Modeling and Simulation of an ADSL Transmitter

by

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Abstract

This paper gives a short introduction of Asymmetric Digital Subscriber Line (ADSL) technology and briefly discusses its importance. ADSL uses Discrete Multitone Modulation technique and enables very high rate data throughput over normal telephone lines. This paper discusses at length the modulation scheme and also describes in detail the functioning of an ADSL transmitter

Asymmetric Digital Subscribe Line (ADSL) is used to deliver high-rate digital data over existing ordinary telephone lines. ADSL facilitates use of normal telephone services, ISDN and high-speed data transmission simultaneously. ETSI and ANSI have defined the current ADSL standard. OFDM or Discrete Multi-tone technology is the modulation technology that allows the transmission of high-speed data. The high bandwidth provided by ADSL makes technologies like Video-on-demand and video-conferencing possible. ADSL is seen as a smooth transition between present day telephone lines to the fiber optic links of the future. It is assuming lot of importance because of its economic viability.

DSL Architecture

ADSL allows data rates of 6Mbps downstream and about 640Kbps upstream. The transmission bandwidth is 1.1MHz downstream and 256KHz upstream. The limit of 1.1MHz is due to power constraints imposed by FCC. The frequency domain is divided into 256 sub-channels of 4kHz. The 0-4kHz channel is dedicated for voice, 2-6 channels are not used. There are two versions of ADSL:

- Frequency Division Multiplexed: Upstream and downstream channels use different frequency range. Upstream uses 7-32 and downstream uses 32-256
- Echo cancelled: Upstream and downstream channels overlap. Upstream uses channels 7-32 and downstream uses channels 7-256. So echo-cancellation techniques are used to make the transmitted and received signals independent of each other

ADSL System Engineering Requirements

The twisted-pair telephone loop between a central office and a subscriber premises can have a length of a few kilofeet to a few tens of kilofeet. A twisted telephone loop consists of many cable sections of sizes from 19 gauge to 26 gauge. Due to installation practice, there are bridged taps associated with telephone subscriber loops. This causes magnitude and phase distortion for the ADSL bandwidth. Reflections due to the bridged taps and impedance mismatch are also potential cause of intersymbol interference. Counter measures such as adaptive channel equalization and multi-carrier modulation become necessary.

Many twisted pairs share the same electrical sheath and plastic covering within a single twisted pair cable. Crosstalk exists between adjacent cables. There is Near End crosstalk (NEXT) and Far End crosstalk (FEXT).

Apart from these, telephone lines attenuate signals and attenuation increases with increasing frequencies. At 1.1MHz, the attenuation of a 24 gauge wire is

Table 1: Attenuation of a 24 gauge wire signal at 1.1MHz

Length of line (in kft)	Attenuation (in dBm/Hz)
10	-70
12	-90
14	-100
16	-110
18	-120

Thus, very efficient schemes for modulation and equalization are a must to counter severe channel attenuation (due to loading coils), inter-symbol interference, cross-talk noise, additive white Gaussian noise and impulse noise. Extensive Digital processing techniques, such as adaptive channel equalizers and adaptive echo cancellers have been adapted for ADSL systems. The advancements in VLSI and the presence of high-speed DSPs have made realization of the above mentioned techniques. Implementation of very good error correction techniques is also made possible.

Discrete Multi-tone Modulation: Discrete Multi-tone Modulation (DMT) technique is chosen as an ANSI standard line code for ADSL system. Multi-carrier modulation techniques are primarily used to partition data transmission channels with inter-symbol interference, into a set of parallel sub-channels, each with its own carrier, while the carriers form an orthogonal set [1]. Data transmission in one sub-channel is (ideally) independent of the data transmitted in other sub-channels and channel is partitioned such that each sub-channel has a near flat response. DMT modulation [4] is the most preferred because of its computational ease. For DMT, the modulation and demodulation (transmitter) schemes can be carried out with Inverse FFT (transmitter), and FFT (receiver). To implement an N sub-channel DMT system, a 2N size FFT is required. The forced conjugate symmetry in the frequency domain will result in real-valued time domain samples.

