel Optimization

The optimal use of the ADSL channels is achieved by making the optimum use of all the subchannels. Each subchannel is associated with two transmission parameters: transmitted power and data rate. Data rate is defined as the number of bits per transmission, e.g., four if QAM16 is used and six if QAM64. Assigning equal data rate and transmitted power to all the subchannels clearly does not provide the optimal subchannel usage. From the information theoretic view, water-filling [67] can achieve the ultimate system capacity, where different transmitted power is allocated to the subchannels according to their channel gain and noise variance. Larger amounts of transmitted power are assigned to the subchannels with higher channel gain and lower noise variance. We name those subchannels as the good subchannels. As a result, the good subchannels should transmit at a higher data rate to maximize the overall transmission data rate for a given power limit.

There has been extensive study on the allocation of power and data rate to the subchannels, known as the loading algorithm [61], [68], [69],

[71], [72]. We will review the loading algorithms in the following sections.

Multimedia over ADSL

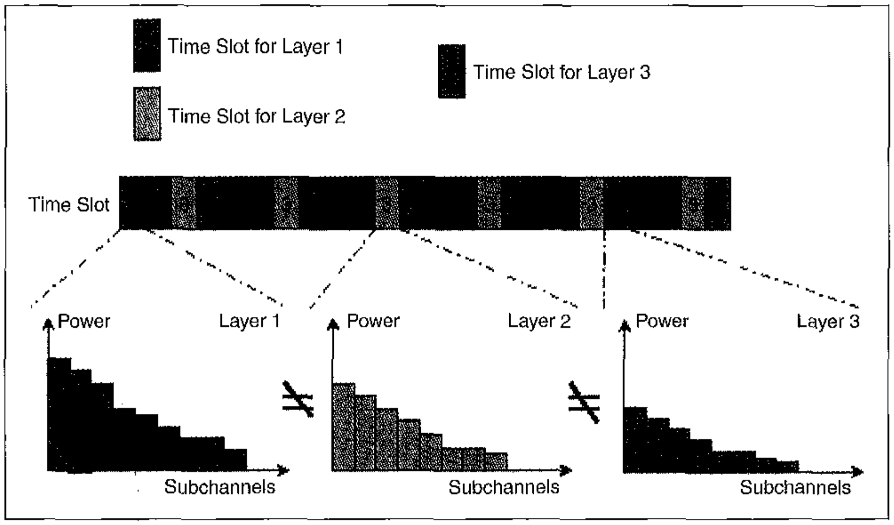
The primary motivation behind ADSL is the delivery of multimedia services. Early ADSL system designs emphasize providing video dial tone (VDT) MPEG-1 and MPEG-2 video to end-users. Given the fast growth of Internet and multimedia applications, the widespread acceptance of ADSL systems will depend on the ability to provide efficient delivery and a refined quality of multimedia data to the subscribers. As we described earlier, multimedia data has quite different characteristics compared to general data. One major characteristic is the layer coded structure where multimedia data is constructed into separate data streams, each representing a layer. The layers have different QoS requirements, i.e., data rate and error performance [BER, symbol error rate (SER)].

With limited communication resources, e.g., bandwidth and transmitted power, a key design issue in multimedia communications is to handle the layers differently and, therefore, efficiently. A larger amount of channel resources should be assigned to the layers with higher importance. For instance, it is well known that the use of scalability can enhance the error robustness of a video service. MPEG-2 video coded using spatial scalability option consists of one base layer and one/multiple enhancement layer(s). The base layer represents the basic shape of the picture and is the most important information, while the enhancement layers carry the details. The layers should be transmitted in such a way that channel loss is concentrated in the enhancement layer(s). For most communication systems, this can be accomplished by assigning enough channel resources to support the base layer transmission before considering the enhancement layers. The design of an ADSL system should follow the same criteria.

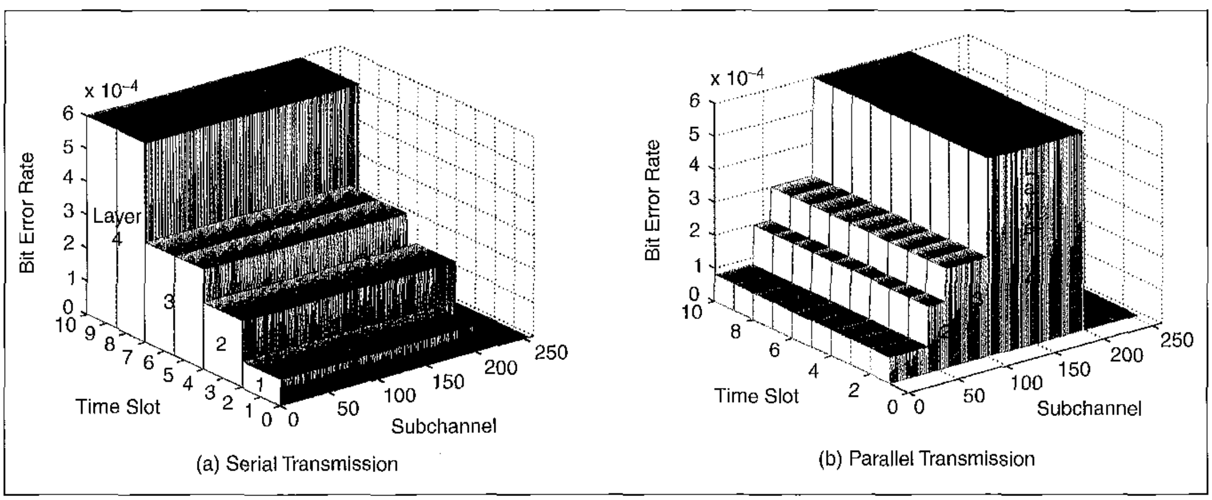
There are two transmission schemes for multimedia data over ADSL.

