NEAR EAST UNIVERSITY

Faculty of Engineering

Department of Electrical and Electronic Engineering

Asymmetric Digital Subscriber Line (ADSL)

Graduation Project

EE-400

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First of all I would you like to thanks ALLAH for guiding me through my steady.

Actually I want to plead abject to my parent that give me all of supporting until this time. If they don't give me spiritual and their prayers I am not successful this much.

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ABSTRACT

Everyday a new technology is being developed in the world. One of these developments has been Asymmetric Digital Subscriber Line (ADSL). This technology has quickly emerged and became a part of our everyday life.

ADSL provides a network for transmission of data. For file download and upload, for example, ADSL uses different speeds to communicate the data.

This project analyzes all the high speed file transfer technologies related to ADSL such as its wideband version, xdsl, as applications comparable to cable and satellite.

OFDM is a widely used and popular technology which has become a part of ADSL technology. Matlab simulations are performed and it is shown that OFDM improves the performance in fading channels as compared to single carrier communication system.

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INTRODUCTION

Data communications and networking are changing the way we do business and the way we live. Business decision have to be made ever more quickly, and the decision makers require immediate access to accurate information. Why wait a week for that report from germany to arrive by mail when it could appear almost instantaneously through computer networks? Business today rely on computer networks and internetworks. but before we ask how quickly we can get hooked up, we need to know how networks operate , what types of technologies are available , and which design best fills which set of needs.

The development of the personal computer brought about tremendous changes for business, industry, science, and education. A similar revolution is occurring in data communications and networking. Technological advantage are making it possible for communications links to carry more and faster signals. As a result, service are evolving to allow use of the expanded capacity, including the extension to established telephone services such as conference calling, call waiting, voice mail, and caller ID.

Data communications and networking are in their infancy. the goal is to be able to exchange data such as text, audio, and video from any point in the world. We want to access the internet to download and upload information quickly and accurately and at any time.

The first chapter describes the elements of a data communication system .

Chapter two describes asymmetric digital subscriber line (ADSL) which is a new modem technology that converts existing twisted-pair telephone lines into access paths for high-speed communications of various sorts.

Chapter three gives the details of Orthogonal Frequency-Division Multiplexing (OFDM) .

Finally, chapter four includes the results obtained through simulation. The MATLAB code is included in the appendix section.

OVERWIEW OF DATA COMMUNICATIONS

Data communication and networking are topics that have moved from the technical world to the public realm. Products such as mp3 players and cellular phone are no longer the manifestation of high tech wizardry, But gadgets are toted by everyone from preteens to grandparents. Progress in data communications and networking technologies is proceeding at a rapid rate. Bunny –ear antennas on the television have gone the way of the dinosaurs, phased out by digital cable and satellite dishes. The home office is moving toward wireless connection as well. The end user of such technologies is only required know how to use the systems. A students in this field however, must be familiar with the issues and concepts as shows (Table 1.1).

NETWORK MODELS	
DATA COMMUNICATIONS	**NETWORKING**
	INTERNET
	LANs and WANs
COMPONENTS	DISTRIBUTION PROCESSING
DATA REPRESENTATION	CRITERIA
DATA FLOW	STRUCTURE

Table 1.1 Overview of Data Communications and Networking [1]

Data communications

Networks exist so that data may be send from one place to another the basic concept of data communication. To fully grasp this subject, we must understand the physical network component how different types of data can be represented, and how to create a data flow.

Networking

Data communications between remote parties can be archived through a process called networking, involving the connection of computers, media, and networking devices. When we talk about networks, we need to keep in mind three concepts: distributed processing, networking criteria, and network structure.

Local and wide area networks

Networks are divided into two main categories: local are networks (LANs) and wide area networks (WANs). These two types of networks have different characteristic and different functionalities. In general, a LAN is a collection of computers and peripheral devices in a limited area such as a company or department. A WAN, however , is a collection of LANs and spans a large geographical distance.

Internet

The internet, the main of the book is a collection of LANs and WANs held together by internetworking devices. in the figure, we demonstrate this relationship by having the box entitled internet enclose LANs and WANs. The internet is, however, more than just a physical connection of LANs and WANs; internetworking protocols and standard are also needed.

Network models

Network models serve to organize, unify, and control the hardware and software components of data communications and networking. Although the term ''network model'' suggest a relationship to networking, the model also encompasses data communications.

1.1 Data communication

When we communicate, we are sharing information. This sharing can be local or remote. Between individuals, local communication usually occurs face to face. While remote communication takes place over distance. The term telecommunication, which includes telephony, telegraphy and television means communications at a distance.

The word data refers to information presented in whatever form is agreed upon by the parties creating and using the data.

Data communications is the exchange of data between two devices via some form of transmission medium such as a wire cable. For data communications to occur, the communication devices must be part of a communications systems made up of a combination of hardware and software. The effectiveness of a data communications system depends on three fundamental characteristics: delivery, accuracy, and timeliness.

1. Delivery. The system must deliver data to the correct destination. Data must be received by the intended device or user and only by that device or user.
2. Accuracy. The system must deliver the data accurately. Data that have been altered in transmission and left uncorrected are unusable.
3. Timeliness. The system must deliver data in a timely manner. Data delivered late are useless. In the case of video and audio, timely delivery means delivering data as they are produced, in the same order that they are produced and without significant delay. This kind of delivery is called real-time transmission.

1.1.1 Components

A data communication systems has five components:

1. Message. The message is the information (data) to be communicated. It can consist of text, numbers, picture, sound or video – any combination of these.
2. Sender. The sender is the device that send the data message. It can be a computer, workstation, telephone handset, video camera and so on.
3. Receiver. The receiver is the device that receives the message. It can be computer, workstation, telephone handset, television and so on.

4. Medium. The transmission medium is the physical path by which a message travels from sender to receiver. It could be a twisted-pair wire, coaxial cable, fiber optic cable, or radio waves .
5. Protocol. A protocol is a set of rules that governs data communications. It represent an arrangement between the communicating devices. Without a protocol, two devices may be connected but not communicating, just as a person speaking French cannot be understood by a person who speaks only Japanese.

1.1.2 Data representation

Information today comes in different forms such as text , numbers, images, audio and video.

1.1.2.1 Text

In data communications, text is represented as a bit pattern.

A sequence of bits (0s or 1s). The number of bits in a pattern depends on the number of symbols in the language. For example, the English language uses 26 symbols (A,B,C….Z) to represent uppercase letters, 26 symbols (a,b,c……..z) to represented lowercase letters, 10 symbols (0,1,2……..9) to represent numeric characters, and symbols (.,?,:,;……) to represent punctuation. Other symbols such as the blank, the newline, and the tab are used for text alignment and readability.

Different sets of bit patterns have been designed to represent text symbols, each set is called a code , and the process of representing symbols is called coding.

ASCII the American National Standards Institute (ANSI) developed a code called the American standard code for information interchange (ASCII). This code uses 7 bits for each symbol. This means 128 different symbols can be defined by this code.

ISO the International Organizations for standardization known as ISO, has designed a code using a 32 –bit pattern.

1.1.2.2 Numbers

Numbers are also represented by using bit pattern, however, a code such as ASCII is not used to represent numbers ; the number is directly converted to a binary number.

1.1.2.3 Images

Images today are also represented by bit pattern. however, the mechanism is different . in its simpler form, an image is divided into a matrix of pixel, where each pixel is a small dot. The size of the pixel depends on what is called the resolution. For example, an image can be divided into 1000 pixel or 10,000 pixel. In the second case, there is better representation of the image (better resolution), but more memory is needed to store the image.

1.1.2.4 Audio

Audio is a representation of sound. Audio is by nature different from text, numbers, or images. It is continuous, not discrete. Even when we use a microphone to change voice or music to an electric signal, we create a continuous signal.

1.1.2.5 Video

Video can be produced either as a continuous entity (by a camera), or it can be a combination of images, each a discrete entity, arranged to convey the idea of motion. Again we can change video to a digital or an analog signal.

1.1.3 Direction of data flow

Communication between two devices can be simplex, half –duplex, or full – duplex.

1.1.3.1 Simplex

In simplex mode, the communication is unidirectional, as on one way street. Only one of the two devices on a link can transmit; the other can only receive.

Keyboard and traditional monitors are both examples of simplex devices. The keyboard can only introduce input; the monitor can only accept output.

1.1.3.2 Half – duplex

In half-duplex mode, each station can both transmit and receive, but not at the same time. When one devices is sending, the other can only receive and vice versa

The half – duplex mode is like a one –lane road with two-direction traffic. While cars are travelling in one direction, cars going the other way must wait. In a half –duplex transmission, the entire capacity of a channel is taken over by whichever of the two devices is transmitting at the time.

1.1.3.3 Full – duplex

In full- duplex (also called duplex), both can transmit and receive simultaneously

The full-duplex mode is like a two way street with traffic in both directions at the time. In full-duplex mode, signals going in either direction share the capacity of the link. This sharing can occur in two ways: either the link must contain two physically separate transmission paths, one for sending and the other for receiving; or the capacity of the channel is divided between signals travelling in the directions.

One common example of full-duplex communication is the telephone network. When two people are communicating by a telephone line, both can talk and listen at the same time.

1.2 What is data communications?

The distance over which data moves within a computer may vary from a few thousandths of an inch, as is the case within a single IC chip, to as much as several feet along the backplane of the main circuit board.

Over such small distances, digital data may be transmitted as direct, two-level electrical signals over simple copper conductors. Except for the fastest computers, circuit designers are not very concerned about the shape of the conductor.

Frequently, however, data must be sent beyond the local circuitry that constitutes a computer. In many cases, the distances involved may be enormous. Unfortunately, as the distance between the source of a message and its destination increases, accurate transmission becomes increasingly difficult. This results from the electrical distortion of signals traveling through long conductors, and from noise added to the signal as it propagates through a transmission medium.

Although some precautions must be taken for data exchange within a computer, the biggest problems occur when data is transferred to devices outside the computer's circuitry. In this case, distortion and noise can become so severe that information is lost.

Data communications concerns the transmission of digital messages to devices external to the message source. "External" devices are generally thought of as being independently powered circuitry that exists beyond the chassis of a computer or other digital message source. As a rule, the maximum permissible transmission rate of a message is directly proportional to signal power and inversely proportional to channel noise.

It is the aim of any communications system to provide the highest possible transmission rate at the lowest possible power and with the least possible noise.

1.3 Communication Channels

A communications channel is a pathway over which information can be conveyed. It may be defined by a physical wire that connects communicating devices, or by a radio, laser, or other radiated energy source that has no obvious physical presence. Information sent through a communications channel has a source from which the information originates, and a destination to which the information is delivered.

Although information originates from a single source, there may be more than one destination, depending upon how many receive stations are linked to the channel and how much energy the transmitted signal possesses. In a digital communications channel, the information is represented by individual data bits, which may be encapsulated into multi bit message units. A byte, which consists of eight bits, is an example of a message unit that may be conveyed through a digital communications channel.

A collection of bytes may itself be grouped into a frame or other higher-level message unit. Such multiple levels of encapsulation facilitate the handling of messages in a complex data. Any communications channel has a direction associated with it (Fig 1.1).

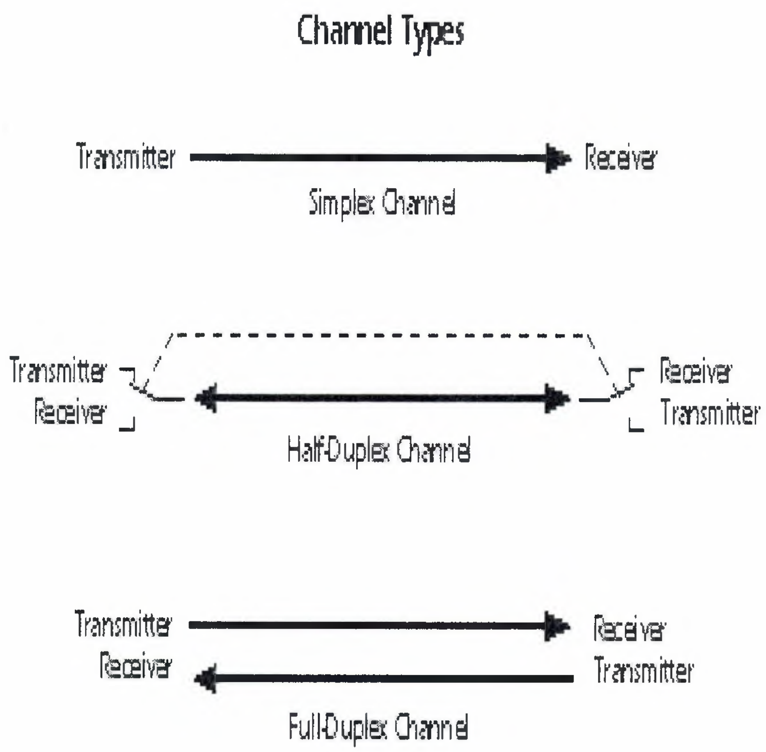

Figure 1.1 Any Communication Channel [2]

The message source is the transmitter, and the destination is the receiver. A channel whose direction of transmission is unchanging is referred to as a simplex channel.

A half-duplex channel is a single physical channel in which the direction may be reversed. Messages may flow in two directions, but never at the same time, in a half-duplex system. In a telephone call, one party speaks while the other listens. After a pause, the other party speaks and the first party listens. Speaking simultaneously results in garbled sound that cannot be understood. A full-duplex channel allows simultaneous message exchange in both directions. It really consists of two simplex channels, a forward channel and a reverse channel, linking the same points.

1.4 Serial Communication

Most digital messages are vastly longer than just a few bits. Because it is neither practical nor economic to transfer all bits of a long message simultaneously, the message is broken into smaller parts and transmitted sequentially. Bit-serial transmission conveys a message one bit at a time through a channel. Each bit represents a part of the message. The individual bits are then reassembled at the destination to compose the message. In general, one channel will pass only one bit at a time. Thus, bit-serial transmission is necessary in data communications if only a single channel is available. Bit-serial transmission is normally just called serial transmission and is the chosen communications method in many computer peripherals. Byte-serial transmission conveys eight bits at a time through eight parallel channels.

Although the raw transfer rate is eight times faster than in bit-serial transmission, eight channels are needed, and the cost may be as much as eight times higher to transmit the message. When distances are short, it may nonetheless be both feasible and economic to use parallel channels in return for high data rates. On the other hand, when communicating with a timesharing system over a modem, only a single channel is available, and bit-serial transmission is required (Figure 1.2).

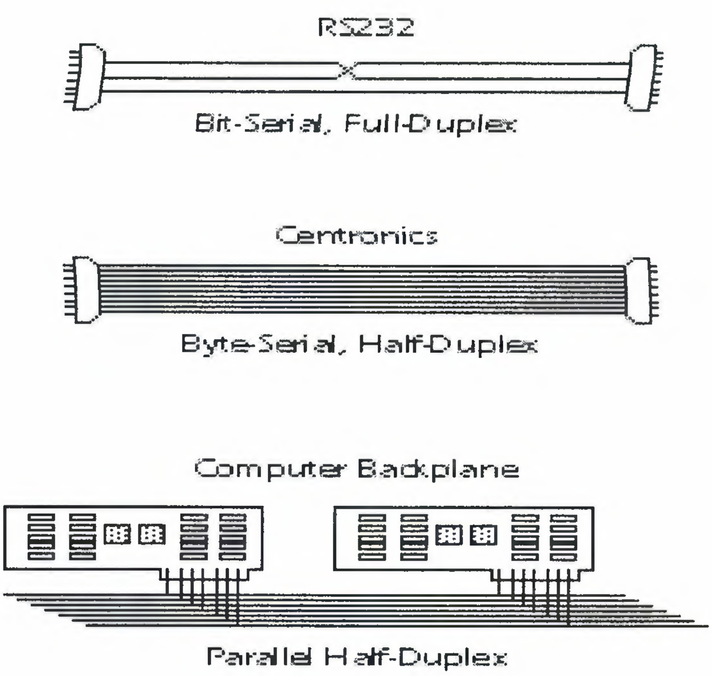

Figure 1.2 Bit serial transmission [2]

The baud rate refers to the signaling rate at which data is sent through a channel and is measured in electrical transitions per second. In the EIA232 serial interface standard, one signal transition, at most, occurs per bit, and the baud rate and bit rate are identical. In this case, a rate of 9600 baud corresponds to a transfer of 9,600 data bits per second with a bit period of 104 microseconds (1/9600 sec.) (Figure1.3). If two electrical transitions were required for each bit, as is the case in non-return-to-zero coding, then at a rate of 9600 baud, only 4800 bits per second could be conveyed. The channel efficiency is the number of bits of useful information passed through the channel per second. It does not include framing, formatting, and error detecting bits that may be added to the information bits before a message is transmitted, and will always be less than one.

