Chapter 1

Introduction

The advancement in multimedia applications and the development of the Internet have created a demand for high-speed digital communications. Sophisticated audio and video coding methods have reduced the bit rate requirements for audio and video transmission. This in turn motivated the development of communication systems to achieve these requirements. Both technologies enabled high-quality audio and video transmission and introduced a number of new applications for businesses and residential consumers.

Key applications other than voice communications include Internet accessΓ streaming audioΓ and broadcast video. Table 1.1 and 1.2 list several residential and business applications and their data rate requirements. Downstream (from the service provider to the consumer) and upstream (from the consumer to the service provider) requirements are listed in separate columns because some applications have asymmetric requirements. For exampleΓ video broadcasting is an asymmetric application requiring a fast downstream link but no upstream link. Table 1.1 shows that the residential consumer application requirements can be satisfied with a data rate of 3 Mb/s with the exception

Application	downstream data rate (kb/s)	upstream data rate (kb/s)
Voice telephony	16 – 64	16 – 64
Internet access	14 – 3Γ000	14 – 384
Electronic Mail	9 – 128	9 – 64
High definition TV	12Γ000 –24Γ000	0
Broadcast video	1Γ500 – 6Γ000	0
Music on demand	384 – 3Γ000	9
Videophone	128 – 1Γ500	128 – 1Γ500
Distance Learning	384 – 3Γ000	128 – 3Γ000
Database Access	14 – 384	9
Software download	384 – 3Γ000	9
Shop at home	128 – 1Γ500	9 – 64
Video games	64 – 1Γ500	64 – 1Γ500

Table 1.1: Some residential consumer applications and their upstream and downstream data rate requirements [1].

of video and TV applications. Most business applications also require a data rate of 3 Mb/s; howeverΓapplications such as supercomputing could require a symmetric data rate of up to 45 Mb/s.

The remainder of this chapter is organized as follows. Section 1.1 describes standards for high-speed wireline transceiversΓincluding digital subscriber lines (DSL) and cable modems. Section 1.2 gives a brief introduction to DMT modulation and Section 1.3 gives a brief introduction to the equalization problems in DMT modulation. Section 1.4 lists the acronyms and abbreviations used in the dissertation. Section 1.5 gives the thesis statement and the organization of this dissertation.

Application	downstream data rate (kb/s)	upstream data rate (kb/s)
Voice telephony	16 – 64	16 – 64
Facsimile	9 – 128	9 – 128
Internet	14 – 3Γ000	14 – 384
Intranet	64 – 3Γ000	64 – 1Γ500
Electronic commerce	28 – 384	28 – 384
Home office	128 – 6Γ000	64 – 1Γ500
LAN interconnection	384 – 10Γ000	384 – 10Γ000
Electronic Mail	9 – 128	9 – 64
Videophone	128 – 1Γ500	128 – 1Γ500
Database Access	14 – 384	9
Software download	384 – 3Γ000	9
Supercomputing	6Γ000 – 45Γ000	6Γ000 – 45Γ000
Collaborative design	128 – 1Γ500	128 – 1Γ500

Table 1.2: Some business consumer applications and their upstream and downstream data rate requirements [1].

1.1 High-speed Wireline Transceivers

Many solutions have been developed for high-speed communicationsΓ as shown in Table 1.3. Voiceband modems could be considered to be the first solution. Due to the bandwidth constraint of traditional telephone voice linesΓ the data rate of voiceband modems cannot achieve the requirements of most of the applications listed in Tables 1.1 and 1.2. Integrated Service Digital Network (ISDN) provides higher bitrates than voiceband modems. InitiallyΓ T1/E1 lines offered the only business solution for having higher bitrates than ISDN. HoweverΓ the installation and maintenance of T1/E1 lines are very highΓ which motivated service providers to search for cheaper alternatives. DSL technol-