Serial Transmission: Time Division Multiplexing

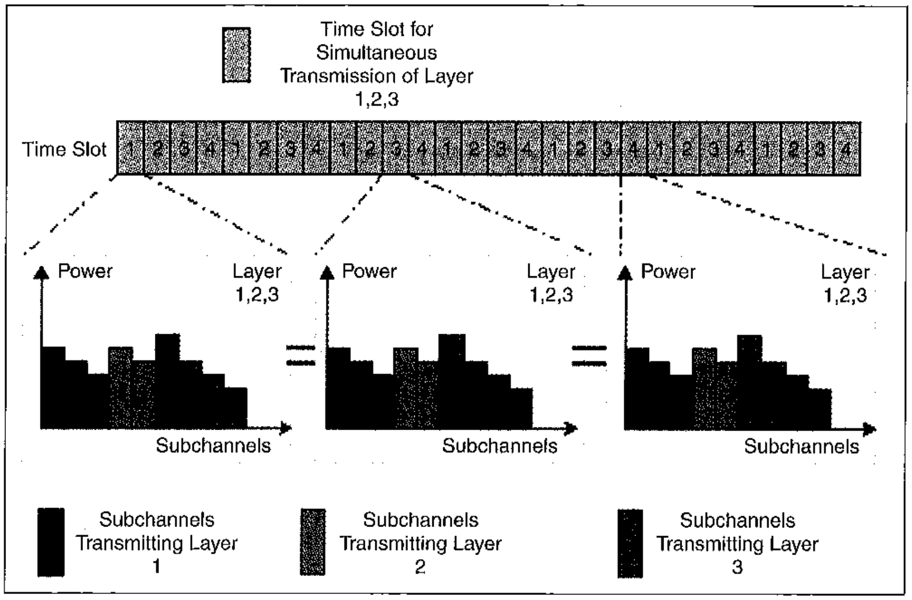
ADSL transmission is divided into time slots. In each time slot, only data from a single layer is transmitted. As



3. *Serial transmission: Every four time slots are grouped into a frame, therefore $T = 4$. This example assumes three source layers, where layer 1 is transmitted in slot 1,2 and layer 2 in slot 3,and layer 3 in slot 4. The power and bit-rate allocation in slot 1,2 and 3,4 are different.*



4. *Error performance across the time slots and subchannels for (a) serial and (b) parallel transmission.*



▲ 5. *Parallel transmission: The three source layers are transmitted in the same slot, each occupying a set of subchannels. Although the time slots are still grouped into frames, for every time slot in a frame, the power and bit-rate allocation remains the same.*

such, the layers are time-division multiplexed. The design task is thus finding a time slot to layer assignment to achieve high efficiency transmission and provide an acceptable QoS to the users. Such a system is named as serial transmission in [83] and [85]. The time slot as well as subchannel structure for the serial transmission is shown in Fig. 3. For three source layers, time slots 1 and 2 are assigned to transmit layer 1 while slots 3 and 4 are transmitting layers 2 and 3, respectively. The subchannel power and bit-rate distributions are different in slots 1 and 2 compared to 3 and 4. It is important to note that within a single time slot, the power and bit rate are allocated so that all the usable subchannels perform at the same error rate. For two time slots transmitting different layers, however, the subchannels error performances are completely different. An error performance distribution across the time slots and subchannels is illustrated in Fig. 4(a) with the system configuration of four layers from single or multiple sources, 256 subchannels, and ten time slots. The BER is constant within the same time slot and different across the time slots.

The design of the loading algorithm is to find the optimal time slot to layer assignment and then distribute the power and bit rate among the subchannels differently for each layer. In this case, the loading algorithm needs to compute the power and bit-rate allocations for every layer. In general the layers differ greatly in data size and error performance requirements, therefore the subchannel power and bit-rate allocation varies greatly from layer to layer. This leads to not only increased computational complexity but also difficulty in the transmitter/receiver design due to the frequent changes of the transmission parameters. We will study the loading algorithm in the next section.

Parallel Transmission: Frequency Division Multiplexing

Most ADSL channels are slow varying or static. The channel gain and noise differences across frequency are more severe compared to those across the time. The serial transmission copes with such differences through the power and bit-rate loading, which eliminates the difference. Such differences, however, can be adopted to provide different error performances to the layers, which inspires the parallel transmission [83], [87].

The essence of the parallel transmission is to utilize the special ADSL channel characteristics by transmitting multimedia layers simultaneously, each occupying a set of subchannels. As shown in Fig. 5, it is equivalent to frequency division multiplexing the data streams corresponding to different multimedia layers. The parallel transmission relies on a subchannel-to-layer assignment that groups the appropriate subchannels together for the layers. In general, data from the important layers are transmitted through the subchannels with better channel performance, i.e., larger channel gain, lower noise variance, or simply the good subchannels. Such assignment provides reliable transmission to the most important layers without large power consumption. It could be more advantageous under low power constraints. In contrast to the serial transmission, the parallel transmission can integrate various traffic flows with different QoS requirements without any frequent channel parameter changes.

The error performance distribution of the parallel transmission is quite different from that of the serial transmission, as can be seen from Fig. 4(b). It achieves constant error performance across the time slots but different error performances at the subchannels transmitting different layers. For this reason, the associated loading algorithm is different from that of the serial transmission. It consists of a subchannel-to-layer assignment scheme which assigns the optimal number of subchannels to each layer, and a subchannel power and bit-rate allocation scheme which realizes different error performances across the subchannels. The loading algorithm is described together with the serial loading algorithm in the following section.

Channel Loading Algorithms

Here we describe the loading algorithms for the serial and parallel transmissions. In general, the optimization problem has two forms: minimizing channel resource usage for a given QoS requirement and maximizing media quality for a given amount of the channel resources.

Problem I: Channel Resource Consumption Minimization for a Given QoS Requirement

The success of multimedia business depends on how to provide user satisfied services at low service cost. To a communication system, the service cost would mainly depend on the channel resource consumption. Hereby we interpret power consumption as a major and informative indicator of the channel resource consumption and emphasize the loading algorithm design on finding the appropriate time slot/subchannel-to-layer assignments as to minimize the total transmitted power usage. The constraints of the problem are the targeted throughput and error performances, defined differently by each layer. A typical application in this case would be an integrated audio, video, and data service where each service is classified as a layer. Or, it can be multiple audiovisual objects from the same source as in the interactive multimedia applications, i.e., MPEG-4 [24], where each object defines a layer. In both cases, each layer is associated with a targeted data rate and BER. The corresponding throughput in terms of the number of bits per transmission can be determined from the data rates and the system sampling rate.