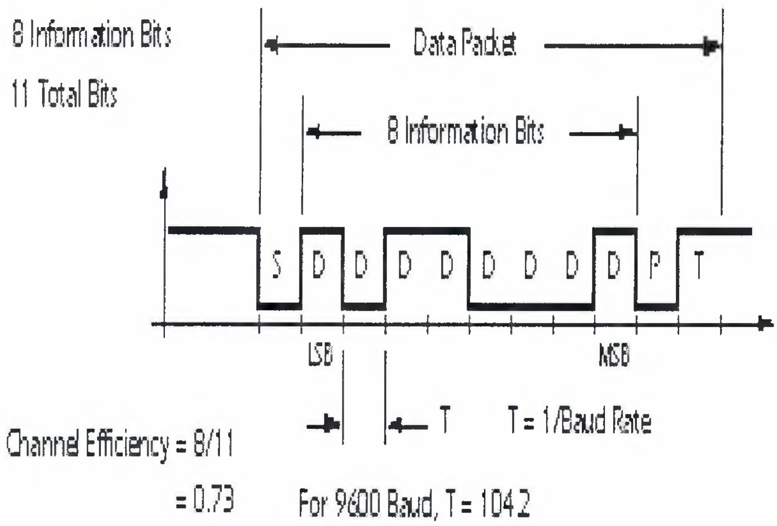

Figure 1.3 Data Rate of the Channel [1]

The data rate of a channel is often specified by its bit rate (often thought erroneously to be the same as baud rate). However, an equivalent measure channel capacity is bandwidth. In general, the maximum data rate a channel can support is directly proportional to the channel's bandwidth and inversely proportional to the channel's noise level. A communications protocol is an agreed-upon convention that defines the order and meaning of bits in a serial transmission. It may also specify a procedure for exchanging messages. A protocol will define how many data bits compose a message unit, the framing and formatting bits, any error-detecting bits that may be added and other information that governs control of the communications hardware. Channel efficiency is determined by the protocol design rather than by digital hardware considerations. Note that there is a tradeoff between channel efficiency and reliability protocols that provide greater immunity to noise by adding error-detecting and correcting codes must necessarily become less efficient.

1.5 Asynchronous vs. Synchronous Transmission

Serialized data is not generally sent at a uniform rate through a channel. Instead, there is usually a burst of regularly spaced binary data bits followed by a pause, after which the data flow resumes. Packets of binary data are sent in this manner, possibly with variable-length pauses between packets, until the message has been fully transmitted. In order for the receiving end to know the proper moment to read individual binary bits from the channel, it must know exactly when a packet begins and how much time elapses between bits. When this timing information is known, the receiver is said to be synchronized with the transmitter, and accurate data transfer becomes possible. Failure to remain synchronized throughout a transmission will cause data to be corrupted or lost. Two basic techniques are employed to ensure correct synchronization. In synchronous systems, separate channels are used to transmit data and timing information. The timing channel transmits clock pulses to the receiver. Upon receipt of a clock pulse, the receiver reads the data channel and latches the bit value found on the channel at that moment. The data channel is not read again until the next clock pulse arrives. Because the transmitter originates both the data and the timing pulses, the receiver will read the data channel only when told to do so by the transmitter (via the clock pulse), and synchronization is guaranteed.

Techniques exist to merge the timing signal with the data so that only a single channel is required. This is especially useful when synchronous transmissions are to be sent through a modem. Two methods in which a data signal is self-timed are non return-to-zero and bi phase Manchester coding.

In asynchronous systems, a separate timing channel is not used. The transmitter and receiver must be preset in advance to an agreed-upon baud rate. A very accurate local oscillator within the receiver will then generate an internal clock signal that is equal to the transmitters within a fraction of a percent. For the most common serial protocol, data is sent in small packets of 10 or 11 bits, eight of which constitute message information. When the channel is idle, the signal voltage corresponds to a continuous logic '1'. A data packet always begins with a logic '0' (the start bit) to signal the receiver that a transmission is starting. The start bit triggers an internal timer in the receiver that generates the needed clock pulses. Following the start bit, eight bits of message data are sent bit by bit at the agreed upon baud rate. The packet is concluded with a parity bit and stop bit. One complete packet is shown in (Figure 1.4)

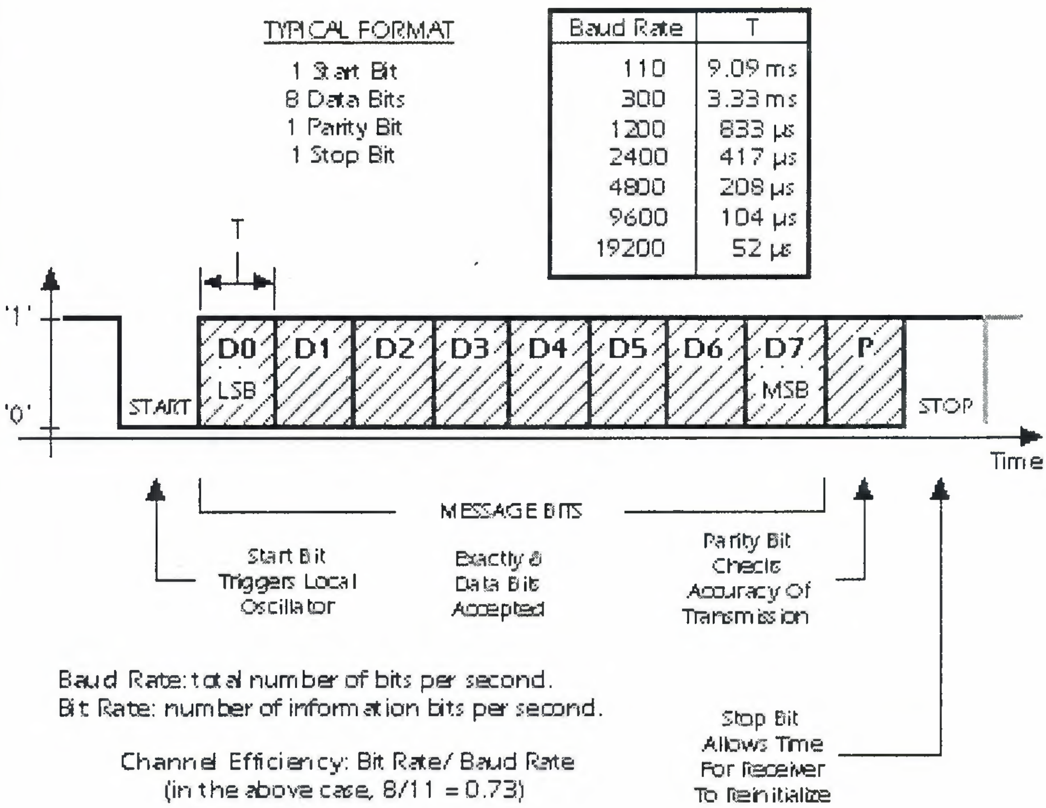

Figure 1.4 One Complete Packet Illustrate [2]

The packet length is short in asynchronous systems to minimize the risk that the local oscillators in the receiver and transmitter will drift apart. When high-quality crystal oscillators are used, synchronization can be guaranteed over an 11-bit period. Every time a new packet is sent, the start bit resets the synchronization, so the pause between packets can be arbitrarily long.

Note that the EIA232 standard defines electrical, timing, and mechanical characteristics of a serial interface. However, it does not include the asynchronous serial protocol shown in the previous figure, or the ASCII alphabet described next.

1.6 The ASCII character set

Characters sent through a serial interface generally follow the ASCII (American Standard Code for Information Interchange) character standard (Figure 1.5).

Nonprintable Control Characters		Special Symbols	Numbers, Special Symbols	Upper-Case Alphabet		Lower-Case Alphabet	
00 NUL	10 DLE	20 SP	30 0	40 @	50 P	60 `	70 p
01 SOH	11 DC1	21 !	31 1	41 A	51 Q	61 a	71 q
02 STX	12 DC2	22 "	32 2	42 B	52 R	62 b	72 r
03 ETX	13 DC3	23 =	33 3	43 C	53 S	63 c	73 s
04 EOT	14 DC4	24 $	34 4	44 D	54 T	64 d	74 t
05 ENG	15 NAK	25 %	35 5	45 E	55 U	65 e	75 u
06 ACK	16 SYN	26 &	36 6	46 F	56 V	66 f	76 v
07 BEL	17 ETB	27 '	37 7	47 G	57 W	67 g	77 w
08 BS	18 CAN	28 (	38 8	48 H	58 X	68 h	78 x
09 HT	19 EM	29)	39 9	49 I	59 Y	69 i	79 y
0A LF	1A SUB	2A *	3A :	4A J	5A Z	6A j	7A z
0B VT	1B ESC	2B +	3B ;	4B K	5B [	6B k	7B {
0C FF	1C FS	2C ,	3C <	4C L	5C \	6C l	7C |
0D CR	1D GS	2D _	3D =	4D M	5D]	6D m	7D }
0E SO	1E RS	2E .	3E >	4E N	5E ^	6E n	7E ~
0F SI	1F VS	2F /	3F ?	4F O	5F #	6F o	7F DEL

Figure 1.5 ASCII [3]

This standard relates binary codes to printable characters and control codes. Fully 25 percent of the ASCII character set represents nonprintable control codes, such as carriage return (CR) and line feed (LF).

Most modern character-oriented peripheral equipment abides by the ASCII standard, and thus may be used interchangeably with different computers.

1.7 Parity and Checksums

Noise and momentary electrical disturbances may cause data to be changed as it passes through a communications channel. If the receiver fails to detect this, the received message will be incorrect, resulting in possibly serious consequences. As a first line of defense against data errors, they must be detected. If an error can be flagged, it might be possible to request that the faulty packet be resent, or to at least prevent the flawed data from being taken as correct. If sufficient redundant information is sent, one- or two-bit errors may be corrected by hardware within the receiver before the corrupted data ever reaches its destination. A parity bit is added to a data packet for the purpose of error detection. In the even-parity convention, the value of the parity bit is chosen so that the total number of '1' digits in the combined data plus parity packet is an even number. Upon receipt of the packet, the parity needed for the data is recomputed by local hardware and compared to the parity bit received with the data. If any bit has changed state, the parity will not match, and an error will have been detected. In fact, if an odd number of bits (not just one) have been altered, the parity will not match. If an even number of bits have been reversed, the parity will match even though an error has occurred. However, a statistical analysis of data communication errors has shown that a single-bit error is much more probable than a multi bit error in the presence of random noise. Thus, parity is a reliable method of error detection.

Even-Parity Computation

Data	Parity Bit
1 0 1 1 0 0 0 1	0
1 0 0 0 0 1 1 0	1

Another approach to error detection involves the computation of a checksum. In this case, the packets that constitute a message are added arithmetically. A checksum number is appended to the packet sequence so that the sum of data plus checksum is zero. When received, the packet sequence may be added, along with the checksum, by a local microprocessor. If the sum is nonzero, an error has occurred. As long as the sum is zero, it is highly unlikely (but not impossible) that any data has been corrupted during transmission.

Checksum Computation

```
             10110001  ┐
             10000110  │
             01001100  ├ Data
             11111111  │
           + 10100000  ┘
          ----------
0011001000010          Arithmetic Sum

             00100010  Sum Truncated to 8 Bits

           + 11011110  Checksum
          ----------
(mod 256)    00000000  Sum plus Checksum Equal Zero
```

Errors may not only be detected, but also corrected if additional code is added to a packet sequence. If the error probability is high or if it is not possible to request retransmission, this may be worth doing. However, including error-correcting code in a transmission lowers channel efficiency, and results in a noticeable drop in channel throughput.

1.8 Data Compression

If a typical message were statistically analyzed, it would be found that certain characters are used much more frequently than others. By analyzing a message before it is transmitted, short binary codes may be assigned to frequently used characters and longer codes to rarely used characters. In doing so, it is possible to reduce the total number of characters sent without altering the information in the message. Appropriate decoding at the receiver will restore the message to its original form. This procedure, known as data compression, may result in a 50 percent or greater savings in the amount of data transmitted. Even though time is necessary to analyze the message before it is transmitted, the savings may be great enough so that the total time for compression, transmission, and decompression will still be lower than it would be when sending an uncompressed message. Some kinds of data will compress much more than others. Data that represents images, for example, will usually compress significantly, perhaps by as much as 80 percent over its original size.

Data representing a computer program, on the other hand, may be reduced only by 15 or 20 percent. A compression method called Huffman coding is frequently used in data communications, and particularly in fax transmission. Clearly, most of the image data for a typical business letter represents white paper, and only about 5 percent of the surface represents black ink. It is possible to send a single code that, for example, represents a consecutive string of 1000 white pixels rather than a separate code for each white pixel.

Consequently, data compression will significantly reduce the total message length for a faxed business letter. Were the letter made up of randomly distributed black ink covering 50 percent of the white paper surface, data compression would hold no advantages.

1.9 Data Encryption

Privacy is a great concern in data communications. Faxed business letters can be intercepted at will through tapped phone lines or intercepted microwave transmissions without the knowledge of the sender or receiver. To increase the security of this and other data communications, including digitized telephone conversations, the binary codes representing data may be scrambled in such a way that unauthorized interception will produce an indecipherable sequence of characters. Authorized receive stations will be equipped with a decoder that enables the message to be restored. The process of scrambling, transmitting, and descrambling is known as encryption. Custom integrated circuits have been designed to perform this task and are available at low cost.

In some cases, they will be incorporated into the main circuitry of a data communications device and function without operator knowledge. In other cases, an external circuit is used so that the device, and its encrypting/decrypting technique.

1.10 Data Storage Technology

Normally, we think of communications science as dealing with the contemporaneous exchange of information between distant parties. However, many of the same techniques employed in data communications are also applied to data storage to ensure that the retrieval of information from a storage medium is accurate.

We find, for example, that similar kinds of error-correcting codes used to protect digital telephone transmissions from noise are also used to guarantee correct read back of digital data from compact audio disks, CD-ROMs, and tape backup systems.

1.11 Data Transfer in Digital Circuits

Data is typically grouped into packets that are either 8, 16, or 32 bits long, and passed between temporary holding units called registers. Data within a register is available in parallel because each bit exits the register on a separate conductor. To transfer data from one register to another, the output conductors of one register are switched onto a channel of parallel wires referred to as a bus. The input conductors of another register, which is also connected to the bus, capture the information (Figure 1.6)

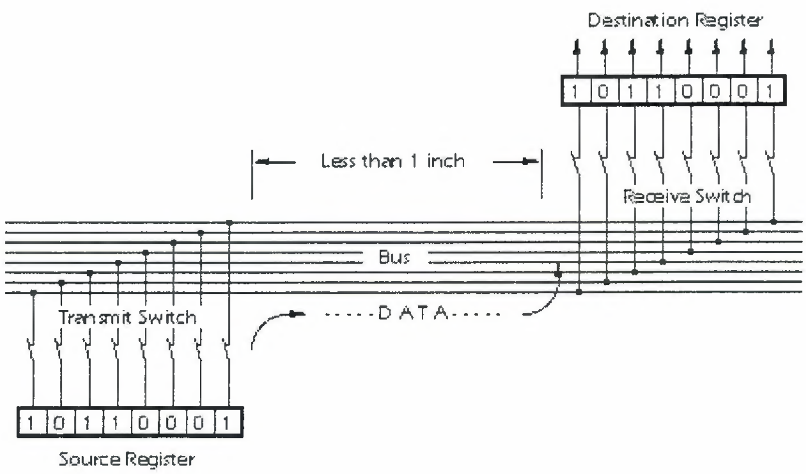

Figure 1.6 Data Transfer [1]

Following a data transaction, the content of the source register is reproduced in the destination register. It is important to note that after any digital data transfer, the source and destination registers are equal; the source register is not erased when the data is sent.

The transmit and receive switches shown above are electronic and operate in response to commands from a central control unit. It is possible that two or more destination registers will be switched on to receive data from a single source. However, only one source may transmit data onto the bus at any time. If multiple sources were to attempt transmission simultaneously, an electrical conflict would occur when bits of opposite value are driven onto a single bus conductor. Such a condition is referred to as a bus contention. Not only will a bus contention result in the loss of information, but it also may damage the electronic circuitry. As long as all registers in a system are linked to one central control unit, bus contentions should never occur if the circuit has been designed properly.

Note that the data buses within a typical microprocessor are fundamentally half-duplex channels.

1.12 Transmission over Short Distance (<2 feet)

When the source and destination registers are part of an integrated circuit (within a microprocessor chip, for example), they are extremely close (thousandths of an inch). Consequently, the bus signals are at very low power levels, may traverse a distance in very little time, and are not very susceptible to external noise and distortion. This is the ideal environment for digital communications. However, it is not yet possible to integrate all the necessary circuitry for a computer (i.e., CPU, memory, disk control, video and display drivers, etc.) on a single chip.

When data is sent off-chip to another integrated circuit, the bus signals must be amplified and conductors extended out of the chip through external pins. Amplifiers may be added to the source register (Figure 1.7)

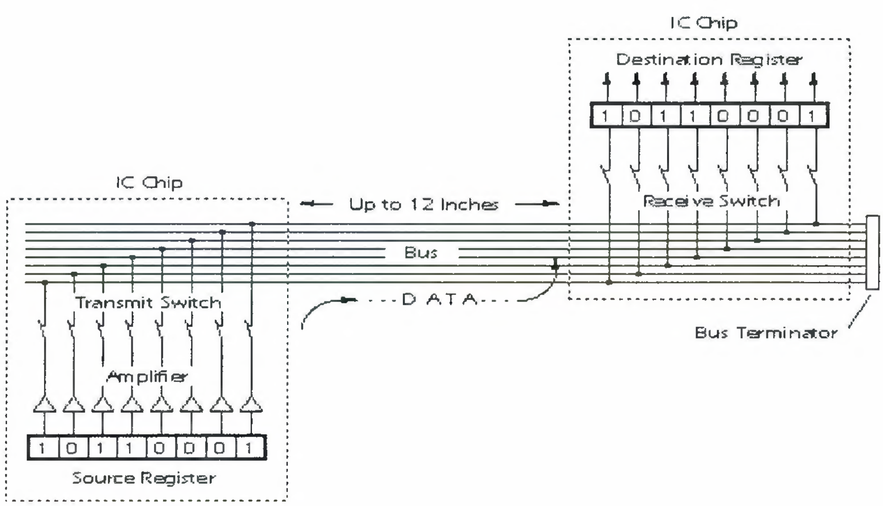

Figure 1.7 Data Transfer With Signals [5]

Bus signals that exit microprocessor chips and other VLSI circuitry are electrically capable of traversing about one foot of conductor on a printed circuit board, or less if many devices are connected to it.