Standard	Data Rates (kb/s)	Description
V.32 (voiceband modem)	9.6	full-duplex using PSK
V.34 (voiceband modem)	33.6	channel precoding
V.90 (voiceband modem)	56 down 33.6 up	pulse code modulation
ISDN (Integrated service digital network)	144	two 64 kb/s and one 16 kb/s channel
HDSL (High-bit-rate DSL)	1,544 / 2,048	two wire pairs, reach of 12,000 feet
HDSL2 (High-bit-rate DSL)	1,544 / 2,048	one wire pair, reach of 12,000 feet
ADSL (Asymmetric DSL)	1,500 – 8,000 down 16 – 640 up	one wire pair, reach of 18,000 feet requires splitters
RADSL (Rate-adaptive ADSL)	1,500 – 8,000 down 16 – 640 up	adaptive rates, one wire pair, requires splitters
G.lite (splitterless ADSL)	up to 1,500 down up to 512 up	no splitter
VDSL (Very high rate DSL)	13,000 – 52,000 down 1,500 – 6,000 up	one wire pair, reach of 4,500 feet
DOCSIS (Data over cable)	27,000 or 36,000 down 320 – 10,240 up	cable TV infrastructure
IEEE 802.14 (Cable Modem)	27,000 or 36,000 down 320 – 20,480 up	cable TV infrastructure

Table 1.3: High-speed data communication standards.

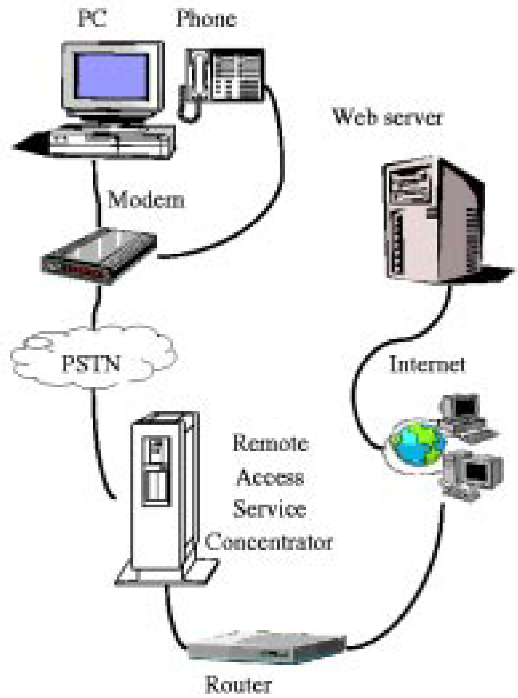

Figure 1.1: Typical dial-up connection via a voiceband modem

ogy offered cheaper alternatives to T1/E1 lines. The first DSL technology was High-bit-rate DSL (HDSL) which was targeted for businesses. The first DSL technology aimed for residential consumers was Asymmetric DSL (ADSL). Initially planned for video-on-demand applicationsΓ ADSL evolved into a key Internet access technology. The emerging Very-high-rate DSL (VDSL) standard is intended to be a bridge between fiber and copper communication technology.

1.1.1 Voiceband transceivers

Voiceband modems were introduced in the 1950s to transmit data through telephone channels. A typical dial-up connection via a voiceband modem is shown in Fig. 1.1. In Fig. 1.1Γa Personal Computer (PC) is connected via the Public Switched Telephone Network (PSTN) to a Remote Access Service (RAS) concentrator which consists of several modems. One of the available modems in the RAS concentrator answers the call from the dial-up modem and directs the call to a router which routes traffic to the desired destination (generally a Web server) over the Internet.

A telephone channel passes only frequencies from about 200 Hz to 3400 Hz. This bandwidth is enough to transmit intelligible voice. A voiceband modem converts the data to be transmitted into a signal which has similar characteristics as a voice signal. Once the data is modulated to fit into the frequency range of the phone channelΓit appears as a voice signal to the channel.

One of the first commercial modems featured a speed of 300 b/s and used frequency shift keying (FSK) modulation. It was introduced by AT&T under the name Bell 103. CCITT (now ITU) standardized V.21 modems for the same data rate. Bell 202 was the first modem to achieve 1Γ200b/s using half duplex FSK modulation. Vadic Inc. introduced the VA3400 modem in 1973 which was the first modem using phase shift keying (PSK) and featured full duplex 1Γ200 b/stransmission [1].

V22.bis doubled the bit rate to 2Γ400b/s in 1981. The V.32 was the first modem to use trellis coding and featured echo cancellationΓwhich enabled transmission of data in both directions at the same time in the same frequency bands. V.32 modems achieved a bit rate of 9Γ600 b/s. Constellation shapingΓbandwidth optimizationΓ andchannel-dependent coding enabled the V.34 modem to achieve 28.8 kb/s which increased to 33.6 kb/s in 1995.