The slot/subchannel-to-layer assignments rely on a novel yet simple bit-rate and power allocation scheme which achieves the same BER across the subchannels to be optimized. For the parallel transmission, these subchannels are simply the ones that transmit the same layer in a single ADSL transmission. This allocation scheme is essential since for each layer it can effectively reduce the transmitted power consumption and achieve the throughput and BER requirements. Several algorithms have been proposed in this area [61], [68], [71], [72]. These techniques assume that all the usable subchannels perform at the same error performance, which stands for the optimal solution for data transmission. We hereby refer to them as data loading algorithms. The algorithms are developed based on the water-filling approach.

Each independent subchannel is equivalent to an AWGN channel, with SNR valued $E_s G$ where E_s represents the transmitted power and G represents the channel gain to noise ratio (CGNR) $G = |H|^2 / N_0$. The rate b provides a measure of the subchannel throughput and is computed as

$$b = \log_2\left(1 + \frac{\text{SNR}}{\Gamma}\right) \tag{2}$$

The widespread acceptance of ADSL systems will depend on the ability to provide efficient delivery and a refined quality of multimedia data to the subscribers.

where Γ is defined as the SNR gap in [60]

$$\Gamma = \frac{\text{SNR}}{2^b - 1}. \tag{3}$$

Γ measures the SNR distance from the capacity ($C = \log_2(1 + SNR)$). For uncoded QAM, Γ is directly related to the bit error performance. For instance, for a BER of 10^{-6}, $\Gamma = 8.8$ dB. Based on the water-filling result [67], the power allocated to each subchannel is proportional to Γ / G_k, where G_k represents the CGNR of subchannel k. Subchannels with large G_k, referred as the good subchannels, are allocated with higher amount of power. The corresponding b_k can be computed from (2), which can be any real number. The current transmission systems, however, support only discrete bit rate. The Hughes-Hartog algorithm suggested a greedy algorithm which assigns the bits successively to the subchannels until the data throughput is reached [61]. In [71], the bit-rate distribution is computed by rounding of approximate water-filling results and finding the optimal Γ. Another power allocation algorithm based on table-lookup is described in [72]. A technique that appeared recently is the Campello algorithm, which achieves the optimal system performance and provides adaptation to channel variations [68]. An important feature of this kind of loading algorithm is that different total power constraints results in different number of the usable or "on" subchannels. A subchannel is usable or "on" if it is assigned with nonzero bit rate therefore carries information.

Serial Transmission: Time Slot Assignment

Time slot assignment generates a set of time slot distributions. For real-time services, a set of T time slots are grouped together into a time frame, where the slot assignment is repeated frame by frame. This is similar to the time division multiplexing (TDM). Within a single time frame, that is T slots, the channel is assumed to be static. In this case, the time slots assignment is to find the optimal distribution of $\{T_k\}_{k=1}^{N}$ with a constraint of $\sum_{k=1}^{N} T_k = T$ for N source layers. The optimization can be represented mathematically as follows:

Given QoS Requirement	$\left\{R_{m,T}, \mathrm{BER}_m\right\}_{m=1}^{N}$
Find	$\{T_m\}_{m=1}^{N}$ where $\sum_{m=1}^{N} T_m = T$ to
Minimize	$E = \sum_{m=1}^{N} \frac{T_m}{T} E_m\left(R_{m,T}, \mathrm{BER}_m, C_m\right), = \sum_{m=1}^{N} \frac{T_m}{T} \sum_{k=1}^{C_m} \frac{\Gamma(BER_m)}{g_k}\left(2^{b_{m,k}} - 1\right),$
where	$\frac{T_m}{T} \sum_{k=1}^{C_m} b_{m,k} = R_{m,T},$ (4)

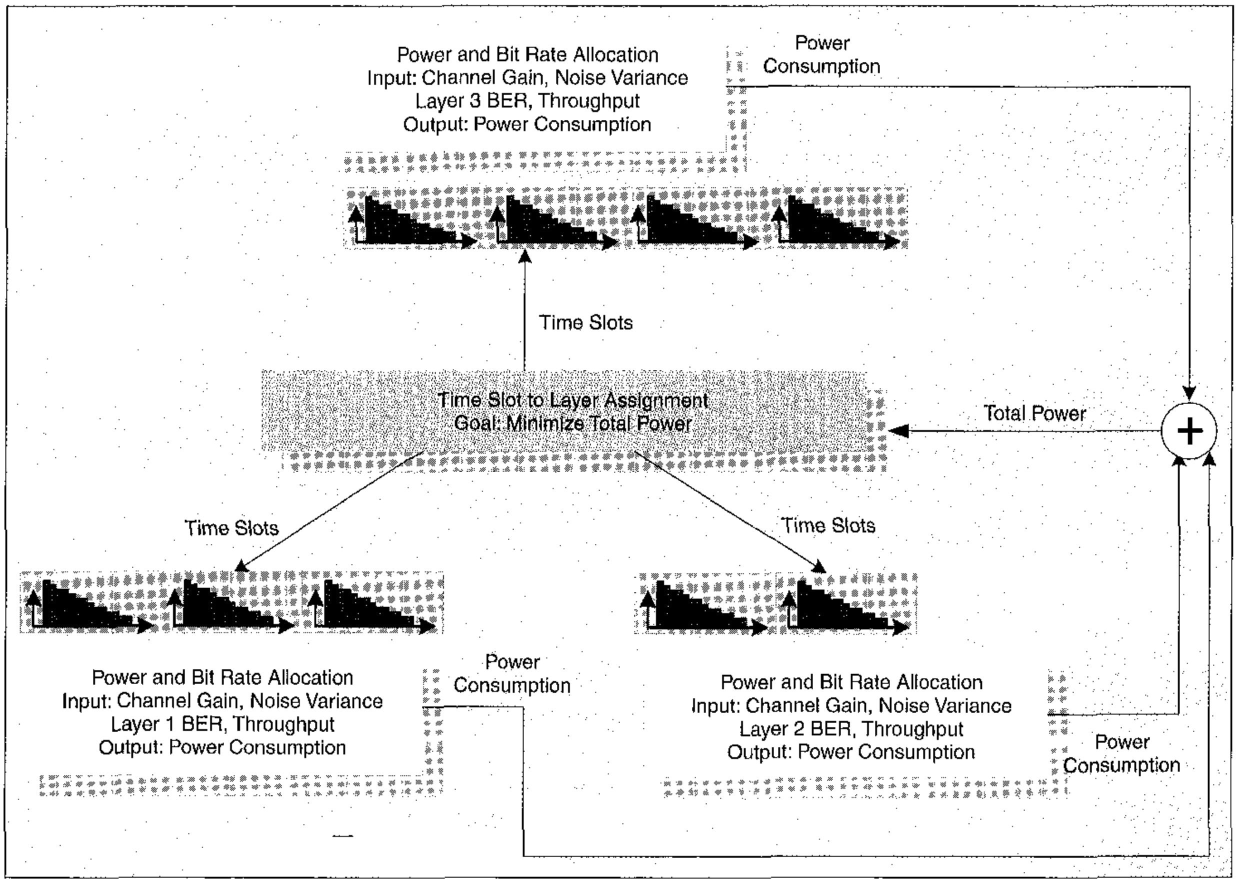
where E_m is the power consumption of all the usable subchannels when transmitting layer m, and E represents the average power consumption per transmission; $R_{m,T}$ is the throughput requirement of layer m and C_m represents the number of subchannels that are "on" during layer m's transmission; g_k and $b_{m,k}$ represent the CGNR and bit rate at subchannel k, respectively.