1.13 Noise and Electrical Distortion

Because of the very high switching rate and relatively low signal strength found on data, address, and other buses within a computer, direct extension of the buses beyond the confines of the main circuit board or plug-in boards would pose serious problems.

First, long runs of electrical conductors (figure 1.8), either on printed circuit boards or through cables, act like receiving antennas for electrical noise radiated by motors, switches, and electronic circuits (Figure 1.8).

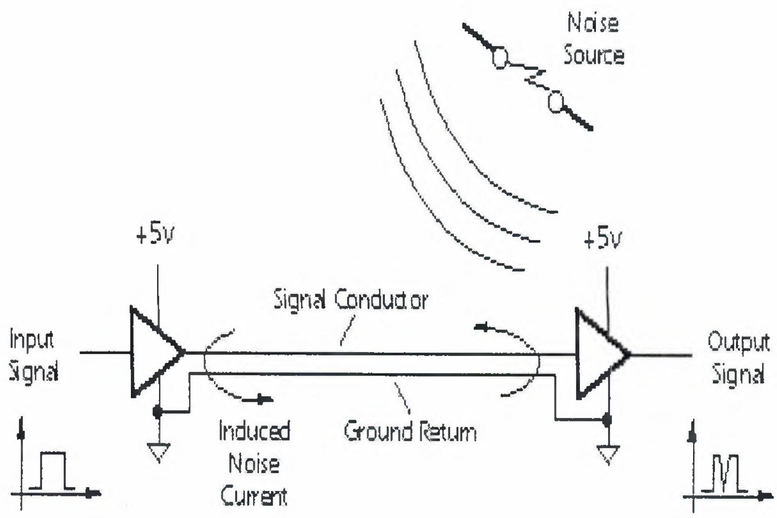

Figure 1.8 Electrical Conductors [5]

Such noise becomes progressively worse as the length increases, and may eventually impose an unacceptable error rate on the bus signals. Just a single bit error in transferring an instruction code from memory to a microprocessor chip may cause an invalid instruction to be introduced into the instruction stream, in turn causing the computer to totally cease operation. A second problem involves the distortion of electrical signals (figure 1.9) as they pass through metallic conductors. Signals that start at the source as clean, rectangular pulses may be received as rounded pulses with ringing at the rising and falling edges (Figure 1.9).

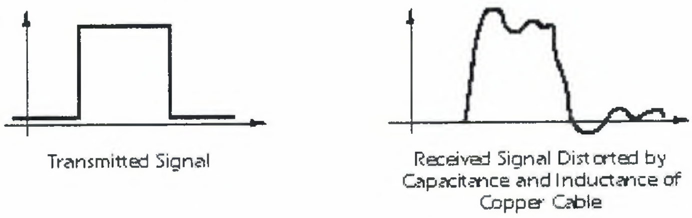

Figure 1.9 Electrical Signals [5]

These effects are properties of transmission through metallic conductors, and become more pronounced as the conductor length increases. To compensate for distortion, signal power must be increased or the transmission rate decreased.

Special amplifier circuits are designed for transmitting direct (un modulated) digital signals through cables. For the relatively short distances between components on a printed circuit board or along a computer backplane, the amplifiers are in simple IC chips that operate from standard +5v power. The normal output voltage from the amplifier for logic '1' is slightly higher than the minimum needed to pass the logic '1' threshold. Correspondingly for logic '0', it is slightly lower. The difference between the actual output voltage and the threshold value is referred to as the noise margin, and represents the amount of noise voltage that can be added to the signal without creating an error (Figure 1.10).

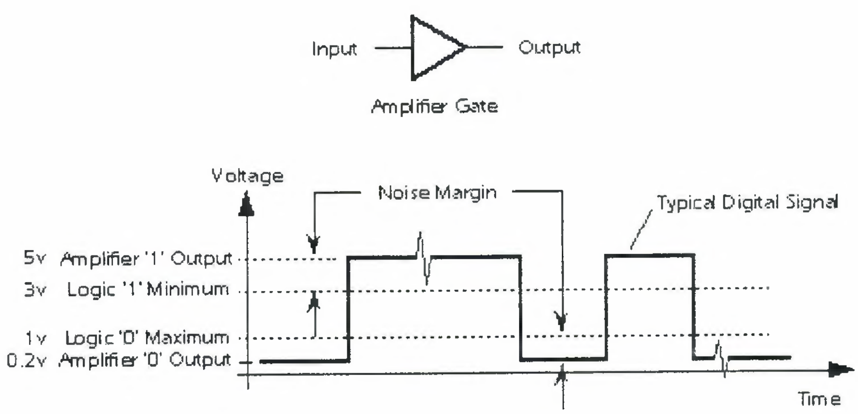

Figure 1.10 Noise Margin [5]

1.14 Transmission over medium distance (< 20 feet)

Computer peripherals such as a printer or scanner generally include mechanisms that cannot be situated within the computer itself. Our first thought might be just to extend the computer's internal buses with a cable of sufficient length to reach the peripheral. However, would expose all bus transactions to external noise and distortion even though only a very small percentage of these transactions concern the distant peripheral to which the bus is connected. If a peripheral can be located within 20 feet of the computer, however, relatively simple electronics may be added to make data transfer through a cable efficient and reliable. To accomplish this, a bus interface circuit is installed in the computer (Figure 1.11).

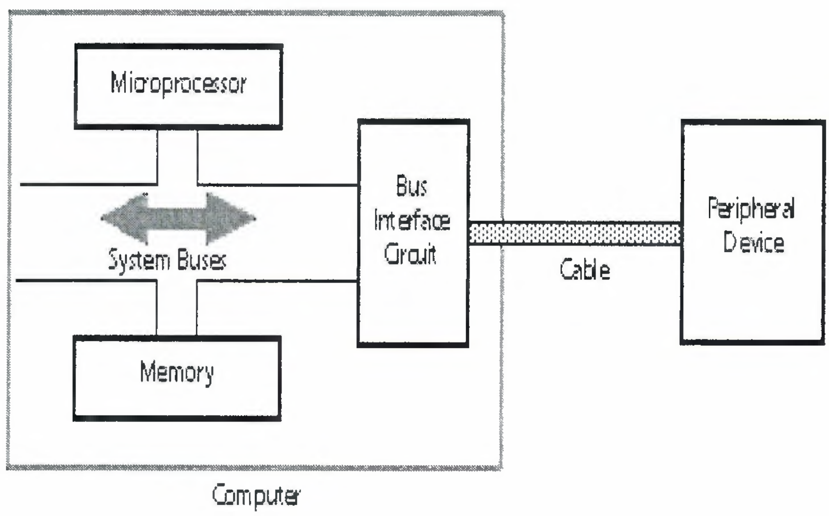

Figure 1.11 Bus Interface Circuit [5]

It consists of a holding register for peripheral data, timing and formatting circuitry for external data transmission, and signal amplifiers to boost the signal sufficiently for transmission through a cable. When communication with the peripheral is necessary, data is first deposited in the holding register by the microprocessor. This data will then be reformatted, sent with error-detecting codes, and transmitted at a relatively slow rate by digital hardware in the bus interface circuit. In addition, the signal power is greatly boosted before transmission through the cable. These steps ensure that the data will not be corrupted by noise or distortion during its passage through the cable.

In either a simple extension cable or a LAN, a balanced electrical system is used for transmitting digital data through the channel. This type of system involves at least two wires per channel, neither of which is a ground. Note that a common ground return cannot be shared by multiple channels in the same cable as would be possible in an unbalanced system. The basic idea behind a balanced circuit is that a digital signal is sent on two wires simultaneously, one wire expressing a positive voltage image of the signal and the other a negative voltage image. When both wires reach the destination, the signals are subtracted by a summing amplifier, producing a signal swing of twice the value found on either incoming line. If the cable is exposed to radiated electrical noise, a small voltage of the same polarity is added to both wires in the cable. When the signals are subtracted by the summing amplifier, the noise cancels and the signal emerges from the cable without noise (Figure 1.13).

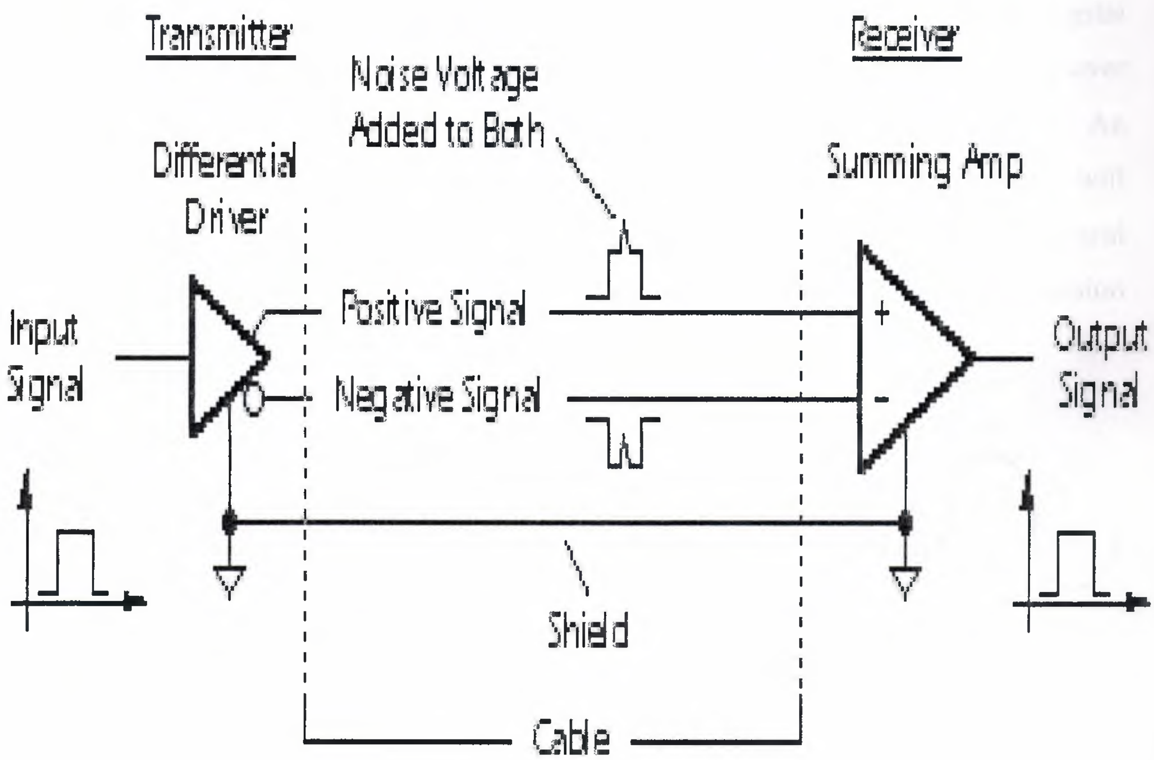

Figure 1.13 The signal emerges from the cable [7]

A great deal of technology has been developed for LAN systems to minimize the amount of cable required and maximize the throughput.

The costs of a LAN have been concentrated in the electrical interface card that would be installed in PCs or peripherals to drive the cable, and in the communications software, not in the cable itself (whose cost has been minimized). Thus, the cost and complexity of a LAN are not particularly affected by the distance between stations.

1.16. Transmission over Very Long Distance (greater than 4000 feet)

Data communications through the telephone network can reach any point in the world. The volume of overseas fax transmissions is increasing constantly, and computer networks that link thousands of businesses, governments, and universities are pervasive. Transmissions over such distances are not generally accomplished with a direct-wire digital link, but rather with digitally-modulated analog carrier signals.

This technique makes it possible to use existing analog telephone voice channels for digital data, although at considerably reduced data rates compared to a direct digital link. Transmission of data from your personal computer to a timesharing service over phone lines requires that data signals be converted to audible tones by a modem. An audio sine wave carrier is used, and, depending on the baud rate and protocol, will encode data by varying the frequency, phase, or amplitude of the carrier. Several modulation techniques typically used in encoding digital data for analog transmission are shown below (Figure 1.14).

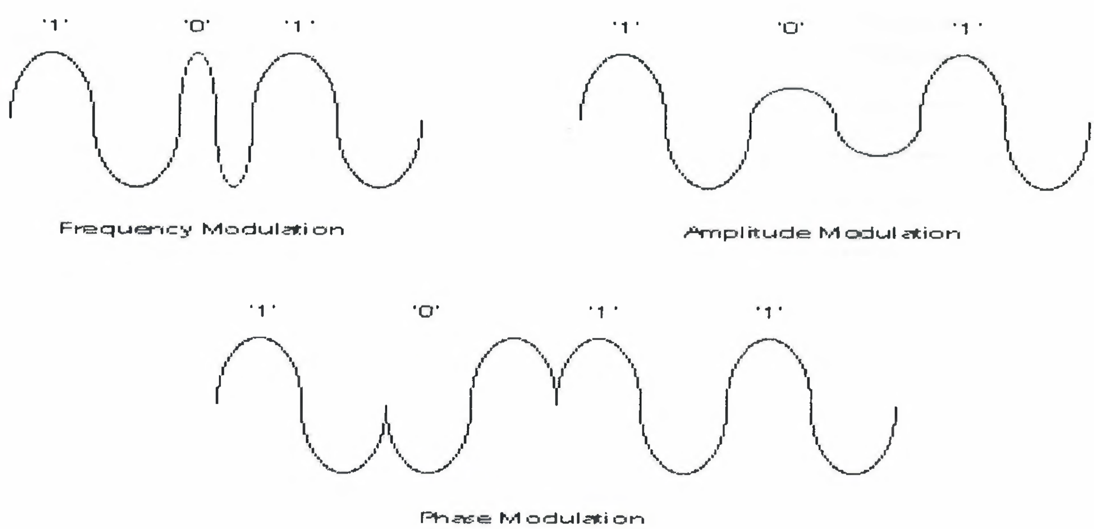

Figure 1.14 Encoding digital data for analog transmission [8]

CHAPTER TWO

Asymmetric Digital Subscriber Line (ADSL)

Definition

Asymmetric digital subscriber line (ADSL) is a new modem technology that converts existing twisted-pair telephone lines into access paths for high-speed communications of various sorts.

Overview

ADSL can transmit more than 6 Mbps to a subscriber enough to provide Internet access, video-on-demand, and LAN access. In interactive mode it can transmit more than 640 kbps in both directions. This increases the existing access capacity by more than fifty-fold enabling the transformation of the existing public network. No longer is it limited to voice, text, and low-resolution graphics. It promises to be nothing less than an ubiquitous system that can provide multimedia (including full-motion video) to the entire country. ADSL can perform as indicated in (*Table* 2.1).

Table 1. ADSL Data Rates As a Function of Wire and Distance

Data Rate (Mbps)	Wire Gauge (AWG)	Distance (ft)	Wire Size (mm)	Distance (km)
1.5-2.0	24	18,000	0.5	5.5
1.5-2.0	26	15,000	0.4	4.6
6.1	24	12,000	0.5	3.7
6.1	26	9,000	0.4	2.7

Table 2.1 ADSL Data Rates As a Function of Wire and Distance

2.1.1. A Short History of Analog Modems

The term modem is actually an acronym which stands for Modulation/demodulation. A modem enables two computers to communicate by using the public switched telephone network. This network can only carry sounds so modems need to translate the computer's digital information into a series of high-pitched sounds which can be transported over the phone lines. When the sounds arrive at their destination, they are demodulated turned back into digital information for the receiving computer

Figure 2.1 Analog Modems [15]

All modems use some form of compression and error correction. Compression algorithims enable throughput to be enhanced two to four times over normal transmission. Error correction examines incoming data for integrity and requests retransmission of a packet when it detects a problem.

2.1.2. The Analog Modem Market

The dynamics of the analog-modem market can be traced back to July 1968 when, in its landmark Carter fone decision, the FCC ruled that "the provisions prohibiting the use of customer-provided interconnecting devices were unreasonable."

January 1, 1969, AT&T revised its tariffs to permit the attachment of customer-provided devices (such as modems) to the public switched network subject to the following three important conditions:

- The customer-provided equipment was restricted to certain output power and energy levels, so as not to interfere with or harm the telephone network in any way.

- The interconnection to the public switched network had to be made through a telephone company-provided protective device, sometimes referred to as a data access arrangement (DAA).

- All network-control signaling such as dialing, busy signals, and so on had to be performed with telephone-company equipment at the interconnection point.