It was believed that 33.6 kb/s was the highest data rate that could possibly be achieved on a telephone voice channel until 56 kb/s modems were introduced in 1996. The V.90 standard for these modems did not appear until 1998. The V.90 modemsΓwhich are also called Pulse Code Modulation (PCM) modemsΓ are asymmetric. They support a downstream bit rate of 56.6 kb/s and an upstream bit rate of 33.6 kb/s. In practiceΓ PCM modems rarely achieve 56.6 kb/s and are generally limited to 50 kb/s [1].

1.1.2 Integrated services digital network (ISDN) transceivers

ISDN was introduced by CCITT in 1976. The first ISDN service in North America was not available until 1986. InitiallyΓ ISDN systems were based on time compression multiplexing (TCM) or alternate mark inversion (AMI). HoweverΓ 2B1Q (2 binaryΓ 1 quaternary) and 4B3T (4 binaryΓ 3 ternary) transmission offered greater loop reach and therefore were adopted in the standard.

Basic rate ISDN supports a bit rate of 160 kb/s symmetric transmission on loops up to 18Γ000feet. This rate is divided into two 64 kb/s B channelsΓ one 16 kb/s D channelΓand one 16 kb/s framing and control channel; henceΓ the transmission data rate is 144 kb/s. Basic rate ISDN uses one of the four available signal levels to represent two bitsΓ hence 2B1Q. It supports full duplex communication using echo cancellation. The bandwidth used is about 80 kHz. Provided that the loops are unloadedΓ basic rate ISDN modems can handle

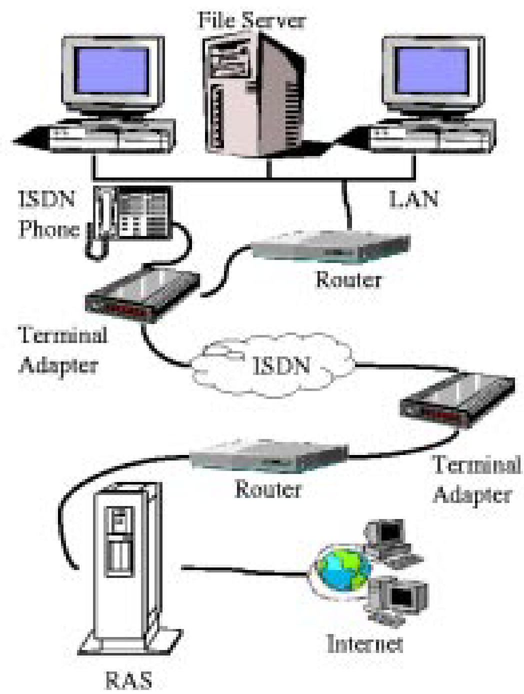

Figure 1.2: Typical ISDN connection

bridge taps by using an adaptive equalizer.

Three methods are used to increase the range of ISDN over 18Γ000 feet – basic rate ISDN transmission extension (BRITE)Γmid-span repeaterΓ and extended-range basic rate ISDN. BRITE uses time division multiplexed channels and digital loop carriers to extend the reach of basic rate ISDN. A mid-span repeater can extend the loop reach by a factor of two. Extended range ISDN uses more advanced digital communication methods to extend the range of basic rate ISDN; e.g.Γtrellis coding enables transmission of 160 kb/s for up to 28Γ000 feetwithout a repeater.

1.1.3 Digital subscriber line transceivers

In the 1960sΓ T1/E1 lines became an efficient way to interconnect two central telephone offices. Voice became digital at a rate of 64 kb/s. In North AmericaΓ 24 voice channels each consisting of 8 bits were grouped to form frames of 193 bits including the framing bit. One T1 line carried these frames at a rate of 8 kHz resulting in a bit rate of 1.544 Mb/s. In EuropeΓ30 voice channels were grouped in addition with two framing and signaling channels which results in 32 64 kb/s channels to be transmitted over one E1 line. HenceΓ the bit rate of an E1 line is 2.048 Mb/s.