For each distribution, the throughput and BER to be achieved in each time slot can be computed based on the required data rate and the number of time slots assigned to the layers. Then, within each time slot, the data loading algorithm is applied to compute the power and bit-rate allocation based on the layer's requirements. The optimal time slot assignment is the one with minimum amount of the transmitted power. Figure 6 shows the structure of the serial loading algorithm. An exhaustive search can be applied to find the solution.

Parallel Transmission: Subchannel-to-Layer Assignment

The subchannels are first sorted in a decreasing CGNR order, and the layers are sorted in an increasing BER order. The subchannels-to-layers assignment is to compute the optimal number of subchannels defined as $\{C_m\}_{m=1}^{N}$ for N source layers to minimize the total power consumption. For a given $\{C_m\}_{m=1}^{N}$, subchannels 1 to C_1 transmit layer 1 while subchannels C_1+1 to C_1+C_2 transmit layer 2, and so on. The corresponding optimization problem is then

Given QoS Requirement	$\left\{R_{m,T}, \mathrm{BER}_m\right\}_{m=1}^{N}$
Find	$\{C_m\}_{m=1}^{N}$ where $\sum_{m=1}^{N} C_m \le C_T$ to
Minimize	$E = \sum_{m=1}^{N} E_m\left(R_{m,T}, \mathrm{BER}_m, C_m\right), = \sum_{m=1}^{N} \sum_{k=C_{m-1}+1}^{C_m} \frac{\Gamma(\mathrm{BER}_m)}{g_k}\left(2^{b_k} - 1\right),$
where	$\sum_{k=C_{m-1}+1}^{C_m} b_k = \mathrm{R}_{m,T}.$ (5)



▲ *6. Loading algorithm of serial transmission.*

Here E represents the total power consumption per ADSL transmission.

For the subchannels transmitting each individual layer, the data loading algorithm allocates the power and bit rate and computes the overall power consumption. The subchannel-to-layer assignment picks the optimal assignment as the one with minimal overall power consumption. The parallel loading algorithm is shown in Fig. 7.

An exhaustive search can be applied to find the best $\{C_m\}_{m=1}^{N}$ distribution by comparing the power consumption. In [84], a successive search algorithm was developed. It defined two efficiency measures to detect whether the optimal solution is reached.

Subchannel Distribution Efficiency

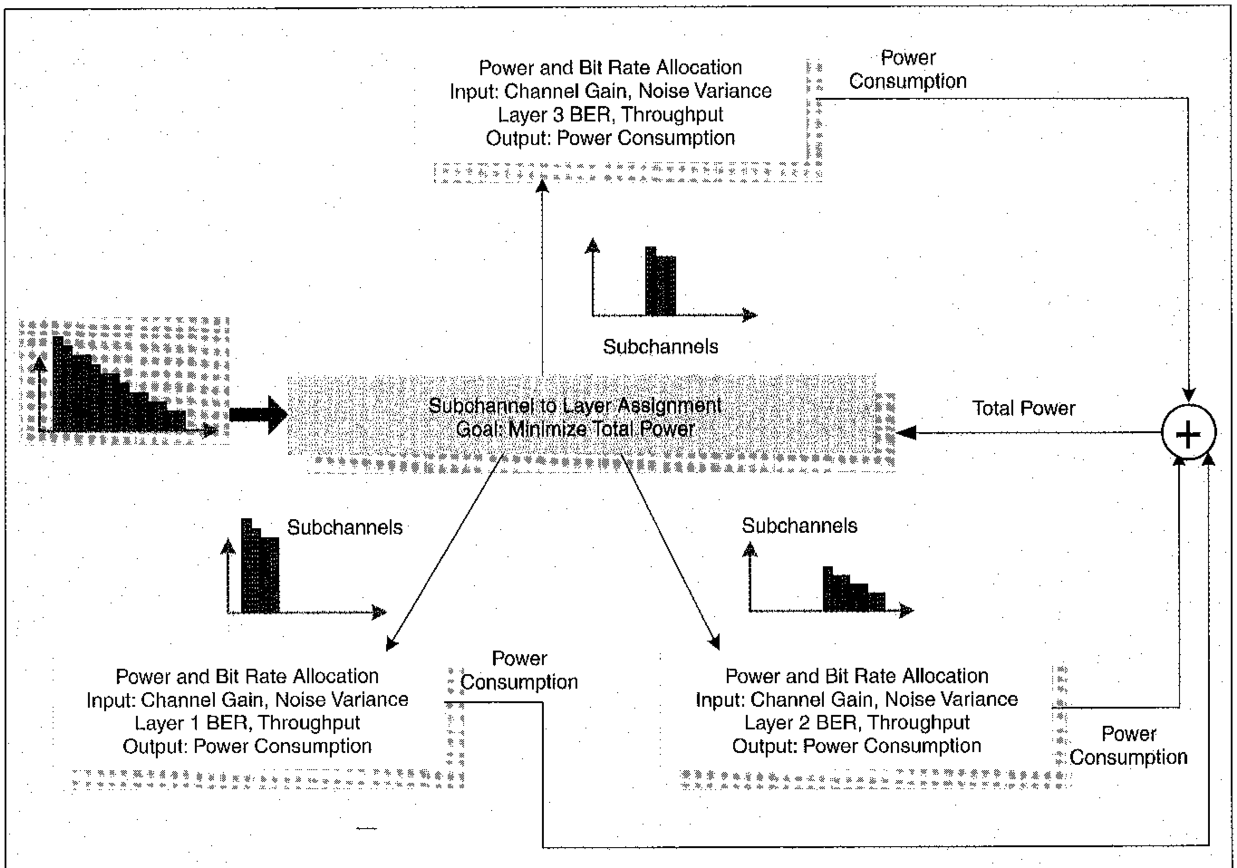
For a given number of total subchannels in use, if no movement of a single subchannel from one layer to another, excluding layer $N+1$, can reduce the power consumption, then the subchannel distribution efficiency is achieved.