By 1976, the FCC had recommended a plan whereby current protective devices would be phased out in favor of a so-called registration plan. Registration would permit direct switched-network electrical connection of equipment that had been inspected and registered by an independent agency such as the FCC as technically safe for use on the switched network.

In the post-war era, heavy emphasis on information theory led to the profound and now famous 1948 paper by Claude Shannon providing us with a concise understanding of channel capacity for power and band limited gaussian noise channels our analog telephone channel.

$$C = Bw * Log2(1+S/N) \qquad (2.1)$$

This simply states that the channel capacity, *C*, is equal to the available channel bandwidth, *Bw*, times the log base 2 of 1 plus the signal-to-noise ratio in that bandwidth. It does not explain "how" to accomplish this, it simply states that this channel capacity can be approached with suitable techniques.

As customers started buying and using modems, speed and reliability became important issues. Each vendor tried to get as close to the limit expressed by Shannon's Law as they could. Until Recommendation V.32, all modem standards seemed to fall short of this capacity by 9 to 10 db S/N. Estimates of the channel capacity used assumed bandwidths of 2400 Hz to 2800 Hz, and S/N ratios from 24 db to 30 db and generally arrived at a capacity of about 24,000 bits per second (bps). It was clear that error-correction techniques would have to become practical before this gap would be diminished.

Modems of the 1950's were all proprietary primarily FSK (300 bps to 600 bps) and vestigial sideband (1200 bps to 2400 bps). These devices used or were built upon technology from RF radio techniques developed during the wartime era and applied to wire line communications.

International standardization of modems started in the 1960s. In the 1964 Plenary, the first CCITT Modem Recommendation, V.21 (1964), a 200 bps FSK modem (and now 300 bps) was ratified and is (still) used in the V.34/V.8 handshake. The preferred modulation progressed to 4 Phase (or 2X2 QAM) in 1968, and to 4X4 QAM with V.22bis in 1984. Additionally, in 1984, the next major technological advancement in modem recommendations came with V.32 and the addition of echo cancellation and trellis coding. Trellis codes, first identified by Dr. Gottfred Unger boeck, were a major breakthrough in that they made it practical to provide a level of forward error correction to modems, realizing a coding gain of 3.5 db, and closing over a third of the "gap" in realizing the Shannon channel capacity. Recommendation V.32bis built on this and realized improvement in typical-connection S/N ratios and increased the data rates to 14,400 bps.

As work on V.34 started in earnest (1989/90), a recognition of further improvement in the telephone networks in many areas of the world was evident. With this recognition, the initial goal of 19,200 bps moved to 24,000 bps and then to 28,800 bps. The newer V.34 (1996) modem supports 33,600 bps. Such modems achieve 10 bits per Hertz of bandwidth, a figure which approaches the theoretical limits. Recently, a number of companies have introduced a 56.6-kbps analog modem designed to operate over standard phone lines. However, the modem is asymmetrical (it operates at normal modem speeds on the upstream end), it requires a dedicated T1/E1 connection to the ISP site to consistently reach its theoretical limits. For users without such a line the modem offers, inconsistently at best according to reports, a modest gain in performance.

However, the bandwidth limitations of voice band lines are not a function of the subscriber line but the core network. Filters at the edge of the core network limit voice-grade bandwidth to approximately 3.3 kHz. Without such filters, the copper access wires can pass frequencies into the MHz regions. Attenuation determines the data rate over twisted-pair wire, and it, in turn, is a function of line length and frequency. Indicated the practical limits on data rates in one direction compared to line length.

2.1.3 Digital Subscriber Line (DSL)

Despite its name, DSL does not refer to a physical line but to a modem or rather a pair of modems. A DSL modem pair creates a digital subscriber line, but the network does not purchase the lines when it buys ADSL it already owns those it purchases modems.

A DSL modem transmits duplex (i.e., data in both directions simultaneously) at 160 kbps over copper lines of up to 18,000 feet. DSL modems use twisted-pair bandwidth from 0 to approximately 80 kHz which precludes the simultaneous use of analog telephone service in most cases (Figure 2.2).

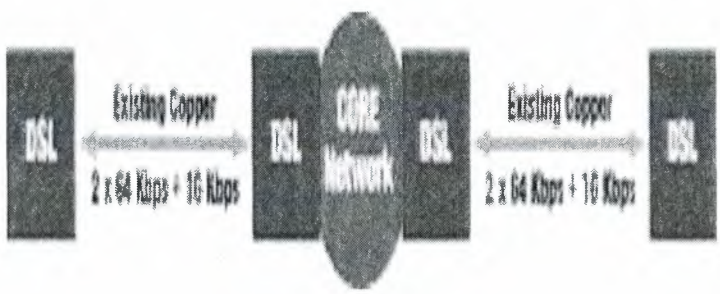

Figure 2.2 Analog Telephone Service [15]

T1 and E1

Engineers created a voice multiplexing system which digitized a voice sample into a 64 kbps data stream (8000 voltages samples per second) and organized these into a 24-element framed data stream with conventions for determining precisely where the 8-bit slots went at the receiving end. The frame was 193 bits long and created an equivalent data rate of 1.544 Mbps. The engineers called their data stream DS1, but it has since come to be known as T1. Technically, though, T1 refers to the raw data rate, with DS1 referring to the framed rate.

In Europe, the world's public telephone networks other than AT&T modified the Bell Lab approach and created E1a multiplexing system for 30 voice channels running at 2.048 Mbps.

Unfortunately, T1/E1 is not really suitable for connection to individual residences. The transmission protocol they used, alternate mark inversion (AMI), required transceivers 3,000 feet from the central office and every 6,000 feet thereafter. AMI demands so much bandwidth and corrupts the cable spectrum so much that telephone companies could use only one circuit in any 50-pair cable and none in any adjacent cables. Under these circumstances, providing high bandwidth service to homes would be equivalent to installing new wire.

2.1.4. xDSL

High Data-Rate Digital Subscriber Line (HDSL)

HDSL is simply a better way of transmitting T1/E1 over copper wires, using less bandwidth without repeaters. It uses more advanced modulation techniques to transmit 1.544 Mbps over lines up to 12,000 feet long.

Single-Line Digital Subscriber Line (SDSL)

SDSL is a single-line version of HDSL, transmitting T1/E1 signals over a single twisted pair, and able to operate over the plain old telephone service (POTS) so that a single line can support POTS and T1/E1 at the same time. It fits the market for residence connection which must often work over a single telephone line. However, SDSL will not reach much beyond 10,000 feet. At the same distance, ADSL reaches rates above 6 Mbps.

Asymmetric Digital Subscriber Line (ADSL)

ADSL is intended to complete the connection with the customer's premise. It transmits two separate data streams with much more bandwidth devoted to the downstream leg to the customer than returning. It is effective because symmetric signals in many pairs within a cable (as occurs in cables coming out of the central office) significantly limit the data rate and possible line length.

ADSL succeeds because it takes advantage of the fact that most of its target applications (video-on-demand, home shopping, Internet access, remote LAN access, multimedia, and PC services) function perfectly well with a relatively low upstream data rate. MPEG movies require 1.5 or 3.0 Mbps down stream but need only between 16 kbps and 64 kbps upstream. The protocols controlling Internet or LAN access require somewhat higher upstream rates but in most cases can get by with a 10 to 1 ratio of downstream to upstream bandwidth.

2.1.5. The Modem Market

Sales in the modem business started out slowly until customers started buying PCs. Likewise, costs were high until the volumes picked up. When the 14.4-kbps modem was first introduced, it cost $14,400 or one dollar per bit. Today, a much faster consumer-level modem with many more features costs only $100- $300, making it unusual for a home PC today to be without a modem.

Over the years, customers watched modem vendors evolve their products on a standards basis. This technique, although somewhat time consuming, was very important and led to significant feature enhancement. Initially, several modulation schemes were in use, but by the time the V.34 modem came out all of the major modem-modulation schemes were combined in that standard_ giving the customer one modem that could be used in many applications. As the modem market matured, customers became less concerned with the internals of standards and more concerned with features, size, and flexibility. As a result of the progress in analog-modem technology and with the advent of mass-market consumer-level PCs, there are over 500 million modems in the world today.

The xDSL modem market will follow similar market patterns. Today, things like modulation schemes, the type of protocol supported to the home or small business, and costs of the units are the main topics. As the xDSL market matures, most likely in a fashion similar to that of the analog modem, customers will become less concerned with modulation and protocols. On the other hand, they will look for vendors that provide plug-and-play interoperability with their data equipment, ease of installation, the best operating characteristics on marginal lines, and minimalist size and power requirements.

2.1.6. ATM versus IP to the Desktop

There is a great debate raging among potential service providers as to whether there should be standard IP 10BT connections or ATM connections to their customers' PCs. The two are very similar the difference is in the specifics of the equipment and not in the amount of equipment required.

There are various advantages to each method of network access:

IP Advantages

- 10BT Ethernet is basically self-learning.
- Inexpensive LAN PC cards already exist.
- 10BT is an industry standard.
- LAN networks are proven and work today.
- There is much expertise in this technology.
- PC software and OS drivers already interface to IP based LANs.

ATM Advantages

- Streaming video transport has already been proven.
- Mixing of services (e.g., video, telephony, and data) is much easier.
- Traffic speeds conform to standard telephony transport rates .
- New PC software and drivers will work with ATM.

The issue actually gets more interesting because both architectures usually interface to an ATM backbone network for high-speed connections over a wide area. Therefore, the real issues are the costs of building the network, the services that are to be carried over it, and the time frame for the implementation. If the need is for data services_ Internet connections, work at home, etc., the obvious choice is an IP network. The hardware and software required to implement this network is available and relatively inexpensive.

ATM would be the solution for multiple mixed QoS service requirements in the near future. It is true that the IP technology is being extended to offer tiered Q o S with RSVP, and IP telephony is being refined to operate more efficiently. The paradox, however, is that these standards do not exist today. ATM standards are quite complete. However, not all may be easily implementable. In spite of this, there are many ATM networks in existence or currently under construction.

This leaves the issue of costs. The true costs of creating and operating a large- scale data-access network are not known. True, there are portions that are understood, but many others are only projected. This creates great debate over which technology is actually less costly. The only way for the costs to be really known is to build reasonably large networks and compare costs. If one technology is a clear winner a somewhat doubtful hypothesis then use that technology. If there is no clear cost advantage, then build the network with the service set that matches the service needs of the potential customers. The issue is to start the implementation phase where the real answers will be determined and subsequently end the interminable discussion phase.

2.1.7. CAP versus DMT

These are the two primary xDSL standards over which much debate has ensued. Although the debate continues, the real action is taking place in the marketplace. CAP demonstrated a clear lead in getting product to market. Chips were available in quantity, and they worked. Numerous products that incorporated these chips are installed in a number of locations by service providers. Standards and interoperability issues between vendors and implementations are now being addressed.

DMT, on the other hand, has been in the standards arena for some time and continues to evolve. It is now considered a standard by a number of service providers. This technology featured some innovations that were not originally in the CAP feature set such as rate adaptation. On the other hand, the chips are just now finding their way into products. Trial activities are only now beginning, and advanced chip sets that match the features of CAP chips are now being promised for 3Q97.

The issue is which will win the market. The service providers who are building the xDSL network will select the technology that meets their needs. Many vendors are offering products that use either technology. Some new chips are being announced that allow adaptation between either technology. The point here is that the technology of xDSL chips is not a roadblock to deployment. Either appears to work well and true interoperability remains in the future much like mid-span meets for SONET equipment.

2.1.8. The Future

Look at the past of analog modems to foretell the future of xDSL. Standards were an issue with modems and will be an issue with xDSL products. However, it is not obvious to a technologist who or what technology will win out. Remember that in the VCR arena, Beta max had the better-quality picture, but VHS eventually won out.

2.2.1. Digital Subscriber Line

*Digital Subscriber Line (*DSL*)* technology is a modem technology that uses existing twisted-pair telephone lines to transport high-bandwidth data, such as multimedia and video, to service subscribers. The term *xDSL* covers a number of similar yet competing forms of DSL, including ADSL, SDSL, HDSL, RADSL, and VDSL. xDSL is drawing significant attention from implementers and service providers because it promises to deliver high-bandwidth data rates to dispersed locations with relatively small changes to the existing telco infrastructure. xDSL services are dedicated, point-to-point, public network access over twisted-pair copper wire on the local loop ("last mile") between a network service provider (NSP's) central office and the customer site, or on local loops created either intra-building or intra-campus. Currently the primary focus in xDSL is the development and deployment of ADSL and VDSL technologies and architectures. This chapter covers the characteristics and operations of ADSL and VDSL.

2.2.2. Asymmetric Digital Subscriber Line (ADSL)

ADSL technology is asymmetric. It allows more bandwidth downstream---from an NSP's central office to the customer site---than upstream from the subscriber to the central office. This asymmetry, combined with always-on access (which eliminates call setup), makes ADSL ideal for Internet/intranet surfing, video-on-demand, and remote LAN access. Users of these applications typically download much more information than they send.

ADSL transmits more than 6 Mbps to a subscriber, and as much as 640 kbps more in both directions (shown in Figure 2.3). Such rates expand existing access capacity by a factor of 50 or more without new cabling. ADSL can literally transform the existing public information network from one limited to voice, text, and low-resolution graphics to a powerful, ubiquitous system capable of bringing multimedia, including full motion video, to every home this century.

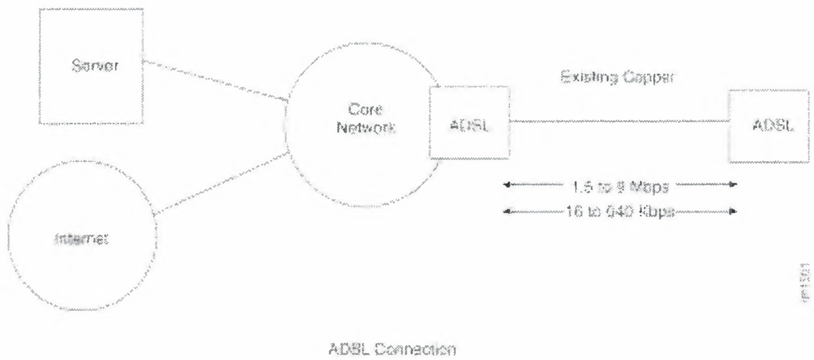

Figure 2.3 The components of a ADSL network include a telco and a CPE.[11]

ADSL will play a crucial role over the next decade or more as telephone companies enter new markets for delivering information in video and multimedia formats. New broadband cabling will take decades to reach all prospective subscribers. Success of these new services will depend on reaching as many subscribers as possible during the first few years. By bringing movies, television, video catalogs, remote CD-ROMs, corporate LANs, and the Internet into homes and small businesses, ADSL will make these markets viable and profitable for telephone companies and application suppliers alike.

ADSL Capabilities

An ADSL circuit connects an ADSL modem on each end of a twisted-pair telephone line, creating three information channels---a high-speed downstream channel, a medium-speed duplex channel, and a basic telephone service channel. The basic telephone service channel is split off from the digital modem by filters, thus guaranteeing uninterrupted basic telephone service, even if ADSL fails. The high-speed channel ranges from 1.5 to 6.1 Mbps, and duplex rates range from 16 to 640 kbps. Each channel can be sub multiplexed to form multiple lower-rate channels.

ADSL modems provide data rates consistent with North American T1 1.544 Mbps and European E1 2.048 Mbps digital hierarchies (see Figure 2.4) and can be purchased with various speed ranges and capabilities. The minimum configuration provides 1.5 or 2.0 Mbps downstream and a 16 kbps duplex channel; others provide rates of 6.1 Mbps and 64 kbps duplex. Products with downstream rates up to 8 Mbps and duplex rates up to 640 kbps are available today ADSL modems accommodate Asynchronous Transfer Mode (ATM) transport with variable rates and compensation for ATM overhead, as well as IP protocols.

Downstream data rates depend on a number of factors, including the length of the copper line, its wire gauge, presence of bridged taps, and cross-coupled interference. Line attenuation increases with line length and frequency and decreases as wire diameter increases. Ignoring bridged taps ADSL performs as shown in (Table 2.4).

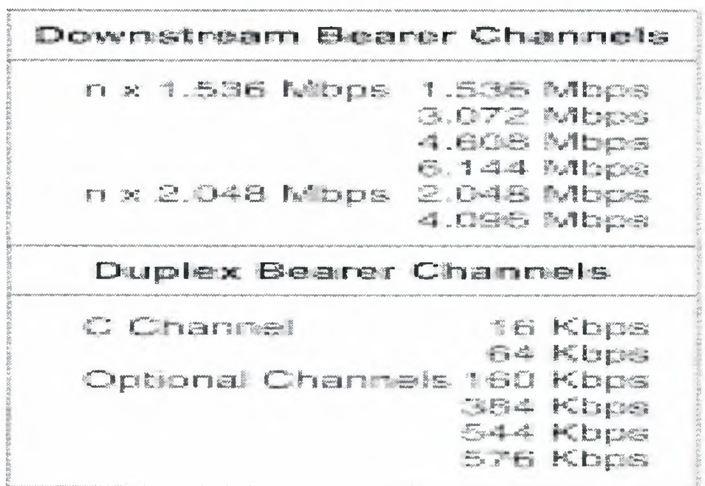

Figure 2.4 This chart shows the speeds for downstream bearer and duplex bearer channels.[12]

Table 2.1: Claimed ADSL Physical-Media Performance

Data rate (Mbps)	**Wire gauge (AWG)**	**Distance (feet)**	**Wire size (mm)**	**Distance (kilometers)**
1.5 or 2	24	18,000	0.5	5.5
1.5 or 2	26	15,000	0.4	4.6

6.1	24	12,000	0.5	3.7
6.1	26	9,000	0.4	2.7

Although the measure varies from telco to telco, these capabilities can cover up to 95% of a loop plant, depending on the desired data rate. Customers beyond these distances can be reached with fiber-based digital loop carrier (DLC) systems. As these DLC systems become commercially available, telephone companies can offer virtually ubiquitous access in a relatively short time.