In the 1980sΓT1/E1 lines became available to businesses for voice and data transmission. T1/E1 lines required repeaters every 3Γ000to 5Γ000feetΓ and all bridged taps had to be removed for proper use of these lines. ThereforeΓa T1/E1 line was expensive to install (on the order of $10Γ000)and the installation as well as maintenance was time consuming. This motivated the search for transmission techniques that would be easy to installΓwould not need repeaters every 5Γ000feetΓand could support bridge taps. This search lead to the development of the first member of the DSL family: HDSL.

HDSL is based on ISDN. HDSL uses 2B1Q transmission which supported repeaterless transmission up to 12Γ000 feet.HDSL splits T1 transmission rate of 1.544 Mb/s into two 784 kb/s signals which require a signal rate of 392 kHz with 2B1Q transmission. In EuropeΓthe E1 transmission rate of 2.048 Mb/s rate was initially split into two three-wire pairs which later was reduced to two pairs each transmitting 1.168 Mb/s.

The acceptance of HDSL lead to the exploration of whether or not a T1/E1 line could be replaced by only one wire pair instead of two wire pairs as

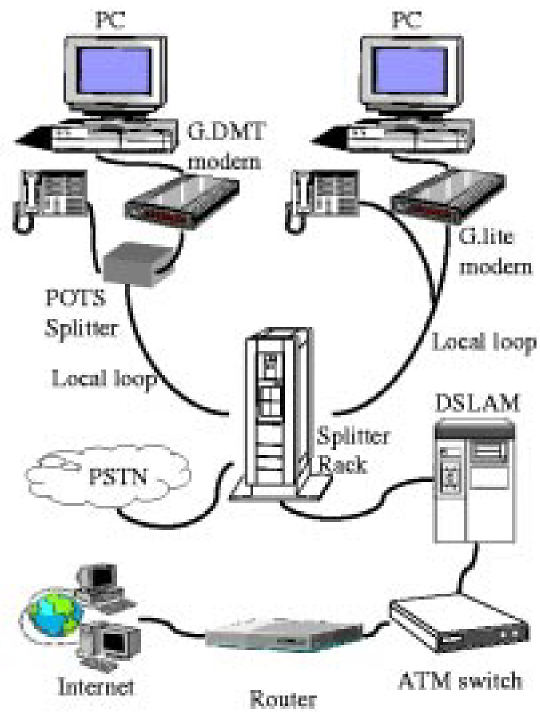

Figure 1.3: ADSL connection with G.DMT and G.lite

in the HDSL standard. This is achieved with the second generation of HDSL or HDSL 2. HDSL 2 uses sophisticated coding and modulation techniques to achieve the required bit rate and reach on a single wire pair.

DSL technology is not limited to businesses and communication between central offices. Consumer applications such as video-on-demand required high throughput in the downstream and smaller throughput in the upstream. ADSL was originally proposed for video-on-demand applications to transmit MPEG-1 video streams. Before the standardizationΓ the bit rate requirements changed due to the development of MPEG-2.

Internet browsing is another example of asymmetric communication.

Unlike video-on-demand applicationsΓ Internet browsing does not have a fixed bit rate requirement that needs to be supported. Rate-adaptive ADSL automatically determines the highest rate it can provide over a given loop. Rate-adaptive ADSL (RADSL) supports downstream rates up to 7 – 10 Mb/s and upstream rates up to 512 – 900 kb/s. To separate the data band from the Plain Old Telephone System (POTS) bandΓ a splitter has to be installed at the consumer side. This expensive process motivated a low rate ADSL standard which does not require a splitter to be installed. Splitterless ADSL or G.lite targets a data rate of 1.5 Mb/s downstream primarily for Internet applications.

ADSL was the first DSL standard to adopt multicarrier modulation. The emerging Very High Bit Rate DSL (VDSL) standards will likely support two line codes – multicarrier modulation and carrierless amplitude/phase (CAP) modulation. Multicarrier modulated VDSL will likely be an extension of ADSL. The goal for VDSL is to achieve up to 52 Mb/s downstream for distances up to 1Γ000 feetand 13 Mb/s downstream for distances up to 3Γ750 feet. VDSL is proposed as a way to connect a consumer to a fiber optical communication network in the consumer's neighborhood.