Total Subchannel Efficiency

This is to decide what is the optimal number of subchannels occupied by all the layers. If no movement of a single subchannel from layer $N+1$ to layer $l \in [1, N]$ can reduce the power consumption, then the total subchannel efficiency is achieved.

To provide fast Internet access, ADSL should support efficient image transmissions.

Here the $(N+1)$th layer is introduced to represent the subchannels that are "off." The optimal solution must satisfy the above two efficiencies. The loading algorithm starts from a particular subchannel distribution $\{C_{1,\min}, C_{2,\min}, \ldots, C_{N,\min}\}$, where $C_{k,\min}$ is the minimal number of subchannels to carry layer k. This initial distribution satisfies the subchannel distribution efficiency. If it also satisfies total subchannel efficiency, the solution is reached. Otherwise the algorithm finds the layer that yields the maximal power reduction by adding one subchannel to it. After updating the layers, the subchannel distribution efficiency may not hold and, if so, $\{C_i\}_{i=1}^{N}$ is optimized by moving the subchannel from layer to layer except layer $N+1$. This is repeated until the distribution satisfies both the subchannel distribution efficiency and the total subchannel efficiency. The detailed optimization can be found in [84].



▲ *7. Loading algorithm of parallel transmission.*

Problem II: Quality Maximization for a Given Amount of Channel Resources

This optimization problem is usually associated with single media applications such as image downloading and streaming video and audio. For image video and audio, the receiving quality is usually measured as the distortion between the original and the reconstructed data. The overall distortion depends on both source coding and channel transmission. Since our interest resides in the study of the effectiveness of ADSL channel transmission, the channel distortion provides a sufficient indicator in the evaluation of the receiving quality. Less distortion indicates better quality. The channel distortion can be approximated as [75]

$$D_c = \sum_{m=1}^{N} Pe_m W_m \tag{6}$$

where W_m represents the average distortion caused by a single bit error at layer m and Pe_m is the BER for layer m, which is a function of power, bit rate, and channel gain. While the W_m is determined mainly by the source layer, the loading algorithm can find the optimal Pe_m distribution so that the distortion D_c can be minimized.

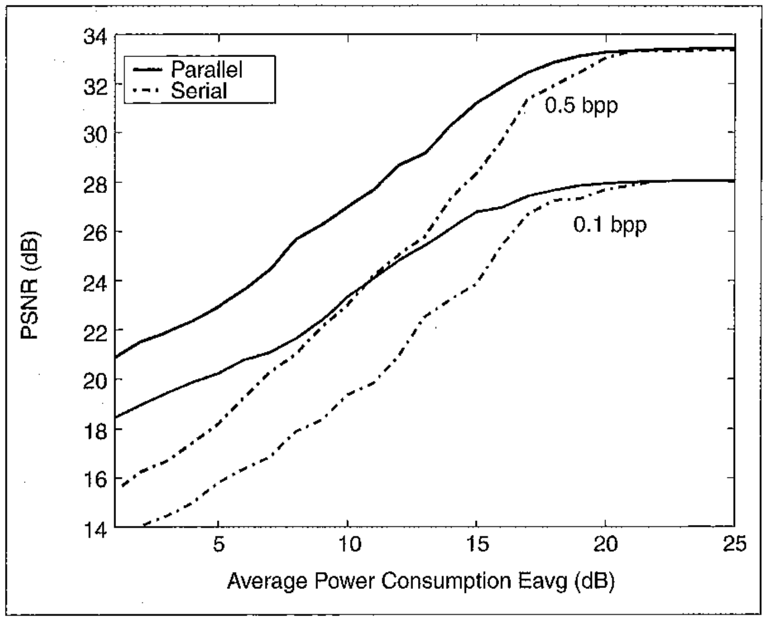
The loading algorithms are based on the fact that the subchannels transmitting the same layer should have the same BER performance. For this optimization, however, the designing criteria of the power and bit-rate allocation scheme is to minimize the BER for a given amount of power consumption while achieving the throughput. As such, the BER for each layer is unknown prior to the optimization. A simple loading algorithm addressing this problem is proposed in [69]. Define $R_{m,T}$ as the throughput associated with layer m in a single ADSL transmission. Using the bit rate and power loading algorithm in [69], Pe_m can be computed as

$$Pe_m\left(R_{m,T}, E_{m,T}, G_m, C_m\right) \approx 4Q\left(\sqrt{\frac{3E_{m,T}G_m/C_m}{\left(2^{R_{m,T}/C_m} - 1\right)}}\right) \tag{7}$$

where

$$G_m = \frac{C_m}{\sum_{i=C_{m-1}+1}^{C_m} 2^{(b_i - R_m)} \Big/ g_i}$$

is the rate averaged channel gain to noise ratio (RACGNR) for layer m and



▲ 8. *Received PSNR performances as a function of the average subchannel power consumption. Both parallel and serial transmissions are shown and compared. Image Lena is compressed to 0.5 bpp and 0.1 bpp using four-level subband coding combined with vector quantization scheme. The simulated ADSL channel is configured as 256 subchannels, 512 bits throughput, QAM4, 8, 16, 32, 64 modulations.*

$$E_{m,T} = \sum_{i=C_{m-1}+1}^{C_m} E_i$$

represents the total power consumption of the subchannels transmitted layer m. We refer $E_{m,T}$ as layer power. And the power constraint is defined as $\sum_{m=1}^{N} E_{m,T} \le E_T$ where E_T represents the total power consumption, a major indicator of the channel resource usage.