Many applications envisioned for ADSL involve digital compressed video. As a real-time signal, digital video cannot use link- or network-level error control procedures commonly found in data communications systems. ADSL modems therefore incorporate forward error correction that dramatically reduces errors caused by impulse noise. Error correction on a symbol-by-symbol basis also reduces errors caused by continuous noise coupled into a line.

2.2.3. ADSL Technology

ADSL depends on advanced digital signal processing and creative algorithms to squeeze so much information through twisted-pair telephone lines. In addition, many advances have been required in transformers, analog filters, and analog/digital (A/D) converters. Long telephone lines may attenuate signals at 1 MHz (the outer edge of the band used by ADSL) by as much as 90 dB, forcing analog sections of ADSL modems to work very hard to realize large dynamic ranges, separate channels, and maintain low noise figures. On the outside, ADSL looks simple---transparent synchronous data pipes at various data rates over ordinary telephone lines. The inside, where all the transistors work, is a miracle of modern technology. Figure 2.5 displays the ADSL transceiver-network end.

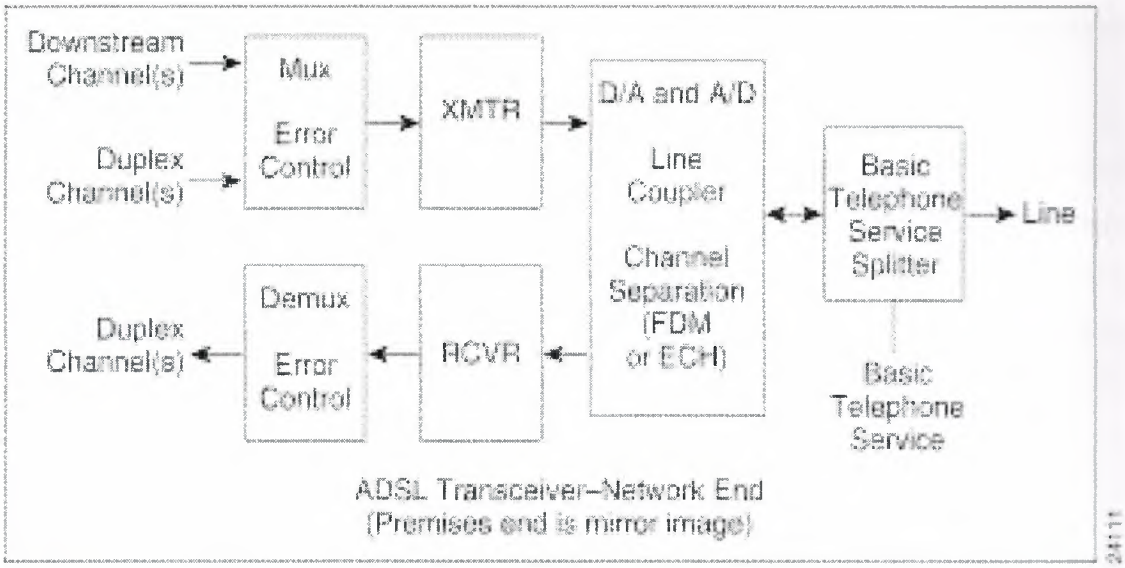

Figure 2.5 This diagram provides an overview of the devices that make up the ADSL transceiver-network end of the topology.**[12]**

To create multiple channels, ADSL modems divide the available bandwidth of a telephone line in one of two ways---frequency-division multiplexing (FDM) or echo cancellation---as shown in (Figure 2.6). FDM assigns one band for upstream data and another band for downstream data. The downstream path is then divided by time-division multiplexing into one or more high-speed channels and one or more low-speed channels. The upstream path is also multiplexed into corresponding low-speed channels. Echo cancellation assigns the upstream band to overlap the downstream, and separates the two by means of local echo cancellation, a technique well known in V.32 and V.34 modems. With either technique, ADSL splits off a 4 kHz region for basic telephone service at the DC end of the band.

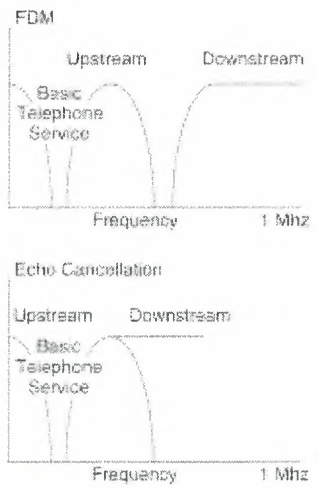

Figure 2.6 ADSL uses FDM and echo cancellation to divide the available bandwidth for services. [5]

An ADSL modem organizes the aggregate data stream created by multiplexing downstream channels, duplex channels, and maintenance channels together into blocks, and attaches an error correction code to each block. The receiver then corrects errors that occur during transmission up to the limits implied by the code and the block length. The unit may, at the user's option, also create superblocks by interleaving data within sub locks; this allows the receiver to correct any combination of errors within a specific span of bits. This in turn allows for effective transmission of both data and video signals.

ADSL Standards and Associations

The American National Standards Institute (ANSI) Working Group T1E1.4 recently approved an ADSL standard at rates up to 6.1 Mbps (ANSI Standard T1.413). The European Technical Standards Institute (ETSI) contributed an annex to T1.413 to reflect European requirements. T1.413 currently embodies a single terminal interface at the premises end. Issue II, now under study by T1E1.4, will expand the standard to include a multiplexed interface at the premises end, protocols for configuration and network management, and other improvements.

The ATM Forum and the Digital Audio-Visual Council (DAVIC) have both recognized ADSL as a physical-layer transmission protocol for UTP media.

The ADSL Forum was formed in December 1994 to promote the ADSL concept and facilitate development of ADSL system architectures, protocols, and interfaces for major ADSL applications. The forum has more than 200 members, representing service providers, equipment manufacturers, and semiconductor companies throughout the world. At present, the Forum's formal technical work is divided into the following six areas, each of which is dealt with in a separate working group within the technical committee:

- ATM over ADSL (including transport and end-to-end architecture aspects)
- Packet over ADSL (this working group recently completed its work)
- CPE/CO (customer premises equipment/central office) configurations and interfaces
- Operations
- Network management
- Testing and interoperability

ADSL Market Status

ADSL modems have been tested successfully in more than 30 telephone companies, and thousands of lines have been installed in various technology trials in North America and Europe. Several telephone companies plan market trials using ADSL, principally for data access, but also including video applications for uses such as personal shopping, interactive games, and educational programming.

Semiconductor companies have introduced transceiver chipsets that are already being used in market trials. These chipsets combine off-the-shelf components, programmable digital signal processors, and custom ASICs (application-specific integrated circuits). Continued investment by these semiconductor companies has increased functionality and reduced chip count, power consumption, and cost, enabling mass deployment of ADSL-based services.

2.2.4. Very-High-Data-Rate Digital Subscriber Line (VDSL)

It is becoming increasingly clear that telephone companies around the world are making decisions to include existing twisted-pair loops in their next-generation broadband access networks. Hybrid fiber coax (HFC), a shared-access medium well suited to analog and digital broadcast, comes up somewhat short when used to carry voice telephony, interactive video, and high-speed data communications at the same time. Fiber all the way to the home (FTTH) is still prohibitively expensive in a marketplace soon to be driven by competition rather than cost. An attractive alternative, soon to be commercially practical, is a combination of fiber cables feeding neighborhood optical network units (ONUs) and last-leg-premises connections by existing or new copper. This topology, which is often called fiber to the neighborhood (FTTN), encompasses fiber to the curb (FTTC) with short drops and fiber to the basement (FTTB), serving tall buildings with vertical drops.

One of the enabling technologies for FTTN is VDSL. In simple terms, VDSL transmits high-speed data over short reaches of twisted-pair copper telephone lines, with a range of speeds depending on actual line length. The maximum downstream rate under consideration is between 51 and 55 Mbps over lines up to 1000 feet (300 m) in length. Downstream speeds as low as 13 Mbps over lengths beyond 4000 feet (1500 m) are also common. Upstream rates in early models will be asymmetric, just like ADSL, at speeds from 1.6 to 2.3 Mbps. Both data channels will be separated in frequency from bands used for basic telephone service and Integrated Services Digital Network (ISDN), enabling service providers to overlay VDSL on existing services. At present the two high-speed channels are also separated in frequency. As needs arise for higher-speed upstream channels or symmetric rates, VDSL systems may need to use echo cancellation.

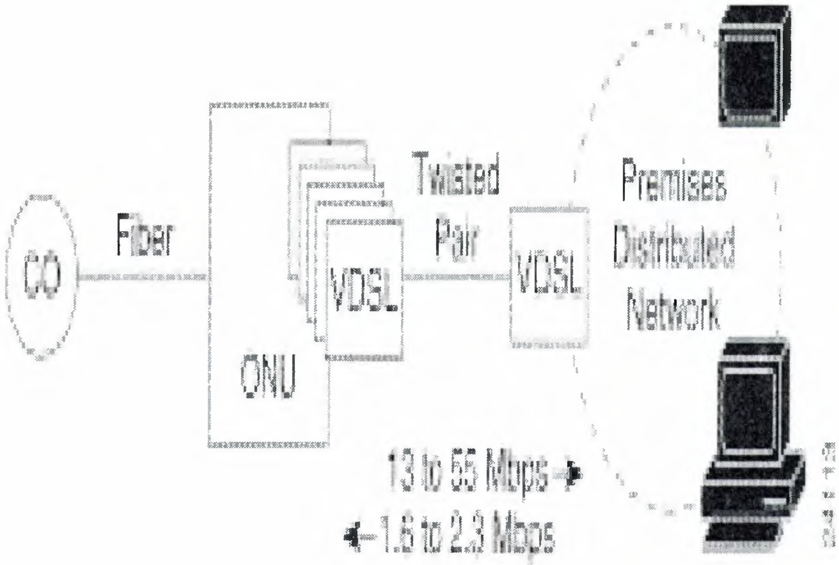

Figure 2.7 This diagram provides an overview of the devices in a VDSL network.[5]

2.2.4.1. VDSL Projected Capabilities

Although VDSL has not achieved ADSL's degree of definition, it has advanced far enough that we can discuss realizable goals, beginning with data rate and range. Downstream rates derive from submultiples of the SONET (Synchronous Optical Network) and SDH (Synchronous Digital Hierarchy) canonical speed of 155.52 Mbps, namely 51.84 Mbps, 25.92 Mbps, and 12.96 Mbps. Each rate has a corresponding target range:

Target Range (Mbps)	**Distance (feet)**	**Distance (meters)**
12.96-13.8	4500	1500
25.92-27.6	3000	1000
51.84-55.2	1000	300

Upstream rates under discussion fall into three general ranges:

- 1.6-2.3 Mbps.
- 19.2 Mbps
- Equal to downstream

Early versions of VDSL will almost certainly incorporate the slower asymmetric rate. Higher upstream and symmetric configurations may only be possible for very short lines. Like ADSL, VDSL must transmit compressed video, a real-time signal unsuited to error retransmission schemes used in data communications. To achieve error rates compatible with those of compressed video, VDSL will have to incorporate forward error correction (FEC) with sufficient interleaving to correct all errors created by impulsive noise events of some specified duration. Interleaving introduces delay, on the order of 40 times the maximum length correctable impulse.

Data in the downstream direction will be broadcast to every CPE on the premises or be transmitted to a logically separated hub that distributes data to addressed CPE based on cell or time-division multiplexing (TDM) within the data stream itself. Upstream multiplexing is more difficult. Systems using a passive network termination (NT) must insert data onto a shared medium, either by a form of TDM access (TDMA) or a form of frequency-division multiplexing (FDM). TDMA may use a species of token control called cell grants passed in the downstream direction from the ONU modem, or contention, or both (contention for unrecognized devices, cell grants for recognized devices). FDM gives each CPE its own channel, obviating a Media Access Control (MAC) protocol, but either limiting data rates available to any one CPE or requiring dynamic allocation of bandwidth and inverse multiplexing at each CPE. Systems using active NTs transfer the upstream collection problem to a logically separated hub that would use (typically) Ethernet or ATM protocols for upstream multiplexing.

Migration and inventory considerations dictate VDSL units that can operate at various (preferably all) speeds with automatic recognition of a newly connected device to a line or a change in speed. Passive network interfaces need to have hot insertion, where a new VDSL premises unit can be put on the line without interfering with the operation of other modems.

2.2.4.2. VDSL Technology

VDSL technology resembles ADSL to a large degree, although ADSL must face much larger dynamic ranges and is considerably more complex as a result. VDSL must be lower in cost and lower in power, and premises VDSL units may have to implement a physical-layer MAC for multiplexing upstream data.

2.2.4.3. Line Code Candidates

Four line codes have been proposed for VDSL:

- *CAP (carrierless amplitude modulation/phase modulation)*---A version of suppressed carrier quadrature amplitude modulation (QAM). For passive NT configurations, CAP would use quadrature phase shift keying (QPSK) upstream and a type of TDMA for multiplexing (although CAP does not preclude an FDM approach to upstream multiplexing).
- *DMT (discrete multitone)*---A multicarrier system using discrete fourier transforms to create and demodulate individual carriers. For passive NT configurations, DMT would use FDM for upstream multiplexing (although DMT does not preclude a TDMA multiplexing strategy).
- *DWMT (discrete wavelet multitone)*---A multicarrier system using wavelet transforms to create and demodulate individual carriers. DWMT also uses FDM for upstream multiplexing, but also allows TDMA.
- *SLC (simple line code)*---A version of four-level baseband signaling that filters the based band and restores it at the receiver. For passive NT configurations, SLC would most likely use TDMA for upstream multiplexing, although FDM is possible.

Channel Separation

Early versions of VDSL will use frequency division multiplexing to separate downstream from upstream channels and both of them from basic telephone service and ISDN(figure2.8). Echo cancellation may be required for later-generation systems featuring symmetric data rates. A rather substantial distance, in frequency, will be maintained between the lowest data channel and basic telephone service to enable very simple and cost-effective basic telephone service splitters. Normal practice would locate the downstream channel above the upstream channel. However, the DAVIC specification reverses this order to enable premises distribution of VDSL signals over coaxial cable systems.

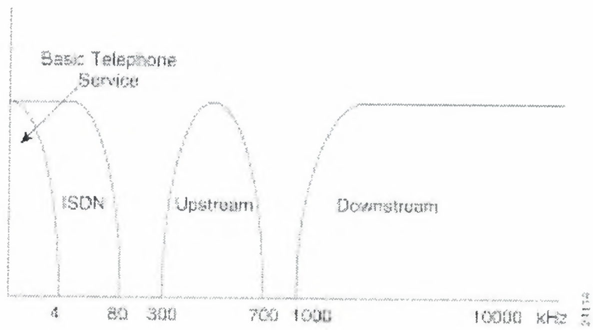

Figure 2.8 Early versions of VDSL will use FDM to separate downstream from upstream channels and both of them from basic telephone service and ISDN, as this example shows.

Forward Error Control

FEC will no doubt use a form of Reed Soloman coding and optional interleaving to correct bursts of errors caused by impulse noise. The structure will be very similar to ADSL, as defined in T1.413. An outstanding question is whether FEC overhead (in the range of 8%) will be taken from the payload capacity or added as an out-of-band signal. The former reduces payload capacity but maintains nominal reach, whereas the latter retains the nominal payload but suffers a small reduction in reach. ADSL puts FEC overhead out of band.

Upstream Multiplexing

If the premises VDSL unit comprises the network termination (an active NT), then the means of multiplexing upstream cells or data channels from more than one CPE into a single upstream becomes the responsibility of the premises network. The VDSL unit simply presents raw data streams in both directions. As illustrated in (Figure 2-8), one type of premises network involves a star connecting each CPE to a switching or multiplexing hub; such a hub could be integral to the premises VDSL unit.

In a passive NT configuration, each CPE has an associated VDSL unit. (A passive NT does not conceptually preclude multiple CPE per VDSL, but then the question of active versus passive NT (Figure 2.9) becomes a matter of ownership, not a matter of wiring topology and multiplexing strategies.) Now the upstream channels for each CPE must share a common wire. Although a collision-detection system could be used, the desire for guaranteed bandwidth indicates one of two solutions.

The first invokes a cell-grant protocol in which downstream frames generated at the ONU or farther up the network contain a few bits that grant access to specific CPE during a specified period subsequent to receiving a frame. A granted CPE can send one upstream cell during this period.

The transmitter in the CPE must turn on, send a preamble to condition the ONU receiver, send the cell, and then turn itself off. The protocol must insert enough silence to let line ringing clear. One construction of this protocol uses 77 octet intervals to transmit a single 53-octet cell.