The general structure of a DMT system is illustrated in Figure 1, where $\{X_{0,} X_1,\ldots, X_{n-1}\}$ Are the original complex, input data symbols, $\{x_k\}$ is the modulated data sequence, $\{y_k\}$ is the received sequence (after the removal of cyclic prefix, which will be discussed very soon), and $\{Y_0, Y_1, \ldots, Y_{n-1}\}$ are the decoded complex data symbols.$\{p_i\}$ and $\{p_i^*\}$ are the modulating and demodulating orthonormal vectors.

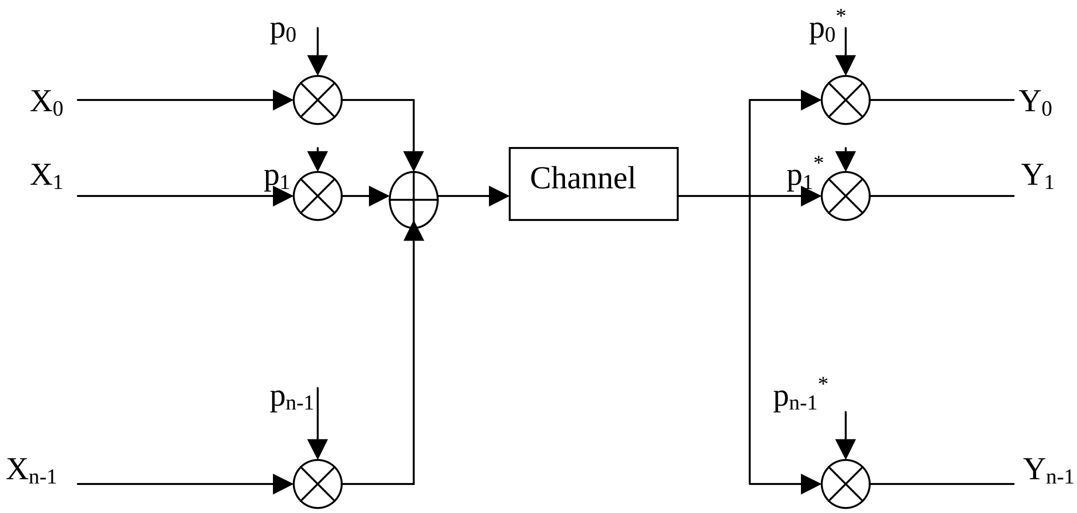

Figure 1: Basic Discrete Multitone Modulation Scheme

DMT also allows for a variable data rate based on signal to noise ratio of individual sub-channel. This allows for optimal usage of bandwidth and is suitable for applications that require bandwidth on demand.

Architecture of an ADSL transmitter:

The block diagram of a basic ADSL DMT transmitter [3] is illustrated in Figure 2.

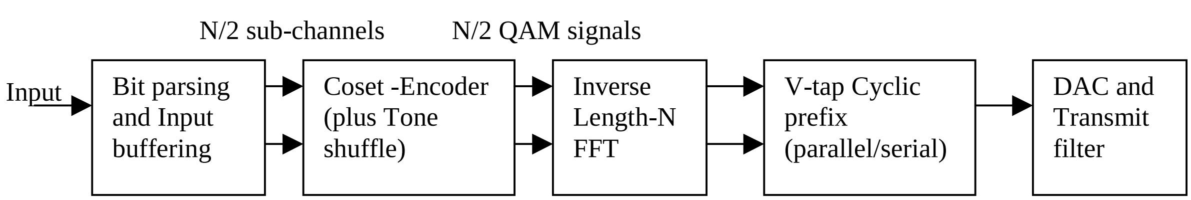

Figure 2: Block Diagram of a DMT Transmitter

Each sub-channel has a width of Δf=4kHz and T=250μsecond is the block symbol period of each multi-carrier channel. To maximize the transport capacity of any given loop, bit loading algorithm [5] is used to allocate “number of bits” to the channels according to the channel noise conditions. The gap approximation is given by:

