Modeling and Simulation of Discretized Data Transmission in Very High-Speed Digital Subscriber Line

Dogu Arifler, Ming Ding, and Zukang Shen

Abstract

Very High-Speed Digital Subscriber Line (VDSL) technology enables very high bit rate applications over ordinary phone lines. In this paper, we present a survey of the recent research and standardization proposals for VDSL modem transceivers. We propose to build a dataflow model that will enable us to simulate and optimize VDSL transceiver designs.

1. INTRODUCTION

Very High-Speed Digital Subscriber Line (VDSL) technology [1] is an emerging solution for the last mile of connection to a residence or small office. VDSL technology enables very high bit rate applications, such as teleconferencing and telecommuting, over ordinary phone lines. Currently, proposed speeds for VDSL are 22/3, 13/3 Mbps for asymmetric (downstream/upstream) operations, and 13, 9, 6 Mbps for symmetric operations [2].

The VDSL technical specification [3, 4] proposes the use of discrete multitone (DMT) as the modulation method in VDSL systems. DMT divides the bandlimited communication channel into a large number of orthogonal, narrowband carriers, or *tones*. The use of DMT in VDSL has important advantages including the ability to maximize the transmitted bit rate and to adapt to changing line conditions.

In this literature survey, we describe the recent research and identify open issues in designing DMT based VDSL modem transceivers. We briefly discuss VDSL channel modeling, DMT, modem initialization, and computational complexity of the transceiver. Designing VDSL modems is inherently complex. Hence, we propose to build a Synchronous Dataflow (SDF) model for the discretized data transmission in VDSL. Our goal is to provide an abstraction for the transmission system that will enable designers to simulate, evaluate, optimize, and possibly synthesize different transceiver designs.

2. BACKGROUND

2.1. Channel Model

VDSL makes high-speed transmission over conventional twisted pair of wires possible. Twisted pair of wires was originally designed for voice band signal transmission. Co-existence of various kinds of services in the same bandwidth makes the VDSL channel very complicated. Furthermore, crosstalk noise, impulse noise, radio noise, bridged-tap sections, and load coins along the transmission line make the situation even worse.

Crosstalk is caused by electromagnetic radiation. Near-end crosstalk (NEXT) is caused by signals traveling in opposite directions, while far-end crosstalk (FEXT) is caused by signals traveling in the same direction [1]. Both NEXT and FEXT increase with frequency f. Typically, NEXT increases with $f^{1.5}$ and FEXT with f^{2}. Compared to other types of digital subscriber lines, VDSL is more susceptible to FEXT since the loop length of a VDSL system is much shorter. NEXT is even worse than FEXT because it does not experience the loop attenuation while the signal from the other direction does.

Interleaving is reordering transmitted bytes over a block of codewords so that adjacent bytes in a data stream are not from the same codeword. If an interleaver is implemented in the transmitter, most of the impulse noise can generally be corrected. However, recent research indicates that impulse noise events are longer than those that can be corrected by the current interleavers. The work in [5] explores the potential benefits of multi-user detection to mitigate crosstalk in VDSL systems with impulse noise.

Figure 1 illustrates a simple channel model that can be used when modeling VDSL systems.

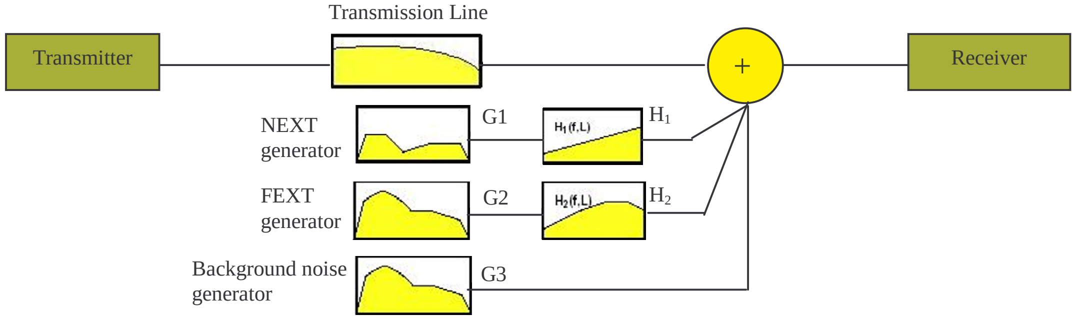


Figure 1. A model for the VDSL channel.