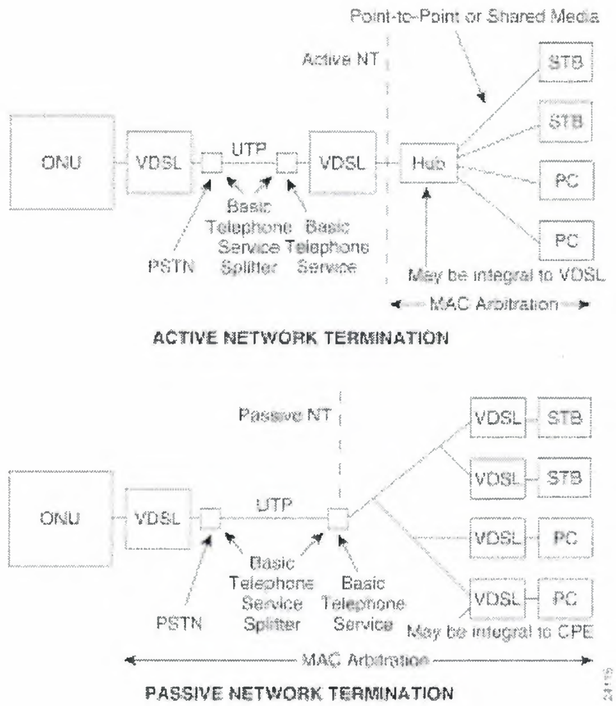

Figure 2.9 This figure shows examples of termination methods in passive and active networks. [5]

The second method divides the upstream channel into frequency bands and assigns one band to each CPE. This method has the advantage of avoiding any MAC with its associated overhead (although a multiplexor must be built into the ONU), but either restricts the data rate available to any one CPE or imposes a dynamic inverse multiplexing scheme that lets one CPE send more than its share for a period. The latter would look a great deal like a MAC protocol, but without the loss of bandwidth associated with carrier detect and clear for each cell.

2.2.4.4. VDSL Issues

VDSL is still in the definition stage; some preliminary products exist, but not enough is known yet about telephone line characteristics, radio frequency interface emissions and susceptibility, upstream multiplexing protocols, and information requirements to frame a set of definitive, standardize able properties. One large unknown is the maximum distance that VDSL can reliably realize for a given data rate. This is unknown because real line characteristics at the frequencies required for VDSL are speculative, and items such as short bridged taps or un terminated extension lines in homes, which have no effect on telephony, ISDN, or ADSL, may have very detrimental affects on VDSL in certain configurations. Furthermore, VDSL invades the frequency ranges of amateur radio, and every above-ground telephone wire is an antenna that both radiates and attracts energy in amateur radio bands. Balancing low signal levels to prevent emissions that interfere with amateur radio with higher signals needed to combat interference by amateur radio could be the dominant factor in determining line reach.

A second dimension of VDSL that is far from clear is the services environment. It can be assumed that VDSL will carry information in ATM cell format for video and asymmetric data communications, although optimum downstream and upstream data rates have not been ascertained. What is more difficult to assess is the need for VDSL to carry information in non-ATM formats (such as conventional Plesiochronous Digital Hierarchy [PDH] structures) and the need for symmetric channels at broadband rates (above T1/E1). VDSL will not be completely independent of upper-layer protocols, particularly in the upstream direction, where multiplexing data from more than one CPE may require knowledge of link-layer formats (that is, ATM or not).

A third difficult subject is premises distribution and the interface between the telephone network and CPE. Cost considerations favor a passive network interface with premises VDSL installed in CPE and upstream multiplexing handled similarly to LAN buses. System management, reliability, regulatory constraints, and migration favor an active network termination, just like ADSL and ISDN, that can operate like a hub, with point-to-point or shared-media distribution to multiple CPE on-premises wiring that is independent and physically isolated from network wiring.

However, costs cannot be ignored. Small ONUs must spread common equipment costs, such as fiber links, interfaces, and equipment cabinets, over a small number of subscribers compared to HFC. VDSL therefore has a much lower cost target than ADSL because VDSL may connect directly from a wiring center or cable modems, which also have much lower common equipment costs per user. Furthermore, VDSL for passive NTs may (only *may*) be more expensive than VDSL for active NTs, but the elimination of any other premises network electronics may make it the most cost-effective solution, and highly desired, despite the obvious benefits of an active NT.

Standards Status

At present five standards organizations/forums have begun work on VDSL:

- *T1E1.4*---The U.S. ANSI standards group T1E1.4 has just begun a project for VDSL, making a first attack on system requirements that will evolve into a system and protocol definition.
- *ETSI*---The ETSI has a VDSL standards project, under the title High-Speed Metallic Access Systems, and has compiled a list of objective, problems, and requirements. Among its preliminary findings are the need for an active NT and payloads in multiples of SDH virtual container VC-12, or 2.3 Mbps. ETSI works very closely with T1E1.4 and the ADSL Forum, with significant overlapping attendees.
- *DAVIC*---DAVIC has taken the earliest position on VDSL. Its first specification due to be finalized will define a line code for downstream data, another for upstream data, and a MAC for upstream multiplexing based on TDMA over shared wiring. DAVIC is only specifying VDSL for a single downstream rate of 51.84 Mbps and a single upstream rate of 1.6 Mbps over 300 m or less of copper. The proposal assumes, and is driven to a large extent by, a passive NT, and further assumes premises distribution from the NT over new coaxial cable or new copper wiring.
- *The ATM Forum*---The ATM Forum has defined a 51.84 Mbps interface for private network UNIs and a corresponding transmission technology. It has also taken up the question of CPE distribution and delivery of ATM all the way to premises over the various access technologies described above.

- *The ADSL Forum*---The ADSL Forum has just begun consideration of VDSL. In keeping with its charter, the forum will address network, protocol, and architectural aspects of VDSL for all prospective applications, leaving line code and transceiver protocols to T1E1.4 and ETSI and higher-layer protocols to organizations such as the ATM Forum and DAVIC.

2.2.5. VDSL's Relationship with ADSL

VDSL has an odd technical resemblance to ADSL. VDSL achieves data rates nearly 10 times greater than those of ADSL (shown in Figure 2.10), but ADSL is the more complex transmission technology, in large part because ADSL must contend with much larger dynamic ranges than VDSL. However, the two are essentially cut from the same cloth. ADSL employs advanced transmission techniques and forward error correction to realize data rates from 1.5 to 9 Mbps over twisted pair, ranging to 18,000 feet; VDSL employs the same advanced transmission techniques and forward error correction to realize data rates from 13 to 55 Mbps over twisted pair, ranging to 4,500 feet. Indeed, the two can be considered a continuum, a set of transmission tools that delivers about as much data as theoretically possible over varying distances of existing telephone wiring.

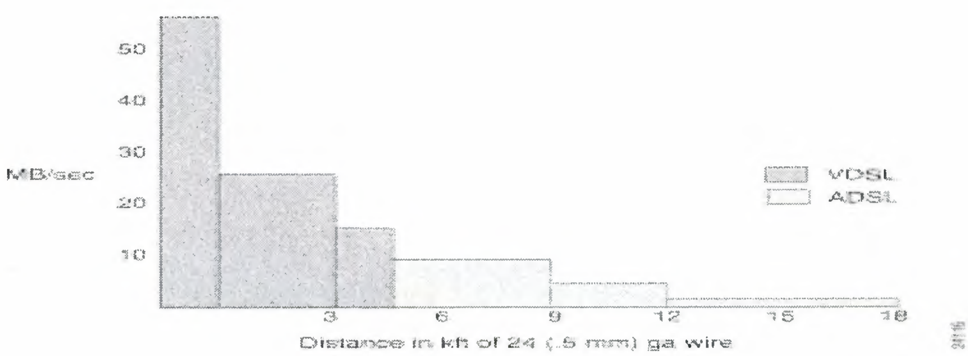

Figure 2.10 This chart provides a comparison of transfer rates between ADSL and VDSL.[12]

VDSL is clearly a technology suitable for a full-service network (assuming that *full service* does not imply more than two high-definition television [HDTV] channels over the highest-rate VDSL). It is equally clear that telephone companies cannot deploy ONUs overnight, even if all the technology were available. ADSL may not be a full-service network technology, but it has the singular advantage of offering service over lines that exist today, and ADSL products are closer in time than VDSL. Many new services being contemplated today such as videoconferencing, Internet access, video on demand, and remote LAN access can be delivered at speeds at or below T1/E1 rates. For such services, ADSL/VDSL provides an ideal combination for network evolution. On the longest lines, ADSL delivers a single channel. As line length shrinks, either from natural proximity to a central office or deployment of fiber-based access nodes, ADSL and VDSL simply offer more channels and capacity for services that require rates above T1/E1 (such as digital live television and virtual CD-ROM access).

CHAPTER THREE

ORTHOGONAL FREQUENCY DIVISION MULTIPLEXING

Essentially identical to Coded OFDM (COFDM) is a digital multi-carrier modulation scheme, which uses a large number of closely-spaced orthogonal *sub-carriers*. Each sub-carrier is modulated with a conventional modulation scheme (such as quadrature amplitude modulation) at a low symbol rate, maintaining data rates similar to conventional *single-carrier* modulation schemes in the same bandwidth. In practice, OFDM signals are generated using the Fast Fourier transform algorithm.

The primary advantage of OFDM over single-carrier schemes is its ability to cope with severe channel conditions for example, multipath and narrowband interference without complex equalization filters. Channel equalization is simplified because OFDM may be viewed as using many slowly-modulated narrowband signals rather than one rapidly-modulated wideband signal.

The orthogonality of the sub-carriers results in zero cross-talk, even though they are so close that their spectra overlap. Low symbol rate helps manage time-domain spreading of the signal (such as multipath propagation) by allowing the use of a guard interval between symbols. The guard interval also eliminates the need for a pulse-shaping filter. OFDM has developed into a popular scheme for wideband digital communication systems.

A major goal of modern communication systems is to allow high-speed communication, regardless of the location or mobility of the system users. However, this goal is difficult to achieve due to the multipath fading that affects wireless communication signals. One alternative for achieving high-speed wireless communication in the presence of multipath fading is to employ a multicarrier system, generally implemented as an orthogonal frequency division multiplexing (OFDM) system, in conjunction with error control coding. Such coded OFDM systems have emerged recently as a serious competitor to single-carrier systems and have been employed or are being considered for a number of applications, including digital audio broadcast and digital video broadcast in Europe, wireless local area networks, broadband fixed wireless access, and cellular data. One of the challenges to be overcome when employing an OFDM system in low-power peer-to-peer wireless communication systems is that the complex envelope of the baseband OFDM signal can demonstrate significant variation; in other words, its peak-to-mean envelope power ratio (PMEPR) can be much larger than that of an analogous single-carrier system.

3.1 Key features

3.1.1 Summary of advantages

- Can easily adapt to severe channel conditions without complex equalization
- Robust against narrow-band co-channel interference
- Robust against Inter symbol interference (ISI) and fading caused by multipath propagation
- High spectral efficiency
- Efficient implementation using FFT
- Low sensitivity to time synchronization errors
- Tuned sub-channel receiver filters are not required (unlike conventional FDM)
- Facilitates Single Frequency Networks, i.e. transmitter macro diversity.

3.1.2 Summary of disadvantages

- Sensitive to Doppler shift.
- Sensitive to frequency synchronization problems.
- Inefficient transmitter power consumption, due to linear power amplifier requirement.

3.2 Characteristics and principles of operation Orthogonality

In OFDM, the sub-carrier frequencies are chosen so that the sub-carriers are orthogonal to each other, meaning that cross-talk between the sub-channels is eliminated and inter-carrier guard bands are not required. This greatly simplifies the design of both the transmitter and the receiver; unlike conventional FDM, a separate filter for each sub-channel is not required.

3.2.1 Orthogonality

The orthogonality also allows high spectral efficiency, near the Nyquist rate. Almost the whole available frequency band can be utilized. OFDM generally has a nearly 'white' spectrum, giving it benign electromagnetic interference properties with respect to other co-channel users. The orthogonality allows for efficient modulator and demodulator implementation using the FFT algorithm. Although the principles and some of the benefits have been known since the 1960s, OFDM is popular for wideband communications today by way of low-cost digital signal processing components that can efficiently calculate the FFT. OFDM requires very accurate frequency synchronization between the receiver and the transmitter; any deviation and the sub-carriers are no longer orthogonal, causing *inter-carrier interference* (ICI), i.e. cross-talk between the sub-carriers. Frequency offsets are typically caused by mismatched transmitter and receiver oscillators, or by Doppler shift due to movement. Whilst Doppler shift alone may be compensated for by the receiver, the situation is worsened when combined with multipath, as reflections will appear at various frequency offsets, which is much harder to correct. This effect typically worsens as speed increases, and is an important factor limiting the use of OFDM in high-speed vehicles. Several techniques for ICI suppression are suggested, but they may increase the receiver complexity.

3.2.2 Guard interval for elimination of inter-symbol interference

One key principle of OFDM is that since low symbol rate modulation schemes (i.e. where the symbols are relatively long compared to the channel time characteristics) suffer less from inter symbol interference caused by multipath, it is advantageous to transmit a number of low-rate streams in parallel instead of a single high-rate stream. Since the duration of each symbol is long, it is feasible to insert a guard interval between the OFDM symbols, thus eliminating the inter symbol interference. The guard-interval also reduces the sensitivity to time synchronization problems.

A simple example: If one sends a million symbols per second using conventional single-carrier modulation over a wireless channel, then the duration of each symbol would be one microsecond or less. This imposes severe constraints on synchronization and necessitates the removal of multipath interference. If the same million symbols per second are spread among one thousand sub-channels, the duration of each symbol can be longer by a factor of thousand, i.e. one millisecond, for orthogonality with approximately the same bandwidth. Assume that a guard interval of 1/8 of the symbol length is inserted between each symbol. Inter symbol interference can be avoided if the multipath time-spreading (the time between the reception of the first and the last echo) is shorter than the guard interval, i.e. 125 microseconds. This corresponds to a maximum difference of 37.5 kilometers between the lengths of the paths. The last 125 microseconds of each symbol are copied and sent in advance of the symbol as a cyclic prefix. The cyclic prefix, which is transmitted during the guard interval, consists of the end of the OFDM symbol copied into the guard interval, and the guard interval is transmitted followed by the OFDM symbol. The reason that the guard interval consists of a copy of the end of the OFDM symbol is so that the receiver will integrate over an integer number of sinusoid cycles for each of the multi paths when it performs OFDM demodulation with the FFT.

3.2.3 Simplified equalization

The effects of frequency-selective channel conditions, for example fading caused by multipath propagation, can be considered as constant (flat) over an OFDM sub-channel if the sub-channel is sufficiently narrow-banded, i.e. if the number of sub-channels is sufficiently large. This makes equalization far simpler at the receiver in OFDM in comparison to conventional single-carrier modulation. The equalizer only has to multiply each sub-carrier by a constant value, or a rarely changed value.

Our example: The OFDM equalization in the above numerical example would require N = 1000 complex multiplications per OFDM symbol, i.e. one million multiplications per second, at the receiver. The FFT algorithm requires $N\log_2 N$ = 10.000 complex-valued multiplications per OFDM symbol, i.e. *10 million multiplications per second*, at both the receiver and transmitter side. This should be compared with the corresponding one million symbols/second single-carrier modulation case mentioned in the example, where the equalization of 125 microseconds time-spreading using a FIR filter would require 125 multiplications per symbol, i.e. *125 million multiplications per second.*

Some of the sub-carriers in some of the OFDM symbols may carry pilot signals for measurement of the channel conditions, i.e. the equalizer gain for each sub-carrier. Pilot signals may also be used for synchronization. If differential modulation such as DPSK or DQPSK is applied to each sub-carrier, equalization can be completely omitted, since these schemes are insensitive to slowly changing amplitude and phase distortion.

3.2.4 Channel coding and interleaving

OFDM is invariably used in conjunction with channel coding (forward error correction) and almost always uses frequency and/or time interleaving. Frequency (subcarrier) interleaving increases resistance to frequency-selective channel conditions such as fading. For example, when a part of the channel bandwidth is faded, frequency interleaving ensures that the bit errors that would result from those subcarriers in the faded part of the bandwidth are spread out in the bit-stream rather than being concentrated. Similarly, time interleaving ensures that bits that are originally close together in the bit-stream are transmitted far apart in time, thus mitigating against severe fading as would happen when travelling at high speed.

However, time interleaving is of little benefit in slowly fading channels, such as for stationary reception, and frequency interleaving offers little to no benefit for narrowband channels that suffer from flat-fading (where the whole channel bandwidth is faded at the same time).

The reason why interleaving is used on OFDM is to attempt to spread the errors out in the bit-stream that is presented to the error correction decoder, because when such decoders are presented with a high concentration of errors the decoder is unable to correct all the bit errors, and a burst of uncorrected errors occurs. A common type of error correction coding used with OFDM-based systems is convolutional coding, which is often concatenated with Reed-Solomon coding. Convolutional coding is used as the inner code and Reed-Solomon coding is used for the outer code usually with additional interleaving (on top of the time and frequency interleaving mentioned above) in between the two layers of coding. The reason why this combination of error correction coding is used is that the viterbi decoder used for convolutional decoding produces short errors bursts when there is a high concentration of errors, and Reed-Solomon codes are inherently well-suited to correcting bursts of errors. Newer systems, however, usually now adopt the near-optimal types of error correction coding that use the turbo decoding principle, where the decoder iterates towards the desired solution. Examples of such error correction coding types include turbo codes and LDPC codes. These codes only perform close to the Shannon limit for the Additive White Gaussian Noise (AWGN) channel, however, and some systems that have adopted these codes have concatenated them with either Reed-Solomon (for example on the Media FLO system) or BCH codes (on the DVB-S2 system) to improve performance further over the wireless channel.