Under the above assumption, the optimization function for this type of problem can be formulated as

$$\begin{array}{ll} \text{Given throughput} & \left\{R_{m,T}, W_m\right\}_{m=1}^{N}, \\ \text{Find} & \left\{E_m, R_m, C_m, T_m\right\}_{m=1}^{N}, \text{ to} \\ \text{Minimize} & D_c = \sum_{m=1}^{N} Pe_m(R_m, E_m, C_m, T_m) W_m, \\ \text{subject to} & \sum_{m=1}^{N} \frac{T_m}{T} E_m \le E_T,\ \sum_{m=1}^{N} C_m \le C_T \end{array} \tag{8}$$

where E_T is the total power constraint, C_T is the maximum number of subchannels allowed to use, and Pe is the BER function.

Parallel Transmission

From (7), for a given subchannel-to-layer assignment and the layer throughput requirements $\{C_m, R_{m,T}\}_{m=1}^{N}$, the BER and thus the power and bit-rate assignment depend on $E_{m,T}$ and E_T. According to [83], $E_{m,T}$s are derived through a layer level power allocation as

$$E_{m,T} = \Phi^{-1}_{\alpha_m}\left(\lambda_{\text{opt}}/W_m\right) \tag{9}$$

where

$$\Phi_\alpha(x) = \sqrt{\frac{\alpha}{x}}\exp(-\alpha x) \tag{10}$$

and λ_{opt} satisfies

$$\sum_{m=1}^{N}\Phi^{-1}_{\alpha_m}\left(\lambda_{\text{opt}}/W_m\right) = E_T$$
$$\text{with } \alpha_m = \frac{3G_m}{2C_m(2^{R_{m,T}/C_m}-1)}. \tag{11}$$

The major task in the loading algorithm is to find the optimal $\{C_m\}_{m=1}^{N}$ distribution to minimize the channel distortion D_c. Similar to the parallel loading algorithm for the resource minimization problem, a successive search can be applied. In this case, the total subchannel efficiency and subchannel distribution efficiency are defined with respect to the goal of distortion minimization. The loading algorithm starts from $\{C_m = C_{m,min}\}_{m=1}^{N}$. The optimal $\{E_{m,T}\}_{m=1}^{N}$ is derived using (9). The algorithm then examines the total subchannel efficiency to decide whether to increase the occupied subchannels. Any change in subchannel assignment would yield a different $\{E_{m,T}\}_{m=1}^{N}$ and thus a different subchannel power and bit-rate assignment. The algorithm then examines the total number of subchannel efficiency and repeats the same procedure until both efficiencies are satisfied. To further reduce the computational complexity, several suboptimal algorithms have been developed in [83].

Serial Transmission

The time slot-to-layer assignment also requires a layer level power allocation to achieve different error performance during different time slots. We define the sum of subchannel power during layer *m*'s transmission to be e_m and the number of slots assigned to layer *m* be T_m. For a given $\{T_m\}_{m=1}^{N}$, the optimal $\{e_m\}_{m=1}^{N}$ can be resolved by finding the λ such that

$$E_T = \sum_{m=1}^{N}\frac{T_m}{T}\Phi^{-1}_{\beta_m}\left(\frac{\lambda}{(1-\rho_m)W_m + \rho_{m+1}W_{m+1}}\right) \tag{12}$$

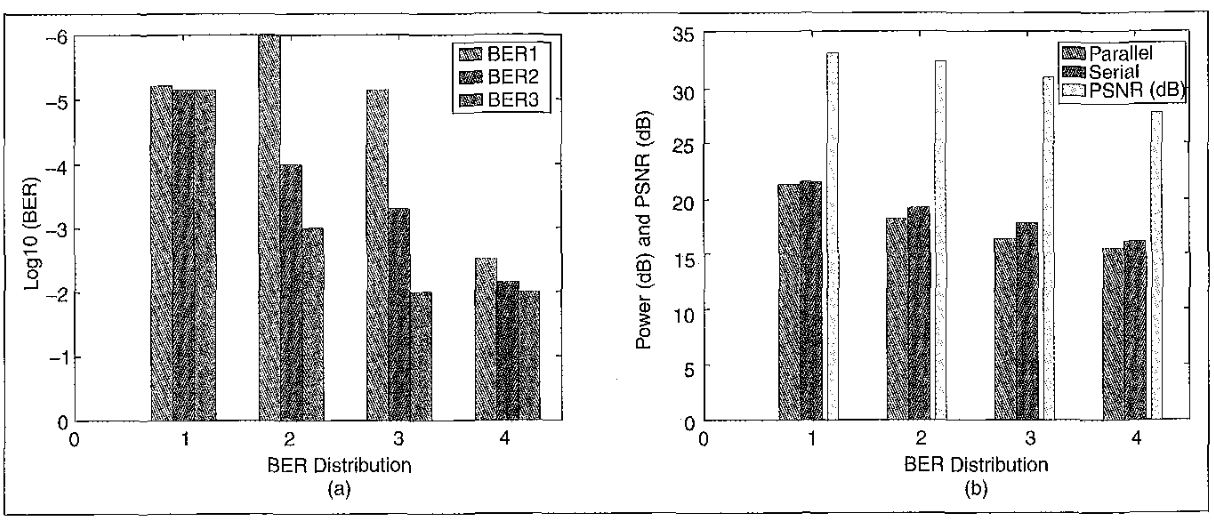
where

$$e_m = \Phi^{-1}_{\beta_m}\left(\frac{\lambda}{(1-\rho_m)W_m + \rho_{m+1}W_{m+1}}\right)$$
$$\beta_m = \frac{3}{C_m\left(2^{\frac{R_{m,T}}{C_m}}-1\right)\frac{1}{C_m}\sum_{i=1}^{C_m}\frac{1}{g_i}2^{(R_i - R_{m,T}/C_m)}}.$$