1.1.4 Cable transceivers

In the 1990sΓ the demand for high-speed communication motivated cable companies to search for ways to use their cable infrastructure for high-speed communications. InitiallyΓ cable TV systems could not support two-way communication. One ad-hoc solution uses POTS voice band modems for the upstream and the cable channel for the downstream. In 1994Γ the IEEE formed the 802.14 Cable TV Media Access Control and Physical Protocol Working Group to form a standard [2]. Their standard was not released until 1997.

Because of the delay of the IEEE standardΓ several vendors formed their own group to decide on a standard for cable modems. They released the Data Over Cable System Interface Specification (DOCSIS) in early 1996. Both the DOCSIS and IEEE standards are similar in the physical layer but have differences in the media access control layer. The IEEE standard is based on Asynchronous Transfer Mode (ATM) whereas the DOCSIS standard supports variable length packets for the delivery of Internet Protocol (IP) traffic. Although better in some aspectsΓ the IEEE ATM approach has a higher implementation cost than DOCSIS. ThereforeΓ nearly all of the major cable modem vendors use the DOCSIS standard [1].

According to the DOCSIS standardΓ the downstream channel uses 64 or 256 Quadrature Amplitude Modulation (QAM) on a carrier of 6 MHz. The data rate is either 27 or 36 Mb/s. The upstream modulation method is Quadrature Phase Shift Keying (QPSK) and 16 QAM on a variable carrier between 200 kHz and 3.2 MHz. Data rate is between 320 kb/s and 10 Mb/s [3]. This data rate is shared by all cable users on the same local area cable network.

1.2 Discrete Multitone Modulation

High-speed communication standards mentioned in the previous section require broadband channels. Inter-symbol interference (ISI) is a major problem associated with broadband channels. This undesirable effect is caused by the spectral shaping of the channel. In other wordsΓ variation of magnitude and phase responses of the channel over frequency causes neighboring symbols to interfere with each other at the receiver. Two approaches to combat ISI are full channel equalization and multicarrier modulation (MCM).

Full channel equalization undoes the spectral shaping effect of a channel using a filter which is called an equalizer. Although linear equalizers are easy to implementΓ they enhance noise and thus degrade the performance of the system. ThereforeΓ more complicated nonlinear equalizersΓ such as the decision feedback equalizerΓ have been proposed. One of the drawbacks of nonlinear equalizers is their computational complexityΓ especially under high sampling rates.

Multicarrier modulation is one possible solution for high-speed digital communications. In contrast to single carrier modulationΓ multicarrier modulationΓ

- avoids full equalization of a channelΓ
- uses available bandwidth efficiently by controlling the power and number of bits in each subchannelΓ
- is robust against impulsive noise and fast fading due to its long symbol durationΓ and
- avoids narrowband distortion by simply disabling one or more subchannels.

Multicarrier modulation has been standardized for G.DMT and G.lite ADSL [4] as well as digital audio/video broadcasting [5Γ6].

In multicarrier modulationΓ the channel is partitioned into a large number of small bandwidth channels called subchannels. If a subchannel were narrow enough so that the channel gain in the subchannel is approximately a complex constantΓ then no ISI would occur in this subchannel. ThusΓ information can be transmitted over these narrowband subchannels without ISIΓ and the total number of bits transmitted is the sum of the number of bits transmitted in each subchannel. If the available power were distributed over the subchannels using the SNR of each subchannelΓ then high spectral efficiency could be achieved. This principle dates back to Shannon's 1948 paper [7] and has been applied in practical systems as early as the late 1950s [8].

Efficiently dividing the channel into hundreds of subchannels became tractable only in the 1990s with the cost vs. performance provided by programmable digital signal processors and the advancement in digital signal processing methods [8]. One of the most efficient ways to partition a channel into large number of narrowband channels is the fast Fourier transform (FFT) [8]. Multicarrier modulation implemented via a FFT is called Discrete Multitone (DMT) modulation or Orthogonal Frequency Division Multiplexing (OFDM). DMT is more common in wireline applicationsΓ whereas OFDM is more common for wireless applications. In transmissionΓ the key difference between the two methods is in the assignment of bits to each subchannel.