3.2.5 Adaptive transmission

The resilience to severe channel conditions can be further enhanced if information about the channel is sent over a return-channel. Based on this feedback information, adaptive modulation, channel coding and power allocation may be applied across all sub-carriers, or individually to each sub-carrier. In the latter case, if a particular range of frequencies suffers from interference or attenuation, the carriers within that range can be disabled or made to run slower by applying more robust modulation or error coding to those sub-carriers. The term discrete multi tone modulation (DMT) denotes OFDM based communication systems that adapt the transmission to the channel conditions individually for each sub-carrier, by means of so called bit-loading. Examples are ADSL and VDSL. The upstream and downstream speeds can be varied by allocating either more or fewer carriers for each purpose. Some forms of Rate-adaptive DSL use this feature in real time, so that bandwidth is allocated to whichever stream needs it most.

3.2.6 OFDM extended with multiple access

OFDM in its primary form is considered as a digital modulation technique, and not a multi-user channel access technique, since it is utilized for transferring one bit stream over one communication channel using one sequence of OFDM symbols. However, OFDM can be combined with multiple access using time, frequency or coding separation of the users.

In Orthogonal Frequency Division Multiple Access (OFDMA), frequency-division multiple access is achieved by assigning different OFDM sub-channels to different users. OFDMA supports differentiated quality-of-service by assigning different number of sub-carriers to different users in a similar fashion as in CDMA, and thus complex packet scheduling or media access control schemes can be avoided. OFDMA is used in the uplink of the IEEE 802.16 Wireless MAN standard, commonly referred to as WiMAX.

3.2.7 Space diversity

In OFDM based wide area broadcasting, receivers can benefit from receiving signals from several spatially-dispersed transmitters simultaneously, since transmitters will only destructively interfere with each other on a limited number of sub-carriers, whereas in general they will actually reinforce coverage over a wide area. This is very beneficial in many countries, as it permits the operation of national single-frequency networks (SFNs), where many transmitters send the same signal simultaneously over the same channel frequency. SFNs utilize the available spectrum more effectively than conventional multi-frequency broadcast networks (MFN), where program content is replicated on different carrier frequencies. SFNs also result in a diversity scheme gain in receivers situated midway between the transmitters. The coverage area is increased and the outage probability decreased in comparison to an MFN, due to increased received signal strength averaged over all sub-carriers. Single-frequency networks is a form of transmitter macro diversity. OFDM may be combined with other forms of space diversity, for example antenna arrays and MIMO channels. This is done in the IEEE802.11n Wireless LAN standard.

3.2.8 Linear transmitter power amplifier

An OFDM signal exhibits a high peak-to-average power ratio (PAPR) because the independent phases of the sub-carriers mean that they will often combine constructively. Handling this high PAPR requires:

- a high-resolution digital-to-analog converter (DAC) in the transmitter
- a high-resolution analog-to-digital converter (ADC) in the receiver
- a linear signal chain.

Any non-linearity in the signal chain will cause inter modulation distortion that raises the noise floor may cause inter symbol interference generates out-of-band spurious radiation.

The linearity requirement is demanding, especially for transmitter RF output circuitry where amplifiers are often designed to be non-linear in order to minimize power consumption. In practical OFDM systems a small amount of peak clipping is allowed to limit the PAPR in a judicious tradeoff against the above consequences. However the transmitter output filter which is required to reduce out-of-band to legal levels has the effect of restoring peak levels that were clipped, so clipping is not an effective way to reduce PAPR.

3.3 Ideal system model

This section describes a simple idealized OFDM system model suitable for a time-invariant AWGN channel.

3.3.1 Transmitter

An OFDM carrier signal is the sum of a number of orthogonal sub-carriers, with baseband data on each sub-carrier being independently modulated commonly using some type of quadrature amplitude modulation (QAM) or phase-shift keying (PSK). This composite baseband signal is typically used to modulate a main RF carrier.

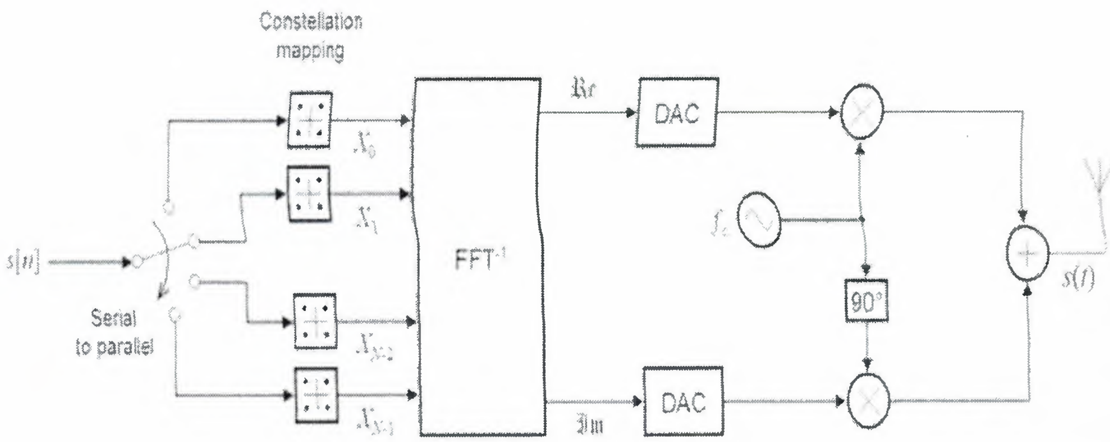

Figure 3.1 *OFDM Transmitter [3]*

$s[n]$ is a serial stream of binary digits. By inverse multiplexing, these are first de-multiplexed into N parallel streams, and each one mapped to a (possibly complex) symbol stream using some modulation constellation (QAM, PSK, etc.). Note that the constellations may be different, so some streams may carry a higher bit-rate than others.

An inverse FFT is computed on each set of symbols, giving a set of complex time-domain samples. These samples are then quadrature-mixed to pass band in the standard way. The real and imaginary components are first converted to the analogue domain using digital-to-analogue converters (DACs); the analogue signals are then used to modulate cosine and sine waves at the carrier frequency, f_c, respectively. These signals are then summed to give the transmission signal, $s(t)$.

3.3.2 Receiver

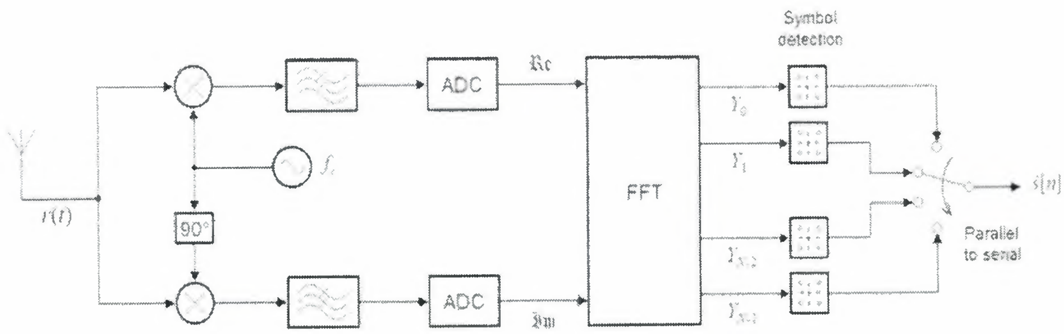

Figure 3.2 OFDM Receiver [5]

The receiver picks up the signal $r(t)$, which is then quadrature-mixed down to baseband using cosine and sine waves at the carrier frequency. This also creates signals centered on $2f_c$, so low-pass filters are used to reject these. The baseband signals are then sampled and digitized using analogue-to-digital converters (ADCs), and a forward FFT is used to convert back to the frequency domain. This returns N parallel streams, each of which is converted to a binary stream using an appropriate symbol detector. These streams are then re-combined into a serial stream, $\hat{s}[n]$, which is an estimate of the original binary stream at the transmitter.

3.4 Mathematical description

If N sub-carriers are used, and each sub-carrier is modulated using M alternative symbols, the OFDM symbol alphabet consists of M^N combined symbols. The low-pass equivalent OFDM signal is expressed as:

$$\nu(t) = \sum_{k=0}^{N-1} X_k e^{i2\pi kt/T}, \quad 0 \le t < T,$$

where $\{X_k\}$ are the data symbols, N is the number of sub-carriers, and T is the OFDM symbol time. The sub-carrier spacing of $1 / T$ makes them orthogonal over each symbol period; this property is expressed as:

$$\frac{1}{T}\int_0^T \left(e^{i2\pi k_1 t/T}\right)^* \left(e^{i2\pi k_2 t/T}\right) dt$$
$$= \frac{1}{T}\int_0^T e^{i2\pi(k_2-k_1)t/T} dt = \begin{cases} 1, & k_1 = k_2 \\ 0, & k_1 \neq k_2 \end{cases}$$

where $(\cdot)^*$ denotes the complex conjugate operator. To avoid inter symbol interference in multipath fading channels, a guard interval of length T_g is inserted prior to the OFDM block. During this interval, a *cyclic prefix* is transmitted such that the signal in the interval $-T_g \le t < 0$ equals the signal in the interval $(T - T_g) \le t < T$. The OFDM signal with cyclic prefix is thus:

$$\nu(t) = \sum_{k=0}^{N-1} X_k e^{i2\pi kt/T}, \quad -T_g \le t < T$$

The low-pass signal above can be either real or complex-valued. Real-valued low-pass equivalent signals are typically transmitted at baseband—wire line applications such as DSL use this approach.

3.5 Usage

3.5.1 ADSL

OFDM is used in ADSL connections that follow the G.DMT (ITU G.992.1) standard, in which existing copper wires are used to achieve high-speed data connections. Long copper wires suffer from attenuation at high frequencies. The fact that OFDM can cope with this frequency selective attenuation and with narrow-band interference are the main reasons it is frequently used in applications such as ADSL modems. However, DSL cannot be used on every copper pair; interference may become significant if more than 25% of phone lines coming into a central office are used for DSL.

3.5.2 Power line Technology

OFDM is used by power line devices to extend Ethernet connections to other rooms in a home through its power wiring. Adaptive modulation is particularly important with such a noisy channel as electrical wiring.

3.5.3 Wireless local area networks (LAN) and metropolitan area networks (MAN)

OFDM is also now being used in some wireless LAN and MAN applications, including IEEE 802.11a/g (and the defunct European alternative HIPERLAN/2) and WiMAX. IEEE 802.11a, operating in the 5 GHz band, specifies airside data rates ranging from 6 to 54 Mbit/s. Below contains a listing of the eight specified PHY data rates. Four different modulation schemes are used: BPSK, 4-QAM, 16-QAM, and 64-QAM, along with a number of convolutional encoding schemes. This allows the system to adapt to the optimum data rate vs. error rate for the current conditions.

3.5.4 Terrestrial digital radio and television broadcasting

Much of Europe and Asia has adopted OFDM for terrestrial broadcasting of digital television (DVB-T, DVB-H and T-DMB) and radio (EUREKA 147 DAB, Digital Radio Mondiale, HD Radio and T-DMB).

3.5.4.1 DVB-T

By Directive of the European Commission, all television services transmitted to viewers in the European Community must use a transmission system that has been standardized by a recognized European standardization body,[1] and such a standard has been developed and codified by the DVB Project, *Digital Video Broadcasting (DVB); Framing structure,* *channel coding and modulation for digital terrestrial television.* Customarily referred to as DVB-T, the standard calls for the exclusive use of COFDM for modulation. DVB-T is now widely used in Europe and elsewhere for terrestrial digital TV.

3.5.4.2 COFDM vs. VSB

The question of the relative technical merits of COFDM versus 8VSB has been a subject of some controversy, especially between Europe and USA. The United States has rejected several proposals to adopt COFDM for its digital television services, and has instead opted for 8VSB (vestigial sideband modulation) operation.

One of the major benefits provided by COFDM is that it renders radio broadcasts relatively immune to multipath distortion and signal fading due to atmospheric conditions or passing aircraft. Proponents of COFDM argue that it resists multipath far better than 8VSB. Early 8VSB DTV (digital television) receivers often had difficulty receiving a signal in urban environments.

However, newer 8VSB receivers are far better at dealing with multipath, hence the difference in performance may diminish with advances in demodulator design. Moreover, 8VSB modulation requires less power to transmit a signal the same distance, i.e., the received carrier-to-noise threshold is lower for the same bit error rate. In less-populated areas, 8VSB may have an advantage because of this. In urban areas, however, COFDM is believed to offer better reception than 8VSB. In practice, it may be impossible to settle this debate without empirical history

3.5.4.3 Digital radio

COFDM is also used for other radio standards, for digital audio broadcasting (DAB), the standard for digital audio broadcasting at VHF frequencies, and also for Digital Radio Mondiale (DRM), the standard for digital broadcasting at shortwave and medium wave frequencies.

The USA again uses an alternate standard, a proprietary system developed by quity dubbed "HD Radio". However, it uses COFDM as the underlying broadcast technology to add digital audio to AM (medium wave) and FM broadcasts. Both Digital Radio Mondiale and HD Radio are classified as in-band on-channel systems, unlike Eureka 147 (DAB: Digital audio broadcasting) which uses separate VHF or UHF frequency bands instead.

3.5.4.4 BST-OFDM used in ISDB

The BST-OFDM (Band Segmented Transmission Orthogonal Frequency Division Multiplexing) system proposed for Japan in the ISDB-T, ISDB-TSB and ISDB-C broadcasting systems improves upon COFDM by exploiting the fact that some OFDM carriers may be modulated differently from others within the same multiplex. Some forms of COFDM already offer this kind of hierarchical modulation, though BST-OFDM is intended to make it more flexible. The 6 MHz television channel may therefore be "segmented", with different segments being modulated differently and used for different services. It is possible, for example, to send an audio service on a segment that includes a segment comprised of a number of carriers, a data service on another segment and a television service on yet another segment - all within the same 6 MHz television channel. Furthermore, these may be modulated with different parameters so that, for example, the audio and data services could be optimized for mobile reception, while the television service is optimized for stationary reception in a high-multipath environment.

3.5.5 Ultra wideband

UWB (ultra wideband) wireless personal area network technology may also utilize OFDM, such as in Multiband OFDM (MB-OFDM). This UWB specification is advocated by the WiMedia Alliance (formerly by both the Multiband OFDM Alliance {MBOA} and the WiMedia Alliance, but the two have now merged), and is one of the competing UWB radio interfaces.

3.5.6 Flash-OFDM

Flash-OFDM (Fast Low-latency Access with Seamless Handoff Orthogonal Frequency Division Multiplexing) is a system that is based on OFDM and specifies also higher protocol layers. It has been developed and is marketed by clarion. Flash-OFDM has generated interest as a packet-switched cellular bearer, on which area it would compete with GSM and 3G networks. As an example, old 450 MHz frequency bands that were used by NMT (a 1G analog network, now decommissioned) in Europe are already being licensed to Flash-OFDM operators. In Finland the license holder Digital has begun deployment of its nationwide "@450" wireless network, planned to be operational in April 2007.

American wireless carrier Sprint Nextel had stated plans for field testing Flash-OFDM (along with other wireless broadband network technologies) for their 4G offering, which will be deployed using the licenses they own nationwide in the 2.5GHz frequency range. Sprint subsequently has decided to deploy the mobile version of WiMAX which is based on SOFDMA, scalable orthogonal frequency division multiple access technology.

T-Mobile already offers Flash-OFDM connection to its subscribers in Slovakia. The maximum download speed is 5.3 Mbit/s, whereas upload speed is limited to 1.8 Mbit/s. Citizens Telephone Cooperative launched a Flash-OFDM service to subscribers in parts of Virginia in March, 2006. The maximum speed available is 1.5 Mbit/s.

Digi web Ltd. is launching 872MHz Flash-OFDM during 2007 in Ireland and also will be launching in Norway. To date most of the 450MHz systems are 1.25MHz. The 872MHz systems are 4Mhz bandwidth.

CHAPTER FOUR

RESULTS

4.1 Additive white Gaussian noise (AWGN) channel results

4.1.1 Simulation of single carrier communication

Figure 4.1 shows the BER (bit error rate) versus SNR (Signal to noise ratio) performance of communication system using a single carrier on AWGN channel. Binary phase shift key (BPSK) modulation is used.