The *linemod* program [6] from Stanford University can be used to model the VDSL transmission line as a finite impulse response (FIR) filter. [3] describes NEXT and FEXT generators, G1, and G2, whose outputs are passed through the crosstalk transfer functions $H_1(f, L)$ and $H_2(f, L)$ that are functions of frequency f and line length L. The models for crosstalk transfer functions [3] are given as follows:

$$H_1(f,L) = K_{xn} \times (f / f_0)^{0.75} \times \sqrt{1 - |S_T(f,L)|^4} \tag{1}$$

$$H_2(f,L) = K_{xf} \times (f / f_0) \times \sqrt{(L / L_0)} \times |S_T(f,L)| \tag{2}$$

where $K_{xn} \cong 0.0032$ and $K_{xf} \cong 0.0056$ (empirically obtained constants that scale the NEXT and FEXT transfer functions respectively), $f_0 = 1$ MHz and $L_0 = 1$ km (chosen reference frequency and length respectively), and $S_T(f, L) = |s_{21}|$ is the test loop transfer function. G3 generates the background noise in the channel.

2.2. Discrete Multitone Modulation

DMT is a form of multi-carrier modulation. DMT divides the bandlimited transmission channel into a set of frequency-indexed sub-channels that are modulated and demodulated independently. The main motivation for DMT is that, if the bandwidths of sub-channels are sufficiently narrow, then no inter-symbol interference (ISI) occurs on any sub-channel. A significant amount of research has shown that such a system is capable of transmitting at the highest performance with proper allocation of bits and energies to these sub-channels [7, 8]. A proper allocation of bits is one in which sub-channels with higher signal-to-noise ratios (SNR) carry proportionately more bits.

In DMT based systems, a bit stream is encoded into a set of Quadrature Amplitude Modulation (QAM) symbols. Each QAM symbol is carried on one sub-channel. The number of bits that can be transmitted on the *n*th sub-channel with an estimated SNR_n by a two-dimensional QAM symbol is approximately given by

$$b_n = \log_2(1 + \frac{SNR_n}{\Gamma}) \tag{3}$$

where Γ = 9.8 dB + γ_m - γ_c for a bit error rate of 10^{-7} [8]. The quantity γ_c is the coding gain of the applied code, and γ_m is the margin, an amount of extra performance that is required to ensure adequate performance in the presence of unforeseen channel impairments. In general, b_n is not an integer. It can either be truncated to an integer, or, in the coded case, be rounded to the nearest fraction that can be implemented by multidimensional Trellis codes. Bit loading algorithms [8] are used to find the optimum assignment of bits to each sub-channel. In VDSL, b_n values range from 0 to 15 [4].

DMT modulation and demodulation are generally implemented using Inverse Fast Fourier Transform (IFFT) and Fast Fourier Transform (FFT). In this paper, we use the terms FFT/IFFT and Discrete Fourier Transform (DFT)/Inverse DFT (IDFT) interchangeably. The IFFT of an N-dimensional sequence $\{X_i\}$ is given by

$$x_n = \frac{1}{\sqrt{N}} \sum_{i=0}^{N-1} X_i e^{j\frac{2\pi}{N}in} \tag{4}$$

Denoting the input QAM symbols, or *sub-symbols*, by X_i for $i = 1, \ldots, M$, where M is the number of sub-channels and $N = 2M$, X_i is given by

$$X_i = \begin{cases} X_i & i = 1, \ldots, M-1 \\ \mathrm{Re}(X_M) & i = 0 \\ \mathrm{Im}(X_M) & i = M \\ X_{N-i}^{*} & i = M+1, \ldots, N-1 \end{cases} \tag{5}$$

imposing the conjugate symmetry conditions on X_i to make $\{x_n\}$ a real sequence.

The FFT of an N-dimensional sequence $\{x_n\}$ is given by

$$X_i = \frac{1}{\sqrt{N}} \sum_{n=0}^{N-1} x_n e^{-j\frac{2\pi}{N}in} \tag{6}$$

Reduced complexity FFT/IFFT implementations for DMT based VDSL modems on digital signal processors are always topics for further research [9].

2.3. Initialization and Channel Identification

The DMT transmitter needs an estimate of the channel's impulse response and crosstalk noise spectrum before the data transmission begins. The most important steps in the identification and initialization process between the transmitter and the receiver, as described by [4] are: (1) Definition of a common mode of operation, and clock and symbol synchronization, (2) Channel and noise identification, (3) Calculation of bit and energy allocations at each sub-channel based on channel and noise identification, and (4) Exchange of parameters, such as *energy* and *bit allocation tables*.