1.2.1 DMT Transmitter

A block diagram of a DMT (or OFDM) transceiver is shown in Fig. 1.4. In the transmitterΓ M bits of the input bit stream are buffered. These bits are then assigned to each of the $N/2$ subchannels using a bit loading algorithm [8]. In DMT systemsΓ bit loading algorithms assign the bits and available power to each subchannel according to the SNR in each subchannelΓ such that high SNR subchannels receive more bits than low SNR subchannels. Extremely

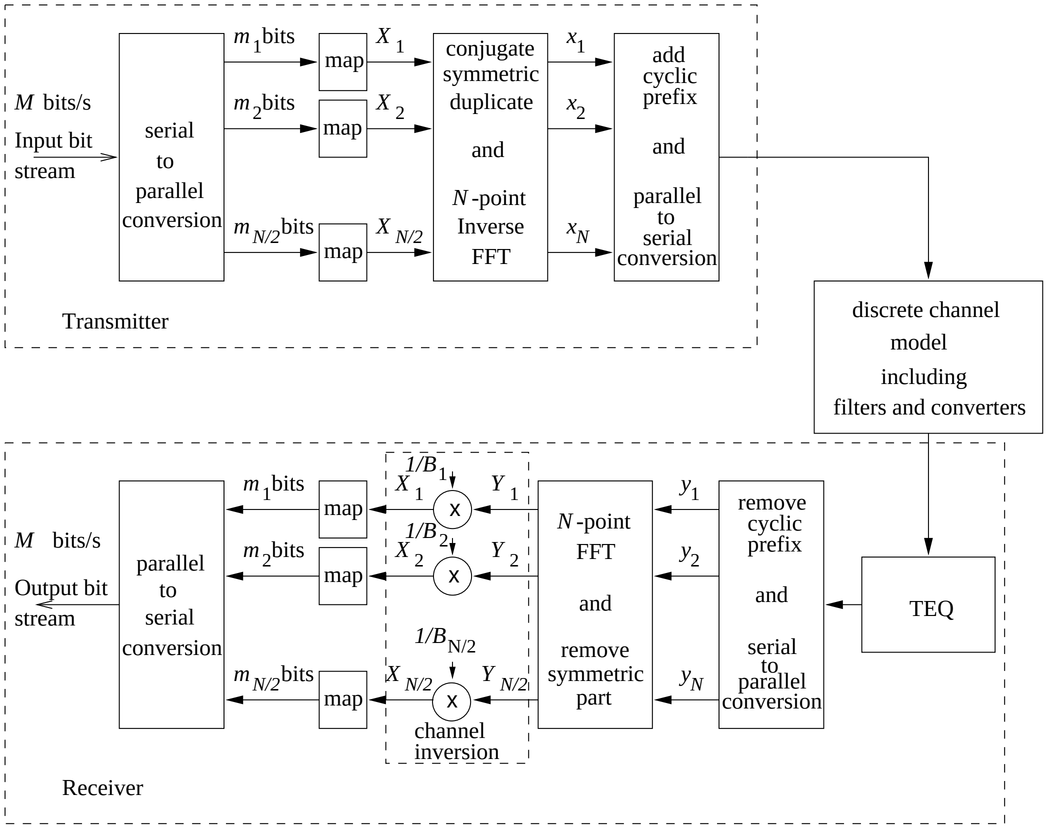

Figure 1.4: Block diagram of a multicarrier modulation system. The transmit filterΓ D/A converterΓchannelΓ A/D converterΓand receive filter are combined into one block.

low SNR subchannels are not used. In OFDM systemsΓthe number of bits in each channel is equal and constant. ThusΓthere is no need for a bit loading algorithm.

The second step is the mapping of the assigned bits to subsymbols using a modulation methodΓsuch as QAM in ADSL modems. These subsymbols are complex-valued in general and can be thought of being in the frequency domain. The efficiency of DMT and OFDM lies in the modulation of the subcarriers. Instead of having $N/2$ independent modulatorsΓthe modulators are implemented with an N-point inverse FFT (IFFT). In order to obtain real samples after IFFTΓthe $N/2$ subsymbols are duplicated with their conjugate symmetric counterparts. The obtained time domain samples are called a DMT symbol.