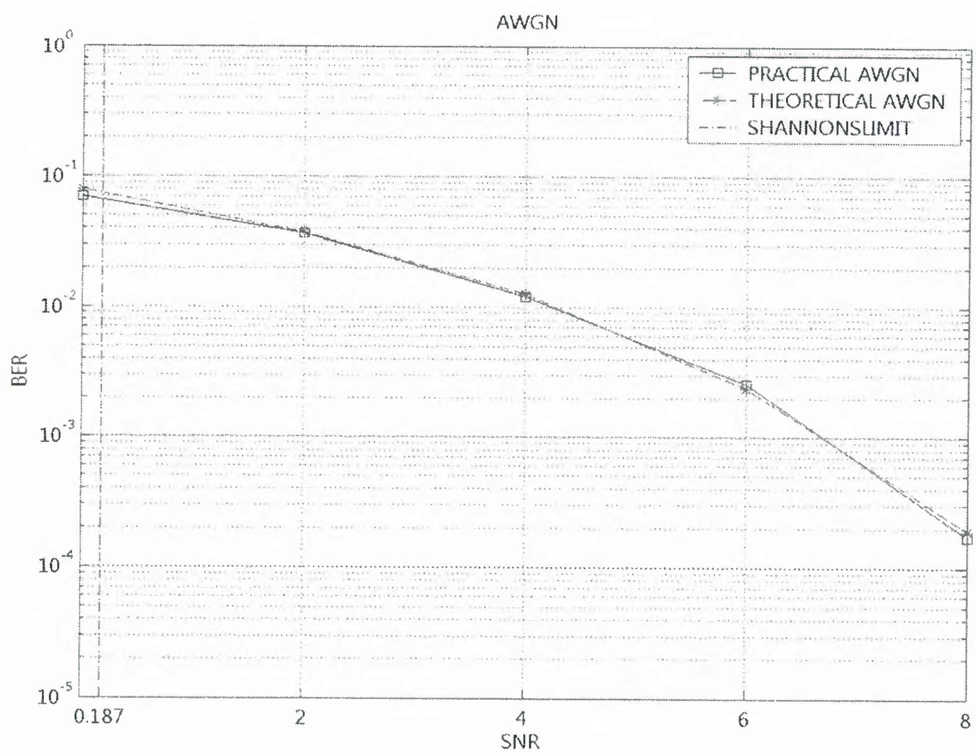

Figure 4.1 Performance of a single carrier communication on AWGN channel

SNR is inversely proportional to the noise variance (equation (4.1)).Therefore as SNR increase the noise corrupts less number of data and BER decrease.

$$variance = \frac{1}{2(SNR)} \tag{4.1}$$

4.1.2 Simulation of OFDM

Figure 4.2 shows the performance of OFDM over AWGN channel. The same results as in single carrier communication is obtained. Therefore, it can be argued that OFDM has no effect the performance over AWGN channel.

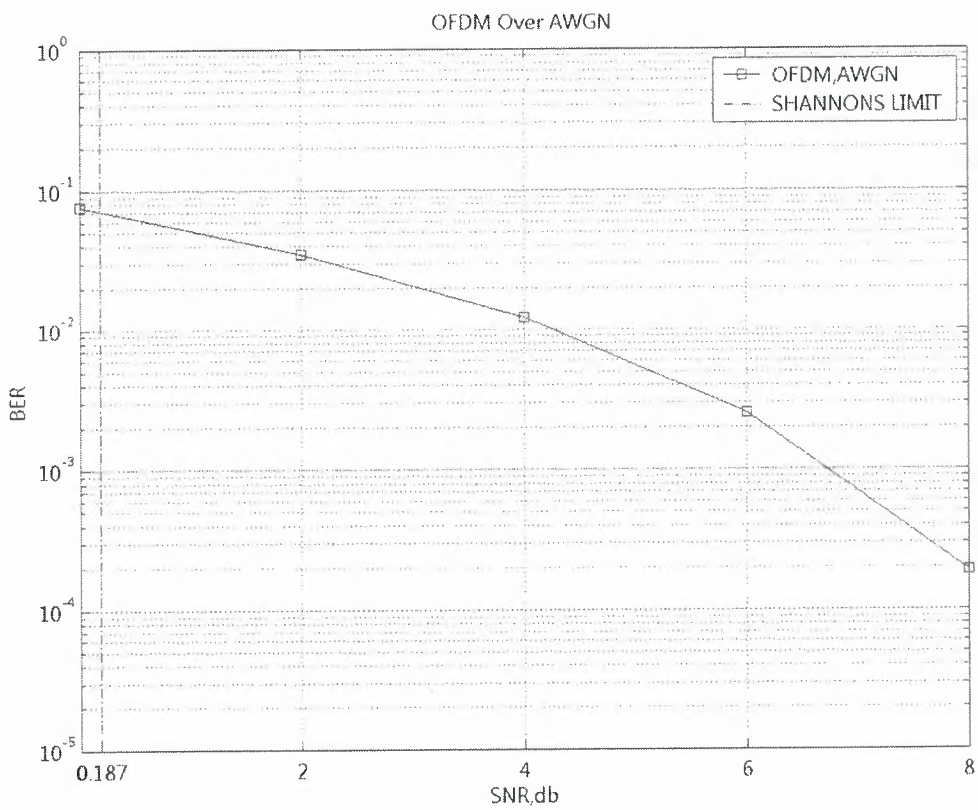

Figure 4.2 Performance of OFDM on AWGN channel

In OFDM N subcarrier are used. Since Fourier Transform (FT) is comed out using IFFT/FFT,N needs to be a power of 2 BPSK modulation is used on each carrier.

4.1.3 Theoretical AWGN

Figure 4.1 shows also the bit theoretical error rate probability for BPSK over AWGN channel.

4.1.4 Shannon's limit

Shannos limit is a signal to noise ratio (SNR) limit required by the ideal system for error free transmission to be possible. For AWGN channel the limit is 0.187

4.2 FADING RESULTS

4.2.1 Simulation single carrier communication

Figure 4.3 shows the performance of BPSK modulated single carrier communication over slow, flat fading channel. It is observed that the performance is worse than that obtained over AWGN channel.

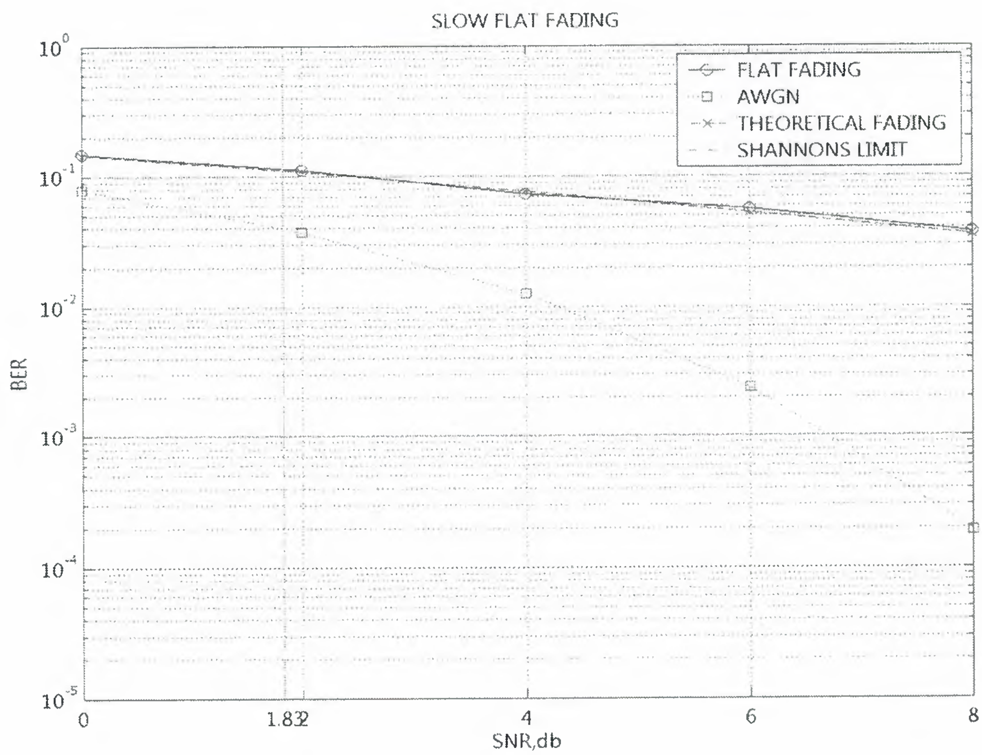

Figure 4.3 Performance of single carrier communication, over slow flat fading channel

4.2.2 Simulation of fading with OFDM

Figure 4.4 shows the performance of the flat fading in OFDM over slow, flat fading channel. When consider to be similar the performance of single carrier communication. It is observed that the performance has slightly improved. This shows that using multicarrier communication is better to battle against fading than using a single carrier.

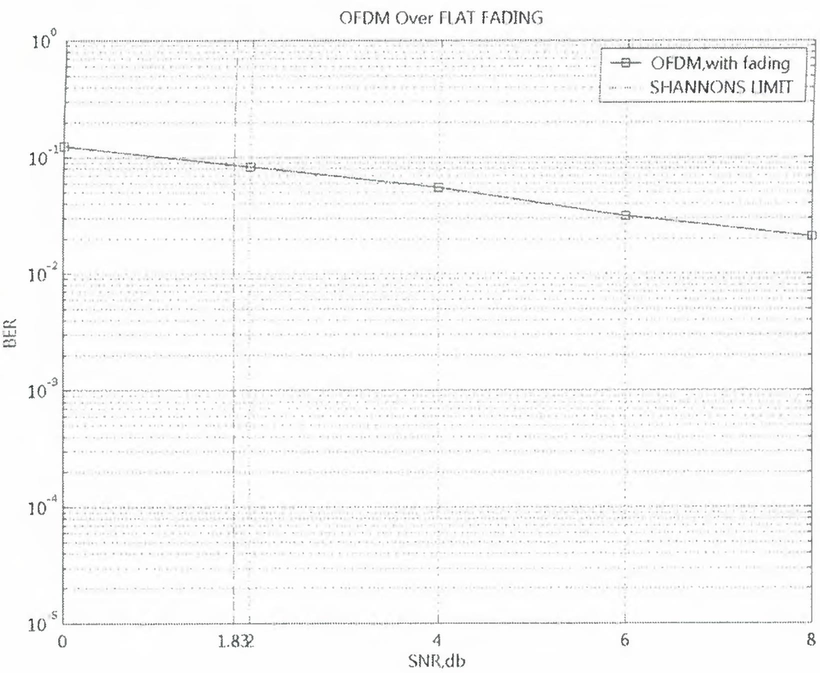

Figure 4.4 performance of flat fading over OFDM

4.2.3 Theoretical fading

In the figure 4.3 the theoretical fading plot is also given. This performance is also given by

$$P_{e(FADING)}=\frac{1}{2}\left(1-\sqrt{\frac{SNR}{1+SNR}}\right)$$

4.2.4 Shannons limit

The shannons limit for flat fading channel is 1.83 db.

CONCLUSIONS

To bring to a conclusion ADSL, to provide to do with phone cheap and quality speaking. And ADSL is new technology and new communication. We have seen network of data and sound that to provide to using same time. Users considering purchasing an ADSL service Users considering purchasing an ADSL service should evaluate their needs, in relation to the service they will receive via ADSL. For instance, analogue or ISDN dial-up users, who currently have a high telephone bill may benefit from the "always on" nature of ADSL, and thus make a significant saving on their telephone bill. The Leased line users however need to be aware of the fact that with ADSL, they will no longer enjoy predictable service levels, which are usually necessary where any kind of business reliance is placed on an Internet access service. Thus, it is advisable to weigh up the potential financial losses, which may result from the unavailability of Internet access, before considering potential price savings, which may be enjoyed by converting to ADSL.

The complex envelope of the transmitted signal in an OFDM system has been shown to converge weakly to a Gaussian random process as the number of subcarriers in the system goes to infinity. In addition, modern extreme value theory has been employed to derive fully analytic and simple yet accurate approximations for the PMEPR for the transmitted signal in OFDM systems. Orthogonal frequency division multiplexing (OFDM) that are several methods exist for remove these limitations, contain a multicarrier modulation technique. Fading and inter symbol interference occurring in wireless and band limited channels are major limiting factors in communication systems.

Performance of single carrier and OFDM modulator communication systems are analyzed in addition white Gaussian noise (AWGN) and flat, slow fading channels. It is explain through MATLAB simulation, that OFDM should be like better in flat, slow fading channels over single carrier systems as it assists to increase the performance. As a future work, guard intervals could be created where cyclic prefix is used to overcome the effect of inter symbol interference.

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[14] http://www.iec.org The International Engineering Consortium

APPENDIX A

MATLAB CODES DESCRIPTION

Appendix A.1 Simulation of single carrier communication over AWGN channel

```
snr=0:2:8;
for i=1:5
err=0;
data=0;
var=1/(2*(10^(snr(i)/10)));
q(i)=0.5*erfc(sqrt(10^(snr(i)/10)));
while err<100
data=data+1;
s=rand;
if s>.5
s=1;
else
s=0;
end
tx=(2*s)-1;
noise=sqrt(var)*randn;
rx=tx+noise;
if rx>0
rx=1;
else
```

```
rx=-1;
end
if rx~=tx
err=err+1;
end
end
ber(1,i)=err/data;
end
x=0.187*ones(1,6);
y=[10^-5 10^-4 10^-3 10^-2 10^-1 10^0];
semilogy(snr,ber,'-s',snr,q,'--*',x,y,'-.');
grid
set(gca,'xtick',[0.187 2 4 6 8]);
xlabel('SNR');
Ylabel('BER');
axis([0 8 10^-5 10^0]);
title('AWGN');
legend('PRACTICAL AWGN','THEORETICAL AWGN','SHANNONSLIMIT');
```

Appendix A.2 Simulation of OFDM over AWGN channel

```
snr_db=0:2:8;
N=512;
for i=1:5
err=0;
data=0;
```

```
snr=10^(snr_db(i)/10);
variance=1/(2*N*snr);
while err<100
data=data+N;
for k=1:N
s(k)=rand;
if s(k)>0.5
s(k)=1;
else
s(k)=0;
end
tx=2*s-1;
t=ifft(tx,N);
for k=1:N
r(k)=(t(k))+((sqrt(variance)*randn)+(sqrt(variance)*rand*j));
end
rx=fft(r,N);
for k=1:N
if real(rx(k))>0
r(k)=1;
else
r(k)=-1
end
if r(k)~=t(k)
err=err+1;
```

```
end
end%while
ber(1,i)=err/data;
end%for
x=0.187*ones(1,6);
y=[10^-5 10^-4 10^-3 10^-2 10^-1 10^0];
semilogy(snr_db,ber,'-s',x,y,'-.');
grid
set(gca,'xtick',[0 0.187 2 4 6 8]);
xlabel('SNR,db')
ylabel('BER')
axis([0 8 10^-5 10^0])
title('OFDM Over AWGN')
Legend('OFDM,AWGN','SHANNONS LIMIT');
```

Appendix A.3 Simulation of fading in single carrier

```
snr=0:2:8;
for i=1:5
err=0;
data=0;
snr_db=10^(snr(i)/10)
var=1/(2*snr_db)
q(i)=0.5*erfc(sqrt(snr_db)
thfade(i)=0.5*(1-(sqrt((snr_db)/(1+snr_db))))
while err<100
```

```
data=data+1
s=rand
if s>.5
s=1
else
s=0
end
tx=(2*s)-1
noise=sqrt(var)*randn
rand1=sqrt(0.5)*rand
rand2=sqrt(0.5)*rand
fade=sqrt((rand1^2)+(rand2^2))
rx=(fade*tx)+noise
if rx>0
rx=1
else
rx=-1
end
if rx~=tx
err=err+1;
end
end
ber(1,i)=err/data;
end
x=1.83*ones(1,6);
```

```
y=[10^-5 10^-4 10^-3 10^-2 10^-1 10^0];

semilogy(snr,ber,'-o',snr,q,':s',snr,thfade,'--x',x,y,'-.')

grid

set(gca,'xtick',[0 1.83 2 4 6 8])

xlabel('SNR,db')

ylabel('BER')

axis([0 8 10^-5 10^0])

title('SLOW FLAT FADING')

legend('FLAT FADING','AWGN','THEROTICAL FADING','SHANNONS
LIMIT')
```

Appendix A.4 Simulation fading over OFDM

```
snr_db=0:2:8;

N=512;

for i=1:5

err=0;

data=0;

snr=10^(snr_db(i)/10);

variance=1/(2*N*snr);

while err<100

data=data+N;

for k=1:N

s(k)=rand;

if s(k)>0.5

s(k)=1;
```

```
else
s(k)=0;
end
end
tx=2*s-1;
t=ifft(tx,N);
for k=1:N
rand1=sqrt(0.5)*randn;
rand2=sqrt(0.5)*randn;
fade=sqrt(rand1^2+rand2^2);
r(k)=(fade*t(k))+((sqrt(variance)*randn)+(sqrt(variance)*rand
*j));
end
rx=fft(r,N);
for k=1:N
if real(rx(k))>0
r(k)=1;
else
r(k)=-1
end
if r(k)~=t(k)
err=err+1;
end
end%while
ber(1,i)=err/data;
```

```
end%for
x=0.183*ones(1,6);
y=[10^-5 10^-4 10^-3 10^-2 10^-1 10^0];
semilogy(snr_db,ber,'-s',x,y,'-.');
grid
set(gca,'xtick',[0 0.183 2 4 6 8]);
xlabel('SNR,db')
ylabel('BER')
axis([0 8 10^-5 10^0])
title('OFDM Over FLAT FADING')
Legend('OFDM,with fading','SHANNONS LIMIT');
```