$$b(i) = \log_2(1+ (SNR(i) / \Gamma))$$

Where b(i) is the number of bits that can be supported by the i^{th} subchannel, SNR(i) is the signal to noise ratio of the i^{th} subchannel, and Γ is the SNR gap, which is a function of target bit error rate (10^{-7}), the chosen coding scheme and the desired system performance margin. The bit allocation found during initialization might become sub-optimal due to varying channel conditions and therefore adaptive bit-loading schemes are used, wherein the transmitter receives information from the receiver about channel characteristics and adapts the bit loading scheme accordingly.

The downstream and upstream data channels are synchronized to the 4kHz ADSL DMT symbol rate, and multiplexed into two separate data buffers: fast and interleaved. ADSL uses a superframe structure, which is composed of 68 ADSL data frames. Each user data is assigned to either fast or interleaved buffer during initialization. The binary data stream output from the fast and interleaved buffers are scrambled using the following algorithm:

$$d_n = D_n \oplus d_{n-18} \oplus d_{n-23}$$

Where D_n is the n-th output from the fast or the interleaved buffer and d_n is the n-th output from the corresponding scrambler. Forward error correction is used to ensure optimal performance. It is based on Reed Solomon coding and must be implemented.

Serial input data, which needs to be transmitted over several channels along with additional control and operational information, at a total rate of B_{total} /T, are then grouped into blocks of B_{total} bits.

These B_{total} bits are appropriately encoded and converted to parallel form and modulated by N different carriers. Bits allocated to each subchannel are Quadrature Amplitude Modulated (QAM). Constellation encoder can be implemented with or without Trellis coding. A DMT time domain signal has a high peak-to-average ratio because its amplitude distribution is almost Gaussian and large values may be clipped by the D/A converter. The error signal caused by clipping can be considered as an additive negative impulse for the time sample that was clipped. The clipping error power is almost equally distributed across all tones in which clipping occurs. Clipping is therefore more likely to cause errors on those tones that have been assigned the largest number of bits. These errors can be corrected by forward error correction coding if the tones with the largest number of bits have been assigned to the interleave buffer.

The discrete multitone modulation is accomplished by a 2N-point IFFT operation (N=256 for an ANSI compliant ADSL system). As a direct consequence of finite IFFT size, inter-block interference sets in and the assumption of independent channels no longer remains valid. In a finite length DMT system, the effective length of the channel impulse response, or the channel constraint length 'V' will cause inter-block interference, as the tail of the previous block multi-carrier symbol will corrupt the beginning of the current block multi-carrier symbol (the sub-channels are, therefore, not strictly independent). To counter the effects of the above mentioned problem, cyclic prefix is applied to the modulated block. The length of cyclic prefix will be the same as 'V'. The cyclic prefix procedure is a simple wrapping around of the current block symbol so that the discrete time equivalent data sequence appears periodic.

The parallel output with cyclic prefix is passed through a Digital to Analog converter that operates at a sampling rate of (2N+V)/T .The resulting analog waveform is low pass filtered to produce an analog transmit line signal.

Further, echo-cancellation comprising of echo-emulation, cancellation and an adaptive update of echo-model parameter is used to improve the system performance.

Scope of the project

Further work would involve a detailed study of ADSL and developing a model of an ADSL transmitter using HP EESof. A subset of G.Lite standard would be implemented. The different comprising blocks of an ADSL transmitter that are described in this document would be modeled as dataflow graphs. We would try to develop an environment for simulating an ADSL transmitter and evaluating the performance of the same in different channel conditions. Project would also involve integration of efforts between teams working on initialization, transmitter, channel and receiver modeling to evolve a complete ADSL system.

References:

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   Notes by Prof. Brian L.Evans based on guest lectures on ADSL by Dr. Sayfe Kiaei (Motorola) and Mr.Guner Arslan (UT Austin).