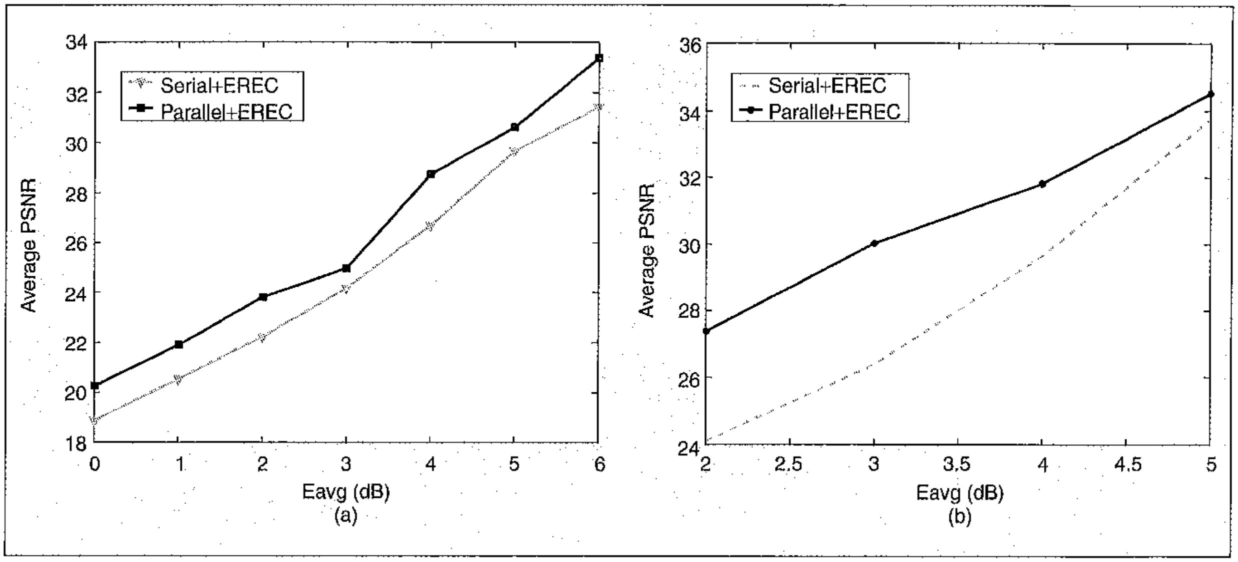
$R_{m,T}$ represents the throughput during the time slots transmitting layer *m*, and C_m represents the total number of usable or "on" subchannels to transmit layer *m* [83].

Some Applications

Having studied the optimization algorithms for both serial and parallel transmissions, we will present some typical applications that utilize those two schemes for ADSL transmission.



▲ 9. *Subband image simulation results: (a) Image subband BER distributions created based on user requirements and (b) average channel power usage for image subbands at different BER distributions. Image Lena is compressed to 0.5 bpp using four-level subband coding, vector quantization scheme. The simulated ADSL channel is configured as 256 subchannels, 512 bits throughput, QAM4-64 modulations. The PSNR of the reconstructed image is also illustrated for reference purpose.*



▲ 10. *Performance comparison of the parallel and serial transmissions combined with EREC using 60 frames of (a) "Trevor" and (b) "Miss America" sequences. The performance measure is the averaged PSNR over 60 frames versus average power consumption per subchannel. (Note: For a color picture, PSNR is an average of Y, C_b, and C_r components, that is $PSNR = 4PSNR_Y + PSNR_{C_b} + PSNR_{C_r}/6$.)*

Image Transmission

Today's key Internet activities such as E-commerce require significant amounts of image downloading. To provide fast Internet access, ADSL should support efficient image transmissions. Subband/wavelet coding has been a well-known scheme for image compression [4], [75], [82]. The compressed image consists of a set of subbands with different levels of perceptual importance. Therefore, they can be classified into layers. We use a typical grayscale image "Lena" in the simulation. The ADSL system consists of 256 subchannels, employing both serial and parallel transmission schemes. The bit rate that each subchannel can transmit is bounded by $R_{max} = 6$ and $R_{min} = 2$. The corresponding modulations are QAM64, QAM32, QAM16, QAM8, and QAM4.

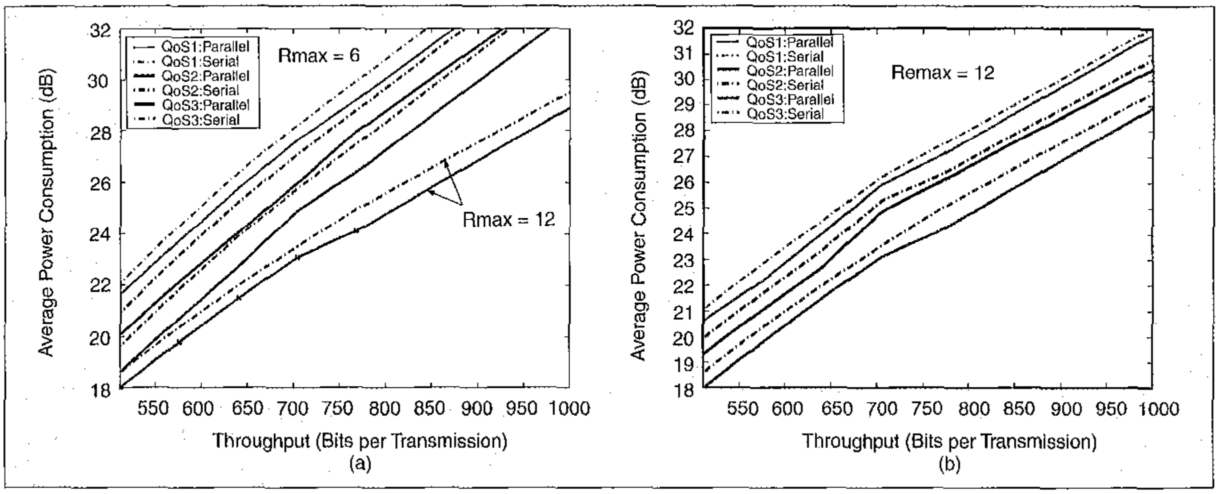
First, the "Quality Maximization" problem is considered where the image quality is measured in terms of peak signal to noise ratio ($PSNR = 10\log(255^2/MSE)$). MSE represents the mean squared error between the original image and the reconstructed image, a distortion measure widely used. The power constraint is represented as the average power consumption per subchannel $E_{avg} = E_T/C_T$ where C_T is the total number of subchannels in the ADSL system, in this case, $C_T = 256$. We refer to the throughput for each ADSL transmission as B_T. For the parallel transmission, the amount of data from each layer in a single transmission is computed proportionally to the bit size of the layer. Figure 8 illustrates the received peak signal-to-noise ratio (PSNR) versus E_{avg} performance. The parallel transmission outperforms the serial transmission. Particularly, for $B_T = 512$, it achieves a 8-10 dB PSNR improvement at 0.5 bits per pixel (bpp) source rate and 4-6 dB at 0.1 bpp source rate. We observe that for the same power constraint, increasing throughput B_T results in decreased PSNR performance, since carrying more information using the same power would suffer from degraded error performance. We also observe that the parallel transmission yields increased performance improvement as the throughput increases, especially for E_{avg} valued at 10 dB and higher.