A guard period between DMT symbols is used to prevent ISI. It is implemented by prepending a symbol with its last v samplesΓ which is called a cyclic prefix. ThusΓ one block consists of $N + v$ samples instead of N samplesΓ which reduces the channel throughput by a factor of $\frac{N+v}{N}$. ISI is completely eliminated for channels with impulse responses of length less than or equal to $v + 1$. The prefix is selected as the last v samples of the symbol in order to convert the linear convolution effect of the channel into circular convolution and help the receiver perform symbol synchronization. Circular convolution can be implemented in the DFT domain by using the FFT. After the FFT in the receiverΓ the subsymbols are the product of the N-point FFT of the channel impulse response and the N-point FFT of the transmitted subsymbols.

1.2.2 DMT Receiver

The receiver is basically the dual of the transmitter with the exception of the addition of time-domain and frequency domain equalizers. The time-domain equalizer ensures that that the equalized channel impulse response is shortened to be less than the length of the cyclic prefix. If the TEQ is successfulΓ then the received complex subsymbols after the FFT are the multiplication of the transmitted subsymbols with the FFT of the shortened (equalized) channel impulse response. The frequency domain equalizer (also called a one-tap equalizer) divides the received subsymbols by the FFT coefficients of the shortened channel impulse response. After mapping the subsymbols back to the corresponding bits using the QAM constellationΓ theyare converted to serial bits.

1.3 Equalization for Discrete Multitone Modulation

With DMTΓthe problem of fully equalizing a channel is converted into partitioning the channel into small subchannels which is more efficient to implement in high-speed transmission. HoweverΓthis does not imply that equalization is not required in an DMT system. The spectra of each inverse FFT (IFFT) modulated subchannel is a sampled sinc function which is not bandlimited. Demodulation is still possible due to the orthogonality between the sinc functions. An ISI causing channelΓhoweverΓdestroys orthogonality between subchannels so that they cannot be separated at the receiver.

One way to prevent ISI is to use a guard period between two successive DMT symbols (one DMT symbol consists of N samples where $N/2+1$ is the number of subchannels). This guard period has to be at least as long as the channel impulse response. Since no new information is transmitted in this guard periodΓ thechannel throughput reduces proportionally to the length of it. If the channel impulse response is relatively long compared to the symbol lengthΓ thenthis performance loss can be prohibitive.

One way to reduce ISI with a shorter cyclic prefix is to use an equalizer. Since the length of a DMT symbol is longer than a symbol in single carrier modulationΓequalization is simpler. AlsoΓnoise enhancement by the equalizer is not an issue because the equalizer does not affect the SNR in each subchannelΓwhich are the primary parameters to determine the performance of a DMT system.

The ADSL standard uses a guard periodΓ time-domain equalizationΓ and frequency-domain equalization. The Time-Domain Equalizer (TEQ) shortens the channel to a length of a predetermined but short guard period. The TEQ can be implemented as an FIR filter whose filter coefficients are trained during initialization. Although this combination has been standardized and is implemented in practical systemsΓon-going research seeks to improve the performance of DMT transceivers. TEQ design is one of the topics promising improvement for DMT transceivers and is the topic of this dissertation.

The major challenge in designing a TEQ is to combine channel capacity optimization into the design procedure. Optimizing channel capacity requires solving a nonlinear optimization problemΓ which raises serious questions about the computational complexity for a real-time solution. A successful TEQ design method mustΓ thereforeΓ not only optimize channel capacity but also must do this with acceptable implementation complexity. The goal of this research is to find such a design method.

1.4 Nomenclature

ADSL	:	Asymmetric Digital Subscriber Lines
AMI	:	Alternate Mark Inversion
ATM	:	Asynchronous Transfer Mode
AWGN	:	Additive White Gaussian Noise
BRITE	:	Basic Rate ISDN Transmission Extension
CAP	:	Carrierless Amplitude/Phase