3. MODELING AND SIMULATION

A VDSL modem transceiver at the Physical Medium Dependent (PMD) layer is illustrated in Figure 2.

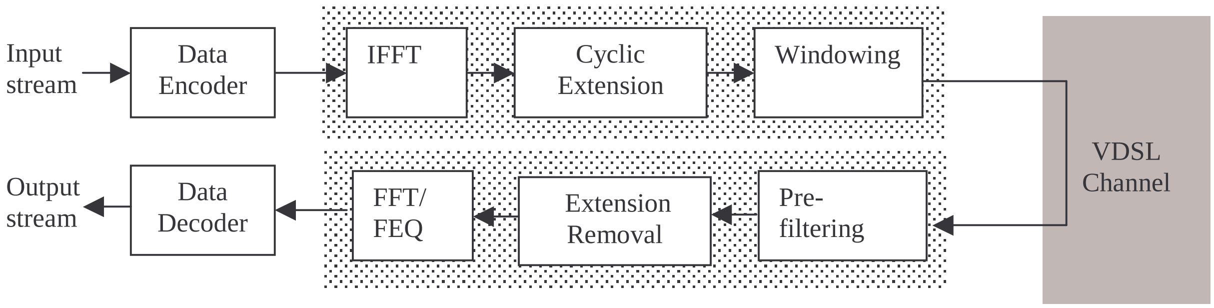


Figure 2. A VDSL modem transceiver. Shaded blocks are the “enhanced” modulation (top) and demodulation (bottom) subsystems.

3.1. VDSL Transmitter

In DMT based VDSL systems, the main subsystems at the transmitter are the data encoder and the enhanced DMT modulator. The former is a QAM-like two-dimensional encoder, which supports up to 2^{15}-point QAM constellations. The encoder transforms the input bit stream from Physical Medium Specific Transmission Convergence (PMS-TC) sub-layer to complex, frequency-domain signal points for each of the $M = N/2$ tones.

The enhanced DMT modulator performs three functions: First, it performs an N-point IFFT on the complex, frequency-domain signal points $\{X_i\}$ (as described in section 2.2) to produce N real, time-domain signal points $\{x_n\}$. Second, the last L_{CP} samples of the IFFT output $\{x_n\}$ are prepended to the time-domain samples as the cyclic prefix (CP). The first L_{CS} samples of $\{x_n\}$ are appended to the block of time-domain samples as the cyclic suffix (CS). When the CP length is equal to or greater than the length of the impulse response of the channel, transmission sub-channels are free from ISI. The CS is added to align the downstream and upstream symbols at both ends of transmission. Finally, a non-rectangular window is applied to the first and last β samples ($\beta < L_{CP}$ and

$\beta < L_{CS}$) on each side of the cyclically extended block. The windowing idea is borrowed from wireless multi-carrier systems, e.g., Orthogonal Frequency Division Multiplexing systems. Windowing smoothes out the transition between successive DMT symbols resulting in lower sidelobes in the transmit spectrum and better spectral confinement [1].

3.2. VDSL Receiver

The main subsystems at the receiver are the enhanced demodulator and the data decoder. Pre-filtering, or time-domain equalization (TEQ) may be implemented if the channel duration is still too long compared to the length of cyclic extension [1]. Another possible use of TEQ is to suppress the radio frequency interference (RFI) from amateur radio bands. A short length TEQ is generally implemented to suppress ISI and RFI. More specifically, for a VDSL system with sampling frequency at 4.4 MHz, 512 sub-carriers and CP length at 80, a 5-tap TEQ may be implemented to shorten the channel impulse response to the CP limit [10, 11].

Windowing in combination with "folding" can also be performed within the cyclic extension range to reduce RFI and inter-carrier interference. Frequency-domain equalization (FEQ) can be implemented to compensate for magnitude and phase distortion from the channel and TEQ [7].