Table 2. QoS Requirements.			
Requirement	Service 1 200 Kb/s	Service 2 64 Kb/s	Service 3 0 Kb/s
QoS1	10^{-6}	10^{-5}	10^{-3}
QoS2	10^{-5}	10^{-3}	10^{-6}
QoS3	10^{-3}	10^{-5}	10^{-6}

Next, we want to define the QoS requirement in terms of the subband BER. For instance, subband 1 requires BER1, subbands 2 and 3 require BER2, and the rest require BER3. In Fig. 9(a), four BER distributions are shown ranging from 10^{-6} to 10^{-2}. The resulting average power E_{avg}s for both the serial and parallel transmissions are computed and compared in Fig. 9(b). We observe that the parallel transmission maintains superior performance for all the scenarios. Comparing the performance improvement achieved for distributions 2 and 3 to that of distributions 1 and 4, we can conclude that the parallel transmission is very useful for delivering a mixture of layers with large BER differences.

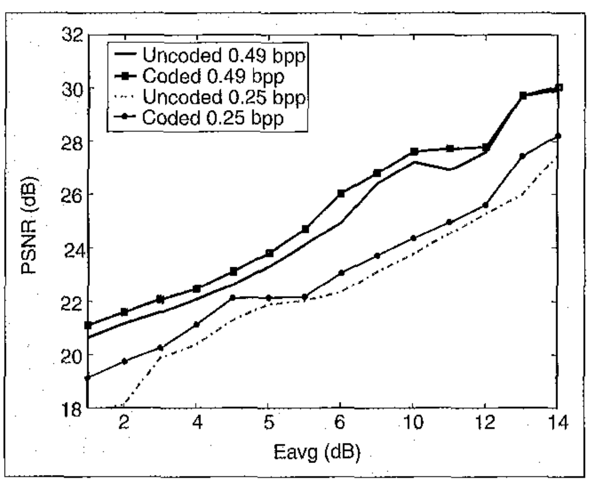
H.263 Low Bit-Rate Video

The recent development of video coding emphasized improving the error resilience of compressed video streams. One of the key features in error resilience is being able to



▲ 11. *Transmitted power consumption versus data throughput for (a) $R_{max} = 6$ and (b) $R_{max} = 12$. The ADSL system is configured to have 256 subchannels and using QAM modulations with $R_{min} = 2$.*

extract and separate the important information and utilize unequal error protection. Within a video bit stream, information bits are of different importance. Using H.263 low bit-rate video as an example, low frequency coefficients, particularly DC coefficients, represent more important information compared to high frequency coefficients; motion vectors have more impact on the decoded video quality than DCT coefficients if corrupted. The error-resilient entropy code (EREC) was applied to separate the information bits into the layers [86]. EREC was originally proposed in [76] and [77] to provide increased resilience to random and burst errors while maintaining high compression. It reorganizes variable length blocks to fixed length slots such that each block starts at a known position and the beginning of each block is more immune to error propagation than those at the end. In [86], each slot is further divided into layers and both the parallel and serial transmission are applied to achieve different error performances for different layers. Figure 10 shows the decoded video quality in terms of PSNR based on 60 frames of the standard QCIF (176×144 pels) color sequences "Trevor" and "Miss America." We observe that the parallel transmission with the EREC system achieves 2-4 dB gain over the system relying on the serial transmission with EREC [86].



▲ 12. *Performance comparison of coded and uncoded system using "Lena."*

Integrated Service of Video, Speech, and Data

To cover a wide variety of multimedia services, we select three services with data rate 200 kb/s, 64 kb/s, and 10 kb/s, respectively. The QoS requirements are listed in Table 2. The QoS1 corresponds to the integrated video, speech, and data scenario in Table 1. The power consumption for the serial and parallel transmission systems are studied in Fig. 11. Clearly, the parallel transmission results in a 0.5-1 dB power reduction compared to the serial transmission. It is not surprising that as the throughput increases, so does the power consumption [88].

We are interested in examining the effect of R_{max} for a fixed R_{min}. By comparing Fig. 11(a) and (b), the power consumption with ($R_{max} - R_{min} = 11$) modulators available at transmitter/receivers is 2-3 dB lower than that with five modulators. This is due to the fact that although the subchannels with higher channel gain are assigned with more power to transmit at higher rate, R_{max} puts a constraint on the highest bit rate, so that any assigned bit rate higher than R_{max} is truncated. Instead, the system has to assign the subchannels with lower channel gain to carry the information which results in an increased power consumption.

Further Channel Optimization Through Coding

Powerful coding techniques, such as trellis codes, are feasible means for further improving the system performance [79]. There can be many ways to concatenate trellis codes on an MCM system. Most of the existing approaches use a single encoder/decoder which codes across the subchannels [81]. It has the advantage of low decoding latency and complexity. It also propagates the error in one subchannel to others. As a result, the subchannel performances depend upon one another. This is not feasible in the parallel transmission system.

In [83], it is proposed to use separate encoder/decoders for the subchannels transmitting different source layers. Since the number of source layers is usually small, this will not yield great increase in complexity and latency. Figure 12 shows the performance comparison of the uncoded system the coded system using Wei's four-dimensional, 16 state codes combined with trellis shaping [80].

Summary and Discussion

This tutorial describes the present transmission systems for delivering multimedia services over ADSL. We emphasize two transmission techniques: the parallel transmission that utilizes frequency division multiplexing and the serial transmission that utilizes time division multiplexing. We have defined two optimization problems: 1) minimizing channel resources for a given set of QoS requirements and 2) maximizing receiving quality for a given amount of channel resources. We then studied the corresponding channel loading algorithms for both serial and parallel transmission. The algorithms are applied to a subband coded image, H.263 video, and integrated services. From the simulation results, the parallel transmission outperforms the serial transmission by matching the channel characteristics to the multimedia characteristics. It achieves unequal error protection naturally and efficiently. We observe that the performance degrades as the channel throughput increases, for a given amount of channel resource. Furthermore, the number of modulators R_{max} puts a limit on the system capacity. Such limitation can be resolved by sophisticated hardware evolutions which allow a large number of modulator/demodulators in the transmitter/receivers. In conclusion, it is clear that an ADSL system with the parallel transmission can provide reliable and yet resource efficient transmissions for both single and integrated multimedia services.

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