CCITT	: Comite Consultatif Internationale de Telegraphie et Telephonie
CP	: Cyclic Prefix
CSA	: Carrier Serving Area
DC	: Divide And Conquer
DFT	: Discrete Fourier Transform
DMT	: Discrete Multitone Modulation
DOCSIS	: Data Over Cable Service Interface Specifications
DSL	: Digital Subscriber Line
FEXT	: Far-End Crosstalk
FFT	: Fast Fourier Transform
FIR	: Finite Impulse Response
FSK	: Frequency Shift Keying
GSNR	: Geometric Signal-to-Noise Ratio
GUI	: Graphical User Interface
HDSL	: High-bit-rate Digital Subscriber Line
ICI	: Interchannel Interference
IEEE	: Institute of Electrical and Electronics Engineers
IFFT	: Inverse Fast Fourier Transform
IP	: Internet Protocol
ISDN	: Integrated Service Digital Network
ISI	: Intersymbol Interference
ITU	: International Telecommunication Union
LAN	: Local Area Network
LMS	: Least Mean Squared
LU	: Lower Upper
MAC	: Multiply and Accumulate

MCC	:	Maximum Channel Capacity
MFB	:	Matched Filter Bound
MGSNR	:	Maximum Geometric Signal-to-Noise Ratio
min-ISI	:	Minimum Intersymbol Interference
MIPS	:	Million Instructions per Second
ML	:	Maximum Likelihood
MMSE	:	Minimum Mean Squared Error
MPEG	:	Moving Picture Experts Group
MSE	:	Mean Squared Error
MSSNR	:	Maximum Shortening Signal-to-noise Ratio
NEXT	:	Near-End Crosstalk
OFDM	:	Orthogonal Frequency Division Multiplexing
PCM	:	Pulse Code Modulation
POTS	:	Plain Old Telephone System
PSK	:	Phase Shift Keying
PSTN	:	Public Switched Telephone Network
QAM	:	Quadrature Amplitude Modulation
QPSK	:	Quadrature Phase Shift Keying
RADSL	:	Rate-adaptive Asymmetric Digital Subscriber Line
RAS	:	Remote Access Service
SIR	:	Shortened Impulse Response
SNR	:	Signal-to-noise Ratio
SSNR	:	Shortening Signal-to-noise Ratio
TCM	:	Time Compression Multiplexing
TEQ	:	Time-Domain Equalizer
TIR	:	Target Impulse Response

UEC : Unit-Energy Constraint

UTC : Unit-Tap Constraint

VDSL : Very-high-speed Digital Subscriber Lines

1.5 Thesis statement and organization of the dissertation

In this dissertationΓ Idefend the following thesis statement:

> DMT TEQ design that minimizes frequency weighted ISI power to push ISI into low SNR frequency bands gives equivalent performance to optimal DMT TEQ design that maximizes channel capacity.

Showing this statement to be true would enable the design of optimal TEQs without directly optimizing channel capacity. Channel capacity optimization generally involves nonlinear programming which is not suitable for real-time implementation due to its computational complexity.

This dissertation is organized as follows. Chapter 2 summarizes previous work on TEQ design. It describes MMSE TEQ design which is currently the most commonly used approach in commercial ADSL modems. The MSSNR design method is followed by a suboptimal but computationally efficient alternative to the MSSNR design method called the divide-and-conquer design method. The MGSNR method follows with the multicarrier channel capacity definition.

Chapter 3 proposes a new subchannel SNR definition. I first motivate the definition by an example and then generalize the example to define equivalent signal, noise, and ISI paths in DMT transceivers. These equivalent paths allow me to write SNR in a subchannel as a function of signal, noise, and ISI power.

Chapter 4 proposes the optimal maximum channel capacity (MCC) TEQ design method. I derive an objective function for channel capacity that is a nonlinear function of TEQ taps based on the SNR definition in Chapter 3. This objective function defines the nonlinear optimization problem for optimizing TEQ design in terms of maximizing channel capacity. The MCC TEQ can be calculated by a unconstrained nonlinear optimization method.

Chapter 5 proposes the near-optimal minimum-ISI (min-ISI) TEQ design method. It is based on the thesis that minimizing the total ISI power maximizes channel capacity. I show that this method generalizes the MSSNR method by adding a frequency domain weighting of the ISI power. Fast iterative and recursive algorithms are presented to reduce the computation complexity of the min-ISI method.

Chapter 6 details the simulation environment and its parameters used in the comparative performance analysis of all of the TEQ design methods. Simulation results show that the min-ISI method gives equivalent performance to the optimal MCC method as claimed in the thesis statement.

Chapter 7 summarizes this dissertation and points out possible areas for further research. Appendix A presents details of the MATLAB DMTTEQ toolbox for TEQ design which we developed during the course of this research.