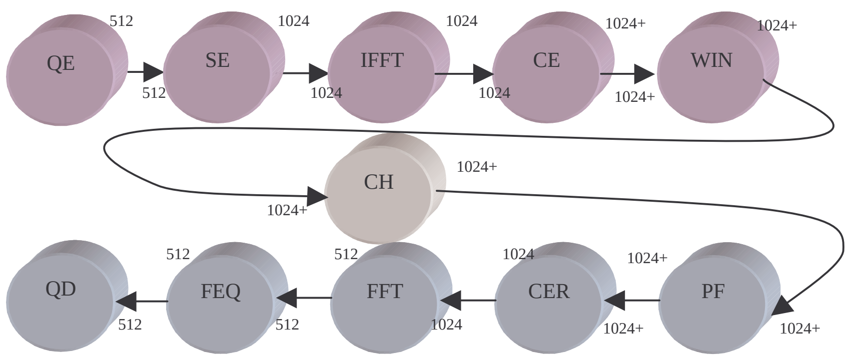
3.3. Computational Complexity

The functions that contribute the most to the processing during data transmission are the data encoder and decoder, and IFFT/FFT blocks for modulation/demodulation of bit streams. In this paper, we consider only the complexity of IFFT/FFT blocks, which are usually implemented in software, and ignore the complexity of QAM encoding and decoding, which can be implemented efficiently by field-programmable gate arrays. It is well known that for an N-point FFT (or N-point IFFT), when N is greater than 128, the computational complexity is $O(1.5N \log_2 N)$ where N is a power of 2.

4. PROPOSED WORK

Our goal in this project is to design low-memory, low-complexity VDSL transceivers that can be implemented on embedded systems.

We will use SDF to model the VDSL modem transceiver blocks as illustrated in Figure 3. SDF is very suitable for modeling signal processing and communication systems in which a stream of tokens is processed by signal processing functional blocks.



QE: Bank of QAM encoders	SE: Conjugate symmetric extension	CE: Cyclic extension
WIN: Windowing	CH: Channel	PF: Pre-filtering
CER: Cyclic extension removal	FEQ: Frequency-domain equalization	QD: Bank of QAM decoders

Figure 3. Top-level SDF model for the VDSL transceiver (M=512,   =CP+CS=80).

In our SDF model, the inputs and outputs of blocks will be sub-symbols. The model conforms to the requirements of SDF that the number of tokens or sub-symbols for each input and output channel must be predetermined.

Each QAM encoder/decoder in the bank of QAM encoders/decoders performs a bit allocation table lookup to determine the number of bits to encode/decode. The total number of bits encoded/decoded by the encoder/decoder bank is fixed.

References

[1] J. M. Cioffi, V. Oksman, J. Werner, T. Pollet, P. M. P. Spruyt, J. S. Chow, and K. S. Jacobsen, "Very-High-Speed Digital Subscriber Lines," *IEEE Communications Magazine*, pp. 72-79, April 1999.

[2] V. Oksman, http://www.ieee802.org/3/efm/public/jul01/presentations/oksman_1_0701.pdf.

[3] "Transmission and Multiplexing (TM); Access transmission systems on metallic access cables; Very high speed Digital Subscriber Line (VDSL); Part 1: Functional requirements," *ETSI TS 101 270-1 V1.2.1* (1999-10), Technical Specification.

[4] "Transmission and Multiplexing (TM); Access transmission systems on metallic access cables; Very high speed Digital Subscriber Line (VDSL); Part 2: Transceiver specification," *ETSI TS 101 270-2 V1.1.1 (2001-02)*, Technical Specification.

[5] H. Dai and H. V. Poor, "Crosstalk mitigation in DMT VDSL with impulse noise," *IEEE Transactions on Circuits and Systems I: Fundamental Theory and Applications*, Volume 48, Issue 10, pp. 1205-1213, Oct. 2001.

[6] *Linemod*, Stanford University, http://www.stanford.edu/class/ee379a.

[7] J. S. Chow, J. C. Tu, and J. M. Cioffi, "A discrete multitone transceiver system for HDSL applications," *IEEE Journal on Selected Areas in Communications*, Volume 9, Issue 6, pp. 895 –908, Aug. 1991.

[8] P. S. Chow, J. M. Cioffi, J.M., and J. A. C. Bingham, "A practical discrete multitone transceiver loading algorithm for data transmission over spectrally shaped channels," *IEEE Transactions on Communications*, Volume 43, Issue 2, Part: 3, pp. 773 –775, Feb.-March-April 1995.

[9] A. Lakhzouri and M. Renfors, "Signal processor implementation of DMT based VDSL modems," *IEEE International Conference on Acoustics, Speech, and Signal Processing, Proceedings*, Volume 4, pp. 2349 –2352, 2001.

[10] G. Arslan, B. L. Evans, and S. Kiaei, "Equalization for Discrete Multitone Receivers To Maximize Bit Rate," *IEEE Transactions on Signal Processing*, Vol. 49, No. 12, pp. 3123-3135, Dec. 2001.

[11] N. Al-Dhahir and J. M. Cioffi, "Optimum Finite-Length Equalization for Multicarrier Transceivers," *IEEE Transactions on Communications*, Vol. 44, No. 1, Jan. 1996.